ENGINEERING ACOUSTICS

MECHENG 4115, 7027

Lecture Notes 2018

Carl Howard

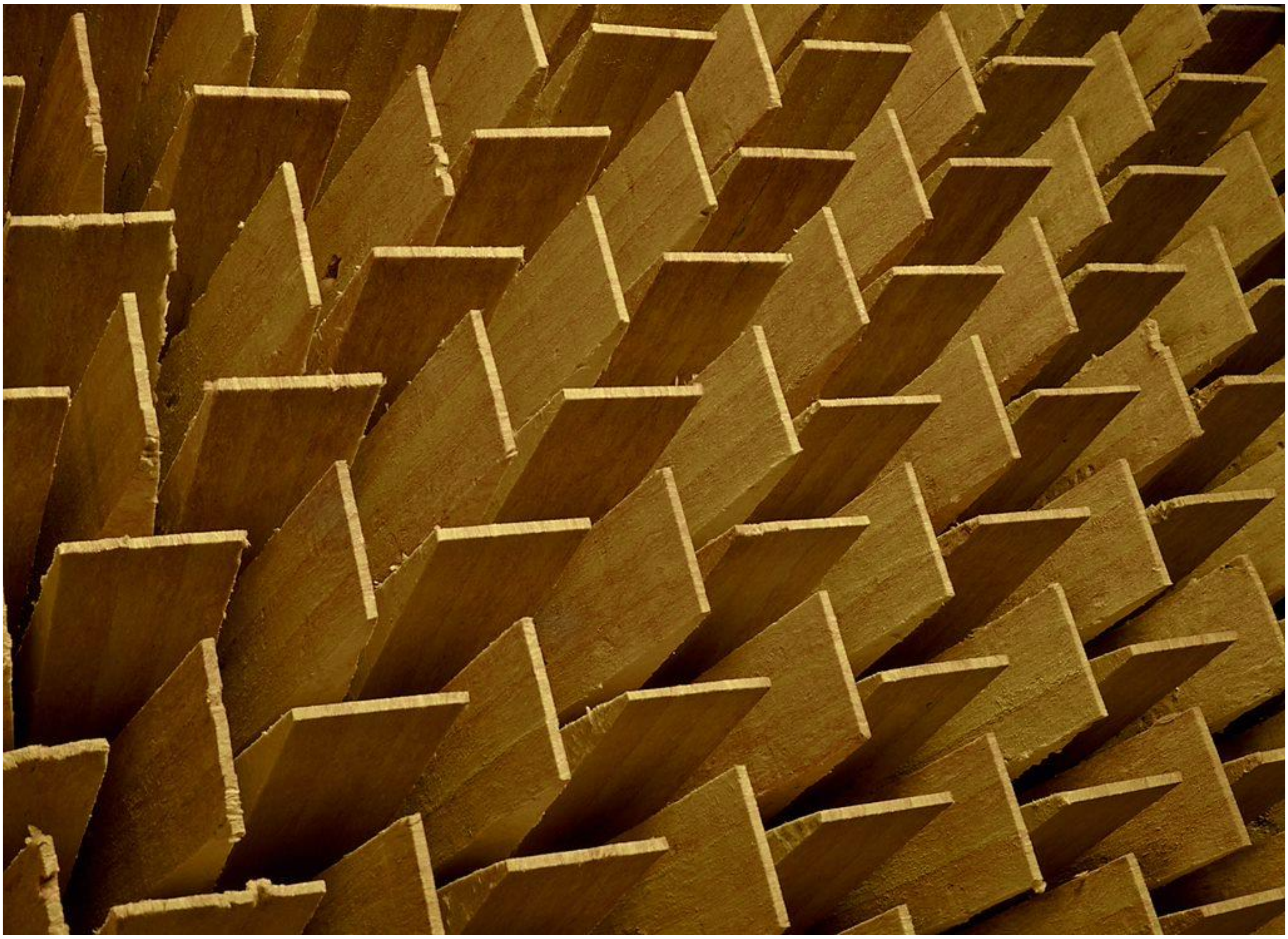
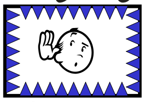


Table of Contents

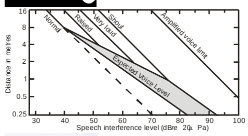
Introductory Lecture  Page 3

Basics  Page 14

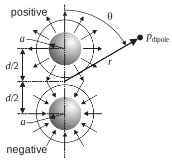
General Noise Control  Page 44

Instrumentation  Page 70

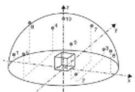
The Ear  Page 86

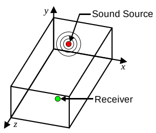
Criteria  Page 98

Psychoacoustics  Page 125

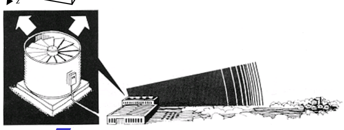
Sound Sources 

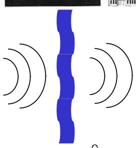
Page 145

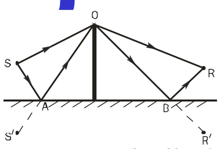
Sound Power  Page 169

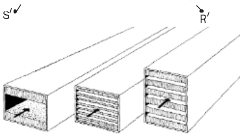
Indoor 

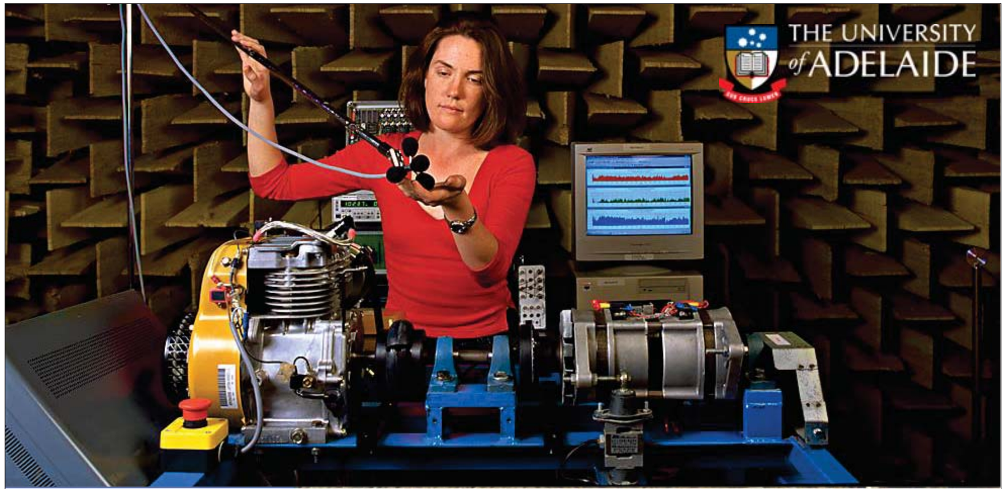
Page 181

Outdoor  Page 206

Acoustic Enclosures and Transmission Loss of Panels  Page 220

Barriers  Page 250

Mufflers  Page 267



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Carl Howard

Engineering Acoustics

adelaide.edu.au

seek LIGHT

Welcome!

- Bio [Who is this guy?]
- About the course [What is this course all about?]
- Visit to Acoustics Chambers

About the Course

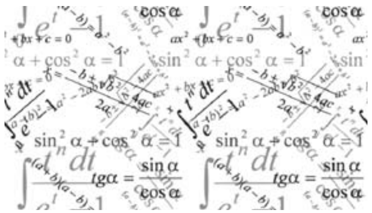
- Comments about the “maths”.
- Vital to use example problems for learning.
- How SELTs are used to improve the course.
- Lifelong learners and adult learners.
- Learning resources.
- Recommended learning procedure.
- Expectations.
- Overview of the course content.
- Schedule of tasks.
- Assessment and late policy.
- Tour of acoustic chambers.

Interact and Participate!

- Throughout the course we’ll use the socrative.com web site. socrative by MasteryConnect
- Get the “app” for iOS or Android.
- Use your smartphone, tablet, or laptop and goto: https://b.socrative.com/login/student/
  Enter the room number as: b069a2aa

The “maths” – DON’T WORRY!

- Maths in the course is not hard… about Year 11 / 12 standard.
- First, it is important to understand the concepts.
  Many concepts will be new, but they are not hard to understand.
- Once you understand the concepts, the maths “makes sense”.
- Don’t fall into the mindset of
  1. find formula
  2. stuff numbers into equation
- Open book exam. Don’t need to memorise formulas.



Example Problems and Answers

- There are over 250 example tutorial problems and fully worked answers available on MyUni.
  - Non-graded practice questions; vital learning resource.
  - These should be used to test your understanding. i.e. Learning Diagnostic Tool.
  - Beware - Don’t fall into the mindset of trying to stuff numbers into a formula. **Understand what is going on**.
- All exams and full solutions from 1996 onwards (13 years!) are available on MyUni.
  - The years that are missing are because it was not taught in that year.
  - The last exam is not online as it will be used as assignment questions for this year.

What has changed based on 2015 SELTs

- Course notes revised
  - Went through entire notes and marked equations with corresponding page number in textbook, annotated symbols and units.
  - Some slides removed in an attempt to reduce page count.
  - Will skip over some content in the lectures but notes still retained.
  - Formula sheet will be made available for YOUR customisation.
  - Many new slides added, for better explanations of some topics.
- New assignment questions.
- Lab class will be run slightly differently.
- New assignment for SoundPlan.

What has changed based on 2016 SELTs

- Course notes revised
  - Almost no comments on notes ! No comments on content, layout, colour, figures … must be nearly perfect !
  - Page numbers added.
  - Module on SoundPlan replaced with Mufflers (originally in the course).
    Very expensive to run (2x external consultants, 1 flew from Melbourne, assignment, software costs, installation, acceptance testing, lab costs, marking, and students were indifferent.)
- New assignment questions and personalised answers.

What has changed for 2018 Course

- SELTs 2017 comments:
  - No mention about course notes – must be perfect ☺
  - A couple of students commented about MapleTA assignments, but appreciated there was no viable alternative.
  - No suggestions for improvements, only praises.
- Course notes revised.
- New demonstrations: quarter wavelength tube, ANC duct.
- New / revised lab practical.
- New assignment questions.
- Scavenger hunt game.
- Change to a "flipped" classroom style where recorded lectures should be viewed before class, then come to class to discuss, ask questions, do problems, see demos... why? See next slide on class attendance

Lifelong Learning

- We are training you to be a PROFESSIONAL ENGINEER
- Graduate Attribute "Lifelong Learning"
  - *student-centred learning*;
  - *a focus on learning so as to equip students with the attitudes and skills to learn for themselves both in formal education and long after they have graduated;*

Lifelong Learners:

- Plan their own learning
- **Assess their own learning**
  - students learn to take control of the crucial first step in learning; finding out what it is they do not know. Peer and self assessment assumes assessment is a skill that is vital for students to learn if they are to monitor their learning in an ongoing way.
- **Are active rather than passive learners**
- Learn in both formal and informal settings
- Learn from their peers, teachers, mentors etc.

From: Knapper, C. and Cropley, A. J. *Lifelong learning in higher education.* London : Kogan Page, 2000.

Kiley, M. and Cannon, R., *Leap into... Lifelong Learning*, Centre for Learning and Professional Development, http://hdl.handle.net/2440/71214

Learning Resources

There is a **vast** range of learning resources that you ***should*** make use of throughout the course.

Print Matter	Digital	Personal
Lecture Notes	Video Recordings	Attend Classes
	Internet References	
Textbooks	Online reSearch Engines	Ask Colleagues
Practice Problems	ENC Software	Discussion Board

Learning Resources

Print Matter

Lecture Notes

Textbooks

Practice Problems

- Hundreds (!) of slides (notes only)
- The course is based on the textbook, which has detailed explanations.
  Other textbooks that should be used as well!
- Hundreds (!!) of practice problems to test your understanding of material. (Not graded)
  Do not just use the assignment as the learning diagnostic tool.

There are questions asked in this course that are intentionally not answered in the notes.
This is to get you to do some research -- to develop a “Graduate Attribute”.
We are training you to be a PROFESSIONAL ENGINEER.
As an engineer you will be faced with this situation all the time.

Learning Resources

Digital

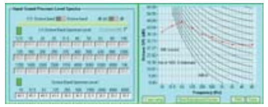
Video Recordings

Internet References

Online reSearch Engines

ENC Software

echo360



- Need to watch the lecture recordings **BEFORE** in-class sessions.
  There are other videos that you should watch as well.
- Online references listed in the notes provide good explanations.
- Use search engines to find answers.
- ENC software in CATS has most of the formulas – use to solve Q’s.

Learning Resources

Personal

Attend Classes

Ask Colleagues

Discussion Board

- Come to classes and participate!
  Be an **ACTIVE** learner,
  Ask questions about the videos,
  Hear the acoustic demonstrations,
  Ask questions about the assignments,
  Get involved.
- Ask your colleagues questions.
- Post questions on the discussion board that your colleagues should answer.
  I will also contribute.
  (Do not email me directly, unless personal nature)

Learning Resources

- Recommended learning process.

Video Recordings

Lecture Notes

Textbooks

Attend Classes

Practice Problems

Discussion Board

Assessments
(Assignments, Lab Prac)

Assessment: Final Exam

Assignments

- Most assignments use MapleTA
  - A specialised online system for mathematical Q&A.
  - Used extensively in maths and other courses.
  - Does not assess your "working out", just the answer.
    "Oh, but even though I got the wrong answer, can't I get some marks?"
    Need to alter your attitude to learning...
- We are training you to be a **PROFESSIONAL** engineer
  - As a professional engineer / doctor / surgeon / accountant etc, is it acceptable to get answers wrong or even part wrong ?

Assignments

- MapleTA is a "**learning diagnostic tool**".
  - You should be using the (non-graded) practice questions **first**, to check / test your understanding.... THEN,
  - MapleTA assignments have low weight grading, intentionally.
    (I would prefer zero weighting, so it is another practice assignment / learning diagnostic.)
  - Life Long Learners == "Assess their own learning"
- Some assignments require grading 1,700 answers and are returned within 1 hour, then discussed at the next class. Not possible with hand-written assignments.
- Look at your personalised "worked solutions" to find out how you could have solved the problem (often more than one way to solve the problem).
- See PDF file on MyUni for instructions on how to access and use MapleTA (via Assignments tab on MyUni).

Actionbound Acoustic Tour

Actionbound

- Smartphone "app" adventure tour of interesting acoustic sites around the University of Adelaide.
- See separate instructions on MyUni.
- You'll need to load the app on your smart-device, follow the tour guide, see acoustic features, listen to sounds, answer questions, score points.
- As part of Assignment 4, due Friday 9am, 2 Feb 2018.
  - MapleTA 3%
  - Actionbound 2%

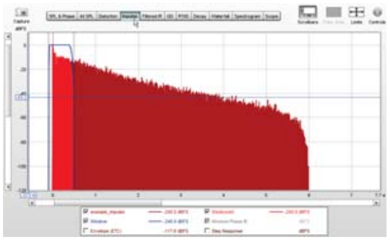
Laboratory Practical

- Measurement of acoustic absorption coefficient using T_{60} and substitution methods in acoustic reverberation chamber.
- 10% of course grade
- No formal lab "report".
  Just need to submit numerical values and answer a couple of questions.

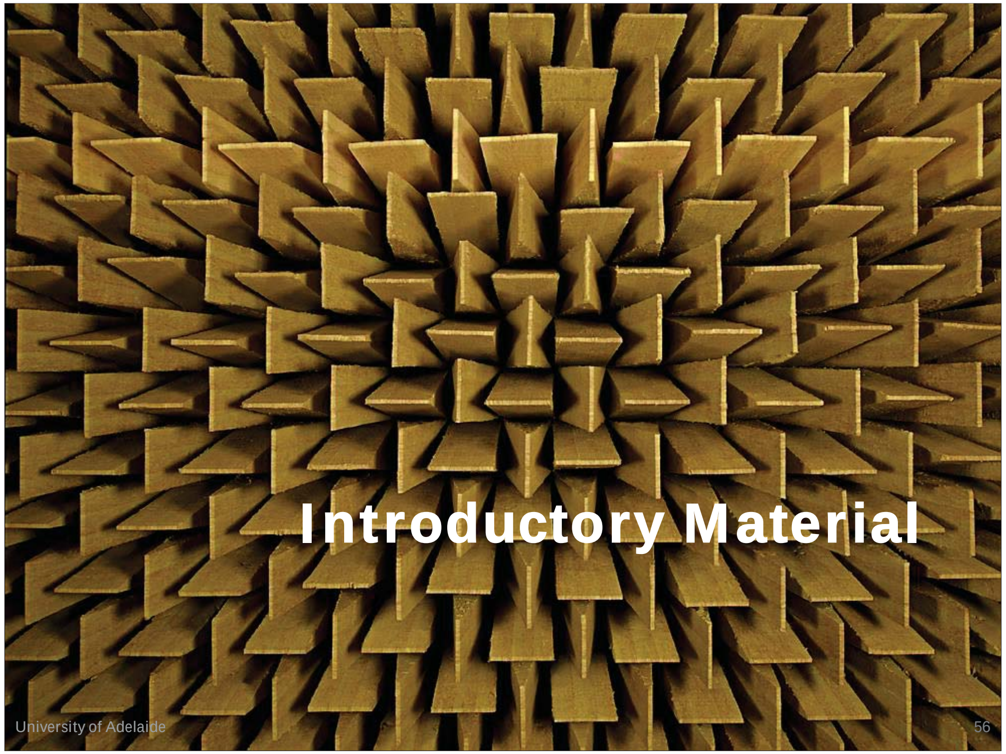
Distance Learning– Lab Prac Alternative

- Acoustics tutorial / assignment

  As you can't attend the acoustic chambers
- Use Room EQ Wizard software to determine the T60 from supplied wave files.
- Determine sound absorption coefficient α.



- Use your own smartphone, tablet, laptop to measure the impulse response in a room of your choice.
- Start straight away!
- Write up due 5pm Thursday 8 February 2018 via MyUni.



Learning

- Take charge of your own learning!
- Use the resources available – see next slides.
  - Read the notes.
  - Read the textbook.
  - Use other resources – see course handbook.
  - Do the practice problems
    - These are VITAL learning resources.
    - If you get questions wrong, learn from it, and move on.
  - Use the internet.
  - Use the discussion board on MyUni.

Learning Resources for the Course

- MyUni – starting point.
- Course *Notes* – they are ***notes,*** NOT a textbook.
- Textbook – details about each concept.
- Reference Materials – still confused? Read another book.
- Practice Problems and Solutions – these should be revised to understand the material.
- Discussion Board – Get help from colleagues and me.
- MapleTA Assignments – Another learning diagnostic tool.
  - Tests your understanding of the concepts. If you get a question wrong, look through the worked solutions and revise the material.
  - Low weighting towards overall grades.
  - This course is training you to be a professional engineer - you don't get part marks for getting an answer wrong.

Learning Resources for the Course

- ENC Software – All formulas in textbook.
- Recorded Lectures – I'll record each lecture.
- Come to Classes – Get involved, ask questions, come participate and learn !
- Demonstrations in Lectures – Reinforces material.
- Laboratory Practical – Reinforces material.

- SELT Questions:

  Q 4. Appropriate strategies to engage me in my learning

  What else would you like and need?

  Q 9. My learning in this course is supported by effective feedback

  What else would you like and need?

Resources

Books

Bies, D.A. and Hansen, C.H., Engineering Noise Control, 4th edition, Spon Press, London, (2009).

Bies, D.A. and Hansen, C.H., Howard, C.Q., Engineering Noise Control, 5th edition, CRC Press, London, (2017).

Hansen, C.H. Noise Control: From Concept to Application (2005).

Frank J. Fahy, "Foundations of Engineering Acoustics."

L.E. Kinsler et. al. 3rd edn. "Fundamentals of acoustics"

M. P. Norton, D. G. Karczub, "Fundamentals of noise and vibration analysis for engineers."

H. Fastl and E. Zwicker, "Psychoacoustics Facts and Models", Springer, 3rd Ed 2007. (free ebook from BSL)

Resources

Internet

http://www.animations.physics.unsw.edu.au/waves-sound/sound/index.html

http://www.acousticalsurfaces.com/acoustic_IOI/101home.htm

http://psysound.org/

http://audacity.sourceforge.net

http://www.psysound.org/ (Psysound and AARAE: Audio and Acoustical Response Analysis Environment)

Resources

iPhone

RTA Audio

iAnalyzer Lite

Decibel 10th

SGenerator Lite

Toolbox acousticassist

Faber Acoustics (best one but expensive)

Spectrum View – Oxford Wave Research

Android

RTA Pro Analyzer

AudioTool

Speedy Spectrum Analyzer

FrequenSee - Spectrum Analyzer

Spectrum Analyzer

Audalyzer

Noise Meter

Resources

Free Online Courses in Acoustics

- Acoustics of Speech and Hearing, MIT
  - http://ocw.mit.edu/courses/electrical-engineering-and-computer-science/6-551j-acoustics-of-speech-and-hearing-fall-2004/
- Advanced Structural Dynamics and Acoustics, MIT
  - http://ocw.mit.edu/courses/mechanical-engineering/2-067-advanced-structural-dynamics-and-acoustics-13-811-spring-2004/
- Introduction to Acoustics (Part 1 & 2), Korea Advanced Institute of Science and Technology
  - https://www.coursera.org/course/acoustics1
  - https://www.coursera.org/course/acoustics2

Expectations

- You should act as a **Professional Engineer.**
- You will have to read the notes and the textbook.
- I won't go through the text on every slide. Why?
  - Because students get bored from PowerPoint slides filled with text and math Greek characters -- "Death by PowerPoint"
    ... a bit hard to escape this at times because there are a lot of concepts and equations in this course.
  - There is a lot to get through.
  - It is often easier to digest material by reading by yourself.
    ... but **PLEASE** ask questions if something is confusing.

"I understand we have limited time to cover an entire course, but when going through the slides I suggest giving a little bit more time to the ambiguous points only, I would say 95% of the course don't require that and they are very easy to understand. So rush through the slides. But keep an eye on the difficult parts."

Yes! But you have to TELL ME if you want more info.

Course Content

- Goal is for our graduates to have a basic understanding in (some) aspects of acoustics.
  - The subject of acoustics is MASSIVE, and we can't cover it all.
  - Provide foundation knowledge.
- We are teaching you to be a *PROFESSIONAL* engineer.
  - Provide knowledge to be an acoustic consultant.
  - Learn and behave like a professional engineer.
    - Learning is in **YOUR** hands.
    - Always be professional. i.e., turn up on time, be prepared, get it right, respectful of others, submit high-quality work, submit work on-time, ...
      Graduate Attribute: *A commitment to the highest standards of professional endeavour and the ability to take a leadership role in the community.*

Famous Acousticians

Amar Bose
BOSE Corp

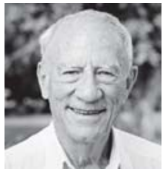
Ray Dolby
Dolby Laboratories

Bolt, Beranek, Newman
Acoustic consulting, concert hall design, JFK Assassination, aircraft noise, interior noise curves, vibration levels for sensitive equipment, ARPANET, Telenet, email...

Ray Tomlinson
Bolt Beranek Newman
Inventor of @ sign for email.

Sir James Lighthill
Cambridge Uni
Aeroacoustics, Concorde, Lighthill's eighth power law, Fluid mechanics, ..

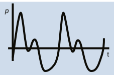
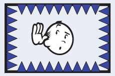
Per Bruel and Viggo Kjaer
Bruel and Kjaer
Inventors and manufacturers of acoustic and vibration instrumentation



Sir Harold Marshall
Marshall Day Acoustics
Concert hall, lateral reflections, Rayleigh Medal, Walter Sabine Medal

Paul Scully-Power
First Australian astronaut, Remote sensing, oceanographer, Distinguished Chair of Underwater Acoustics in the US Navy

Course Content

TOPIC	
Basics	
General Noise Control	
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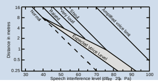
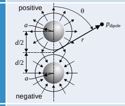
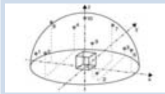
Criteria	

Course Content

TOPIC	
Psychoacoustics	
Sound Sources	
Sound Power	

Course Content

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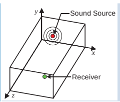
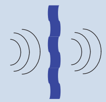
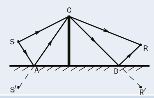
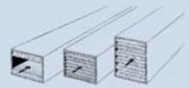
Indoor	
Outdoor	

Course Content

TOPIC	
Enclosures and Transmission Loss of Panels	
Barriers	
Mufflers	

Schedule 2018

	2018						
	Monday	Tuesday	Wednesday	Thursday	Friday	Saturday	Sunday
Jan	22	23	24	25	26	27	28
	4 lectures		Assign't 1 Due 4 lectures		Australia Day		
Feb	29	30	31	01	02	03	04
	Assign't 2 Due 4 lectures		Assign't 3 Due 4 lectures		Assign't 4 Due 4 lectures		
Feb	05	06	07	08	09	10	11
	Assign't 5 Due 4 lectures	Lab Pracs	Assign't 6 Due Lab Pracs	Lab Pracs	4 lectures		
Feb	12	13	14	15	16	17	18
	EXAM WEEK	EXAM WEEK	EXAM WEEK	EXAM WEEK	EXAM WEEK		

Assessment – Onsite Course

Assessment task	Weighting %	Description	Due date
Assignment 1	5	Assignment on Chapter 1 in the text. Refer to Practise problems 1 in MyUni.	Wed 9am 24 January
Assignment 2	5	Assignment on Chapters 2, 3 and 4 in the text. Refer to Practise problems 2 in MyUni.	Monday 9am 29 January
Assignment 3	5	Assignment on Chapters 5 and 6 in the text. Refer to Practise problems 3 in MyUni.	Wednesday 9am 31 January
Assignment 4	5	Assignment on Chapter 7 in the text. Refer to Practise problems 4 in MyUni. + ActionBound Scavenger Hunt Game.	Friday 9am 2 February
Assignment 5	5	Assignment on Chapters 8 in the text. Refer to Practise problems 5 in MyUni.	Monday 9am 5 February
Assignment 6	5	Assignment on Chapters 9 in the text. Refer to Practise problems 6 in MyUni	Wednesday 9am 7 February
Laboratory	10	Acoustics laboratory practical	2 days after lab
Final Exam	60	Exam on all parts of the course	Exam period

Assessment – Distance Learning

Assessment task	Weighting %	Description	Due date
Assignment 1	5	Assignment on Chapter 1 in the text. Refer to Practise problems 1 in MyUni.	Wed 9am 24 January
Assignment 2	5	Assignment on Chapters 2, 3 and 4 in the text. Refer to Practise problems 2 in MyUni.	Monday 9am 29 January
Assignment 3	5	Assignment on Chapters 5 and 6 in the text. Refer to Practise problems 3 in MyUni.	Wednesday 9am 31 January
Assignment 4	5	Assignment on Chapter 7 in the text. Refer to Practise problems 4 in MyUni.	Friday 9am 2 February
Assignment 5	5	Assignment on Chapters 8 in the text. Refer to Practise problems 5 in MyUni.	Monday 9am 5 February
Assignment 6	5	Assignment on Chapters 9 in the text. Refer to Practise problems 6 in MyUni	Wednesday 9am 7 February
Laboratory	10	Acoustics laboratory practical	Thursday 5pm 8 February
Final Exam	60	Exam on all parts of the course	Exam period

Extensions and Late Penalties

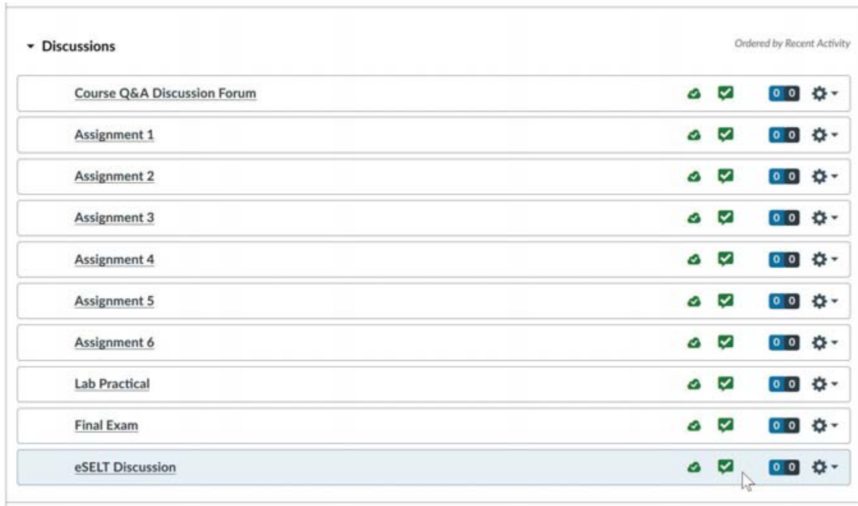
- **The answers to assignments will be discussed at the class following the due date, and hence submissions will not be accepted after the discussion of the answers to assignments.**
- **ALL assignments must be submitted electronically through MyUni. No paper assignments are accepted.**
- **For assignments that are late and the answers have not been discussed or published, will be penalised 10% per day.**
- **Extensions for assignments will only be given in exceptional circumstances and a case for this with supporting documentation can be made in writing after a lecture or via email to the lecturer who set the assignment. Assignments will be assessed and returned in 2 weeks of the due date. There will be no opportunities for re-submission of work of unacceptable standard.**

Critical Dates

https://www.adelaide.edu.au/student/dates/critical/2018/

Term	Last Day to Add Online	Census Date	Last Day to WNF	Last Day to WF
Summer School	Wed 24/01/2018	Fri 26/01/2017	Sun 4/02/2017	Fri 9/02/2017

Subscribe to MyUni Discussion Boards



Applications of acoustics

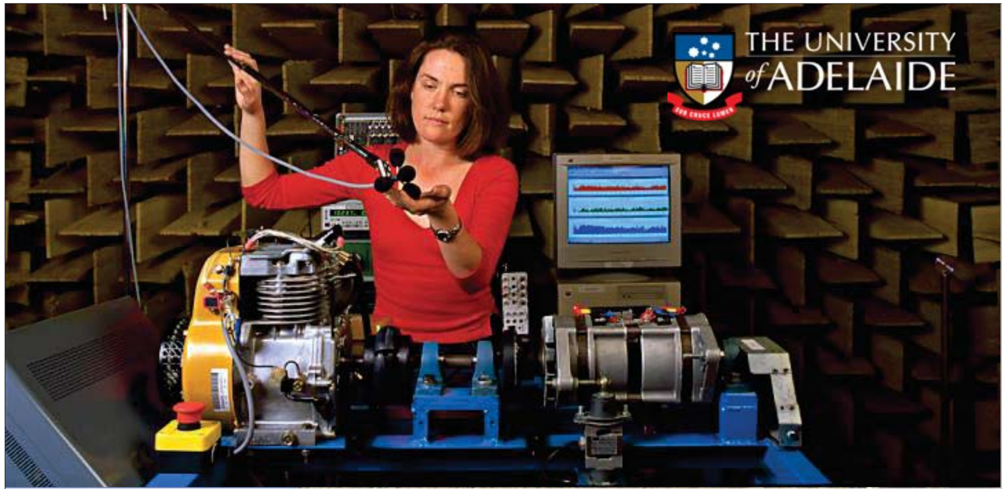
- One hour video describing applications of acoustics, on MyUni.
- Please watch it!
- The applications of acoustics are enormous!
- You probably never knew where acoustics is used, as most people take acoustics for granted.

360 Deg Video Tours of Acoustic Chambers

- There are 360 degree video tours of the anechoic and reverberation chambers:

https://youtu.be/VH5G9ucwPC4 https://youtu.be/y5V_boCjins

- Playback using a computer, smartphone, or VR headset



Fundamentals

Introduction

To start off, we need to talk about some basics:

- Acoustics – so that we understand some basic concepts
- Terminology – we need to speak the same language
- Simple mathematics – how to add up dB etc.

- A lot of this material is from the Acoustics module in the Dynamics and Control 2 (also Sustainability & the Environment).
  – We can't assume that all students have done the D&C2 course (or remember it !), so have to cover it here.

Contents

- Amplitude, frequency, wavelength
- Speed of sound
- Impedance
- Wave equation
- Plane waves, spherically spreading waves
- Resonance
- Octave bands
- Sound pressure level
- Sound intensity and sound power
- Addition and subtraction of dBs
- Directivity and reflection effects
- Flow resistivity and absorption

Learning Outcomes

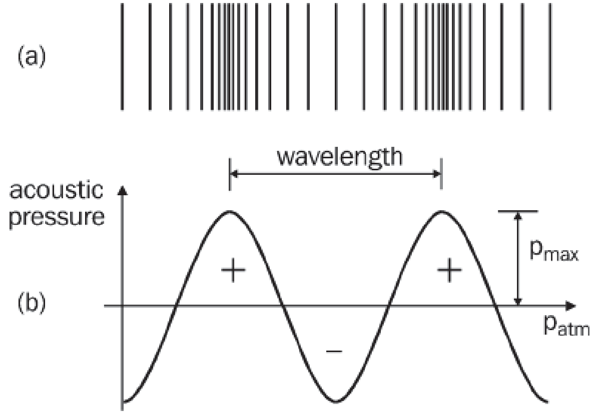
- Understand some basic terminology used in acoustics.
- Be able to calculate frequency, wavelength, speed of sound.
- Understand resonance
- Good understanding of octave bands.
- Be able to calculate sound pressure level, add and subtract SPLs, NRs.
- Basic understanding of sound intensity and sound power
- Understand how reflecting planes alter SPL from a radiating source.
- Understanding of flow resistivity and absorption.

Log into socrative.com

- Use your smartphone, tablet, or laptop and go to: https://b.socrative.com/login/student/
- Enter the room number as: b069a2aa
- You will be asked to answers 3 questions this lecture.

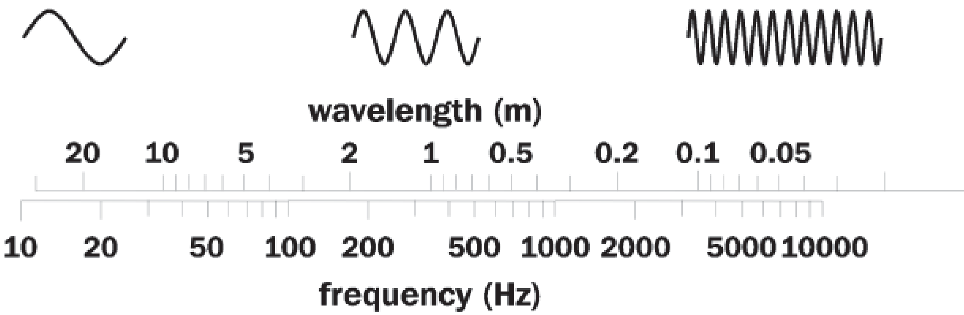
Frequency, Wavelength, Amplitude

- Amplitude, frequency (pitch) and wavelength



Frequency, Wavelength, Amplitude

Amplitude, frequency (pitch) and wavelength

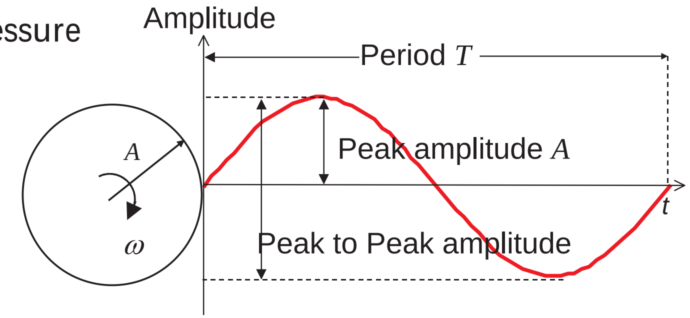


period, $T = 1/f$, where f = frequency [Hz]

Harmonic Pressure

REVISION

- Complex Pressure



- The period, T, is the time between two consecutive repetitive points on the waveform.
- The frequency, f, is the reciprocal of the period $f = \frac{1}{T} = \frac{\omega}{2\pi}$

Speed of Sound Relationship

$$\boxed{c = f\lambda}$$

ENC p24

- c = speed of sound [m/s]
- f = frequency [Hz]
- λ = wavelength [m]

Speed of Sound for Gases

- Speed of sound, c for gases.

$$\boxed{c = \sqrt{\frac{\gamma P}{\rho}} = \sqrt{\frac{\gamma RT}{M}}}$$

ENC p17

- P = pressure [Pa]
- T = temperature [K]
- R = universal gas constant [J/mol.K] (8.314 J/mol.K)
- M = molecular weight [kg/mol] (for air = 0.029 kg/mol)
- Υ = ratio of specific heats (for air = 1.402)

Example:
Air at 22°C -> $c = \sqrt{\frac{\gamma RT}{M}} = \sqrt{\frac{1.402 \times 8.314 \times (273.15 + 22)}{0.029}} = 344 \text{ m/s}$

Speed of Sound

- Mach 1.0 = speed of sound at the location.

- Inhaling
  - Helium: makes your voice higher pitch.
  - Sulfur hexafluoride: makes your voice deeper.
  - https://youtu.be/EA9QU0rdb9k

Speed of sound in solids

- In solids it is the speed of propagation of ***longitudinal*** waves not bending waves
- Speed is independent of frequency

$$c = \sqrt{D_F / \rho} \quad \text{for liquids}$$

$$c_L = \sqrt{E / \rho} \quad \text{for solid rod}$$

$$c_L = \sqrt{\frac{E}{\rho(1 - \nu^2)}} \quad \text{for thin panel}$$

D_F = bulk modulus of the fluid
E = Young's modulus
ρ = density
ν = Poission's ratio

ENC p15-17

Wavenumber *k*

- Another term that is often used in acoustics is *WAVENUMBER k* (don't get confused with stiffness)

$$\boxed{k = \frac{\omega}{c} = \frac{2\,\pi f}{c}}$$

ENC p24

k = wavenumber in 1/m
ω = circular frequency in radians / s
c = speed of sound in m/s
f = frequency in Hz

Demo:

- Distance between ears: about 20cm
- Corresponds to a frequency of

$$f = c/\lambda = 344/0.2 = 1720 \text{ Hz}$$

So if we play a sound at about 1700 Hz, move your head around slightly and very slowly....

You should notice that the sound level gets louder and quieter.

Density of a Gas

- Using the gas law

$$\boxed{p = \rho \frac{R}{M} T}$$

- The equation can be re-arranged to find density as:

$$\boxed{\rho = \frac{pM}{RT}}$$

p = pressure [Pa] (atmospheric air pressure is 101,325 Pa)
T = temperature [K]
R = universal gas constant [J/mol.K] (8.314 J/mol.K)
M = molecular weight [kg/mol] (for air = 0.029 kg/mol)
ρ = density of gas

Acoustic Field Variables

- Sound sensed by the ear is due to VERY small pressure fluctutions.
- The expressions for pressure, particle velocity, temperature and density can be written in terms of steady state (mean) values, and the perterbation.

Pressure: $P_{tot} = P + p(r, t)$

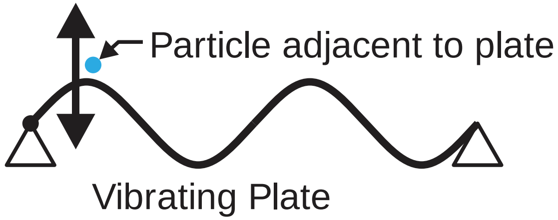
Velocity: $U_{tot} = U + u(r, t)$

Temperature: $T_{tot} = T + \tau(r, t)$

Density: $\rho_{tot} = \rho + \sigma(r, t)$

Acoustic Field Variables

- Pressure, Temperature, Density should be familiar.
- Particle velocity requires further discussion.
- Think of a "particle" as a small bit of the medium.
- Usually an oscillating velocity – not a mean flow.
- Consider a small parcel of air adjacent to a vibrating panel.



Wave Equation

- Acoustic propagation is mathematically described by the wave equation

$$\nabla^2 p = \frac{1}{c^2}\frac{\partial^2 p}{\partial t^2}$$

- See Appendix A in your text for a full derivation.
- Wave equation is derived using:
  - Conservation of Mass equation
  - Eulers Equation of motion
  - Equation of State

ENC p21--

Wave Equation

- Often written in terms of the acoustic potential function φ as

$$\nabla^2 \varphi = \frac{1}{c^2}\frac{\partial^2 \varphi}{\partial t^2}$$

where

$$u = -\nabla\varphi$$

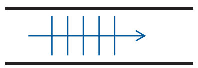
Particle velocity is the negative gradient of the potential function.

$$p = \rho\frac{\partial\varphi}{\partial t}$$

Pressure is the differential wrt to time of the potential function.

Plane and Spherical Spreading Waves

- Sound wave propagation is complicated.
- It can often be described as propagating as
- Plane waves
  - wavefront is flat

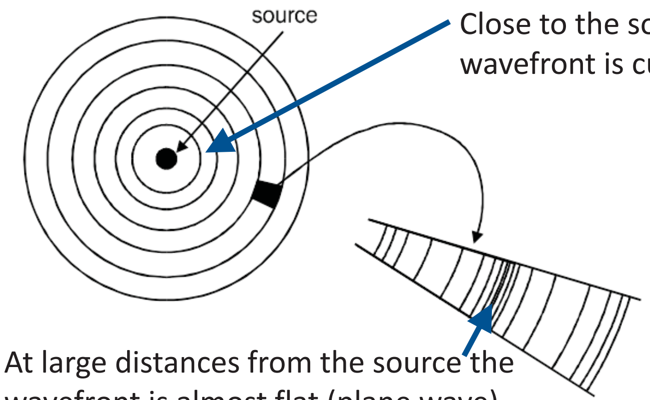


Plane wave in a duct

- Spherically spreading waves
  - Curved wavefront.

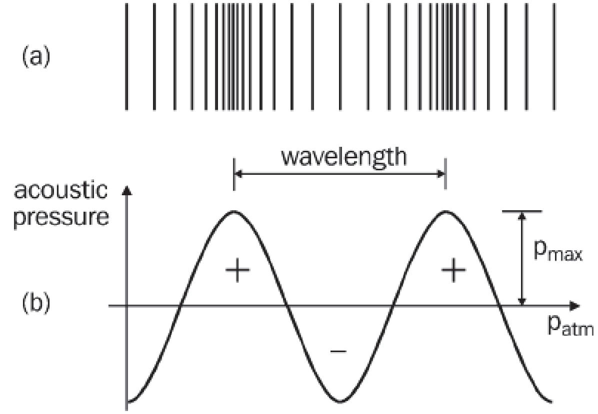
Plane and Spherical Spreading Waves

- Consider a sound source that generates spherically spreading waves.



Plane Waves

- Longitudinal pressure fluctuations



Plane Waves

- Start with wave equation

$$\frac{\partial^2 \varphi}{\partial x^2} = \frac{1}{c^2}\frac{\partial^2 \varphi}{\partial t^2}$$

- Define the acoustic potential function where *f* describes a distribution along *x*

$$\varphi = f\ (ct \pm x)$$

- Define pressure and particle velocity as

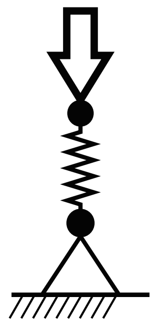
$$p = \rho \frac{\partial \varphi}{\partial t} = \rho c\ f'(ct \pm x)$$
$$u = -\nabla \varphi = \mp\ f'(ct \pm x)$$

- Divide the two, which leads to an important definition

$$\boxed{\frac{p}{u} = \rho c \quad = \text{characteristic impedance}}$$

What does impedance mean?

The simplest mechanical analogy is a force on a spring.



Force on a spring $f = kx$

Stiffness $k = \frac{f}{x}$

Velocity can be written as $v = j\omega x$

Mechanical Impedance $Z = \frac{f}{v} = \frac{k}{j\omega}$

Ratio of force and velocity

Impedance

- Spring

$$Z = \frac{f}{v} = \frac{k}{j\omega}$$

- Plane Wave

$$\boxed{\frac{p}{u} = \rho c}$$

- i.e. acoustic pressure divided by acoustic particle velocity.

Other Types of Impedances

Mechanical Impedance

$$Z_m = \frac{F}{u} = \frac{pA}{u}$$

Used to describe sound radiation from a vibrating structure.

Specific Acoustic Impedance

$$Z_s = \frac{p}{u}$$

Used to describe propagation of sound in free space, reflection and transmission of sound at interfaces

ENC p53

Other Types of Impedances

Acoustic Impedance

$$Z_a = \frac{p}{UA} = \frac{p}{v}$$

Used to describe propagation in ducts and mufflers

F = force (N, Units MLT^{-2})
u = acoustic particle velocity (m/s, Units LT^{-1})
p = pressure (Pa, Units $MT^{-2}L^{-1}$)
v = acoustic volume velocity (m^3/s, Units L^3T^{-1})
S = area (m^2, Units L^2)

ENC p53

Spherical Wave Solution to Wave Eqn

- For a harmonic solution $\phi = \frac{A}{r} e^{j(\omega t - kr)}$
- Pressure is $p = \rho \frac{\partial \phi}{\partial t} = \frac{j\omega A \rho}{r} e^{j(\omega t - kr)}$
- Velocity is $u = -\nabla\phi = \frac{A}{r^2} e^{j(\omega t - kr)} + \frac{jkA}{r} e^{j(\omega t - kr)}$
- Impedance is $\frac{p}{u} = \rho c \frac{jkr}{1 + jkr}$

ENC p28

Spherical Wave Solution to Wave Eqn

- Multiply previous equation by $\frac{(1-jkr)}{(1-jkr)}$ gives

$$\frac{p}{u} = \rho c \frac{kr\,(kr + j)}{1 + k^2 r^2}$$

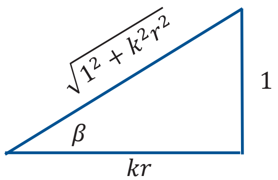
which can be re-written as $\frac{p}{u} = \rho c e^{j\beta} \cos\beta$

where the phase angle $\beta = (\theta_p - \theta_u) = \tan^{-1}\left[\frac{1}{kr}\right]$

θ_p = phase angle of pressure

θ_u = phase angle of velocity

ENC p36



Spherical Wave Solution to Wave Eqn

- So from these previous equations we said that $\frac{p}{u} = \rho c e^{j\beta} \cos\beta$
  where $\beta = (\theta_p - \theta_u)$
  θ_p = phase angle of pressure
  θ_u = phase angle of velocity

- We said that for a plane wave $\frac{p}{u} = \rho c$

Q: So looking at this equation, what is the phase angle between pressure and velocity for a plane wave?

Why is impedance an important concept?

- Knowing the relationship between pressure and particle velocity (impedance) is vital for
  - Understanding the acoustic absorption of materials
  - Acoustic response of an enclosed system

Hearing Aids

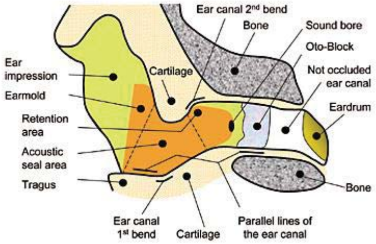
- "Super Power Behind-The-Ear Hearing Instruments
  - Sound pressure level (SPL) output typically over 135dB" (!!!)
- A "Sound Garden" concert might be around 120 dB up close.
- If I turned on this hearing aid in this room, would I get 135 dB ??
- What acoustic difference is there to cause 135 dB in the ear and the low noise level in the room ?

knowles

http://www.knowles.com/eng/Applications/Specialty-components/Hearing-aid-components#Behind-the-ear

Impedance of the ear

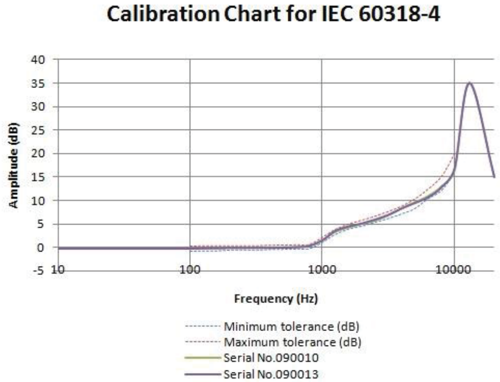
- An example you can relate to... earbud headphones
- Ever noticed you that the sound is different depending on how you insert the earbuds ?



Insertion depth and leakage will change the response

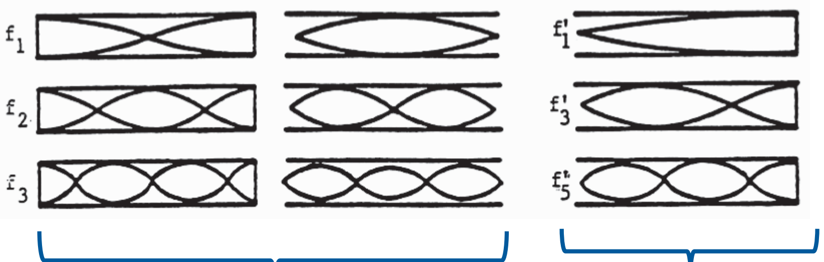
Impedance of the ear

Simulator with a standardised impedance of a human ear.



Resonance / Natural Frequencies

- organ pipe resonances (duct diameter d, length L)
- bends not seen by waves for f < 0.5 c/d
- closed both ends | open both ends | closed one end

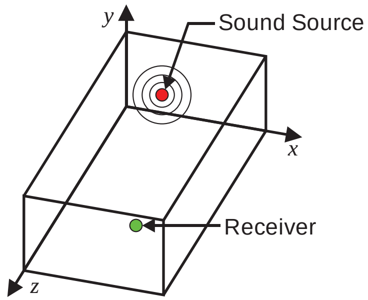


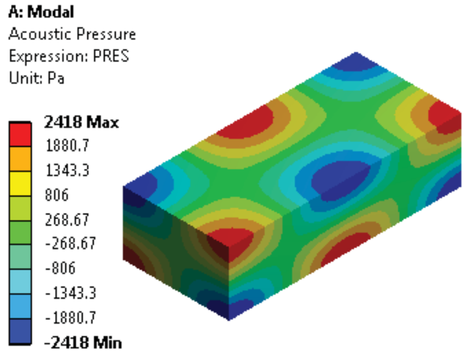
$$f = \frac{nc}{2L}$$

$$f = \frac{(2n-1)c}{4L}$$

Resonance / Natural Frequencies

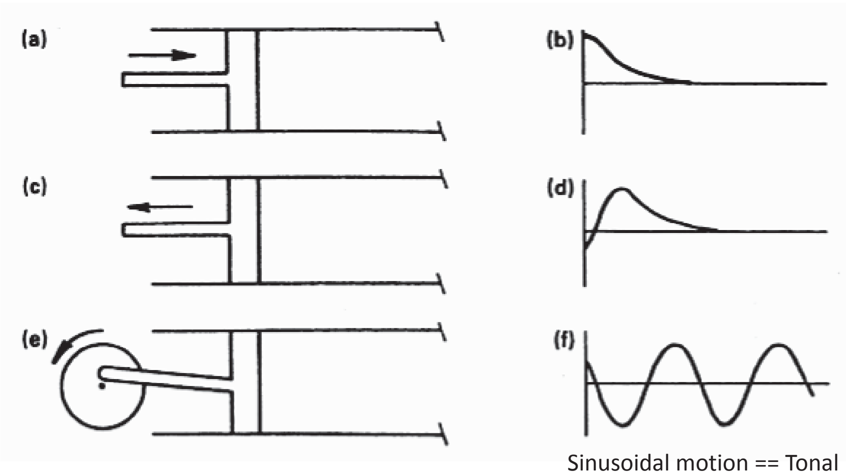
- Acoustic Modes of a Room





Acoustic pressure is longitudinal waves

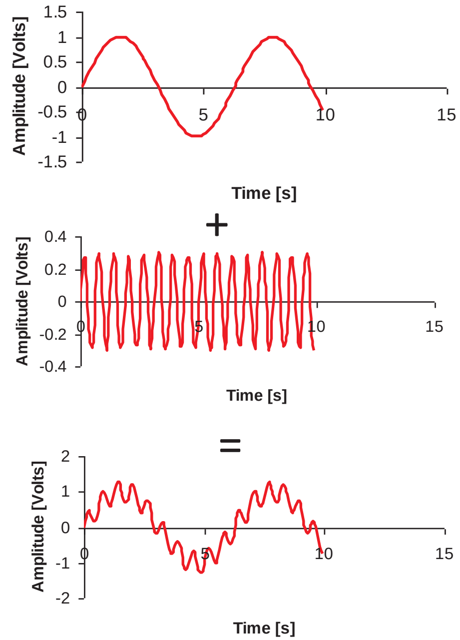
- Consider a moving piston — Equivalent pressure



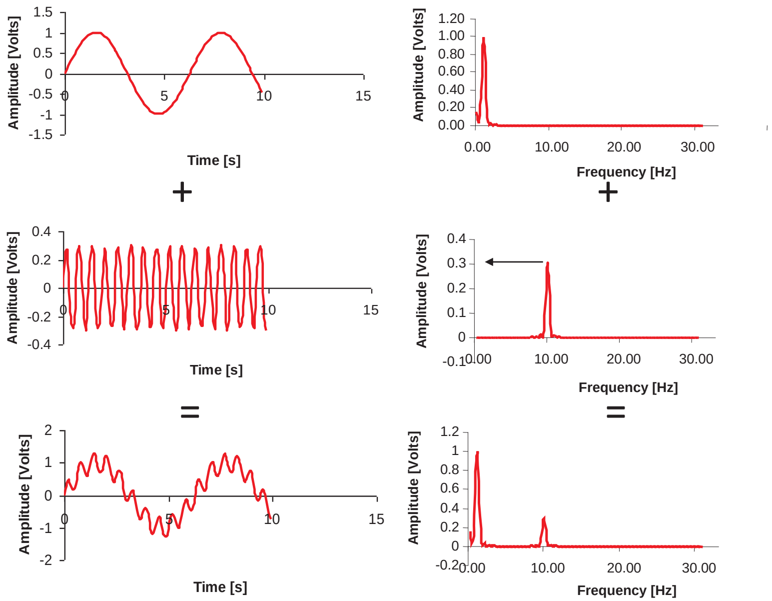
Sinusoidal motion == Tonal

Addition

- Example of adding two signals in the ***time domain***

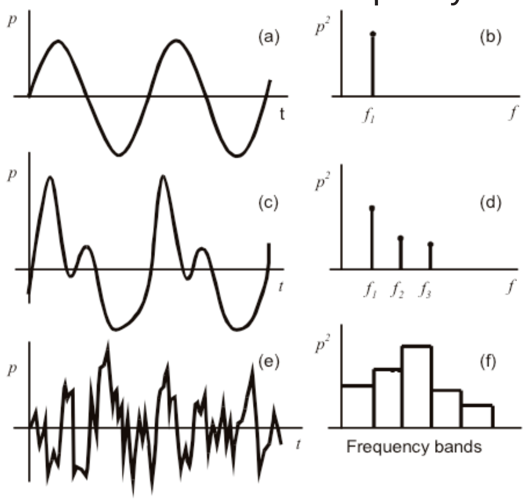


Addition in Time & Frequency Domains



Spectral Analysis

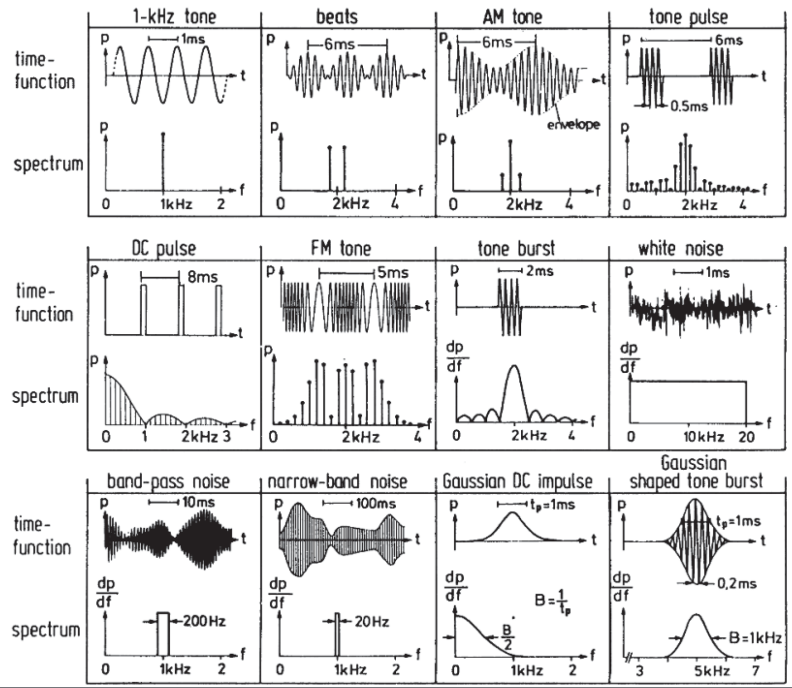
- Time Domain — Frequency Domain



ENC p42

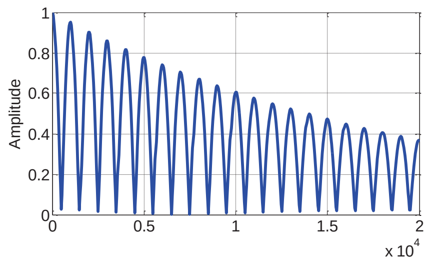
Standard Noise Types

- There are some standard noise test signals used in Psychoacoustics.
  - Pure tone, beats, amplitude-modulated tone, tone pulse, DC pulse, frequency-modulated tone, tone burst, white noise, band-pass noise, narrow-band noise, Gaussian DC impulse, Gaussian-shaped tone burst.
- The time functions and associated frequency spectra can be seen in the following figure.

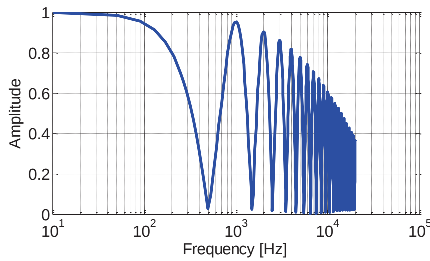


Logarithmic Frequency Scale

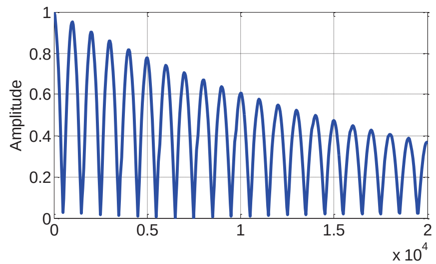
- Human ear has a HUGE frequency range:
  - 20Hz to 20,000Hz
  - (we'll talk about the huge sound level range a bit later)



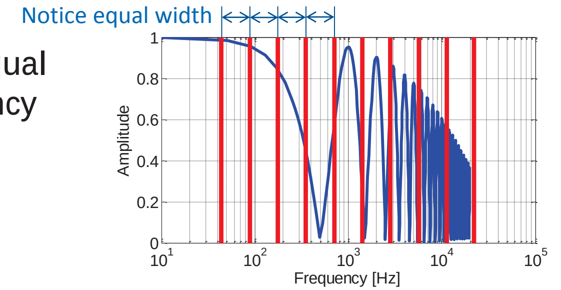
- Frequency scale is often represented on a *logarithmic* scale.
- ... which is often grouped into *octave bands* ...



Octave Bands



- Octave Bands have equal widths on log-frequency scale.



ENC4 p44, ENC5 p43

Octave Bands

- Group range of frequencies into bands.
- Most common are
  - Octave
  - One-Third Octave
- 1/3 octave band number ID (integer)
  - take 10 $\log_{10}$ of band centre frequency and round off.
  - E.g. 10 $\log_{10}$ (250)=23.979

Band Number	centre frequency	centre frequency	Lower	Upper
	OCTAVE	ONE-THIRD		
		OCTAVE		
14		25	22	28
15	31.5	31.5	28	35
16		40	35	44
17		50	44	57
18	63	63	57	71
19		80	71	88
20		100	88	113
21	125	125	113	141
22		160	141	176
23		200	176	225
24	250	250	225	283
25		315	283	353
26		400	353	440
27	500	500	440	565
28		630	565	707
29		800	707	880
30	1 000	1 000	880	1 130
31		1 250	1 130	1 414
32		1 600	1 414	1 760
33	2 000	2 000	1 760	2 250
34		2 500	2 250	2 825
35		3 150	2 825	3 530
36	4 000	4 000	3 530	4 400
37		5 000	4 400	5 650
38		6 300	5 650	7 070
39	8 000	8 000	7 070	8 800
40		10 000	8 800	11 300
41		12 500	11 300	14 140
42	16 000	16 000	14 140	17 600
43		20 000	17 600	22 500

Octave Bands

Band Number	Octave band center frequency	One-third octave band centre frequency	Band limits	
			Lower	Upper
20		100	88	113
21	125	125	113	141
22		160	141	176
23		200	176	225
24	250	250	225	283
25		315	283	353

2x

One-Third Octave Bands

Band Number	Octave band center frequency	One-third octave band centre frequency	Band limits	
			Lower	Upper
20		100	88	113
21	125	125	113	141
22		160	141	176
23		200	176	225
24	250	250	225	283
25		315	283	353

Octave Bands

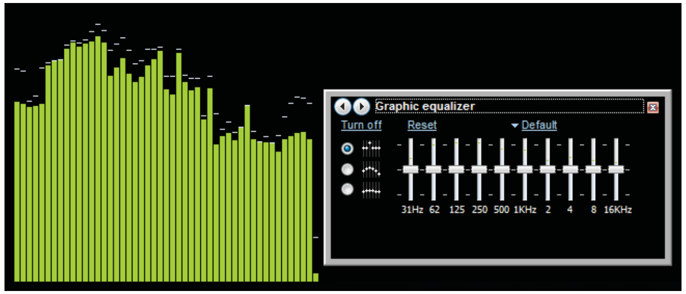
Band Number	Octave band center frequency	One-third octave band centre frequency	Band limits	
			Lower	Upper
20		100	88	113
21	125	125	113	141
22		160	141	176
23		200	176	225
24	250	250	225	283
25		315	283	353

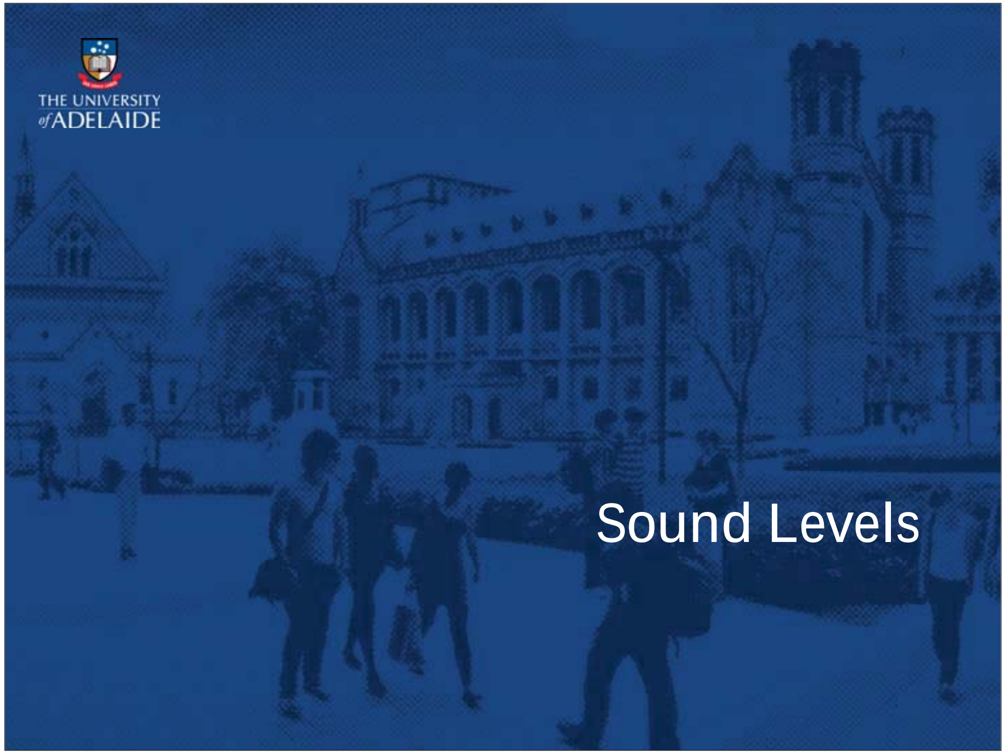
Band Number	Octave Band Center Frequency	1/3 Octave Band Center Frequency	Band Limits Lower	Band Limits Upper
-1		0.8	0.7	0.9
0	1	1	0.9	1.1
1		1.25	1.1	1.4
2		1.6	1.4	1.8
3	2	2	1.8	2.2
4		2.5	2.2	2.8
5		3.15	2.8	3.5
6	4	4	3.5	4.4
7		5	4.4	5.6
8		6.3	5.6	7
9	8	8	7	9
10		10	9	11
11		12.5	11	14
12	16	16	14	18
13		20	18	22
14		25	22	28
15	31.5	31.5	28	35
16		40	35	44
17		5	44	57
18	63	63	57	71
19		80	71	88
20		100	88	113
21	125	125	113	141
22		160	141	176

Band Number	Octave Band Center Frequency	1/3 Octave Band Center Frequency	Band Limits Lower	Band Limits Upper
23		200	176	225
24	250	250	225	283
25		315	283	353
26		400	353	440
27	500	500	440	565
28		630	565	707
29		800	707	880
30	1000	1000	880	1130
31		1250	1130	1414
32		1600	1414	1760
33	2000	2000	1760	2250
34		2500	2250	2825
35		3150	2825	3530
36	4000	4000	3530	4400
37		5000	4400	5650
38		6300	5650	7070
39	8000	8000	7070	8800
40		10000	8800	11300
41		12500	11300	14140
42	16000	16000	14140	17600
43		20000	17600	22500

Application: Filters

- Demonstration - Graphic Equaliser on WMP





Sound Pressure

- Sound pressure, p
  - VERY small fraction of atmospheric pressure (101325Pa).
    - e.g. speech about 60dB re 20μPa = 0.02 Pa (RMS), factor of 101325/0.02= 5 × 10^6
  - varies cyclically.
- Sound Pressure Level (SPL) expressed in decibels

$$\mathrm{SPL} = L_p = 20\log_{10}\frac{p_{\mathrm{RMS}}}{p_{\mathrm{ref}}}$$

Always RMS

where p_{RMS} = RMS acoustic pressure
p_{ref} = reference pressure = 20 × 10^{-6} Pa.

See also http://en.wikipedia.org/wiki/Decibel ENC p38

What does RMS mean?

- The mean square value of the pressure is defined by

$$\bar{p}^2 = \frac{1}{T}\int_0^T p^2(t)\,dt$$

- The square root of this value, called the root mean square (rms) value
  - denoted as p_{rms}
- For harmonic pressure $\boxed{p_{rms} = \frac{p_{peak}}{\sqrt{2}} = \frac{A}{\sqrt{2}}}$

Sound Pressure Level (dB)

- From previous slides, we define sound pressure level in decibels, which is defined as

$$\mathrm{SPL} = L_p = 10\log_{10}\left(\frac{p_1}{p_{\mathrm{ref}}}\right)^2 = 20\log_{10}\left(\frac{p_1}{p_{\mathrm{ref}}}\right)$$

and values are shown followed by dB re something, e.g., 50 dB re 20 μPa

[We'll see a bit later that for sound power levels we write it with a difference reference value
e.g., 63 dB re 1pW]

Range of Hearing in Humans

- 0dB re 20μPa = 0.000020 Pa
  - ½ drop of water over 10m²

- 140dB re 20μPa = 200 Pa
  - 100 x 2 litre bottles over 10m²

ENC p39

Subjective Assessment of Change in SPL

Change in sound pressure level (dB)	Change in power Decrease	Change in power Increase	Change in apparent loudness
3	1/2	2	just perceptible
5	1/3	3	clearly noticeable
10	1/10	10	half or twice as loud
20	1/100	100	much quieter or louder

ENC p85

[We're going to see this table many times in this course!]

Amplitude Descriptors

- Consider a harmonic function for pressure, say

$$p(t) = A\sin(\omega\, t + \phi\,)$$

where
A = magnitude of the pressure
ω = circular frequency (radians / s) remember $\omega = 2\pi f$
t = time
ϕ= phase angle shift

Amplitude Descriptors

- Note: these relationships only apply for harmonic pressure!
- Do not use them for other signal types

- A_{peak} is the peak value of the pressure
- $A_{peak\text{-}to\text{-}peak}$ is the difference between the positive and the negative maximum values of the pressure
- $A_{peak\text{-}to\ peak} = 2 \times A_{peak}$
- $A_{average}$ is the average value defined by:

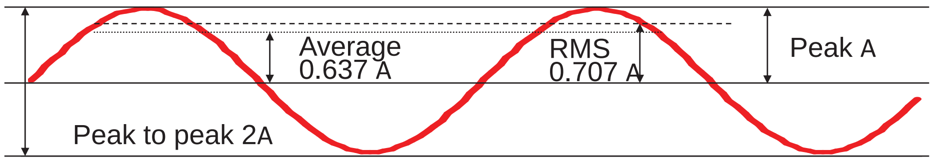
$$A_{average} = \frac{1}{T}\int_0^T |p(t)|\mathrm{d}t$$

Amplitude Descriptors

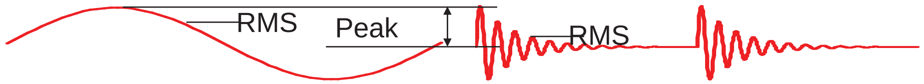
- A_{RMS} is the root mean square value (RMS) defined by:

$$A_{RMS} = \sqrt{\frac{1}{T}\int_0^T x^2(t)dt}$$

$$\begin{aligned}
A_{peak} &= 1.414 \times A_{RMS} = 1.57 \times A_{average} \\
A_{peak\text{-}to\text{-}peak} &= 2 \times A_{peak} \\
A_{average} &= 0.9 \times A_{RMS} = 0.637 \times A_{peak} \\
A_{RMS} &= 0.707 \times A_{peak} = 1.11 \times A_{average}
\end{aligned}$$



Amplitude Descriptors



Same peak values

BUT

different RMS values

Addition and Subtraction

- Addition of noise levels
  - Two coherent sources
    (tonal noise sources of the same frequency)
    the phase angle must be taken into account.

$$\langle p_t^2 \rangle = \langle p_1^2 \rangle + \langle p_2^2 \rangle + 2\langle p_1 \, p_2 \rangle \cos(\beta_1 - \beta_2)$$

ENC p47

Addition and Subtraction

Example: Two identical speakers generate a pure tone signal, each producing 85dB re 20μPa at a microphone.

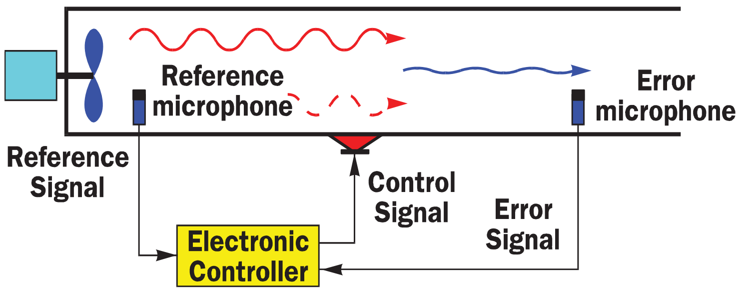
a) What would the SPL be if there were a 180 deg phase shift between the two signals?

b) What would the SPL be if both speakers operated together with no phase difference?

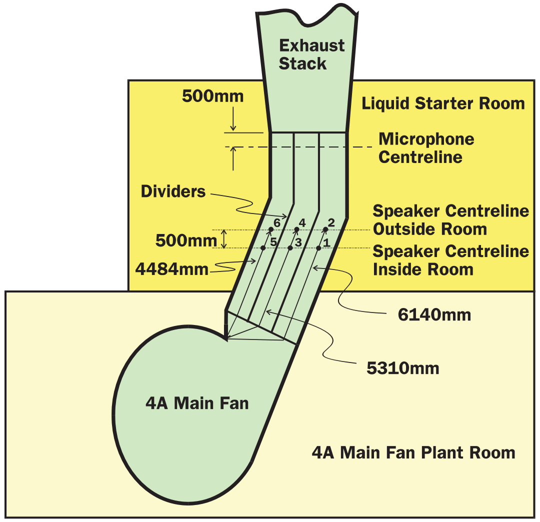
Addition and Subtraction

a) phase angle = 180 degrees

This technique is used in Active Noise Control (cancellation)



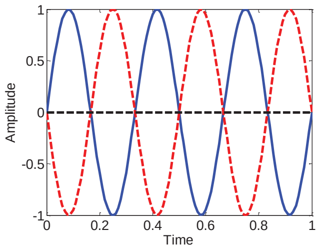
ANC in Exhaust Stack



Addition and Subtraction

a) phase angle = 180,

hence cos(180)= -1



$$\langle p_t^2 \rangle = \langle p_1^2 \rangle + \langle p_2^2 \rangle + 2\langle p_1 p_2 \rangle \cos(\beta_1 - \beta_2)$$

$$= \langle p_1^2 \rangle + \langle p_1^2 \rangle - 2\langle p_1^2 \rangle = 0$$

$$L_{p,\,\mathrm{total}} = 10\log_{10}\left[\frac{p_{\mathrm{RMS}}^2}{p_{\mathrm{ref}}^2}\right]$$

$$= -\infty \text{ dB re } 20\mu\text{Pa}$$

Addition and Subtraction

b) phase angle =0, hence cos(0)=1

$$\mathrm{SPL} = L_p = 20\log_{10}\frac{p_{\mathrm{RMS}}}{p_{\mathrm{ref}}} = 10\log_{10}\left[\frac{p_{\mathrm{RMS}}}{p_{\mathrm{ref}}}\right]^2$$

$$p_{1,\,\mathrm{RMS}}^2 = p_{2,\,\mathrm{RMS}}^2 = p_{\mathrm{ref}}^2 10^{L_p/10}$$

$$\langle p_t^2 \rangle = \langle p_1^2 \rangle + \langle p_2^2 \rangle + 2\langle p_1 p_2 \rangle \cos(\beta_1 - \beta_2)$$

$$= 4\langle p_1^2 \rangle = 4\times\left[p_{\mathrm{ref}}^2 10^{L_p/10}\right]$$

$$L_{p,\,\mathrm{total}} = 10\log_{10}\left[\frac{p_{\mathrm{RMS}}^2}{p_{\mathrm{ref}}^2}\right] = 10\log_{10}\left[\frac{4\times\left(p_{\mathrm{ref}}^2 10^{L_p/10}\right)}{p_{\mathrm{ref}}^2}\right]$$

$$= L_{p,1} + 10\log_{10}(4) = 85 + 6 = 91 \text{ dB re } 20\mu\text{Pa}$$

Addition and Subtraction

Addition of noise levels

– Incoherent sources (random, unrelated)

$$L_p = 10\log_{10}\left[10^{L_{p1}/10} + 10^{L_{p2}/10} + 10^{L_{p3}/10} + \dots\right]$$

ENC p48

Why are we adding p² 's together?

Addition and Subtraction

Example of addition of incoherent noise levels:

Q: What is the combined noise level due to 3 incoherent sound sources if each produces a sound pressure level at a given location of 93, 90 and 87 dB respectively, when operating alone?

A: $$L_p = 10\log_{10}\left[10^{93/10} + 10^{90/10} + 10^{87/10}\right]$$

$$= 95.4 \text{ dB re } 20\mu\text{Pa}$$

Note: very rare to quote values more accurate than 0.1dB

Addition and Substraction

Subtraction of SPLs

$$L_p = 10\log_{10}\left[10^{L_{pt}/10} - 10^{L_{pb}/10}\right]$$

ENC p50

Addition and Subtraction

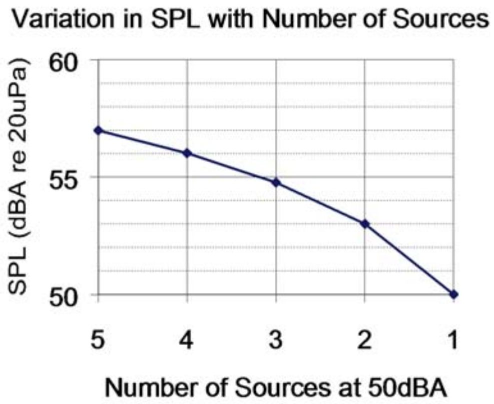
Example of subtraction of SPLs :

Q: The sound pressure level measured at the operator's position on a machine was 94dB in the 500 Hz octave band with the machine operating and 90dB with the machine turned off. What is the contribution due to the machine alone?

A: $$L_p = 10\log_{10}\left[10^{94/10} - 10^{90/10}\right] = 91.8 \text{ dB re } 20\mu\text{Pa}$$

Addition and Subtraction

- This subtraction maths is REALLY important and deceptively misleading.
- In order to achieve an overall Reduction in SPL requires reduction of ALL noise sources
- A 20dB reduction of one of five sources, might only mean an overall reduction 1 dB.



Addition and Subtraction

Combining sound pressure level reductions

- used when sound can get to a receiver from a source along more than one path. Each path has a noise attenuation associated with it which is the reduction in level (in dB) over that for direct line-of-sight propagation. Sometimes the overall reduction may be negative (i.e. increases noise level).

- Overall Noise Reduction

$$NR = L_{pR} - L_p = -10\log_{10}\sum_{i=1}^{n}10^{-(NR_i/10)}$$

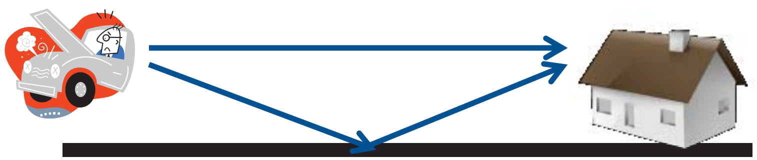
ENC p51

- L_{PR} = level due to direct line of sight propagation
- L_p = observed level
- Note the direct line of sight path (NR_1 =0) must be included in the sum if it exists

Addition and Subtraction

EXAMPLE:

Q: Sound propagates over the ground from a noisy diesel engine to a residence 100m away. If the wave reflected from the ground suffers an attenuation of 3dB, what is the overall noise reduction at the residence over what it would be if there were no ground reflected wave?



Addition and Subtraction

$$NR = L_{pR} - L_p = -10\log_{10}\sum_{i=1}^{n}10^{-(NR_i/10)}$$

$$NR = -10\log_{10}\left[10^{-3/10} + 10^{-0/10}\right] = -1.8dB$$

- i.e. a 1.8 dB increase.

Addition and Subtraction

Combining level reductions

- If it is desired to calculate the effect of an alteration to either the number of paths or the attenuation due to any of the paths then

$$NR = 10\log_{10}\sum_{i=1}^{n_A} 10^{-(NR_{Ai}/10)} - 10\log_{10}\sum_{i=1}^{n_B} 10^{-(NR_{Bi}/10)}$$

ENC p51

NR = noise reduction due to alteration

A = paths before alteration

B = paths after alteration

Addition and Subtraction

- EXAMPLE: Suppose there were two paths in a product by which noise escaped. One was a direct path with 0dB noise reduction and the other path had a 3dB noise reduction. Applying noise control eliminated both of these paths but introduced 4 new paths for which the noise reductions were 8, 12, 12 and 16 dB respectively. What was the overall noise reduction achieved?

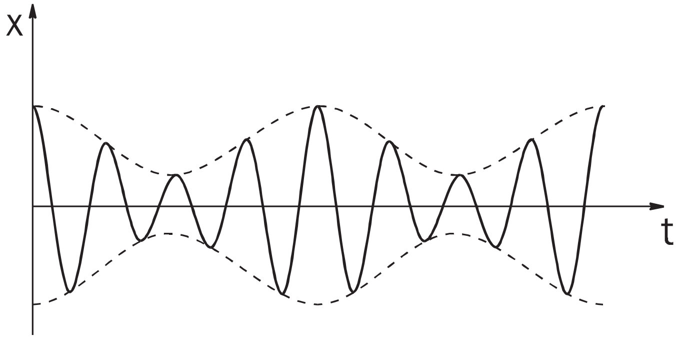
$$NR = 10\log_{10}\left[10^{-3/10} + 10^{0/10}\right]$$

$$-10\log_{10}\left[10^{-8/10} + 10^{-12/10} + 10^{-12/10} + 10^{-16/10}\right]$$

$$NR = 1.8 + 5.1 = 6.9\ \text{dB}$$

Beating

- When two sounds of slightly different frequencies are added the total mean square sound pressure rises and falls cyclically with time and the phenomenon known as ***beating....*** VERY annoying.



Beating

Beating

Beating

	F1 + F2	
	400 & 400.5 Hz	1 beat every 2 seconds
	400 & 401 Hz	1 beat per second
	400 & 403 Hz	3 beats per second
	400 & 410 Hz	10 beats per second
	400 & 420 Hz	can you still hear …
	400 & 430 Hz	can you still hear …
	400 & 440 Hz	interference beats?

Psycho-Acoustics

Equivalent SPL

- Equivalent continuous sound level [dB]

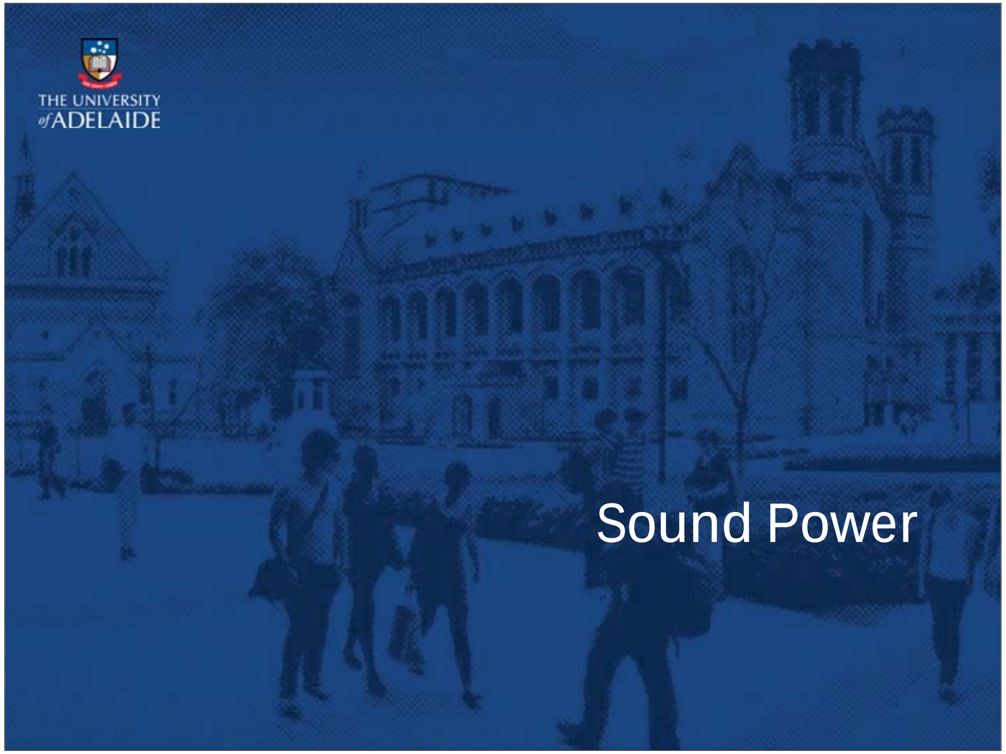
$$L_{eq} = 10\log_{10}\left[\frac{1}{T}\int_0^T 10^{Lp/10}\,dt\right]$$

ENC p106

For example,
Q: What is the equivalent SPL over an 8-hour period?

Very important concept for quantifying noise emission and noise exposure for people.

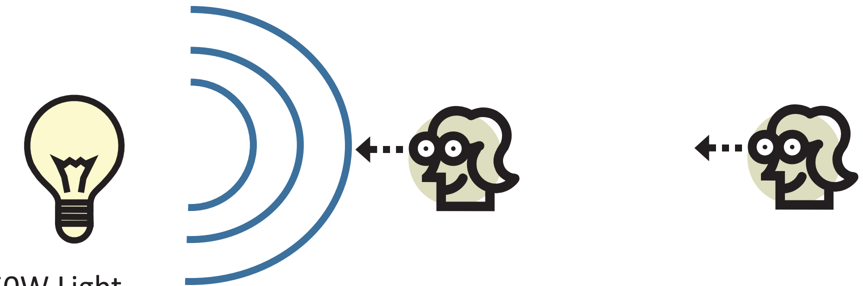
[We'll see a lot more on this when we get to the Noise Criteria section]



Sound Power

Light Globe Analogy

- A 60W light globe always has the same power rating, but the brightness of the light will depend on the distance from the light source.



60W Light Globe

Acoustic Intensity

In order to calculate sound power, we need to know about *acoustic intensity*

- defined as the product of pressure and particle velocity and is thus a vector quantity $I = p \times u$
- real part (product of acoustic pressure, p and in phase part of acoustic particle velocity, u) is a measure of the energy of the disturbance
- can be used for source sound power measurements in noisy environments
- identifies sources of sound (as it is a vector)

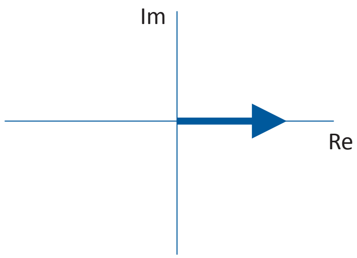
ENC p34

Acoustic Intensity

Acoustic pressure can be written as

$$p = P\sin(\omega t + \theta_p)$$

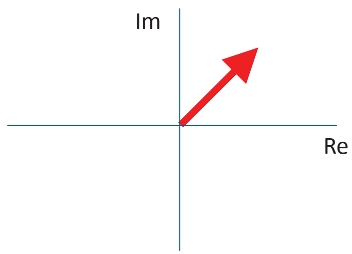
which represents a rotating vector



Likewise, acoustic velocity can be written as

$$u = U\sin(\omega t + \theta_u)$$

which represents a rotating vector



Acoustic Intensity

- time averaged or active intensity is

$$I(r) = \langle p(r,t)u(r,t)\rangle = \lim_{T\to\infty}\frac{1}{T}\int_0^T p(r,t)u(r,t)dt$$

ENC p34

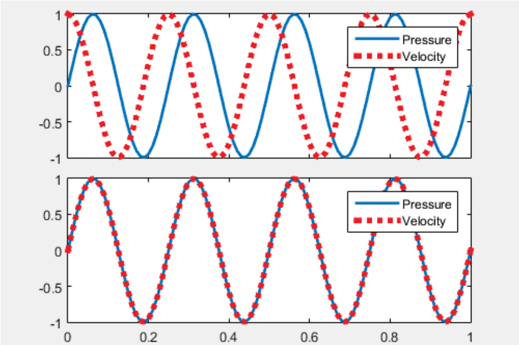
- For sinusoidal waves

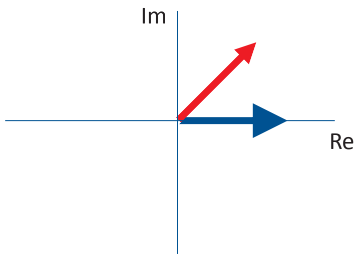
$$p = P\sin(\omega t + \theta_p) \qquad u = U\sin(\omega t + \theta_u)$$

- The active (real) intensity is $I = \dfrac{PU}{2}\cos(\theta_p - \theta_u)$
- The reactive (imaginary) intensity is $I_r = \dfrac{PU}{2}\sin(\theta_p - \theta_u)$

Acoustic Intensity

What is important is the magnitude and phase angle between p and u. $I = \frac{PU}{2}\boxed{\cos(\theta_p - \theta_u)}$

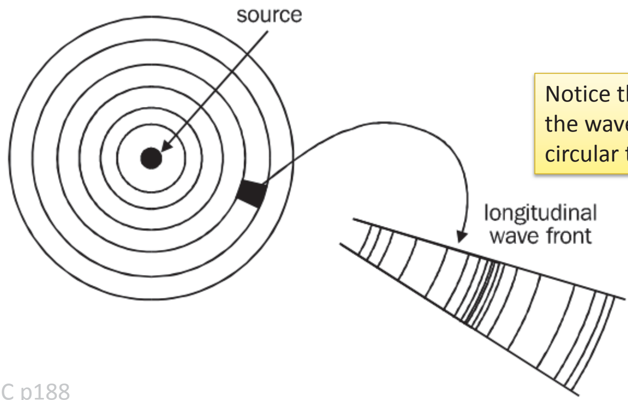




If $(\theta_p - \theta_u) = 90°$ then the intensity is ___ ?

Acoustic Intensity

- Waves radiating from a source become planar at a great distance (far field)
  – In this case *p* and *u* are in-phase



Notice that at large distances the wavefront goes from circular to nearly straight.

ENC p188

Acoustic Intensity

- Waves radiating from a source become planar at a great distance (far field)
  – In this case *p* and *u* are in-phase and

$$\boxed{I = \frac{p_{rms}^2}{\rho c}}$$

p_{rms} = root mean square pressure
ρ = density of medium (kg/m^3)
c = speed of sound (m/s)

ENC p36

- Sound intensity level

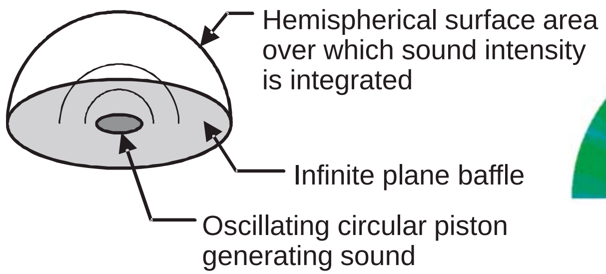
$$\boxed{L_I = 10\log_{10}\left[\frac{I}{I_{ref}}\right]}$$

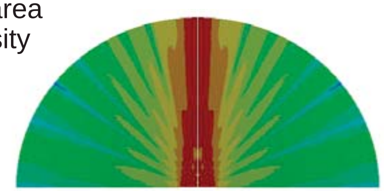
ENC p40

L_I = sound intensity level (dB)
I = sound intensity (W / m^2)
I_{ref} = reference intensity ($10^{-12}\ W/m^2$)

Sound Power

- Sound power is basically the integration of sound intensity over an imaginary surface that surrounds the source.
- Example: sound power from a loudspeaker (which is modelled as an oscillating piston).





Sound Power

- Average intensity measured on an imaginary surface (in a direction, *n*, normal to the surface) surrounding a source multiplied by the imaginary surface area

$$W = \int_A I \cdot n \, dA$$

- Sound power level

$$\boxed{L_W = 10\log_{10}\left[\frac{W}{W_{ref}}\right]}$$

ENC p39

L_W = sound power level (dB)
W = sound power (W)
W_{ref} = reference power ($10^{-12} W$)

Sound Power

- How are sound pressure level and sound power level related?

$$p_{rms}^2 = \rho c I = \frac{\rho c W}{4\pi r^2}$$

r is distance from sound source at which the acoustic pressure = *p*

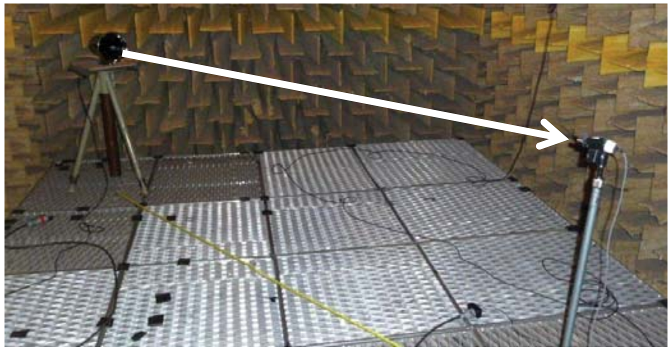
$$L_p = L_w + 10\log_{10}\left(\frac{\rho c}{400}\right) - 10\log_{10}\left(4\pi r^2\right)$$

- In air at 20°C $L_p = L_w - 10\log_{10}\left(4\pi r^2\right)$

Sound Power

This will be covered in more detail later

- How is sound power measured?
  - Measure the SPL at a known distance from the source

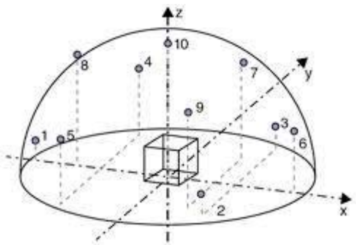


Sound Power

- How is sound power measured?
  - Use a reverberation chamber to measure the average SPL in the room.

Sound Power

- How is sound power measured?
  - Use a sound intensity probe and scan over an imaginary surface.



SPL Decreases with Distance

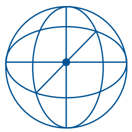
- If the sound pressure level, L_m, is measured at some reference distance, r_m, from the noise source (usually far enough to avoid source near field effects which complicate the sound field close to a source), then the sound pressure level at some other distance, r, may be estimated using:

$$L_p = L_m - 20\log_{10}\left(\frac{r}{r_m}\right)$$

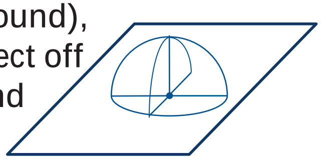
Assuming spherical spreading

Reflection Effects

- If there are reflecting surfaces nearby, then the sound power that is radiated is confined.
- Imagine a simple source that radiates sounds equally in all directions.



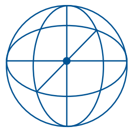
- If there is a reflecting plane (e.g. the ground), then the radiated sound power will reflect off the surface into the air, so that the sound pressure level at a point will be louder compared to the omni-directional case.



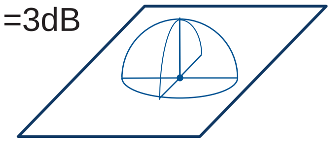
ENC p220

Directivity

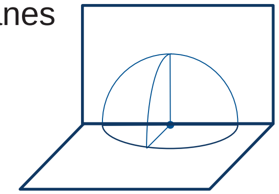
- Free Space
  D=1
  DI=0dB



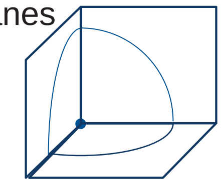
- Centre large plane
  D=2
  DI=3dB



- Two planes
  D=4
  DI=6dB



- Three Planes
  D=8
  DI=9dB



Reflection Effects

Situation	Directivity factor, Q	Directivity Index, DI (dB)
free space	1	0
centred in a large flat surface	2	3
centred at the edge formed by the junction of two large flat surfaces	4	6
at the corner formed by the junction of three large flat surfaces	8	9

ENC p221

Directivity

- Directivity is defined in terms of the sound intensity measured at an angle I_θ, divided by the average intensity I_{av}

Directivity factor $$Q_\theta = \frac{I_\theta}{I_{av}}$$

where $$I_{av} = \frac{W}{4\pi r^2}$$

- Directivity Index, $DI = 10\log_{10}(Q_\theta)$

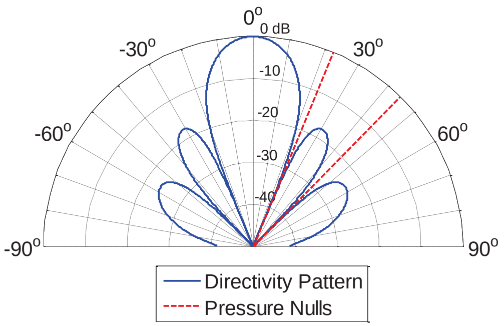
ENC p219

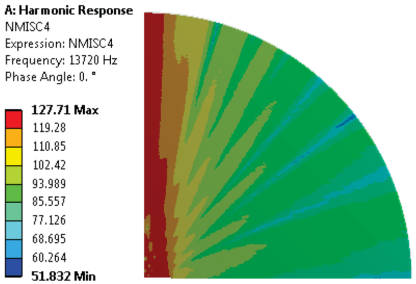
Baffled Piston

Covered Later in Sound Sources

- At low frequencies it radiates omni-directionally.
- At higher frequencies the angular directivity also has pressure zeros at certain angles.

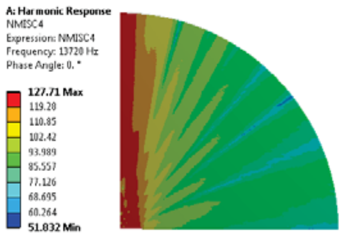
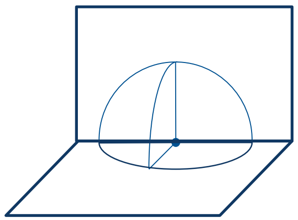
Beam Pattern of a Baffled Circular Piston





Directivity and Reflection Effects

- Directivity and reflection effects are related.
- Nearby surfaces will reflect sound and this changes the *effective* directivity of the source.
- Don't get confused between the two concepts.

the source can have directivity such that it radiates sound better at a particular angle.	a nearby reflecting surface can change the radiated sound.
	

Reflection Effects

- Consider a sound power source ...

$$W = I\frac{4\pi r^2}{Q} = p_{rms}^2\frac{4\pi r^2}{\rho c Q}$$

- This can be re-arranged to

$$p_{rms}^2 = \frac{W\rho c Q}{4\pi r^2}$$

Reflection Effects

- Divide by the appropriate reference values

$$\frac{p_{rms}^2}{p_{\text{ref}}^2} \times \frac{1}{W_{\text{ref}}} = \frac{W\rho c Q}{4\pi r^2} \times \frac{1}{W_{\text{ref}}} \times \frac{1}{p_{\text{ref}}^2}$$

- Do a bit of a shuffle ...

$$\left[\frac{p_{rms}^2}{p_{\text{ref}}^2}\right] = \left[\frac{Q}{4\pi r^2}\right] \times \left[\frac{W}{W_{\text{ref}}}\right] \times \left[\frac{\rho c \times W_{\text{ref}}}{p_{\text{ref}}^2}\right]$$

Reflection Effects

- Then take $10 \times \log_{10}$ on each side

$$\underbrace{10\log\left[\frac{p_{rms}^2}{p_{\text{ref}}^2}\right]}_{L_p} = 10\log\left[\frac{Q}{4\pi r^2}\right] + \underbrace{10\log\left[\frac{W}{W_{\text{ref}}}\right]}_{L_w} + 10\log\left[\frac{\rho c \times W_{\text{ref}}}{p_{\text{ref}}^2}\right]$$

$$L_p = L_w + 10\log_{10}\left[\frac{Q}{4\pi r^2}\right] + 10\log\left[\frac{\rho c \times 10^{-12}}{\left(20\times 10^{-6}\right)^2}\right]$$

$$\boxed{L_p = L_w + DI + 10\log_{10}\left[\frac{1}{4\pi r^2}\right] + 10\log\left[\frac{\rho c}{400}\right]}$$

ENC p220, p263

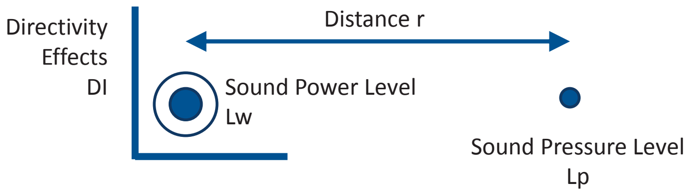
Reflection Effects

- For "normal" atmospheric conditions $\rho = 1.21\ \text{kg}/\text{m}^3$, $c = 343\ \text{m/s}$, so the last term
$$10\log_{10}\left[\frac{1.21 \times 343}{400}\right] = 0.16\ \text{dB}$$
and is often ignored.
- So what does all this mean ....

$$\underbrace{L_p}_{\text{Sound pressure level}} = \underbrace{L_w}_{\text{Sound power level}} + \underbrace{DI}_{\text{Directivity}} + \underbrace{10\log_{10}\left[\frac{1}{4\pi r^2}\right]}_{\text{Distance from source}} + \underbrace{10\log\left[\frac{\rho c}{400}\right]}_{\text{Acoustic medium}}$$

Reflection Effects

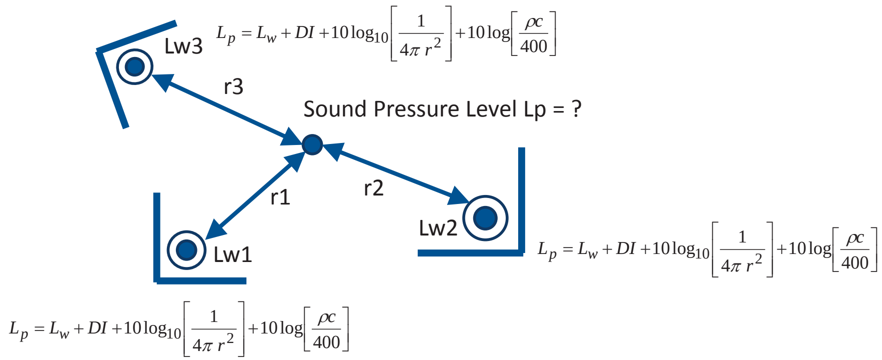
- Try to translate the maths into physical concepts, and then it is much easier to understand !



$$L_w + DI + 10\log_{10}\left[\frac{1}{4\pi\, r^2}\right] + 10\log\left[\frac{\rho c}{400}\right] = L_p$$

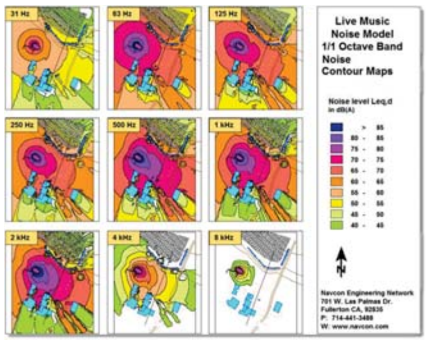
Predicting Sound Pressure Level

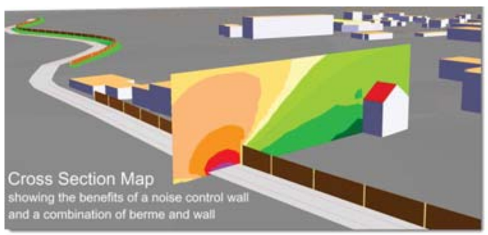
- This equation is important for estimating the SPL at a location due to multiple sources.



Predicting Sound Pressure Level

But when there are many sources, many reflections, many receivers, it gets cumbersome to do by hand calcs, so we'd use engineering software, e.g., SoundPlan, CadnaA





Subjective Assessment of Change in SPL

- To get a *clearly noticeable* change in SPL, it is necessary to reduce sound power level by 1/3.

Change in sound pressure level (dB)	Change in power Decrease	Change in power Increase	Change in apparent loudness
3	1/2	2	just perceptible
5	1/3	3	clearly noticeable
10	1/10	10	half or twice as loud
20	1/100	100	much quieter or louder

ENC p85

Questions

So now I could ask you some interesting questions …

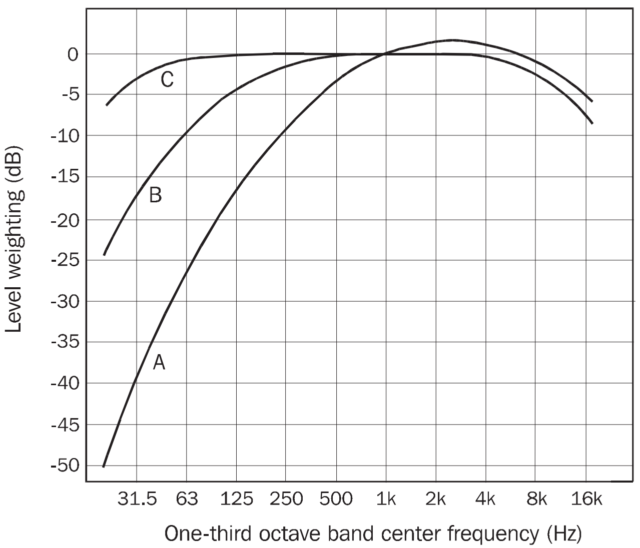
- How much more sound power is required for a source to sound twice as loud?
  - A disappointed car audio enthusiast has a 100W audio amplifier installed, but wants it to sound twice as loud as currently. How many 100W amplifiers should he have installed for it to sound subjectively twice as loud as currently?

- How far away from a source would you need to be for it to subjectively sound half as loud?

Questions

- The sound pressure level measured from a noise source radiating omni-directionally will decrease at a rate of ___ dB per doubling of distance?
- If a noise source initially located in the middle of a concrete floor were relocated to the corner between two reflective planes, one would expect the SPL to increase by __ dB ?
- What is the frequency range of the 250 Hz one-third octave band?
- A quarter-wavelength-tube muffler of length 1m can be modelled as a closed-open pipe. What is the 2nd resonance frequency of the closed-open pipe, assuming the air temperature is 400°C?

Weighting Scales



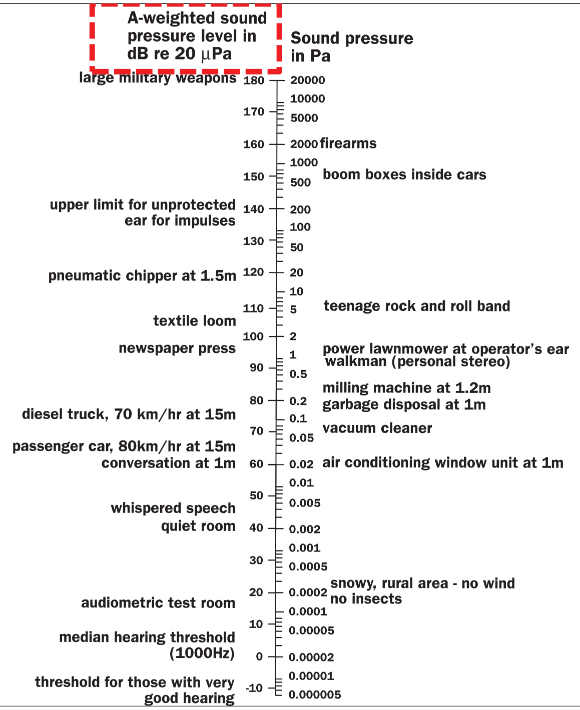
- **A-weighting** most common.
- Low frequencies have low weighting.
- *Roughly* describes human hearing response.

ENC p104

A-Weighting Adjustments

Frequency (Hz)	*A-weighting correction*	*Frequency (Hz)*	*A-weighting correction*	*Frequency (Hz)*	*A-weighting correction*
10	-70.4	160	-13.4	2500	1.3
12.5	-63.4	200	-10.9	3150	1.2
16	-56.7	250	-8.6	4000	1.0
20	-50.5	315	-6.6	5000	0.5
25	-44.7	400	-4.8	6300	-0.1
31.5	-39.4	500	-3.2	8000	-1.1
40	-34.6	630	-1.9	10000	-2.5
50	-30.2	800	-0.8	12500	-4.3
63	-26.2	1000	0.0	16000	-6.6
80	-22.5	1250	0.6	20000	-9.3
100	-19.1	1600	1.0		
125	-16.1	2000	1.2		

ENC p104



A-Weighting Adjustments

How to calculate A-weighted SPL

Freq [Hz]	400	500	630
Linear SPL [dB]	50	60	70
A-weighting correction [dB]	-4.8	-3.2	-1.9
Corrected value [dB]	45.2	56.8	68.1
Squared pressure p^2	$10^{(45.2/10)}$	$10^{(56.8/10)}$	$10^{(68.1/10)}$

$$L_p = 10\log_{10}\left[10^{L_{p1}/10} + 10^{L_{p2}/10} + 10^{L_{p3}/10} + \ldots\right]$$

$$L_p = 10\log_{10}\left[10^{45.2/10} + 10^{56.8/10} + 10^{68.1/10} + \ldots\right] = 68.4\,\text{dB(A) re } 20\,\mu\text{Pa}$$

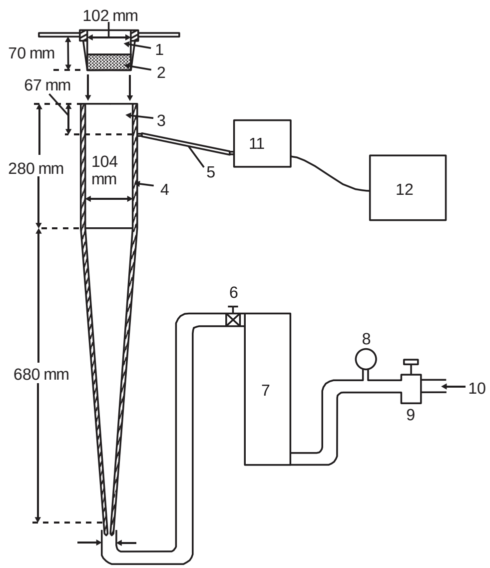
Flow Resistance

ENC p53

- Porous materials absorb sound. VITAL for reducing noise.
- The property of a porous material which indicates its usefulness as an acoustic absorber is **flow resistance**, which is the resistance of the material to an induced flow as a results of a pressure gradient.
- (See next slide for how it is measured)

Flow Resistance

- Example of apparatus used to measure flow resistivity.
- Sample is placed in top holder (1).
- Air is blown into system (10).
- Pressure is measured (11).



Flow Resistance

- Flow resistance of material is calculated as:

$$R_f = \Delta P\, S / V_0 \text{ mks rayls}$$

ENC p53

$$R_1 = R_f \;/\, \ell \quad \text{flow resistivity (mks rayls/m)}$$

R_f = flow resistance of material (mks Rayls)
R_1 = flow resistance of unit thickness (mks Rayls/m)
ΔP = pressure drop across layer (Pa)
S = area (m^2)
V_0 = mean volume flow (m^3/s)
ℓ = material thickness (m)

- R_f has units same as for specific acoustic impedance ρc.
- R_1 is flow resistance of unit thickness.

Absorption

- The flow resistance can be related to the acoustic sound absorption property of the material.
  See Appendix C in textbook.

- Sound absorption property of a material can also be measured by
  - Impedance tube.
  - Reverberation chamber measurements (which you will do in a laboratory practical!).

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Carl Howard

General Noise Control

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Contents

- Basics of Acoustics
- Vibro-acoustic Noise Control
- Air-borne Noise Control
- Liquid-borne Noise Control
- Building Acoustics

- Some demonstrations along the way...

Demonstrations

- Hypersonic Sound Source
  - Directional audio

- Vibrating magnet attached to a plate
  - Vibrating structures generate sound

- Electric shaver in a box
  - An acoustic enclosure and vibration isolation

Learning Outcomes

- Understand noise and vibration transmission concepts
- Understand commonly used noise control solutions
- Ability to select an appropriate noise control technique

Purpose of Teaching and Learning This Material:

- Develop basic conceptual understanding of noise and vibration transmission.
- Understand some common noise control techniques.
- Prevent costly mistakes.

If you only remember one thing from this lecture ....

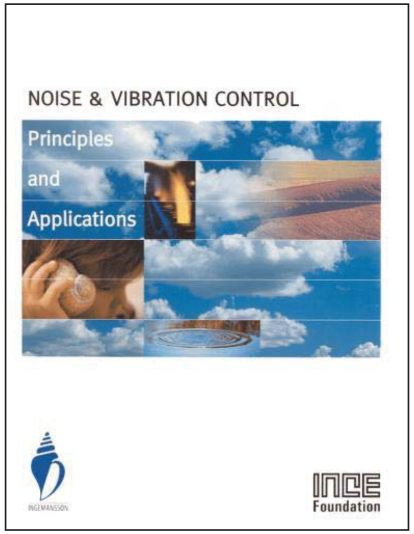
- If noise or vibration will be important, you must **PLAN** for it at the very start of the project.
- Noise and vibration is a ***SYSTEM*** level problem.
- It is cheaper and easier to account for N&V issues in the design stages that a patch-up job after the system is built.

Questions to Think About

- A roof mounted air-conditioner is causing complaints by a neighbour. What could be wrong? How could you fix the issue?
- A factory has metal parts that fall into a collection bin that generates loud noise. How could this problem be addressed?
- An air-conditioning duct is generating whistling noise. What is the likely cause of the noise?
- How does a silencer attenuate sound when there is a line-of-sight hole straight through the device?

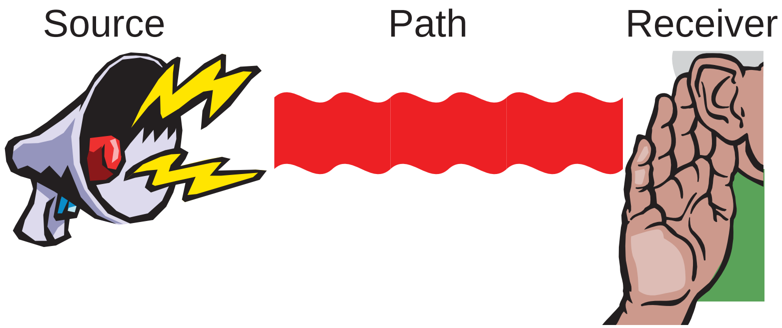
References

- This excellent "book" contains principles of noise and vibration control.
- Buy from: www.ingemansson.com



Noise Propagation Model

Noise and vibration can be reduced at the



The best method is to reduce the noise at the source.

Don't just look at noise control at the receiver for a "band-aid" solution (eg wear earmuffs)

Comment About Understanding Acoustics

- You could get overwhelmed with the number of formulas in this course.... Honestly, I don't remember all of them, and it is not necessary to know them – just look it up.
- I said in intro lecture to focus on the concepts, then you can find the appropriate formula. (It took me a while to figure that out)
- All the acoustics concepts that we'll cover in this course follow this basic noise propagation model, and there is a bit of physics to understand for each segment, and then you can look up the formula.
  Keep this in mind, so you focus on the "big picture".
  Let me elaborate....

Noise Propagation Model

Each segment, Source / Path / Receiver, can be described by a particular system / mechanism.

Noise Source
Lp, Lw, DI
Machine
Monopole / dipole / quadrupole / line source / area source / Piston: baffled / unbaffled

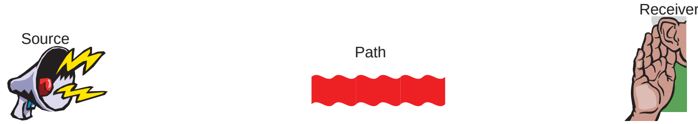
Path
Reflections
Diffraction
Outdoor
Room acoustics
Sound absorbers
TL through panel
Barrier
NR through duct and silencer

Receiver
Person / animal
Microphone

Noise Propagation Model

Let's try an example:
A car driving with passengers and pedestrians next to road.

List the noise sources, paths, and receivers.



Noise Sources

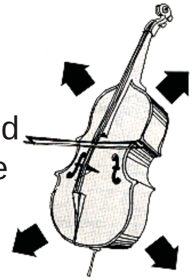
Paths

Receivers

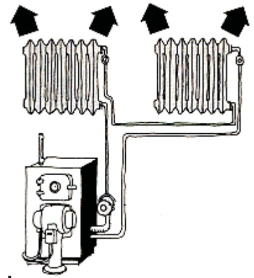
Airborne sound is usually produced by vibration of solids and fluids

- Typically a surface vibrates and pushes the air in front of it and it generates sound.

Bowing a bass causes the bridge and the sound box to vibrate that radiates sound.

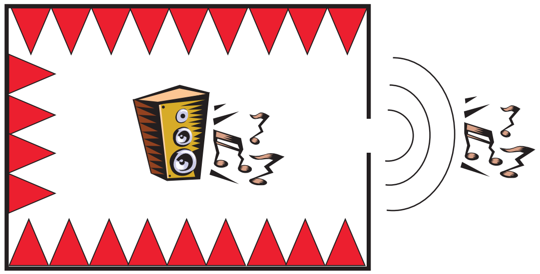


A pump generates pressure pulses in the fluid, which causes the radiator to vibrate and radiate sound.



Gaps in an enclosure are extremely bad for noise reduction

- Small openings in an enclosure can severely reduce the effectiveness of an enclosure.
- Fix the holes first before attempting to fix other problems.

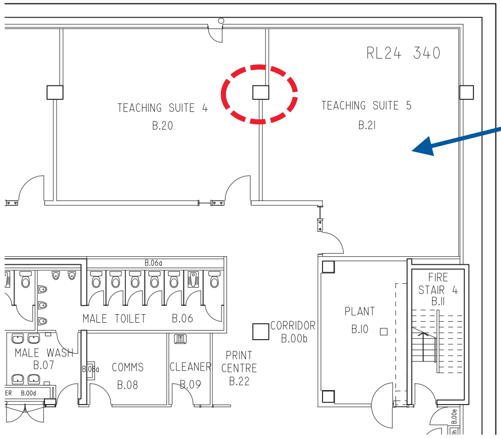


Common Mistakes

- Gaps are the most common mistake in noise control.
  - Holes to allow piping into enclosures.
  - Ventilation that is not designed correctly.
  - Doors without seals.

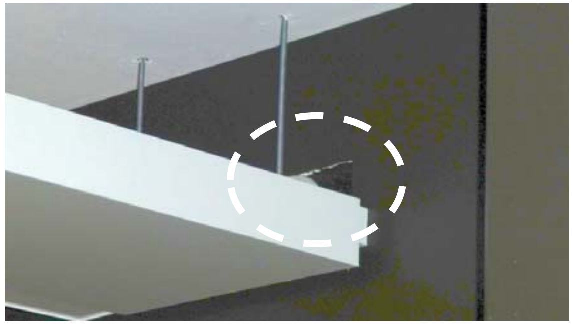
Practical Example

- Hole between lecture rooms B.20 and B.21 in basement of Ingkarni Wardli building.



Practical Example

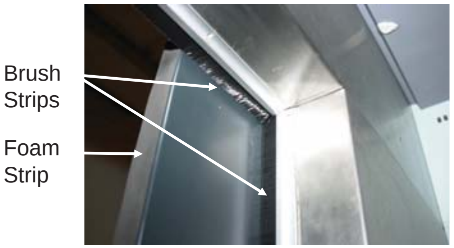
- Hole in wall allows sound to penetrate between rooms
  - Noise causes interference between lectures.

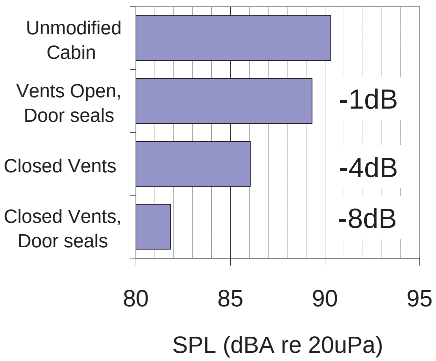


Used this as an assignment problem for 4th year Eng Acoustics course.

Practical Application

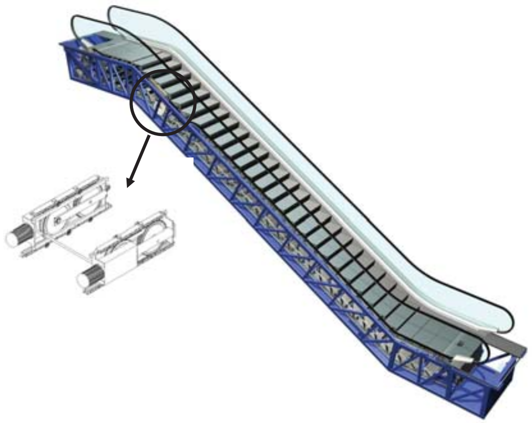
- The noise inside an elevator cabin can be reduced by closing all the gaps.
- Door seals + treated ventilation reduces cabin noise by 8dB.

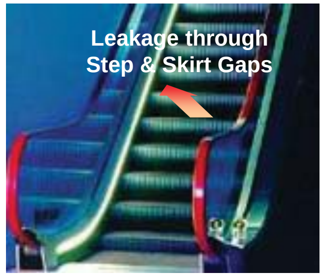




Practical Application

- The drive on an escalator generates noise that escapes through the gaps between the steps.

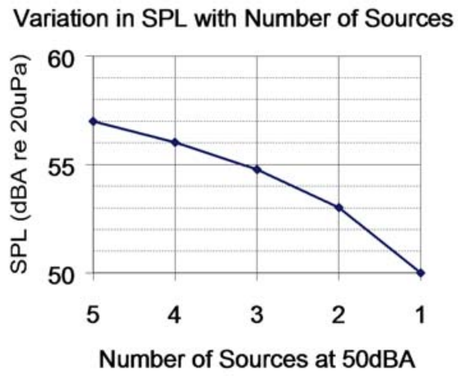




Overall Reduction in SPL Requires Reduction of ALL Noise Sources

Remember this slide from the lecture on Fundamentals

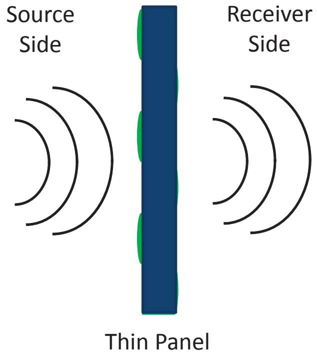
- A 20dB reduction of one of five source, might only mean an overall reduction 1 dB.

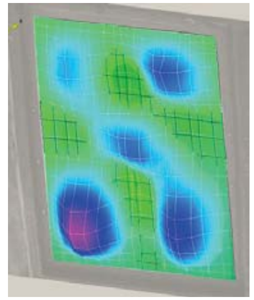


Transmission Loss of a Panel

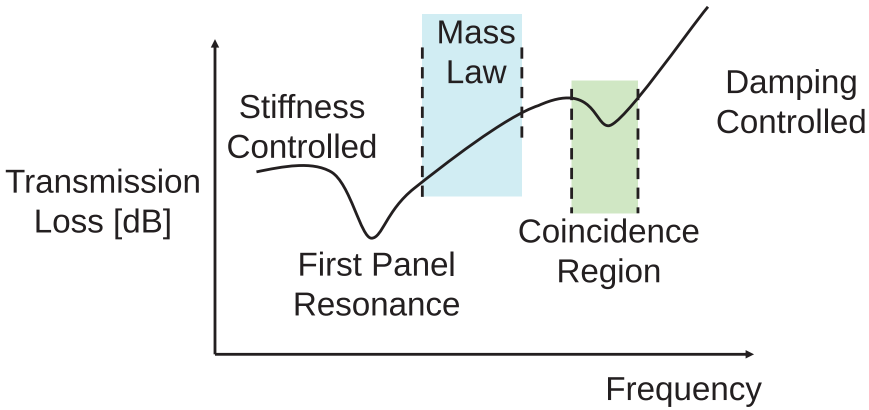
To be covered in Topic 12 Enclosures

1. Incident sound strikes panel.
2. Panel vibrates.
3. Generates sound on opposite site.





Typical Transmission Loss of a Panel



Transmission Loss in Mass Law Range

- The TL of a simple panel in the mass law frequency range can be approximated as

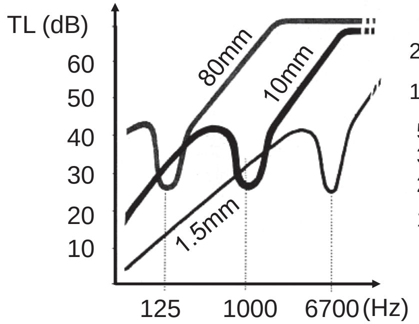
We'll cover this later in depth

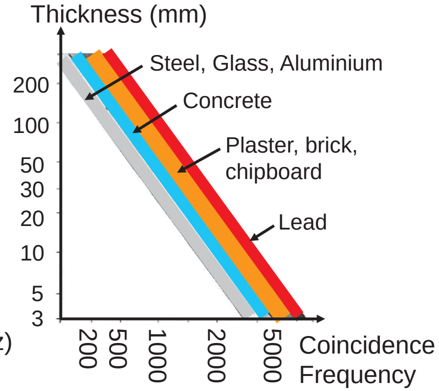
$$\boxed{TL = 20\log_{10}(f\ \rho_s) - 47}$$

TL = Transmission loss in dB
f = frequency in Hz
ρ_s = surface density = density * thickness in kg / m^2

A single wall has poor insulation around a certain frequency

- Near the coincidence frequency, the transmission loss of a wall is reduced. At 1000Hz, a 1.5mm thick steel plate has better insulation than a 10mm thick plate.



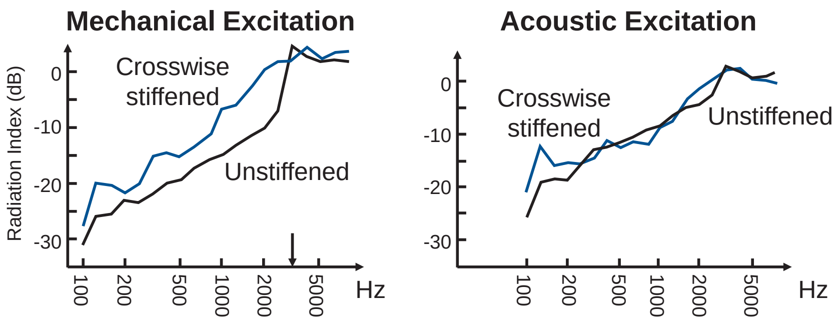


Acoustic behaviour of a panel

- Reducing the stiffness of a panel lowers it's resonance frequency and raises it's critical frequency, basically increasing the region for which the mass law applies.
- Increasing panel mass also lowers resonance frequencies and raises the critical frequency.
- Decreasing panel thickness raises the critical frequency but generally reduces panel mass.
- Increasing the amount of damping applied to the panel will not alter the frequencies of resonance and coincidence but will act to reduce their effect.
- Good insulation is therefore a combination of low stiffness, high mass and high damping (given cost constraints).

Stiffening a panel can INCREASE the noise level

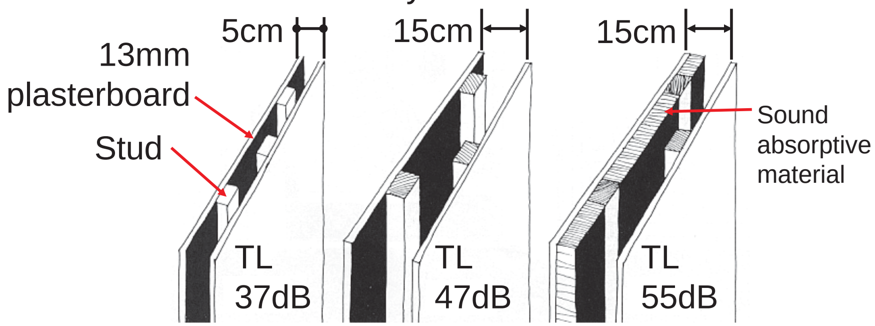
- A stiffened panel that is excited mechanically radiates more noise than an unstiffened panel.



See Fahy, Sound and Structural Vibration, Fig 47, p89

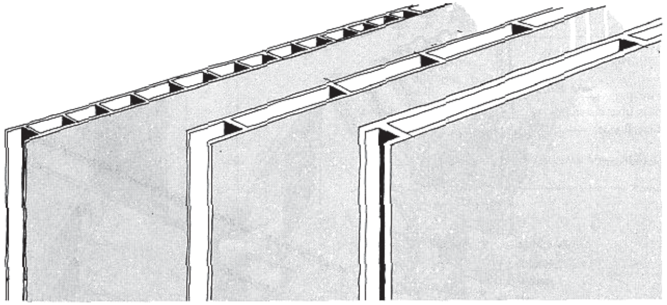
Light double walls provide good sound insulation

- Two lightweight walls separated by an air gap provide good transmission loss. The TL increases as the spacing increases.
- Double walls can provide the same TL as a single wall that is 5x to 10x as heavy.



Double walls should have few connections

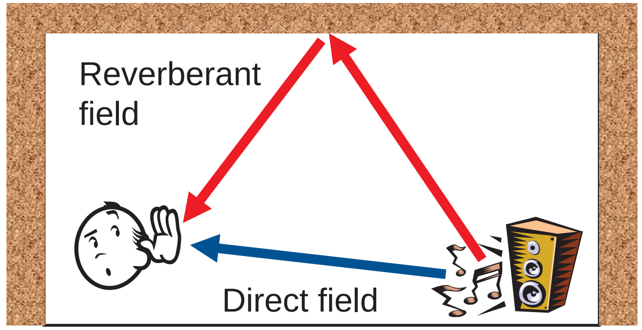
- A double wall provides the best TL if each wall is heavy and has few connections between them.



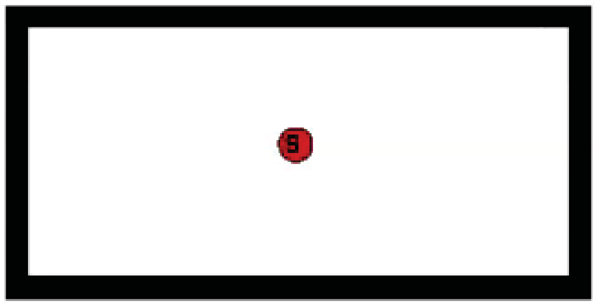
Sound absorption only reduces noise from indirect paths

To be covered in Topic 10 Indoor Sound

- The noise at a receiver in a room is due to the direct and reverberant field.
- Sound absorptive material will only affect the reverberant field!



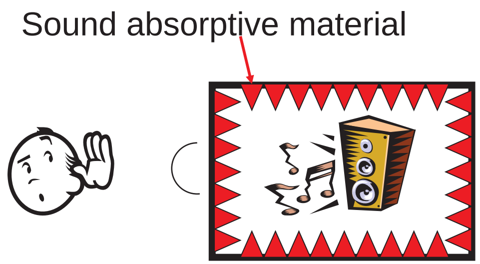
Sound Absorbed When Striking Walls



Sound absorption is used in enclosures to “soak” up sound

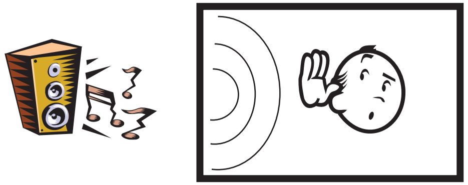
- When a noise source is enclosed, the sound level inside the cavity is higher than if the source was not enclosed.

- Absorptive material is used to “soak” up the sound level, to decrease the sound that impinges on the wall.

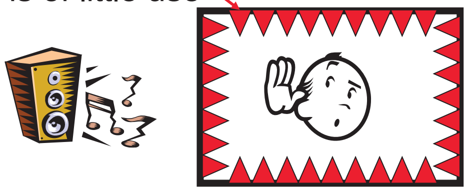


Common mistake

- When the receiver is isolated, sound absorptive material is of little use to reduce the noise.
- It is better to increase the transmission loss of the enclosure.



Sound absorptive material is of little use

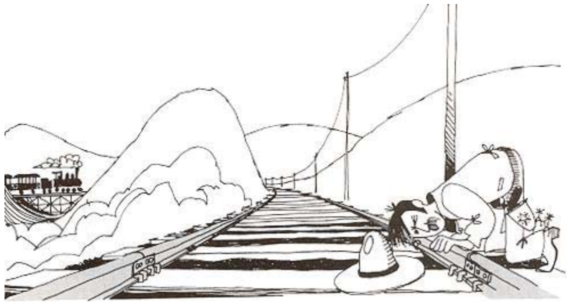


Common Mistakes

- Misunderstanding the difference between stiffness, damping and mass and how they change a panel's response.
- Stiffening panel thinking it will make noise quieter – can make noise worse.
- Applying noise absorption material thinking it will increase the transmission loss – can make no difference.

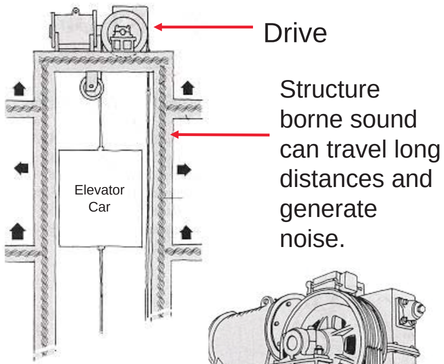
Structure borne sound can travel long distances

- Vibration in solids can travel long distances without being attenuated.
- This is a problem for buildings and ships.

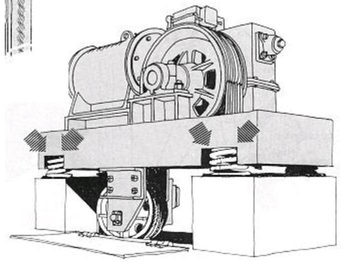


Application

- Problem: The drive for an elevator causes the building to vibrate and noise is generated on many floors below.



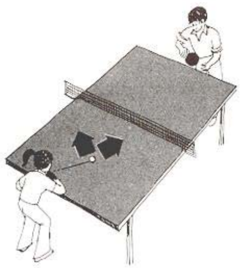
- Solution: The drive is vibration isolated from the building.



The rate of change determines the amount of high frequency noise

- Rapid change in force, pressure, speed, direction, the more dominant is the high frequency noise. A rapid change produces a short pulse, which is dominated by high frequency noise.

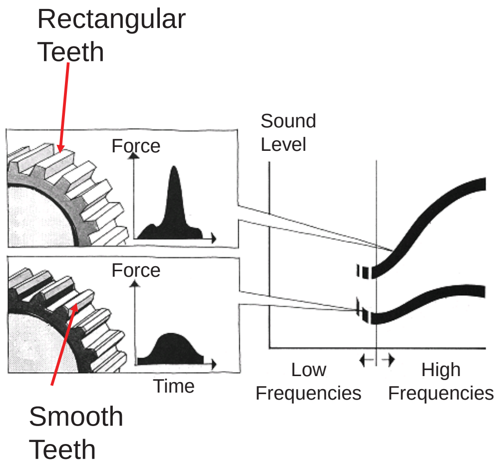
A ping pong ball is contact with the table for a relatively short time and generates high frequency noise.



A bouncing basketball is contact with the floor for a relatively long time and generates low frequency noise.

Application for Gear Teeth

- The force on the teeth rises and falls rapidly, for teeth with rectangular shape. High frequency noise is generated.
- Control Measure: A tooth shape that is smooth has a continuous force transfer and the overall noise is reduced.



Practical Application

- Helicopter gearboxes generate loud noise and are often attached to the fuselage.

- Changing the tooth profile can reduce the noise in the cabin.

Example: Tooth Profile for Escalator Drive

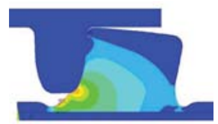
Before

AT20

- Profile caused high stress concentrations and high vibration levels

After

ContiTech

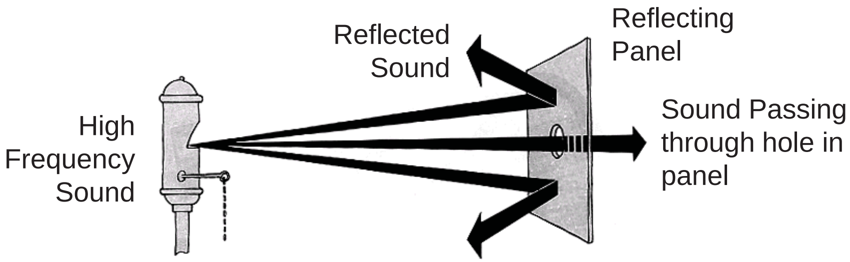


- Redesigned profile caused a gradual application of loading resulting in lower vibration levels.

High Frequency Sound is Directional and Easy to Reflect

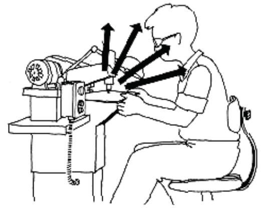
To be covered in Topic 8 Sound Sources

- When high frequency sound strikes a hard surface, it is reflected, but passes directly through any holes in the surface. High frequency sound does not bend around corners.

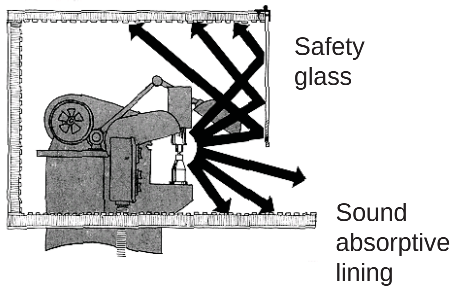


Application

- High frequency noise travels directly from the high speed riveting machine to the worker's ears.

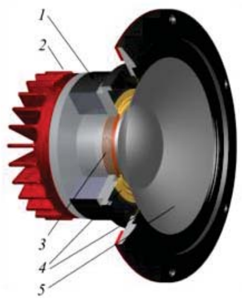


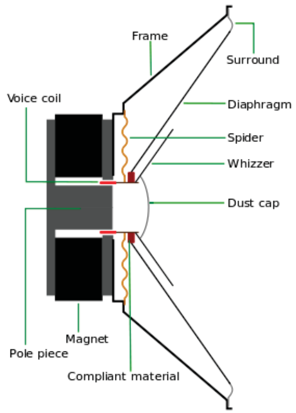
- The machine is enclosed with an opening facing the operator. The inside surfaces are lined with sound absorptive material. A safety shield is installed that reflects sound back into the enclosure.

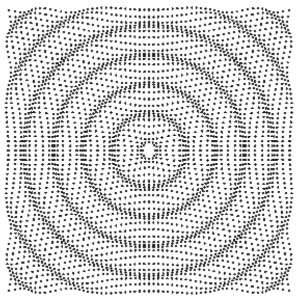


Demonstration of HyperSonic Sound

Typical Loudspeaker





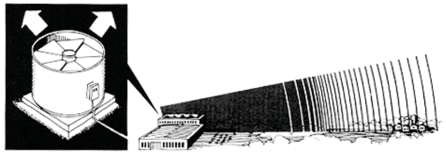


http://www.acs.psu.edu/drussell/demos/rad2/mdq.html

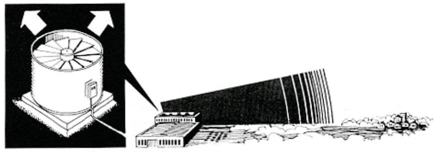
Higher frequency sound is reduced by passing through air

To be covered in Topic 11 Outdoor Sound

- The low frequency noise from rooftop fans disturbs residential homes 400m away.



- Control Measure: Number of fan blades increased.
  No low frequency noise, and high frequency noise is attenuated with distance.



Practical Application

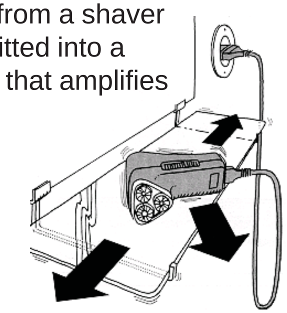
- Roof top air-conditioners.

Global Chiller

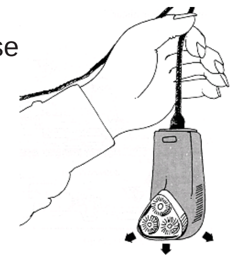
Vibrating plates generate noise

- The noise from a vibrating machine is amplified when it is in contact with objects that have large surface area.

Vibrations from a shaver are transmitted into a glass plate that amplifies the noise.



Vibrations are not transmitted to the plate and the noise is lower.



Demonstration Vibrating Plate Generates Noise

Vibrating structures generate sound

Demonstration

- Distributed Mode Loudspeakers

Demonstration Throat Microphone

Normal Mic vs Throat Mic

A counter example of how detecting vibration can be useful

- Normal Mic

- Throat Mic

Bone Conduction Headphones

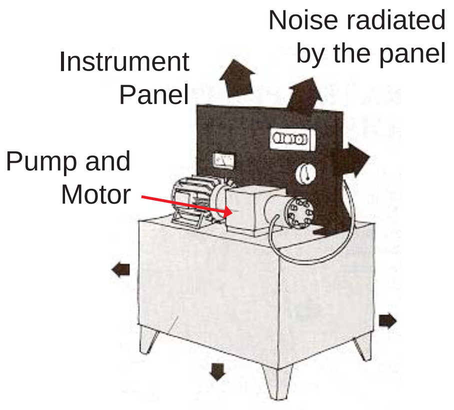
A counter example of how vibrations can be useful

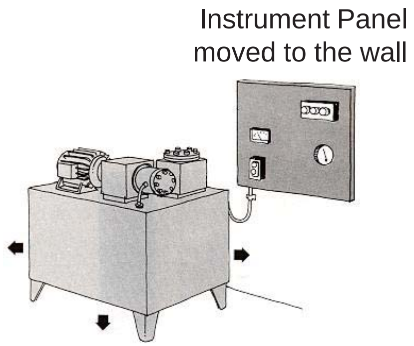
- Sound transmitted as vibration into skull so that you can hear sound.

Applications

Problem: A noisy hydraulic system. Most of the noise comes from the vibrating instrument panel

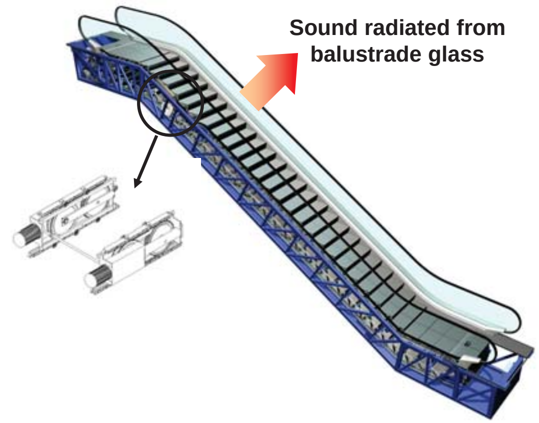
Solution: break the transmission path and the noise level decreases.





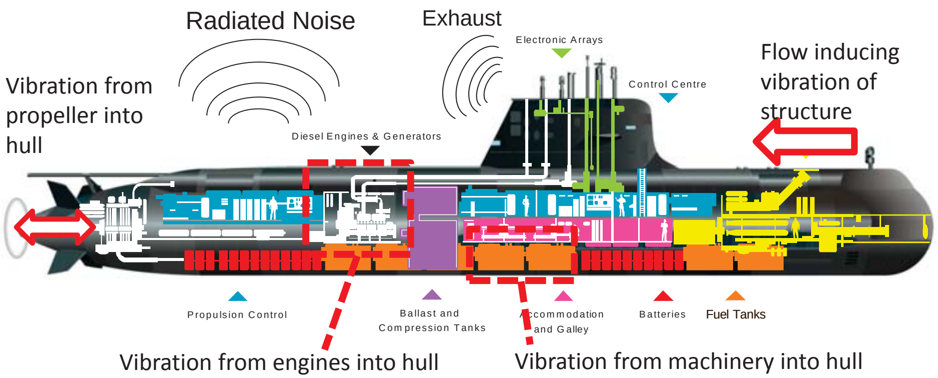
Practical Application

- Escalator has a drive module that couples with the balustrade glass.
- Break the transmission path by using vibration isolators.



Practical Application

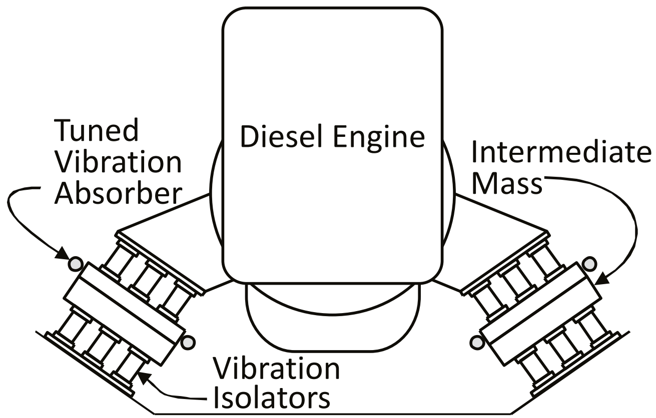
- Vibration from machinery on a submarine must be vibration isolated from the hull.



From: Keith Wood, presentation to AMC, www20100528.amc.edu.au/system/files/asc.pdf

Practical Application

- Install vibration isolators between engine and hull.

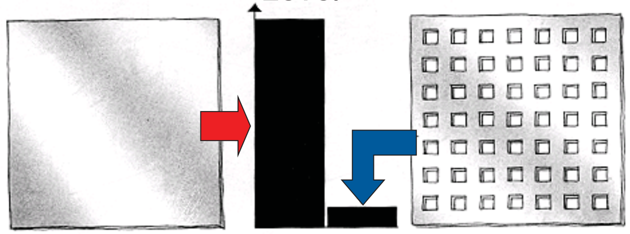


Common Mistakes

- Attaching piping or ducting to the walls of enclosures.
- Attaching instrument panels to vibrating parts.
- Once vibration gets into a structure, the energy can travel and re-radiate from panels.

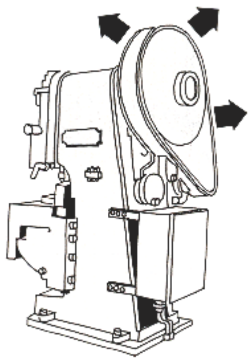
Densely perforated plates produce less noise

- A vibrating large solid plate can generate noise. An alternative is to use a perforated metal plate or mesh.

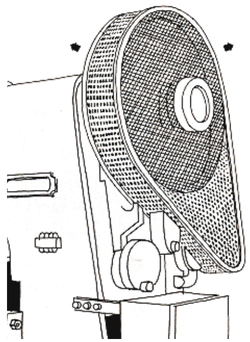


Application

Problem: The protective cover over a flywheel and belt drive radiates noise.

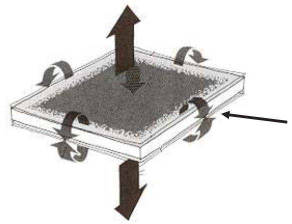


Solution: Replace the solid cover with perforated sheet metal.



Long narrow plate radiates less noise than a square one

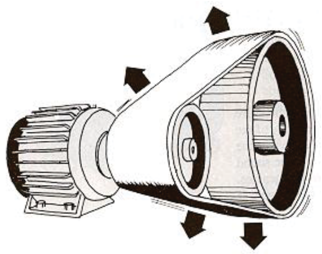
- When a plate vibrates sound comes from both sides. At the edges the pressure difference balances so radiated noise is low.



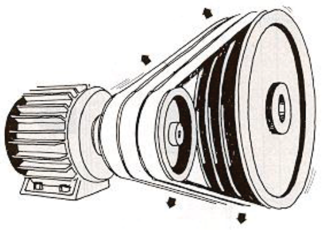
- A long narrow plate radiates less noise than a square plate of the same area.

Applications

- A belt creates low frequency noise because of the broad area.



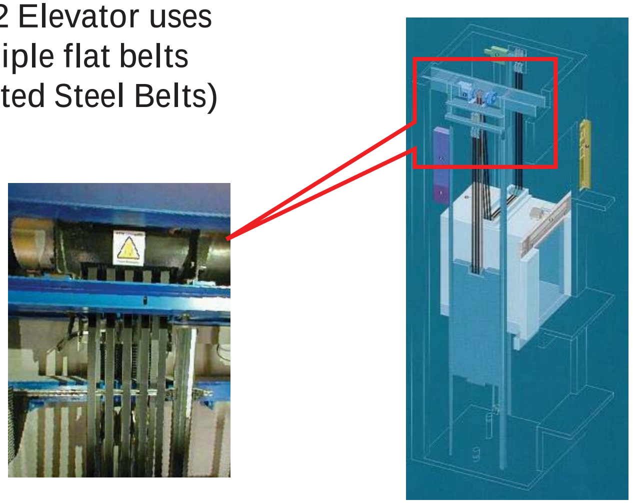
- Control Measure is to replace the belt with narrow separated belts, which reduces the radiated noise.



Practical Example

See video on new Engineering South Lift

- Gen2 Elevator uses multiple flat belts (Coated Steel Belts)



Pictures of Lift Motor in Eng South

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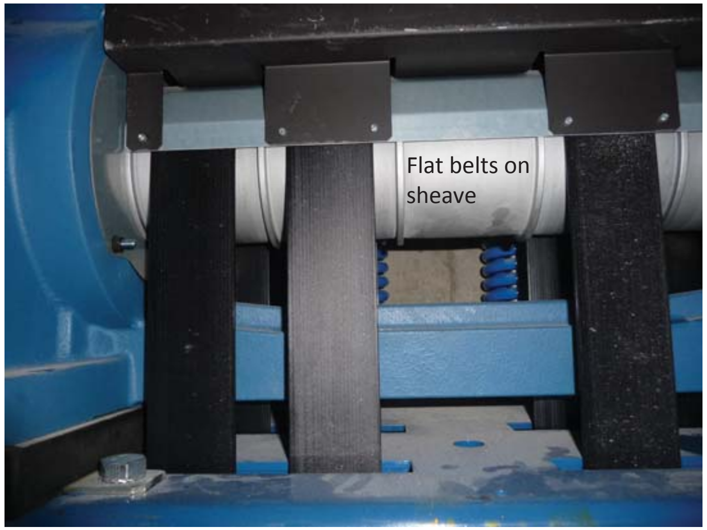

Pictures of Lift Motor in Eng South

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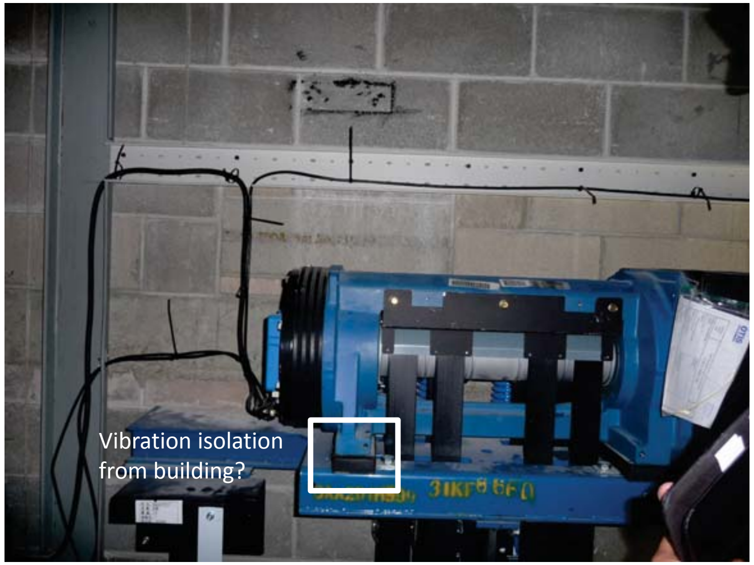
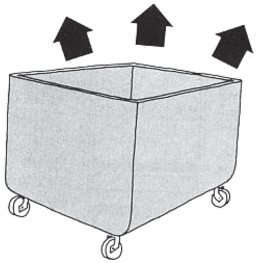
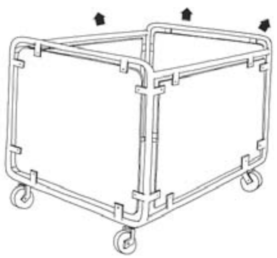


Application

- When parts strike the cart, loud noise is radiated.

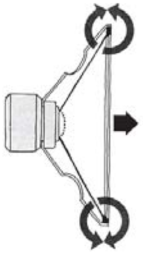


- Control Measure: Replace the walls with open panels. Pressure equalization occurs on all edges and reduces the radiated noise.

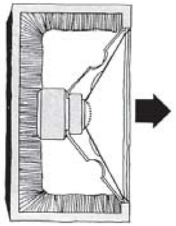


Plates with free edges radiate less low frequency noise

- Pressure equalization occur at the edges of plates.
  - Enclosing the edges prevents pressure equalization.

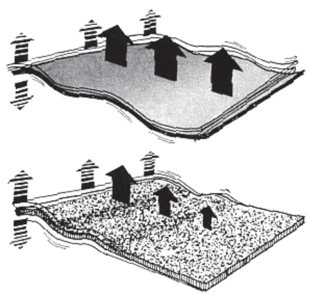


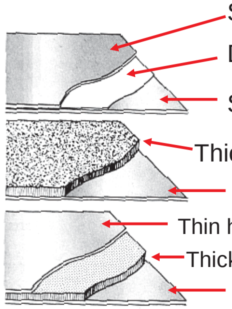
- Loudspeakers produce more bass if they are enclosed in a cabinet.



A Damped Surface Radiates Less Sound

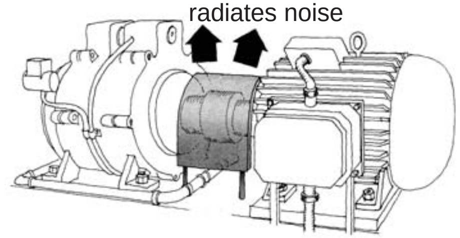
- Vibration decreases as it moves across a plate. The amount it decreases depends on the internal damping. Steel plate has extremely poor damping. Damping can be improved by adding coatings or intermediate layers.



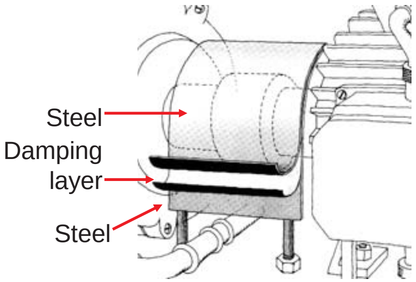


Application

Problem: the loudest noise from a pump and motor comes from the coupling guard made of sheet metal



Solution: replace the guard with a damped panel, or use perforated sheet metal.

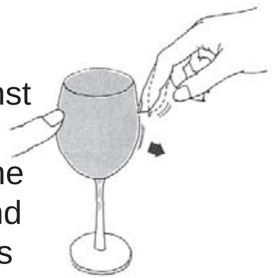


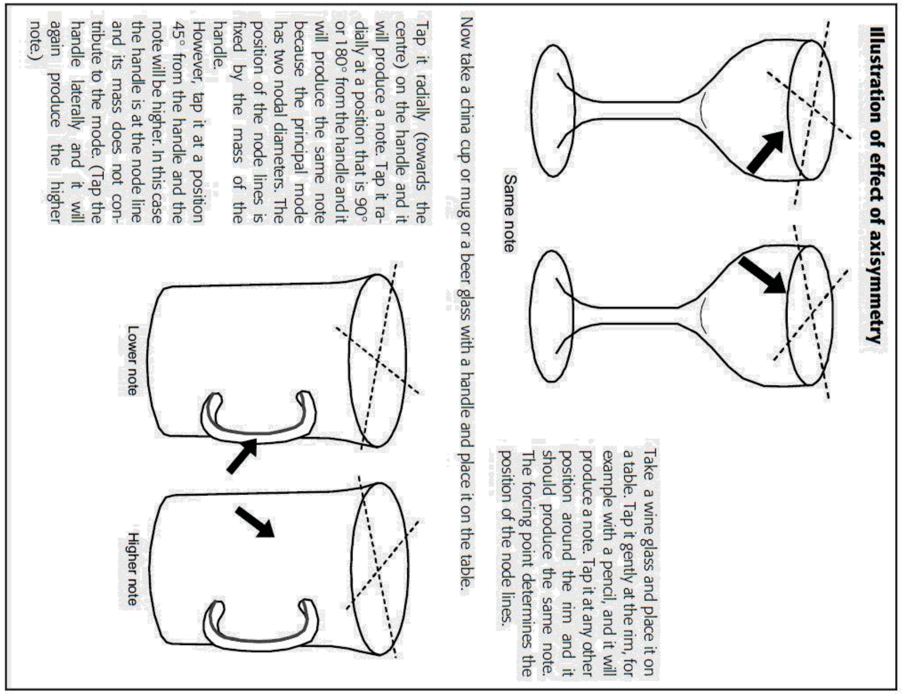
Resonance amplify noise, ... but it can be damped

- Resonance greatly increases noise from a vibrating plate, but it can be suppressed by damping the plate.

Tap against a glass and produces a loud ring

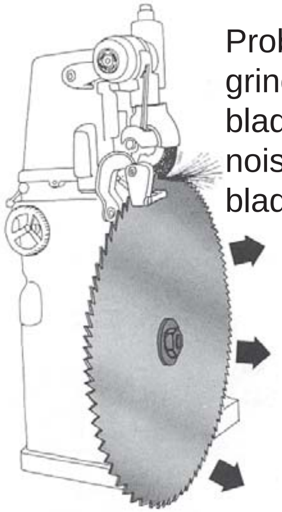
Placing a finger against the glass dampens the vibration and less noise is generated.



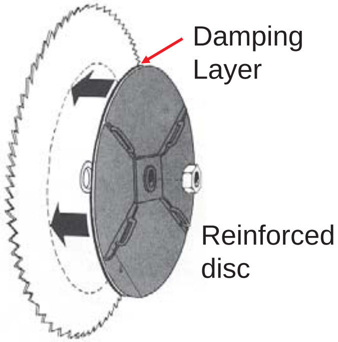


Application

Problem: A tooth grinder for saw blades radiates noise from the blade.

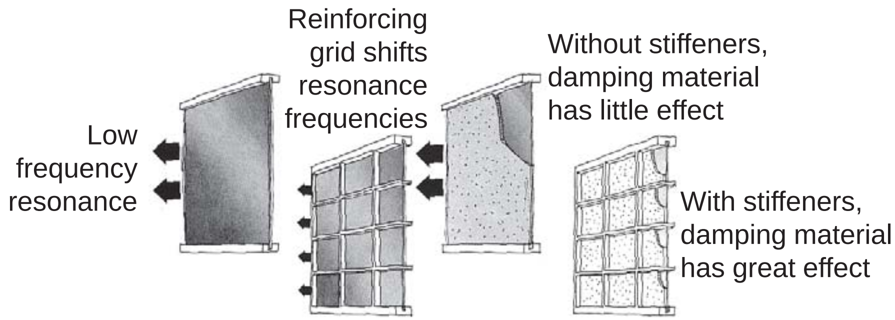


Solution: A rubber sheet is clamped to the blade to damp the vibration



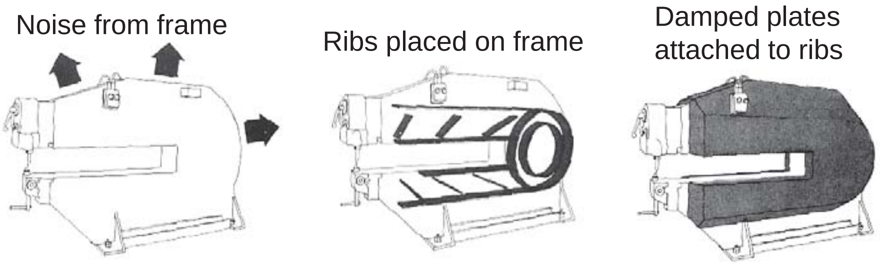
Resonance frequency shifted to higher frequency is more easily damped

- Large vibrating plates have low frequency resonances that are hard to damp.
- If the plate can be stiffened the resonances are shifted higher in frequency which is easier to damp.



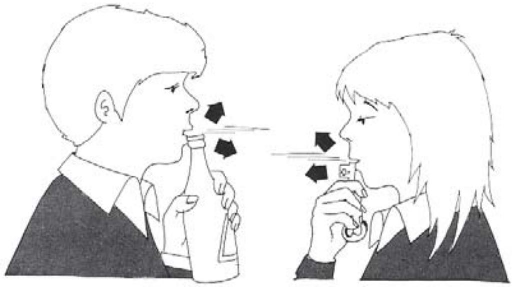
Application

- Problem: A nibbler frame generates low frequency sound from the frame.
- Solution: The machine is enclosed with stiffened and damped panels.



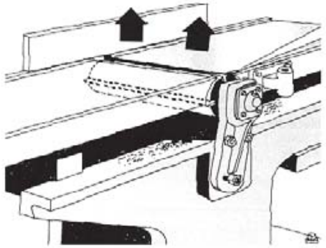
Air flow past hollow openings should be avoided

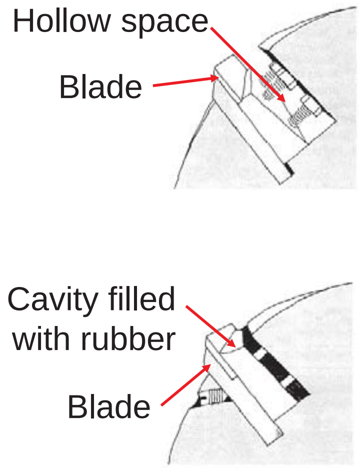
- When air flows past the edge of an opening to a cavity, loud tones are generated. This is how wind instruments work. The larger the volume of the cavity, the lower the frequency will be.



Application

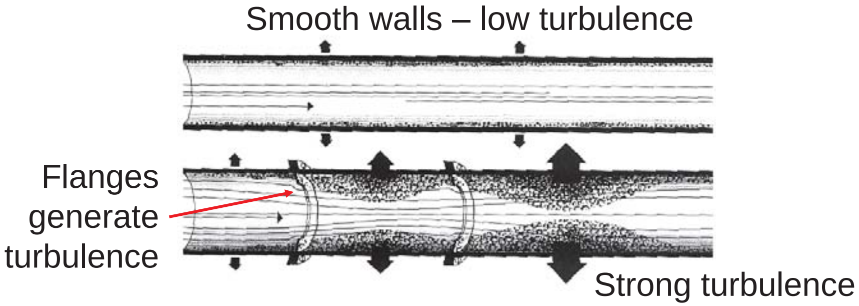
- Problem: A cutter wheel generates loud tones as the blade rotates. The noise may be amplified by the cavity resonance.





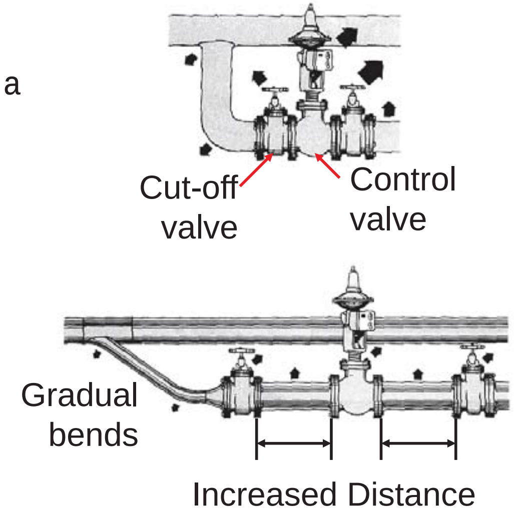
Ducts without obstructions produce the least amount of noise from turbulence

- When liquid or gas flows in a pipe, there is always some turbulence along the walls. The noise from turbulence increases if the flow abruptly changes in direction, speed, or if objects interfere with the flow.



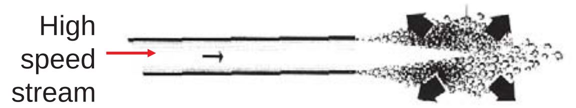
Application

- Problem: A branch in a steam line has three valves that produce a loud shrieking sound.
- Solution: Piping replaced with gradual bends and distance increased between valves.

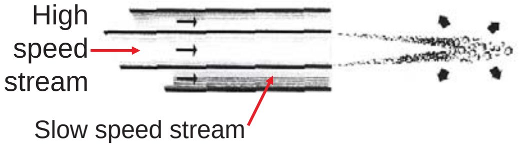


Jet noise can be reduced by using an extra air stream

- High speed air (>100m/s) generates significant turbulence and noise at the exit.

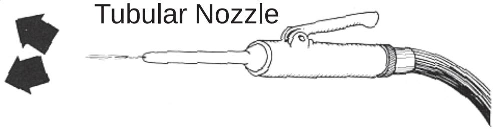


- Reducing the exit speed by one half may decrease the noise level by about 20dB. The noise level is determined by the speed of the jet relative to the surrounding air.

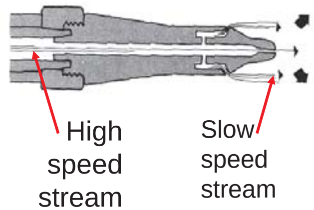


Application

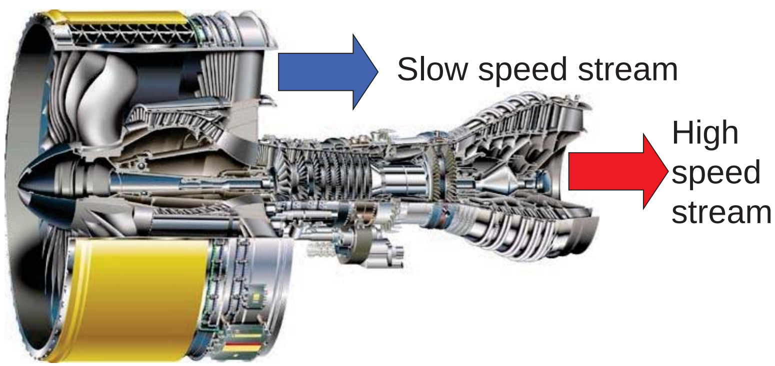
- Problem: A nozzle for compressed air generates loud noise.



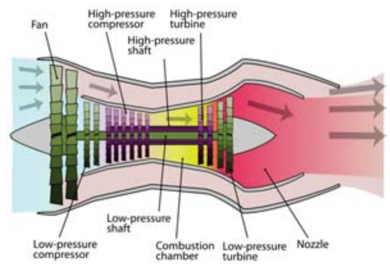
- Solution: The simple tube is replaced by a dual flow nozzle.



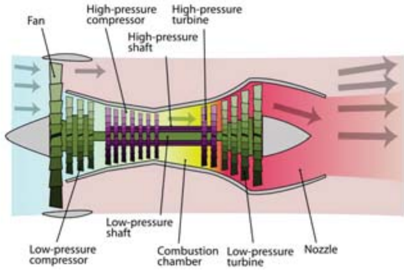
Application



Comparison of By-Pass Ratios



- Low By-Pass [lots of noise]
- Examples:
  - Pratt & Whitney JT8D
  - DC-9, MD-80, 727, 737
  - By-Pass Ratio 0.96:1

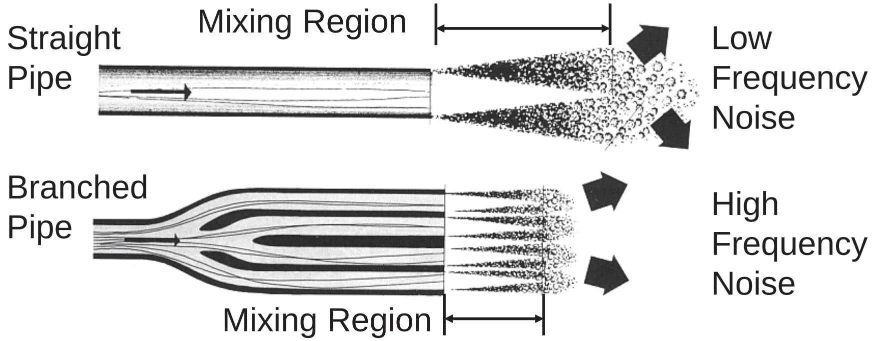


- High By-Pass [lower noise]
- Examples:
  - Rolls-Royce Trent 1000
  - 787
  - By-Pass Ratio 10:1

http://en.wikipedia.org/wiki/Bypass_ratio

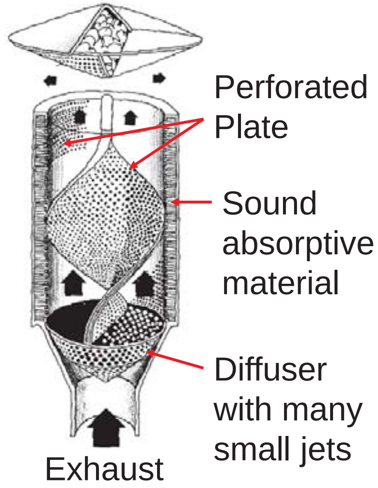
Low frequency jet noise is easier to reduce if converted to high frequency

- If the diameter of the pipe is large, the noise will be loudest at low frequencies.
- If the diameter is small, the noise will be loudest at high frequencies.



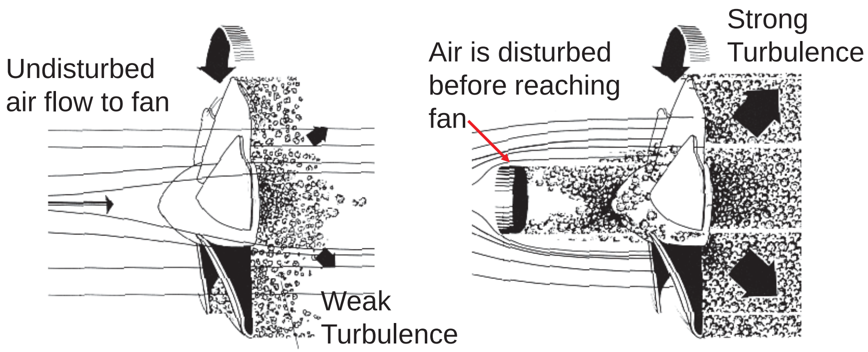
Application

- Problem: Steam safety valve produces high noise levels.
- Solution: A diffuser with perforated cone and high frequency muffler allows high flow rate. Jet speed is reduced by a factor of 4 and total noise is reduced by about 20dB.

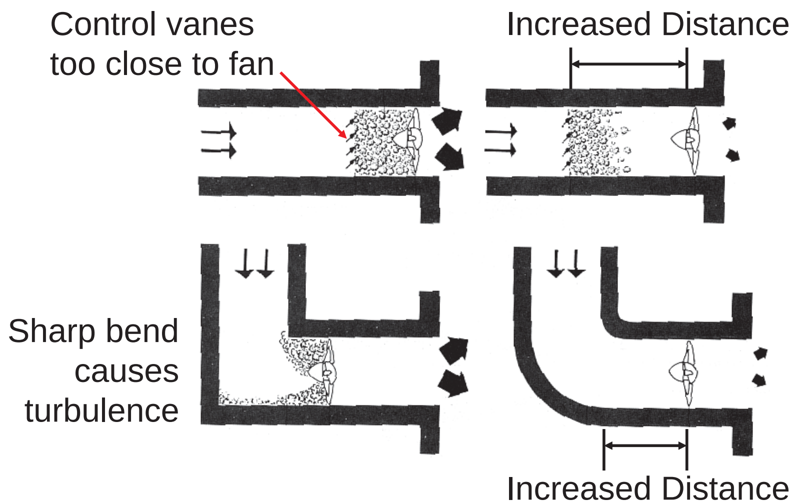


Fans make less noise if placed in undisturbed flow

- If the fan inlet has turbulent air, the sound will be more intense than if the flow is undisturbed.

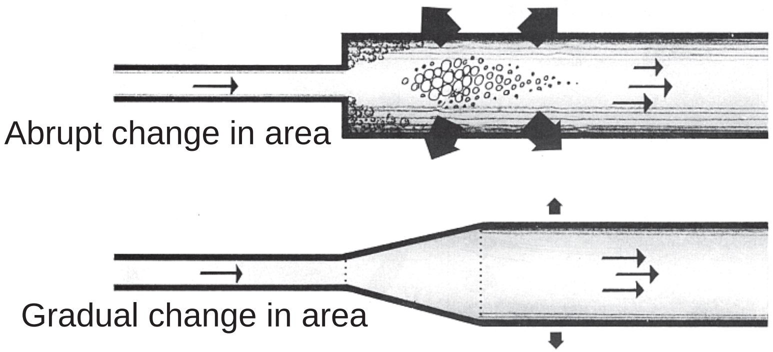


Application



Abrupt changes in area produce noise

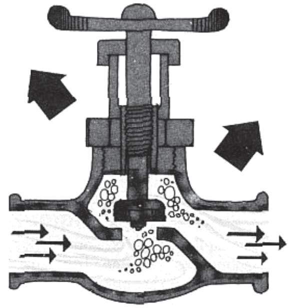
- Turbulence and noise will be created if the cross-sectional area increases abruptly. A gradual change in area by using a gradual change in area.

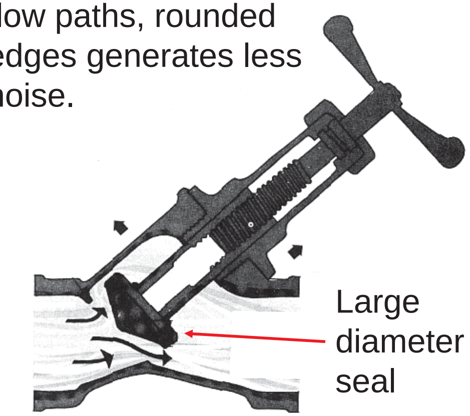


Application

Problem: Control valves in liquid systems often have small valve seats with sharp edges, which results in turbulence and noise.

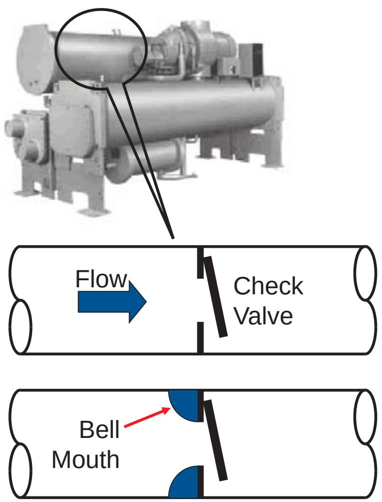
Solution: Control valve with larger cone diameter and straighter flow paths, rounded edges generates less noise.





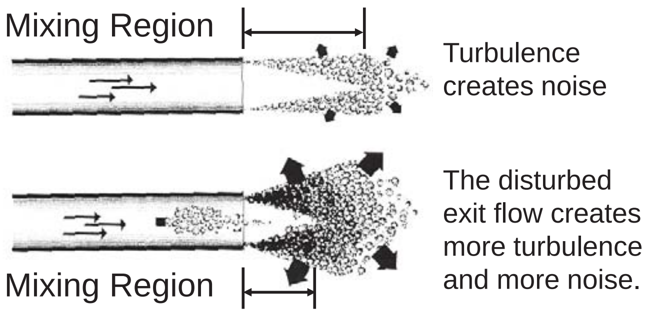
Practical Application

- The original design of a check valve for a chiller had sharp edges that generated noise.
- A bell mouth was installed on the front of the valve to smooth the flow.



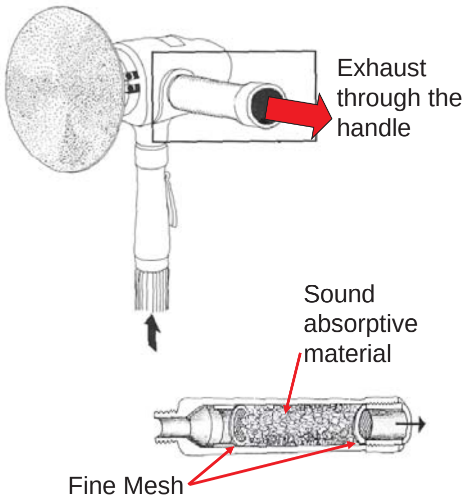
Undisturbed flow produces the least amount of exit noise

- Objects in the flow generate turbulence and noise. A smooth undisturbed flow generates the least amount of noise.



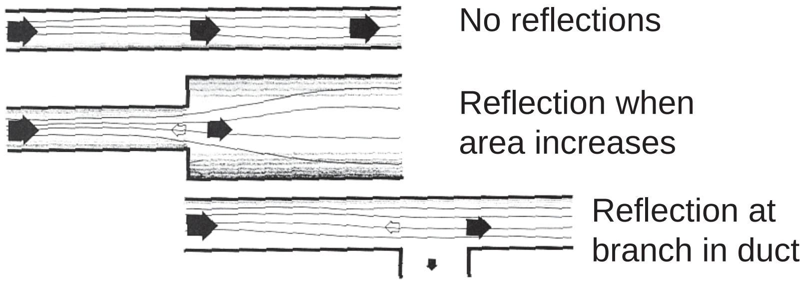
Application for venting compressed air grinder

- Problem: The exhaust from a grinder generates a loud noise.
- Solution: A new handle is designed that has sound absorptive material and fine mesh, that acts to reduce the turbulence, and hence the noise level.



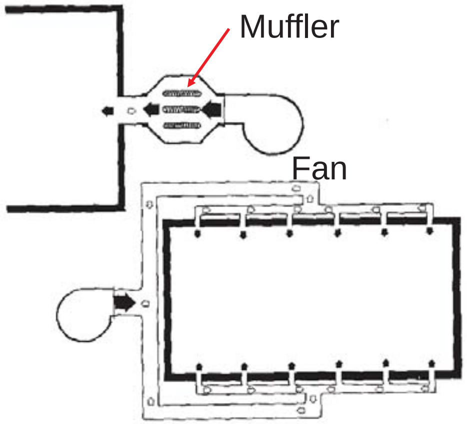
All duct changes reduce sound transmission

- Every change in the pathway causes some sound to be reflected back. A muffler that reflects sound energy back to the source is called a reactive muffler.One that converts sound into heat is a dissipative muffler.



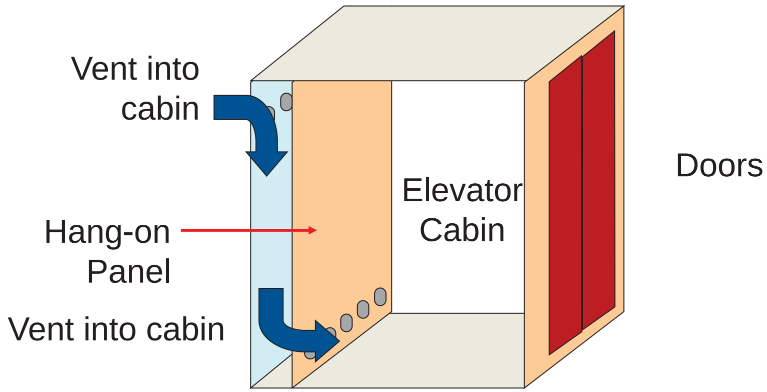
Application

- Problem: Insufficient space for fan and muffler in a ventilation system.
- Solution: Several smaller vents are installed. Sound is reflected at each bend and branch.



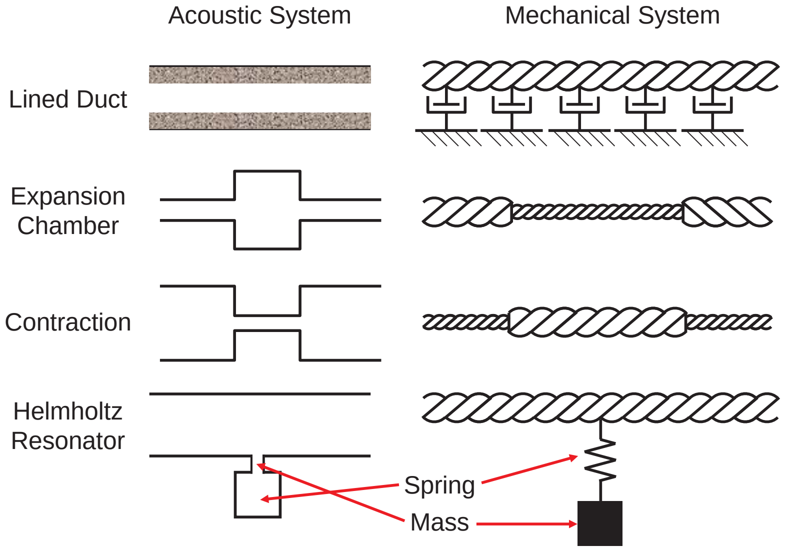
Practical Application

- Convoluted path for the ventilation in an elevator cabin.



How Silencers Work

To Be Covered in Topic 14

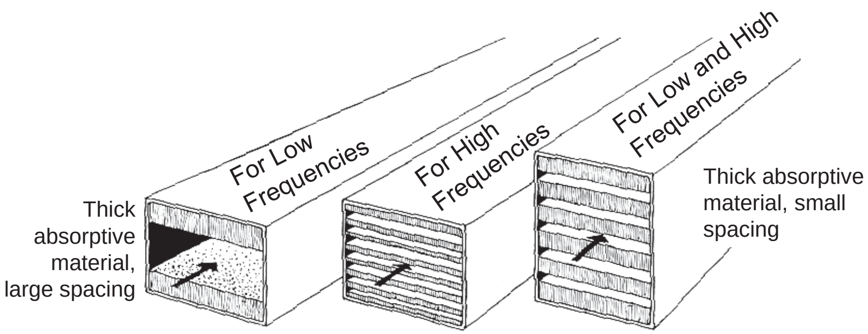


Practical Applications

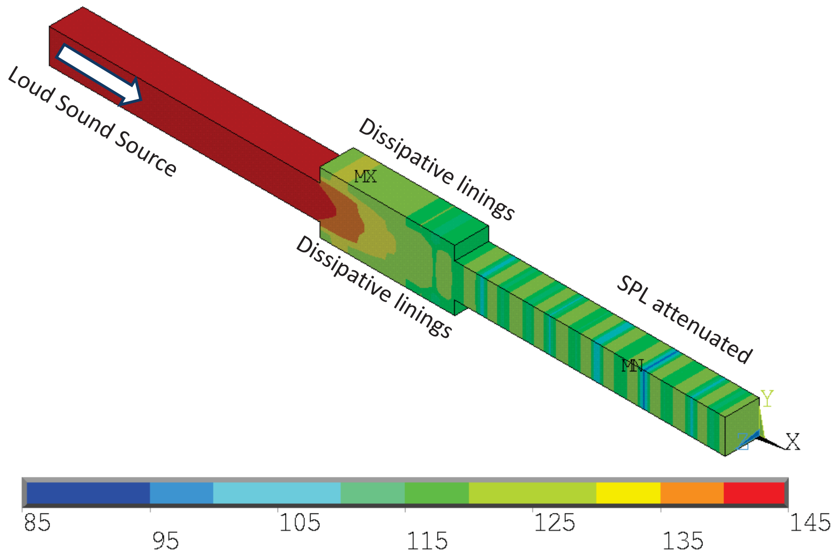
- Reactive Silencer for Power Station

Dissipative mufflers are effective over a broad range of frequencies

- The thickness and spacing of absorptive material effects the frequencies that are absorbed. The thicker the material, the lower the frequency that is absorbed. Smaller spacing is used for higher frequencies.



Example: Absorptive Silencer

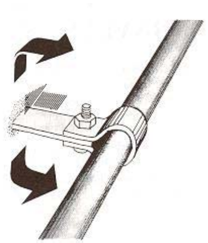


Application

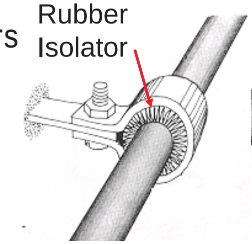
- Equipment room West of The Hub

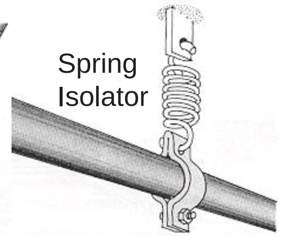
Application

- The pressure pulsation from piping causes the walls to vibrate. If the pipes are attached to a large surface area, it will radiate noise.



- Solution:
  Use flexible connectors between the pipe support and the wall





Demonstration - Shaver in a Box

Demonstration - Shaver in a Box

Points to note from demo …

- Leakage can kill the benefit of an acoustic enclosure.
- Vibration isolation of machinery is very important.
- A vibrating machine touching the enclosure walls can generate a lot of noise.
- Sound absorptive material acts to “soak up” sound inside enclosure.
- Sound absorptive material by itself has very little benefit for improving transmission loss.

The TAKE AWAY

If you only remember one thing from this lecture....

- If noise or vibration will be important, you must ***PLAN*** for it at the very start of the project.

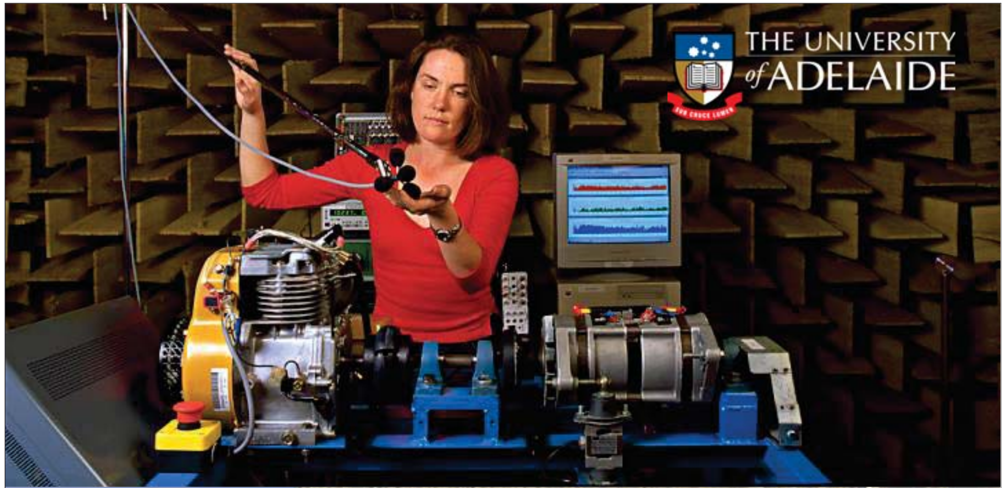
Sorry, there is no noise control fairy.

- Noise and vibration is a ***SYSTEM*** level problem.

- Get help from acoustic and vibration consultants.

Summary

- Basics of Acoustics
- Vibro-acoustic Noise Control
- Air-borne Noise Control
- Liquid-borne Noise Control
- Building Acoustics
- Demonstrations:
  - Audio Spotlight
  - Vibrating magnet attached to object
  - Shaver in a box
  - Noise reduction of a quarter-wave tube attached to a large engine.



CRICOS PROVIDER 00123M

Carl Howard

Instrumentation

adelaide.edu.au seek LIGHT

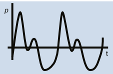
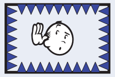
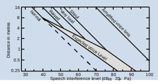
Instrumentation

We need to be able to measure and quantify sound.

This lecture covers:

- What types of instruments are available and applicable for various situations.

Course Content – How it fits

TOPIC	
Basics	
General Noise Control	
Instrumentation	
The Ear	
Criteria	

Contents

- Microphones: Condenser, Piezoelectric
- Sound Level Meters
- Spectrum analysers
- Statistical analysers
- Noise dose meters
- Intensity meters
- Instrumentation for sound source localisation

Learning Outcomes

Be able to:

- Recognise various instruments that are available for conducting acoustic measurements.
- Select appropriate instrumentation for measuring sound.
- Have an understanding of practical measurement procedures.

Microphones

Instrument grade

Usually around $1500

Usually around $4500

Consumer grade

$1

Microphones

Microphones

Microphone types.

- condenser
  - externally polarised
    - some mics need to have a 200V polarisation voltage applied
  - prepolarised
- piezoelectric

Microphones

- condensers or electrets used for precision measurements
- sensitivity

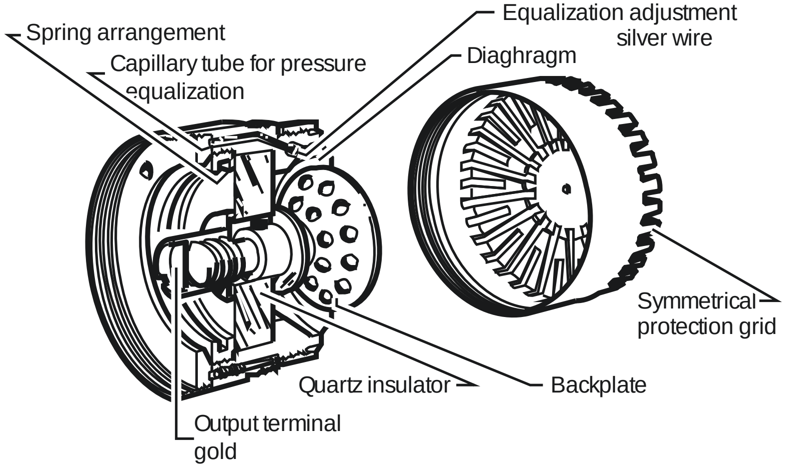
$$S = 20\log_{10} E - L_p + 94 \quad \text{dB re 1V/Pa}$$

where E is the voltage generated for a sound pressure level of Lp.

S = sensitivity of the microphone in dB re 1V/Pa
E = the voltage that is produced in V
L_p = incident sound pressure level in dB re 20 micro-Pa

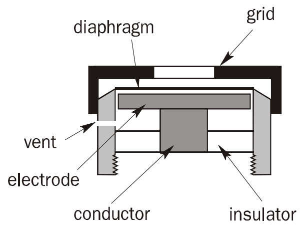
ENC p100

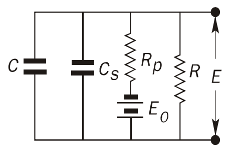
Microphones - Condenser



Condenser Microphone Cartridges — Types 4133 to 4181, Bruel and Kjaer

Microphones - Condenser





- Output voltage $$E \approx \frac{K_1 Q x}{1 + C_s / C} \quad \frac{K_1 Q x^2}{h(1 + C_s/C)^2} \approx \frac{K_1 Q x}{1 + C_s / C}$$
- K_1 is geometry dependent and determined by calibration
- C_s/C and x/h made small to minimise non-linear terms
- $x = -k_3 Ap$, where A is the diaphragm area
- k_3 is the diaphragm stiffness

ENC p97

Microphones - Condenser

- Vent provides mean pressure equalisation across microphone diaphragm
- Size of vent controls mic. low frequency response limit

$$f_{co} = \frac{1}{2\pi R_v C_{dc}}$$

where
$$\frac{1}{C_{dc}} = \frac{1}{C_c} + \frac{1}{C_d} = \frac{C_d + C_c}{C_d C_c}$$

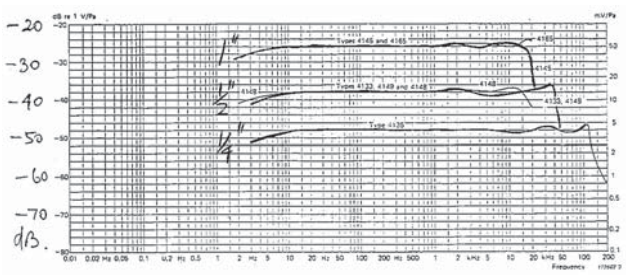
$$R_v = \frac{8 \mu \ell}{\pi a^4}$$

$$C_c = \frac{V_c}{\gamma P}$$

ENC p97

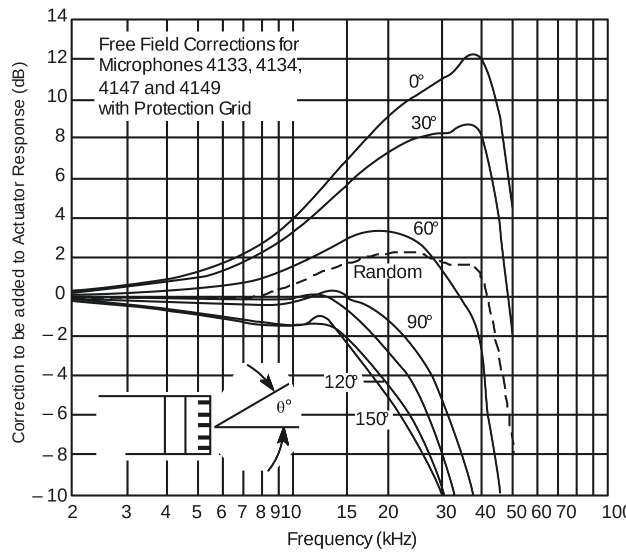
Microphones - Condenser

- vent is a tube of radius, a, and length, l
- γ = 1.4 = ratio of specific heats for air
- P is the mean atmospheric pressure (Pa).
- typical value for C_c is 2.8×10^{-12}
- typical value for the diaphragm compliance, C_d, is 0.3×10^{-12} m⁵/N
- μ is the dynamic viscosity for air = 1.84×10^{-5} N-s/m^5 at 20°C



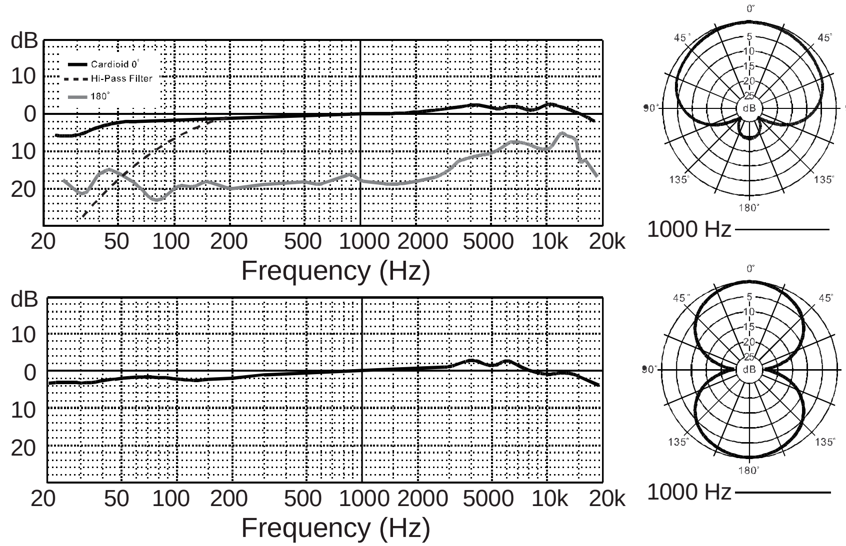
Microphones - Condenser

Response varies with angle of incidence



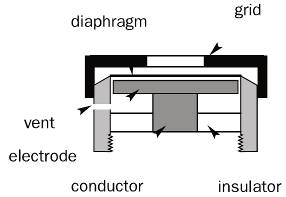
Condenser Microphone Cartridges — Types 4133 to 4181, Bruel and Kjaer

Frequency Response & Directivity

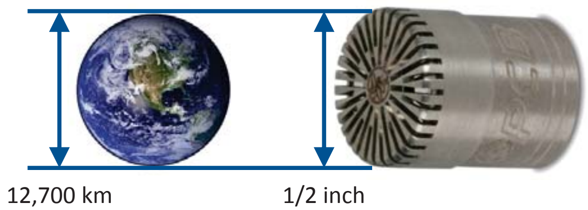


Samson CL8 Microphone Technical Specifications

Diaphragm Moves Tiny Amount



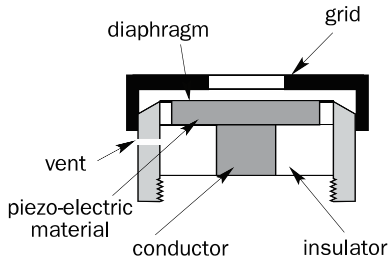
- 40dB Sound
  - About the level in a quiet living room
- ½ inch microphone diaphragm moves about 10^{-11}m
- To put this into perspective, if the diameter of a ½ inch mic were equivalent to diameter of Earth, then the diaphragm moves about ... HOW MUCH ?

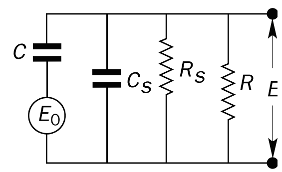


Microphone - Calibrators

- Microphone calibration
  - amplitude calibration:
  - use pistonphone (calibrates SLM too)
  - frequency response calibration

Microphone - Piezoelectric





- Microphone output voltage

$$E = \frac{K_2 x}{1 + C_S / C}$$

ENC p99

Sound Level Meters

- used to measure sound pressure level
- 2 grades of sound level meter
  - type 1 - precision (± 0.5 dB)
  - type 2 - general purpose (± 1.0 dB)

Sound Level Meter

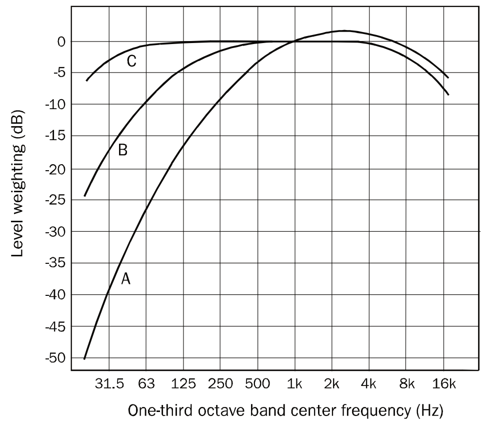
Possible sources of error:

- microphone vibration
- SLM vibration
- background noise
- overloading input amplifier while using filters
- temperature out of -40 to 150°C range
- moisture and dust on microphone diaphragm
- reflections from nearby surfaces
- wind noise due to wind blowing over microphone or nearby objects

-> Learning Outcome: practical use of equipment. Listen to discussion for details

Weighting Networks

- A-weighting most common
  - reflects ear response at low levels
  - discriminates against low frequencies which are not heard as easily



ENC p104

Quantifying Fluctuating Sound

- Most sound is unsteady in amplitude, duration, frequency.
- How is an average value calculated?
- L_{Aeq} , A-weighted equivalent continuous level

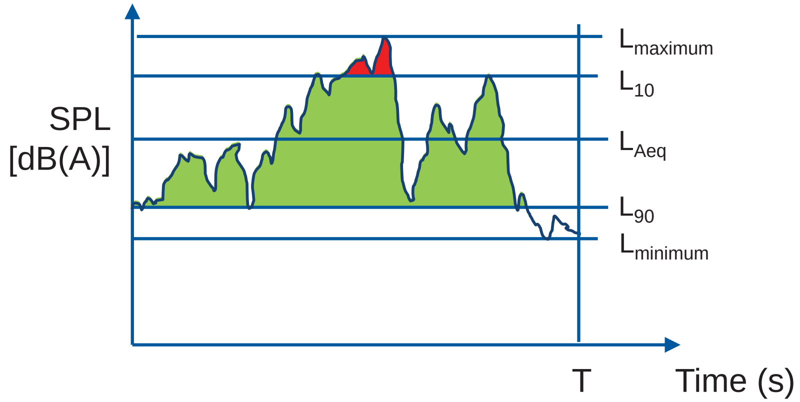
$$\boxed{L_{Aeq,T} = 10 \log_{10} \frac{1}{T} \int_0^T 10^{(L_{pA}(t)/10)}\, dt}$$

ENC p139

- L_{pA} = A-weighted sound pressure level
- L_x where ***x*** is the % of time the level exceeds $\boldsymbol{L_x}$
- L_{10} L_{90} are most common

Statistics

- L_{10} L_{90} are most common.
- Note $L_{10} > L_{90}$ for the same noise.



Example

- Adelaide City Council has noise provisions for music noise

- Music noise criteria:

  The noise level in any bedroom is to be no more than 8 dB above the level of background noise (L90,15min) in any octave band of the sound spectrum when exposed to music noise from any existing entertainment premises. Additionally, the music nose (LA10,15min) must be less than 5 dB(A) above the background noise (LA90,15min) for the overall (sum of all octave bands) A-weighted levels.

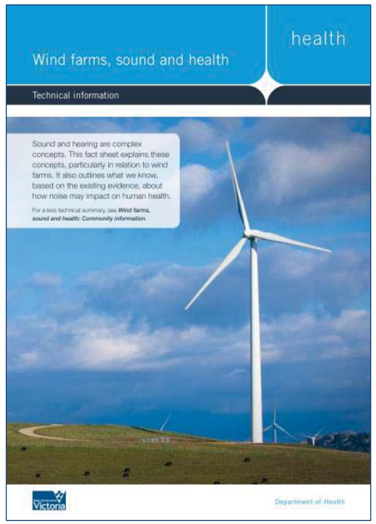
  https://www.cityofadelaide.com.au/assets/documents/noise_fact_sheet_12_frequently_asked_questions.pdf

Example

Another good document:

Wind farms, sound and health: Technical information

Victorian Department of Health



http://docs.health.vic.gov.au/docs/doc/5593AE74A5B486F2CA257B5E0014E33C/$FILE/Wind%20farms,%20sound%20and%20health%20-%20Technical%20information%20WEB.pdf

Example Animations

Have a look at the animated examples on this web site:

http://www.epd.gov.hk/epd/noise_education/web/ENG_EPD_HTML/m2/types_3.html

Example

EXAMPLE L_{Aeq} CALCULATION

- 90 dB(A) for 3 hours
- 94 dB(A) for 2 hours
- 100 dB(A) for 1 hour

What is $\boldsymbol{L_{Aeq}}$ (averaged over 6 hours)?

$$L_{Aeq} = 10\log_{10}\frac{1}{6}\left[3\times 10^{(90/10)} + 2\times 10^{(94/10)} + 1\times 10^{(100/10)}\right]$$
$$= 95.4\ \mathrm{dB(A)}$$

Example

- Calculation of overall A-weighted sound level from octave band linear levels :

Freq	31.5	63	125	250	500	1000	2000
Lin Level	95	95	90	85	80	81	75
A-wt	-39	-26	-16	-9	-3	0	+1
A-level	56	69	74	76	77	81	76

- Overall A-level:

$$= 10\log_{10}[10^{5.6} + 10^{6.9} + 10^{7.4} + 10^{7.6} + 10^{7.7} + 10^{8.1} + 10^{7.6}]$$
$$= 84.6\ \mathrm{dB(A)}$$

Other Types of Acoustic Instruments

- DATA LOGGERS
  - measure distribution of fluctuating noises over long time periods, generally over several weeks.

https://www.bksv.com/en/products/environment-management/noise-and-vibration-monitoring-terminals/noise-monitoring-terminal-family-3639

Other Types of Acoustic Instruments

- SPECTRUM ANALYSERS
  - provide a detailed description of average or instantaneous noise level vs frequency
  - useful for analysis of signals containing tones
  - frequency of tones can indicate sources of noise.




Other Types of Acoustic Instruments

- Personal Sound Exposure Meters (Noise Dosimeters)
  - measure cumulative noise dose to assess hearing damage risk
  - worn on the person - small and simple to operate

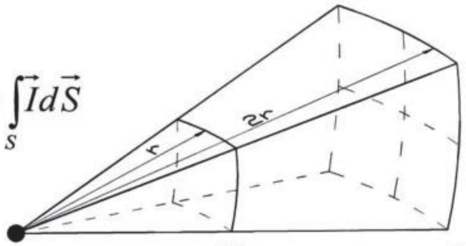


Instrumenting a person

Measurement of Sound Power

- Previous instruments were used to measure sound pressure levels.
- How can we measure sound power levels?
- Couple of ways...

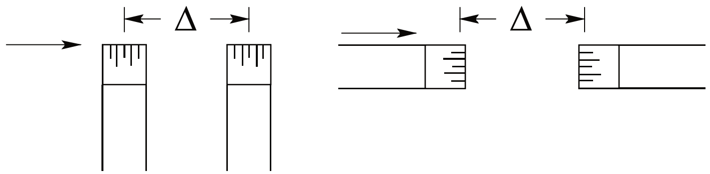
$$W = \int_S \vec{I} d\vec{S}$$



$$I_r = \frac{W}{4 \cdot \pi \cdot r^2} \qquad I_{2r} = \frac{W}{16 \cdot \pi \cdot r^2}$$

Measurement of Sound Power

- Sound Intensity Meter
  - may be special purpose
  - may be configured using software and a dual channel spectrum analyser
  - require measurement of the time integral of the product of the instantaneous acoustic
  - pressure and particle velocity
  - two microphones needed for p-p measurement
  - particle velocity related to the pressure gradient



Sound Intensity Probe

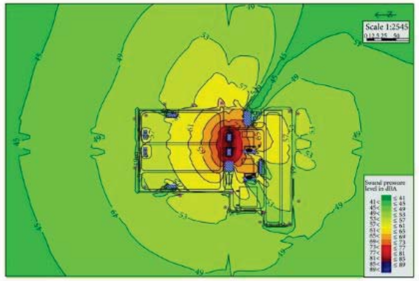
Sound Intensity Calibrator

Measurement of Sound Intensity

Measurement of Sound Intensity

Example: measuring sound power from an exhaust on a roof

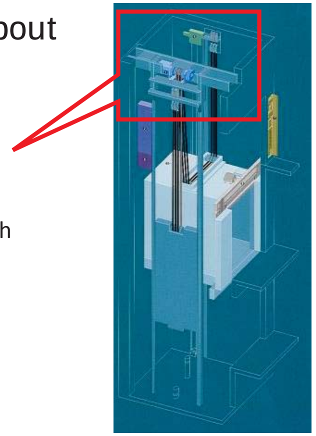
This technique can be used for to measure noise sources in a large industrial factory, and then feeds into a computational model for noise prediction (e.g. SoundPlan)



Reason for Measuring Sound Power

- It is the fundamental measure of how much noise a machine is making.
- Can then make future predictions about the expected sound level.

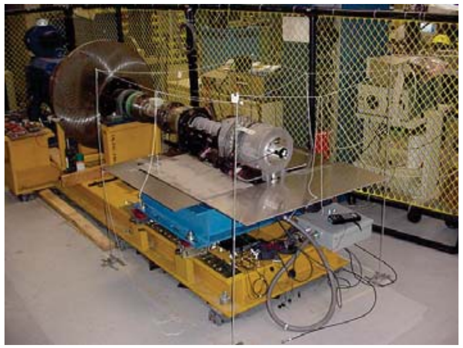
Electric hoist motors in elevator make noise. Elevator hoistway is concrete reverberant room, which is noisy. Thin doors on the floors can let sound into office floors that annoys building occupants.

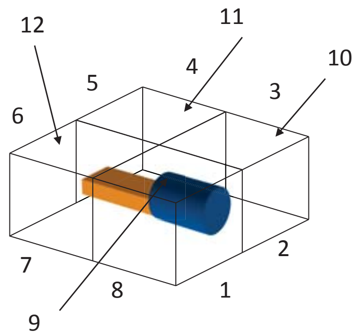


Pictures of Lift Motor in Eng South

Engineering South Lift

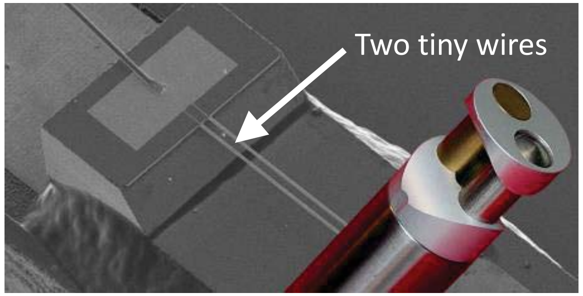
- Conducted evaluation of many configurations of motors for OTIS
  - Aluminum (sic) or Steel, thick or thin casings
  - Round or Square





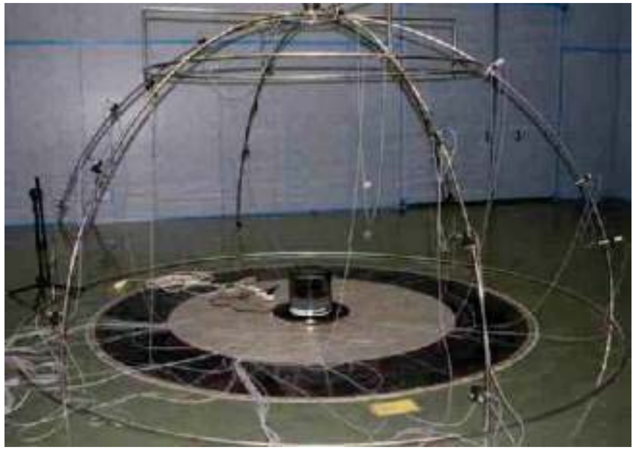
Microflown

- Measures acoustic particle velocity
- Only instrument that directly measures particle velocity



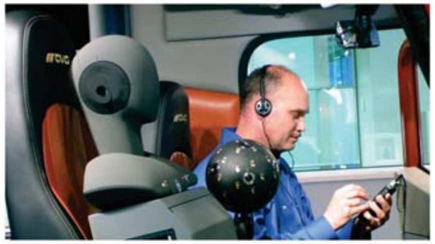
http://www.microflown.com/files/media/library/presentations/microflown-anewcategoryofsensors.pdf

Multiple microphones



Other Instruments

- Artificial Headforms



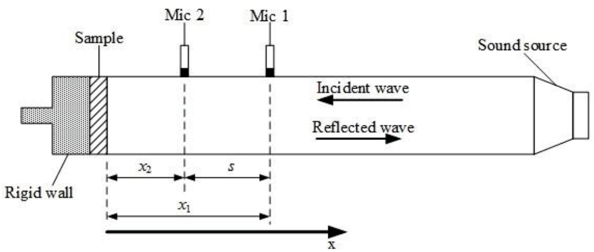
Other Instruments

- Binaural Microphones (for audio applications)

Impedance Tube

Used to measure **absorption coefficient** of materials.

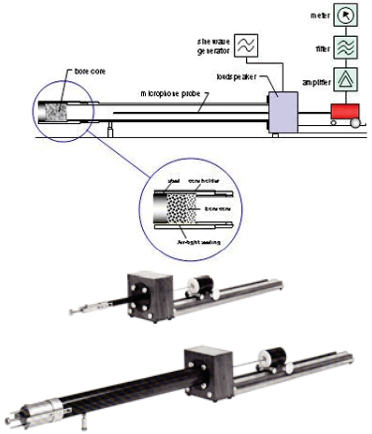
- Sample is placed at the end (or with a small air gap) of a closed tube.
- Loudspeaker generates white / pink noise.
- 2 or 4 microphones used to determine the incident and reflected sound power, and hence calculates the sound power absorbed by the sample, and the absorption coefficient.



Impedance Tube

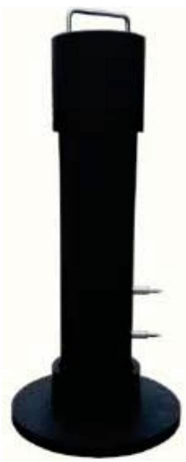
Typical modern impedance tube

Old school setup that uses measurement of amplitudes of standing waves.



Impedance Tube

- Impedance tube for measurement of absorption coefficient of pavements.
- Used for designing quiet road surfaces.



BSWA SW420R Impedance Tube System
For Pavements Absorption Testing - ISO13472-2

Microphone Arrays

- Used to determine location of sound sources

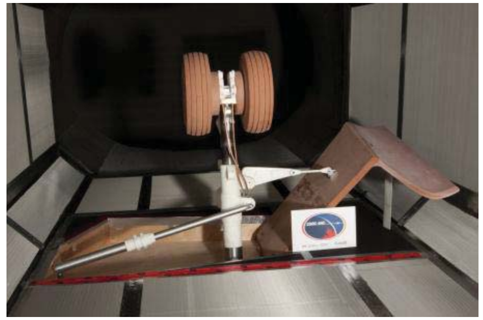
Ring Array

Spherical Array

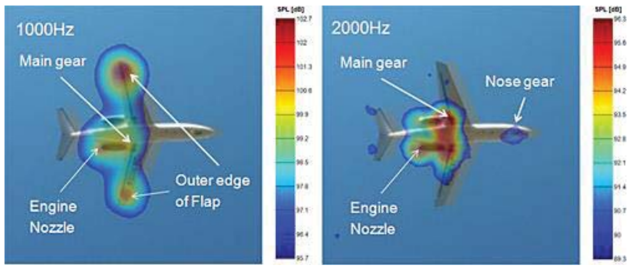
Star Array

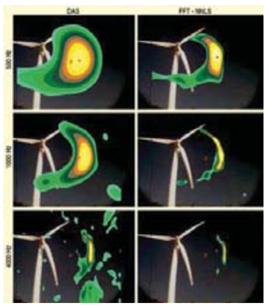
Statistical Acoustic Holography

Acoustic Arrays in Wind Tunnels



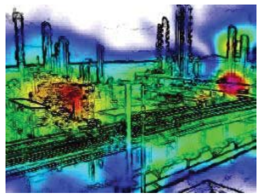
Example of Locating Sound Sources



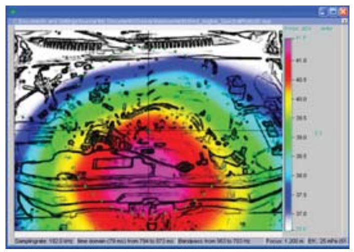


Acoustic Camera Examples

Industrial noise at a refinery



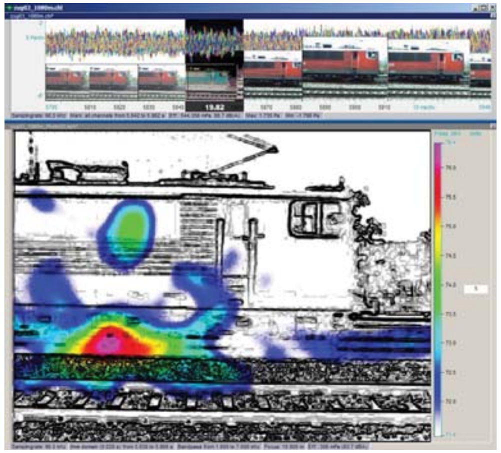
Engine development



Acoustic Camera images courtesy of gfaitech

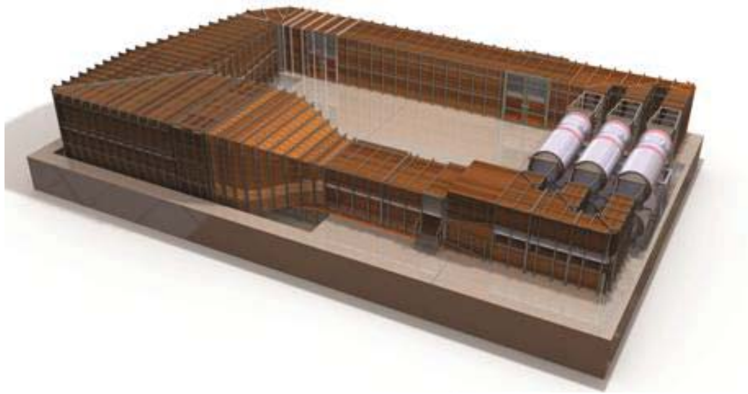
Acoustic Camera Examples

- Can locate sound sources on moving objects



Acoustic Camera images courtesy of gfaitech

Thebarton Wind Tunnel



Acoustic Holography

- Diagnosis of noise source from wind tunnel.

Figure 1: $V = 20\,\mathrm{m/s}$ at $f = 10992\,\mathrm{Hz}$, $A_{ref} = 6.5\,\mathrm{dB}$

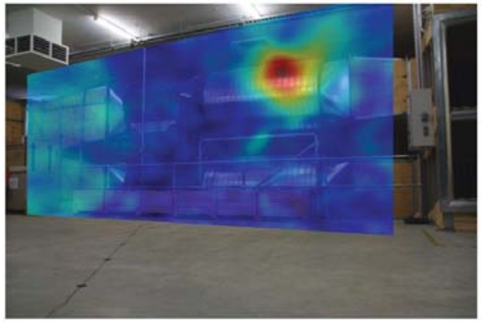
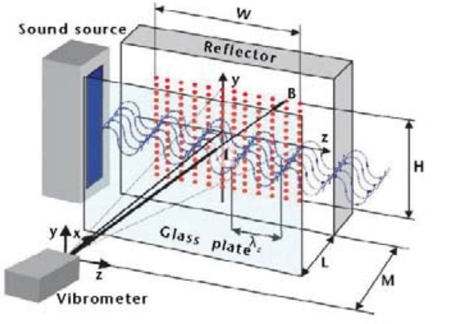
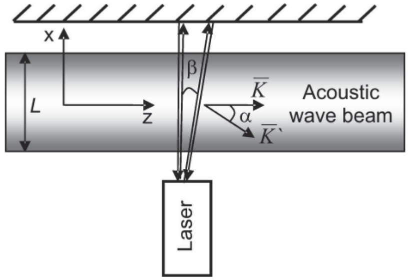
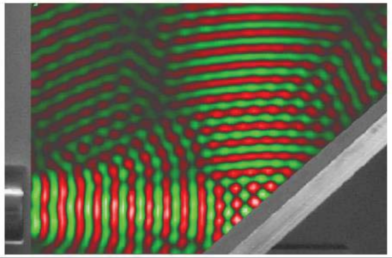


Figure 2: $V = 20\,\mathrm{m/s}$ at $f = 12336\,\mathrm{Hz}$, $A_{ref} = 8.2\,\mathrm{dB}$

Refracto-Vibrometry







Omni-Directional Speakers

Bruel and Kjaer OmniSource Loudspeaker - Type 4295

Bruel and Kjaer OmniPower Sound Source - Type 4292-L

Reference Sound Sources

Bruel and Kjaer Reference Sound Source - Type 4204

Acculab RSS Type 600

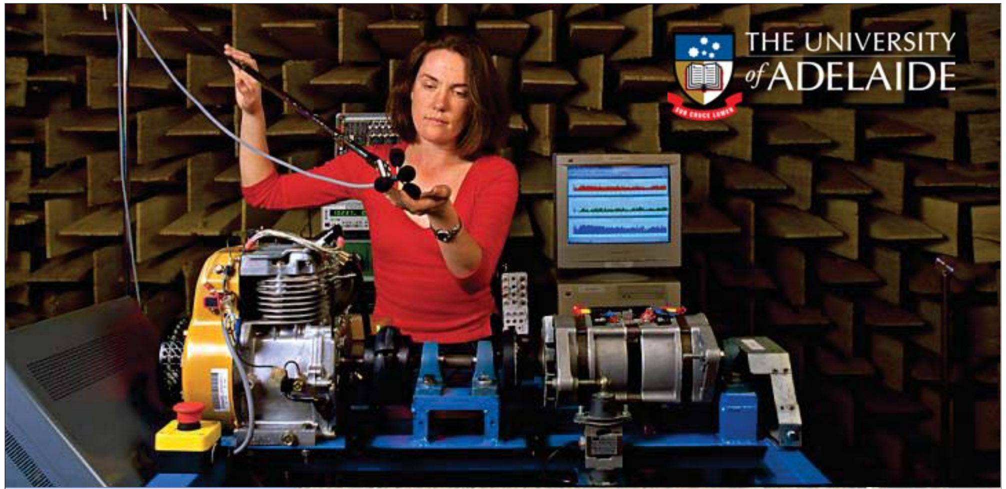
You'll use a calibrated fan sound source in the lab practical

Questions

- What instrument would you use to determine the noise exposure of a person?
- What instrument would you use to measure sound intensity?
- Describe 3 methods to measure sound power of a noise source.
- What device could you use to measure the sound absorption coefficient of acoustic foam?
- The sound pressure level measured from a noise source radiating omni-directionally will decrease at a rate of ___ dB per doubling of distance?

Questions

- What instrument would you use to record the sound levels a person is exposed to over a week period?
- Which instruments can be used to measure sound power levels?
- A microphone and pre-amplifier generates a voltage when exposed to sound. What instrument is used to calibrate the measurement system?



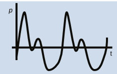
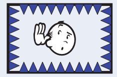
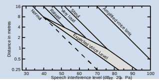
CRICOS PROVIDER 00123M

The Human Ear

adelaide.edu.au

seek LIGHT

Course Content – How it fits

TOPIC	
Basics	
General Noise Control	
Instrumentation	
The Ear	
Criteria	

Contents

- Anatomy of the human ear
- Outer ear
- Middle ear
- Inner ear
- Hair cells
- Neural encoding
- Noise induced hearing loss
- Pitch and source localisation
- Basilar membrane response

Learning Outcomes

- Anatomy of the human ear.
- Understand how the human ear converts sound into an auditory response.

Questions To Think About

- What is the mechanical gain by the bones in the middle ear?
- In which frequency band does hearing loss first occur?
- What would be the expected hearing range of a mouse, if the size of the anatomical parts scaled accordingly?
- If the bones in the middle ear are seized, is it possible to hear?

Range of Hearing in Humans

From the lecture on Fundamentals

- 0dB re 20μPa = 0.000020 Pa
  - ½ drop of water over 10m²

- 140dB re 20μPa = 200 Pa
  - 100 x 2 litre bottles over 10m²

$$\frac{200\ \text{Pa}}{20\ \mu\text{Pa}} = \frac{2\times10^{2}}{2\times10^{-5}} = 10^{7}$$

From the lecture on Fundamentals

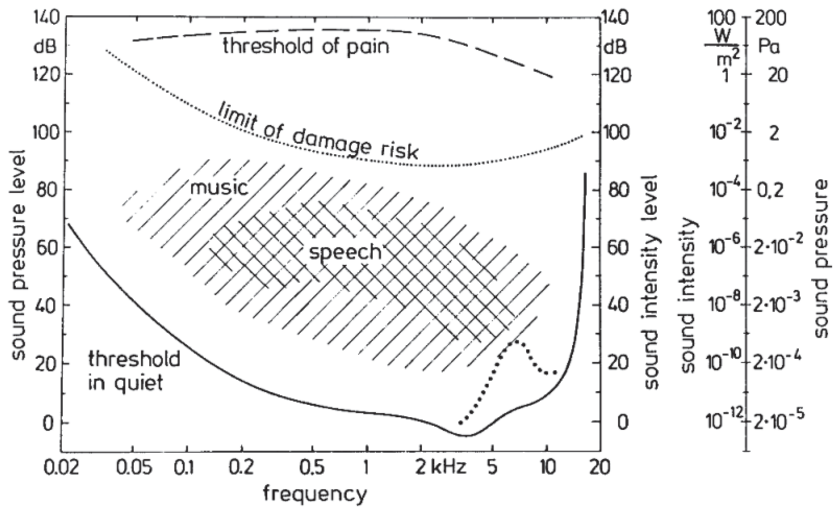
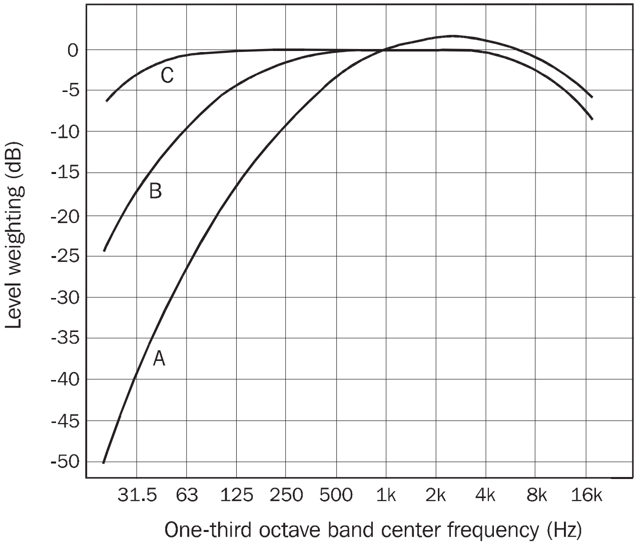


Fig 2.1, Fastl and Zwicker, Psychoacoustics, Facts and Models (2007)

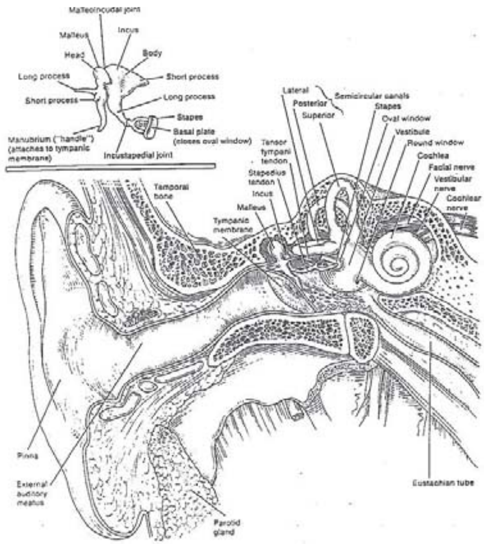
Weighting Scales



- **A-weighting** most common.
- Low frequencies have low weighting.
- *Roughly* describes human hearing response.

ENC p104

Anatomy of the Human Ear



http://ocw.usu.edu/Electrical_and_Computer_Engineering/Science_of_Sound/ECE_3260__ear.swf-view.html

Other Animations on the Internet

- Excellent animation with lots of details
  http://www.sumanasinc.com/webcontent/animations/content/soundtransduction.html
- Virtual Tour of the Ear
  http://ctl.augie.edu/perry/ear/hearmech.htm
- Good, but not quite enough details
  https://www.wisc-online.com/learn/natural-science/life-science/ap14204/the-sense-of-hearing

Outer Ear

- PINNA
  - Antenna with directional properties above 5000 Hz (λ = 69mm) to 17,000 - 20,000 Hz (λ = 17mm)

- AUDITORY CANAL
  - average length, 23mm
  - closed organ pipe resonance ≈ 4,000 Hz corresponding to $\lambda/4$ = 23mm
  - resonance amplification at 4,000 Hz ≈ 3 due to pressure maximum at eardrum (tympanic membrane)

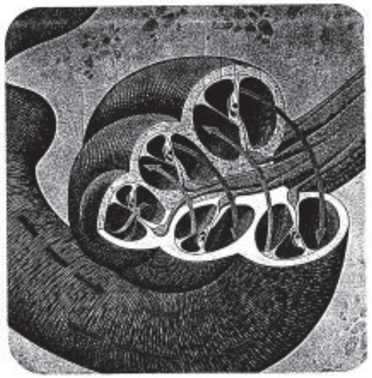
- EARDRUM
  - cone shaped diaphragm, 9mm dia.

Middle Ear

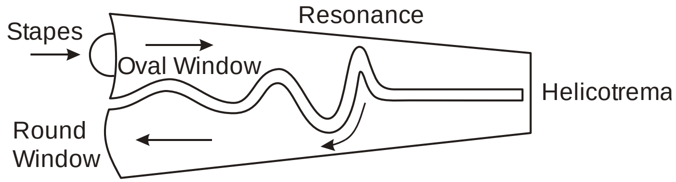
- connected to throat through eustacean tube
- air-filled: volume ≈ 2cc
- mechanical transformer
  - converts small pressures and large displacements into vice versa
  - consists of 3 tiny bones (hammer, anvil, stirrup / stapes) which connect the eardrum to the oval window
  - mechanical advantage of 3:1
  - $\frac{\text{area of eardrum}}{\text{area of oval window}} = 5$
  - total mechanical advantage = 15:1

- muscles attached to stapes stiffen when subjected to loud noise to protect inner ear
  - NOT effective for impulse noise

Inner Ear - Cochlea



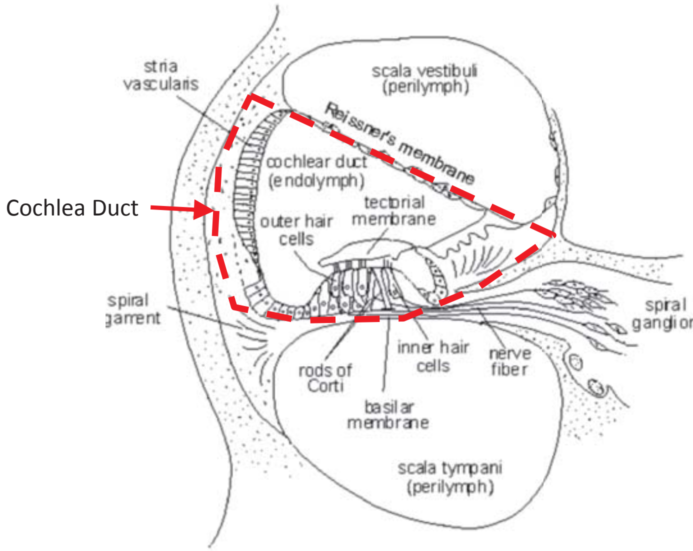
Cross-Section of Cochlear Duct



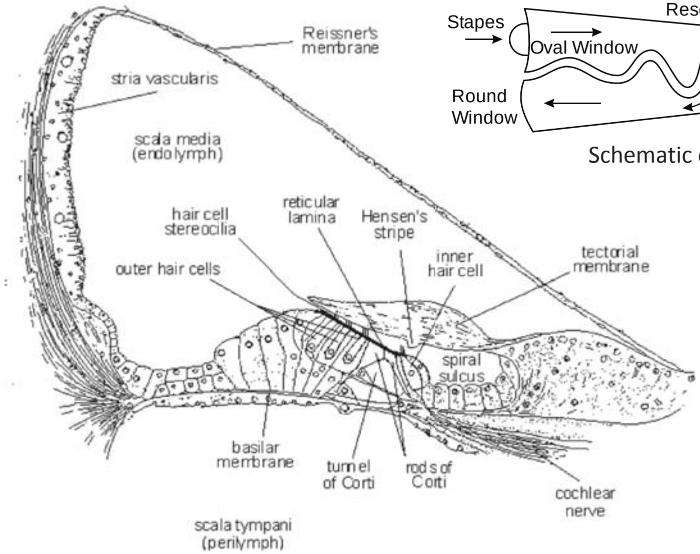
Schematic of cochlea (unrolled)

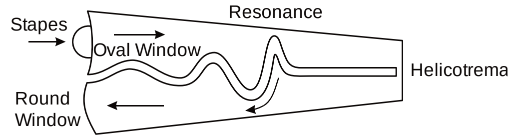
- If you were unroll the cochlea, it would look like a long duct with an upper and lower gallery.
- Vibration of Stapes causes Oval Window to vibrate which in turns displaces fluid in cochlea.

Cross-Section of Cochlea



Cross-Section of Cochlear Duct



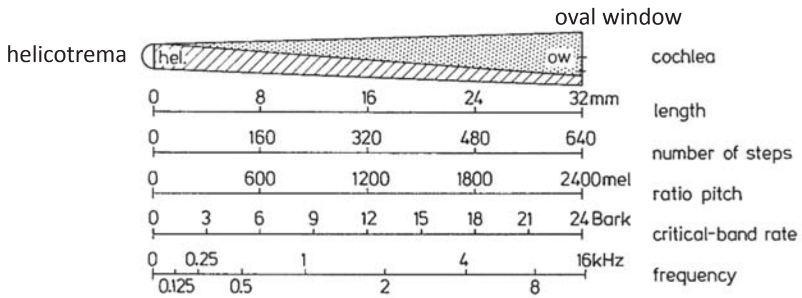


Schematic of cochlea (unrolled)

Inner Ear - Cochlea

- spiral chamber $2\frac{5}{8}$ turns
- houses hearing organs
- divided into 3 chambers, 34 mm long
  - upper gallery
  - lower gallery
  - cochlear partition
- upper and lower galleries joined at apex
- flexible walls of cochlea duct result in dispersive sound propagation
  - wavelength dependent on location
  - speed of sound goes to zero at a particular basilar membrane location for every frequency, resulting in large amplification of basilar membrane motion at that point
  - location depends on frequency

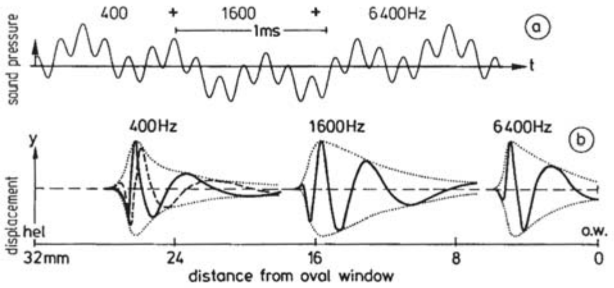
Cochlea



- High frequency sounds excite start of cochlea, low frequency at the end of cochlea.

Fig 6.1, Fastl and Zwicker, Psychoacoustics, Facts and Models (2007)

Basilar Membrane



- The basilar membrane vibrates at different locations depending on frequency [frequency – place transformation]
- Figure shows vibration response due to simultaneous 400Hz + 1600 Hz + 6,400Hz sound

Fig 3.5, Fastl and Zwicker, Psychoacoustics, Facts and Models (2007)

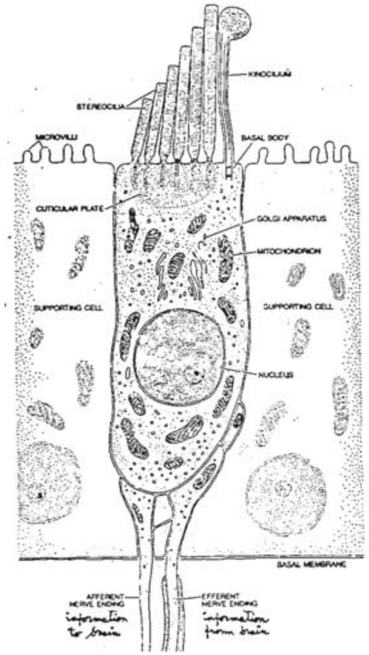
Inner Ear - Cochlea (Cont.)

- High frequencies only excite that part of gallery near oval window
- Round window - pressure release mechanism
- Basilar membrane
  - separates upper and lower galleries
  - controls overall stiffness of cochlear partition
  - its vibration determines how we hear
  - stiffness decreases as apex of the cochlea is approached such that different parts of the membrane are resonant at different audio frequencies

Hair Cells

- Sound sensing organs
- Transform shear force (or displacement) to an analog voltage
- Attached to basilar membrane
- Contained in organ of corti
- Outer hair cells
  - sense fluid displacement
  - end of stereocelia attached to tectorial membrane
  - extension and contraction of hair cells as a result of the generated voltage causes additional movement of the tectorial membrane, resulting in about 25dB amplification of motion of inner hair cell stereocelia - feedback mechanism to amplify low level sound
  - responsible for selective amplification allowing us to focus on a particular conversation in a noisy environment such as a party.

- **Inner hair cells**
  - sense velocity
  - ends of stereocelia are free
  - responsible for detecting the sound



Neural Encoding

- Cochlear nerve (attached to each hair cell)
  - efferent (signals from brain to ear)
  - afferent (signals from ear to brain)
- Cells of afferent nervous system convert analog signal from hair cells to digital code
  - fire once each cycle in phase up to 5 kHz; helps in determining frequency
  - Random phase above 5 kHz
  - fire when threshold voltage exceeded
  - not all fire
  - probability of firing increases as voltage increases
- encoding rate to brain by nervous system is 170Hz max.
- ear does all frequency analysis and encoding rate is proportional to loudness

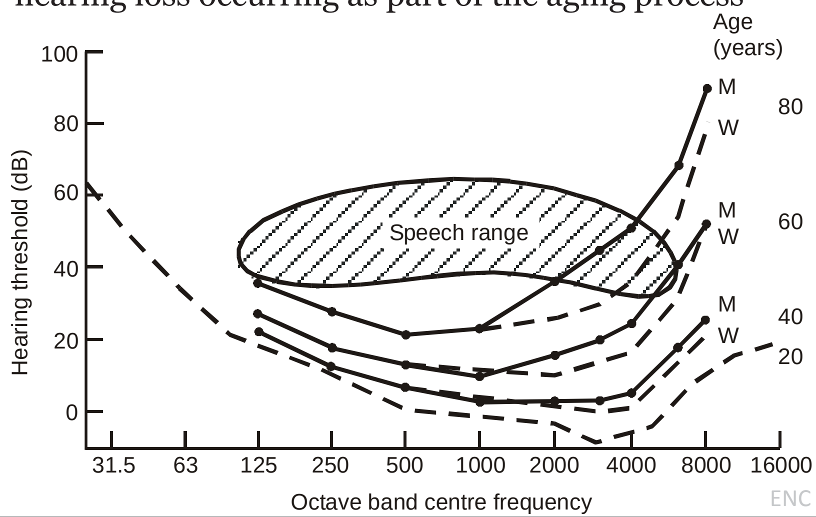
- hair cells are characterised by resonance frequencies
  - correspond to basilar membrane resonance at same location
  - effectively increases "Q" of basilar membrane and thus sensitivity of ear to changes in pitch
- efferent nerve system carrying instructions from brain to outer hair cells
  - can control state of outer hair cells (turn on or off)
  - can also amplify the motion of the outer hair cell stereocelia (referred to as “un-damping”) and increase fluid velocity past the inner hair cells which actually sense the sound
  - act as a volume control to maintain inner hair cell neuron firing within the dynamic range of brain

Noise Induced Hearing Loss

- stereocelia on outer hair cells fatigue and break first when subjected to excessive noise resulting in loss of ability to
  - locate noise sources
  - converse in a noisy environment
  - first sign of noise induced loss
  - not helped by conventional hearing aid
- following outer hair cell loss, inner hair cells lost progressively
  - first noticed in 4000Hz to 6000Hz range (see next 2 slides)
  - speech range (500Hz to 2000Hz) eventually affected
- half octave shift
  - experiments show that when people are subjected to intense tonal sound, the temporary threshold shift occurs at a frequency one half octave lower than the test frequency. WHY? See text, pp71 to 73.

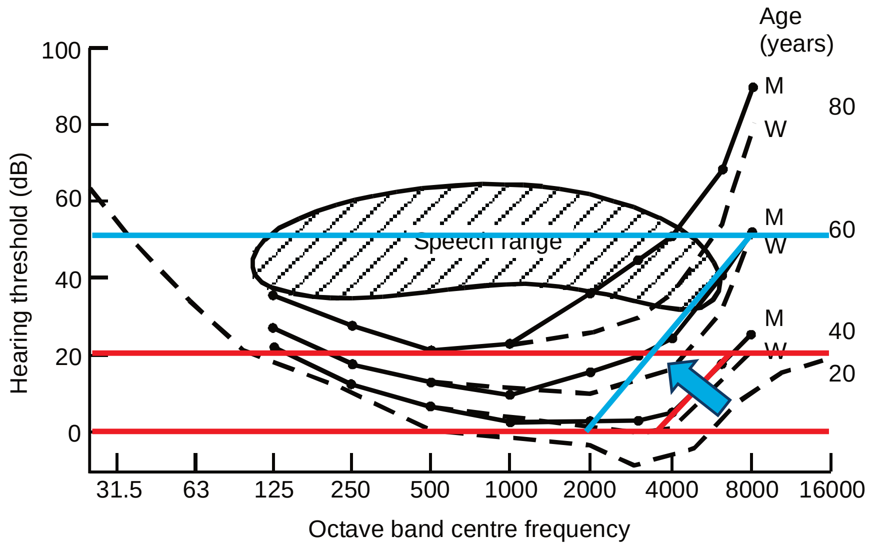
Presbyacusis

- hearing loss occurring as part of the aging process

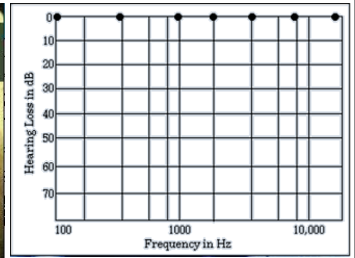


ENC p145

Change in Hearing Threshold

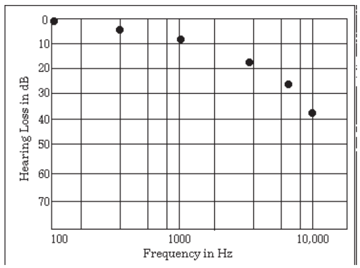


Normal Hearing



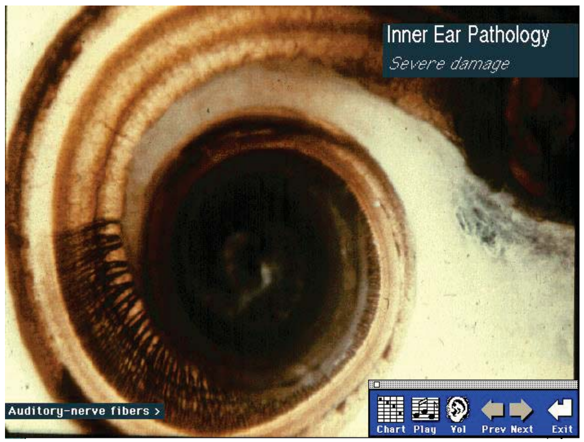
http://www.neurophys.wisc.edu/~ychen/auditory/audiogram/audiogrammain.html

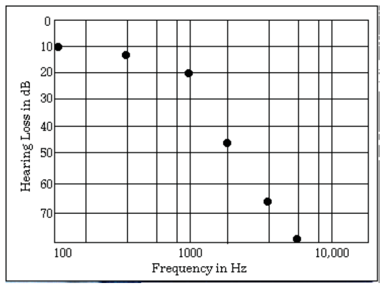
Moderate Damage



http://www.neurophys.wisc.edu/~ychen/auditory/audiogram/audiogrammain.html

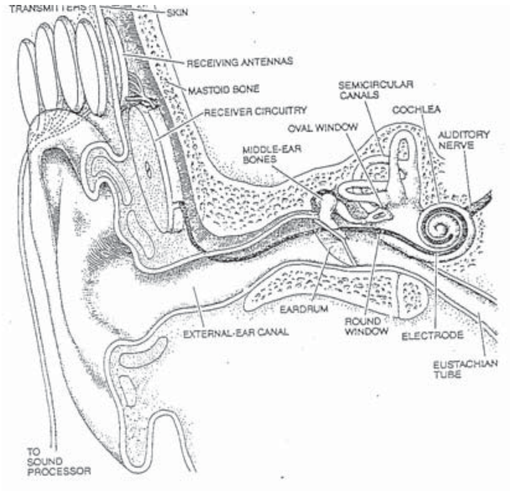
Severe Damage



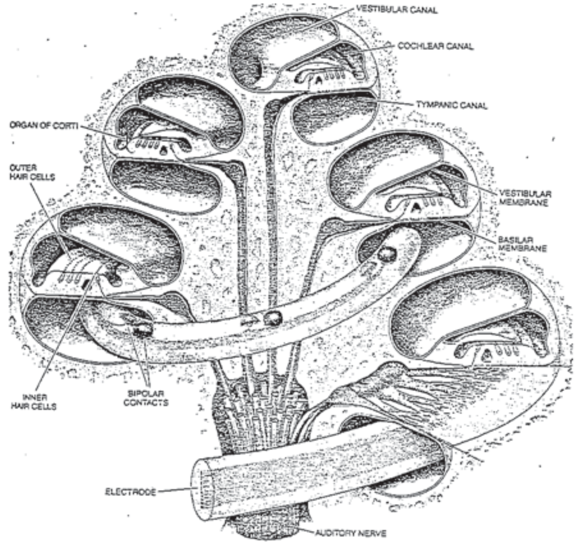


http://www.neurophys.wisc.edu/~ychen/auditory/audiogram/audiogrammain.html

- possible help
  - prosthesis implanted in cochlear
  - electrodes excite hair cells selectively and cause neurons to fire
  - problems in fitting and properly localising stimuli



Prosthesis In Cochlear



Hearing with a Cochlear Implant

http://www.youtube.com/watch?v=SpKKYBkJ9Hw

The psychological and social consequences of

Hearing loss

Reduced hearing is a very common condition among our working population. Work induced hearing loss is, of course, a kind of physical injury. But what about the psychological aspects?

To be slowly, steadily and irrevocably more and more isolated from the surrounding world is a nightmarish experience. No wonder the psychological effects of hearing loss can be just as traumatic as the physical condition.

Hearing: a cornerstone of human communication

Hearing is a prerequisite for natural, spontaneous development of speech. Hearing and speech are the natural links in the contact between people.

Therefore, reduced or lost hearing abilities bring about psychological stress that can be unbearable for the patient, and that can badly affect family, friends and acquaintances as well.

Many people don't want to talk about their hearing problem. Hearing loss is an invisible condition. Most sufferers try to live up to normal expectations from themselves and from their surroundings for as long as possible. This can lead to severe problems that may not be evident in visible handicap.

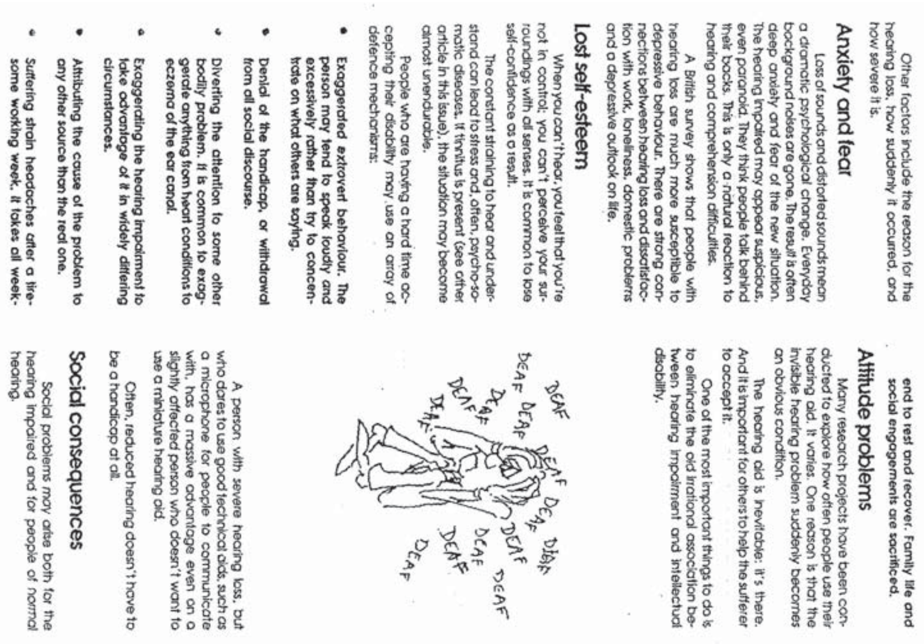
Three groups

Hearing experts talk about three different kinds of deafness:

- Childhood deafness
  People who were born deaf, or who become deaf before learning to talk. People in this group rely almost completely on sign language. They have never heard their own voice, and never had the opportunity to control it.
- Adulthood deafness
  People in this group suffer from severely impaired hearing or complete deafness. They have lost their hearing at an adult age, know how to speak, and have language comprehension. However, the hearing damage is so severe that modern hearing aids cannot improve the hearing. They are isolated from the surrounding hearing world, but also from childhood deafness sufferers, who communicate on a completely different basis.
- Hearing impairment
  The hearing may be more or less reduced, but has occurred late in life: normal language development has not been affected. The hearing damage may have occurred at one specific point in time, or can be a gradually worsening condition.

Psychological ramifications

Many factors will affect a person who suffers from impaired hearing. The attitudes of the rest of the community, family and friends, and, of course, work environment are all important to the way the problem can be dealt with. If the sufferer is treated with understanding and respect, it becomes much easier to accept and cope with reduced hearing.



Other factors include the reason for the hearing loss, how suddenly it occurred, and how severe it is.

Anxiety and fear

Loss of sounds and distorted sounds mean a dramatic psychological change. Everyday background noises are gone. The result is often deep anxiety and fear of the new situation. The hearing impaired may appear suspicious, even paranoid. They think people talk behind their backs. This is only a natural reaction to hearing and comprehension difficulties.

A British survey shows that people with hearing loss are much more susceptible to depressive behaviour. There are strong connections between hearing loss and dissatisfaction with work, loneliness, domestic problems and a depressive outlook on life.

Lost self-esteem

When you can't hear, you feel that you're not in control; you can't perceive your surroundings with all senses. It is common to lose self-confidence as a result.

The constant straining to hear and understand can lead to stress and, often, psycho-somatic diseases. If tinnitus is present (see other article in this issue), the situation may become almost unendurable.

People who are having a hard time accepting their disability may use an array of defence mechanisms:

- **Exaggerated extrovert behaviour. The person may tend to speak loudly and excessively rather than try to concentrate on what others are saying.**
- **Denial of the handicap, or withdrawal from all social discourse.**
- **Diverting the attention to some other bodily problem. It is common to exaggerate anything from heart conditions to eczema of the ear canal.**
- **Exaggerating the hearing impairment to take advantage of it in widely differing circumstances.**
- **Attributing the cause of the problem to any other source than the real one.**
- **Suffering strain headaches after a tiresome working week. It takes all weekend to rest and recover. Family life and social engagements are sacrificed.**

Attitude problems

Many research projects have been conducted to explore how often people use their hearing aid. It varies. One reason is that the invisible hearing problem suddenly becomes an obvious condition.

The hearing aid is inevitable: it's there. And it is important for others to help the sufferer to accept it.

One of the most important things to do is to eliminate the old irrational association between hearing impairment and intellectual disability.

A person with severe hearing loss, but who dares to use good technical aids, such as a microphone for people to communicate with, has a massive advantage even on a slightly affected person who doesn't want to use a miniature hearing aid.

Often, reduced hearing doesn't have to be a handicap at all.

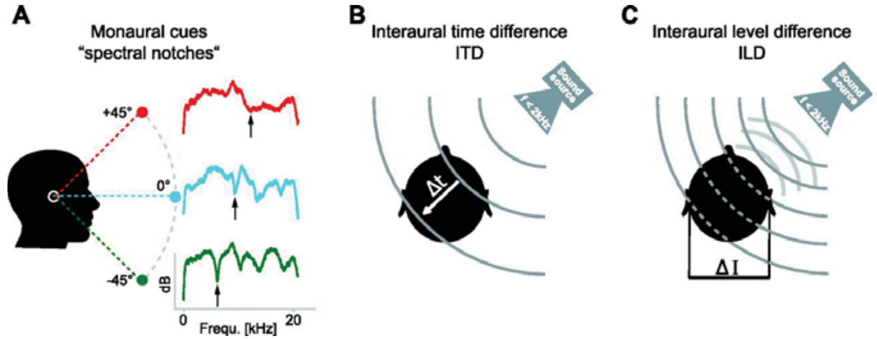
Social consequences

Social problems may arise both for the hearing impaired and for people of normal hearing.

Source Localisation

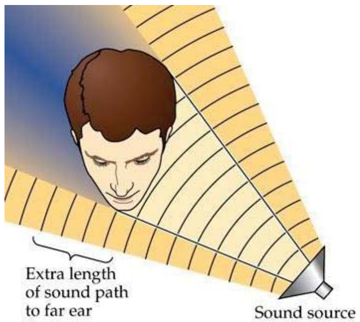
The way humans can determine the location of a noise source depends on several "clues"

A. Spectral cues
B. Interaural Time Difference – it takes a bit longer for sound to reach the other ear
C. Interaural Level Difference – there is shadowing by the head.



Source Localisation

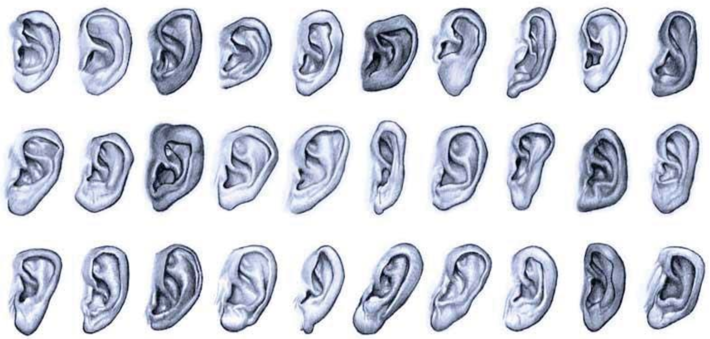
In the horizontal plane the ITD and ILD help determine location of sound.



In the vertical plane, sound bounces off the pinna and gives us clues.

Source Localisation

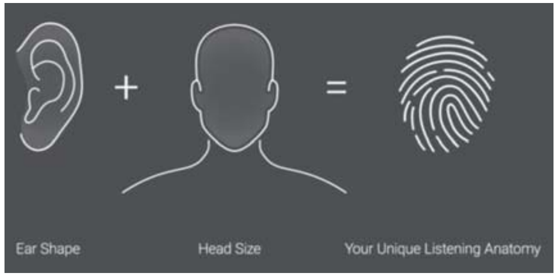
But everyone has a unique pinna.



Hence, a person's hearing is unique.

Source Localisation

- The combined response is called the Head Related Transfer Function HRTF.
- A person's measured HRTF varies between individuals.



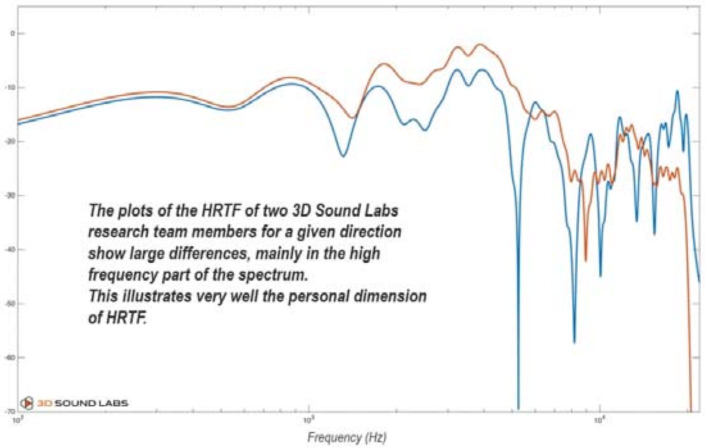
https://www.ossic.com/blog/2017/10/4/what-is-a-hrtf

Personal HRTF

- It can be measured by
  - using an anechoic chamber and placing numerous loudspeakers around a person that is wearing in-ear microphone.

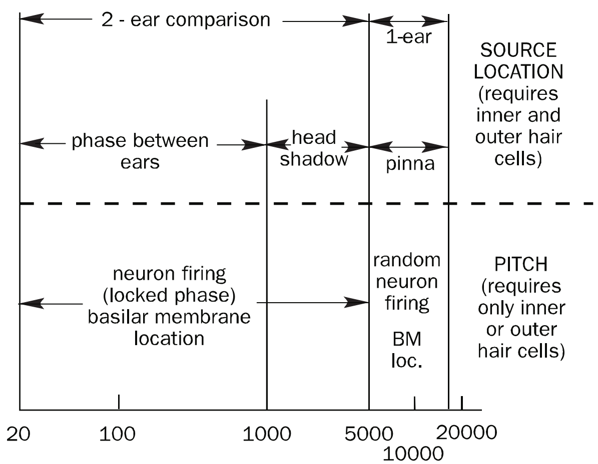
Personal HRTF

Measured HRTFs vary between individuals. http://www.3dsoundlabs.com/hrtf-101/



Hence, generic HRTF filters are not excellent for creating believable Virtual Reality experiences.

Pitch and Source Location



Binaural Hearing

From the Psychoacoustics section a bit later ...

- Localization
  - Some sources always sound like they are originating from a particular direction.

Fig 15.3, Fastl and Zwicker, Psychoacoustics, Facts and Models (2007)

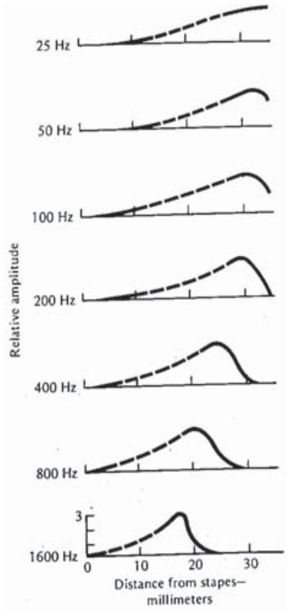
Binaural Hearing

- Hence, the sound recorded by a microphone “sounds” very different compared to a binaural artificial headform.
- The microphone has no diffraction by shape of head, no reflections, no ears, no .... so the response is very different.

Masking

- Low frequencies mask high frequencies
  - Notice how low frequency response will interfere with high frequency response.

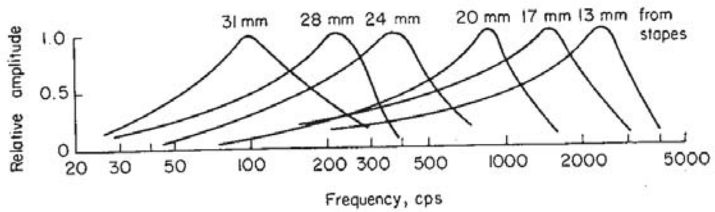
- High frequency hearing lost first



Basilar membrane response

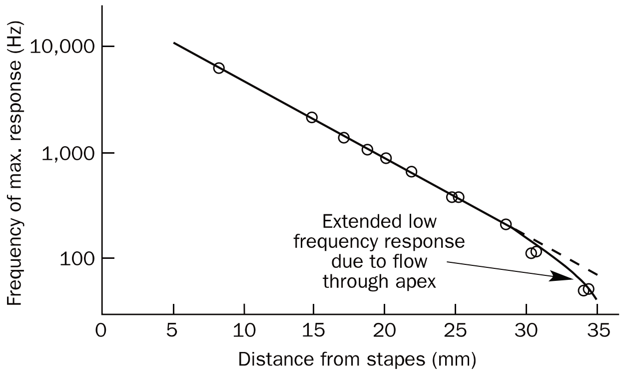
Masking

Basilar membrane frequency response at 6 locations



Location of maximum response as a function of frequency

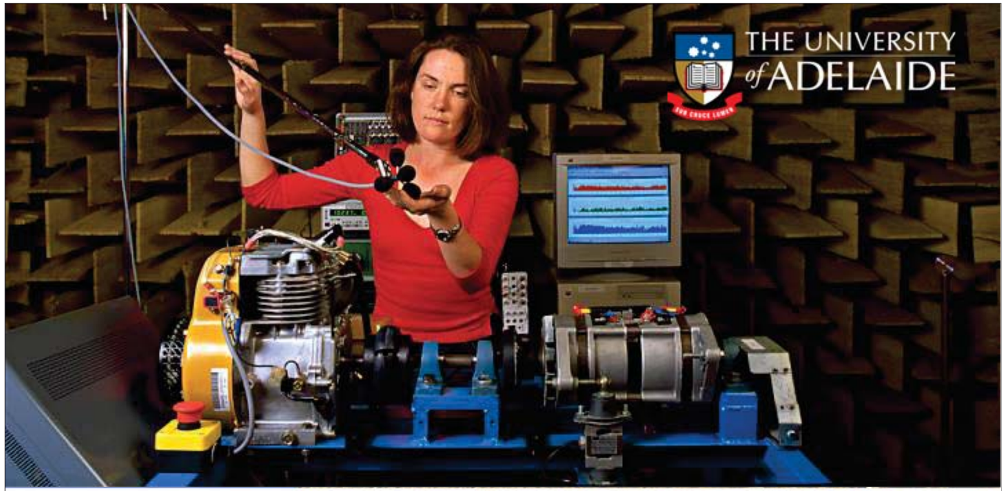
- Previous slide showed response vs. distance.
- This slide shows response vs frequency.



- Following from previous figures...
  Location of maximum response from stapes end depends on frequency.

Summary

- Anatomy of the human ear
- Outer ear
- Middle ear
- Inner ear
- Hair cells
- Neural encoding
- Noise induced hearing loss
- Pitch and source localisation
- Basilar membrane response



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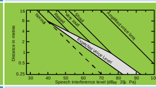
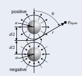
Noise Criteria

adelaide.edu.au seek LIGHT

Course Content – Where Are We?

TOPIC	
The Ear	
Criteria	
Psychoacoustics	
Sound Sources	
Sound Power	

Contents

- Noise Descriptors
- Noise Criteria
- Noise Induced Hearing Loss and Criteria
- Speech Interference
- Building Acoustics Criteria
- Environmental Noise Criteria
- Environmental Noise Surveys

Learning Outcomes

On completion of the course, students should:

- **be able to assess occupational and environmental noise problems,**

Details:

- Various metrics used to quantify occupational and environmental noise.
- Hearing loss associated with age and exposure.
- Hearing damage risk and trading rules.
- Architectural noise ratings: Noise Rating (NR), Noise Criteria (NC), Room Criteria (RC), Balanced Noise Criteria (NCB), Room Noise Criteria (RNC).
- Environmental noise criteria.

Questions to Think About

- How do you determine the noise exposure a person receives?
- What are the noise limits for workplaces?
- How do you rate the noise level in a room?
- What level of sound is going to cause communication problems?
- How do you measure noise levels outside due to industrial operations?
- What are the limits on these noise levels?

Questions Asked This Lecture

- Q1 An employee working at a bar in a night club is exposed to 105 dB(A) for a 4 hour shift.
  a) What is the A-weighted Sound Exposure?
  b) What is the 8-hour equivalent level?
  c) How much would the shift length have to be reduced to comply with the 85 dB(A) 8 hour requirement?

- Q2 SPL is measured as

Freq [Hz]	63	125	250	500	1000	2000	4000	8000
SPL [dB]	48	48	43	38	30	20	16	12

  What is the NCB rating?

- Q3 Exit Survey

Relevant Standards

- National Code of Practice for Noise Management and Protection of Hearing at Work - 3rd Edition
  https://www.safeworkaustralia.gov.au/system/files/documents/1702/nationalcodeofpractice_noisemanagementandprotectionofhearingatwork_3rd_edition_nohsc2009-2004_pdf.pdf

- Australian Standards: AS/NZS 1269.1:2005 (R2016) Occupational noise management Measurement and assessment of noise immission and exposure

Resources

- Bies and Hansen, "Engineering Noise Control"
- National Acoustics Laboratory, Hearing Loss Simulator: https://www.nal.gov.au/about-hearing-loss/hearing-loss-simulations/
- National Acoustics Laboratory, Binge Listening report

Noise and Health Issues

- World Health Organisation (WHO), noise is the second largest environmental cause of health problems, just after the impact of air quality (particulate matter).
- An EU study on the Health implication of road, railway and aircraft noise in the European Union found that exposure to noise in Europe contributes to:
  - about 910,000 additional prevalent cases of hypertension,
  - 43,000 hospital admissions per year, and
  - at least 10,000 premature deaths per year related to coronary heart disease and stroke.

http://ec.europa.eu/environment/noise/health_effects_en.htm

Noise and Health Issues

- In the USA, hearing loss is the *third-most common* chronic physical condition among adults, after hypertension and arthritis, and is more prevalent than diabetes or cancer.
- About 11% of the U.S. working population has hearing difficulty.
- About 24% of the hearing difficulty among U.S. workers is caused by occupational exposures.
- About 8% of the U.S. working population has tinnitus ('ringing in the ears') and 4% has both hearing difficulty and tinnitus.

http://www.cdc.gov/niosh/topics/ohl/

Background

- BHP Billiton (2003), a major mining and manufacturing company in Australia (and globally), conducted an extensive assessment of exposures in the workplace, and reported that **51% of employees** are potentially exposed above the noise exposure limits (if no hearing protection is used).

BHP Billiton (2003) Health, Safety, Environment and Community Report, 2003. Performance Summaries.

Background

- BHP report:

In FY2015, the incidence of employee occupational illness was 4.93 per million hours worked, an increase of 74 per cent compared with FY2014. Noise induced hearing loss cases increased significantly due to a more accurate assessment triggered by incorrectly applying our hearing loss criteria in previous years at some assets. Our reduction in musculoskeletal illnesses was primarily due to the introduction of a multifaceted control program at one of our assets.

Year ended 30 June [(a)]	2015	2014	2013
Noise induced hearing loss	**3.05**	0.68	0.51
Musculoskeletal	**1.52**	1.61	1.24
Other illnesses	**0.36**	0.55	0.64
Total	**4.93**	2.84	2.39

(a) Includes data for Continuing and Discontinued operations for the financial years being reported.

Massive increase!

Twice as big as musculoskeletal !

http://www.bhp.com/-/media/bhp/documents/investors/annual-reports/2015/bhpbillitonstrategicreport2015.pdf?la=en

Background

- When personal music players are set at maximum volume, it is possible to exceed the recommended standard limits in as little as 5 minutes .
  Cory D.F. Portnuff and Brian J. Fligor. Sound output levels of the ipod and other mp3 players: Is there potential risk to hearing? In Noise-Induced Hearing Loss In Children At Work And Play, Embassy Suites River Center, Cincinnati, Ohio, October 19-20 2006. National Hearing Conservation Association. http://www.hearingconservation.org/docs/virtualPressRoom/portnuff.htm.
- Hearing loss occurs due to both advancing age and environmental exposure (e.g. loud noise, smoking, chemicals, drugs, ...)
  - Commonly used medicines that may cause hearing loss include: Aspirin, when large doses (8 to 12 pills a day) are taken. Nonsteroidal anti-inflammatory drugs (NSAIDs), such as ibuprofen and naproxen. Certain antibiotics, especially aminoglycosides (such as gentamicin, streptomycin, and neomycin).

Noise Descriptors

- A-weighted Equivalent Continuous Noise Level (times are in hours)
  - used for occupational and environmental noise

$$L_{Aeq,T} = 10\log_{10}\left[\frac{1}{T}\int_0^T 10^{L_A(t)/10}\,dt\right]$$

  - which can be written in summation form as

$$L_{Aeq,T} = 10\log_{10}\left\{\frac{\left[t_1 10^{L_{A1}/10} + t_2 10^{L_{A2}/10} + \ldots + t_m 10^{L_{Am}/10}\right]}{\left[t_1 + t_2 + \ldots + t_m\right]}\right\}$$

Noise Descriptors

- For occupational noise, most common descriptor is $L_{Aeq,8h}$
- For sound experienced over T hours:

$$L_{Aeq,8h} = 10\log_{10}\left[\frac{1}{8}\int_0^T 10^{L_A(t)/10}dt\right]$$

ENC p139

- If the SPL (L_{Ai}) is measured using a sound level meter at *m* different locations where an employee may spend some time, then the above integral equation can be written as:

$$\boxed{L_{Aeq,8h} = 10\log_{10}\frac{1}{8}\left[t_1 10^{L_{A1}/10} + t_2 10^{L_{A2}/10} + \ldots + t_m 10^{L_{Am}/10}\right]}$$

ENC p139

Example

- An employee in a factory spends…

Activity	Time [hrs]	SPL [dB(A)]
Breaks	1.5	65
Grinding	2	100
Machine press	2	93
Paint shop	1.5	70
Assembly line	1.0	90
TOTAL	8	

$$L_{Aeq,8h} = 10\log_{10}\frac{1}{8}\left[t_1 10^{L_{A1}/10} + t_2 10^{L_{A2}/10} + \ldots + t_m 10^{L_{Am}/10}\right]$$

$$= 10\log_{10}\frac{1}{8}\left[\begin{array}{l}1.5\times10^{65/10} + 2\times10^{100/10} + 2\times10^{93/10} \\ +1.5\times10^{70/10} + 1.0\times10^{90/10}\end{array}\right]$$

$$= 94.95\ \text{dB(A)}$$

$$= 95.0\ \text{dB(A)}$$

Noise Descriptors

- A-weighted **Sound Exposure** (where times are in hours)

$$E_{A,T} = \int_{t_1}^{t_2} p_A^2(t)dt = \sum_i p_A^2 \times t_i$$

ENC p139

  – Remember from the fundamentals section

$$L_p = 10\log_{10}\frac{p_{\text{RMS}}^2}{p_{\text{ref}}^2}$$

$$p_{\text{RMS}}^2 = p_{\text{ref}}^2 \times 10^{L_p/10}$$

  – Using previous equations, the following can be shown to be true:

$$E_{A,T} = 4T\times10^{(L_{Aeq,T}-100)/10}$$

ENC p140

Noise Descriptors

- A-weighted Sound Exposure Level, L_{AE} or SEL, where times are in seconds.

$$L_{AE} = 10\log_{10}\left[\int_{t_1}^{t_2}\frac{p_A^2(t)}{p_{ref}^2}dt\right] = 10\log_{10}\left[\frac{E_{A,T}\times3600}{p_{ref}^2}\right]$$

ENC p140

  – used mainly for assessment of transient environmental noise, such as traffic noise, aircraft noise and train noise.

Noise Descriptors

- From the preceding equations we can write:

$$L_{Aeq,8h} = 10\log_{10}\left(\frac{E_{A,8h}}{3.2\times10^{-9}}\right) = 10\log_{10}\left(\frac{1}{28{,}800}10^{L_{AE,8h}/10}\right)$$

ENC p141

- Day–Night Average Sound Level, L_{dn} or DNL
  - used sometimes to quantify traffic noise

$$L_{dn} = 10\log_{10}\frac{1}{24}\left[\int_{22:00}^{07:00} \underline{10}\times10^{L_A(t)/10}dt + \int_{07:00}^{22:00} 10^{L_A(t)/10}dt\right] \quad \text{dB}$$

Ouch!

ENC p141

Noise Descriptors

- Community Noise Equivalent Level, L_{den} or CNEL
  - used sometimes to quantify industrial noise and traffic noise

$$L_{den} = 10\log_{10}\frac{1}{24}\left[\begin{array}{l}\int_{22:00}^{07:00} 10\times10^{L_A(t)/10}dt + \\ \int_{07:00}^{19:00} 10^{L_A(t)/10}dt + \\ \int_{19:00}^{22:00} 3\times10^{L_A(t)/10}dt\end{array}\right] \quad \text{dB}$$

ENC p141

Example

- An employee in a factory spends...

Activity	Time [hrs]	SPL [dB(A)]	p^2	$E_{A,8hr}$
Breaks	1.5	65	0.001265	0.001897
Grinding	2	100	4	8
Machine press	2	93	0.798105	1.59621
Paint shop	1.5	70	0.004	0.006
Assembly line	1.0	90	0.4	0.4
TOTAL	8			10.0

Example

- Or using the formula

$$E_{A,T} = \sum_i 4T\times10^{(L_{Aeq,T}-100)/10}$$

$$= 4\times1.5\times10^{(65-100)/10} + 4\times2\times10^{(100-100)/10} +$$
$$4\times2\times10^{(93-100)/10} + 4\times1.5\times10^{(70-100)/10} +$$
$$4\times1.0\times10^{(90-100)/10}$$
$$= 10.0\ \text{Pa}^2\text{h}$$

Example

- Which can be converted back into an $L_{Aeq,8h}$ value

$$L_{Aeq,8h} = 10\log_{10}\left(\frac{E_{A,8h}}{3.2\times10^{-9}}\right)$$
$$= 10\log_{10}\left(\frac{10.0}{3.2\times10^{-9}}\right)$$
$$= 94.95\,\text{dB(A)}$$
$$= 95.0\,\text{dB(A)}$$

For practical purposes a noise exposure of 1.0 Pa²h corresponds to $L_{Aeq,8h}$ of 85 dB(A).

Example

Day	$L_{Aeq,8h}$ dB(A)	Pa²	Hours	Pa²h
Monday	85	0.126	8	1
Tuesday	97	2.0	8	16
Wednesday	100	4.0	8	32
Thursday	102	6.3	8	50.7
Friday	85	0.126	8	1
Total				101

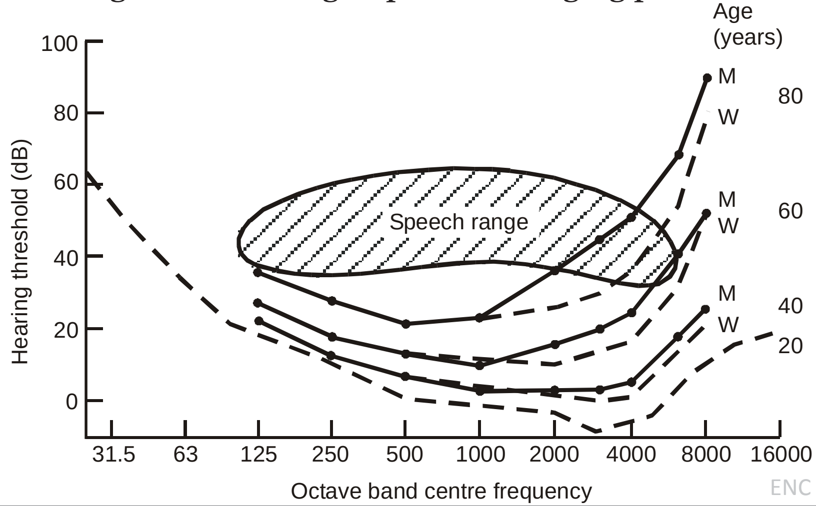
The average Pascal-squared hour value over 5 working days is therefore 101/5 which is equal to 20.2 Pa²h.

Noise Criteria

- provide a guide for acceptable noise levels
- provide a means for estimating required noise reduction
- provide the means for determining feasibility of alternative solutions
- provide a means for estimating the cost of noise control
- Industrial criteria fall into 4 categories
  - hearing damage risk
    - steady state noise
    - impulse and impact noise
  - speech interference
  - work efficiency
  - environmental

Presbyacusis

- hearing loss occurring as part of the aging process



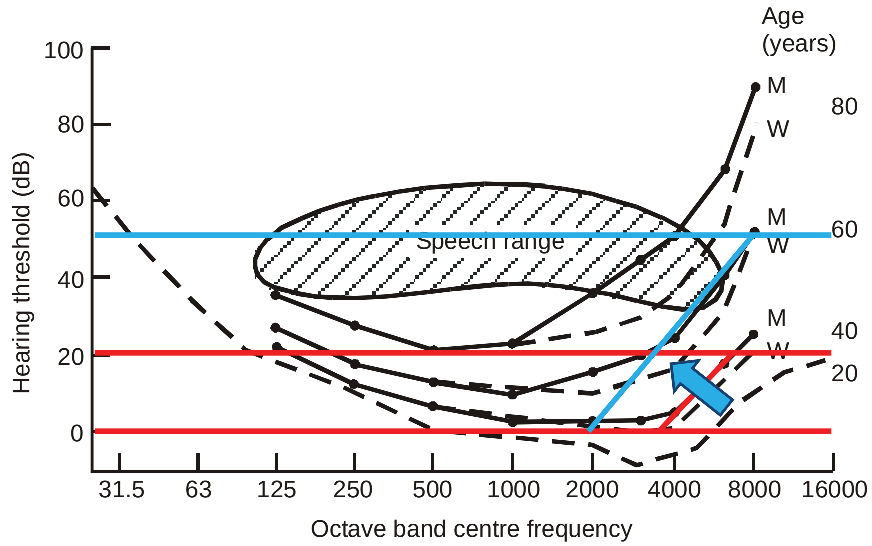
ENC p145

Presbyacusis

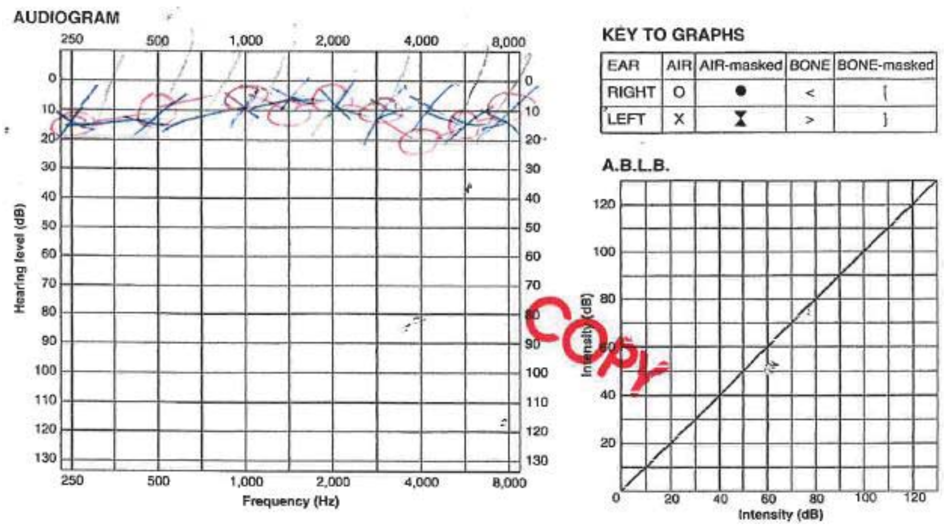
Things to note about the graph on previous slide:

- increasing loss with increasing frequency
- loss accelerates with age
- men worse off than women
- notice the frequency range that drops off first

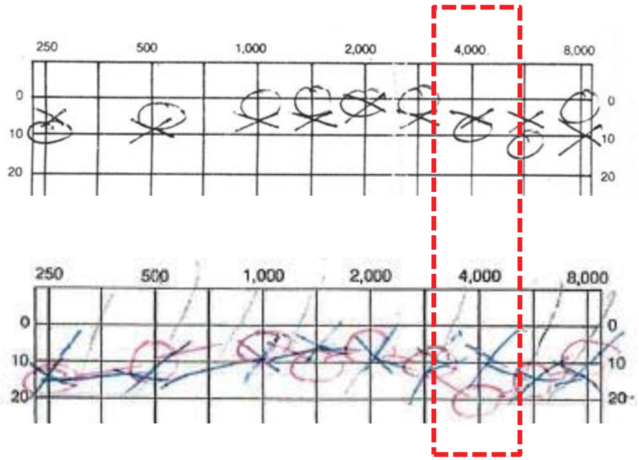
Change in Hearing Threshold



My audiogram ...



My audiogram ...



Age 26

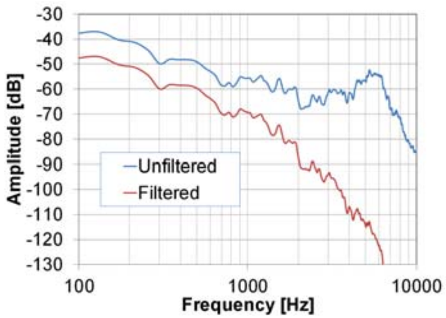
Age 40 ?

More hearing loss on right ear…
Audiologist said probably caused by driving with window down!

Demonstration

- Simulation of degradation in ability to understand speech.
- "I went down to the beach and sat on a chair and decided to eat a refreshing peach. Suddenly the sun shone on my shoes and I thought, I would like to teach people about acoustics. I went for a walk to Grand Central Station and caught a train."

Unfiltered Filtered



Noise Induced Hearing Loss

- What happens?
- Speech recognition requirements
  - frequencies 500 - 2000Hz critical – have a look at the graph
  - "beach", "teach", "peach".
  - Quote from Madagascar movie - Alex the Lion: "Did he just say Grand Central Station, or my aunt's constipation?"
- Loss occurs first between 4kHz and 6kHz
- Short exposure - temporary loss (temporary threshold shift) ... been to a loud nightclub?
  - complete recovery if not too severe
  - permanent loss if exposure severe or repeated sufficiently often
  - gradually spreads to lower and higher frequencies (than 4-6 kHz)

Noise Induced Hearing Loss

- Subjective perception of loss
  - some loss of hearing for high pitched squeaks in machinery
  - difficult to understand speech in a noisy environment
- Hearing damage risk
  - cannot protect everyone over entire audio range
  - aim is to protect most individuals against loss for everyday speech
    - arithmetic average of loss at 500, 1000 and to 2000 Hz used in USA
    - in Australia weighted average of loss in range 500 to 6000 Hz is used
    - arithmetic average loss of 25dB in specified frequency range defines beginning of significant loss
    - loss of 92 dB corresponds to total loss
- Risk can be calculated using ISO 1999 standard. Alternative formulations described in text book are not widely used.

ISO 1999 Damage Risk Assessment

- calculate normalised 8 hour equivalent noise level $L_{Aeq,8h}$

$$L_{Aeq,8h} = 10\log_{10}\left[\frac{1}{8}\int_0^T 10^{L_A(t)/10}\,dt\right]$$

  - in practice this may be estimated using a sound level meter
  - measure noise level at locations where employee spends time

$$L_{Aeq,8h} = 10\log_{10}\frac{1}{8}\left[t_1 10^{L_{1A/10}} + t_2 10^{L_{2A/10}} + \ldots + t_m 10^{L_{mA/10}}\right]$$

t_i = time in hours at location i

L_{iA} = sound pressure level (A-weighted) at location i

ISO 1999.2 (cont.)

- Hearing loss H′ (or Hearing Level) is

$$H' = H + N - HN/120$$

ENC p149

H = loss due to age
N = noise induced loss

- 50% fractile means that 50% of population will suffer a loss equal or in excess of it

$$N_{50} = \left(u + v\log_{10}\Theta\right)\left(L_{Aeq,8h} - L_0\right)$$

Θ = exposure time in years
u, v, L_o from Table 1

ISO 1999.2 (cont.)

- Table 1. Values of the coefficients u, v and L_o used to determine the NIPTS for the median value of the population, $N_{o,50}$

Frequency Hz	*u*	*v*	L_0 dB
500	-0.033	0.110	93
1000	-0.02	0.07	89
2000	-0.045	0.066	80
3000	+0.012	0.037	77
4000	+0.025	0.025	75
6000	+0.019	0.024	77

ENC p149

ISO 1999.2 (cont.)

- Other fractiles N_Q from
  - k from Table 2

$$N_Q = N_{50} + kd_u \quad 5 < Q < 50$$

$$N_Q = N_{50} - kd_L \quad 50 < Q < 9$$

$$d_u = \left(X_u + Y_u\log_{10}\Theta\right)\left(L_{Aeq,8h} - L_o\right)^2$$

$$d_L = \left(X_L + Y_L\log_{10}\Theta\right)\left(L_{Aeq,8h} - L_o\right)^2$$

where X_L Y_L X_u Y_u from Table 3.

- For exposure times less than 10 years

$$N_{50} = \frac{\log_{10}(\Theta+1)}{\log_{10}(11)} N_{50}; \quad \Theta \le 10$$

Θ = exposure time in years

ENC p150

ISO 1999.2 (cont.)

- Table 2. Values of the multiplier k for each fractile Q.

Q		*k*
0.05	0.95	1.645
0.10	0.90	1.282
0.15	0.85	1.036
0.20	0.80	0.842
0.25	0.75	0.675
0.30	0.70	0.524
0.35	0.65	0.385
0.40	0.60	0.253
0.45	0.55	0.126
0.50		0.0

ENC p150

ISO 1999.2 (cont.)

- Table 3. Constants for calculating N_Q fractiles.

Frequency (Hz)	X_u	Y_u	X_L	Y_L
500	0.044	0.016	0.033	0.002
1000	0.022	0.016	0.020	0.000
2000	0.031	-0.002	0.016	0.000
3000	0.007	0.016	0.029	-0.010
4000	0.005	0.009	0.016	-0.002
6000	0.013	0.008	0.028	-0.007

ENC p151

ISO 1999.2 (cont.)

Threshold shift H_{50}, for the 50% fractile due to age alone:

$$H_{50} = a(Y - 18)^2$$

Y=age in years

$H_Q=H_{50}+kS_u$ 5 < Q < 50

$H_Q=H_{50}- kS_L$ 50 < Q < 95.

$S_u=b_u + 0.445\ H_{50}$

$S_L=b_L + 0.356\ H_{50}$

constants a, b_u, b_L in Table 4.

ISO 1999.2 (cont.)

- Table 4. Values of the parameters and used to determine respectively the upper and lower parts of the statistical distribution H_Q.

Frequency	Value of b_u		Value of b_L		Value of a	
Hz	Males	Females	Males	Females	Males	Females
125	7.23	6.67	5.78	5.34	0.0030	0.0030
250	6.67	6.12	5.34	4.89	0.0030	0.0030
500	6.12	6.12	4.89	4.89	0.0035	0.0035
1000	6.12	6.12	4.89	4.89	0.0040	0.0040
1500	6.67	6.67	5.34	5.34	0.0055	0.0050
2000	7.23	6.67	5.78	5.34	0.0070	0.0060
3000	7.78	7.23	6.23	5.78	0.0115	0.0075
4000	8.34	7.78	6.67	6.23	0.0160	0.0090
6000	9.45	8.90	7.56	7.12	0.0180	0.0120
8000	10.56	10.56	8.45	8.45	0.0220	0.0150

Allowable $L_{Aeq,8h}$

- Currently almost universally **85 dB(A) for 8 hours**
- Australia and Europe
  - all noise exposure is counted towards daily noise dose, irrespective of the level
  - all noise levels (even below 85 dB(A)) must be used to calculate $L_{Aeq,8h}$
- USA - OHSA and military exclude levels below 80 dB(A) in calculation of $L_{Aeq,8h}$.

Allowable $L_{Aeq,8h}$

Relevant Standards:

- National Code of Practice for Noise Management and Protection of Hearing at Work - 3rd Edition
  http://www.safeworkaustralia.gov.au/sites/swa/about/publications/pages/cp2004noisemanagementandprotectionofhearing
- National Standard for Occupational Noise
  http://www.safeworkaustralia.gov.au/sites/swa/about/publications/pages/ns2000occupationalnoise
- Australian Standards: AS/NZS 1269:2005 Occupational noise management

Trading Rules for Non-continuous Exposure

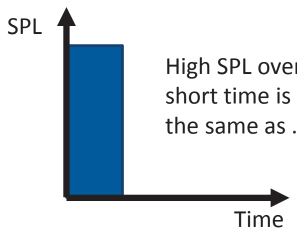
- ***Trading*** rules for non-continuous exposure
  - defined as the decibel increase in noise level which is equivalent to an exposure time increase by a factor of 2
  - In Australia (and Europe) **3 dB(A) rule:** means that **halving** *exposure time* is equivalent to **increasing** *exposure level by 3 dB*
- Examples:
  - 8 hours at 85 dB(A) is same noise dose as 4 hours at 88 dB(A)

$L_{Aeq,8hr}$	Time (hours)
91	2
88	4
85	8
82	16

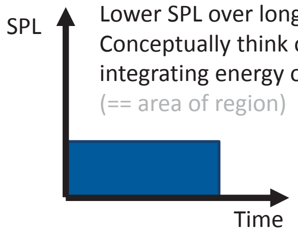
This is a good "sanity check" for your calculations

Trading Rules for Non-continuous Exposure

Think of these noise exposure problems as ears getting cumulative energy (damage) = (pressure)^2 x time and can be "**traded**" or exchanged.
The analogy I like to use is "the area under the graph".



High SPL over short time is the same as ...



Lower SPL over longer time... Conceptually think of it as integrating energy over time (== area of region)

... the areas under the graphs are the same.

Trading Rules for Non-continuous Exposure

- Defined as the decibel increase in noise level which is equivalent to an exposure time increase by a factor of 2.
- Equation for equivalent sound level can be rewritten more <u>generally</u> as

$$L'_{Aeq,8h} = \frac{10}{n}\log_{10}\left\{\frac{1}{8}\int_0^T \left[10^{L_A(t)/10}\right]^n dt\right\}$$

ENC p156

where the value of n is related to the trading rule, L in dB, by $n = 3/L$

- Introducing a base level, L_B which $L'_{Aeq,8h}$ should not exceed, previous equation can be written as

$$L'_{Aeq,8h} = \frac{L}{0.301}\log_{10}\left\{\frac{1}{8}\int_0^T \left[2^{\left(L_{pA}(t)-L_B\right)/L}\right] dt\right\} + L_B$$

ENC p157

Trading Rules for Non-continuous Exposure

Example: how much time at 93 dB(A) is equivalent to 1 hour at 90 dB(A)?

- Australia and Europe - 3 dB(A) rule
  - means that halving exposure time is equivalent to increasing exposure level by 3 dB.
  - 90 dB(A) for 1hr = 93 dB(A) for 1/2 hour.

- USA - OHSA 5 dB trading rule
- USA - Military 3 dB trading rule

Trading Rules for Non-continuous Exposure

- Maximum allowed exposure time (L is trading rule)

$$T_a = 8 \times 2^{-\left(L'_{Aeq,8h} - L_B\right)/L}$$

ENC p157

If the number of hours of exposure is different to 8, then to find the actual allowed exposure time to the given noise environment, the “8” in the above equation is replaced by the actual number of hours of exposure.

Trading Rules for Non-continuous Exposure

- Daily noise dose (DND) $$DND = 2^{\left(L'_{Aeq,8h} - 90\right)/L}$$

ENC p158

- Australian regulations once used L_B = 85dB(A) for DND calculations – no longer used. Hence, won’t be using it in this course, but included for reference only.

Exposure for Number of Hours Different Than 8 Hours Per Day

- Example: 85 dB(A) for 16 hours per day

$$L'_{Aeq,8h} = \frac{3}{0.301}\log_{10}\frac{1}{8}\left[16 \times 2^{(85-85)/3}\right] + 85\text{dB(A)} = 88 \ \text{dB(A)}$$

- allowable exposure time

$$T_a = 16 \times 2^{-\left(L'_{Aeq,8h} - L_B\right)/L} = 8 \ \text{hours}$$

Note that the "8" in the above equation has been replaced by the actual number of hours of exposure.

Question …

An employee working at a bar in a night club is exposed to 105 dB(A) for a 4 hour shift.

(a) What is the A-weighted Sound Exposure?

(b) What is the 8-hour equivalent level?

(c) How much would the shift length have to be reduced to comply with the 85 dB(A) 8 hour requirement?

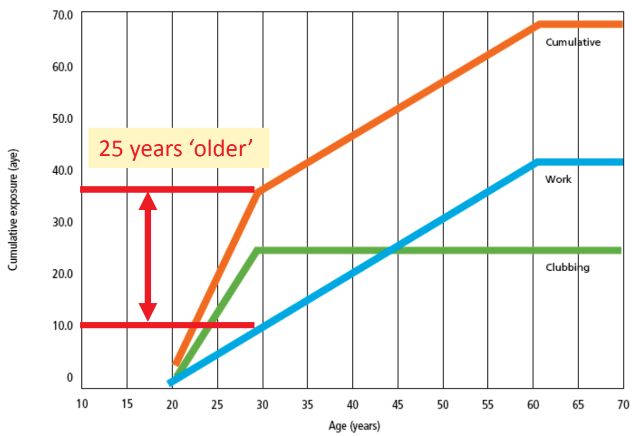
Comments …

- Research conducted by The National Acoustic Laboratories in Australia measured noise levels at nightclubs.
- **The average noise level for the clubs visited was 98 dBA, and measurements ranged from 91 dBA to 106 dBA** throughout the time period from 9pm to 3am.
- For every hour past 9pm, the average level of noise consistently rose by 4 dB.

Williams, W., Beach, E. & Gilliver, M., "Clubbing – the cumulative effect of noise exposure from attendance at dance clubs and nightclubs on whole-of-life noise exposure", Noise and Health. DOI: 10.4103/1463-1741.64970

Comments …

Graph Five: The cumulative effect of noise exposure from work, clubbing and a combination of both



The clubber who attends clubs on a weekly basis for 10 years has, at their 30th birthday, ears that are **25 years 'older'** (in terms of accumulated noise exposure) than their non-clubbing equivalent

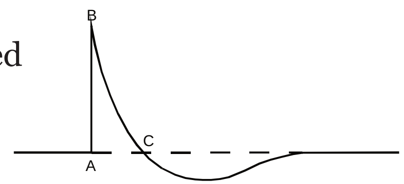
Protect your ears! Wear ear plugs.

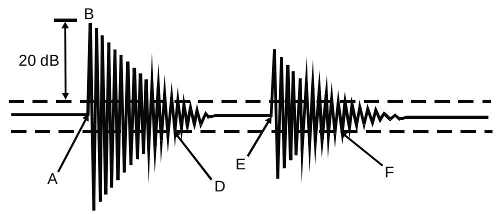
AYE = Allowable Yearly Exposure, based on acceptable workplace levels

National Acoustics Laboratory, Binge Listening report, p7-8.

Impulsive Noise

- Very short duration characterised by a shock front.
- Usually only people exposed to explosions e.g. gunshots, quarry blasting.
- 10's or 100's per exposure
- B-duration is time for peak to decay by 20 dB



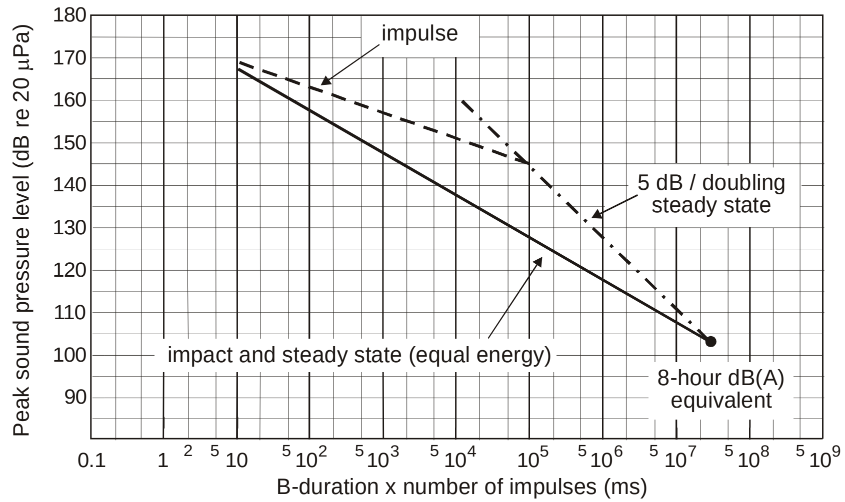


ENC p160

Impact Noise

- Produced by non-explosive means e.g. metal-to-metal impacts.
- Characteristic shock front is not always present as often occurs in a reverberant industrial environment.

Impulse and Impact Damage Risk



ENC p162

Impulse and Impact Damage Risk

- IMPULSE NOISE (explosive phenomenon) Daily Noise Dose (DND)

$$DND = \frac{\text{no. of impulses}}{\text{allowed number}}$$

- IMPACT NOISE (non-explosive noise)

$$DND = \frac{\text{no. of impacts}}{\text{allowed number}} = 2^{(L_{peak} - L_{allowed})/3}$$

Speech Interference Criteria

- There are important frequency ranges for oral communication. (see table on next page)
- Background noise level can make it difficult to understand speech.
- Speech level of a talker depends on the talker’s subjective response to the level of the background noise.
  - If you go to a loud nightclub, you raise your voice to be communicate with your friends.

ENC p164

Speech Interference Criteria

Table 5. Significant frequency ranges for speech communication

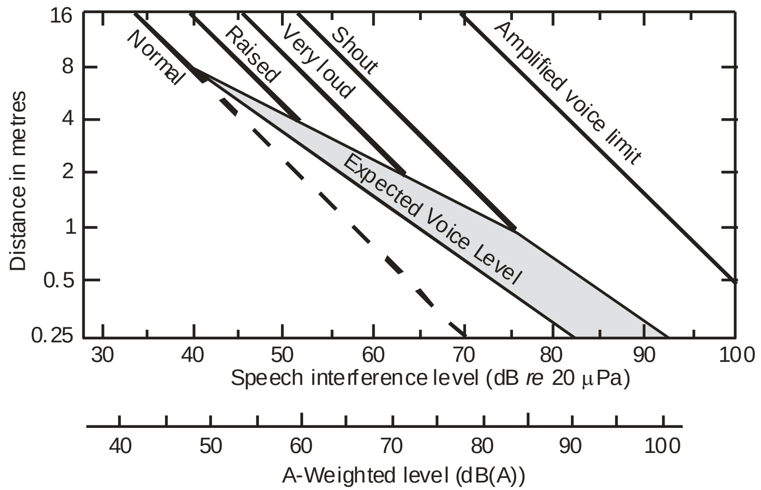
ENC p165

	Approximate frequency range (Hz)
Range of hearing	20 to 18,000
Speech range	200 to 6,000
Speech intelligibility (containing the frequencies most necessary for understanding speech)	500 to 4,000
Speech privacy range (containing speech sounds which intrude most objectionably into adjacent areas)	250 to 2,500
Male voice (peak frequency of energy output)	350
Female voice (peak frequency of energy output)	700

Speech Interference Criteria

- See chart (next page) in ANSI S12.65-2006 - For Rating Noise with Respect to Speech Interference.
- Originally devised from measurements outdoors for speech communication.
- Curves for four voice levels are shown: *normal*, *raised*, *very loud*, and *shout*.
- The shaded triangular region shows the range of *expected voice levels* due to the normal raising of one's voice in noisy surroundings

Speech Interference Criteria



See also ANSI S12.65-2006, page 4

ENC p165

Speech Interference Criteria

- The first four of the voice levels listed above may be obtained by placing the talker in quiet surroundings and instructing him to use one of these four levels.
- If, however, the talker is in the same noisy surroundings as the listener he will automatically raise his voice in attempting to communicate with the listener.
- The increase in voice level for most persons is between 3 and 5 dB for every 10 dB increase in speech interference level above 40 dB until the maximum attainable vocal effort is reached.

Speech Interference Criteria

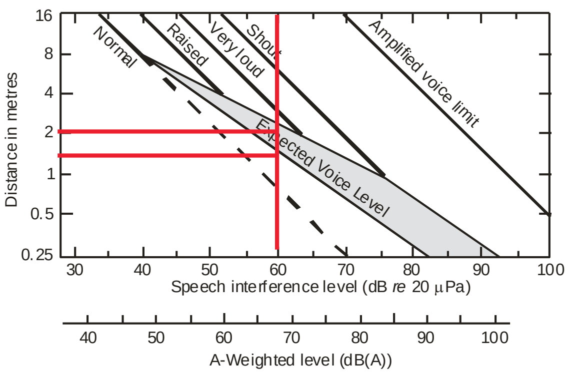
- For communication indoors, where reflections off walls or other surfaces reinforce the speech signal, the distance for just-reliable communication may be larger than that shown in the chart, provided the indoor space is not highly reverberant.
- For example, in a typical office an increase in talker-listener distance beyond about 1 meter does not necessarily result in a decrease in the speech level at the listener's ear.

Speech Interference Criteria

- Speech Interference Level (SIL) is the *arithmetic average* of the background SPL (in dB) in the four octave bands 500Hz, 1000 Hz, 2000 Hz, 4000Hz.
- Expected Voice Level region is the voice level that the talker would automatically use as a result of the background noise level.
  - decreases by 4 dB if talker wearing ear muffs.
- Generally acceptable criterion for foreman's offices, control rooms, etc. is 70 dB(A) to allow telephone comms.

Example

- Average background noise level 60 dB.
- At 1.5 m – Normal speech level
- At 2.0m – Raise voice needed.



Question …

- What voice level is **expected** (give a range) and **required** if a talker is to make himself understood to someone 3 metres away in a noise environment of 75dB(A)?

Speech Interference - Intense Tones

- Speech range is generally 200 Hz to 6000 Hz.
- Masking effect is best when on sounds that have a higher frequency than the masking tone.
  - i.e. low frequency tones can be effective in masking speech.
  - Most effective masking for speech is 500 Hz tone rich in harmonics.

- Masking will covered more in the section on *Psychoacoustics*.

Speech Privacy

- Needed for conversation privacy in offices.
- Generally, the higher the background noise levels from air conditioning and other mechanical equipment, the less one has to worry about speech privacy and the more flimsy can be the office partitions.
- Likely to be a problem in a building with no air conditioning or forced ventilation systems.

ENC p179

Speech Privacy

- To avoid speech privacy problems:-
  - Use partitions or separation walls with an adequate sound insulation (see Table).
  - Ensure that there are no air gaps between the partitions and the permanent walls and floor
  - Ensure that the partitions extend all the way to the ceiling, roof or the underside of the next floor
  - Ensure that acoustic tiles with absorption coefficients of at least 0.1 to 0.4 are used for the suspended ceiling to reduce the reverberant sound level in the office spaces, and also to increase the overall noise reduction of sound transmitted from one office to another.

Speech Privacy

- If speech privacy is a problem, then there are two alternative approaches possible:
  - increase the sound insulation of the walls (for example, by using double stud instead of single stud walls so that the same stud does not contact both leaves)
  - make sure that construction has been carried out properly (ie good mortar in brick walls and no gaps between brick and plaster board)
  - add acoustic ‘perfume to the corridors and offices adjacent to those where privacy is important.

Speech Privacy

Speech privacy noise insulation requirement.

ENC p180

Sound as heard by occupant	Average sound insulation [a] plus ambient noise (dB(A))
Intelligible	70
Ranging between intelligible and unintelligible	75-80
Audible but not obtrusive (unintelligible)	80-90
Inaudible	90

[a] Average sound insulation is the arithmetically averaged sound transmission loss over the 1/3 octave bands from 100 Hz to 3150 Hz.

Psychological Effects Of Noise

- long term exposure to excessive levels can cause
  - permanently elevated blood pressure
  - stress
  - reduced work efficiency, resulting in tense, jittery workers
- optimal noise level depends on task
  - simple tasks should be accompanied with higher noise levels to maintain worker arousal
  - complex tasks - quiet environment needed

Noise Level Goals

Example of guideline noise levels in a factory

Type of Room	Guideline Highest Sound Level dB(A)
Conference room	35
Office	40
Workshop office, rest room	45
Laboratory, measurement or inspection room	50
Canteen	50
Changing room	55
Repair workshop	60
Production areas	75
Fan room, compressor room etc. normally unmanned	90

Noise Criteria

- Commonly used criteria in acoustics :

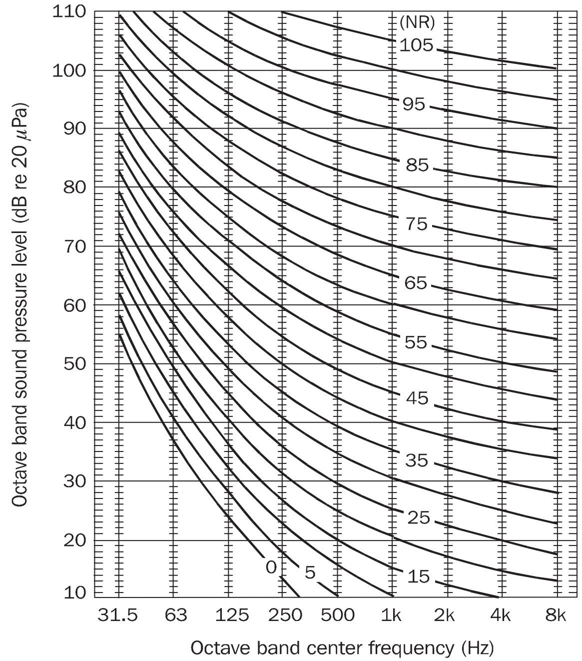
Criteria	Description
NCB curves	office and commercial buildings
RC curves	office and commercial buildings
NR curves	community noise - includes noise character - single number descriptor
dB(A)	community noise

- Noise weighting curves are better than dB(A) for specifying acceptable noise limits in offices, as they take into account frequency content of the noise

Noise Rating (NR) Curves

- used for community noise specification
- To determine NR ratings plot octave band levels on NR chart NR level is highest curve which envelopes data.

ENC p169



NR Curves

The NR index is the value of that curve which lies just above the spectrum of the measured noise.

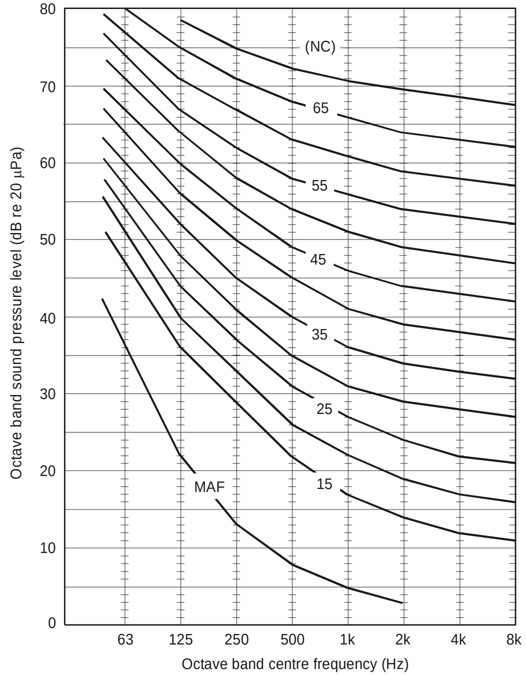
ENC p170

Noise Criteria (NC) Curves

- Used for interior noise specification in building industry.
  - Largely superseded by NCB curves.
- A number of problems
  - do not account for low frequency rumble noise
  - too permissive at 2000 Hz and higher
  - do not correlate well with subjective response to air con noise
- To determine NC ratings plot octave band levels on NC chart NC level is highest curve which envelopes data.

ENC p171

NC Curves



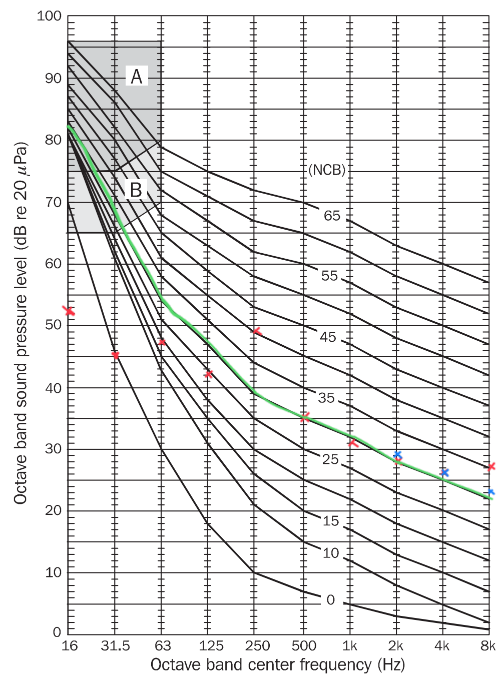
ENC p171

Balanced Noise Criteria (NCB) Curves

- Better than NC for interior noise specification but more difficult to use.
- Designation number is the SIL of a noise (arithmetic average of 500 Hz, 1 kHz, 2 kHz, and 4 kHz octave bands).
- Rumbly if any values in 500 Hz band or lower exceed the NCB curve by more than 3 dB.
- Hissy if NCB curve of best fit between 125 Hz and 500Hz is exceeded by any band levels between and incl. 1 kHz and 8 kHz.

ENC p175

ENC p175



NCB Curves
Balanced Noise Criterion Curves are used to specify acceptable background noise levels in occupied spaces. More detailed information on NCB curves may be found in American National Standard ANSI S12.2-1995, "Criteria for Evaluating Room Noise". The designation number of an NCB curve is equal to the Speech Interference Level (SIL) of a noise with the same octave band levels as the NCB curve.

The SIL of a noise is the arithmetic average of the 500 Hz, 1 kHz, 2 kHz and 4 kHz octave band levels.

To determine whether the background noise is "rumbly", plot the octave band sound levels of the measured noise on a chart containing a set of NCB curves. If any values in the 500 Hz octave band or lower exceed by more than 3dB the curve corresponding to the NCB rating of the noise, then the noise is labelled "rumbly".

To determine if the noise is "hissy", the NCB curve which is the best fit of the octave band sound levels between 125Hz and 500Hz is determined. If any of the octave band sound levels between 1000Hz and 8000Hz inclusive exceed this curve, then the noise is rated as "hissy".

Question ....

- On socrative.com, Room b069a2aa
- SPL is measured as

Freq [Hz]	63	125	250	500	1000	2000	4000	8000
SPL [dB]	48	48	43	38	30	20	16	12

- What is the NCB rating?
  - NCB = 48
  - NCB = 37
  - NCB = 35
  - NCB = 26

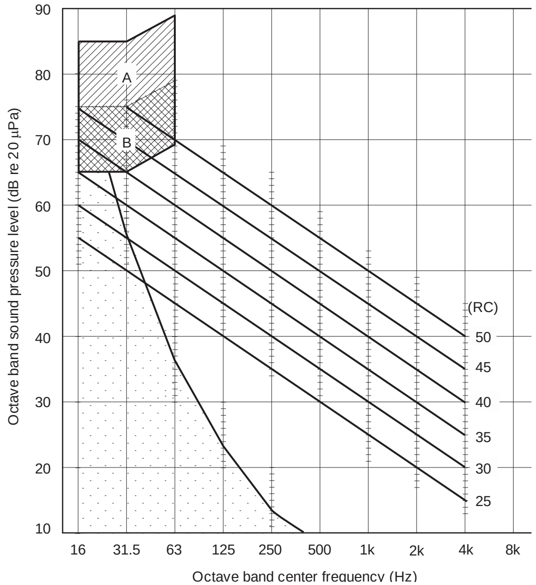
Room Criteria (RC) Curves

- Originally called "Revised Criteria"
- Basic was measurements made in 68 offices in the 1970s where noise levels from HVAC were judged as satisfactory.
  - Is it still applicable 40 years later ?
- Part of standard "ANSI/ASA S12.2-2008 Criteria for Evaluating Room Noise"

ENC p172

See Beranek and Ver (2006), Noise and Vibration Control Engineering, 2nd ed., p895.

Room Criteria (RC) Curves

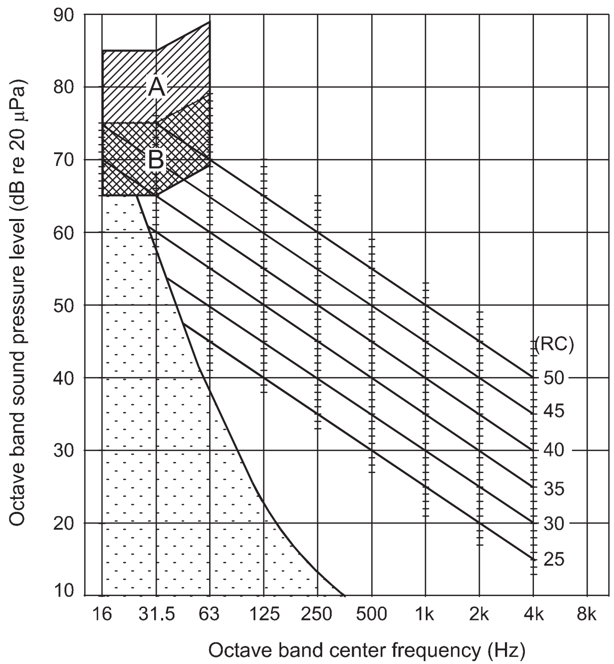


Room Criterion (RC) curves.
Regions A and B: noise-induced feelable vibration in lightweight structures and induced rattles in light fixtures, doors, windows, etc.
Region A: High probability.
Region B: Moderate probability.

ENC p172

RC Rating

- RC rating = average of the 500 Hz, 1000 Hz and 2000 Hz octave band sound levels.
- If any octave band level below 500Hz exceeds the RC curve by more than 5 dB, noise is denoted "rumbly" (e.g. RC 29(R)).
- If any octave band level above 500Hz exceeds the RC curve by more than 3 dB, noise is denoted, "hissy". (e.g. RC 29(H)).
- If neither of the above occurs, the noise is denoted "neutral" (e.g. RC 29(N)).
- If the sound pressure levels in any band between and including 16 Hz to 63 Hz lie in the cross hatched regions, perceptible vibration can occur in the walls and ceiling and rattles can occur in furniture. In this case, the noise is identified with "RV" (e.g. RC 29(RV)).
- RC curves are suitable for specifying the introduction of acoustic "perfume", and noise with a spectrum of that shape has been found to be the least objectionable.

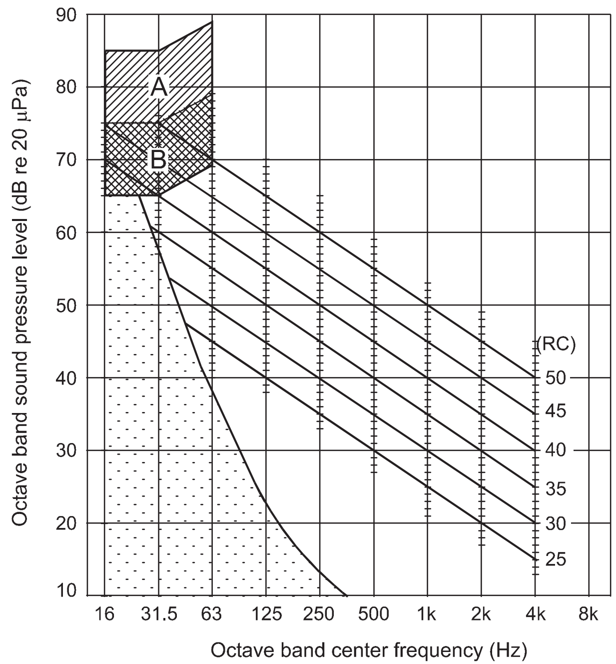


RC Curves
RC curves are used solely to evaluate air conditioning noise.
The RC number is the average of the 500Hz, 1000Hz and 2000Hz octave band sound levels.
RC curves represent the shape of the least objectionable noise spectrum.

If any octave band level below 500Hz exceeds the RC curve by more than 5dB, the noise is denoted "rumbly".

If any octave band level above 500Hz exceeds the RC curve by more than 3dB, the noise is denoted, "hissy".

If neither of the above occurs, the noise is denoted "neutral".



RC Curves

If the SPL is in regions A and B , then it is probably that the noise will induce vibration in lightweight structures such as doors and windows.

Region A: high probability

Region B: moderate probability

The RC curves are only defined up to RC 50.

Example Problem:

The sound pressure level in a room is measured in octave bands as

Freq [Hz]	31.5	63	125	250	500	1000	2000	4000
SPL [dB]	73	57	52	45	40	35	30	35

a) What is the RC rating of the room?
b) Is it rumbly, hissy, or neutral ?
c) Is it likely that the windows will rattle?

Example Problem:

Answer:

First calculate the average:

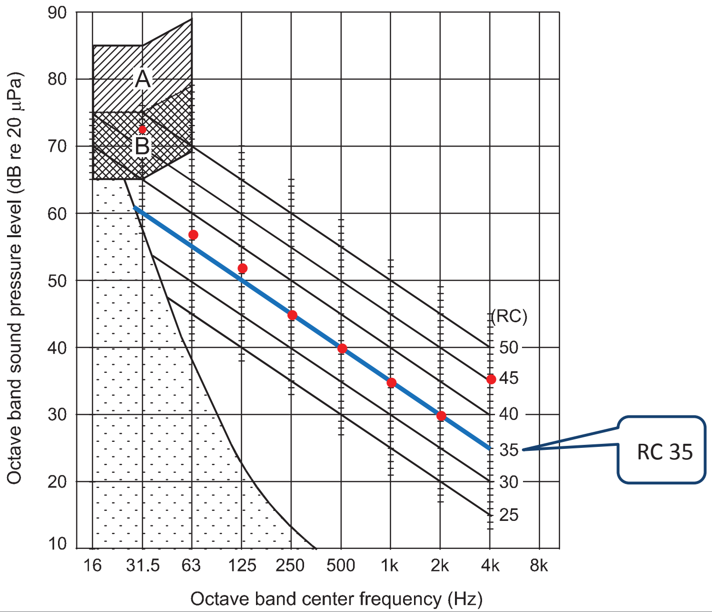
> "The RC number is the average of the 500Hz, 1000Hz and 2000Hz octave band sound levels."

Freq [Hz]	31.5	63	125	250	500	1000	2000	4000
SPL [dB]	73	57	52	45	40	35	30	35

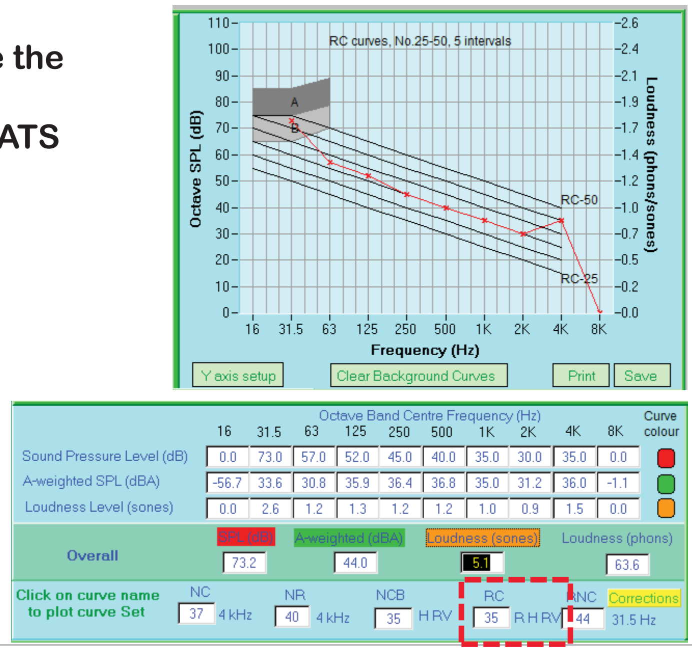
The average is: $[40 + 35 + 30]/3 = 35$

... so the RC rating is 35.

Freq [Hz]	31.5	63	125	250	500	1000	2000	4000
SPL [dB]	73	57	52	45	40	35	30	35



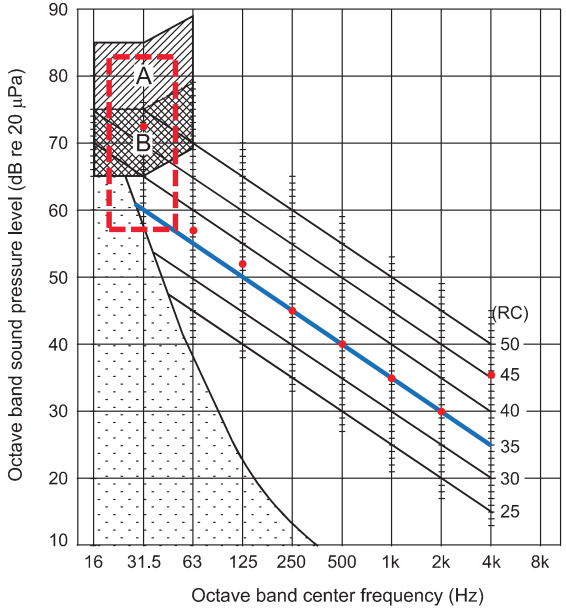
OR .. Use the software ENC in CATS



Is it rumbly, hissy, or neutral ?

If any octave band level below 500Hz exceeds the RC curve by more than 5dB, the noise is denoted “rumbly”.

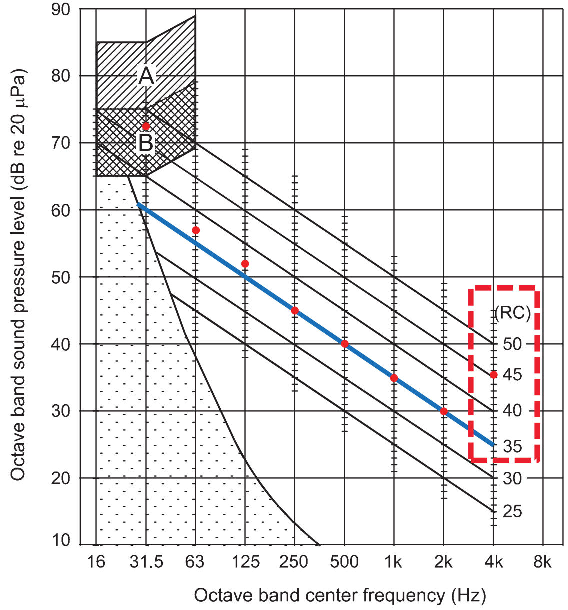
Yes, look at the 31.5 Hz band.



Is it rumbly, hissy, or neutral ?

If any octave band level above 500Hz exceeds the RC curve by more than 3dB, the noise is denoted, “hissy”.

Yes, look at the 4 kHz band.



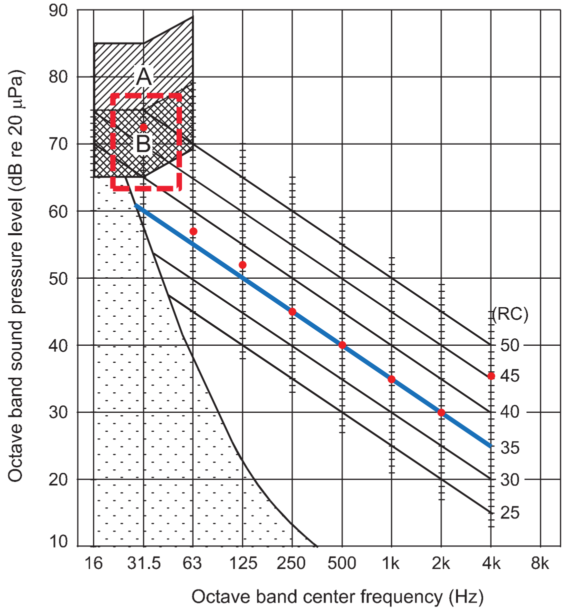
Is it likely that the windows will rattle?

If the SPL is in regions A and B , then it is probably that the noise will induce vibration in lightweight structures such as doors and windows.

Region A: high probability

Region B: moderate probability

Yes, moderate probability.



Noise Specification Comparison

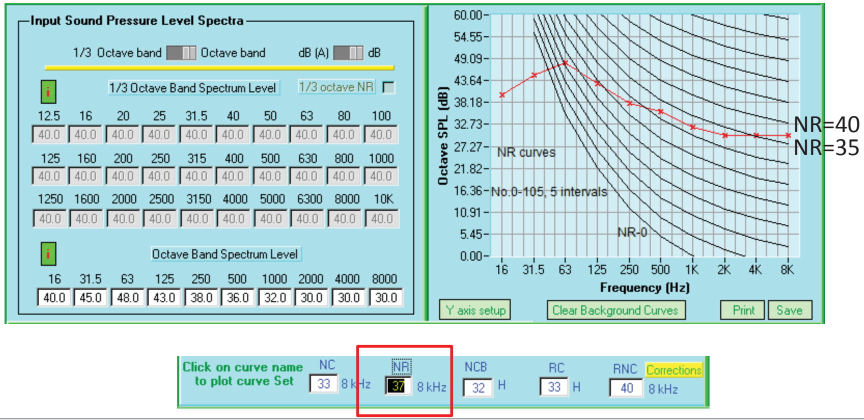
ENC p178

	Specification			
dB(A)	NR	NC	NCB	Comment
25-30	20	20	20	very quiet
30-35	25	25	25	
35-40	30	30	30	quiet
40-45	35	35	35	
45-50	40	40	40	moderately
50-55	45	45	45	noisy
55-60	50	50	50	noisy
60-65	55	55	55	
65-70	60	60	60	very noisy

- rough guide only - exact relationship depends upon frequency spectrum of noise

Previous "Exit Ticket" Question

- What does Tangency Method mean?
  - Means the curve that envelopes all the data
    Example using ENC software



Community Noise

Community Noise

- basis for regulation - dB(A)
  - usually 40 dB(A) + corrections for special conditions
- for fluctuating noise, determine equivalent L_{Aeq}
- recommendations in standards intended only as a guide
  - noise regulations used for enforcement

ENC p180

Adjustments to base level of 40 dB(A)
(adapted from Australian Standard AS1055).

Character of the sound Choose if applicable	
Tones or impulsive noise readily detectable	-5
Tones or impulsive noise just detectable	-2
Time of day Choose if applicable	
Evening (6 pm to 10 pm)	-5
Night time (10 pm to 6 am)	-10
Early morning (6 am to 7 am)	-5
Neighbourhood Select one from here	
Rural and outer suburban areas with negligible traffic	0
General suburban areas with infrequent traffic	+5
General suburban areas with medium density traffic or suburban areas with some commerce or industry	+10
Areas with dense traffic or some commerce or industry	+15
City or commercial areas or residences bordering industrial areas or very dense traffic	+20
Predominantly industrial areas or extremely dense traffic	+25

Estimated public reaction to noise when the adjusted measured noise level exceeds the acceptable noise level (see above table for adjustments to base level 40 dB(A))

Amount in dB(A) by which adjusted measured noise level exceeds the acceptable level	Public reaction	Expression of public reactions in a residential situation
0-5	Marginal	From no observed reaction to sporadic complaints
5-10	Little	From sporadic complaints to widespread complaints
10-15	Medium	From sporadic complaints to threats of community action
15-20	Strong	From widespread complaints to threats of community action
20-25	Very Strong	From threats of community action to vigorous community action
25 and over	Extreme	Immediate direct community and personal action

Question:

- Noise in the community is described as tonal
- Time of day is 9pm
- The place is in the suburbs with low traffic volume.

- The measured SPL was 45 dB(A) re 20 micro-Pa. What would be the community response?

Answer:

• Base Level	40 dB
• Character of Sound: Tones or impulsive noise just detectable	
• Time of Day: Evening (6 pm to 10 pm)	
• Neighbourhood: General suburban areas with infrequent traffic	
TOTAL:	

Measured SPL was 45 dB(A) re 20 micro-Pa.
What would be the community response?

Environmental Noise Surveys

- If a new development will take place, usually have to measure the existing environment noise level.
  - To be able to show before / after noise level.

- Examples: factories, wineries, wind farms (!).

- Measurement locations are usually at the closest residential housing.

Environmental Noise Surveys

- The overall noise impact of an industry on the surrounding community depends on the number of people exposed to noise levels.
- Total Weighted Population (TWP) is

$$TWP = \sum_i W_i P_i$$

ENC p185

- where P_i = Number of people associated with a weighting factor W_i, which depends on the day-night average sound level L_{dn}.

Environmental Noise Surveys

Annoyance weighting factors corresponding to values of L_{dn}

Range of L_{dn} (dB)	W_i
35–40	0.01
40–45	0.02
45–50	0.05
50–55	0.09
55–60	0.18
60–65	0.32
65–70	0.54
70–75	0.83
75–80	1.20
80–85	1.70
85–90	2.31

Environmental Noise Surveys

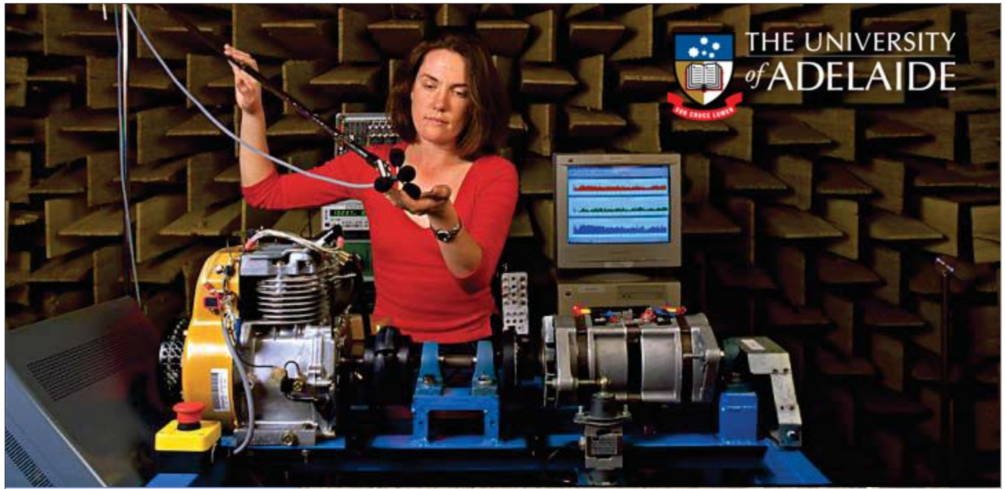
The relative impact of one particular noise environment may be compared with another by comparing the *Noise Impact Index* for each environment defined as:

$$NII = \frac{TWP}{\sum_i P_i}$$

ENC p185

Summary

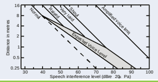
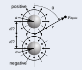
- Noise Descriptors
- Noise Criteria
- Noise Induced Hearing Loss and Criteria
- Speech Interference
- Building Acoustics Criteria
- Environmental Noise Criteria
- Environmental Noise Surveys



Carl Howard

Psychoacoustics

Course Content – Where Are We?

TOPIC	
The Ear	
Criteria	
Psychoacoustics	
Sound Sources	
Sound Power	

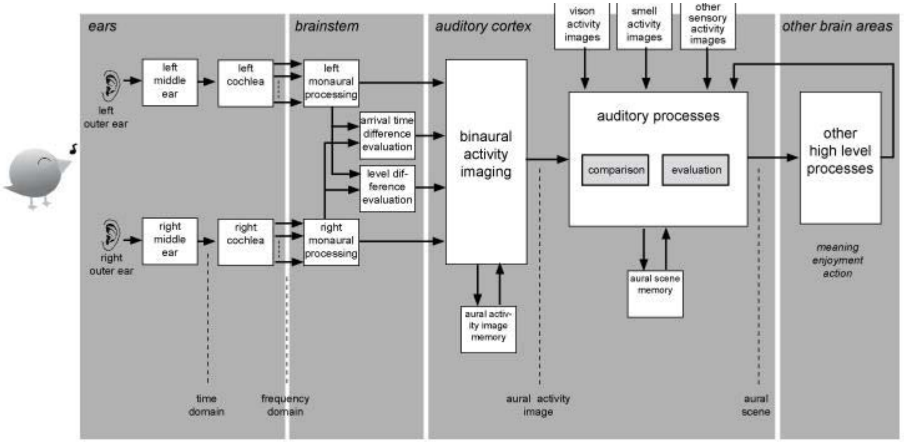
Contents

- Introduction
- Loudness
- Pitch
- Masking
- Critical bands, bandwidth, and rate
- Sharpness
- Roughness
- Binaural hearing
- Annoyance
- Jury testing

Learning Outcomes

- Understand the psychoacoustic terms and able to calculate the metrics for loudness, pitch, critical bands bandwidth rate, sharpness, roughness, annoyance.
- Understand the concept of masking.

Many processes involved in hearing



http://www.yamahaproaudio.com/global/en/training_support/selftraining/audio_quality/chapter4/03_auditory/

Questions to Think About

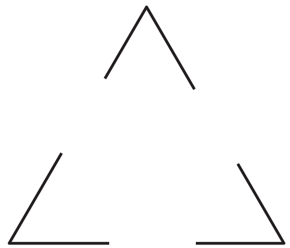
- How is it possible to describe the characteristics of a sound?
- Although two sounds have the same loudness, they can sound completely different. How do you differentiate between the sounds?
- What constitutes a pleasant and appealing sound?
- What is meant by “sound quality” ?
  - The term “quality” can have several interpretations.

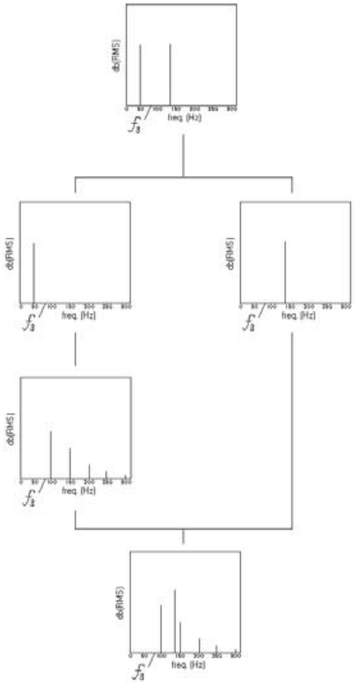
Introduction

Example applications:
- Acoustic trickery
  - False bass, virtual pitch, pitch circularity
- Assessment of loudness
  - Actual measured sound level and perceived loudness
- Sound quality:
  - that sounds good
- Localisation:
  - where is that sound coming from?
- Hearing aids
- Sound design:
  - what is an appropriate sound for taking a picture with a camera?
- Speech recognition:
  - can you hear what I am saying in this background noise?

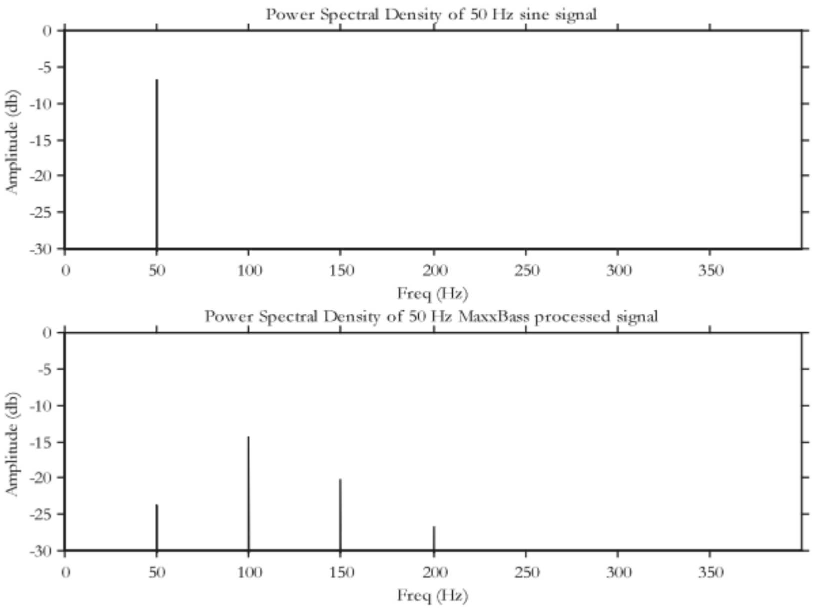
Acoustic Trickery

- False Bass
  - If the fundamental is removed, your brain fills in the missing fundamental.
  - Used in products for bass enhancement.
  - Visual analogy:





The Effect of the MaxxBass Psychoacoustic Bass Enhancement System on Loudspeaker Design



The Effect of the MaxxBass Psychoacoustic Bass Enhancement System on Loudspeaker Design

False Bass

Loudspeakers Duntech PCL-10

SRS WOW
software bass enhancement
http://www.dts.com/audioessentials/
Included in Windows Media Player

MaxxBass by Waves
http://www.waves.com/plugins/maxxbass

Demonstration - Missing Fundamental

Windows Media Player has inbuilt SRS Trubass

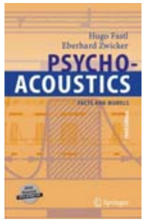
Introduction

- Psychoacoustics is a whole course in itself!
- This lecture will only provide a very brief overview of some of the concepts that are relevant in the field.

Introduction

References

- Hugo Fastl and Eberhard Zwicker (2007), "Psychoacoustics Facts and Models," Springer, 3rd edition.
  Free access to e-book at:
- http://adelaide.hosted.exlibrisgroup.com/SUA:ALMA51139397530001811
- http://link.springer.com.proxy.library.adelaide.edu.au/book/10.1007/978-3-540-68888-4/page/1
- All audio files are on MyUni.



- Designing for Product Sound Quality
  http://adelaide.hosted.exlibrisgroup.com/SUA:ALMA2191997010001811

Introduction

References

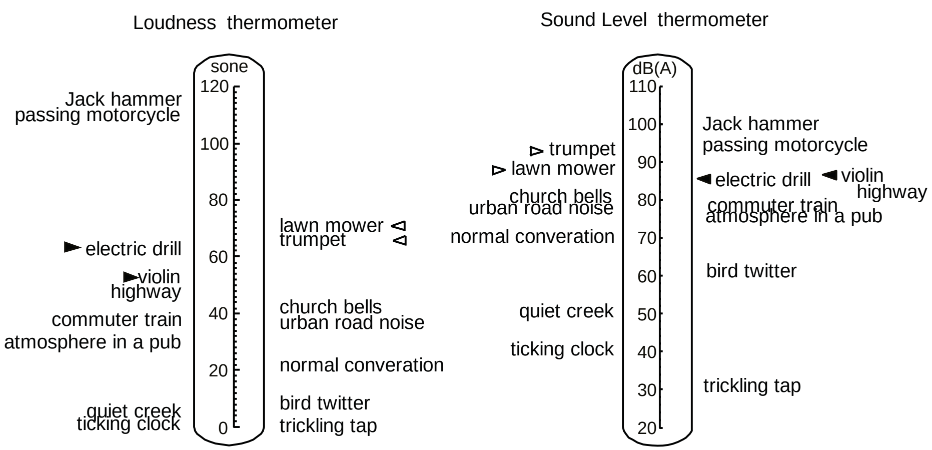
- Online courses
  http://clas.mq.edu.au/perception/psychoacoustics/chapter2.html

- Psysound3: free Matlab toolbox
  http://psysound.wikidot.com/

Loudness

- We can measure sound pressure levels with a SLM, but how does this correspond with the human perception of loudness?
- Fastl and Zwicker, p320.
  - "For example it is stated frequently in advertisings that the noise emission of printer G is by 90% less than that of printer F. This is correct in physical terms of A-weighted sound power. However, the loudness difference perceived by the customer is only about 35%, in line with the indications by a loudness meter."

Loudness



Loudness

- Subjective effect of changes in sound pressure level.

Change in sound level (dB)	Change in power Decrease	Change in power Increase	Change in apparent loudness
3	1/2	2	just perceptible
5	1/3	3	clearly noticeable
10	1/10	10	half or twice as loud
20	1/100	100	much quieter or louder

ENC4 p85, Table 2.1
ENC5, p74, Table 2.2

Loudness - Demonstration

Track 6: Magnitude estimation with anchor sound

This demonstration illustrates the method of magnitude estimation with anchor sound. Four pairs of stimuli are presented.

Write down numbers 1 to 4.

The first sound in each pair, called ***anchor sound***, is always an excerpt of a car sound with 70 dB.

The second sound in each pair should be assigned a numerical value which represents the relation in perceived loudness between the first (anchor) sound and the second sound. If for example the second sound in the second pair would be perceived as being 40 % louder than the anchor sound, then for the second sound the numerical value 140 should be given.

1	2	3	4

Loudness

- Anchor sound = 70dB.
- The second sound within a pair had levels of
  - 60 dB,
  - 75 dB,
  - 70 dB and
  - 68 dB.
- There are no right answers ...
  Maybe you wrote something like ..

1	2	3	4
80	120	100	100

Loudness

Audio Demonstrations:
Track 33: Loudness as a function of burst duration

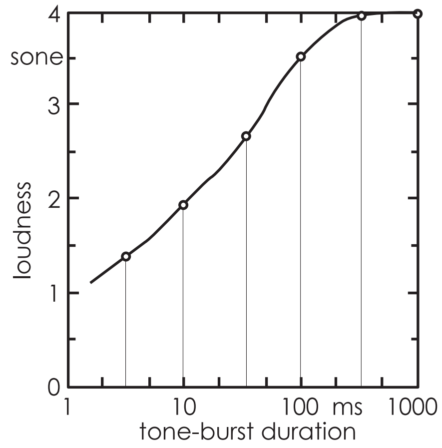
In this demonstration, the dependence of loudness on duration is illustrated. You will hear pairs of 3kHz tones (60 dB), the first tone of a pair is always 1000ms long, the second tone will get progressively shorter. Each pair is presented twice.

Q: What is the trend in loudness?

Loudness

The second tone has durations of: 1000, 300, 100, 30, 10, and 3 ms.

You can hear that below 100 ms, the loudness of the second tone decreases.



Loudness

- How is loudness determined?
  - Still use a microphone and instrumentation to measure amplitude and frequency, but use a different scale (metric) based on the type of sound.

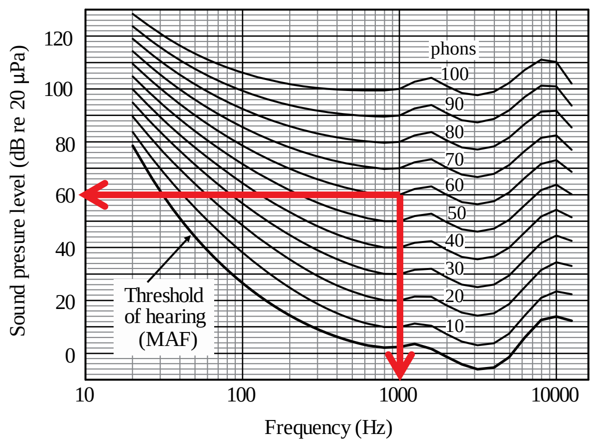
- Read Zwicker and Fastl, p317, Section 16.1.1.
- Also described in ENC p84, Section 2.4.2.

Loudness

Tonal sound

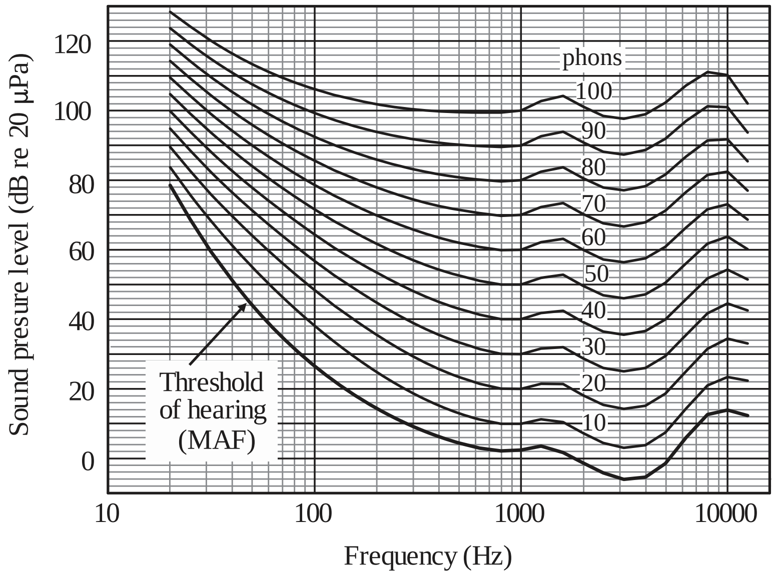
- Equal loudness contours for frontally incident tonal sound
- MAF = minimum audible field
- At 1000 Hz, the SPL value is the same as Phons curve

For Tonal Noise ....



Loudness

For Tonal Noise ....



Loudness

- We said before that:

Change in sound level (dB)	Change in power Decrease	Change in power Increase	Change in apparent loudness
3	1/2	2	just perceptible
5	1/3	3	clearly noticeable
10	1/10	10	half or twice as loud
20	1/100	100	much quieter or louder

- What we really want is a scale for apparent loudness.

Loudness

- Introduce the "sone" - 40-phon contour has arbitrarily been labelled **1 sone**.
- Then the 50-phon contour of the figure, which, according to previous table, would be judged twice as loud, has been labelled **2 sones**, etc.

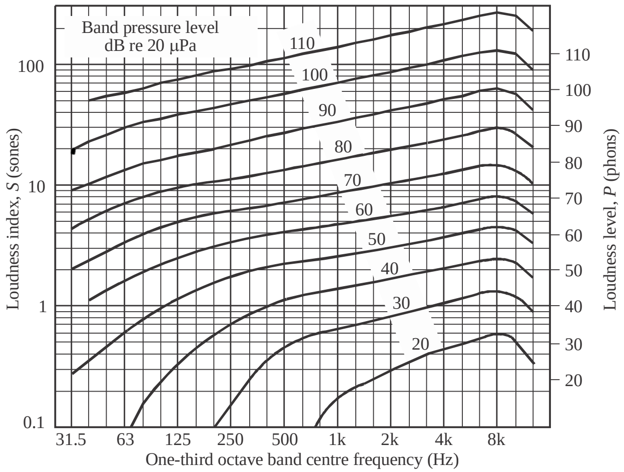
- The unit "sone" is used for the hearing sensation loudness, in just the same way as the unit "Pa" is used for the sound pressure.
  - It is most important not to mix up stimulus magnitudes such as "Pa" or "dB" and sensation magnitudes such as "sone".

Loudness

- Sone - includes equal loudness curves plus subjective overall loudness effects
- 2 Sones sound twice as loud as one

$$S = 2^{(P-40)/10}$$

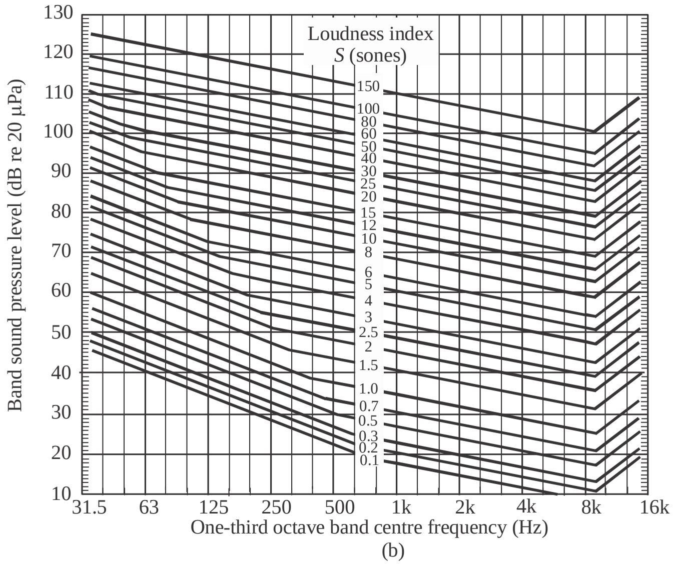
- (above P = 40 up to P=100)
- above 100 phons, subjective loudness increases less rapidly due to saturation
- Below 40 phons, opposite occurs.



(a)

ENC4 Fig 2.10(a), p91. ENC5 Fig 2.13(a), p80

Loudness



ENC4 Fig 2.10(b), p91.
ENC5 Fig 2.13(b), p80.

Example

ENC5, p80.

Given the octave band sound pressure levels shown in the example table in row 1, determine

- the loudness index for each band,
- the composite loudness in sones and in phons, and
- rank order the various bands in order of descending loudness.

Row	Octave band centre frequencies (Hz)	31.5	63	125	250	500	1k	2k	4k	8k
1	Band level (dB re 20 µPa)	57	58	60	65	75	80	75	70	65
2	Band loudness index (sones)	0.8	1.3	2.5	4.6	10	17	14	13	11
3	Ranking	9	8	7	6	5	1	2	3	4

Example

1. Enter the band levels in row 1 of the table, calculated using Fig 2.10(b). Read the loudness indices Si and write them in row 2.
2. Rank the indices as shown in row 3.
3. Determine loudness level (SONES) for each band using Eq. (2.33) where B = 0.3 (octaves) or 0.15 (1/3 octaves)

$$L = S_{\max} + B\sum_i{}' S_i \text{ sones}$$

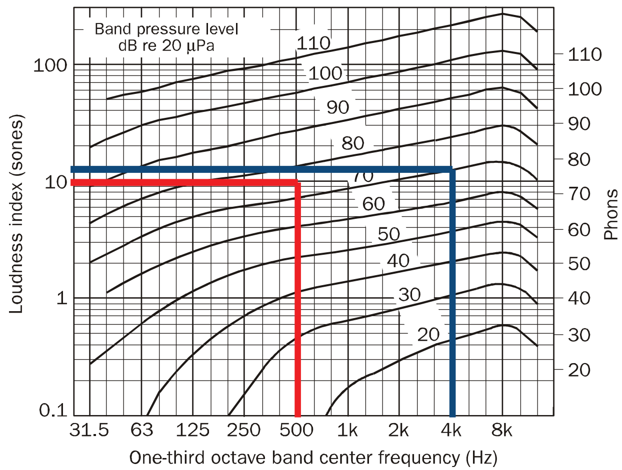
$$L = 17 + 0.3 \times 57.2 = 34 \text{ sones}$$

4. Enter the loudness (**34 sones**) in the scale on the left of Fig 2.10(a), and read loudness on the right, as **91 phons**.

$$P = 40 + 33.323 \times \log_{10} L$$

Example

1. Enter the band levels in row 1 of the table, calculated using Fig 2.10(b).
   Read the loudness indices Si and write them in row 2.
   E.g. 500 Hz, 75 dB == 10 sones;
   4kHz, 70 dB == 12.5 sones.



Example

... we get this for row 2.

Row	Octave band centre frequencies (Hz)	31.5	63	125	250	500	1k	2k	4k	8k
1	Band level (dB re 20 µPa)	57	58	60	65	75	80	75	70	65
2	Band loudness index (sones)	0.8	1.3	2.5	4.6	10	17	14	123	11
3	Ranking	9	8	7	6	5	1	2	3	4

Example

2. Rank the indices, as shown in row 3.

Row	Octave band centre frequencies (Hz)	31.5	63	125	250	500	1k	2k	4k	8k
1	Band level (dB re 20 µPa)	57	58	60	65	75	80	75	70	65
2	Band loudness index (sones)	0.8	1.3	2.5	4.6	10	17	14	13	11
3	Ranking	9	8	7	6	5	1	2	3	4

Example

3. Determine loudness level (SONES) for each band using Eq. (2.33) where B = 0.3 (octaves) or 0.15 (1/3 octaves)

$$L = S_{max} + B\sum_i {}' S_i \text{ sones}$$

' means don't include the max value in sum again!

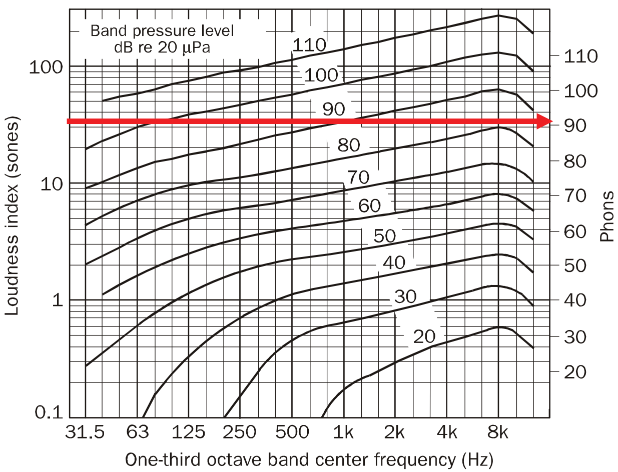
$$L = 17 + 0.3 \times 57.2 = 34 \text{ sones}$$

Row	Octave band centre frequencies (Hz)	31.5	63	125	250	500	1k	2k	4k	8k
1	Band level (dB re 20 µPa)	57	58	60	65	75	80	75	70	65
2	Band loudness index (sones)	0.8	1.3	2.5	4.6	10	17	14	13	11
3	Ranking	9	8	7	6	5	1	2	3	4

Example

4. Enter the loudness (34 sones) in the scale on the left of Fig 2.10(a), and read loudness on the right, as 91 phons.
OR ....

$$P = 40 + 33.323 \times \log_{10} L$$
$$= 40 + 33.323 \times \log_{10} 34$$
$$= 91 \text{ phons}$$

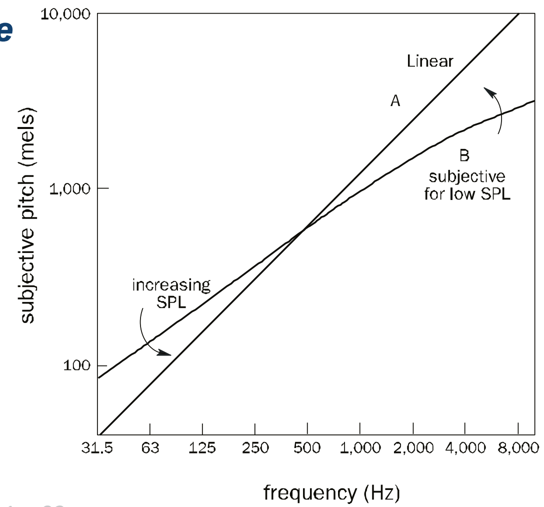


Pitch

- Range:
  - Lowest frequency a person with normal hearing can identify is about 20Hz.
  - Highest, normal hearing about 18kHz. Acute hearing 20kHz.

Pitch

- Pitch is the ***subjective*** response to frequency.
  - Not linearly related.
  - Dependent on sound level.



ENC Fig 2.11, p92

Pitch

Audio Demonstration
Track 15: Pitch shift of pure tones as a function of level

The *pitch* of a pure tone also depends on the *level* of the tone. In this demonstration three tones with frequencies of 200 Hz, 1000 Hz and 6000 Hz, respectively are played at levels of 50 dB and 75 dB. Each pair is repeated three times.

What is your perception of the pitch?

Q: Does the louder sound have a Lower / Same / Higher Pitch compared to the quieter sound?

Freq [Hz]	200	1000	6000
Lower / Same / Higher			

Pitch

- As displayed in below, The 1000 Hz tone and in particular the 200 Hz tone produces a *lower pitch* when played at the higher level of 75dB, whereas
- The 6000 Hz tone elicits a *higher pitch* when played louder.

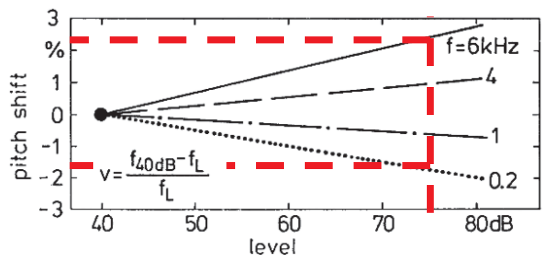


Fig 5.3, Psychoacoustics Facts and Models, Fastl & Zwicker

Pitch

- Lots of pop music uses Autotune by Antares for pitch and timing correction. http://www.antarestech.com/products/index.php

Pitch – Autotune Demo

https://www.youtube.com/watch?v=HGiaAilELME

Masking Demo

Track 9: Pure tones masked by white noise

In this demonstration a pure tone is masked by white noise. The tone has a frequency of 500Hz, the white noise has an overall level of 63dB.

You will hear six presentations, each with triplets of pure tones, masked by white noise.

Write down numbers 1 to 6.

If you hear the tone, write a Y/N beneath the number.

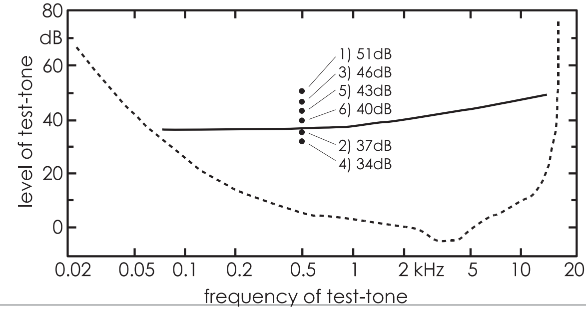
1	2	3	4	5	6
Y	N	Y	N	N	Y

Masking Demo

The level of the tone triplets is (in this order)

51dB, 37dB, 46dB, 34dB, 43dB, 40dB

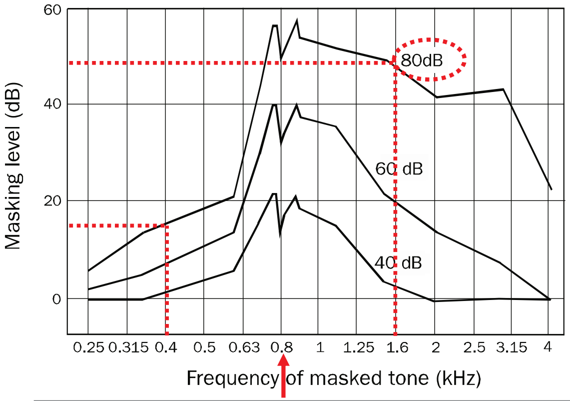
Except for the second (37 dB) and fourth triplet (34 dB), the tones are above masked threshold and should be audible.



Masking Effects Of Tonal Sound

ENC p82

- Masking level is equivalent to threshold shift
- Level of interfering sound shown parametrically
- 800Hz tone used as interfering sound at 40, 60, 80 dB



Play masking tone at 800Hz, 80 dB

The threshold for hearing another tone:

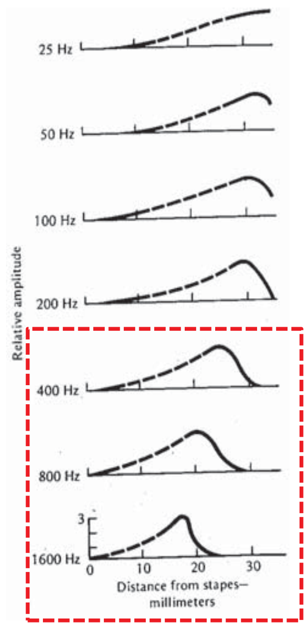
- at 400Hz has increased by 15dB.
- at 1.6kHz has increased by 50dB.

See how the curve above 800 Hz is much higher than below 800Hz…
-> i.e. High freqs are masked!

Masking

Remember this slide from topic on Human Hearing

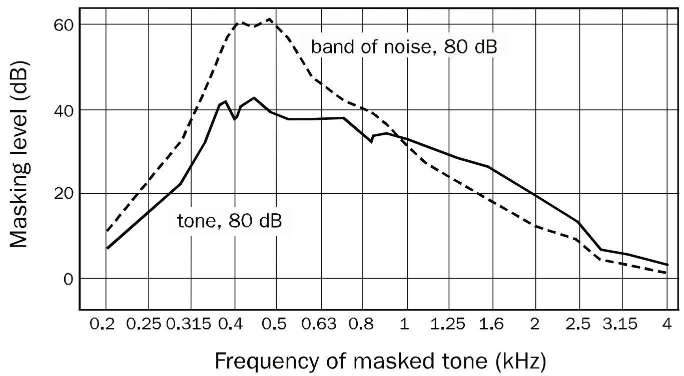
- Low frequencies mask high frequencies
  - Notice how low frequency response will interfere with high frequency response.
- High frequency hearing lost first



Basilar membrane response

Masking Effects of Tonal and Band Limited Sound

- Level of interfering sound shown parametrically
- Solid line corresponds to 400Hz tone 80dB used as interfering sound
- Dashed line corresponds to band of noise centred on 400Hz and 90Hz wide, 80dB as interfering sound



Even though both maskers are at 80dB, the *Narrow Band* of noise is much more effective at masking than the *Tone* at 80dB.

Critical Bands

- Our hearing system analyses a broad spectrum into parts that correspond to critical bands.
- Zwicker (1961) proposed dividing the frequency scale into bands called “Bark”
  (in memory of Barkhausen, a scientist who introduced the “phon”, a value describing loudness level for which the critical band plays an important role).
- The audible frequency range to 16 kHz can be subdivided into 24 abutting critical bands.
  (See handout notes for the scale.)

Critical-band rate z (in Barks), lower (F lower Hz) and upper (F upper Hz) frequency limit of critical bandwidths, Δf G (Hz), centred at (F centre Hz)

Bark	F_lower, F_upper	F_centre	z	Δf G	Bark	F_lower, F_upper	F_centre	z	Δf G
0	0				12	1720			
		50	0.5	100			1850	12.5	280
1	100				13	2000			
		150	1.5	100			2150	13.5	320
2	200				14	2320			
		250	2.5	100			2500	14.5	380
3	300				15	2700			
		350	3.5	100			2900	15.5	450
4	400				16	3150			
		450	4.5	110			3400	16.5	550
5	510				17	3700			
		570	5.5	120			4000	17.5	700
6	630				18	4400			
		700	6.5	140			4800	18.5	900
7	770				19	5300			
		840	7.5	150			5800	19.5	1100
8	920				20	6400			
		1000	8.5	160			7000	20.5	1300
9	1080				21	7700			
		1170	9.5	190			8500	21.5	1800
10	1270				22	9500			
		1370	10.5	210			10500	22.5	2500
11	1480				23	12000			
		1600	11.5	240			13500	23.5	3500
12	1720				24	15500			
		1850	12.5	280					

Critical Bands

The concept of critical bands was proposed by Fletcher. He assumed that the part of a noise that is effective in masking a test tone is the part of its spectrum lying near the tone.

In order to gain not only relative values but also absolute values, the following additional assumption was made:

- masking is achieved when the power of the tone and the power of that part of the noise spectrum lying near the tone and producing the masking effect are the same;
- parts of the noise outside the spectrum near the test tone do not contribute to masking.

Characteristic frequency bands defined in this way have a bandwidth that produces the same acoustic power in the tone and in the noise spectrum within that band when the tone is just masked.

Critical Band Demo 1

- ***Critical Bandwidth*** is the frequency separation needed between two frequencies in order for a difference in pitch to be perceived.
  - Two frequencies must be at least one critical bandwidth apart.

- Demo: Two sine waves at 400Hz. One gradually increases to 510Hz.
  - Initially hear beating;
  - Then distorted;
  - At the end you will hear two tones.

www.sfu.ca/sonic-studio/handbook/Sound/Critical_Band.aiff

Critical Band Demo 2

Track 10: Pure tones masked by narrow-band noise

The masked threshold of pure tones masked by critical-band wide noise (1 kHz, 70 dB) is illustrated. You will hear three series of tone triplets:

the first series is played at a level of 75 dB,

the second at a level of 60 dB,

the third at a level of 40 dB.

Each series consists of six tone triplets with the frequencies: 600 Hz, 800 Hz, 1000 Hz, 1300 Hz, 1700 Hz, and 2300 Hz.

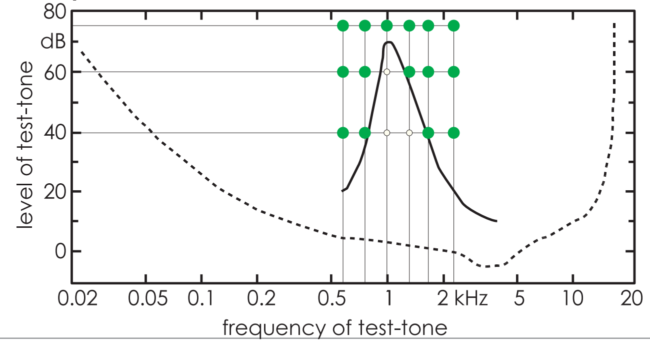
Critical Band Demo 2

- Write down these freqs and Y/N if you can hear the tone. [7 cols x 4 rows]

	600 Hz	800 Hz	1000 Hz	1300 Hz	1700 Hz	2300 Hz
75dB	Y	Y	Y	Y	Y	Y
60dB	Y	Y	N	Y	Y	Y
40dB	Y	Y	N	Y	Y	Y

Critical Band Demo 2

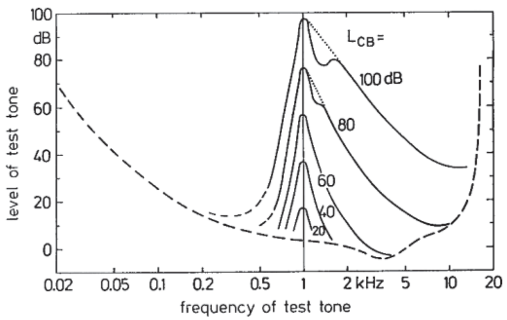
- In the second series the third tone triplet at 1000 Hz is masked by the narrow-band noise,
- In the third series the third and fourth triplet at 1000 Hz and 1300 Hz (for some persons also the fifth triplet at 1700 Hz) are masked.



Critical Band

This figure shows the masked threshold on the level of noise centred at 1kHz.

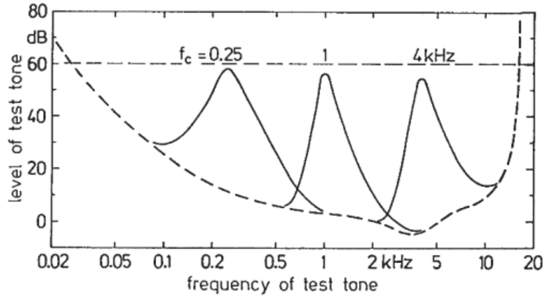
The maximum is always 3dB less than the level of the masking noise.



Fastl and Zwicker (2007), “Psychoacoustics Facts and Models”, Fig 4.4, p65.

Critical Band

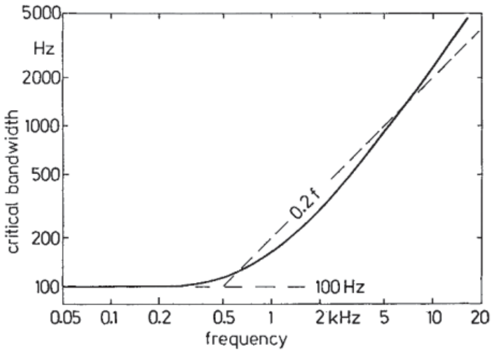
- Similar responses can be determined across the frequency range of hearing.
- This graph shows responses at 250Hz, 1kHz, 4kHz.
- The corresponding bandwidths of noises are: 100Hz, 160Hz, and 700Hz.



Fastl and Zwicker (2007), “Psychoacoustics Facts and Models”, Fig 4.3, p64.

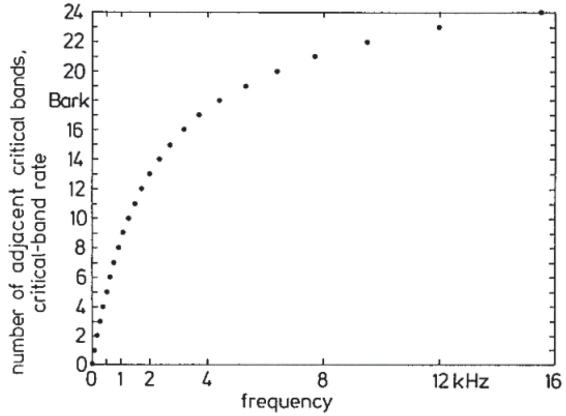
Critical Bandwidth

- Plot of the critical centre frequency and the bandwidth looks like:



Fastl and Zwicker (2007), “Psychoacoustics Facts and Models”, Fig 6., p159.

Critical Band Rate



Fastl and Zwicker (2007), “Psychoacoustics Facts and Models”, Fig 6.9, p161.

Critical Band Rate

- To convert frequency to Bark (Fastl Eq. 6.1, p164)

$$z(\text{Bark}) = 13\arctan(0.00076 f) + 3.5\arctan\left(\left[\frac{f}{7500}\right]^2\right)$$

- Critical Bandwidth

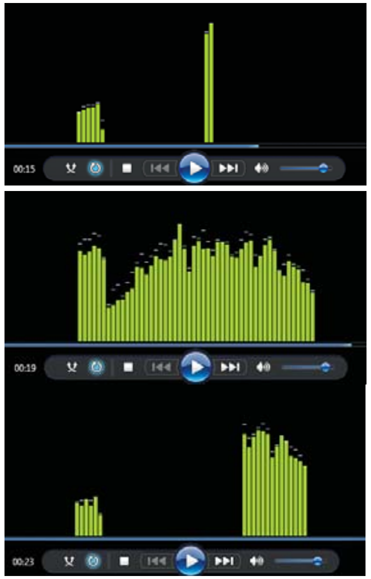
$$\Delta f_G(\text{Hertz}) = 25 + 75\left[1 + 1.4\left(f/1000\right)^2\right]^{0.69}$$

Sharpness

- See Chapt 9 Fastl and Zwicker (2007).
- Most important parameters are:
  - Spectral content
  - Centre frequency of narrow-band sounds.

- Sharpness is a measure of the *high frequency* content of a sound:
  - the greater the proportion of high frequencies the ‘*sharper*’ the sound.
- When investigating the “sound quality” of products, the sharpness can be a useful measure of high freq content.

Sharpness Demonstration



- Should hear increasing sharpness ...
  - Narrow Band noise centred at 1 kHz
  - Uniform Broad band
  - Noise High-pass filtered above 3kHz

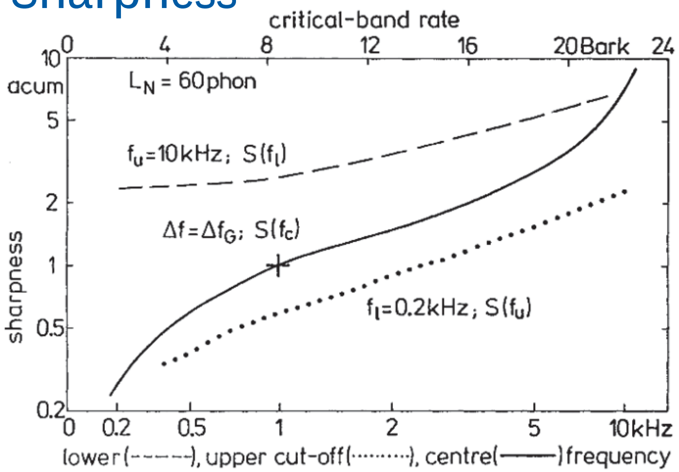
Track 34

Sharpness: Demonstration 2

Noise Type	Sharp	Not Sharp
Broadband		
Tonal		

http://www.salford.ac.uk/computing-science-engineering/research/acoustics/psychoacoustics/sound-quality-making-products-sound-better/sound-quality-testing/sharpness-booming

Sharpness



Fastl & Zwicker Fig 9.1, p240.

Reference point of 1 acum indicated by the + marker. The reference sound producing 1 acum is a narrow-band noise one critical-band wide at a centre frequency of 1 kHz (920-1080Hz) having a level of 60dB.

Sharpness

- Calculated as

$$S = 0.11 \times \frac{\left[\int_0^{24\,\text{Bark}} N'\,g(z)\,z\,dz\right]}{\left[\int_0^{24\,\text{Bark}} N'\,dz\right]} \quad \text{acum}$$

Eq (9.1), p242 Fastl & Zwicker

where denominator give the total loudness N (showed earlier how to calculate) ,

Numerator is like a weighted first moment of area involving loudness N, and g is from the graph

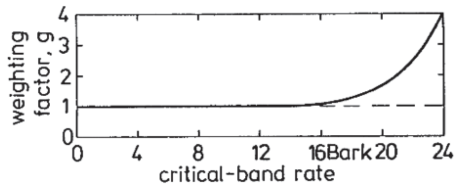
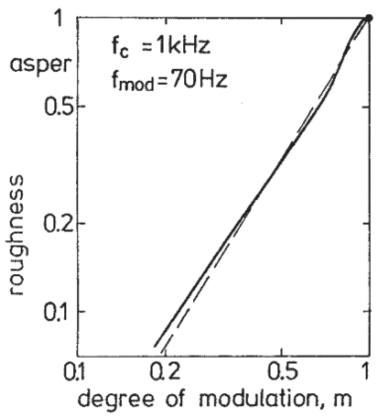


Fig 9.2, p242 Fastl & Zwicker

Roughness

- See Chapt 11 Fastl and Zwicker (2007).
- Units of asper.
- roughness of 1 asper is the 60-dB kHz tone that is 100% modulated amplitude at a modulation frequency of 70 Hz.



Fastl Fig 11.1, p258

Roughness Audio Demo

Track 38: Roughness of pure tones as a function of the degree of modulation

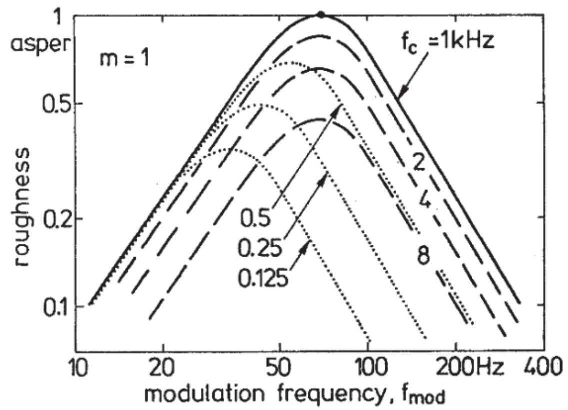
You will hear an amplitude modulated 1 kHz tone with a modulation frequency of 70 Hz and a varying degree of amplitude modulation of

1.0, 0.7, 0.4, 0.25, 0.125, 0.1, and 0.

The roughness of the sound will decrease with smaller degrees of modulation.

Roughness

For other freqs, the peak roughness varies with modulation frequency.



Fastl & Zwicker Fig 11.2, p259

Roughness Audio Demo

Track 39: Roughness as a function of modulation frequency

Three different sounds are presented with different modulation frequencies:

1. amplitude modulated broad-band noise,
2. amplitude modulated pure tones, and
3. frequency modulated pure tones

Each sound is presented with modulation frequencies.

Q: Which one sounds the most "rough" (1st, 2nd, 3rd) ?

Roughness Demo

Each sound is presented with modulation frequencies of 20 Hz, 70 Hz and 200 Hz.

At 70 Hz, the sensation of roughness reaches a maximum. (2nd sound in each group)

Binaural Hearing

- Recordings made with microphones placed at the position of the ears on an artificial headform or on a human.
- Head Related Transfer Function (HRTF) – effect of shape of head (and torso)

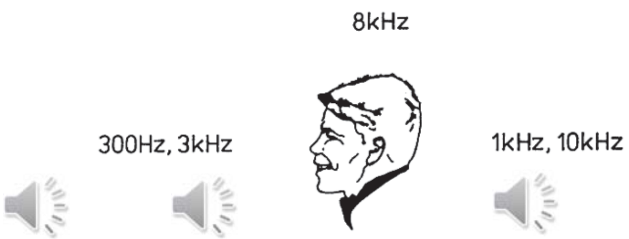
Binaural Hearing

- Localization
  - Ability to determine origin of sound in 3D.
  - Inversions can occur: a sound source is situated in front of the subject but a subject perceives the sound as coming from *behind*.

    For example: if the subject faces the direction 0° and the sound source is presented at an angle of 30° , the sound may be perceived at an angle of 150°. This means that the sound source is situated in front of the subject somewhat to the right, but that the sound is perceived as coming from a position behind the subject.

Binaural Hearing

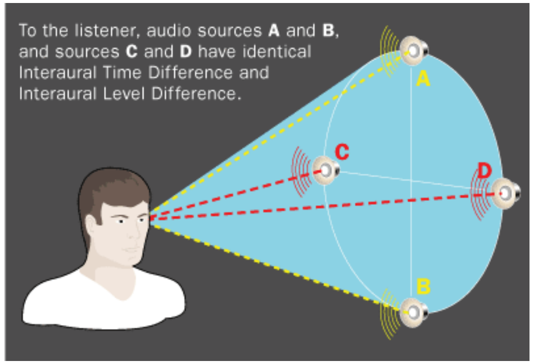
- Localization
  - Some sources always sound like they are originating from a particular direction.



Binaural Hearing

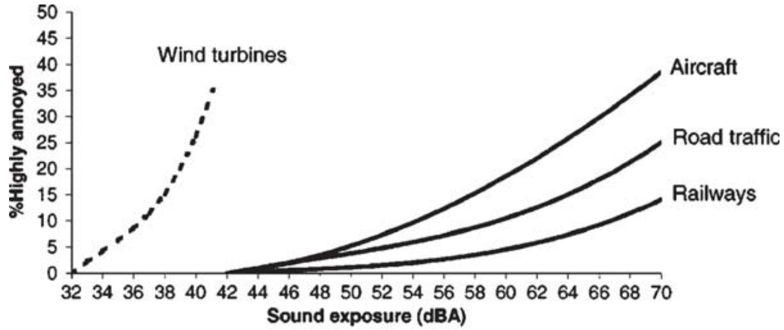
Cone of confusion

- A sound source on the surface of an imaginary cone cannot easily be localised because of similar Interaural Time Differences and Interaural Level Differences.



Annoyance

- Depends on many factors



Sound exposure is for wind turbines calculated A-weighted L_{eq} for a hypothetical time period and for transportation DNL.

Pedersen and Waye (2004), "Perception and annoyance due to wind turbine noise—a dose–response relationship", JASA

Annoyance

- Psychoacoustic elements can be combined (with caution) to determine Psychoacoustic Annoyance (PA), Eq. (16.1)

$$PA \cong N\left(1+\sqrt{[g_1(S)]^2+[g_2(F,R)]^2}\right)$$

where
N hearing sensations loudness
S sharpness
F fluctuation strength
R roughness

Annoyance

- As an example (Eq. 16.2)

$$PA \cong N_5\left(1+\sqrt{[w_S]^2+[w_{F\,R}]^2}\right)$$

where
N_5 is the percentile loudness is sone

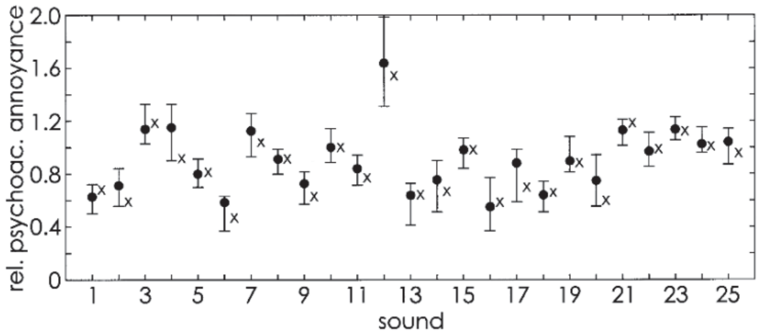
$$w_S = \left[\frac{S}{\mathrm{acum}}-1.75\right]\times 0.25\times\log\left[\frac{N_5}{\mathrm{sone}}+10\right] \text{ for } S > 1.75\,\mathrm{acum}$$

describes the effects of sharpness S

$$w_{F\,R} = \left[\frac{2.18}{(N_5/\mathrm{sone})^{0.4}}\right]\times\left[0.4\times\frac{F}{\mathrm{vacil}}+0.6\times\frac{R}{\mathrm{asper}}\right]$$

Annoyance

- Example: Car sounds (Fig 16.15, Psychoacoustic annoyance of car sounds Dots: Data from psychoacoustic experiments. Crosses: PA value



=> Good correlation between jury trials and predicted values

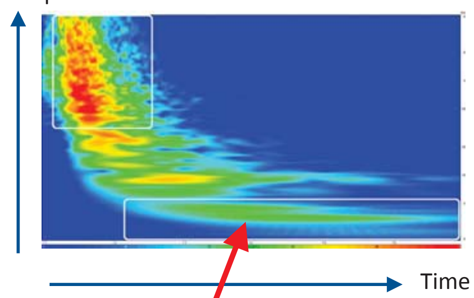
Sound Quality in Product Design

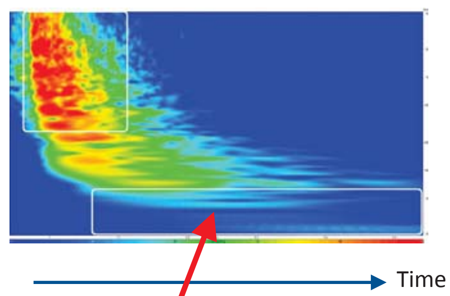
- For example, the new BMW 4 Series Gran Coupe is "a sporty car, so the door isn't supposed to sound too heavy. It can't sound too light, because a light door wouldn't convey the right aspects of quality and safety. But it's not supposed to sound too heavy, either. It should have a precise sound."
- By contrast, "the BMW 7 Series would be a bit softer maybe, a bit darker in the sound as it's our flagship sedan."

Sound Quality in Product Design

Final

Prototype





The production car has the right low and high freq balance.

There is low freq content missing in the prototype.

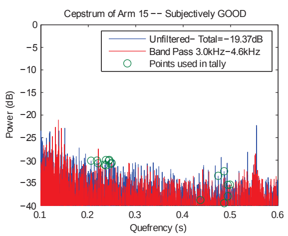
https://medium.com/re-form/click-clack-clunk-how-the-perfect-car-door-sound-is-made-and-why-it-matters-2cf867983a34

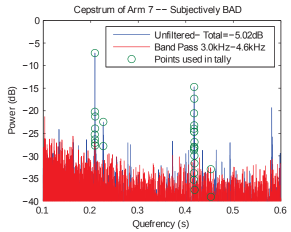
Sound Quality Checking for Defects

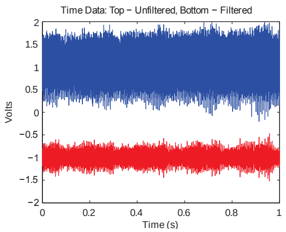
- Sound quality can also be used for production manufacturing quality to ensure there are no defects.
- Example of checking for defective side mirror mechanisms.

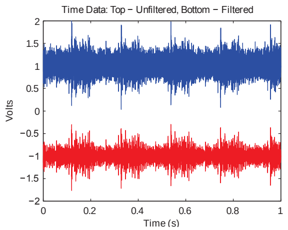
Sound Quality Checking for Defects

- Good
- Defective







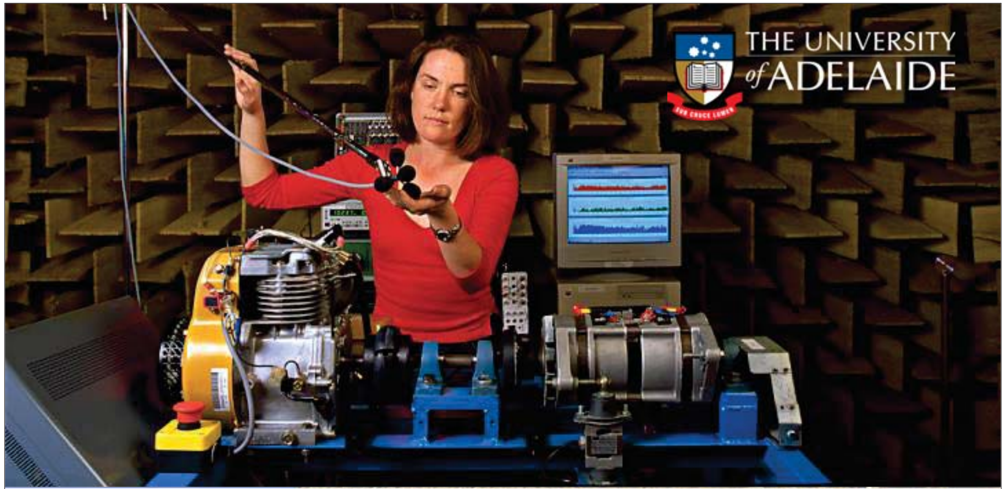


Jury Testing

- Used extensively to evaluate consumer response of sound made by products.
- Play sample sounds, either real or synthesized, to an audience, under very controlled conditions, and obtain subjective responses.
- We will conduct a jury trial ...

Summary

- There are many useful psychoacoustic terms for characterising sounds.
- These terms have metrics (values) that can be used to compare sounds.
- The metrics can be combined into a weighted sum to give a value of annoyance or pleasantness.
- Determining the appropriate weighting factors usually often involves doing jury testing.



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Carl Howard

Sound Sources

adelaide.edu.au

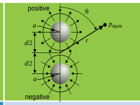
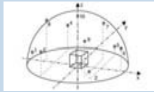
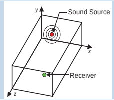
seek LIGHT

Course Content – How it fits

Where are we going?

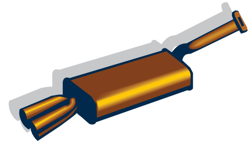
Need to know about sources and power for the next topics…

- Sound indoors
- Outdoor sound propagation

TOPIC	
Sound Sources	
Sound Power	
Indoor	
Outdoor	

Sound Sources

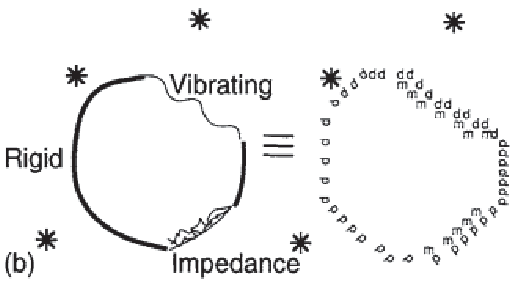
- Need for simple acoustic source models

Global Chiller


Sound Sources

- A complex acoustic system can be modelled accurately by combining a number of simple sound sources (monopoles, dipoles, quadrupoles)



m=monopole,
d=dipole

From: Fahy, Foundations of Engineering Acoustics, Fig 6.11

Contents

- Monopoles, dipoles and quadrupoles
- Line sources
- Baffled piston
- Coherent and incoherent plane sources
- Directivity and reflection effects
- Plane wave reflection and transmission

Learning Outcomes

- Understand and apply theory for the various acoustic source types.
- Be able to select an acoustic source type to represent a real noise source.

Resources

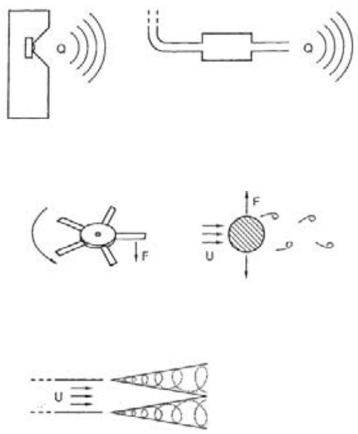
Lots of good resources on the internet about acoustic sources.

See

- http://www.acs.psu.edu/drussell/demos/rad2/mdq.html
- http://www.acs.psu.edu/drussell/Publications/MDQSources.pdf
- http://resource.isvr.soton.ac.uk/spcg/tutorial/tutorial/Tutorial_files/Web-further-dipoles.htm
- http://www.animations.physics.unsw.edu.au/waves-sound/

Categories

- Fluctuating volume/mass displacement/injection
- Accelerating/fluctuating force on fluid
- Fluctuating fluid shear stress

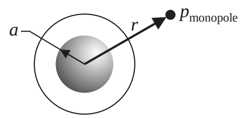


From: Fahy, Foundations of Engineering Acoustics, Fig 6.1

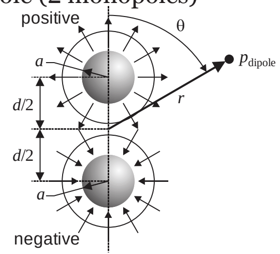
What we'll cover

Basic sound source types include:

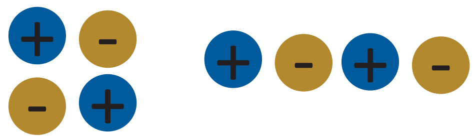
- Monopole



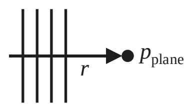
- Dipole (2 monopoles)



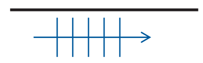
- Quadrupole (2 dipoles)



- Plane Wave



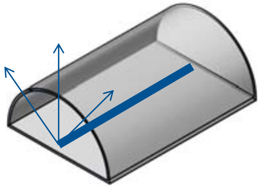
Plane wave in free field



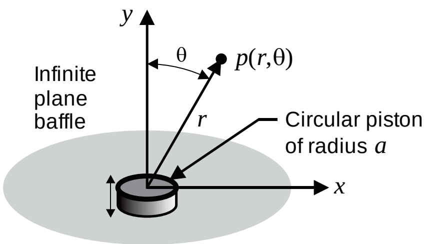
Plane wave in ducts

What we'll cover

- Line Source

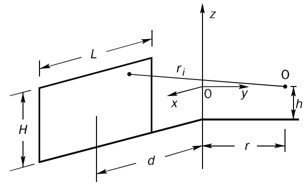


- Baffled Piston

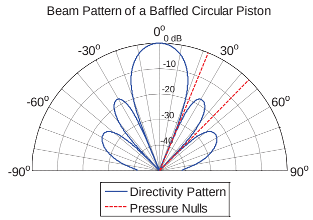


What we'll cover

- Incoherent Plane Radiator

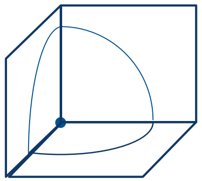


- Directivity of sources

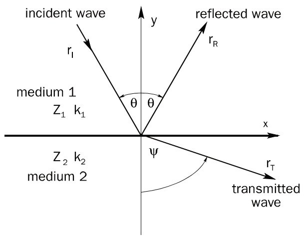


What we'll cover

- Reflections off boundaries



- Reflection, Transmission at boundary (Snell's law)

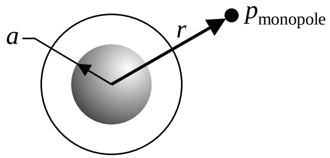


Sound Source Types and Sound Power

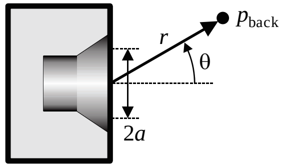
- MONOPOLES - power ∝ U^4
  - where U is the velocity
  - engine exhaust
  - cavitation noise
- DIPOLES - power ∝ U^6
  - spoiler and grille noise
  - valve noise (fluctuating forces on fluid boundaries)
- QUADRUPOLES - power ∝ U^8
  - valve noise (turbulence alone)
  - jet noise

Monopole

- Expanding rigid sphere
- Radiates equally in all direction.

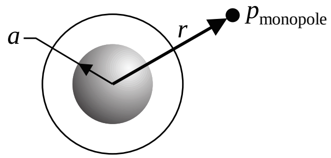


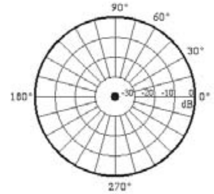
- A loudspeaker in a sealed enclosure at low frequencies is similar to monopole.

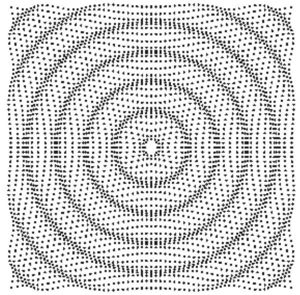


Simple Source

- Monopole
  - Radiates equally in all directions





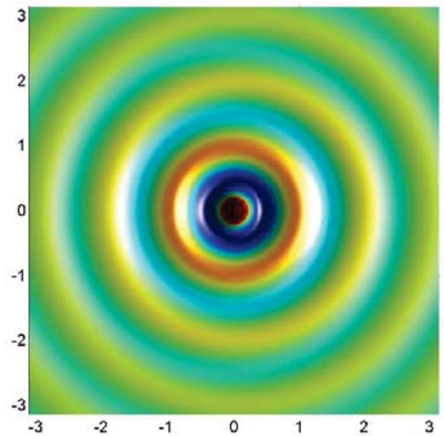


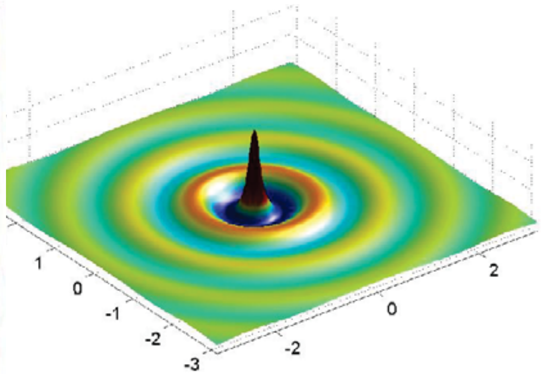
http://www.acs.psu.edu/drussell/demos/rad2/mdq.html

Monopole

- Sub-woofers are used to generate low frequency bass.
- As shown, the sound field is *omni-directional*.
- What this means is that you can place and point them anywhere in a room!

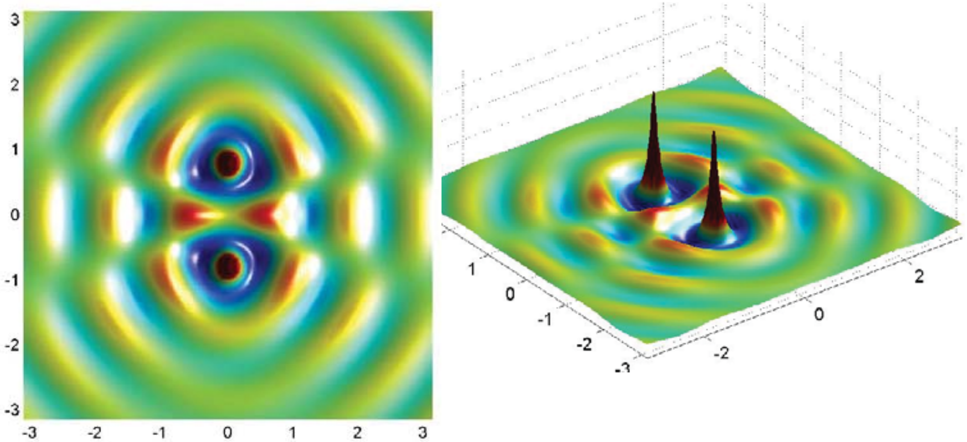
Monopoles





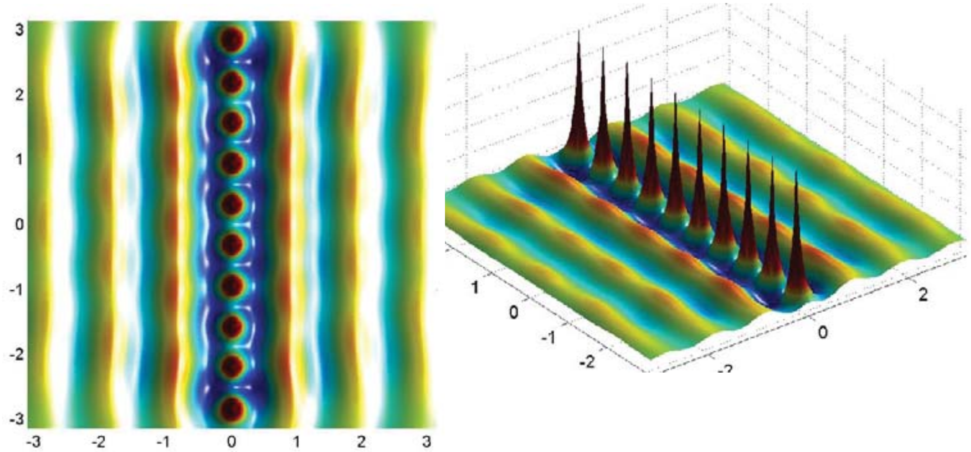
Two Monopoles in Phase

- Superposition two monopoles.



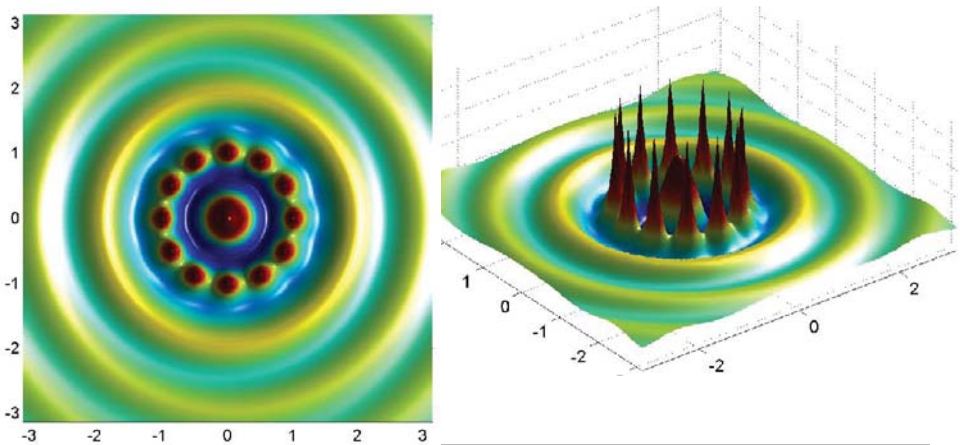
Monopoles

- A linear array of monopoles



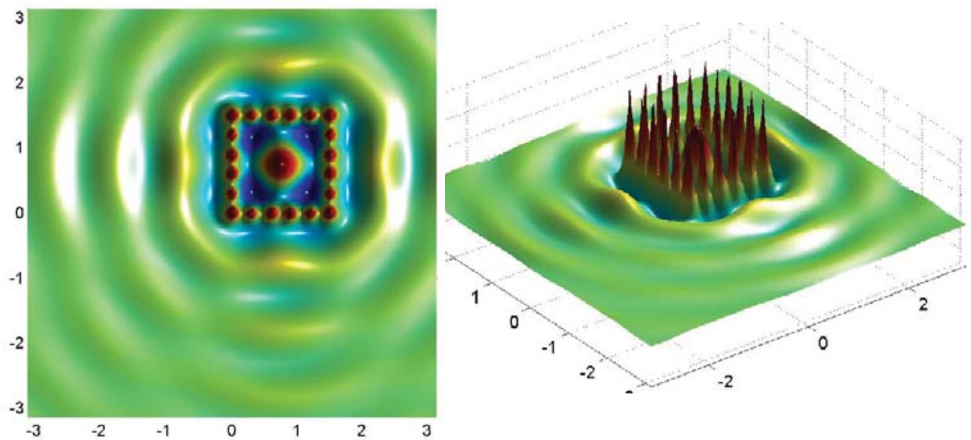
Monopoles

- Ring of monopoles can simulate another monopole in the far-field.



Monopoles

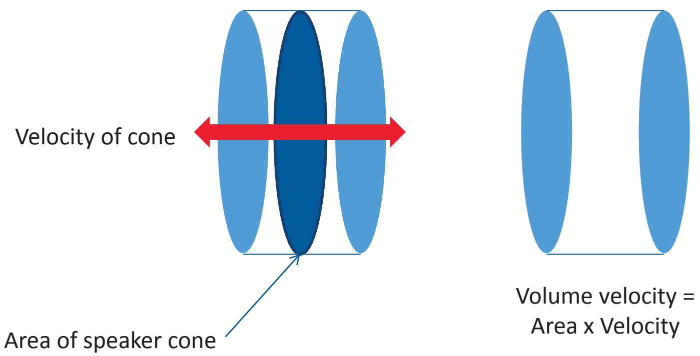
- Can create complex acoustic fields



Monopole

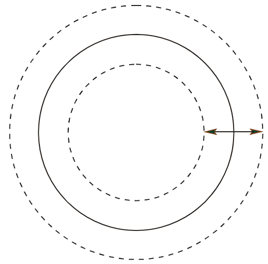
- An important concept to understand is

VOLUME VELOCITY



Simple Source (monopole or pulsating sphere)

- spherical wave potential function



$$\varphi = f(ct - r)/r$$

$$p = \rho \frac{\partial \varphi}{\partial t} = \rho c f'(ct - r)/r$$

$$u = -\frac{\partial \varphi}{\partial r} = [f(ct - r) + r f'(ct - r)]/r^2$$

$$I = \frac{1}{T}\int_0^T pu\, dt = \frac{\rho c}{T}\int_0^T [ff' + rf'f']/r^3 dt$$

Note: $f'(w) = \frac{\partial}{\partial w} f(w)$

Require: $W = \int_S I\, dS = I\, 4\pi r^2 = \text{constant}$

thus: $\frac{1}{T}\int_0^T ff'\,dt = 0$ and $I = \rho c \langle f'f' \rangle / r^2$

Simple Source (Cont.)

$$\langle u^2 \rangle = \frac{1}{Tr^4}\int_0^T \left[ff + 2rff' + r^2 f'f'\right]dt = \langle ff \rangle / a^4 + \langle f'f' \rangle / a^2$$

For source with radius $r = a$ *the* rms source flux defined as

$$\boxed{Q_{rms} = 4\pi a^2 \sqrt{\langle u^2 \rangle}}$$

ENC p190

Q_{rms} = RMS volume velocity (m^3/s)
a = radius of source (m).
$\sqrt{\langle u^2 \rangle}$ = RMS velocity of surface (m/s)

Simple Source (Cont.)

- For pure tone or narrow band of noise

$\langle ff \rangle = \langle f'f' \rangle / k^2$ as $f(ct - r) = e^{jk(ct-r)}$ and $f' = jkf$

Thus $Q_{rms} = 4\pi\sqrt{\langle ff \rangle + \langle f'f' \rangle}\, a^2$

$$Q_{rms} = 4\pi\sqrt{\langle f'f' \rangle\left(1 + k^2 a^2\right)}/k$$

- Intensity of a monopole

$$\boxed{I_M = \frac{Q_{rms}^2 k^2 \rho c}{(4\pi r)^2 \left(1 + k^2 a^2\right)} = \frac{\langle p^2 \rangle}{\rho c}}$$

ENC p190

Q_{rms} = RMS volume velocity (m^3/s)
a = radius of source (m)
r = distance from source (m)
$k = 2\pi f / c$ = wavenumber (rad/m)
ρ = density (kg/m^3)
c = speed of sound (m/s)
$\sqrt{\langle u^2 \rangle}$ = RMS velocity of surface (m/s)

Simple Source

- **Sound Power:**
  ENC p190

$$W_M = \int I dS = \frac{\rho c Q_{rms}^2 k^2}{4\pi\left(1+k^2a^2\right)}$$

- **Sound Pressure:**
  ENC p190

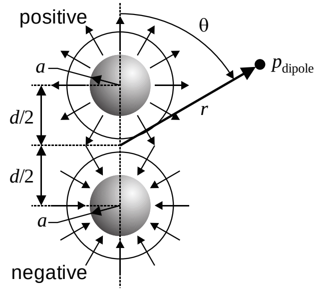
$$\left\langle p^2 \right\rangle = \frac{\left(Q_{rms} k\rho c\right)^2}{\left(4\pi r\right)^2\left(1+k^2a^2\right)}$$

$$\left\langle p^2 \right\rangle = \frac{W_M \rho c}{4\pi r^2}$$

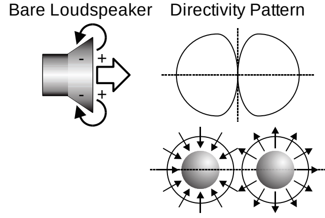
W_M = sound power from monopole in W
Q = volume velocity m³/s
k = wavenumber = $\omega/c = 2\pi f/c$ in rad/m
a = radius of source in m

Dipole

- Two monopoles that have opposite phase.
- Acoustic cancellation normal to the axis of the dipole.

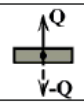
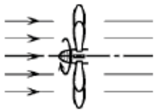


- An unsealed loudspeaker is similar to a dipole. Pressure equalisation around edges causes cancellation.



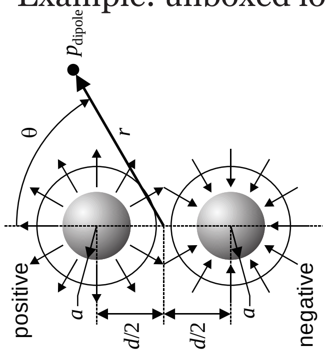
Dipole

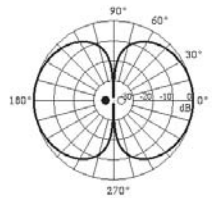
- Many sound sources including aerodynamic dipole sources can be modelled as point dipoles.

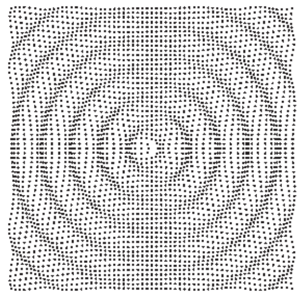
Dipole	Physical situation	Sketch
 Fluctuating force	• Transversally oscillating bodies	
	• Bodies in a flow field	
	• Propellors	

Dipole Source

- Characterised by two identical monopoles, spaced a distanced apart, where they have a 180° phase shift between them.
- Example: unboxed loudspeaker.



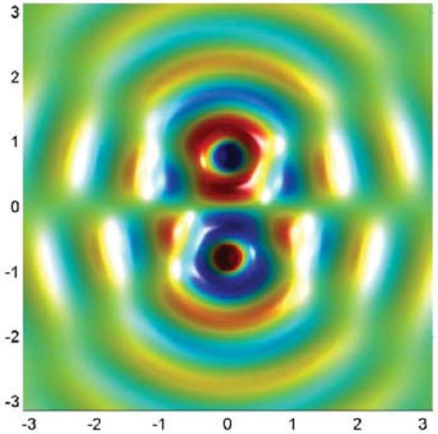


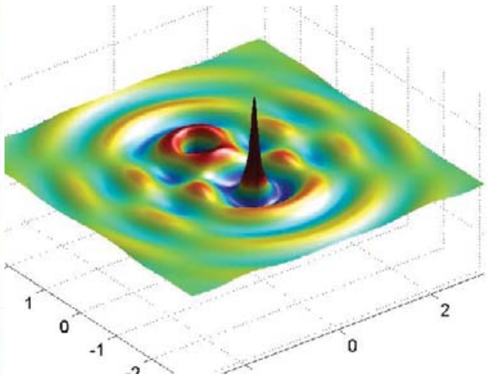


http://www.acs.psu.edu/drussell/demos/rad2/mdq.html

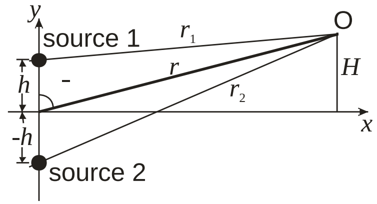
Dipole

- 2 monopoles closely spaced, 180° out of phase
- Cancels normal to axis





Dipole Source



- for $h/r << 1$, $r_1 \approx r - h\cos\theta$ and $r_2 \approx r + h\cos\theta$

$$\phi = \frac{\mathrm{f}\,(ct - r + h\cos\theta)}{r} - \frac{\mathrm{f}\,(ct - r - h\cos\theta)}{r}$$

- Using a Taylor series expansion,

$$\mathrm{f}\,(x + \Delta) = \mathrm{f}\,(x) + \frac{\mathrm{f}\,'(x)}{1!}\Delta + \frac{\mathrm{f}\,''(x)}{2!}\Delta^2 + ...$$

- Thus: $$\varphi = \frac{1}{r}\left[f'(ct-r) + f'(ct-r)h\cos\theta + ...\right.$$
$$\left. - f'(ct-r) + f'(ct-r)h\cos\theta - ...\right]$$
$$= 2f'(ct-r)(h/r)\cos\theta$$

$$p = \rho\frac{\partial\phi}{\partial t} = 2\rho c\frac{h}{r}f''(\mathrm{ct-r})\cos\theta$$

$$u = -\frac{\partial\varphi}{\partial r} = \frac{2h}{r^2}\left[f'(ct-r) + rf''(ct-r)\right]\cos\theta$$

$$\langle f'f'\rangle k^2 = \langle f''f''\rangle \qquad f'(ct-r) = jkAe^{jk(ct-r)}$$

and $$f''(ct-r) = -k^2Ae^{jk(ct-r)}$$

Dipole Source

- **Intensity,**

ENC p194

$$I_D = \rho c\frac{h^2Q_{rms}^2k^4}{(2\pi r)^2(1+k^2a^2)}\cos^2\theta$$

Notice, it depends on angle.

- **Radiated sound pressure**

ENC p194

$$\langle p^2\rangle = \rho cI_D = \left[\frac{hQ_{rms}k^2\rho c\cos\theta}{2\pi r}\right]^2\frac{1}{(1+k^2a^2)}$$
$$= 3W_D\frac{\rho c}{4\pi r^2}\cos^2\theta$$

Notice, it depends on angle.

Dipole Source

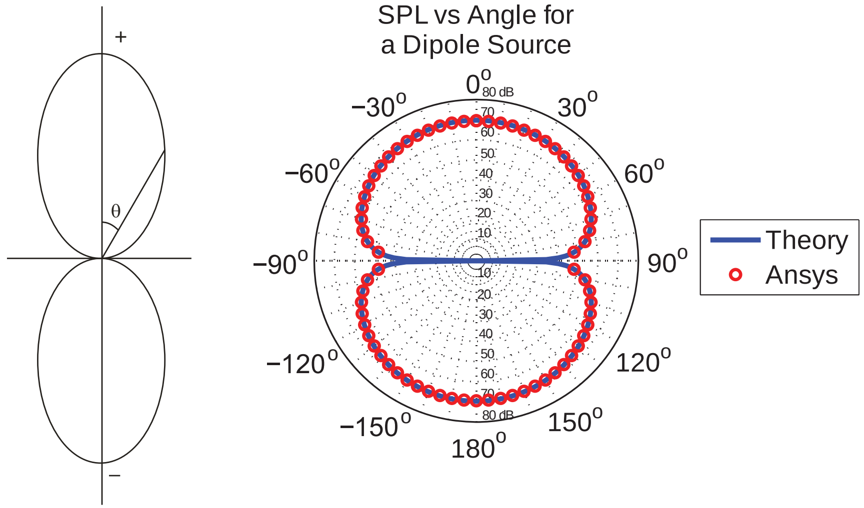
- **source strength,**

$$\boxed{Q_{rms} = 4\pi a^2 \sqrt{\langle u^2 \rangle}}$$

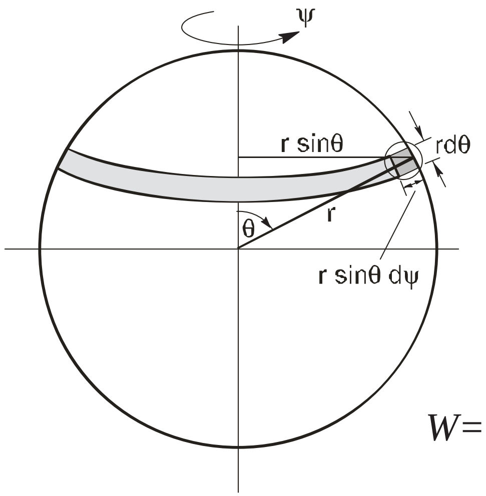
$$Q_{rms} = 4\pi \sqrt{\langle ff \rangle + \langle f' f' \rangle} a^2$$

= strength of one of the simple sources making up the dipole

Dipole Source - Radiation Pattern



Integration Over a Spherical Surface to Calculate Radiated Sound Power



$$W = \int_S I\, dS = \int_0^{2\pi} d\psi \int_0^{\pi} I r^2 \sin\theta d\theta$$

Dipole Source

- **Radiated power,**

ENC p195

$$\boxed{W_D = \rho c \frac{k^4 h^2 Q_{rms}^2}{3\pi(1 + k^2 a^2)}}$$

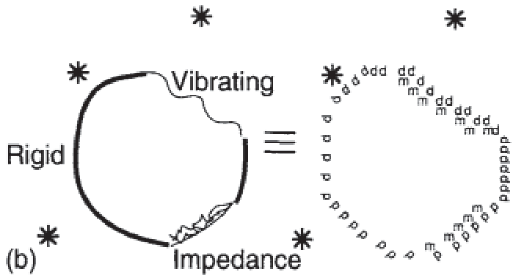
$$\boxed{\frac{W_D}{W_M} = 4(kh)^2/3 \qquad [\text{for} \quad h/r << 1]}$$

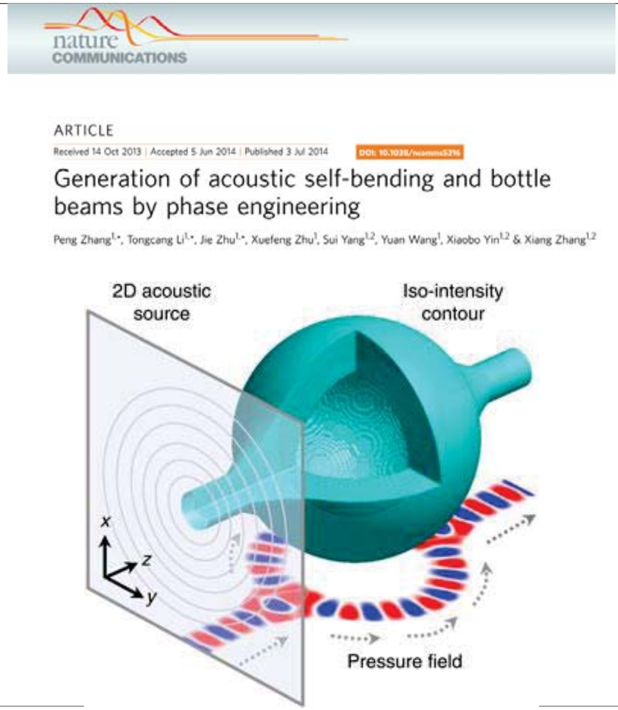
W_D = sound power from dipole in W
W_M = sound power from monopole in W
Q = volume velocity in m³ /s
k = wavenumber = ω/c = $2\pi f/c$ in rad /s
a = radius of source in m
h = half distance between two monopoles in m

Complex Sound Fields

- We can create ***ANY*** sound field if we know
  - Position,
  - Strength, and
  - Phase

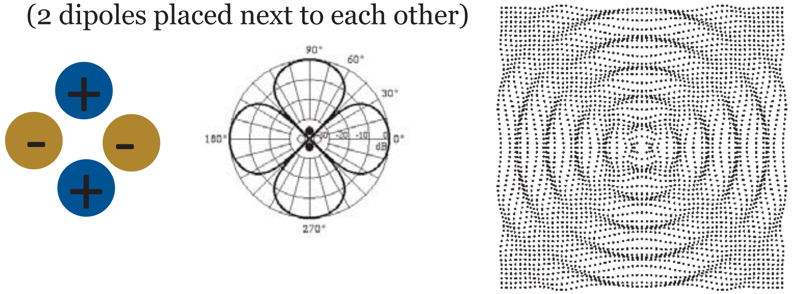
  of monopoles and dipoles.





Quadrupole Source

- Quadrupole (2 dipoles)
  - Lateral Quadrupole
    (2 dipoles placed next to each other)

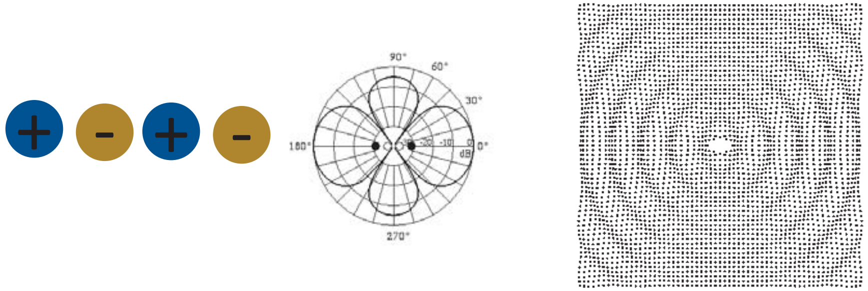


http://www.acs.psu.edu/drussell/demos/rad2/mdq.html

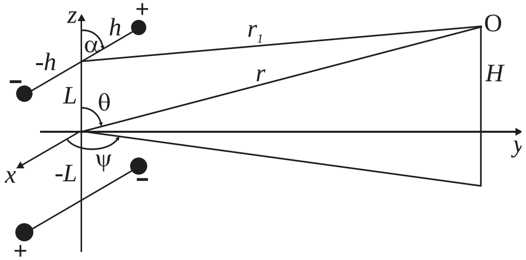
- Examples:
  - Rotating propellor

Quadrupole Source

- Linear Quadrupole
  (2 dipoles placed end-to-end)
- Example: Tuning fork.



Quadrupole Source



- lateral quadruplole: $\alpha = \pi/2$
- longitudinal quadrupole: $\alpha = 0$
- Intensity, power and radiated pressure

$$I_q = \rho c\left[\frac{k^3 hLQ}{\pi r\left(1+k^2a^2\right)}\right]^2 \left[H(\theta,\psi,\alpha)\right]^2$$

ENC p201

where, $H(\theta,\psi,\alpha) = \cos\theta\left(\sin\theta\sin\psi\sin\alpha + \cos\theta\cos\alpha\right)$

Quadrupole Source (Cont.)

$$W_q = \rho c\frac{\left(2k^3hLQ\right)^2}{5\pi\left(1+k^2a^2\right)}\left(\cos^2\alpha + \frac{1}{3}\sin^2\alpha\right)$$

ENC p201

$$\left\langle p_q^2\right\rangle = \left[\rho c\frac{k^3hLQ}{\pi r\left(1+k^2a^2\right)}\right]^2 \left[H(\theta,\psi,\alpha)\right]^2$$

Lateral quadrupole ($\alpha = \pi/2$) (Jet mixing process)

$$W_{lat} = \rho c\frac{\left[2k^3hL\,Q_{rms}\right]^2}{15\pi\left(1+k^2a^2\right)}$$

or $$\left\langle p_{lat}^2\right\rangle = \left[\rho c\frac{k^3hL\,Q_{rms}\sin 2\theta\sin\psi}{2\pi r\sqrt{\left(1+k^2a^2\right)}}\right]^2$$

$$\left\langle p_{lat}^2\right\rangle = \rho c\frac{15W_{lat}\sin^2 2\theta\sin^2 2\psi}{16\pi r^2}$$

Quadrupole Source

Longitudinal Quadrupole ($\alpha = 0$)

$$W_{long} = \rho c\frac{\left[2k^3hL\,Q_{rms}\right]^2}{5\pi\left(1+k^2a^2\right)}$$

ENC p202

and the radiated mean square acoustic pressure is:

$$\left\langle p_{long}^2\right\rangle = 5\rho c\frac{W_{long}\cos^4\theta}{4\pi r^2}$$

Question ....

- A back-enclosed loudspeaker with a cone diameter of 10cm is radiating sound at 50Hz in a free field, and the sound pressure level measured 2m away is 90 dB re 20μPa.

  What is the peak volume velocity of the source?

Answer ....

- What do we know...
  - SPL, so we know the pressure
  - We know the distance r
  - We will assume ρ=1.21, c = 343
- We want to know Q_{peak}

$$\langle p^2\rangle = \frac{W_M \rho c}{4\pi r^2}$$

$$W_M = \frac{\rho c Q_{rms}^2 k^2}{4\pi\left(1+k^2a^2\right)}$$

$$Q_{rms}^2 = \frac{4\pi\left(1+k^2a^2\right)W_M}{\rho c k^2}$$

Answer ...

$$\langle p^2\rangle = \frac{W_M \rho c}{4\pi r^2}$$

$$W_M = \langle p^2\rangle \frac{4\pi r^2}{\rho c}$$

$$L_p = 10\log_{10}\left[\frac{\langle p^2\rangle}{p_{\text{ref}}^2}\right]$$

$$\langle p^2\rangle = p_{\text{ref}}^2 \times 10^{(L_p/10)}$$

$$W_M = \left[p_{\text{ref}}^2 \times 10^{(L_p/10)}\right]\frac{4\pi r^2}{\rho c}$$

$$Q_{rms}^2 = \frac{4\pi\left(1+k^2a^2\right)}{\rho c k^2}\left\{\left[p_{\text{ref}}^2 \times 10^{(L_p/10)}\right]\frac{4\pi r^2}{\rho c}\right\}$$

$$Q_{rms}^2 = \frac{4\pi\left(1+k^2a^2\right)}{\rho c k^2}\left\{\left[p_{\text{ref}}^2 \times 10^{(L_p/10)}\right]\frac{4\pi r^2}{\rho c}\right\}$$

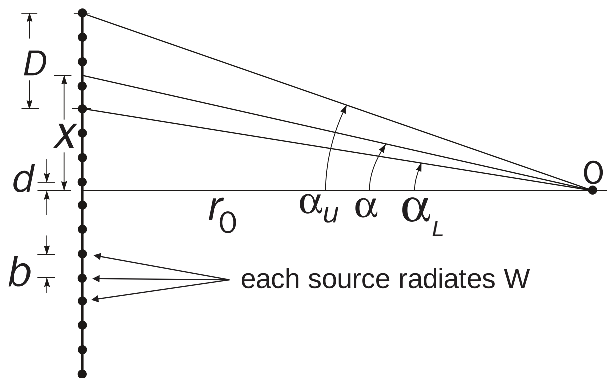
$$Q_{\text{peak}} = \sqrt{2} \times \sqrt{\frac{4\pi\left(1+k^2a^2\right)}{\rho c k^2}\left\{\left[p_{\text{ref}}^2 \times 10^{(L_p/10)}\right]\frac{4\pi r^2}{\rho c}\right\}}$$

$$Q_{\text{peak}} = \sqrt{2} \times \frac{4\pi r p_{\text{ref}}}{\rho c k}\sqrt{\left(1+k^2a^2\right)10^{(L_p/10)}}$$

$$Q_{\text{peak}} = \sqrt{2} \times \frac{4\pi 2 \times 20 \times 10^{-6}}{415 \times \left(2\pi 50/343\right)}\sqrt{\left(1+\left(2\pi 50/343\right)^2(0.1/2)^2\right)10^{(90/10)}}$$

$$= 0.0592 \text{ m}^3/s$$

Incoherent Line Source



$$\langle p^2\rangle = \rho c \frac{W}{4\pi}\sum_{n=-\infty}^{\infty} r_n^{-2}$$

where

$$r_n^2 = r_0^2 + (nb+d)^2$$

ENC p203

Incoherent Line Source (Cont.)

$$\langle p^2 \rangle \approx \rho c \frac{W}{4\pi} \sum_{n=-\infty}^{\infty} \left[r_0^2 + n^2 b^2 \right]^{-1}$$

$$\langle p^2 \rangle \approx \rho c \frac{W\pi}{4b^2} \left| \frac{b}{\pi r_0} \coth\left[\frac{\pi r_0}{b} \right] \right|$$

or, $\langle p^2 \rangle \approx \rho c \frac{W}{4br_0}$, $r_0 \geq b/2$ for error < 0.5 dB

$$L_p = L_w - 6 - 10\log_{10} r_0 - 10\log_{10} b$$

ENC p204

Incoherent Line Source (Cont.)

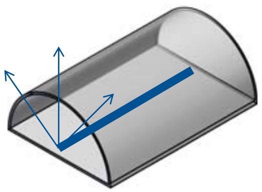
for $r<b/\pi$ only one source contributes

$$\langle p^2 \rangle = \rho c \frac{W}{4\pi r_0^2}$$

$$L_p = L_w - 11 - 20\log_{10} r_0$$

Coherent Line Source

- assumes source radiates cylindrical waves of surface area $2\pi r_0 bn$



- must satisfy $r_o/\lambda >> 1$

$$\langle p^2 \rangle = \rho c \frac{W}{2\pi b r_0}$$

$$L_p = L_w - 8 - 10\log_{10} r_0 - 10\log_{10} b + 10\log\left(\frac{\rho c}{400} \right)$$

ENC p205

Continuous Line Sources

- radiated power expressed in terms of power per unit source length, W_L

$$W_L = \lim_{b \to 0} (W/b)$$

- Thus replace W/b with W_L in all equations

Finite Length Line Source

- incoherent sources spaced b apart

$$\langle p^2 \rangle = \int_{x_1}^{x_2} \frac{W}{b} \frac{\rho c}{4\pi r^2} dx$$

$$x_1 = r_0 \tan \alpha_L, \quad x_2 = r_0 \tan \alpha_u,$$

$$r = r_0 / \cos \alpha, \quad x = r_0 \tan \alpha \ ,$$

$$dx = r_0 d\alpha \sec^2 \alpha$$

$$\langle p^2 \rangle = \rho c \int_{\alpha_L}^{\alpha_u} \frac{W}{b} \frac{\cos^2 \alpha}{4\pi r_0^2} r_0 \sec^2 \alpha d\alpha$$

or

$$\boxed{\langle p^2 \rangle = \rho c \frac{W}{4\pi r_0 b} \left[\alpha_u - \alpha_L\right]}$$

ENC p206

valid if no. of sources > 3 and $r_o > (b/2) \cos \alpha_L$

In General

- infinite number of *incoherent* sources in length D, the *total power*, W_t

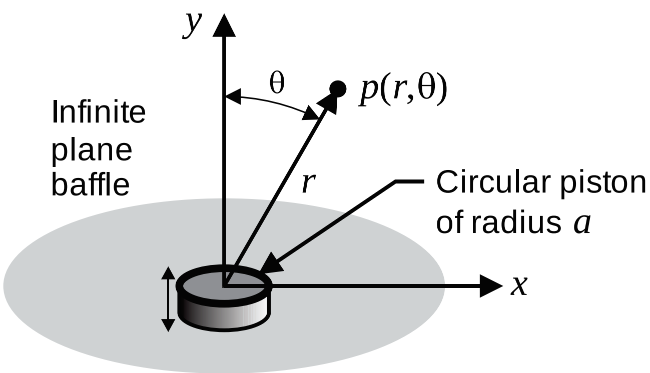
$$\boxed{\langle p^2 \rangle = \rho c \frac{W_t}{4\pi r_0 D} \left[\alpha_u - \alpha_L\right]}$$

ENC p206

- *coherent* source of length D
  - sound pressure level will be the same as for an incoherent source for large distances away compared with the source length.
  - At distances less than $D/2$ away, the sound pressure level (normal to the centre of the source) will be the same as for an infinite coherent source.
  - In between, the sound pressure level will be between the two levels

Baffled Piston

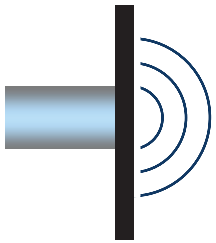
- Consider an oscillating piston mounted in an infinite baffle.

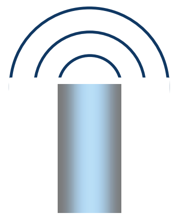


Baffled Piston

- A baffled piston is a good model for:
  - Loudspeakers

  - Sound radiated from the end of a pipe in a wall (or with a slight modification from a chimney / car exhaust)

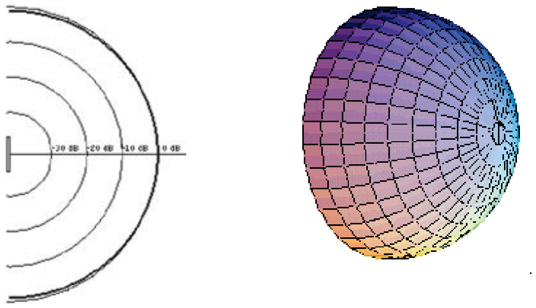




Baffled Piston

Low frequency (ka<1) $k = 2\pi f / c$, hence $ka = \frac{2\pi f}{c}\, a$

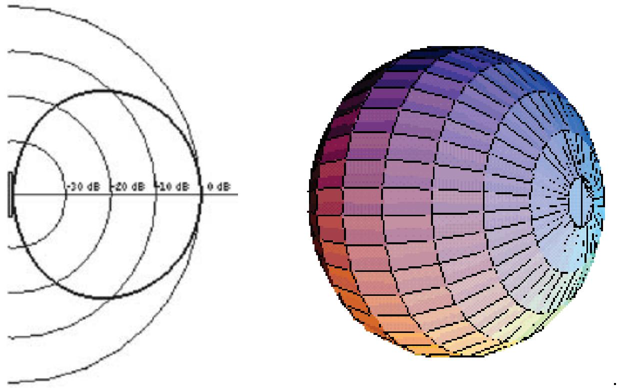
Radiates equally in all directions –
"omni-directional"



http://www.acs.psu.edu/drussell/demos/baffledpiston/baffledpiston.html

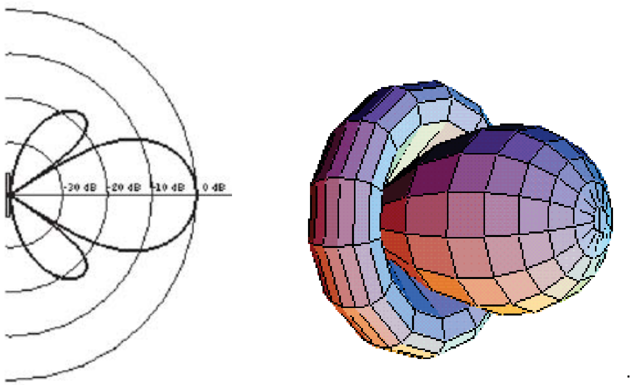
Baffled Piston

Medium frequency (ka>1)
Starts to become directional



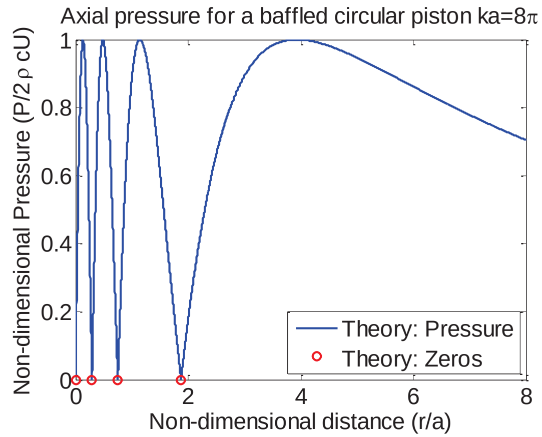
Baffled Piston

High frequency (ka>>1)
Very directional, side lobes start to appear.



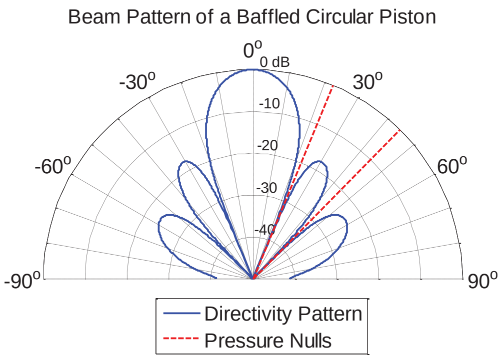
Baffled Piston

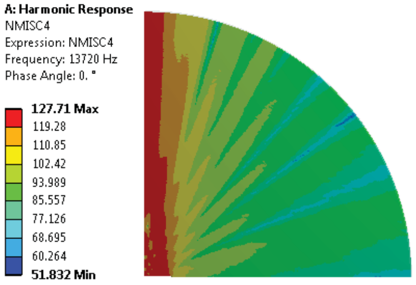
- If you were to examine the pressure on the axis of the piston, it would vary between maximums and minimums.
- There are pressure zeros directly in-front of the piston. (weird)



Baffled Piston

- At low frequencies it radiates omni-directionally.
- At higher frequencies the angular directivity also has pressure zeros at certain angles.

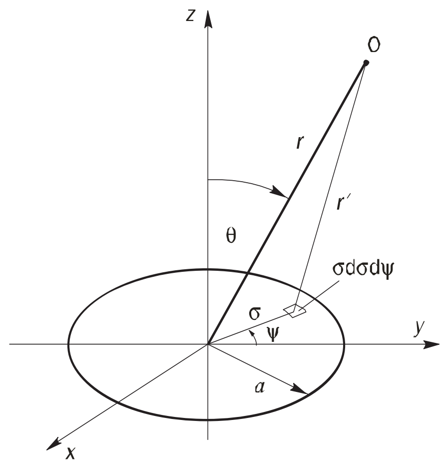




Piston in an Infinite Plane Baffle

- constructed as an integral of in-phase simple sources
- begin with spherical wave solution to wave equation

$$\varphi = \frac{A}{r} e^{j(\omega t - kr)}$$



Piston Radiation (Cont.)

$$u = -\frac{\partial \varphi}{\partial r} = \frac{A}{r^2} e^{j(\omega t - kr)} + \frac{jkA}{r} e^{j(\omega t - kr)}$$

$$p = \rho \frac{\partial \varphi}{\partial t} = \frac{j\rho ck A}{r} e^{j(\omega t - kr)}$$

Using previous simple source analysis and for $kr << 1$,

$$A = \frac{Q_0}{4\pi}$$

let $r = a_i$ be the radius of the spherical source and assume $ka_i << 1$

$$u = \frac{(1 + jka_i)}{a_i^2} \frac{Q_0}{4\pi} e^{j(wt - ka_i)} \approx \frac{Q_0}{4\pi a_i^2} e^{j(wt - ka_i)}$$

$$p = \frac{j\rho ck\, Q_0}{4\pi r} e^{j(\omega t - kr)}$$

Piston Radiation (Cont.)

- incremental pressure contribution from each infinitessimally small source at observation point r_0 (remembering that $Q_0 = 2U\sigma\, d\sigma d\psi$), the 2 being there due to an image source in the baffle and rigid piston

$$dp = \frac{j\rho ckU}{2\pi r'} \sigma d\sigma d\psi e^{j(\omega t - kr')}$$

where U is the piston velocity amplitude

- distance r' $\quad r' = \left[r - \sigma^2 - 2r\sigma \sin\theta \cos\psi\right]^{1/2}$

- for $r >> a$ (far field)

$$r' \approx r - \sigma \sin\theta \cos\psi$$

$$p = \frac{j\rho ck}{2\pi r} U e^{j(\omega t - kr)} \int_0^a \sigma d\sigma \int_0^{2\pi} e^{jk\sigma \sin\theta \cos\psi} d\psi$$

Piston Radiation (Cont.)

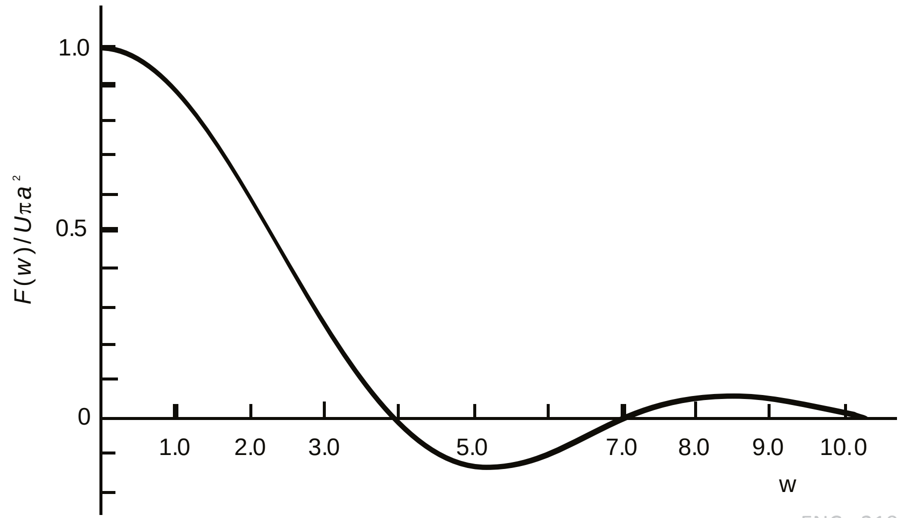
$$p = \frac{j\rho c k}{2\pi r} F(w) e^{j(\omega t - kr)}; \qquad w = ka \sin\theta$$

$$F(w) = U\pi a^2 [J_0(w) + J_2(w)] = U\pi a^2 \, 2 \frac{J_1(w)}{w}$$

ENC p209

where F is plotted on the following page.

Piston Radiation



ENC p210

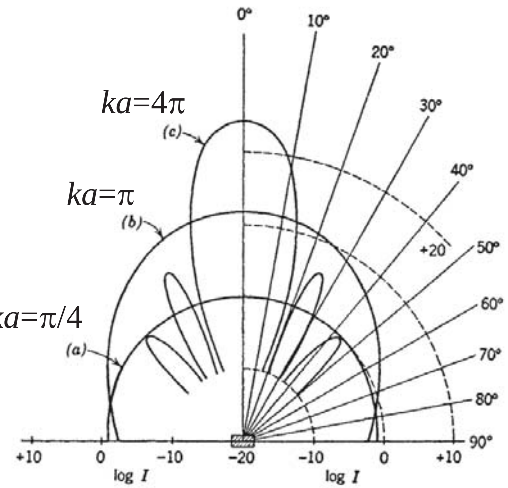
Piston Radiation

- acoustic intensity in the far field is:

$$I = \frac{\rho c k^2}{8\pi^2 r^2} F^2(w)$$

ENC p209

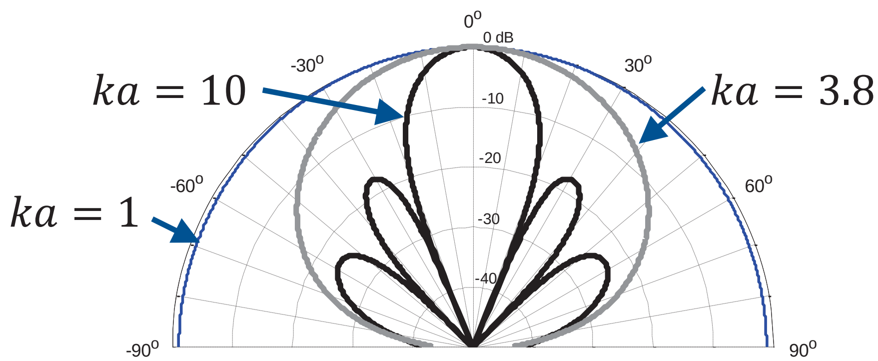
The directivity plot shows the change in radiation pattern as the ka value increases.



Piston Radiation

- Radiation pattern changes from
  - omni-directional $ka = 1$
  - first lobe appears about $ka = 3.8$

Beam Pattern of a Baffled Circular Piston



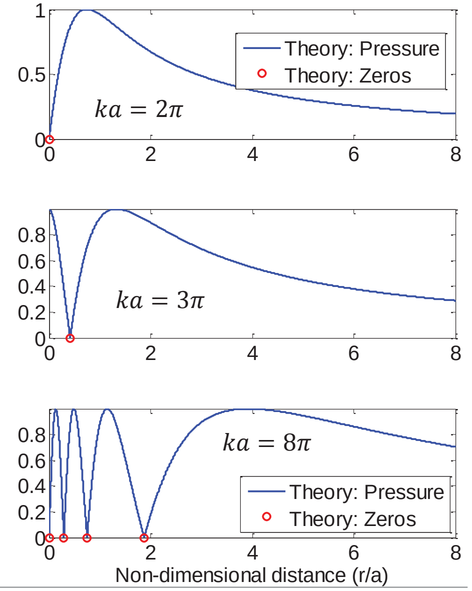
Piston Near Field On-axis Sound Pressure

Remember previously we said that there are pressure nulls on axis.
This is how it is calculated....

On-axis sound pressure nulls in near field exist at axial distances r

$$r = \frac{(a/\lambda)^2 - n^2}{2n/\lambda}$$

ENC p211



Piston Radiation

- Another thing we are often interested in knowing is how much the piston will vibrate depending on the applied force.... Which is impedance...
- piston radiation impedance, $Z = F_R / U$
- The force is calculated by integrating the pressure over the surface area of the piston:

$$F_R = \iint_S p_s dS'$$

$$= \frac{j\rho c k U e^{j\omega t}}{2\pi} \int_0^a r dr \int_0^{2\pi} d\varphi \int_0^a \sigma d\sigma \int_0^{2\pi} \frac{e^{-jkh}}{h} d\psi$$

$$Z_R = \frac{F_R}{Ue^{j\omega t}} = \rho c \pi a^2 \left[R_R(2ka) + jX_R(2ka)\right]$$

ENC p213

Piston Radiation Impedance (Cont.)

- where $z = 2ka$.

$$R_R(z) = \frac{z^2}{2\times4} - \frac{z^4}{2\times4^2\times6} + \frac{z^6}{2\times4^2\times6^2\times8} - \dots$$

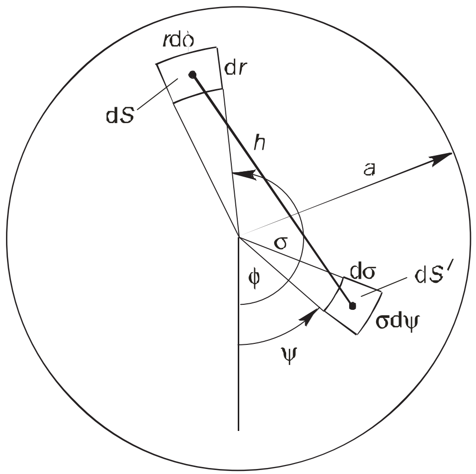
$$X_R(z) = \frac{4}{\pi}\left[\frac{z}{3} - \frac{z^3}{3^2\times5} + \frac{z^5}{3^2\times5^2\times7} - \dots\right]$$

- radiated power

$$\boxed{W = \frac{1}{2}\mathrm{Re}\{Z_R\}U^2 = \frac{1}{2}\rho c R_R \pi a^2 U^2}$$

ENC p213

Piston Radiation (Cont.)



ENC p212

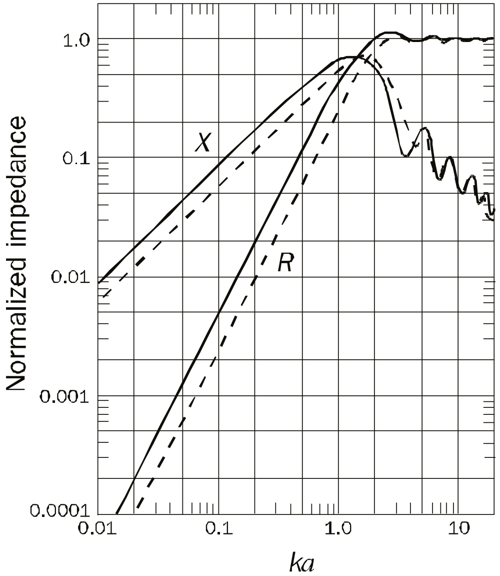
Piston Radiation Impedance (Cont.)

Plotted on a log-log scale

k = wavenumber in rad/s
a = radius of piston in m

Solid line:
piston in an infinite baffle (i.e. a large wall)

Dashed line:
piston at end of long tube

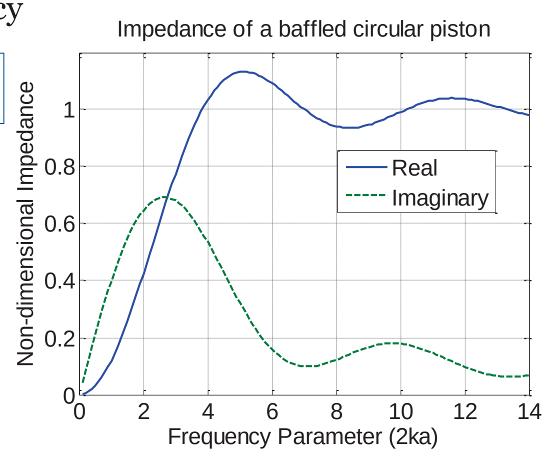


ENC p214

Baffled Piston

- The mechanical impedance of a piston varies with frequency

Plotted on a linear scale

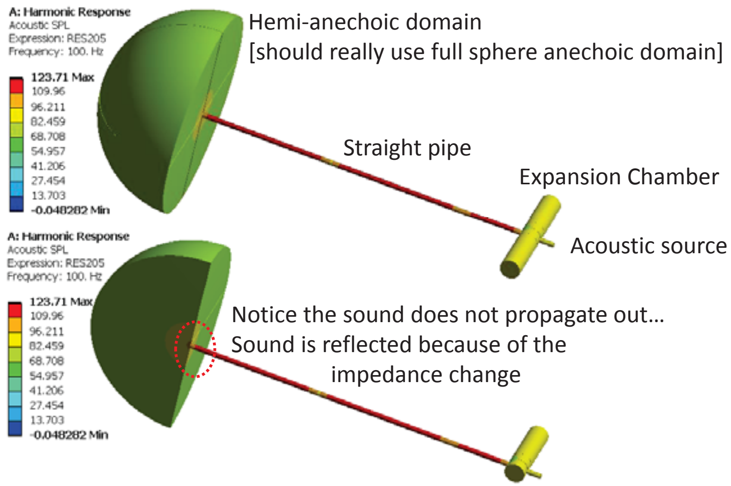


Example

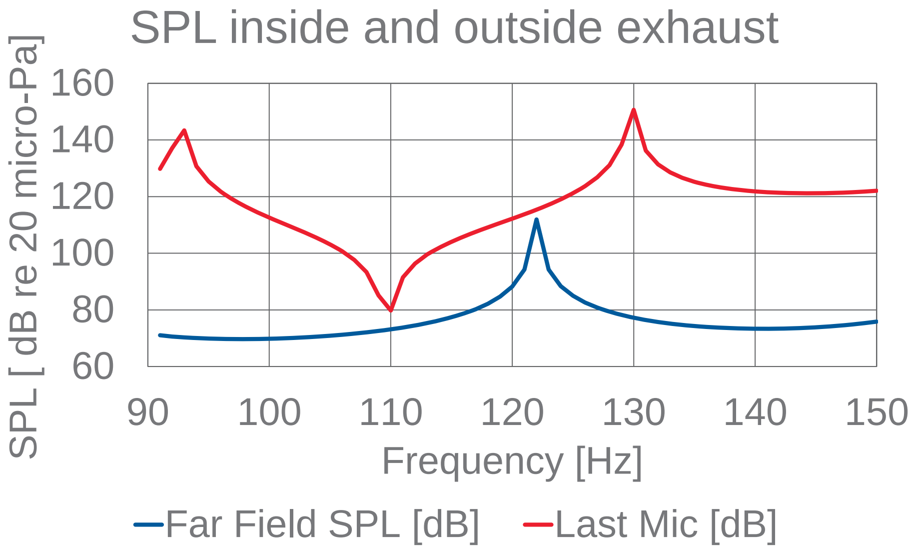
- Exhaust pipe with an expansion chamber silencer radiates into an open field

Example

- Can model acoustics using ANSYS

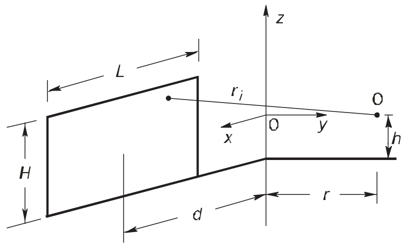


Example



Incoherent Plane Radiator

- plane divided into array of incoherent point sources
  - assume cosine directivity



$$D = r\left(x^2 + z^2 + r^2\right)^{-1/2} = r / r_i$$

- sound pressure at O due to source I

$$\left\langle p_i^2 \right\rangle = \frac{W_i \rho c D}{2\pi r_i^2}$$

ENC p216

  - assumes hemispherical radiation

Incoherent Plane Radiator (Cont.)

- sound power W_i of each source in terms of total power, W

$$W_i = \frac{W dx dz}{HL}$$

- Substituting for W_i and r_i in equation for $\left\langle p_i^2 \right\rangle$ and integrating over area of source

$$\left\langle p^2 \right\rangle = \frac{\rho c W}{2\pi HL} \int_{-h}^{H-h} dz \int_{d-L/2}^{d+L/2} \left[x^2 + z^2 + r^2\right]^{-3/2} r dx$$

- Introduce $\alpha = H/L$, $\beta = h/L$, $\gamma = r/L$ and $\delta = d/L$:

Incoherent Plane Radiator (Cont.)

$$\left\langle p^2 \right\rangle = \frac{\rho c W}{2\pi HL}\left[\tan^{-1}\frac{(\alpha-\beta)(\delta+1/2)}{\gamma\sqrt{(\alpha-\beta)^2+(\delta+1/2)^2+\gamma^2}} + \tan^{-1}\frac{\beta(\delta+1/2)}{\gamma\sqrt{\beta^2+(\delta+1/2)^2+\gamma^2}} - \tan^{-1}\frac{(\alpha-\beta)(\delta-1/2)}{\gamma\sqrt{(\alpha-\beta)^2+(\delta-1/2)^2+\gamma^2}} - \tan^{-1}\frac{\beta(\delta-1/2)}{\gamma\sqrt{\beta^2+(\delta-1/2)^2+\gamma^2}}\right]$$

- on axis of symmetry, $\delta = 0, \quad \beta = 0.5\,\alpha$

$$\boxed{\left\langle p^2 \right\rangle = \frac{2\rho c W}{\pi HL}\tan^{-1}\left[\frac{HL}{2r\sqrt{H^2+L^2+4r^2}}\right]}$$

ENC p217

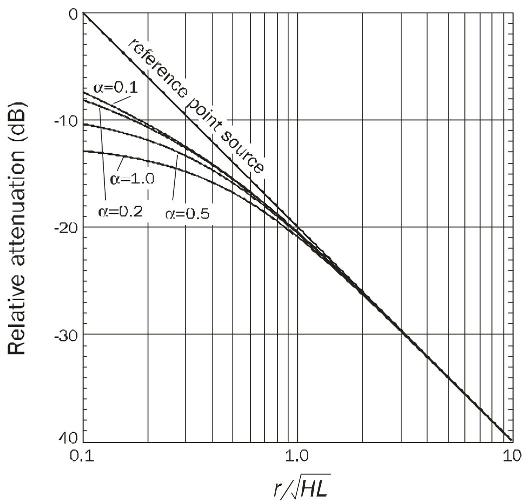
Incoherent Plane Radiator (Cont.)

- at large r,

$$\langle p^2 \rangle = \frac{\rho c W}{2\pi r^2}$$

- as r tends to zero,

$$\langle p^2 \rangle = \frac{\rho c W}{HL}$$



ENC p218

Sound Radiated From Walls Of A Building

- average sound radiated by 4 walls and roof

$$\boxed{L_p = L_{wt} - 10\log_{10} S + 10\log_{10}\frac{\rho c}{400}}$$

ENC p218

- sound radiated normal to wall i

$$L_{p_i} = L_p + L_{w_i} - \left(L_{w_t} - 7\right)$$

(Note that $10\log_{10}(5) = 7$)

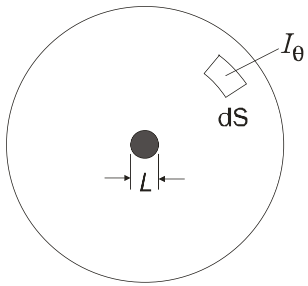
Leads to concept of directivity… We've already seen radiation patterns, but lets formalise it.

Source Directivity

- far field concept,

$$r >> \pi L^2/2\lambda$$

$$W = \int_S I_\theta \mathrm{d}S$$



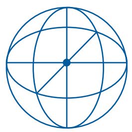
- Mean intensity $\langle I \rangle = \dfrac{W}{4\pi r^2}$
- Directivity factor in direction (θ,ψ) $D_\theta = \dfrac{I_\theta}{\langle I \rangle}$

ENC p219

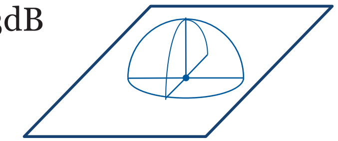
- directivity index, $DI_\theta = 10\log_{10} D_\theta$

Directivity

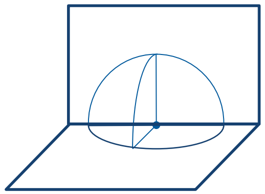
- Free Space
D=1
DI=0dB



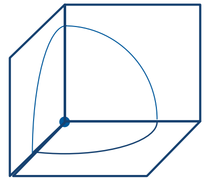
- Centre large plane
D=2
DI=3dB



- Two planes
D=4
DI=6dB



- Three Planes
D=8
DI=9dB



Reflection Effects

- constant power source

Situation	*D*	*DI* (dB)
free space	1	0
centered in a large flat surface	2	3
centered at the edge formed by the junction of two large flat surfaces	4	6
at the corner formed by the junction of three large flat surfaces	8	9

$$W = 4\pi r^2 I / D$$

ENC p221

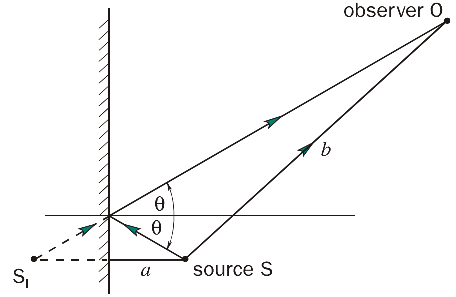
Source Near A Reflecting Plane

- for a simple, *constant power source* within λ/10 of surface or $a << b$

$$\boxed{W = I\frac{4\pi r^2}{D} = \left\langle p^2 \right\rangle \frac{4\pi r^2}{\rho c D}}$$

ENC p220

- for sources further from the surface than, $\lambda/10$, and where $a >> b$ or $a \approx b$, add intensity contributions from source and its image



Observer and Source Near a Reflecting Plane

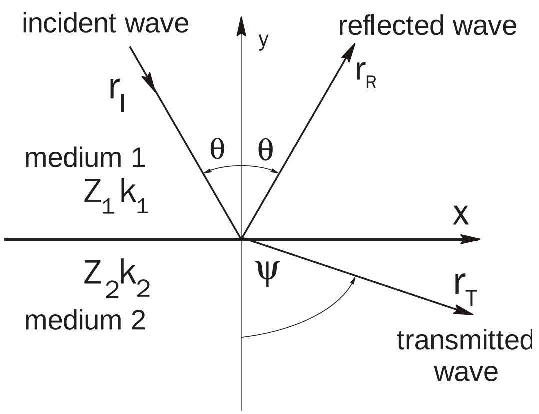
- coherent addition (i.e. add pressures not intensities) of direct and reflected waves if
  - no turbulence
  - no temperature gradients
  - source / observer separation distance large compared to distance from reflecting surface
  - And if the real part of the impedance of the reflecting plane Zs satisfies $\mathrm{Re}\{Z_S\}/\rho c > L/(H+h)$
  - Then coherent addition of source and image results in a 6dB rather than 3dB increase is SPL over that for a free field. Hence, directivity factors need to be multiplied by 2.

Reflection and Transmission at Interface Between 2 Media

- choice of 3 assumptions re interface
  - locally reactive if $f < 10^{-3} R_1$
  - extended reactive
  - modally reactive not usually used for porous surfaces
- choice of two assumptions re incident wave
  - plane (relatively simple analysis)
  - spherical (complicated analysis)
- reflection from porous earth interface
  - earth characterised by 2 quantities which can be calculated from surface flow resistance, R_1 (see App. 3 in text)
  - impedance Z_m
  - propagation constant k_m

ENC p222

Plane Wave Reflection and Transmission



$$k_{1x}=k_1\sin\theta,\quad k_{1y}=k_1\cos\theta$$

$$k_{2x}=k_2\sin\psi,\quad k_{2y}=k_2\cos\psi$$

$$p_I=A_I e^{-jk_1 r_I}=A_I e^{-j\left(k_{1x}x-k_{1y}y\right)}$$

$$p_R=A_R e^{-j\left(k_{1x}x-k_{1y}y\right)}$$

$$p_T=A_T e^{-j\left(k_{2x}x-k_{2y}y\right)}$$

ENC p225

- at $y = 0$, continuity of pressure requires $p_I + p_R = p_T$
  - thus, $A_I e^{-jk_{1x}x}+A_R e^{-jk_{1x}x}=A_T e^{-jk_{2x}x}$
  - and, $A_I + A_R = A_T$
  - so, $(A_I+A_R)\left(e^{-jk_{1x}x}-e^{-jk_{2x}x}\right)=0$
  - and thus, $k_{1x} = k_{2x}$
- above result leads to Snell's Law $n=\dfrac{c_2}{c_1}=\dfrac{k_1}{k_2}=\dfrac{\sin\psi}{\sin\theta}$
- continuity at particle velocity at $y = 0$ $\dfrac{p_I-p_R}{Z_1}\cos\theta=\dfrac{p_T}{Z_2}\cos\psi$
- use of preceding equations gives

$$A_I\left[\frac{\cos\theta}{Z_1}-\frac{\cos\psi}{Z_2}\right]=A_R\left[\frac{\cos\theta}{Z_1}+\frac{\cos\psi}{Z_2}\right]$$

- for air / earth interface
  $Z_1 = \rho c$
  $Z_2 = Z_m$
- Reflection coefficient, R_p given by

$$\boxed{R_p=\frac{A_r}{A_I}=\frac{Z_m\cos\theta-\rho c\cos\psi}{Z_m\cos\theta+\rho c\cos\psi}}$$

- when , $k_m = k_2 >> k_1 = k$
  ψ tends to zero

$$\boxed{R_p=\frac{Z_m\cos\theta-\rho c}{Z_m\cos\theta+\rho c}}$$

ENC p226

which is for a *locally reactive surface*

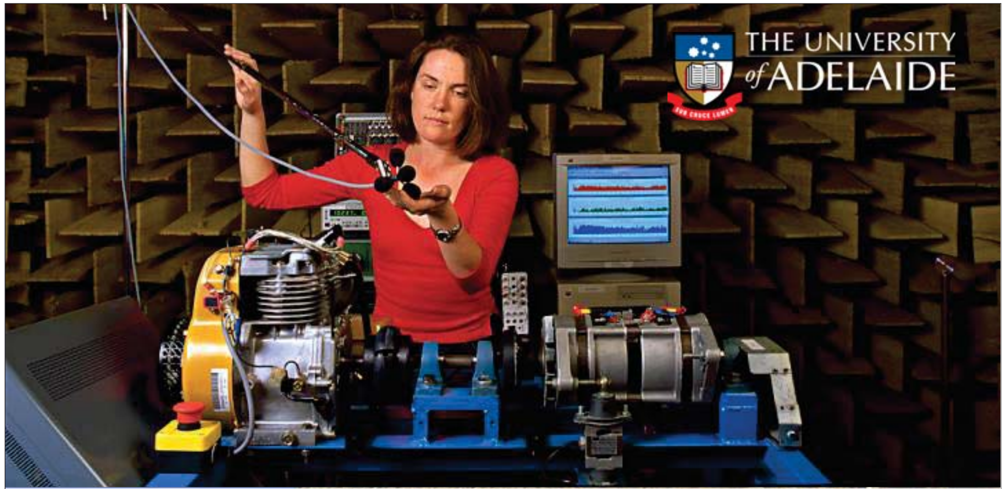
- Z_m calculated from surface flow resistance, R_1 as in App. C
  - see text, Ch. 5 for table of R_1 values

Summary

Covered LOTS of acoustic sources types...

I bet you are surprised by the number of acoustic source types.

- Monopoles, dipoles and quadrupoles
  - Combinations can be used to model a complex sound field
- Line sources
  - Several configurations
- Baffled piston
- Coherent and incoherent plane sources
- Directivity and reflection effects
- Plane wave reflection and transmission



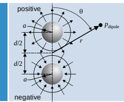
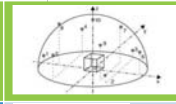
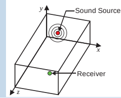
Carl Howard

Sound Power

Course Content – How it fits

We know a bit about sound sources.
Now we need to learn a bit more about
Sound Power
before proceeding to

- Sound indoors
- Outdoor sound propagation

TOPIC	
Sound Sources	
Sound Power	
Indoor	
Outdoor	

Contents

- Effects of nearby surfaces on radiated sound power.
- Near Field, Far Field
- Sound Intensity Measurements
- Methods of Determining Sound Power Level

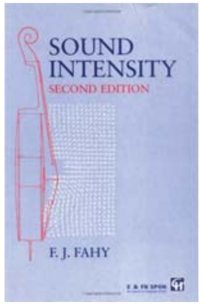
Learning Outcomes

- Understand the acoustic regions of near field and far field are based on characteristic dimensions, frequency, and distance from the source.
- Understand the various ways that sound power can be determined.

Sound Power

- We've already seen sound power many times in the course.
- There is a bit more we need to know…
- See Chapter 6 of text book.

Excellent reference book on the topic -
"Sound Intensity", 2nd Edition
Frank J. Fahy



Sound Power

- Allows comparison between machines.
- Allows verification of specifications
- Once Sound Power Levels are specified, if we know the directivity, we can predict the Sound Pressure Level anywhere.
  i.e., L_w + directivity , can predict L_p anywhere.

Sound Power

- <u>Constant Power</u> Source (omnidirectional) $I = \dfrac{DW}{4\pi r^2}$

$$\boxed{L_p = L_w + 10\log_{10} D - 10\log_{10}\left(4\pi r^2\right) + 10\log_{10}\frac{\rho c}{400}}$$

ENC p263

L_p = sound pressure level (dB re 20µPa)
L_w = sound power level (dB re 10^{-12}W)
D = directivity (no units)
r = distance from source (m)
ρ = density (kg / m^3)
c = speed of sound (m/s)
I = intensity (W/m^2)
W = sound power (W)

- with one plane surface - same power radiated into ½ area so 3dB increase in L_p.

Sound Power

- <u>Constant Volume</u> Source $I = \dfrac{D^2 W}{4\pi r^2}$

ENC p262

$$\boxed{L_p = L_w + 20\log_{10} D - 10\log_{10}\left(4\pi r^2\right) + 10\log_{10}\frac{\rho c}{400}}$$

ENC p263

L_p = sound pressure level (dB re 20µPa)
L_w = sound power level (dB re 10^{-12}W)
D = directivity (no units)
r = distance from source (m)
ρ = density (kg / m^3)
c = speed of sound (m/s)
I = intensity (W/m^2)
W = sound power (W)

- with one plane surface, get pressure doubling so 6dB increase in L_p.

Sound Power

- <u>Constant Pressure</u> Source

$$I = \frac{W}{4\pi r^2}$$

Same pressure, regardless of presence of plane surface
so L_p unchanged

W = free field power
D = directivity due to reflecting surfaces

Above 2 Eqs. only hold if source < λ/10 from reflecting surface. Effect reduces to constant power source equation at λ/2 for a band of noise and 2λ for a tone.

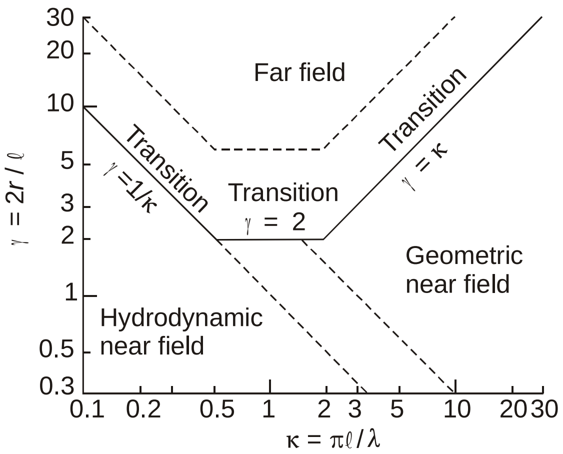
For broadband (⅓ octave or greater) noise add contribution from direct and reflected waves incoherently.

Sound Power

	Increase in L_p and Intensity (dB) for Constant:			
	D	Power	Velocity	Pressure
Free space	1	0	0	0
Flat plane	2	3	6	0
Junction of 2 planes	4	6	12	0
Corner of 3 planes	8	9	18	0

Acoustic Regions

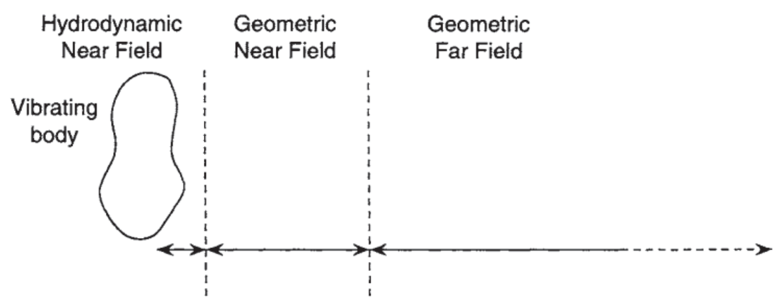
Classified based on
- λ wavelength
- r distance from object
- ℓ characteristic dimension of object (typically the largest dimension)
- Some sources have no hydrodynamic near field.



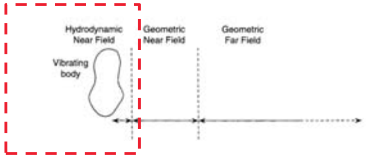
ENC p266

Acoustic Regions

- The sound field radiated by a source in a free field may be divided into three regions:
  - **hydrodynamic near field**,
  - **geometric near field**,
  - **far field**.

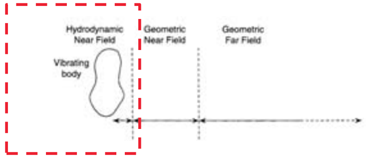


Hydrodynamic Near Field

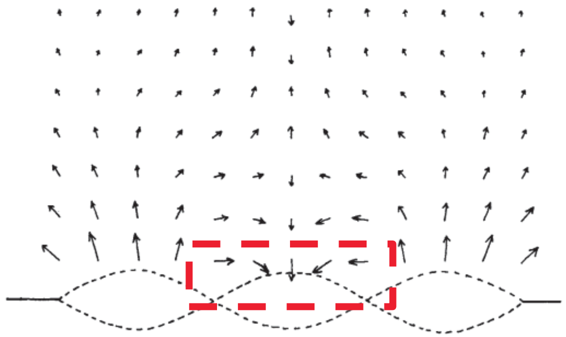


- The hydrodynamic near field is the region immediately adjacent to the vibrating surface.
- This region extends outward much less than one wavelength.
- This region has fluid motion that is not directly associated with sound propagation.
- Acoustic pressure is out of phase with local particle velocity (no measurement of sound power possible).

Hydrodynamic Near Field

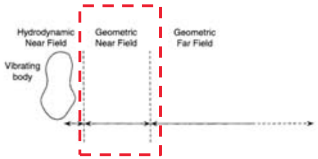


- For example, region next to a vibrating plate has regions where particle velocity is *tangential* to the plate.



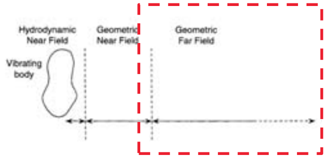
From: Fahy, Foundations of Engineering Acoustics, Fig 6.2

Geometric Near Field



- The geometric near field is the region where interference between contributing waves from various parts of the sound source lead to sound pressure levels that do not decrease monotonically at a rate of 6dB per doubling of the distance from the source.
- Local maxima and minima are common. This effect is greater for pure tones than for bands of noise.
- Particle velocity and pressure are in phase which means that a sufficient number of sound pressure level measurements in the geometric near field will enable the sound power to be determined.

Far Field



- The far field is the region beyond the geometric near field where there are no reflecting surfaces and sound pressure level decreases monotonically at 6dB per doubling of the distance from the source.

- Far field characterised by:

$$\text{FAR FIELD:}\quad \begin{array}{ll} r >> \lambda/2\pi & ; \lambda >> 1/K \\ r >> \ell & ; \gamma >> 2 \\ r >> \pi\,\ell^2/2\lambda & ; \gamma >> K \end{array}$$

ENC p265

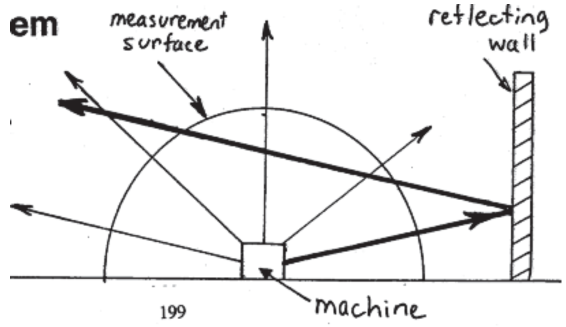
Sound Intensity Measurement

- In order to measure the sound power of an object, we need to measure the sound intensity, and then integrate over an imaginary surface.
- Interested in determining magnitude and direction of acoustic energy flow.
- See text, Ch. 3 and Ch. 1 for details and definitions of intensity.
- Measurements can be made in near field of machine.
- Sound power determined by averaging sound intensity over an imaginary surface surrounding the source
- Nearby reflecting surfaces cause no problems

Sound Intensity

- The reason that reflecting surfaces do not cause a problem is that the sound energy is still captured by the measurement process and cancels out when needed.

$$W = I_n S$$



Sound Intensity Measurement

- Can obtain good measurements by manual or robotic scanning of intensity probe over the measurement surface or by taking stationary measurements at a number of fixed locations on the measurement surface as described in ISO 9614 standards.
- intensity is a vector quantity so only measures net energy flow through surface
  - thus measurements can be made in presence of other noise sources but contribution of these other sources to L_p cannot be more than 10dB above source to be measured due to instrumentation limitations.
- Measurement errors
  - mainly associated with difficulty in accurately measuring u and p-u phase.

Sound Intensity Measurement

For information only...

- When measuring sound intensity, the best spacing between the microphones depends on the frequency range to be measured.

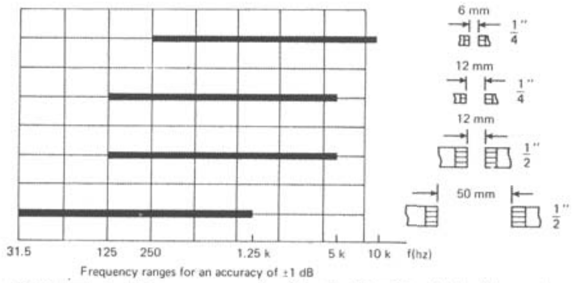
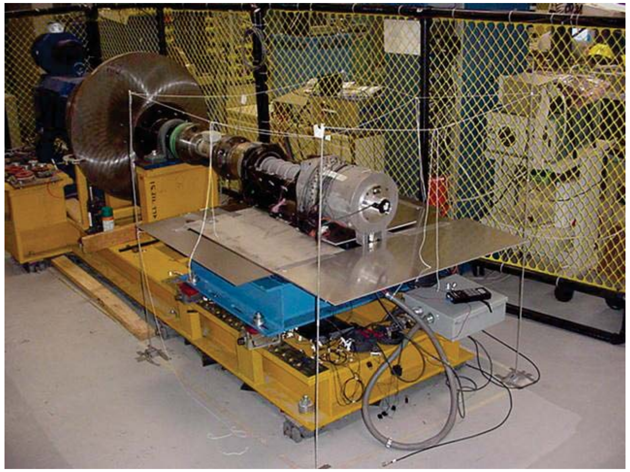


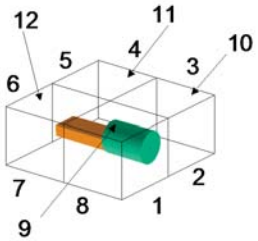
FIG. 14.4 Applicable frequency ranges for a Type (p-p) intensity probe for different microphone diameters and various spacings between microphone diaphragms. Some probe manufacturers insert a solid space between the microphone diaphragms. The upper frequency limit is governed by the finite difference approximation error, i.e., where the microphone spacing becomes comparable to the wavelength. The lower frequency limit is controlled by phase errors between microphones, which approach true field phase shifts at low frequency. *(After W. P. Waser and M. J. Crocker.[1])*

Harris, C.M., "Handbook of Acoustic Measurements and Noise Control: Third Edition", Acoustical Society of America, New York, USA 1998.

Examples ....

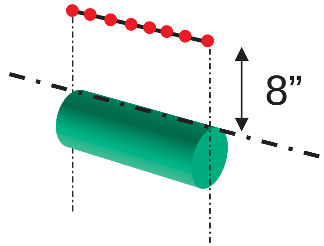
- Measurement of sound power of the GEN2 elevator motor.





Examples ....

- Measurement of sound power of the GEN2 elevator motor.

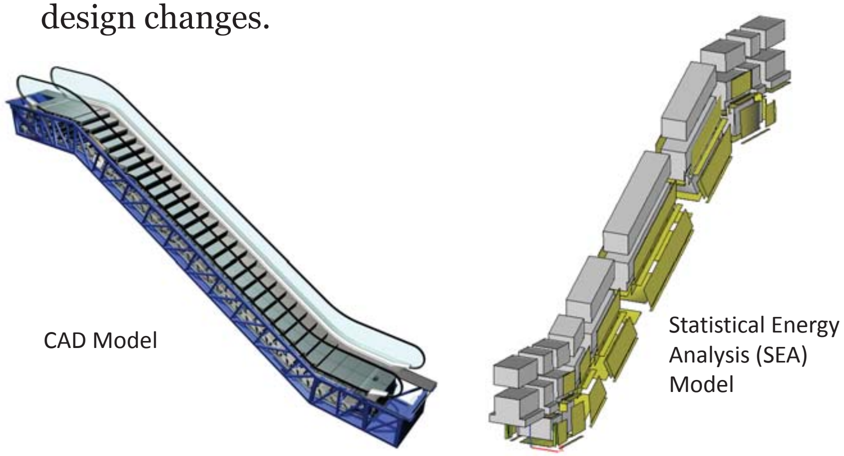


Examples ....

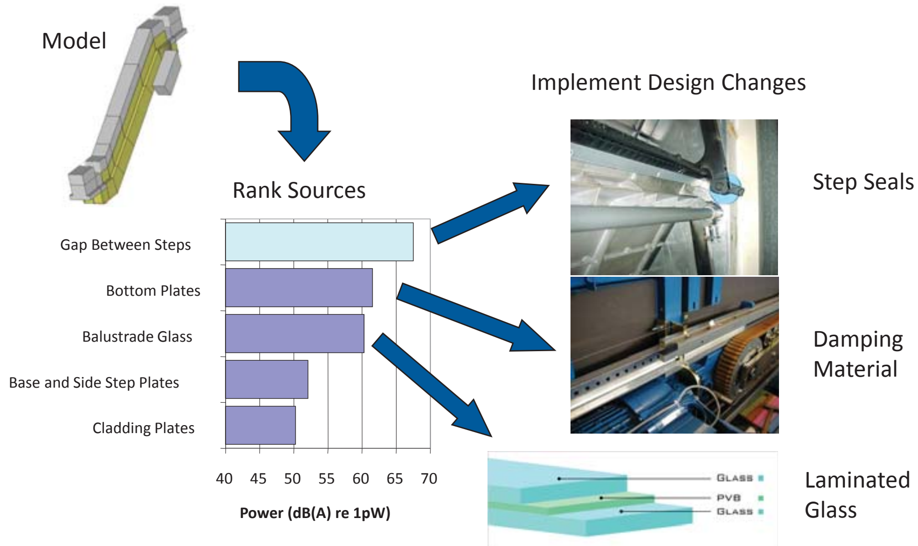
- Measurement of sound power of an escalator motor.

Example ..

- Can use sound power in computer models to guide design changes.



AutoSEA Model Used to Rank Noise Sources



Sound Power from SPL Measurements

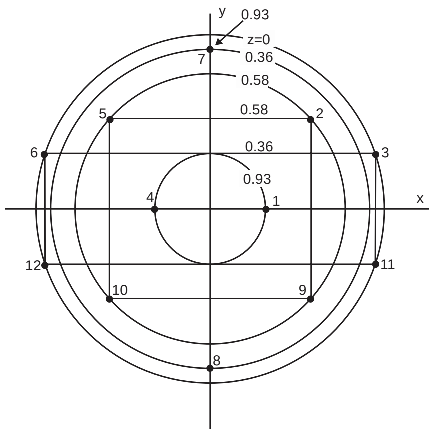
Many ways to determine sound power:

- Laboratory measurements
  - using anechoic or reverberant rooms
- Far field pressure measurements
  - substitution method
  - room absorption source
    - reference sound source
    - reverberation times
  - two separate measurement surfaces
- Near field pressure measurements
- Surface vibration measurements
  (not discussed here - see text book)

Sound Power from SPL Measurements

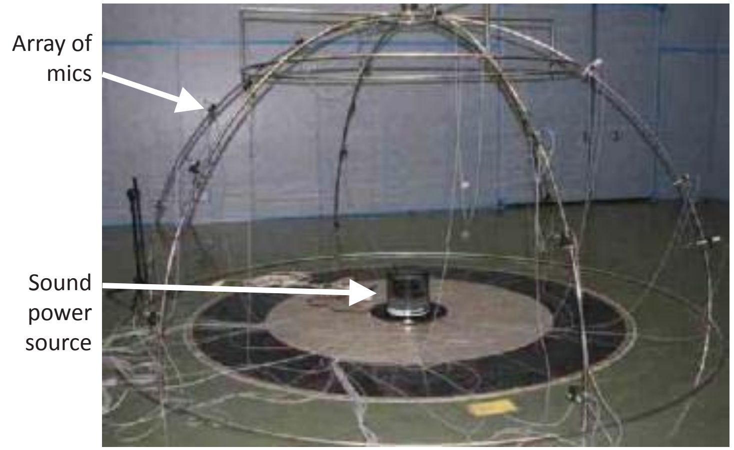
ANECHOIC ROOM
(free and semi-free field)

- sound power determined from far field measurements in a grid pattern.
- See textbook for details.



ENC p271

Sound Power from SPL Measurements



Measurement in a Diffuse Field

- required minimum room volume
  - octave bands - $1.3\lambda^3$
  - 1/3 octave bands - $4.6\lambda^3$
  - discrete tone $\left[\frac{2000}{f}\right]^2 T_{60}$
- optimum room dimensions - 2:3:5
- required mean absorption coefficient
  - mid to high frequencies, $\bar{\alpha} < 0.06$
  - low frequencies, $\bar{\alpha} \approx 0.15$

Measurement in a Diffuse Field

Two measurement methods

1) reverberation time measurement

$$L_w = L_p + 10\log_{10} V - 10\log_{10} T_{60} + 10\log_{10}(1 + S\lambda/8V) - 13.9 \quad (\text{dB re } 10^{-12}\ \text{W})$$

ENC p275

L_w = sound power level in dB
L_p = sound pressure level in dB
V = volume of room in m^3
T_{60} = reverberation time in seconds
S = surface area of room in m^2
λ = wavelength of sound in m

Measurement in a Diffuse Field

2) substitution method

$$L_w = L_{wR} + (L_p - L_{pR})$$

ENC p275

L_w = sound power level in dB
L_{wR} = sound power level of reference source in dB
L_p = sound pressure level in room in dB
L_{pR} = sound pressure level in room due to reference source in dB

- sound pressure level, L_p
  - space averaged at distances > $\lambda/4$ from walls, floor and ceiling
  - excludes direct field

METHOD 1: ROOM ABSORPTION MEASUREMENT

A. Reference Sound Source

- Placed in same room as equipment to be measured.
- L_{pR} measured on a test hemisphere of radius r surrounding the reference source.
- Area of hemisphere, $S_H = 2\pi r^2$
- L_{p2} (value of L_{pR} in free field) calculated from reference source power level L_{wR}

$$L_{p2} = L_{wR} - 20\log_{10} r - 8 + 10\log_{10}\frac{\rho c}{400}$$

- L_{wR} in terms of L_{pR} is

$$L_{wR} = L_{pR} - 10\log_{10}\left[\frac{D}{4\pi r^2} + \frac{4}{R}\right] - 10\log_{10}\frac{\rho c}{400}$$

- substituting for L_{wR} gives

$$L_{p2} + 10\log_{10} S_H = L_{pR} - 10\log_{10}\left[\frac{1}{S_H} + \frac{4}{R}\right]$$

- Rearranging gives

$$\frac{4}{R} = \left[10^{(L_{pR}-L_{p2})/10} - 1\right] / S_H \quad (\mathrm{m}^{-2})$$

- R is the room constant proportional to the room absorption.
- Sound power level of noise source is

$$\boxed{L_w = L_p - 10\log_{10}\left[\frac{D}{4\pi r^2} + \frac{4}{R}\right] - 10\log_{10}\frac{\rho c}{400} \quad (\mathrm{dB\ re\ } 10^{-12}\,\mathrm{W})}$$

B. Reverberation time

Room absorption, $4/R$ determined using:

$$\boxed{\frac{4}{R} = \frac{4 c_o T_{60}}{55.3\,V}}$$

R = room constant in m²
T_{60} = reverberation time in s
c = speed of sound in m/s
V = volume of room in m³

(See Chapter 7 for more information)

Substitution Method

METHOD 2. Substitution

- Measure L_p on a test hemisphere surrounding the machine.
- Replace machine with reference source of power L_{wR} and measure L_{pR} on the same test hemisphere.
- Machine sound power is then

$$\boxed{L_w = L_{wR} + L_p - L_{pR} \quad \mathrm{dB}}$$

ENC p275

L_w = sound power level in dB
L_{wR} = sound power level of reference source in dB
L_p = sound pressure level in room in dB
L_{pR} = sound pressure level in room due to reference source in dB

Substitution Method

- This is clever in its simplicity …

$$L_w = L_{wR} + \boxed{L_p - L_{pR}} \quad \mathrm{dB}$$

This just means the change in SPL that occurs from when the sound source is on compared to when the reference sound power source is used.

…. then we just add this delta in SPL to the sound power level of the reference source…

and presto, that is the sound power level of the device under test.

Reference Sound Sources

Remember this from Lecture on Instrumentation

Bruel and Kjaer Reference Sound Source - Type 4204

Acculab RSS Type 600

Application of Substitution Method

1. Put the reference sound power source in the room and measure the SPL. We measure L_{pR} and know L_{wR} .

2. Replace with the device under test and measure SPL. We measure L_p .

3. The sound power of the device under test is then

$$L_w = L_{wR} + \underbrace{L_p - L_{pR}}_{\Delta L_p} \text{ dB}$$

METHOD 3. Two Hemispherical Test Surfaces

- Smaller test surface area S1
- Larger test surface area S2
- Measure L_{p1} on smaller, L_{p2} on larger

$$L_w = L_{p1} - 10\log_{10}\left[\frac{D}{4\pi r_1^2} + \frac{4(1-\bar{\alpha})}{S\bar{\alpha}}\right]$$

$$= L_{p2} - 10\log_{10}\left[\frac{D}{4\pi r_2^2} + \frac{4(1-\bar{\alpha})}{S\bar{\alpha}}\right]$$

- Rearranging gives

$$L_w = L_{p2} - 10\log_{10}\left[S_1^{-1} - S_2^{-1}\right] + 10\log_{10}\left[10^{(L_{p1}-L_{p2})/10} - 1\right]$$

ASSUMPTIONS FOR FAR FIELD DETERMINATION OF SOUND POWER

- Background noise negligible or taken into account by repeating measurements with machine and reference source turned off.
- Power measurement will not be extrapolated to machine located close to a different number of reflecting surfaces.
- Measurements made in far field (or geometric field) of sound sources.
- Same number of measurements used as for a free field test.
- Measurements done in 1/3 or 1/1 octave bands 63Hz to 8kHz.

Near Field Measurements

- Measurement surface 1m from machine surface; approximately same shape as machine surface; usually parallelepiped (i.e. a rectangular box) .
- Sound absorbing material should be placed on nearby reflective surfaces.
- 5-16 measurements needed on test surface, depending on sound field uniformity

$$L_w = L_p + 10\log_{10} S - \Delta_1 - \Delta_2 \text{ dB re } 10^{-12} W$$

Δ_1 = correction for room effect in dB (0 – 3 dB)
Δ_2 = correction for near field or tangential sound propagation (0 – 3 dB)
S = are of the test surface m^2

ENC p280

Δ_1 Estimation

1. Two Test Surfaces

Neglecting Δ_2 for now, we have

$$L_w = L_{p1} + 10\log_{10} S_1 - \Delta_1$$

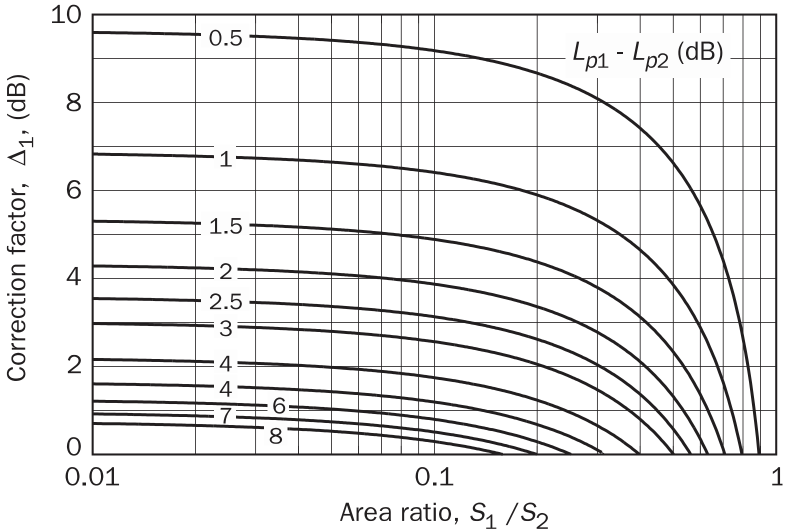
From before, for far field measurements

$$L_w = L_{p2} - 10\log_{10}\left[S_1^{-1} - S_2^{-1}\right] + 10\log_{10}\left[10^{(L_{p1} - L_{p2})/10} - 1\right]$$

Rearranging gives

$$\Delta_1 = L_{p1} - L_{p2} - 10\log_{10}\left[10^{(L_{p1} - L_{p2})/10} - 1\right] + 10\log_{10}\left[1 - \frac{S_1}{S_2}\right]$$

Δ_1 Graphical Estimation



Δ_1 Estimation

2. From Surface Absorption Coefficients

$$\Delta_1 = 10\log_{10}\left[1 + 4S_1 / A\bar{\alpha}\right]$$

A = area of room surfaces
S_1 = area of imaginary measurement surface

See following table for approximate value of the mean acoustic absorption coefficient $\bar{\alpha}$

Mean acoustic absorption Coefficient $\bar{\alpha}$	Description of room
0.05	nearly empty room with smooth hard walls made of concrete, bricks, plaster or tile
0.1	partly empty room, room with smooth walls
0.15	room with furniture, rectangular machinery room, rectangular industrial room
0.2	irregularly shaped room with furniture, irregularly shaped machinery room or industrial room
0.25	room with upholstered furniture, machinery or industrial room with a small amount of acoustical material, e.g., partially absorptive ceiling, on ceiling or walls
0.35	room with acoustical materials on both ceiling and walls
0.5	room with large amounts of acoustical materials on ceiling and walls

Δ_1 Estimation

3. From Table Below

Characteristics of production or test room	Ratio of room volume to test surface area V/S (m)			
usual production room without highly reflective surfaces.	20-50	50-90	90-3,000	over 3,000
room with highly reflective surfaces, with no sound-absorbing treatment.	50-100	100-200	200-600	over 600
Δ_1 (dB)	3	2	1	0

ENC p281

Δ_2 Estimation

Ratio of test surface area to machine surface area S_1/S_m	Near field correction factor Δ_2 (dB)
1-1.1	3
1.1-1.4	2
1.4-2.5	1
2.5-∞	0

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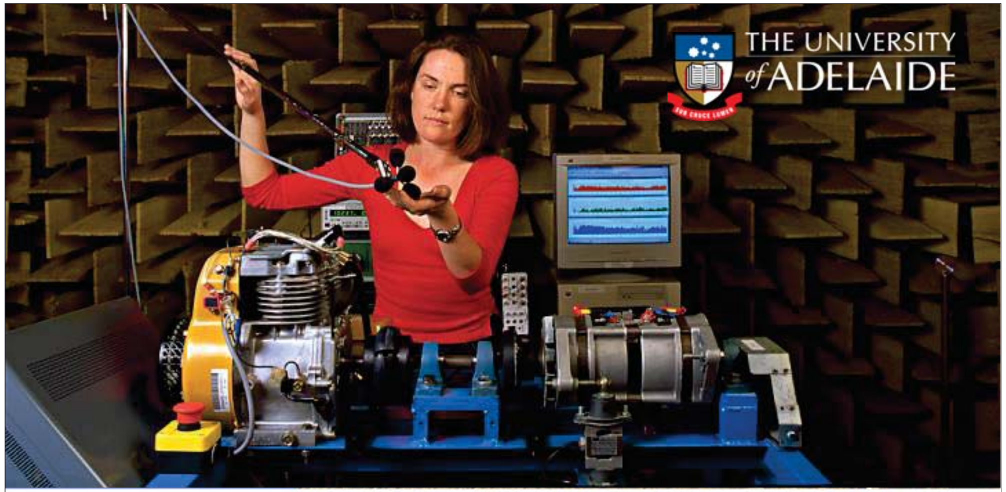
Summary

- Effects of nearby surfaces on radiated sound power.
- Definitions of Near Field, Far Field
- Sound Intensity Measurements
- Several methods for measuring Sound Power Level



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Carl Howard

Indoor Sound

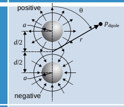
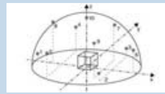
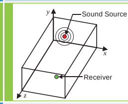
adelaide.edu.au

seek LIGHT

Course Content – How it fits

We know about sound sources, and sound power now we can look at …

- Sound indoors
- Outdoor sound propagation

TOPIC	
Sound Sources	
Sound Power	
Indoor	
Outdoor	

Contents

- Modal response of rooms.
- Diffuse sound field.
- Sabine rooms.
- Reverberation time.
- Panel absorbers.
- Response of long and flat rooms.

Learning Outcomes

- Understand the acoustic response within a room due to direct and reverberant field.
- Understand how to calculate the reverberation time.
- Understand how to design a panel absorber.
- Be able to calculate the change in SPL by installing acoustic absorption.

- Let’s discuss the **CONCEPTS** first, then the maths will make more sense.

Question …

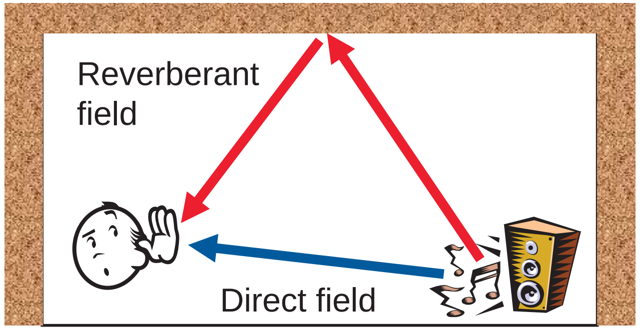
www.socrative.com
Room b069a2aa

A room 10 × 10 × 4m has an average Sabine absorption coefficient $\alpha = 0.1$. The steady state reverberant field pressure level is 60dB.

a) What is the lowest resonance frequency of the room (excluding the 0,0,0 mode)?
b) At what rate (in W/m^2) is the sound energy incident on the walls of the room?
c) What is the acoustic power output level (dB re 10^{-12} W) of the noise source producing this pressure level?
d) At what distance from the noise source is the reverberant field pressure level equal to the direct field pressure level? (Assume that the noise source is on the floor in the centre of the room).
e) Calculate the room reverberation time (seconds).

Sound in Enclosed Spaces

- A sound field comprises
  - ***Direct field*** and
  - ***Reflected (Reverberant) field***



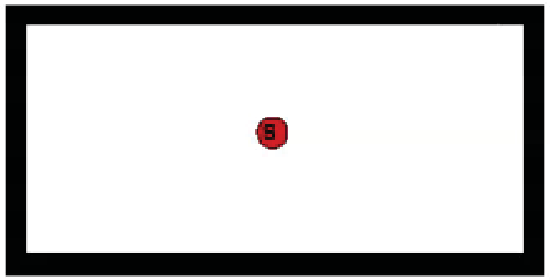
Sound absorption is used in enclosures to "soak" up sound

- Absorptive material is used to "soak" up the sound level, to decrease the sound that impinges on the wall.

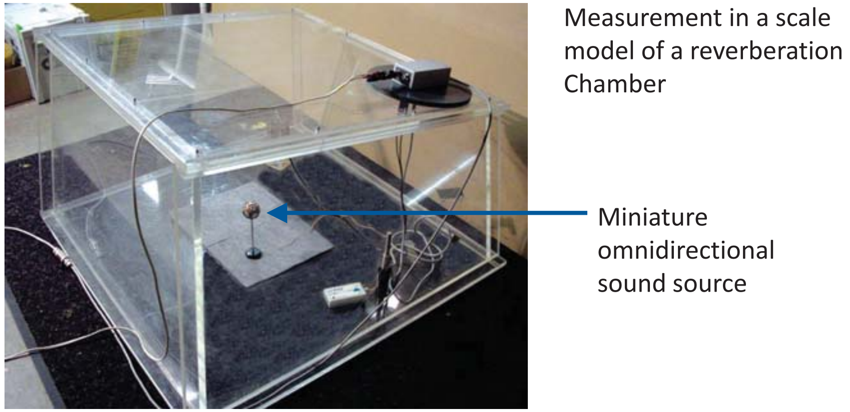
- Sound absorptive material will only affect the reverberant field!

Sound in Enclosed Spaces

- Sound is only absorbed when it strikes a wall. [animation]



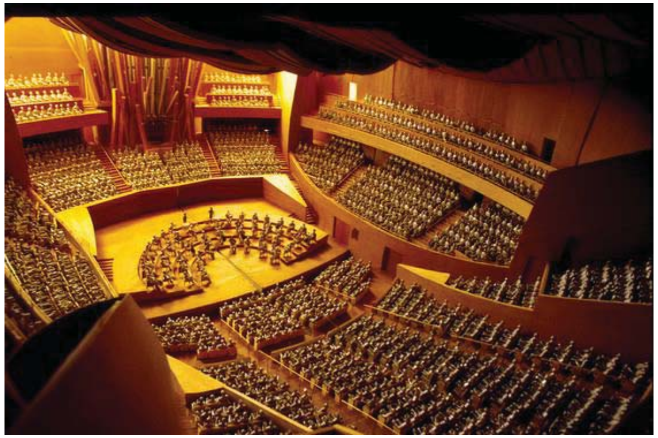
Scale Model Testing



DIRAC Room Acoustics Software Type 7841, bp1974.pdf

Scale Model Testing

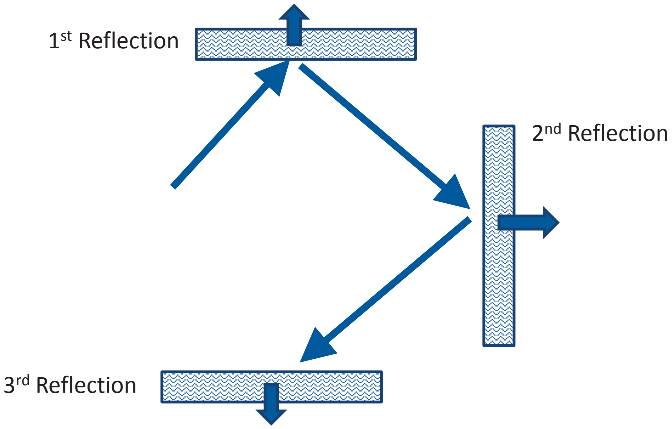
- Walt Disney Concert Hall: one-tenth-scale model



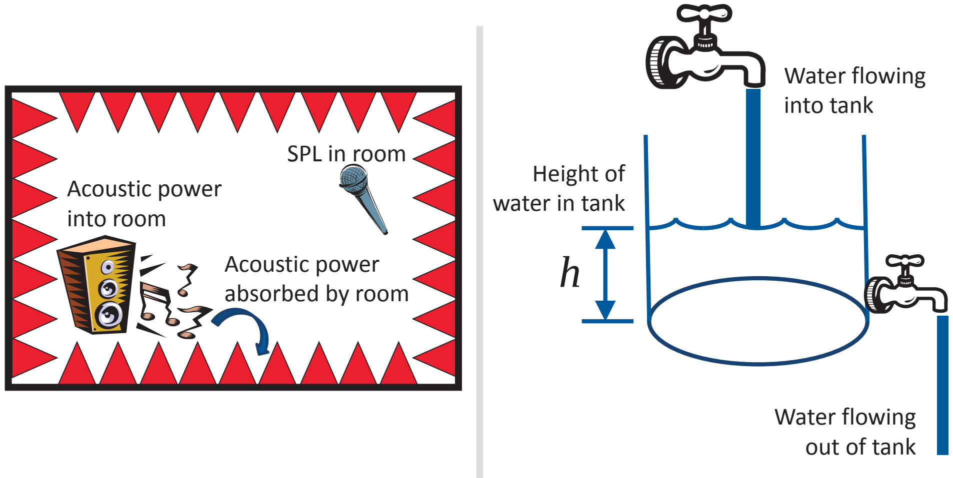
http://wdch10.laphil.com/wdch10/wdch/acoustics.html

Absorption by Walls

- Each time an acoustic wave strikes an absorbing wall, a little bit more of sound is absorbed



Analogy of a Water Tank



- Power Balance:
  "Power In" related to "Energy Stored" plus "Power Out"

Energy Balance

- In maths speak we could write that the sound power P injected into the room during time Δt must equal the change of stored acoustic energy in the room ΔE plus the sound power flowing out or absorbed by the room P_L during time Δt.

Just like any other energy balance for thermodynamics, mass flow, electrical systems etc.

$$\underbrace{P\Delta t}_{\text{Power into room}} = \underbrace{V\Delta E}_{\text{Sound Power level in room}} + \underbrace{P_L\Delta t}_{\text{Sound Power absorbed by room}}$$

Transient Response

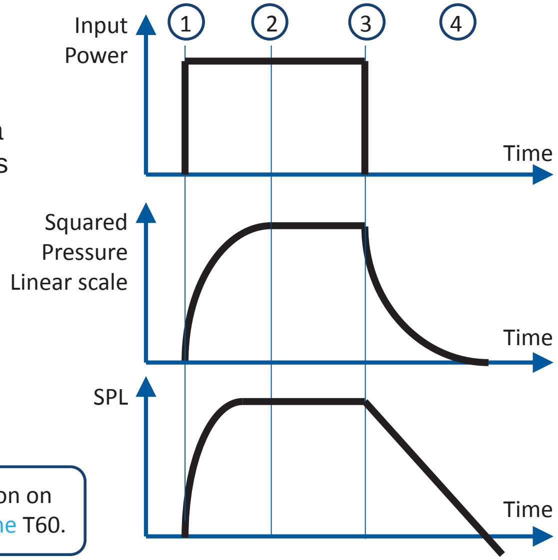
1. Sound power turned on.

1.- 2. Sound level rises up to a maximum level where power is balanced (steady-state).

3. Sound source turned off abruptly.

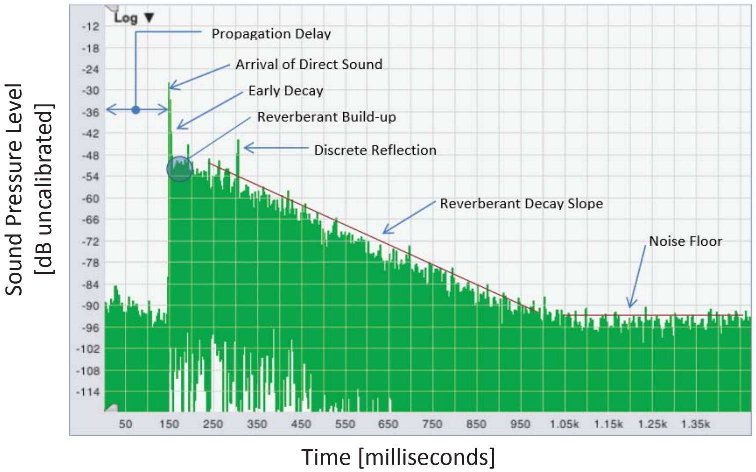
3.- 4. Sound pressure level decays linearly.

Leads to discussion on reverberation time T60.



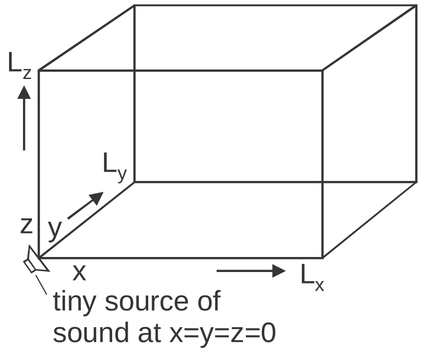
Transient Response

- Actual measured decay response of a room



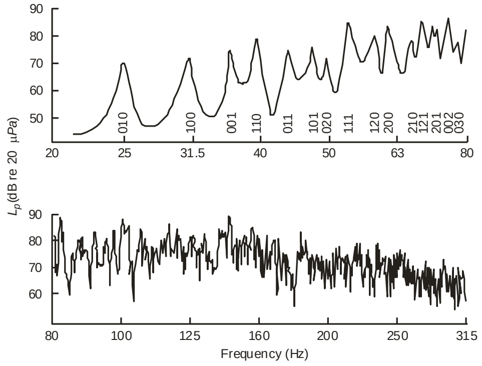
Sound In Enclosed Spaces

- Following analysis assumes that coupling between enclosed sound field and enclosing wall vibration is small
- *Sabine* rooms are close to cubic such that no room dimension exceeds either of the other two by more than a factor of 3.
- If room is not *Sabine*, it is "*flat*" or "*long*"
- Following modal analysis uses rectangular rooms for mathematical convenience
  - same principles apply to rooms of any shape



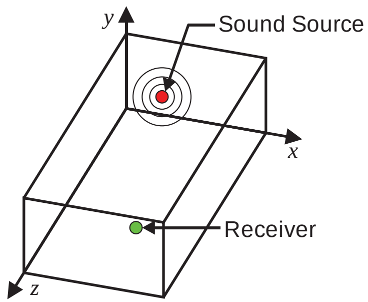
Modal Response at Low Frequencies

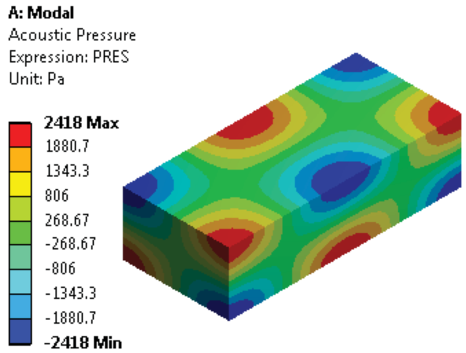
- Low frequencies
  - modal response
  - room resonances well separated
  - sound field characterized by nodes and antinodes
  - SPL varies dramatically as a function of location and frequency
- High frequencies
  - statistical response
  - SPL varies to a lesser extent with location and frequency
  - resonances not as well defined - too many



ENC p292

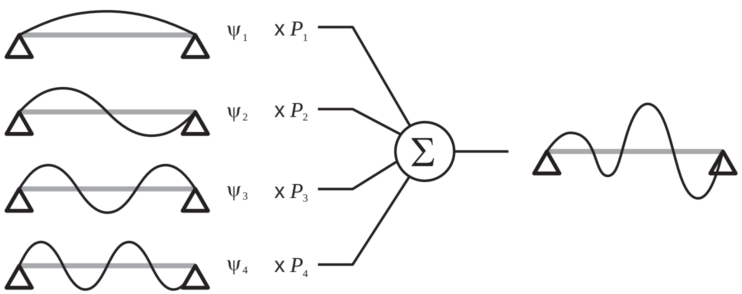
Acoustic Modes of a Room





Modal Summation

- Individual modal responses can be summed to form the total response of a system.



Theory ...

PROBLEM: Derive expressions for modal resonance frequencies and modal pressure distributions for a rectangular cavity by solving the wave equation with appropriate boundary conditions

SOLUTION:

Wave equation: $$\nabla^2 \varphi = \frac{1}{c^2}\frac{\partial^2 \varphi}{\partial t^2}$$

Assume solution $$\varphi = X(x)Y(y)Z(z)e^{j\omega t}$$

Substitute solution into wave equation: $$\frac{X''}{X} + \frac{Y''}{Y} + \frac{Z''}{Z} = -k^2; \quad \left[k = \frac{\omega}{c}\right]$$

Rectangular Room

As k is constant for fixed ω,

$$\frac{X''}{X} = -k_x^2\,;\ \frac{Y''}{Y} = -k_y^2\,;\ \frac{Z''}{Z} = -k_z^2\,;$$

$$k_x^2 + k_y^2 + k_z^2 = k^2$$

Solutions: $X = A_x e^{jk_x x} + B_x e^{-jk_x x}$

Similar for Y and Z

Rigid walls: u = 0 at wall

$$u_x = -\partial\varphi/\partial x = 0 \quad at \quad x=0 \text{ or } L_x$$

$$-jk_x YZe^{j\omega t}\left[A_x e^{jk_x x} - B_x e^{-jk_x x}\right]_{x=0,L_x} = 0$$

Using x = 0, gives $A_x = B_x$

Using $x = L_x$, and $A_x = B_x$ gives:

$$e^{jk_x L_x} - e^{-jk_x L_x} = 2j\sin(k_x L_x) = 0$$

Thus: $k_x = \dfrac{n\pi}{L_x}$;similar results apply for k_y, k_y

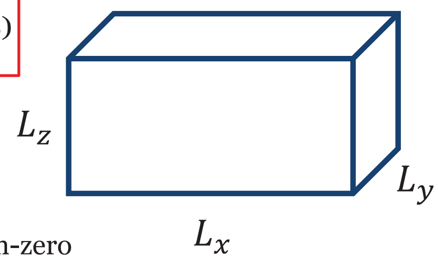
Rectangular Room

Thus: $$\left(\frac{n_x\pi}{L_x}\right)^2 + \left(\frac{n_y\pi}{L_y}\right)^2 + \left(\frac{n_z\pi}{L_z}\right)^2 = \frac{\omega^2}{c^2}$$

Modal resonance frequencies

$$\boxed{f_n = \frac{c}{2}\sqrt{\left[\frac{n_x}{L_x}\right]^2 + \left[\frac{n_y}{L_y}\right]^2 + \left[\frac{n_z}{L_z}\right]^2}\quad \text{(Hz)}}$$

$f_{nx,ny,nz}$ = natural frequency in Hz
c = speed of sound in m/s
n = mode number in a particular axis 0 to ∞
L = length along a particular axis in m



Axial modes: only one of n_x, n_y or n_z non-zero
Tangential modes: two non-zero
Oblique modes: all non-zero
All modes have anti-nodes at room corners

ENC p294

Rectangular Room

How to calculate the resonance frequencies …

- Write out a table with columns for n_x, n_y, n_z with all combinations from 0 to N. For example if N=2.
- Calculate f for each case, then sort the rows based on frequencies from lowest to highest.

n_x	n_y	n_z	f [Hz]
0	0	0	
0	0	1	
0	0	2	
0	1	0	
0	1	1	
0	1	2	
0	2	0	
0	2	1	
0	2	2	
1	0	0	
1	0	1	

.... And so on.

Rectangular Room

Modal pressure distribution: $p = \rho\dfrac{\partial\varphi}{\partial t} = j\omega\rho\, X(x)Y(y)Z(z)e^{j\omega t}$

Using previous solutions for X, Y and Z gives:

$$\boxed{p = p_0\cos\left[\frac{\pi n_x x}{L_x}\right]\cos\left[\frac{\pi n_y y}{L_y}\right]\cos\left[\frac{\pi n_z z}{L_z}\right]e^{j\omega t}}$$

Axial modes: only one of n_x, n_y or n_z non-zero
Tangential modes: two non-zero
Oblique modes: all non-zero
All modes have anti-nodes at room corners

ENC p294

Question …

A room 10 × 10 × 4m has an average Sabine absorption coefficient α = 0.1. The steady state reverberant field pressure level is 60dB.

a) What is the lowest resonance frequency of the room (excluding the 0,0,0 mode)?
b) At what rate (in W/m^2) is the sound energy incident on the walls of the room?
c) What is the acoustic power output level (dB re 10^{-12} W) of the noise source producing this pressure level?
d) At what distance from the noise source is the reverberant field pressure level equal to the direct field pressure level? (Assume that the noise source is on the floor in the centre of the room).
e) Calculate the room reverberation time (seconds).

Question

On Socrative.com Room b069a2aa

- What is the lowest resonance frequency of the room (excluding the 0,0,0 mode)?

A. 1715 Hz
B. 49.3 Hz
C. 24.25 Hz
D. 17.15 Hz

Cylindrical Room

$$\boxed{f(n_z,m,n)=\frac{c}{2}\sqrt{\left(\frac{n_z}{L}\right)^2+\left(\frac{\psi_{m,n}}{a}\right)^2}}$$

ENC p296

n_z = number of nodal planes normal to the axis of the cylinder 0 to ∞
m = number of diametral pressure nodes
n = number of circumferential pressure nodes
$\Psi_{m,n}$ = coefficient based on m,n from table below
L = length of cylinder
a = radius of cylinder

Values of $\psi_{m,n}$

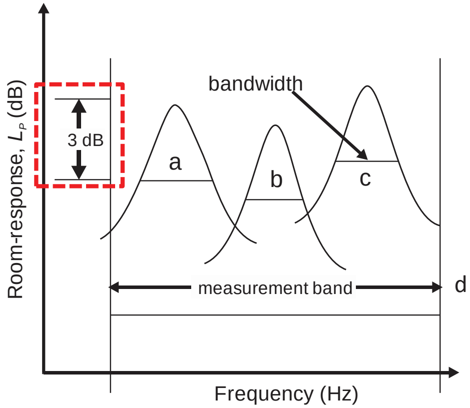
m\n	0	1	2	3	4
0	0.0000	1.2197	2.2331	3.2383	4.2411
1	0.5861	1.6971	2.7172	3.7261	4.7312
2	0.9722	2.1346	3.1734	4.1923	5.2036
3	1.3373	2.5513	3.6115	4.6428	5.6623
4	1.6926	2.9547	4.0368	5.0815	6.1103

Cylindrical Room

The SpaceX Falcon 9 Payload Fairing was successfully tested at the NASA Glenn Research Center Plum Brook Station's Reverberant Acoustic Test Facility (RATF) during May-June 2013.

From Hozman, Hughes, "An Evaluation of the Additional Acoustic Power Needed to Overcome the Effects of a Test-Article's Absorption during Reverberant Chamber Acoustic Testing of Spaceflight Hardware", Noise-Con 2014, Fort Lauderdale, Florida, September 8-10, 2014

Modal Overlap



For this sketch
M=0.3

- in above figure, Modal Overlap $M = (a+b+c)/d$

$$M = \Delta f \; dN/df$$

- Δf = average modal bandwidth for modes in the measurement band
- dN/df is the modal density (see next page ...)

ENC p299

Rectangular Room

- Number of modes with frequencies below f

$$N = \frac{4\pi f^3 V}{3c^3} + \frac{\pi f^2 S}{4c^2} + \frac{f L}{8c}$$

V = volume m^3
S = surface area m^2
L = perimeter m
f = frequency Hz
c = speed of sound m/s

- Modal Density

$$\frac{dN}{df} = \frac{4\pi f^2 V}{c^3} + \frac{\pi f S}{2c^2} + \frac{L}{8c}$$

- Modal Damping

$$\frac{\Delta f}{f} = \frac{2.2}{T_{60} f} = \frac{1}{Q} = \eta = \frac{2\zeta}{\sqrt{1-\zeta^2}} = \frac{\delta}{\pi}$$

Q = quality factor
η = modal loss factor
δ = logarithmic decrement
ζ = critical damping ratio
T_{60} = reverberation time
Δf = bandwidth
f = centre frequency

ENC p299

Cross-Over Frequency

From low to high frequency (statistical) behaviour

- 1/3 or octave band noise / analysis
  - 3-6 modes must be resonant in frequency band for high frequency analysis
- Single frequency
  - Modal overlap $M > 3$ for high frequency analysis

Diffuse Sound Field

See p300 in textbook

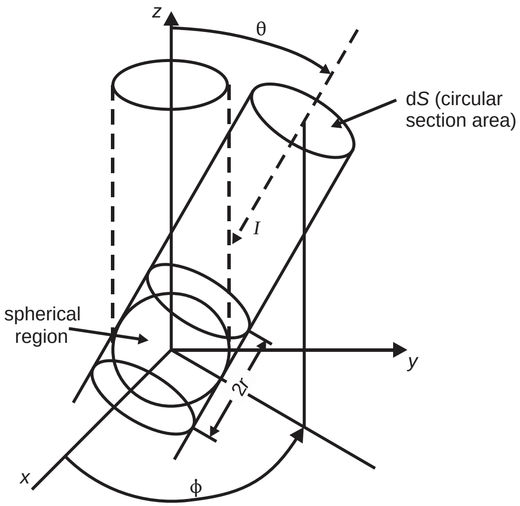
- Intensity = 0
- Require Modal Overlap $M \geq 3$
- does NOT imply uniform spatial pressure distribution
- p^2 fluctuates in time and space

$$\sigma \cong 5.57\left(1+0.238BT_{60}\right)^{-1/2} \quad (dB)$$

ENC p300

Effective Intensity in a Diffuse Sound Field

- Intensity flowing in any one particular direction such as into a wall



ENC p301

- ratio of the volume of the spherical region to the cylindrical section of column which just encompasses the spherical region is:
$$\frac{4\pi r^3}{3}\cdot\frac{1}{2\pi r^3}=\frac{2}{3}$$
- Time for sound to travel through sphere = $2r/c$
- Incremental contribution per unit area to the energy E in the spherical region due to any beam is: $\Delta E = \frac{2}{3} I \frac{2r}{c}$
- total energy is obtained by integrating the incremental energy contribution per unit area of sphere over the area of the sphere. The incremental area of sphere for use in integration is, $\mathrm{d}S = r^2 \sin\theta d\theta \mathrm{d}\varphi$
- Thus,
$$E = \iint_S \Delta E \mathrm{d}S = \Delta \mathrm{E} \int_0^{2\pi} r\mathrm{d}\varphi \int_0^{\pi} r\sin\theta \mathrm{d}\theta$$
$$= \frac{4I}{3c}\int_0^{2\pi}\mathrm{d}\varphi\int_0^{\pi} r^3 \sin\theta \mathrm{d}\theta = \frac{16I\pi r^3}{3c}$$

- Energy Density, $\psi = \frac{E}{V} = \frac{16I\pi r^3/3c}{4\pi r^3/3}$
- Thus, $I = \psi c/4$
- Would like to get diffuse field intensity (in a single direction) defined in terms of the sound pressure
  - Consider a plane wave of unit cross-sectional area travelling a unit distance:
    - Contribution of plane wave to the energy in the resulting unit volume is $E = I \times 1 \times 1/c$
    - Also, $E = V\psi = 1\times1\times1\times\psi$
    - thus for a plane wave, $\psi = I/c$
    - However, for a plane wave, $I = \langle p^2\rangle/\rho c$
    - Thus, $\psi = p^2/\rho c^2$

- In a diffuse field, $\boxed{I = \psi c/4}$

ENC p302

  - Thus, in a diffuse field, $\boxed{I = \frac{\langle p^2\rangle}{4\rho c}}$

I = intensity in any direction W/m²
ψ = energy density W / m³
c = speed of sound m/s
p = pressure Pa
ρ = density kg/m³

Question …

A room 10 × 10 × 4m has an average Sabine absorption coefficient $\alpha = 0.1$. The steady state reverberant field pressure level is 60dB.

a) What is the lowest resonance frequency of the room (excluding the 0,0,0 mode)?
b) At what rate (in W/m²) is the sound energy incident on the walls of the room?
c) What is the acoustic power output level (dB re 10^{-12} W) of the noise source producing this pressure level?
d) At what distance from the noise source is the reverberant field pressure level equal to the direct field pressure level? (Assume that the noise source is on the floor in the centre of the room).
e) Calculate the room reverberation time (seconds).

Question

- On Socrative.com
- At what rate (in W/m²) is the sound energy incident on the walls of the room?

A. 1.21×10^{-12} W/m^2
B. 2.41×10^{-7} W/m^2
C. 9.64×10^{-7} W/m^2
D. 4.15×10^{-2} W/m^2

Sound Absorption In Rooms

- Assume walls locally reactive
  - walls characterized by impedance (surface property)
  - local wall response independent of response at other locations
- locally reactive assumption *invalid* in:
  - vehicles, planes
  - very low frequencies in rooms
- Surface absorption
  - fraction of incident energy absorbed by room surfaces, $\overline{\alpha_w}$

Air Absorption

- Air absorption, $\overline{\alpha_a}$
  - mean free path in empty room =4V/*S*
  - $\overline{\alpha_a} = 4m'V/S$

    where *m'* is the energy attenuation constant defined using:

$$p_1^2 = p_0^2 e^{-m'x}$$

where *x* is the distance travelled from position 0 to position 1

$$10\log_{10} p_1^2 = 10\log_{10} p_0^2 - 10m'x\log_{10}e$$

$$10m'\log_{10}e = \frac{1}{x}\left[10\log_{10}(p_0/p_1)^2\right] = \frac{m}{1000}$$

Where *m* is the attenuation in air in dB per 1000 metres

Air Absorption

– Thus: $4m' = 9.21\times10^{-4}m$
where m is in dB/1000 metres (see table 5.3 in ENC book)

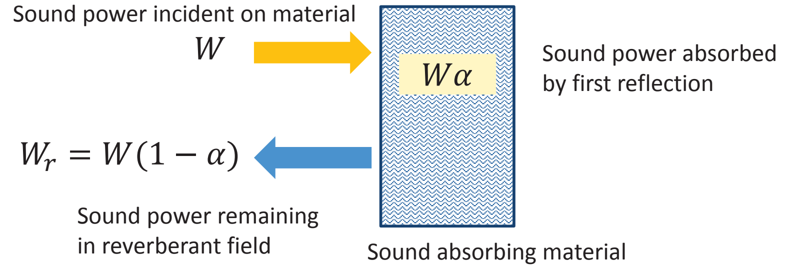
- Total absorption coefficient

$$\boxed{\begin{aligned}\overline{\alpha} &= \overline{\alpha_w} + \overline{\alpha_a} \\ &= \overline{\alpha_w} + 9.21\times10^{-4} mV/S\end{aligned}}$$

$\bar{\alpha}_w$ absorption by walls
$\bar{\alpha}_a$ absorption by air

Absorption by Walls

- Every time a sound wave strikes the wall some energy is absorbed.

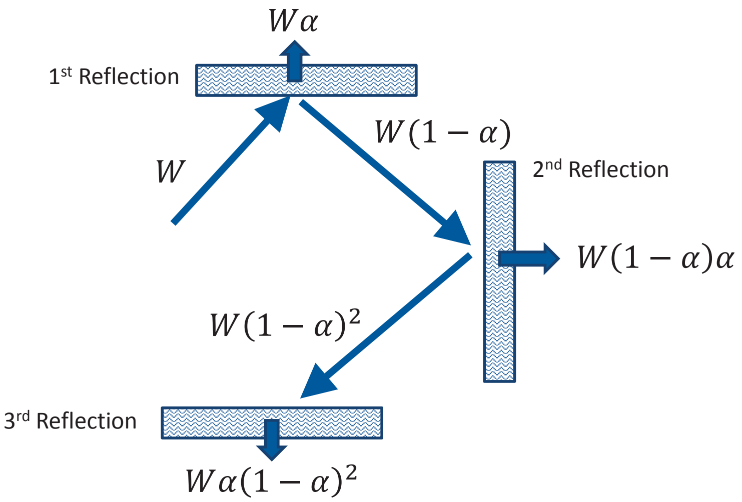


Sanity check:

$$\overbrace{W}^{\text{Incident}} = \overbrace{W\alpha}^{\text{Absorbed}} + \overbrace{W(1-\alpha)}^{\text{Reflected}} + \overbrace{0}^{\text{Transmitted}} = W\alpha + W - W\alpha = W$$

Absorption by Walls

- ... and this continues on and on and on ....



Absorption by Walls ...

- Derivation

1st Reflection

$$\overbrace{W}^{\text{Incident}} = \overbrace{[W]\alpha}^{\text{Absorbed}} + \overbrace{[W](1-\alpha)}^{\text{Reflected}}$$

2nd Reflection

$$\begin{aligned}\overbrace{W(1-\alpha)} &= \overbrace{[W(1-\alpha)]\alpha} + \overbrace{[W(1-\alpha)](1-\alpha)} \\ &= [W(1-\alpha)]\alpha + W(1-\alpha)^2\end{aligned}$$

3rd Reflection

$$\begin{aligned}\overbrace{W(1-\alpha)^2} &= \overbrace{[W(1-\alpha)^2]\alpha} + \overbrace{[W(1-\alpha)^2](1-\alpha)} \\ &= [W(1-\alpha)^2]\alpha + W(1-\alpha)^3\end{aligned}$$

Absorption by Walls …

- Just look at the power absorbed

1st Reflection

$$\overbrace{W}^{\text{Incident}} = \overbrace{\boxed{[W]\alpha}}^{\text{Absorbed}} + \overbrace{[W](1-\alpha)}^{\text{Reflected}}$$

2nd Reflection

$$\overbrace{W(1-\alpha)} = \overbrace{[W(1-\alpha)]\alpha} + \overbrace{[W(1-\alpha)](1-\alpha)}$$
$$= \boxed{[W(1-\alpha)]\alpha} + W(1-\alpha)^2$$

3rd Reflection

$$\overbrace{W(1-\alpha)^2} = \overbrace{[W(1-\alpha)^2]\alpha} + \overbrace{[W(1-\alpha)^2](1-\alpha)}$$
$$= \boxed{[W(1-\alpha)^2]\alpha} + W(1-\alpha)^3$$

Absorption by Walls …

- Just look at the power absorbed by the walls

$$W_a = \overbrace{W\alpha}^{1^{st}\text{ Reflection}} + \overbrace{W\alpha(1-\alpha)^2}^{2^{nd}\text{ Reflection}} + \overbrace{W\alpha(1-\alpha)^3}^{3^{rd}\text{ Reflection}} + \cdots$$

$$W_a = W\alpha \sum_{k=0}^{\infty}(1-\alpha)^k \qquad \sum_{k=0}^{\infty} x^k = \frac{1}{1-x} \text{ for } |x| < 1$$

$$W_a = W\alpha \frac{1}{1-(1-\alpha)} = W\alpha\frac{1}{\alpha} = W$$

- What was that all about??
  – We just proved that the power absorbed must equal the power injected into the room. $W_a = W$

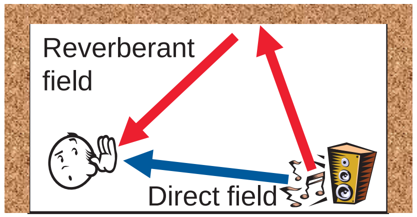
Steady State Response

- What we want is an equation for the sound pressure level in an enclosure with absorbing materials on the walls, when there is a sound power source in the enclosure.

- We need an equation for the overall the SPL due to the
  – Direct field
  – Reverberant field

- Total mean square pressure

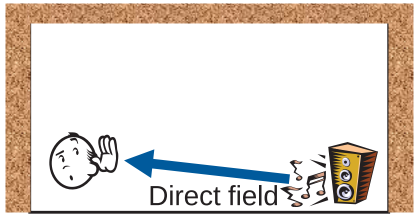
$$\left\langle p^2 \right\rangle_{s,t} = \underbrace{\left\langle p_R^2 \right\rangle_{s,t}}_{\text{reverberant}} + \underbrace{\left\langle p_D^2 \right\rangle_{t}}_{\text{direct}}$$



Steady State Response

- Direct field

$$\boxed{\left\langle p_D^2 \right\rangle_t = \frac{D_\theta}{4\pi r^2} \times W\rho c}$$

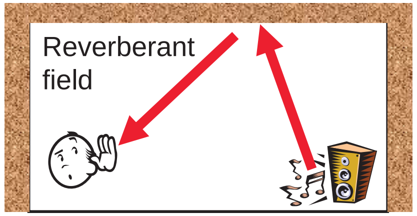


D_θ = directivity factor (no units)
r = distance from source in m
W = sound power of source in Watts
ρ = density in kg / m³
c = speed of sound in m/s

ENC p304

Steady State Response

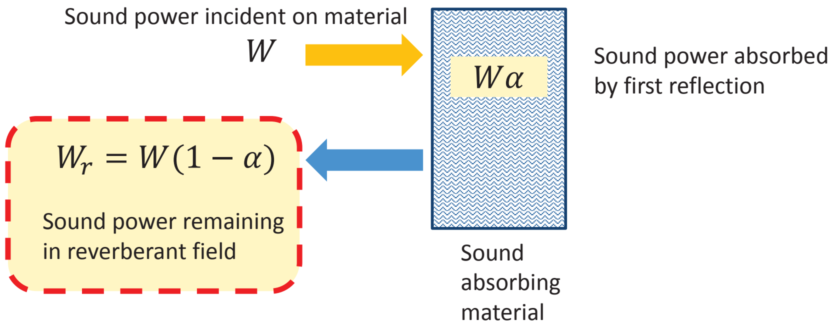
- Reverberant field



- Firstly we need to distinguish how much sound power is
  - Absorbed by walls
  - Injected into the reverberant field

Absorption by Walls

- We can say that the power injected into the reverberant field is everything after the first strike with the wall (which is due the direct field)



Absorption by Walls ...

- Power absorbed, W_a, by surfaces and air is rate of energy absorbed, given by:

$$W_a = S\bar{\alpha} \times \text{[intensity in direction of absorbing surface]}$$

$$\boxed{W_a = S\bar{\alpha}\left[\frac{\langle p^2 \rangle}{4\rho c}\right]}$$

- power introduced into reverberant field = power absorbed

$$W(1-\bar{\alpha}) = S\bar{\alpha}\left[\frac{\langle p^2 \rangle}{4\rho c}\right]$$

- Rearranging give $\boxed{\langle p_R^2 \rangle_{s,t} = \dfrac{4\rho c W(1-\bar{\alpha})}{S\bar{\alpha}}}$

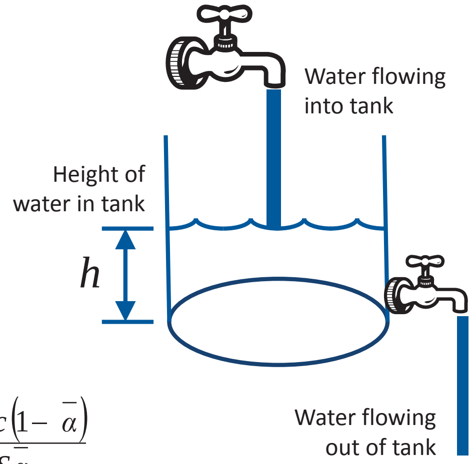
Water Tank Analogy ....

See Fahy, "Foundations of Engineering Acoustics", p265
Bies & Hansen, "Engineering Noise Control", p304

- At steady-state: Water In = Water Out

Sound Power flowing into reverberant field $= W(1-\bar{\alpha})$

Sound pressure squared in room $= \langle \bar{p}_R^2 \rangle$



Sound Power absorbed by walls

$$W_a = S\bar{\alpha}\left[\frac{\langle \bar{p}_R^2 \rangle}{4\rho c}\right]$$

$$\langle p_R^2 \rangle_{s,t} = \frac{4W\rho c(1-\bar{\alpha})}{S\bar{\alpha}}$$

Steady State Response

- Total sound pressure is

$$\langle p^2 \rangle_{s,t} = \langle p_D^2 \rangle_t + \langle p_R^2 \rangle_{s,t}$$
$$= W\rho c \left[\frac{D_\theta}{4\pi r^2} + \frac{4(1-\bar{\alpha})}{S\bar{\alpha}} \right]$$

- Taking logs gives: $\boxed{L_p = L_w + 10\log_{10}\left[\frac{D_\theta}{4\pi r^2} + \frac{4}{R}\right] + 10\log_{10}\left[\frac{\rho c}{400}\right]}$

ENC p305

Steady-State Response

- The sound pressure in an enclosed space due to a sound power source is

$$\boxed{L_p = L_w + 10\log_{10}\left[\frac{D_\theta}{4\pi\, r^2} + \frac{4}{R}\right] + 10\log_{10}\left[\frac{\rho c}{400}\right]}$$

L_p = sound pressure level in dB re 20μPa
L_w = sound power level of source in dB re 10^{-12}W
D_θ = directivity factor (no units)
r = distance from source in m
R = room constant in m^2
ρ = density in kg / m^3
c = speed of sound in m/s

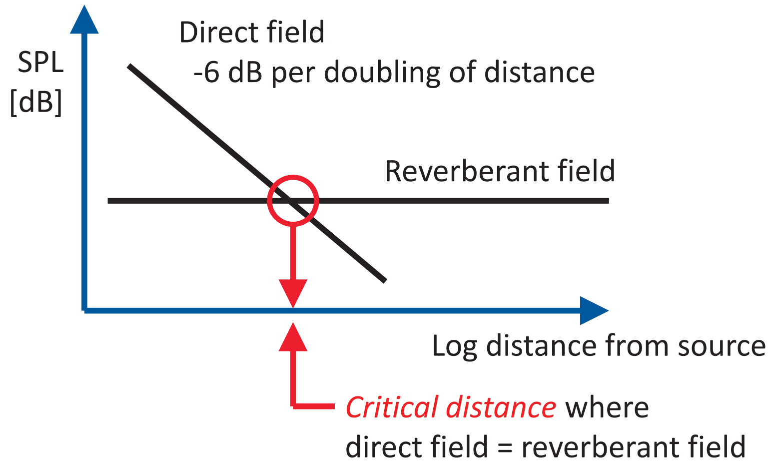
ENC p305

Reflection Effects

Remember this table from the lecture on Fundamentals

Situation	Directivity factor, Q	Directivity Index, DI (dB)
free space	1	0
centred in a large flat surface	2	3
centred at the edge formed by the junction of two large flat surfaces	4	6
at the corner formed by the junction of three large flat surfaces	8	9

Steady-State Response



Question ...

A room 10 × 10 × 4m has an average Sabine absorption coefficient α = 0.1. The steady state reverberant field pressure level is 60dB.

a) What is the lowest resonance frequency of the room (excluding the 0,0,0 mode)?

b) At what rate (in W/m²) is the sound energy incident on the walls of the room?

c) What is the acoustic power output level (dB re 10^{-12} W) of the noise source producing this pressure level?

d) At what distance from the noise source is the reverberant field pressure level equal to the direct field pressure level? (Assume that the noise source is on the floor in the centre of the room).

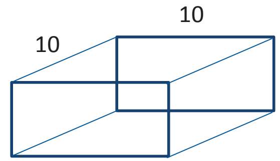
e) Calculate the room reverberation time (seconds).

Answer ....

What is the acoustic power output level (dB re 10^{-12} W) of the noise source producing this pressure level?

$$\langle p_R^2 \rangle_{s,t} = 4W\rho c(1-\bar{\alpha})/S\bar{\alpha}$$

$$W = \frac{\langle p_R^2 \rangle S\bar{\alpha}}{4\rho c(1-\bar{\alpha})}$$



$$S = 2 \times (10 \times 10) + 2 \times (10 \times 4) + 2 \times (10 \times 4) = 360 \text{ m}^2$$

$$W = \frac{\langle p_R^2 \rangle S\bar{\alpha}}{4\rho c(1-\bar{\alpha})} = \frac{4 \times 10^{-4} \times 360 \times 0.1}{4 \times 1.21 \times 343 \times (1-0.1)} = 9.637\,\mu\text{W}$$

$$L_w = 10\log_{10}(9.637 \times 10^{-6}) + 120 = 69.8 \text{ dB re } 10^{-12}\text{ W}$$

Question ...

A room 10 × 10 × 4m has an average Sabine absorption coefficient α = 0.1. The steady state reverberant field pressure level is 60dB.

a) What is the lowest resonance frequency of the room (excluding the 0,0,0 mode)?

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e) Calculate the room reverberation time (seconds).

Answer ....

At what distance from the noise source is the reverberant field pressure level equal to the direct field pressure level?

$$\langle p_R^2 \rangle_{s,t} = 4W\rho c(1-\bar{\alpha})/S\bar{\alpha} \quad = \quad \langle p_D^2 \rangle_t = \frac{D_\theta}{4\pi r^2} \times W\rho c$$

$$\frac{4W\rho c(1-\bar{\alpha})}{S\bar{\alpha}} = \frac{W\rho c D_\theta}{4\pi r^2}$$

$$r = \left[\frac{D_\theta S\bar{\alpha}}{16\pi(1-\bar{\alpha})}\right]^{1/2} = \left[\frac{2 \times 360 \times 0.1}{16\pi(1-0.1)}\right]^{1/2} = 1.26\text{ m}$$

Critical Distance

- This is an important result...
  It tells us the distance that the reverberant field is equal to the direct field.

$$\boxed{r_{critical} = \left[\frac{D_\theta S\bar{\alpha}}{16\pi(1-\bar{\alpha})}\right]^{1/2}}$$

$r_{critical}$ = distance that direct field = reverb in m
D_θ = directivity factor (no units)
S = surface area of room with absorption m^2
$\bar{\alpha}$ = average sound absorption coeff (no units)

- It is important because it means that if the receiver at a distance less than *r* from the source, then the direct field will dominate. Hence, installing sound absorption in the room, which is used to control the reverberant field, won't do any good!

Transient Response (Classical Desc.)

PROBLEM: Relate room absorption $S\bar{\alpha}$, to room reverberation time, T_{60} (time for sound field to decay by 60 dB when sound is turned off)

SOLUTION

- Rate of change of energy, W, in the room = rate of supply, W_o minus rate of absorption, W_a $\quad W = V\partial\psi/\partial t = W_0 - W_a = W_0 - \psi Sc\bar{\alpha}/4$
- Introduce $X = \left[4W_0/Sc\bar{\alpha}\right] - \psi$

then:
$$\frac{dX}{dt} = (-1)\frac{\partial\psi}{\partial t} = \frac{-1}{V}\left[W_0 - \psi S\bar{\alpha}c/4\right]$$
$$\frac{1}{X} = \frac{S\bar{\alpha}c}{4}\left[W_0 - \psi Sc\bar{\alpha}/4\right]^{-1}$$

Thus, $$\frac{1}{X}\frac{dX}{dt} = -\frac{Sc\bar{\alpha}}{4V}$$

Integrate: $$\int_{X_0}^{X}\frac{dX}{X} = \int_0^t -\frac{S\bar{a}c}{4V}dt$$

Transient Response (Cont.)

Thus, $\log_e(X) - \log_e(X_0) = -S\bar{\alpha}ct/(4V)$

and $X = X_0 e^{-S\bar{\alpha}ct/4V}$

- Relate the preceding analysis to reverberation time
  - shut off power at time, t = 0. That is,
    $$W_0|_{t=0} = 0 \Rightarrow X_0 = -\psi_0$$
  - At time , t, $\quad \psi = \frac{4W_0}{S\bar{\alpha}c} - X = \frac{4W_0}{S\bar{\alpha}c} + \psi_0 e^{-S\bar{\alpha}ct/4V}$
  - As $\psi \propto \langle p^2 \rangle \quad \langle p^2 \rangle = \langle p_0^2 \rangle e^{-Sc\bar{\alpha}t/4V}$
    $$L_{p0} - L_p = 1.086 Sc\bar{\alpha}t/V$$

$60 = 1.086 S\bar{\alpha}cT_{60}/V$ where $\boxed{\frac{1}{T_{60}} = \frac{1}{N}\sum_{i=1}^{N}\frac{1}{T_{60i}}}$

Thus, $\boxed{T_{60} = \frac{55.25V}{Sc\bar{\alpha}}}$

ENC p307

Question ...

A room 10 × 10 × 4m has an average Sabine absorption coefficient α = 0.1. The steady state reverberant field pressure level is 60dB.

a) What is the lowest resonance frequency of the room (excluding the 0,0,0 mode)?
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e) Calculate the room reverberation time (seconds).

Question

- On Socrative.com Room b069a2aa

- Calculate the room reverberation time (seconds).

  A. 0.107 s
  B. 1.79 s
  C. 44.5 s

Determination of $S\bar{\alpha}$ - Method 2

- Reference source of known sound power level, L_w
- Measurements taken remote from sound source so reverberant field dominates

$$L_p = L_w + 10\log_{10}\left[\frac{4(1-\bar{\alpha})}{S\bar{\alpha}}\right]$$

- Thus, $$\boxed{R = \frac{S\bar{\alpha}}{(1-\bar{\alpha})} = 4\times 10^{(L_w - L_p)/10}}$$

ENC p313

and $$L_p = 10\log_{10}\left[\frac{1}{N}\sum_{i=1}^{N} 10^{L_{pi}/10}\right]$$

Transient Response - Modal Description

See text book for a detailed explanation of this.

- This analysis leads to the following reverberation time equation: $$T_{60} = -\frac{55.25V}{Sc\log_e(1-\bar{\alpha}_{st})}$$
  ENC p309
  - Norris-Eyring equation
  - used in architectural acoustics
  - air absorption must be included in $\bar{\alpha}_{st}$
- If α < 0.4, error < 0.5 dB if we set $\bar{\alpha}_{st} = \log_e(1-\bar{\alpha}_{st})$
  - Setting $\bar{\alpha} = \bar{\alpha}_{st}$,
  - gives the previous Sabine equation:

$$T_{60} = \frac{55.25V}{Sc\bar{\alpha}}$$

- If $(1-\bar{\alpha}_{st})$ is replaced with $\bar{\beta}_{st}$ in Norris-Eyring equation, then the equation of Millington and Sette is obtained:

$$T_{60} = -\frac{55.25V}{Sc\log_e\bar{\beta}_{st}}$$

ENC p309

- where $$\bar{\beta}_{st} = \left[\prod_{i=1}^{n} \beta_i^{S_i/S}\right]$$

- is the average room surface reflection coefficient

Empirical Expressions

- Previous equations based on assumptions:
  - room dimensions satisfy the conditions for Sabine rooms
  - absorption is reasonably well distributed over the room surfaces
- for rooms which do not meet this criterion, the Sabine expression for reverberation time is used with $\bar{\alpha}$ replaced with α which is defined as follows:

$$\alpha = -\log_e\left(1-\bar{\alpha}_{st}\right)\left[1+0.5\gamma^2\log_e\left(1-\bar{\alpha}_{st}\right)\right] + \frac{\sum_{i=1}^{n}\beta_i\left(\beta_i-1+\bar{\alpha}_{st}\right)S_i^2}{S^2\left(1-\bar{\alpha}_{st}\right)^2}$$

- where n is the number of room surfaces (or part room surfaces if whole surfaces are subdivided)

ENC p311

Empirical Expressions (Cont.)

See p310 in textbook

Fitzroy equation (used in auditoria design)

$$T_{60} = \frac{0.16\,V}{S^2}\left[\frac{-S_x}{\log_e\left(1-\bar{\alpha}_{xst}\right)} + \frac{-S_y}{\log_e\left(1-\bar{\alpha}_{yst}\right)} + \frac{-S_z}{\log_e\left(1-\bar{\alpha}_{zst}\right)}\right]$$

where

V is the room volume (m³),

S_x, S_y and S_z are the total areas of two opposite parallel room surfaces (m²),

$\bar{\alpha}_{xst}, \bar{\alpha}_{yst}$ and $\bar{\alpha}_{zst}$ are the average statistical absorption coefficients of a pair of opposite room surfaces

S is the total room surface area.

See ENC text pp. 310 - 311 for alternative empirical expressions.

ENC p310

Mean Free Path, Λ

- In length of time Λ/c, all sound energy will be once reflected
  - initial energy $\propto \langle p_0^2 \rangle$
  - final energy $\propto \langle p^2 \rangle$
  - Previously we had, $\langle p^2 \rangle = \langle p_0^2 \rangle e^{-Sc\bar{\alpha}t/4V}$;

    and for $t = \Lambda/c$,

    this becomes $\langle p^2 \rangle = \langle p_0^2 \rangle e^{-S\bar{\alpha}\Lambda/4V}$
- If all energy had been reflected once, $\langle p^2 \rangle = \langle p_0^2 \rangle e^{-\bar{\alpha}}$
- Equating the preceding two expressions for gives:

$$\boxed{\Lambda = 4V/S}$$

ENC p303

  - which is valid for rooms of any shape

Images from Pyrotek

Sound Absorbers

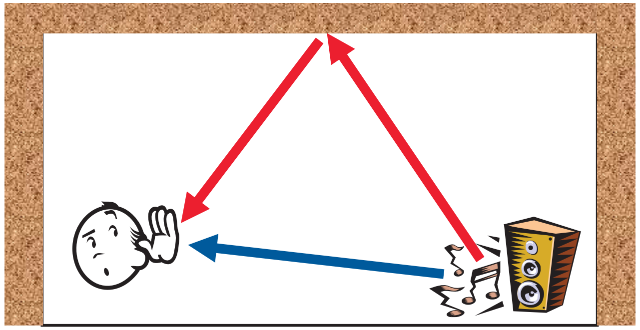
Example Products

- Micro-perforated panels and films

… The trick is that you don't even notice them!

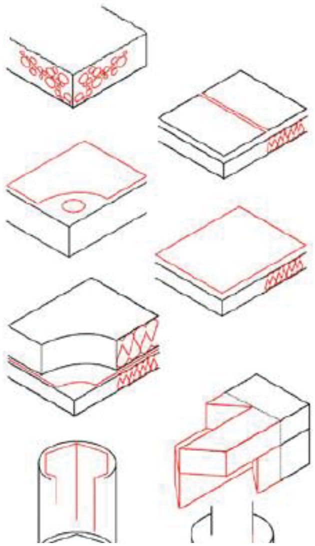
Sound Absorbers

- If the noise level in a room is too loud, what options have you got ????

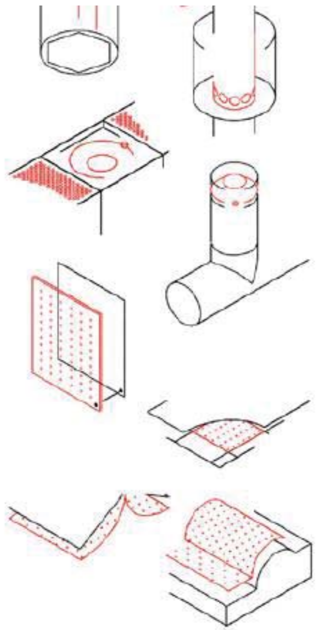


Sound Absorbers - Examples

1. Porous recycling glass
2. Slotted panel absorber
3. Membrane absorber box
4. Compound panel absober
5. Broadband compact absorber
6. Asymmetric structured absorber
7. Angular stack silencer



8. Tubular reactive silencer
9. Active silencer cassette
10. Active side-branch resonator
11. Microperforated panel absorber
12. Microperforated suspended ceiling
13. Microperforated foil absorber
14. Microperforated sheet-metal absorber
15. Microperforated glass absorber



Helmut Fuch (2013), "Applied Acoustics: Concepts, Absorbers, and Silencers for Acoustical Comfort and Noise Control" Springer, ISBN 978-3-642-29366-5, page vii.

Sound Absorbers

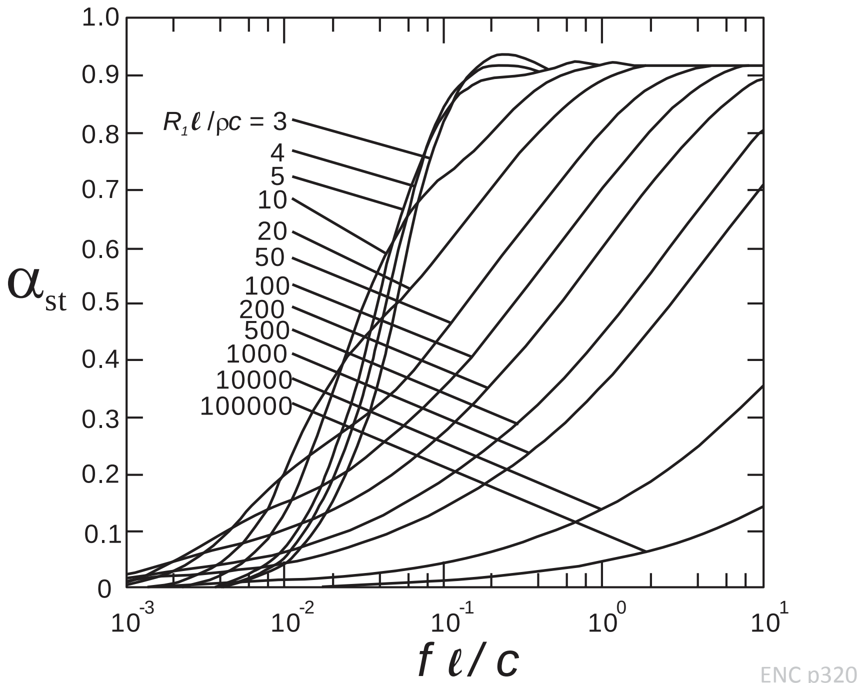
ENC p315

- characterised by absorption coefficients
- Sabine absorption coefficient, α
  - measured in reverberant room
  - measured values up to 1.2

$$S\bar{\alpha} = \frac{55.3V}{c}\left[\frac{1}{T_{60}} - \frac{(S'-S)}{S'T'_{60}}\right] \quad \left(\mathrm{m}^2\right)$$

S' =total area of walls, floor and ceiling and the sample material under test
S = sample material area (10-12 m²)

- Statistical absorption coefficient, α_{st}
  - optimum for $\dfrac{R_1\ell}{\rho c} = 2$ to 5
  - ℓ = porous material thickness (m)
  - α_{st} can also be calculated from normal incidence absorption coefficient measured using an impedance tube (Appendix C of text)



ENC p320

- Noise Reduction Coefficient (NRC)

$$\boxed{\mathrm{NRC} = \frac{\left(\bar{\alpha}_{250} + \bar{\alpha}_{500} + \bar{\alpha}_{1000} + \bar{\alpha}_{2000}\right)}{4}}$$

ENC p319

Porous Absorbers (Cont.)

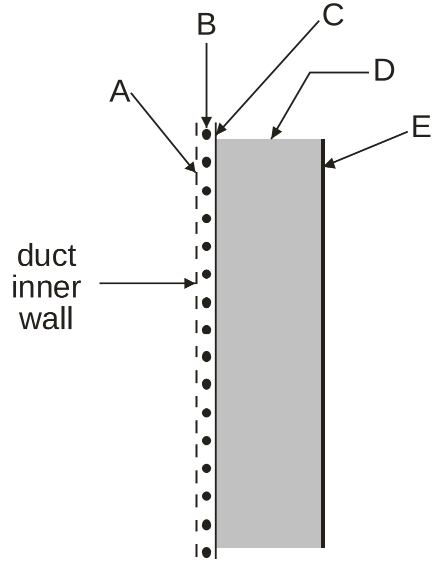
A: Perforated steel or Aluminium panel
- mechanical protection
- > 20% open, with less open area increasing low frequency absorption and decreasing high frequency absorption

B: spacer (12 mm mesh×2mm thick is OK)

C: Plastic wrapping
- moisture barrier
- dust, oil and chemical protection
- 20 - 30 μm thick polyester OK
- thicker materials increase low frequency absorption and decrease high frequency absorption

D: porous material absorber plus backing cavity

E: solid backing wall



Porous Absorbers (Cont.)

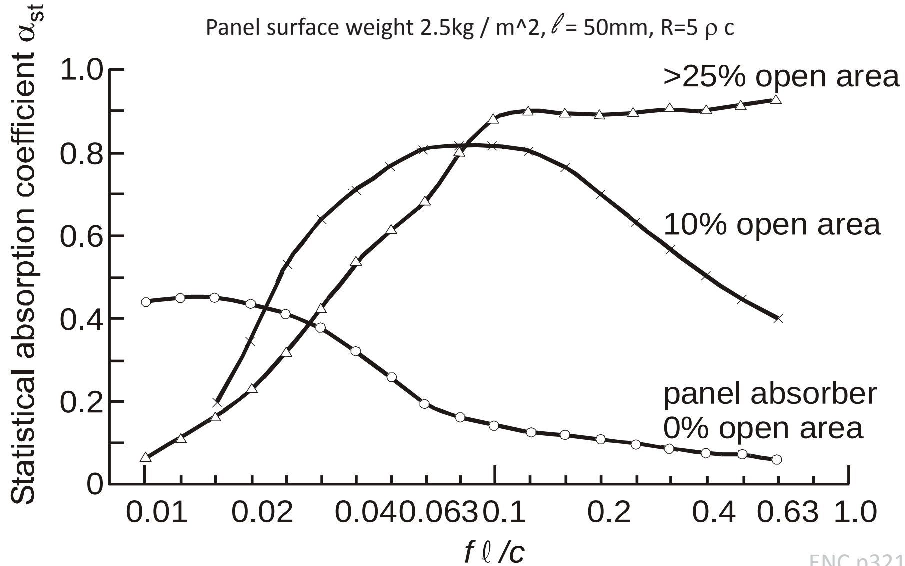
- frequency of maximum absorption calculated using:

$$\boxed{f_{\max} = \frac{c}{2\pi}\left[\frac{P/100}{L\left[t+0.85d\left(1-0.22d/q\right)\right]}\right]^{1/2}}$$

  - P = % open area
  - L = depth of porous material plus backing cavity (between perforated facing and solid backing wall)
  - d = diameter of holes in perforated panel
  - q = distance between centres of holes in perforated panel
  - t = thickness of perforated panel

ENC p320

Example – Measured Responses



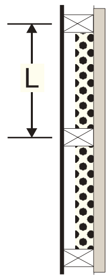
ENC p321

Panel Absorber Design

- Perforated panel
  - surface weight = 2.5 kg/m²
  - thickness = 3mm

- Porous liner
  - 50 mm thick
  - flow resistance = 5ρc

Panel Absorber Design

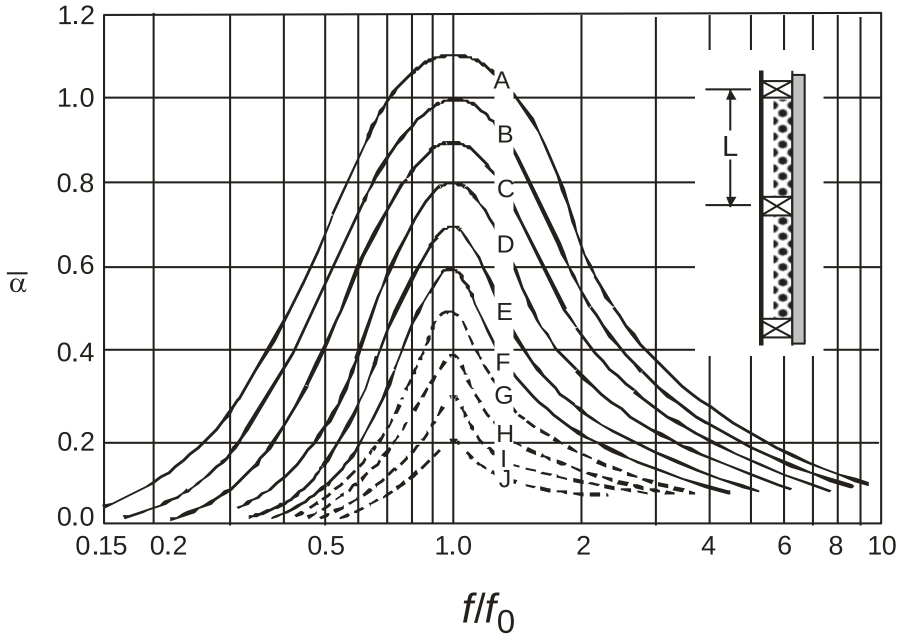


DESIGN PROCEDURE

- choose desired absorption curve (on figure with "hills", next page)
  - Panel supports L > 0.4m
  - Solid line represents porous material in cavity
  - Dashed line represents bare cavity
- Choose fundamental resonance of panel f_o
  - (should be the same as frequency trying to attenuate)
- Select cavity depth, D, and panel mass per unit area, m (on figure with criss-cross pattern)

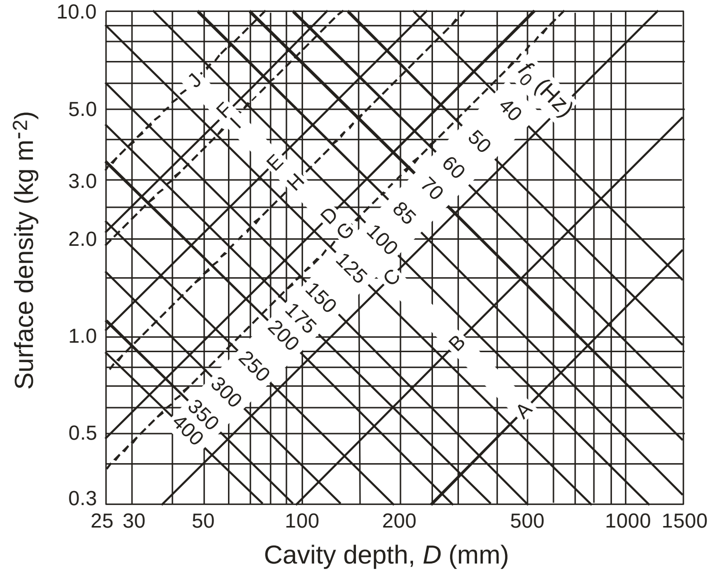
$$f_0 = \frac{1}{2\pi}\sqrt{\frac{\rho c^2}{mD}}$$

m = surface density of panel (kg / m²)
D = depth backing cavity



ENC p323

Panel Absorber Design



ENC p324

Combining Sound Absorption Coefficients

- The composite average absorption coefficient is

$$\boxed{\overline{\alpha} = \frac{\sum_{i=1}^{q} S_i \overline{\alpha}_i}{\sum_{i=1}^{q} S_i}}$$

$\bar{\alpha}$ = composite average absorption coeff
S_i = area of absorptive patch m^2
α_i = coeff of absorption of patch
q = number of patches

ENC p321

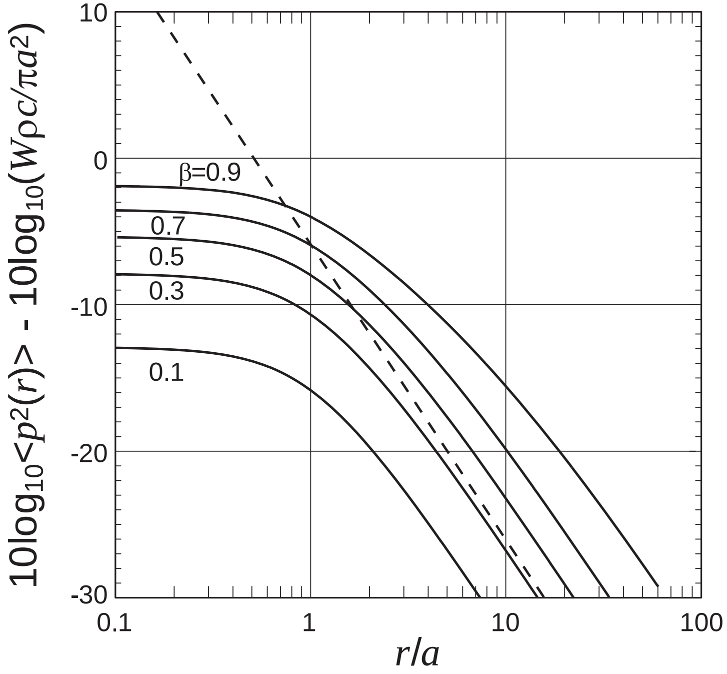
Flat Rooms

- Flat rooms are large in width and length compared to their height a
  - specularly reflecting floor and ceiling with energy reflection coefficients β_1, β_2 .

$$\langle p^2(\mathrm{r})\rangle = \mathrm{W}\frac{\rho c}{4\pi}\left[\frac{1}{r^2}+\sum_{k=1}^{\infty}\left(\frac{1/\beta_1+1/\beta_2}{\mathrm{r}_{2k-1}^2}+\frac{2}{\mathrm{r}_{2k}^2}\right)(\beta_1\beta_2)^k\right]$$

ENC p329

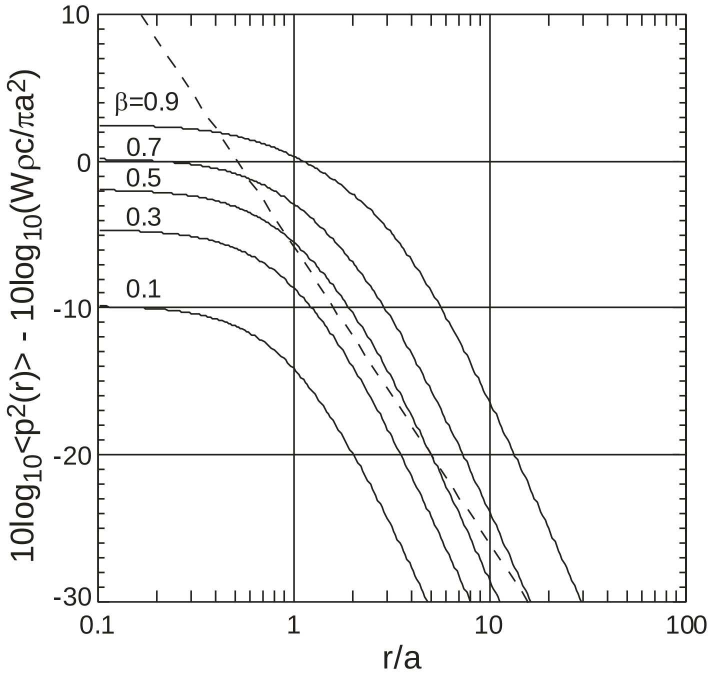
  - See chart on following page.



FLAT ROOMS

- Diffusely reflecting floor and ceiling
  - E.g. Furnished offices, factories with machinery or objects causing scattering of sound.

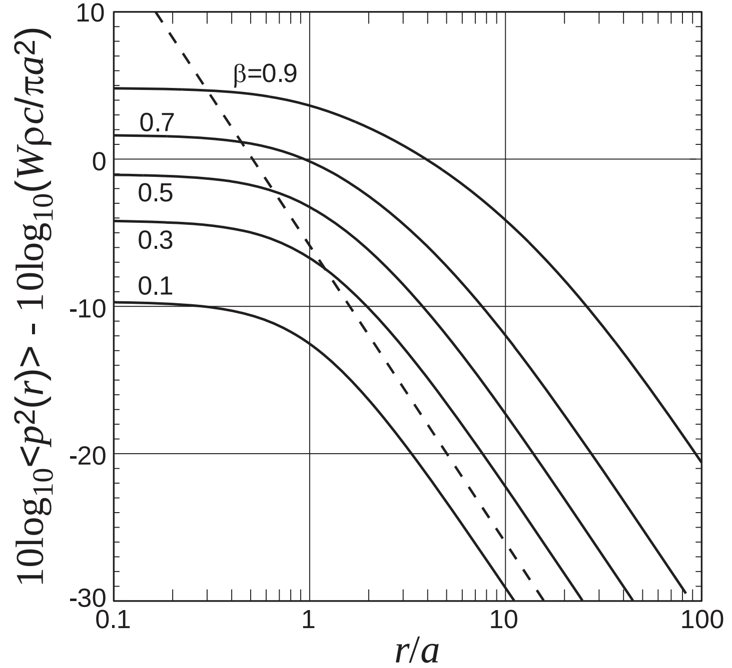
$$\left\langle p^2(\mathrm{r})\right\rangle_{\mathrm{R}} = \frac{\mathrm{W}\rho c\beta}{\pi a^2}\int_0^\infty \frac{\mathrm{e}^{-\mathrm{z}}}{1-\beta z K_1(z)} J_0(r\,z/a)\,\mathrm{zdz}$$



LONG ROOMS

- Rectangular cross section dimensions $a \times b$, specularly reflecting walls, point source

$$\left\langle p^2(\mathrm{r})\right\rangle = \frac{\mathrm{W}\rho c}{4\pi}\left[\frac{1}{r^2} + \sum_{m=1}^{\infty}\sum_{n=1}^{\infty}\frac{4\beta^{m+n}}{(ma)^2+(nb)^2+r^2} + \sum_{n=1}^{\infty}\frac{2\beta^n}{(nb)^2+r^2} + \sum_{m=1}^{\infty}\frac{2\beta^m}{(ma)^2+r^2}\right]$$

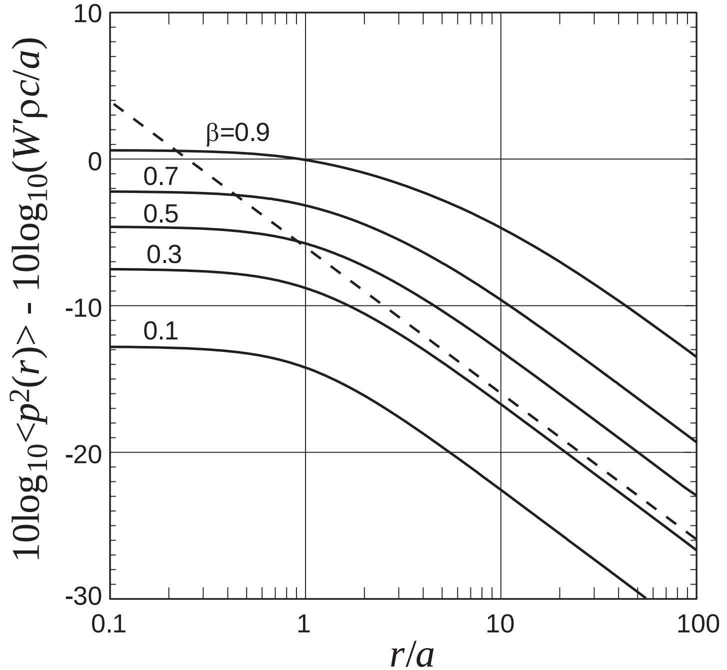


LONG ROOM

Rectangular cross section, specularly reflecting walls, line source

$$\left\langle p^2(\mathrm{r})\right\rangle = \frac{W'\rho c}{4}\left[\frac{1}{r} + \sum_{n=1}^{\infty}\frac{2\beta^n}{\left[(na)^2 + r^2\right]^{1/2}}\right]$$

see text for some other examples.



Will Sound Absorbing Material Reduce Noise Levels At A Particular Location In A Room?

- Sound absorbing material only reduces reverberant field levels
  - to determine potential effectiveness, compare existing values of

    $$\frac{4}{R} \text{ and } \frac{D\theta}{4\pi r^2}$$

  - If the 2 terms are equal, then the maximum possible noise reduction achieved by lining walls ceiling and floor and/or hanging absorbing baffles is 3dB

- Decrease in reverberant sound level is:

$$\Delta L_p = 10\log_{10}\left[\frac{R_f}{R_i}\right]$$

ENC p344

R_i = initial room constant m^2
R_f = room constant after addition of absorbing material m^2

  - best to treat hard surfaces
  - should distribute material uniformly

Optimising Reverberation Times

$$T_{60} = K\left[0.0118\ V^{1/3} + 0.1070\right]$$

ENC p345

T_{60} = reverberation time in seconds
V = volume of room in m^3
K = 4 for speech
K = 5 for orchestras
K = 6 for choirs

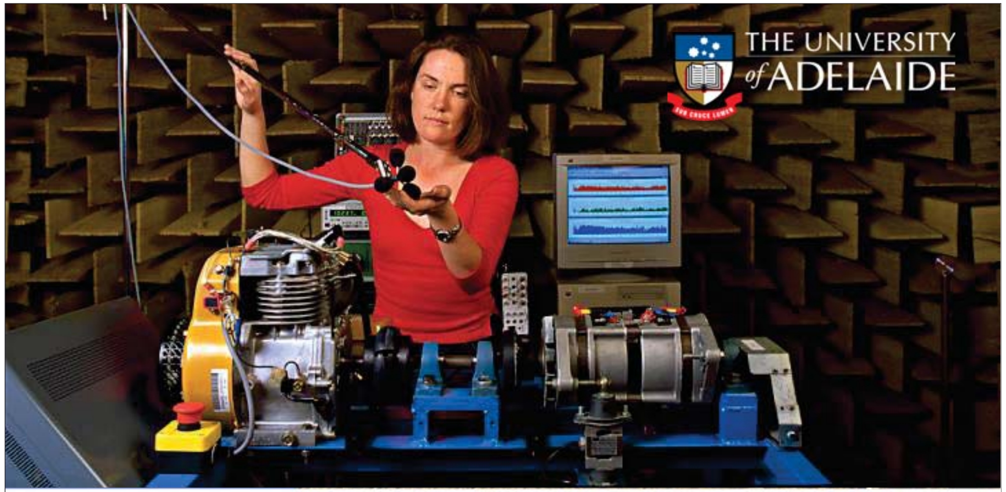
Increase T_{60} by 40% to 100% at 250Hz and below.

Other Topics

- AUDITORIUM DESIGN - IMPORTANT PARAMETERS (See text book for this discussion)
- SOUND REINFORCEMENT (See text book for this discussion)

Summary

- Modal response of rooms.
- Diffuse sound field.
- Sabine rooms.
- Composite Average Absorption.
- Reverberation time.
- Panel absorbers.
- Response of long and flat rooms.



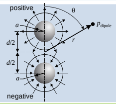
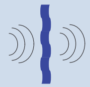
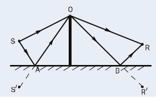
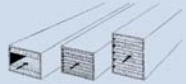
Carl Howard

Outdoor Sound Propagation

adelaide.edu.au

seek LIGHT

Course Content – How it fits

TOPIC	
Sound Sources	
Outdoor	
Acoustic Enclosures and Transmission Loss of Panels	
Barriers	
Mufflers	

Contents

- Geometric spreading factors.
- Standard prediction schemes.
- Factors affecting transmission and attenuation over long distances.

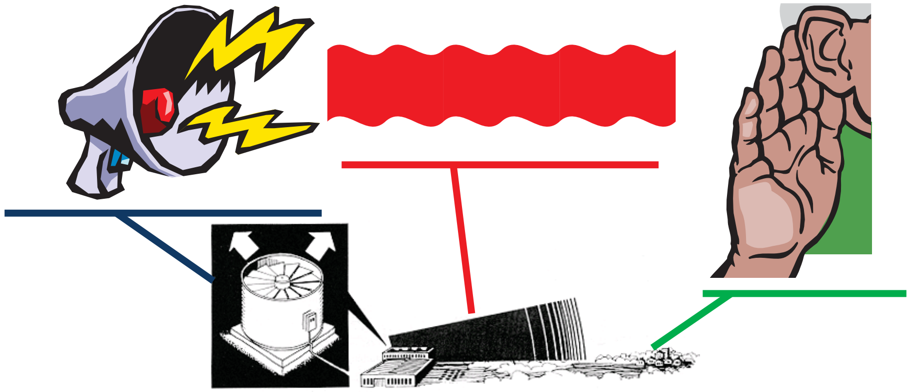
Learning Outcomes

- Be able to calculate the sound pressure level from an acoustic source due to propagation outdoors.
- Understand the factors that affect sound propagation outdoors.

Noise Control Methods

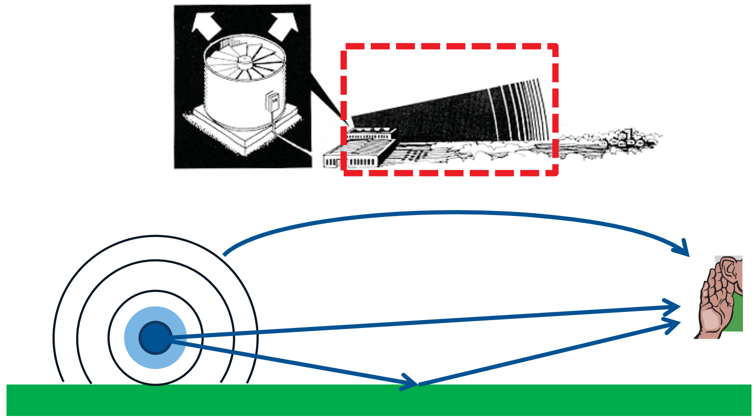
- Noise and vibration can be reduced at the

Source Path Receiver



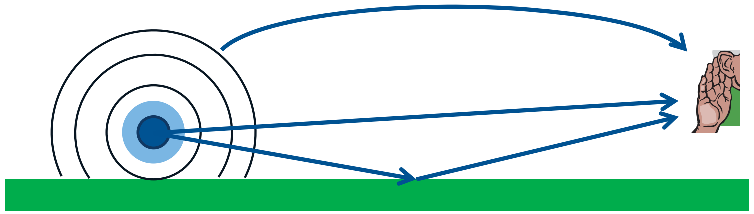
What is important?

- What factors are important?



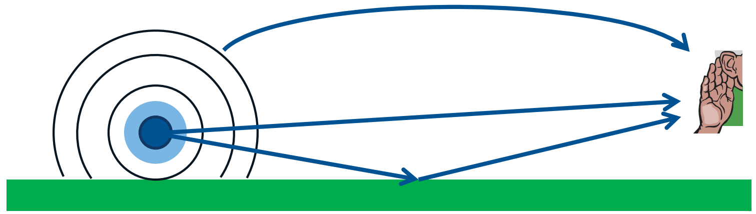
What is important?

- Source
  - Sound power
  - Spreading: Hemispherical, Line Source, Incoherent Plane
  - Directivity



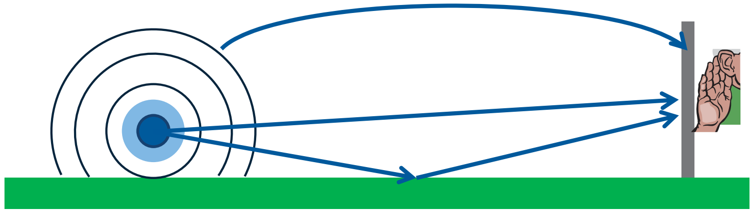
What is important?

- Path
  - Attenuation from reflection off ground
  - Losses from transmission through air
  - Meteorological: wind, temperature gradients, turbulence
  - Trees and foliage



What is important?

- Path
  - Barriers (discussed in separate lecture)



Question …

A sound power source 3m above ground level radiates of 2 watts out the side of a building in the 2000 Hz octave band. The closest community location is at a distance of 150m from the building in a direction normal to the plane of the opening. The ground between the opening and the community is grass covered. The following questions refer only to the 2000Hz octave band.

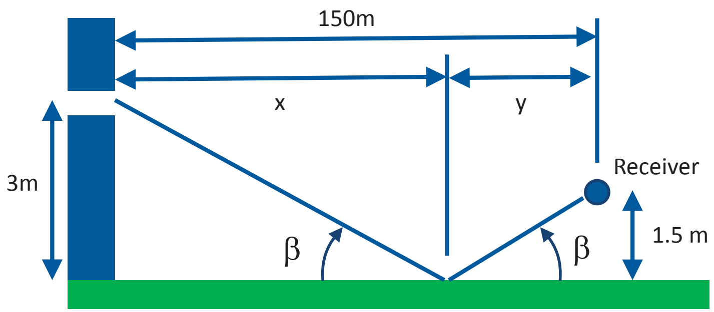
a) Calculate the sound power level L_w radiated from the source.

b) Calculate the excess attenuation A_g due to ground reflection for sound travelling from the opening to the nearest community location (1.5m above ground) (use Fig. 5.20 in your text, R1= 2.25×10^5).

c) Calculate the loss due to atmospheric absorption (in dB). Assume RH = 25%, and a temperature of 20°C.

d) Ignoring all other losses not mentioned above, calculate the sound pressure level at the community location of (b) above.

Question ….



Outdoor Sound Propagation

- Determine power levels L_w of all sources.
- calculate individual components of excess attenuation A_{Ei} for all sources, due to air absorption, ground reflections, barriers, etc.
- Compute L_{pi} at selected points in the environment for each source *i*.
- At each location, logarithmically combine individual source contributions

Question ...

A sound power source 3m above ground level radiates of 2 watts out the side of a building in the 2000 Hz octave band. The closest community location is at a distance of 150m from the building in a direction normal to the plane of the opening. The ground between the opening and the community is grass covered. The following questions refer only to the 2000Hz octave band.

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d) Ignoring all other losses not mentioned above, calculate the sound pressure level at the community location of (b) above.

Answer...

Calculate the sound power level L_w radiated from the source.

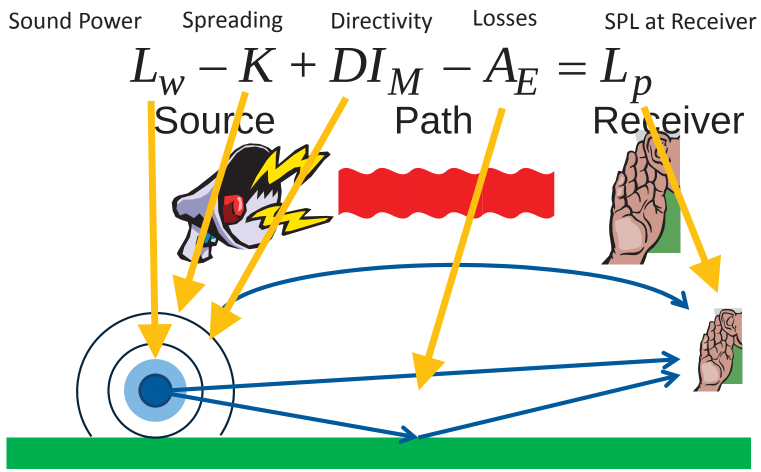
$$\begin{aligned} L_w &= 10\log_{10} W + 120 \\ &= 10\log_{10} 2 + 120 \\ &= 123 \text{ dB re } 10^{-12}\text{W} \end{aligned}$$

Prediction Accuracy Influences

- accuracy of L_w and DI_M
- variability in weather
  - calculate L_{eq}, L_{10}, L_{90}
- hope for ±5dB

Calculating L_{pi} From L_{wi}

- for a single source $L_p = L_w - K + DI_M - A_E$ ENC p234



Calculating L_{pi} From L_{wi}

- For a single source $\boxed{L_p = L_w - K + DI_M - A_E}$ ENC p234
- geometrical spreading factor, *K*
  - simple source $K = 10\log_{10} 4\pi + 20\log_{10} r$ ENC p235
  - simple source radiating hemi-spherically from ground or wall
    $K = 10\log_{10} 2\pi + 20\log_{10} r$ ENC p235
  - line source
    $K = 10\log_{10}\left[4\pi r_0 D/(\alpha_u - \alpha_l)\right]$ ENC p235

Calculating L_{pi} From L_{wi}

- geometrical spreading factor, *K* (cont.)
  incoherent plane source (wall)

$$K = 10\log_{10} 2\pi + 20\log_{10} r - 10\log_{10} F(\alpha, \beta, \gamma, \delta)$$

where

$$F = \frac{\gamma^2}{\alpha}\left[\tan^{-1}\frac{(\alpha-\beta)(\delta+1/2)}{\gamma\sqrt{(\alpha-\beta)^2+(\delta+1/2)^2+\gamma^2}} + \tan^{-1}\frac{\beta(\delta+1/2)}{\gamma\sqrt{\beta^2+(\delta+1/2)^2+\gamma^2}} - \tan^{-1}\frac{(\alpha-\beta)(\delta-1/2)}{\gamma\sqrt{(\alpha-\beta)^2+(\delta-1/2)^2+\gamma^2}} - \tan^{-1}\frac{\beta(\delta-1/2)}{\gamma\sqrt{\beta^2+(\delta-1/2)^2+\gamma^2}}\right]$$

ENC p235

Calculating L_{pi} From L_{wi}

- directivity index, DI_M
  - accounts for source directivity
  - excludes ground reflection
  - includes effect of other reflecting surfaces
- excess attenuation factor, A_E

$$\boxed{A_E = A_a + A_b + A_f + A_g + A_m}$$ ENC p236

A_E = excess attenuation in dB
A_a = atten due to air absorption in dB
A_b = atten due to barriers, houses, process equipment in dB
A_f = atten due to forests in dB
A_g = atten due to ground reflection in dB
A_m = atten due to meteorological effects in dB

Prediction Schemes For Outdoor Sound Propagation

Oil Companies Materials Association
OCMA, (1972)

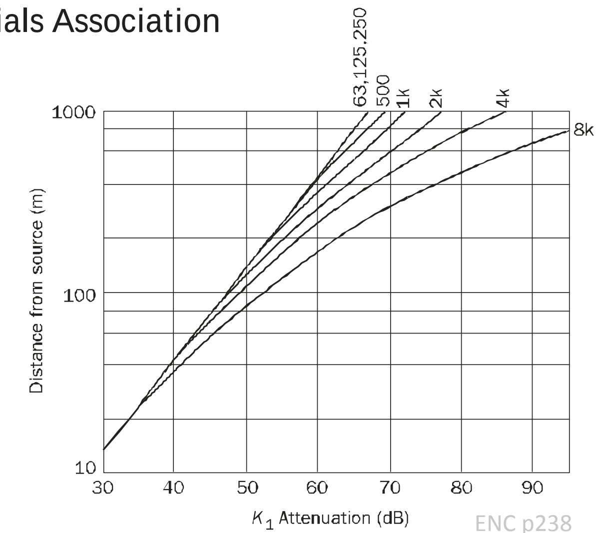
$$L_p = L_w - K_1 - K_2$$

where

$$K_1 = 10\log_{10} 2\pi + 20\log_{10} r + A_a$$

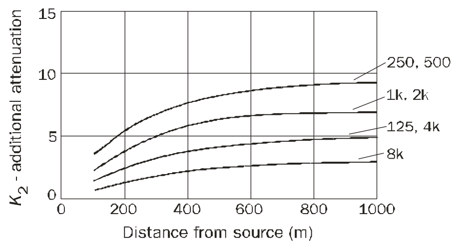
and

$$K_2 = A_b + A_g + A_m$$

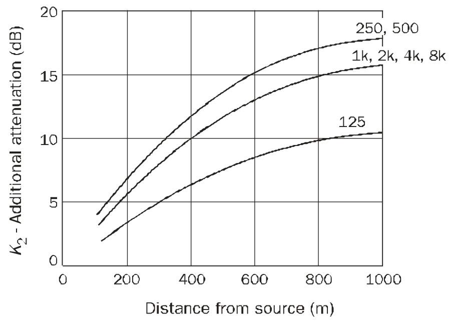


ENC p236 ENC p238

- OCMA (cont.)



minimal shielding



significant shielding

ENC p239

More Recent Prediction Schemes

- excess attenuation effects, A_E, calculated individually and decibels are added arithmetically (except barrier, wind and temperature gradient, and ground effects are combined)
- atmospheric absorption,

$$\boxed{A_a = mX}$$ ENC p237

  - see text, p240 for values of m
  - function of temperature and RH
- shielding by barriers, A_b - see Ch. 8
- attenuation due to forests and foliage
- meteorological influences
  - turbulence (usually included in ground effect)
  - wind gradients
  - temperature gradients
- ground effects

Atmospheric Absorption A_a

Table 5.3 Attenuation due to atmospheric absorption (calculated from Sutherland and Bass, 1979)

Relative humidity (%)	Temp-erature (°C)	m (dB per 1000 m) 63 Hz	125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz
25	15	0.2	0.6	1.3	2.4	5.9	19.3	66.9
	20	0.2	0.6	1.5	2.6	5.4	15.5	53.7
	25	0.2	0.6	1.6	3.1	5.6	13.5	43.6
	30	0.1	0.5	1.7	3.7	6.5	13.0	37.0
50	15	0.1	0.4	1.2	2.4	4.3	10.3	33.2
	20	0.1	0.4	1.2	2.8	5.0	10.0	28.1
	25	0.1	0.3	1.2	3.2	6.2	10.8	25.6
	30	0.1	0.3	1.1	3.4	7.4	12.8	25.4
75	15	0.1	0.3	1.0	2.4	4.5	8.7	23.7
	20	0.1	0.3	0.9	2.7	5.5	9.6	22.0
	25	0.1	0.2	0.9	2.8	6.5	11.5	22.4
	30	0.1	0.2	0.8	2.7	7.4	14.2	24.0

ENC p240

Question ...

A sound power source 3m above ground level radiates of 2 watts out the side of a building in the 2000 Hz octave band. The closest community location is at a distance of 150m from the building in a direction normal to the plane of the opening. The ground between the opening and the community is grass covered. The following questions refer only to the 2000Hz octave band.

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c) Calculate the loss due to atmospheric absorption (in dB). Assume RH = 25%, and a temperature of 20°C.

d) Ignoring all other losses not mentioned above, calculate the sound pressure level at the community location of (b) above.

Answer ...

Calculate the loss due to atmospheric absorption (in dB). Assume RH = 25%, and a temperature of 20°C.

atmospheric absorption, $A_a = mX$

Table 5.3 Attenuation due to atmospheric absorption (calculated from Sutherland and Bass, 1979)

Relative humidity (%)	Temp-erature (°C)	63 Hz	125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz
		m (dB per 1000 m)						
25	15	0.2	0.6	1.3	2.4	5.9	19.3	66.9
	20	0.2	0.6	1.5	2.6	5.4	$\boxed{15.5}$	53.7
	25	0.2	0.6	1.6	3.1	5.6	13.5	43.6
	30	0.1	0.5	1.7	3.7	6.5	13.0	37.0

$$A_a = 15.5 \times \frac{150}{1000} = 2.3 \text{ dB}$$

Trees And Foliage Attenuation

For a heavily wooded area (Hoover, 1961) obtained:
Other work has produced the following conclusions:

$$A_f = 0.01r\, f^{1/3}$$

ENC p241

- a single row of trees along the highway or near houses results in negligible attenuation of the noise;
- a continuous strip of oleander or equivalent shrubs, at least 2.5 m high and 4.5 to 6 m wide, planted along the edge of a highway shoulder, provides noise attenuation of 1-3 dBA at distances of up to 15 m from the rear edge of vegetation;
- a strip of trees, 60 m wide can attenuate traffic noise by up to 10dB(A); and
- vegetation is not generally considered an effective traffic noise barrier, although it does have an effect in attenuating noise at frequencies above 2 kHz.

Trees And Foliage Attenuation (Cont.)

- psychological effect of vegetation as a barrier is important - in many cases if the noise source is not visible, it is less noticeable and annoying, even if the level is not significantly changed.

- ISO 9613-2 (1996) gives the attenuation values in Table 5.4 for sound propagation through dense foliage. For distances of less than 20 m, the values given are absolute dB. For distances between 20 and 200 m, the values given are dB/m and for distances greater than 200 m, the value for 200 m is used.

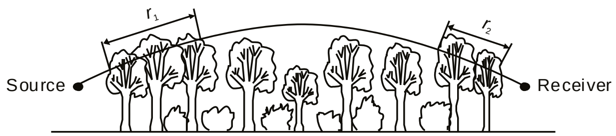
	Octave band centre frequency (Hz)							
	63	125	250	500	1000	2000	4000	8000
A_p (dB/m)	0	.015	.025	.025	.02	.02	.015	.015
A_f (dB) for 10m ≤ r_f ≤ 20m	0	0	1	1	1	1	2	3
A_f (dB/m) for 20m ≤ r_f ≤ 200m	0.02	0.03	0.04	0.05	0.06	0.08	0.09	0.12

A_p = attenuation due to process equipment (see next page)

ENC p242

Trees And Foliage Attenuation (Cont.)

- The distance of travel through the foliage is not equal to the extent of the foliage between the source and receiver
  - depends on the height of the source and receiver and the radius of curvature of the propagating wave as a result of wind and temperature gradients - see figure



Shielding By Barriers

$$A_{bhp} = A_b + A_h + A_p$$

ENC p237

- A_b , the attenuation due to large barriers and large extended buildings - see Ch. 8.
- calculation method modified to account for wind and temperature gradients (Sect. 8.5).
- The excess attenuation due to housing (A_h) may be calculated using (ISO 9613-2 (1996):

$$A_h = 0.1 B r_b - 10\log_{10}\left[1-(P/100)\right] \quad \text{(dB)}$$

where, B is the density of buildings along the path (total plan area of buildings divided by the total ground area)

r_b is the distance that the curved sound ray travels through the houses

P is the percentage ($\leq$ 90%) of the length of housing facades relative to the total length of a road or railway in the vicinity. The term containing P is only used if there are well defined rows of houses perpendicular to the direction of sound propagation.

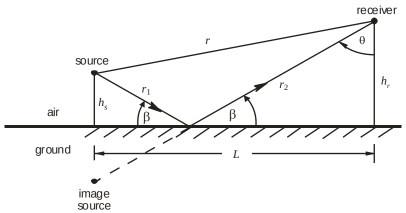
Shielding By Barriers (Cont.)

- The excess attenuation due to process equipment (A_p) may be calculated using the values in line 1 of the table in the foliage section
  - The distance (r_1) to be used for the calculation is only that part of the curved sound ray (close to the source) that travels through the process equipment and the maximum attenuation that is expected is 10dB.
  - The distance is equivalent to distance r_1 in the figure in the foliage attenuation section with the foliage replaced by process equipment.

Ground Effects - 5 Methods

- SIMPLE METHOD
  - Hard ground, A_g = -3dB
  - Soft ground, A_g = 0dB
- CONCAWE method (CONservation of Clean Air and Water in Europe)
- PLANE WAVE REFLECTION METHOD
- ISO 9613 METHOD
- SPHERICAL WAVE REFLECTION METHOD INCLUDING EFFECTS OF TURBULENCE

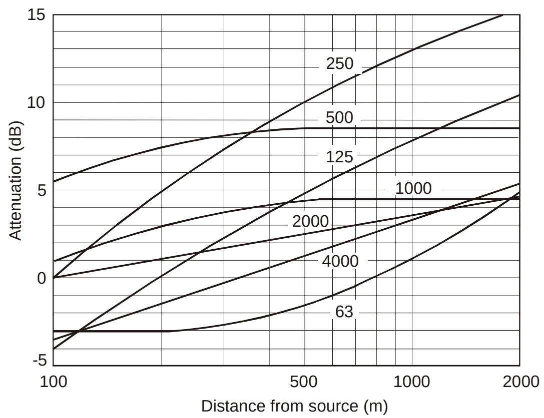
ENC p244



(not covered here but discussed in text book)

CONCAWE Method

- Based on experimental data taken in vicinity of 3 processing plants in Europe.



ENC p243

Plane Wave Reflection Method For Ground Attenuation Calculations

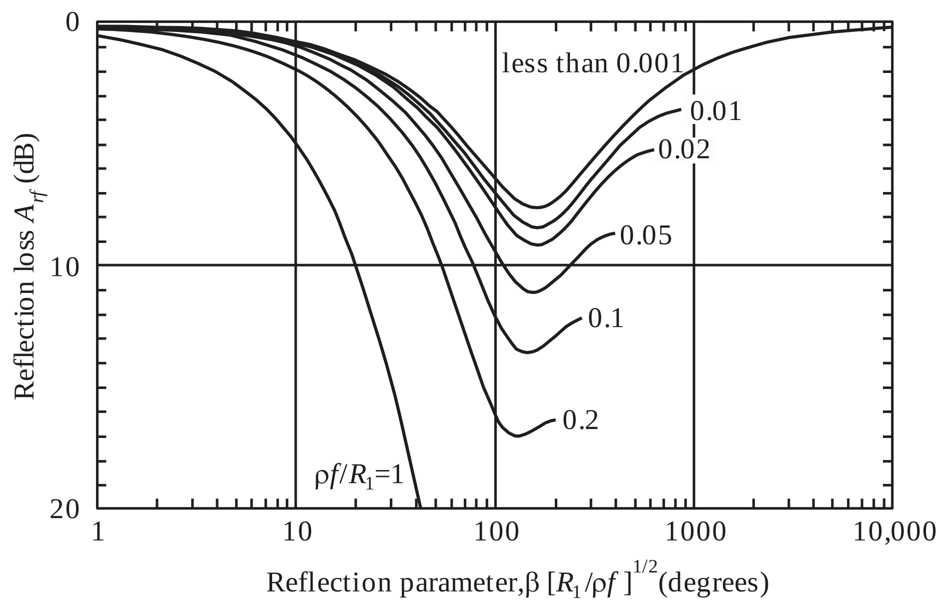
- assumes incoherent combination of direct and reflected waves, A_{RL} = -20log$_{10}|R_p|$ dB

$$R_p = \frac{Z_m \cos\theta - \rho c \cos\psi}{Z_m \cos\theta + \rho c \cos\psi}$$

- A_g varies between 0 and -3dB, depending on value of A_R

$$A_g = -10\log_{10}\left[1+|R_p|^2\right] = -10\log_{10}\left[1+10^{-A_R/10}\right]$$

ENC p244



ENC p245

Question ...

A sound power source 3m above ground level radiates of 2 watts out the side of a building in the 2000 Hz octave band. The closest community location is at a distance of 150m from the building in a direction normal to the plane of the opening. The ground between the opening and the community is grass covered. The following questions refer only to the 2000Hz octave band.

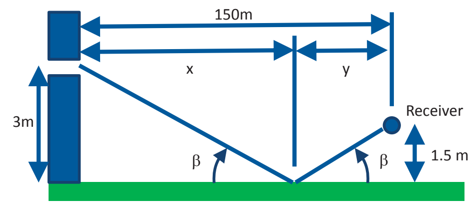
a) Calculate the sound power level L_w radiated from the source.

b) Calculate the excess attenuation A_g due to ground reflection for sound travelling from the opening to the nearest community location (1.5m above ground) (use Fig. 5.20 in your text, R1= 2.25 × 10^5).

c) Calculate the loss due to atmospheric absorption (in dB). Assume RH = 25%, and a temperature of 20°C.

d) Ignoring all other losses not mentioned above, calculate the sound pressure level at the community location of (b) above.

Answer ....



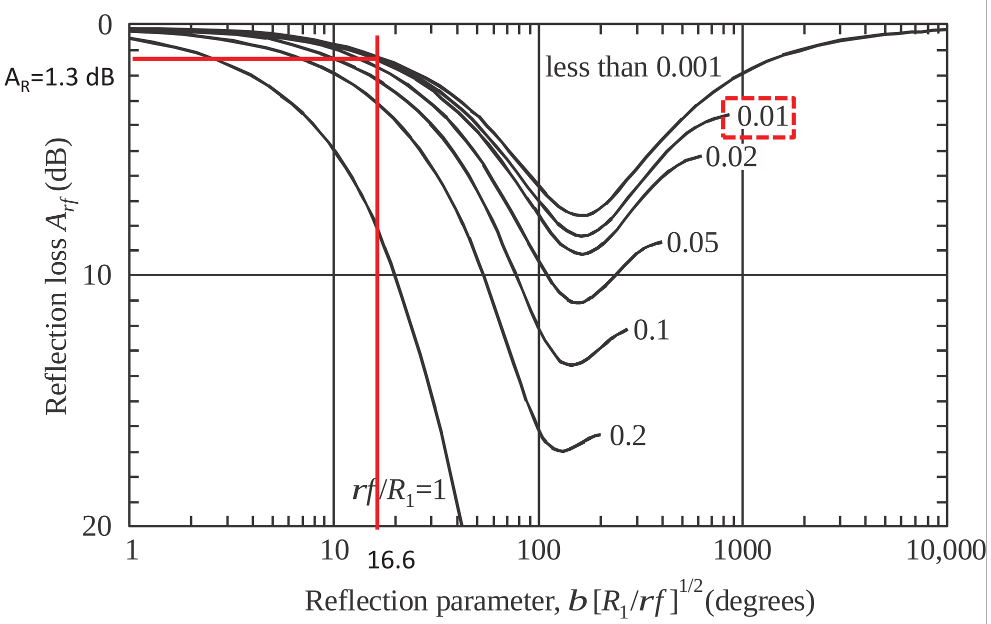
Calculate the excess attenuation A_g due to ground reflection for sound travelling from the opening to the nearest community location (1.5m above ground) (use Fig. 5.20 in your text, R1= 2.25 × 10^5).

Calculate x,y,β using similar triangles. ... x+y=150, x=2y = 100, y=50.

$\tan\beta = 3/100 \qquad \beta = 1.72°$

$$\frac{\rho f}{R_1} = \frac{1.21 \times 2000}{2.25 \times 10^5} = 0.01075$$

$$\beta\left[\frac{R_1}{\rho f}\right]^{1/2} = 1.72\left[\frac{1}{0.01075}\right]^{1/2} = 16.6$$



ENC p245

Answer ....

$$A_g = -10\log_{10}\left[1+10^{-A_R/10}\right]$$
$$= -10\log_{10}\left[1+10^{-1.3/10}\right] = -2.4\text{ dB}$$

Thus the effect of the ground is to ***increase*** the level at the receiver by 2.4dB.

$$A_E = A_a + A_b + A_f + A_g + A_m$$

$$L_p = L_w - K + DI_M - A_E$$

Question ...

A sound power source 3m above ground level radiates of 2 watts out the side of a building in the 2000 Hz octave band. The closest community location is at a distance of 150m from the building in a direction normal to the plane of the opening. The ground between the opening and the community is grass covered. The following questions refer only to the 2000Hz octave band.

a) Calculate the sound power level L_w radiated from the source.
b) Calculate the excess attenuation A_g due to ground reflection for sound travelling from the opening to the nearest community location (1.5m above ground) (use Fig. 5.20 in your text, R1= 2.25 × 10^5).
c) Calculate the loss due to atmospheric absorption (in dB). Assume RH = 25%, and a temperature of 20°C.
d) Ignoring all other losses not mentioned above, calculate the sound pressure level at the community location of (b) above.

Answer ...

$$\langle p_i^2 \rangle = \frac{W_i \rho c D_i}{2\pi r_i^2} = \frac{2 \times 1.21 \times 343 \times 1}{2\pi \times 150^2} = 5.87 \times 10^{-3}\,\text{Pa}^2$$

$$L_p = 10\log_{10}\frac{5.87 \times 10^{-3}}{(20 \times 10^{-6})^2} = 71.66\text{ dB}$$

$$A_E = A_a + A_b + A_f + A_g + A_m$$
$$= A_a + A_g = 2.3 + (-2.4) = -0.1\text{ dB}$$

$$L_p = L_w - K + DI_M - A_E$$
$$= 71.7 - (-0.1) = 71.8\text{ dB}$$

ISO 9613 Method For Ground Attenuation

- moderately complex and yields results of moderate accuracy.
- space between the source and receiver is divided into three zones
  - source zone extends a distance of $30h_s$ from the source towards the receiver
  - receiver zone extends $30h_r$ from the receiver towards the source.
  - middle zone includes the remainder of the path between the source and receiver
    - will not exist if the source / receiver separation is less than $r = 30h_s + 30h_r$.
- The total excess attenuation due to the ground is the sum of the excess attenuations for each of the three zones. That is:

$$A_g = A_s + A_{mid} + A_r$$

- Values for each of the three quantities on the right hand side of Equation (5.176) may be calculated using the table on the next page

ENC p245

ISO 9613 Method For Ground Attenuation

Octave band ground attenuation contributions, A_s, A_r and A_{mid} (after ISO 9613-2, 1996)

	Octave band centre frequency (Hz)							
	63	125	250	500	1000	2000	4000	8000
A_s (dB)	-1.5	$-1.5+G_s a_s$	$-1.5+G_s b_s$	$-1.5+G_s c_s$	$-1.5+G_s d_s$	$-1.5(1-G_s)$	$-1.5(1-G_s)$	$-1.5(1-G_s)$
A_r (dB)	-1.5	$-1.5+G_r a_r$	$-1.5+G_r b_r$	$-1.5+G_r c_r$	$-1.5+G_r d_r$	$-1.5(1-G_r)$	$-1.5(1-G_r)$	$-1.5(1-G_r)$
A_{mid} (dB)	3q	-3q(1-Gm)	$-3q(1-G_m)$	$-3q(1-G_m)$	$-3q(1-G_m)$	$-3q(1-G_m)$	$-3q(1-G_m)$	$-3q(1-G_m)$

- parameter, G has a value of 0.0 for hard ground, a value of 1.0 for soft (or porous) ground and for a mixture of hard and soft ground it is equal to the fraction of ground that is soft.

ENC p246

ISO 9613 Method For Ground Attenuation

- G_s, G_r and G_m are the values of G corresponding to the source zone, the receiver zone and the middle zone

$$a_s, a_r = 1.5 + 3.0e^{-0.12(h_{s,r}-5)^2}\left(1-e^{-r/50}\right) + 5.7e^{-0.9h_{s,r}^2}\left(1-e^{-2.8\times10^{-6}\times r^2}\right)$$

$$b_s, b_r = 1.5 + 8.6e^{-0.09h_{s,r}^2}\left(1-e^{-r/50}\right)$$

$$c_s, c_r = 1.5 + 14.0e^{-0.46h_{s,r}^2}\left(1-e^{-r/50}\right)$$

$$d_s, d_r = 1.5 + 5.0e^{-0.09h_{s,r}^2}\left(1-e^{-r/50}\right)$$

ENC p246

ISO 9613 Method For Ground Attenuation

- A_{mid} is zero for source / receiver separations of less than $30h_s + 30h_r$ and for greater separation distances it is calculated using:

$$q = 1 - \frac{30(h_s + h_r)}{r}$$

ISO method assumes the worst case of downwind propagation so effectively it includes meteorological effects and these should not be added separately

ENC p246

Meteorological Influences

Excess attenuation A_m (dB) due to wind and temperature gradients at two distances, 110 m and 616 m.

Total vertical gradient	**Octave band centre frequency (Hz)**									
(s^{-1})	31.5	63	125	250	500	1000	2000	4000	8000	16000
	110 meters									
+0.075	-2	-2	-0.5	3	-2	-5	-2	-2	-2	-2
-0.075	1	1	2.5	0	2	6	10	6	6	6
	616 meters									
+0.075	-5	-5	-2	0	-9	-9	-6	-7	-7	-7
-0.075	5	5	6	4	7	7	7	6	6	6

ENC p254

Meteorological Influences

- total sonic gradient due to wind and temp.

$$\frac{dc}{dh} = \frac{dU}{dh} + \left[\frac{\partial c}{\partial h}\right]_T$$

where, $\frac{dU}{dh} = \xi \frac{U(h)}{h}$

$U(h) = U_0 h^{\xi}$; h=10m and ξ=0.15

$U(h)$ is wind speed at height h

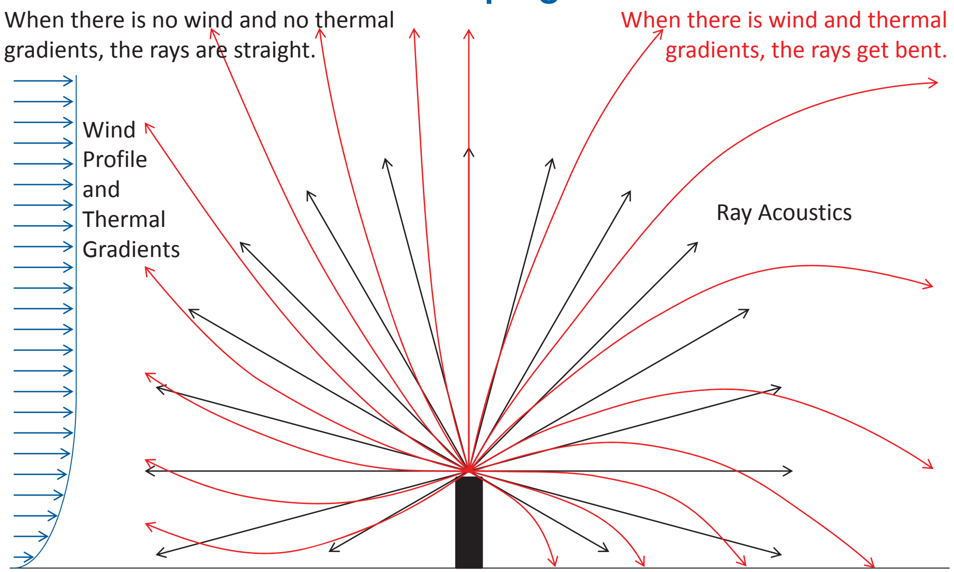
$$\left[\frac{\partial c}{\partial h}\right]_T = 10.3 \frac{dT}{dh}[T_0 + 273]^{-1/2}$$

T_0 is temperature at 1m height

Meteorological Influences

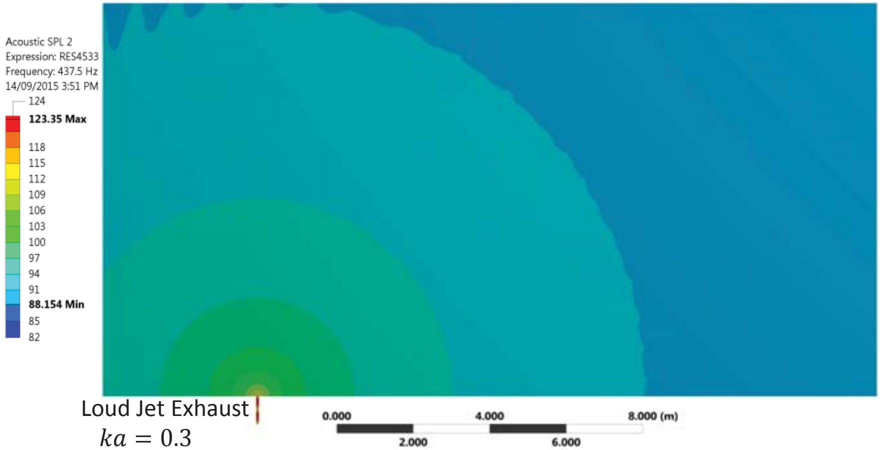
- radius of curvature of sound waves $R = c_0 \left[\frac{dc}{dh}\right]^{-1}$ ENC p251
  - c_0 = speed of sound at sea level and 0°C (331m/sec)
  - if r +ve, rays curve down
  - if r -ve, rays curve up
- Shadow zone (receiver upwind of source) - see text book for detailed discussion)
  - direct sound cannot penetrate due to upwards diffraction
  - Some penetration due to scattering
  - increase in attenuation in the shadow zone to be up to 30 dB
  - requires a negative sonic gradient to occur (no temp. inversion or upwind)

Far-Field Sound Propagation Model



Current Research

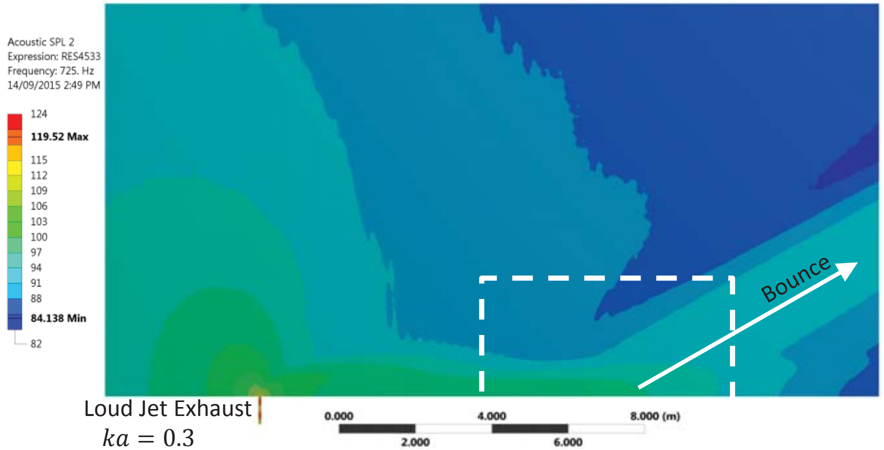
- When there is **no** wind and temperature gradients there is hemispherical spreading.



"Directivity analysis of sound in turbulent exhaust jets with laminar cross-flow: A numerical study" Orddom Y. Leav, Benjamin S. Cazzolato, Carl Q. Howard (2015), Acoustics 2015 conference , Hunter Valley, NSW, Australia, November

Current Research

- When there is wind and temperature gradients the sound can beam onto a community.



"Directivity analysis of sound in turbulent exhaust jets with laminar cross-flow: A numerical study" Orddom Y. Leav, Benjamin S. Cazzolato, Carl Q. Howard (2015), Acoustics 2015 conference , Hunter Valley, NSW, Australia, November

Alternative Calculation of Meteorological Influences

- 6 categories based on total sonic gradient

Meteor-ological Category	Pasquill Stability Category A,B	C,D,E wind speed, v (ms^{-1})	F,G
1	v < - 3.0	-	-
2	-3.0 < v < -0.5	v < - 3.0	-
3	-0.5 < v < +0.5	-3.0 < v < -0.5	v < -3.0
4*	+0.5 < v < +3.0	-0.5 < v < +0.5	-3.0 < v < +0.5
5	v > +3.0	+0.5 < v < +3.0	-0.5 < v < +0.5
6	-	v > + 3.0	+5.0 < v < +3.0

* Category with assumed zero meteorological influence.

ENC p255

Pasquill Stability Category

Wind Speed	Day Time Incoming Solar Radiation mW/cm2				1 hour before sunset or after sunrise	Night-Time Cloud Cover (octas)		
m/s	>60	30-60	<30	O'cast		0-3	4-7	8
≤1.5	A	A-B	B	C	D	F or G**	F	D
2.0-2.5	A-B	B	C	C	D	F	E	D
3.0-4.5	B	B-C	C	C	D	E	D	D
5.0-6.0	C	C-D	D	D	D	D	D	D
>6.0	D	D	D	D	D	D	D	D

** Category G is restricted to nighttime with less than 1 octa of cloud and a wind speed of less than 0.5 m/s.

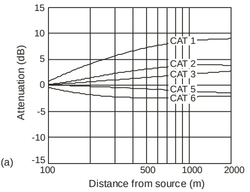
- attenuations for each category given in figures 5.24a-g in 3rd edition of text

ENC p255

Attenuation Cuves

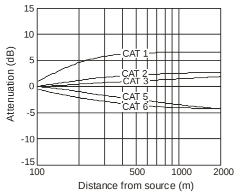
- See text book p256 for graphs

63 Hz



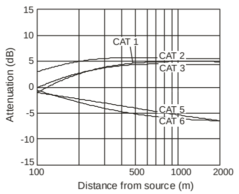
(a)

125 Hz



(b)

250 Hz



(c)

Meteorological Influences

- Variability in sound level predictions due to meteorological influences.

Octave band centre	Distance from source (m)			
frequency (Hz)	100	200	500	1,000
	(dB)			
63	±1	+4, -2	+7, -2	+8, -2
125	±1	+4, -2	+6, -4	+7, -4
250	+3, -1	+5, -3	+6, -5	+7, -6
500	+3, -1	+6, -3	+7, -5	+9, -7
1,000	+7, -1	+11, -3	+12, -5	+12, -5
2,000	+2, -3	+5, -4	+7, -5	+7, -5
4,000	+2, -1	+6, -4	+8, -6	+9, -7
8,000	+2, -1	+6, -4	+8, -6	+9, -7

ENC p258

Subjective Assessment of Change in

Remember this from Lecture on Fundamentals

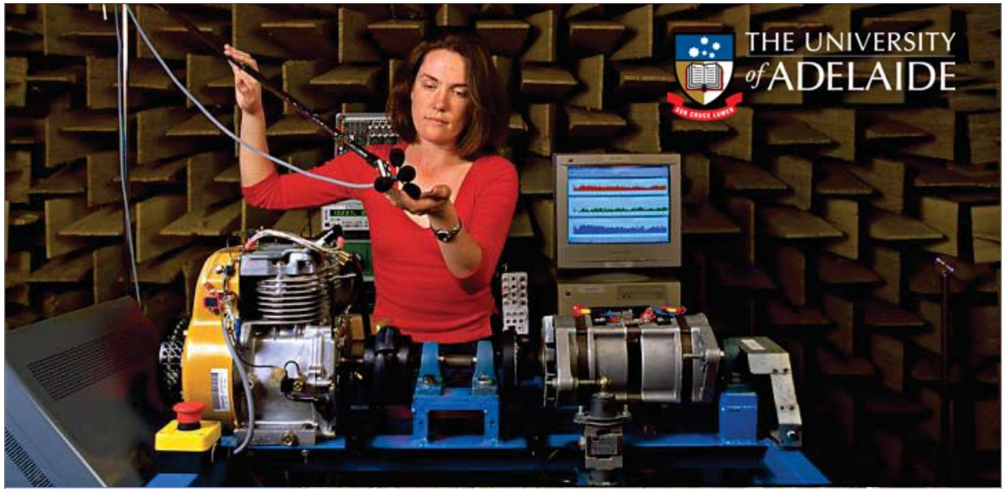
- To get a *clearly noticeable* change in SPL, it is necessary to reduce sound power level by 1/3.

Change in sound pressure level (dB)	Change in power Decrease	Change in power Increase	Change in apparent loudness
3	1/2	2	just perceptible
5	1/3	3	clearly noticeable
10	1/10	10	half or twice as loud
20	1/100	100	much quieter or louder

- Combined ray tracing
  - recent work in this area is focussed on the use of ray tracing of all reflected sound paths to determine the combined effect of ground, barriers, meteorological influences (turbulence, wind and temperature gradients) and air absorption. Too complex to be practical.
  - Note that ground and barrier effects are already combined in current models.

Accuracy?

- Can expect ± 3 dB for A-weighted levels if all the procedures discussed are followed. The most important effect is the meteorological effect - this causes ± 10 dB fluctuations in level at distances greater than 500 m from the noise source.



CRICOS PROVIDER 00123M

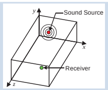
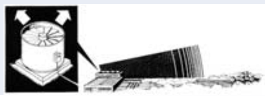
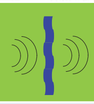
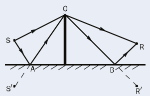
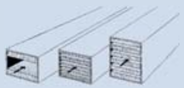
Carl Howard

Acoustic Enclosures and Transmission Loss of Panels

adelaide.edu.au

seek LIGHT

Course Content – Where Are We?

TOPIC	
Indoor	
Outdoor	
Acoustic Enclosures and Transmission Loss of Panels	
Barriers	
Mufflers	

Contents of Module

- General TL features and coincidence frequency
- How TL is measured
- STC and RW ratings
- Impact Isolation Class
- How to predict TL of panels
  - (isotropic, orthotropic, double wall)
- Composite TL
- Flanking TL
- Use TL to predict the NR of an enclosure
- Enclosure leakage causing reduction of TL

Learning Outcomes

Be able to:

- calculate sound transmission loss of single and double walls
- measure the sound transmission loss of a panel
- calculate STC and Rw rating of a panel
- design an acoustic enclosure

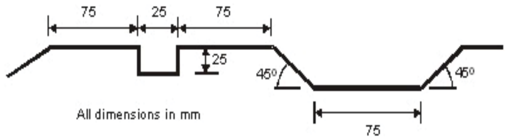
Typical Enclosures

Typical Enclosures

Personnel Enclosures: stop noise getting in

Machinery Enclosures: stop noise getting out

Question ...



Consider the steel panel with the cross-section shown in the figure, having E = 207 GPa and Poisson's ratio ν< = 0.3.

a) Calculate the bending stiffness in two directions: across the ribs and along the ribs. The panel thickness is 1.2mm.

b) Calculate the bending wave speed in both directions for the panel at a frequency of 1000Hz.

c) Calculate the range of critical frequencies for the panel.

d) If the panel is one wall of an enclosure and has dimensions 2m × 2m, calculate its lowest resonance frequency.

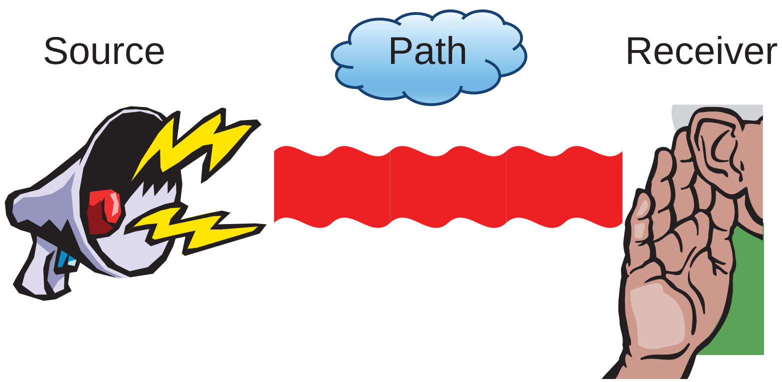
e) Calculate the transmission loss for the panel in octave bands from 63Hz to 8000Hz.

From Practice Problems 5. Q5.1

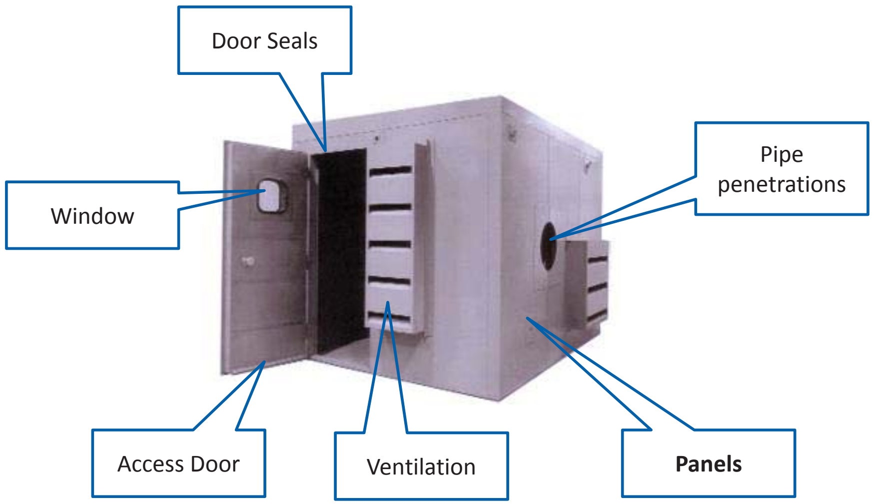
Noise Control Method

Remember this slide from the noise control section?

- Noise and vibration can be reduced at the



A Typical Enclosure

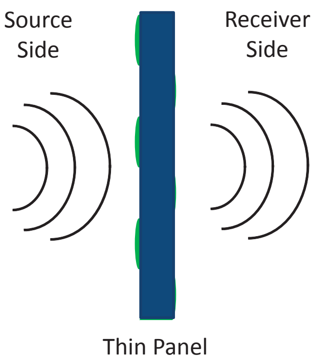


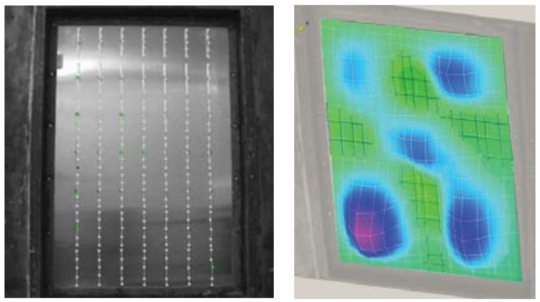
Focus on TL of panels first

Transmission Loss of a Panel

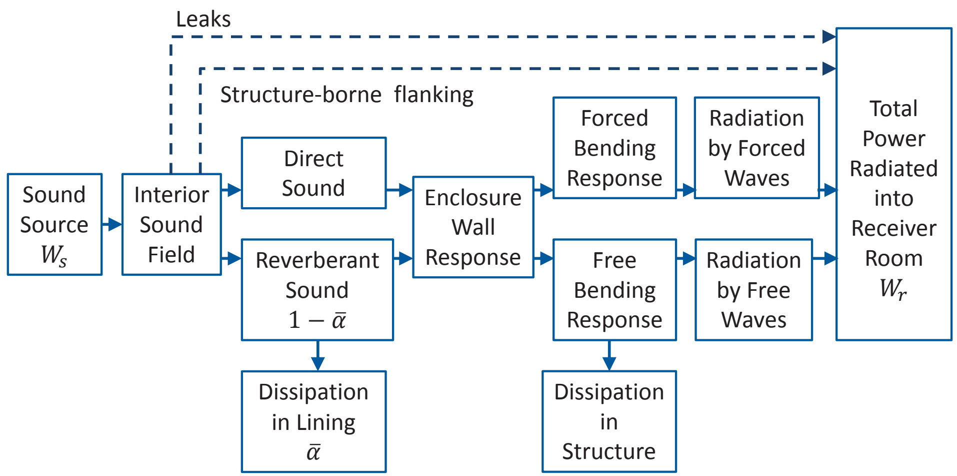
Remember this slide from the noise control section?

1. Incident sound strikes panel.
2. Panel vibrates.
3. Generates sound on opposite site.





Transmission Mechanisms of an Acoustic Enclosure

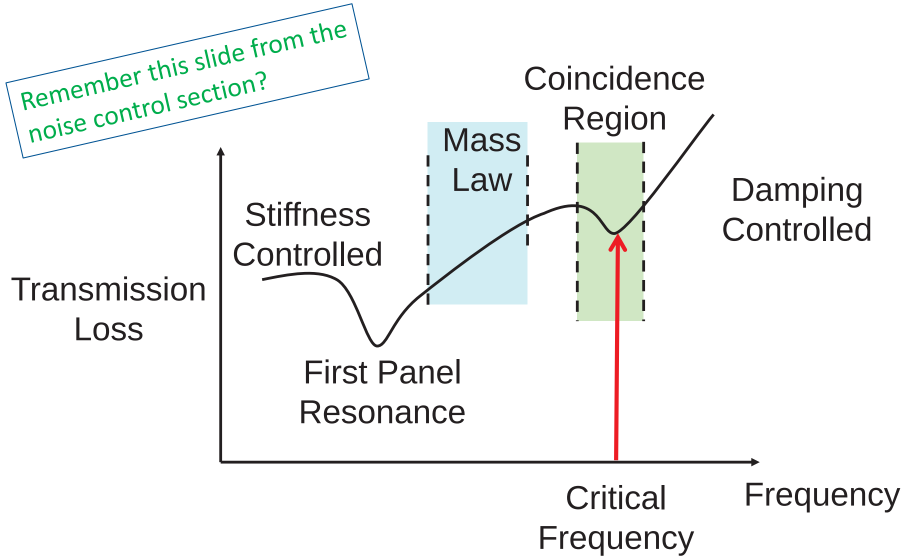


Beranek and Ver p544

How is TL Measured?

- Measure the difference in sound power levels in a source room and receiver room, where the rooms are separated by the test partition.

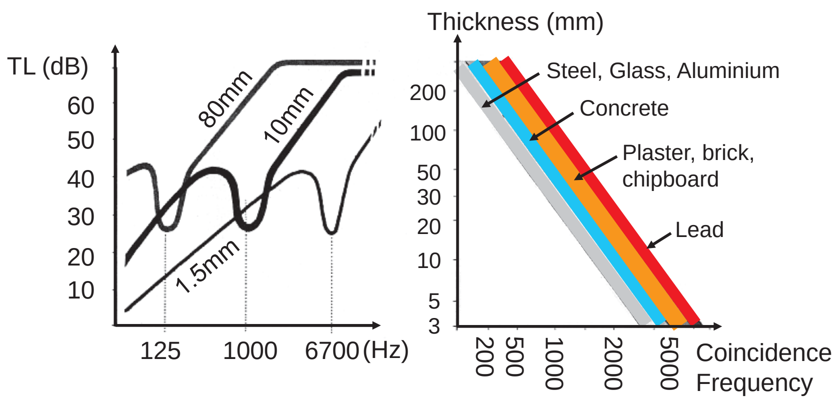
Typical Transmission Loss of a Panel



A single wall has poor insulation around a certain frequency

Remember this slide from the noise control section?

- Near the coincidence frequency, the transmission loss of a wall is reduced. At 1000Hz, a 1.5mm thick steel plate has better insulation than a 10mm thick plate.



Example: Research to Improve TL of a panel

- Used a Polytec laser vibrometer to measure the out-of-plane vibration of the plate.

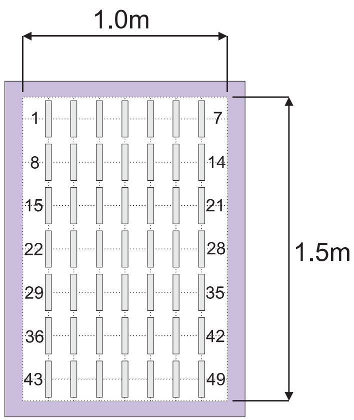
Experimental Setup

- Transmission loss measurements of a panel between two reverberation chambers.
  - 1.0 x 1.5m x 1.5mm
  - Aluminium
  - BCs somewhere between clamped and SS
- Three configurations of panels tested:
  - Bare panel
  - Panel with an array of cantilever resonators
  - Panel with array of rigid blocks (same mass as cantilevers)

Cantilevers and Rigid Blocks

Experimental Setup

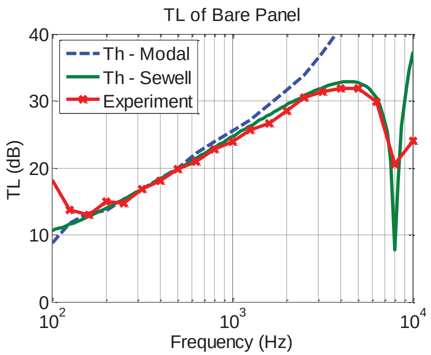
- 49 Devices were attached to the panel in a 7x7 grid pattern
  - Locations not optimised
  - Highest freq / lowest mass top left.
  - Lowest freq / highest mass bottom right.



Panel mass = 6.1kg

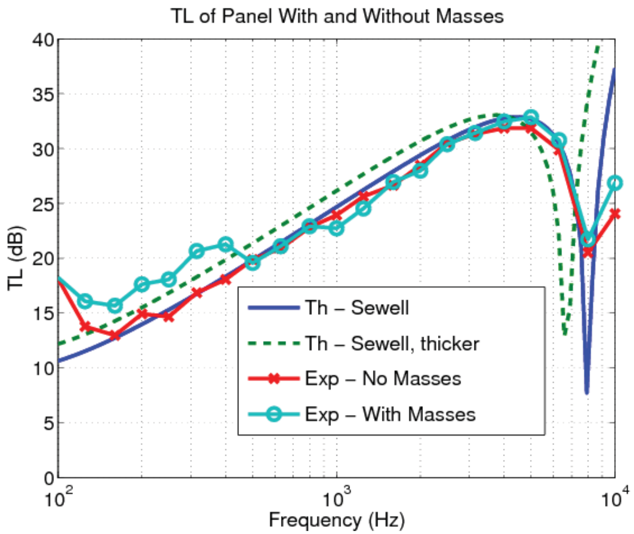
Results of Bare Panel

- Experimentally measured and theoretically predicted transmission loss of a plate without an array of rigid masses, using
  - Modal method
  - Sewell (1970)



Sewell (1970): See Fahy's book "Sound and Structural Vibration: Radiation, Transmission and Response"

Results w and w/o masses



Acoustic Enclosures

- See Chapter 8 , p353 in textbook
- form of *path* control
- sound is transmitted through wall by bending waves in the wall
  - excited by incident pressure field
  - propagate parallel to surface
  - characterised by displacements normal to surface
  - propagation speed, c_B frequency dependent

$$\boxed{c_b = \left[\frac{B\omega^2}{m}\right]^{1/4}}$$

c_b = bending speed of panel (m / s)
B = bending stiffness of panel (kg.m^2.s^{-2})
ω = circular frequency ($2\pi f$ in rad /s)
m = mass per unit area of panel (kg / m^2)

- sound transmission loss (TL) characterises ability of wall to insulate sound
  - important parameters used for calculating TL are
    - wall critical frequency, wall bending stiffness

ENC p354

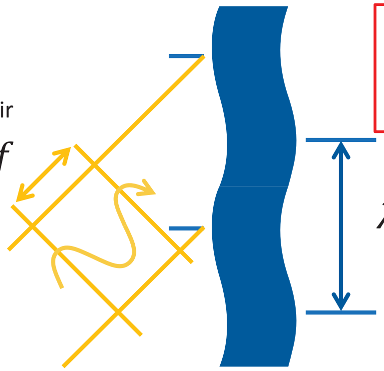
Coincidence Frequency

- The critical (coincidence) frequency is when the bending speed in panel matches speed of sound in surrounding medium
  - λ_{panel} bending wavelength = λ_{air} acoustic wavelength in air

$$\lambda = \lambda_B : c_B = c$$

Wavelength in Air

$$\lambda_{air} = c/f$$



$$\boxed{c_b = \left[\frac{B\omega^2}{m}\right]^{1/4}}$$

$$\lambda_{panel} = c_b/f$$

Bending Wavelength in Panel

Sound Transmission Loss

- wall critical frequency f_c (frequency of poor sound insulation)
  - bending wavelength = acoustic wavelength in air:

$$\lambda = \lambda_B : c_B = c$$

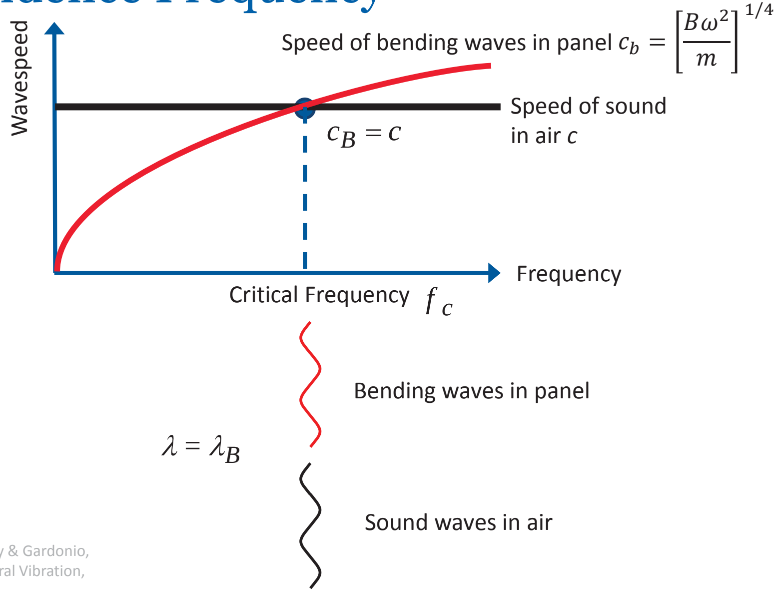
$$f_c = \frac{c^2}{2\pi}\sqrt{\frac{m}{B}} \approx \frac{0.55c^2}{c_L h}$$

- m = mass per unit area of panel (kg / m²)
- B = bending stiffness of panel (kg m² s⁻²)
- c_L = longitudinal wave speed in panel (Appendix 2, text) (m/s)

  divide tabulated values by $(1-v^2)^{1/2}$
  to be applicable to panels, v is Poisson's ratio)
- h = panel thickness (m)

ENC p355

Coincidence Frequency



Compare with Fahy & Gardonio, Sound and Structural Vibration, p289, Fig 5.4

Flat Panel

- Bending stiffness B
  - isotropic (flat) panels

$$\boxed{B = \frac{EI'}{\left(1-\nu^2\right)} = \frac{Eh^3}{12\left(1-\nu^2\right)}}$$

B = bending stiffness of panel (kg.m^2.s^{-2})
E = Young's modulus (Pa)
I' = second moment of area per unit width. (m^4)
h = plate thickness (m)
ν = Poisson's ratio (no units)

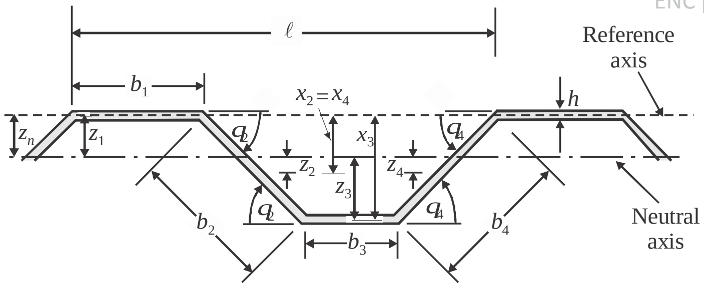
ENC p354

Orthotropic Panel

- Bending stiffness B
  - corrugated panels (orthotropic)
    - characterised by a range of bending stiffnesses depending on direction of bending wave travel in panel
    - results in a range of critical frequencies. Thus good sound transmission over wide frequency range. This is undesirable for noise control.

ENC p358

A typical cross-section of a ribbed panel.



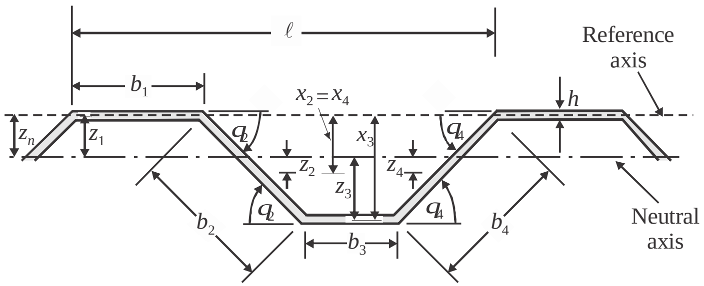
Orthotropic Panel

- Bending stiffness B
  - corrugated panels (orthotropic)

$$B = \frac{Eh}{\left(1-\nu^2\right)\ell}\sum_n b_n\left(z_n^2 + \frac{h^2+b_n^2}{24} + \frac{h^2-b_n^2}{24}\cos 2\theta_n\right) \quad (\mathrm{kg\ \ m^2 s^{-2}})$$

  - must satisfy $\lambda_B >> b_n$

A typical cross-section of a ribbed panel.



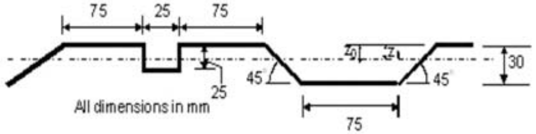
Orthotropic Panel

Neutral axis: $\sum b_n z_n = 0$

or if we choose an arbitrary reference axis, then the distance from the reference axis to the neutral axis, z_n, is:

$$z_n = \frac{\sum_{i=1}^{N} x_i b_i h_i}{\sum_{i=1}^{N} b_i h_i}$$

Answer …



a) Calculate the bending stiffness in two directions: across the ribs and along the ribs. The panel thickness is 1.2mm.

First find location of neutral axis by taking moments about an axis through the centre of the angled section and shown as zo in the figure. In the following equations bi is the length of the ith section. If the neutral axis is denoted as zn where z is the vertical coordinate on the figure, then:

$$z_0 - z_n = \frac{\sum_i b_i z_{i0}}{\sum_i b_i}$$

$$= \frac{75 \times 15 + 2 \times 25 \times 2.5 - 25 \times 10 + 75 \times 15 - 75 \times 15}{75 + 3 \times 25 + 75 + 2\sqrt{2} \times 30 + 75}$$

$$= \frac{1000}{384.9} = 2.6\text{mm}$$

Answer …

Thus the neutral axis is (30/2 - 2.6=) 12.4mm from the centre of the top of the section.

The section thickness, h = 1.2mm and the horizontal length, ℓ, before repeating itself is 250 + 60mm = 0.31m.

The bending stiffness in the direction along the ribs may be calculated with E = 207GPa and ν= 0.3 using

$$B = \frac{Eh}{(1-\nu^2)\ell}\sum_n b_n\left(z_n^2 + \frac{h^2 + b_n^2}{24} + \frac{h^2 - b_n^2}{24}\cos 2\theta_n\right) \quad (\text{kg m}^2\text{s}^{-2})$$

Answer …

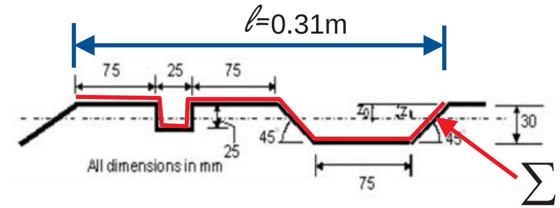
$$B_1 = \frac{207 \times 10^9 \times 0.0012}{0.91 \times 0.31}\left[0.15\left(0.0124^2 + \frac{0.0012^2}{12}\right)\right.$$
$$+ 0.05\left(0.0001^2 + \frac{0.025^2}{12}\right) + 0.025\left(0.0126^2 + \frac{0.0012^2}{12}\right)$$
$$+ 0.075\left(0.0176^2 + \frac{0.0012^2}{12}\right)$$
$$\left.+ 0.06\sqrt{2}\left(0.0026^2 + \frac{0.0012^2 + 2 \times 0.03^2}{24}\right)\right]$$
$$= 8.805 \times 10^8 (2.3082 \times 10^{-5} + 2.6046 \times 10^{-6} + 3.9720 \times 10^{-6}$$
$$+ 2.3241 \times 10^{-5} + 6.9427 \times 10^{-6})$$
$$= 5.27 \times 10^4 \text{ kg m}^2 \text{ s}^{-2}$$

Answer ...

The bending stiffness per unit width in the direction across the ribs may be calculated using

$$B = \frac{Eh^3}{12\left(1-\nu^2\right)l} \times \sum_n b_n$$

$$B_2 = \frac{207 \times 10^9 \times 0.0012^3}{12 \times 0.91} \times \frac{0.385}{0.31} = 40.7\,\mathrm{kgm^2s^{-2}}$$



$$\sum_n b_n = 0.385\,\mathrm{m}$$

Answer ...

b) Calculate the bending wave speed in both directions for the panel at a frequency of 1000Hz.

The bending wave speed is $c_b = \left[\frac{B\omega^2}{m}\right]^{1/4}$

The surface mass of the panel is
m = 7800 × 0.0012 × 0.385/0.31 = 11.62 kg m^{-2}
and the frequency is 1000 Hz.

Thus the lower and upper bending wave speeds corresponding to waves propagating parallel and perpendicular to the ribs respectively are:

$$c_{B1} = \left(\frac{5.27 \times 10^4 \times 4\pi^2 \times 10^6}{11.62}\right)^{1/4} = 650\,\mathrm{m/s}$$

$$c_{B2} = \left(\frac{40.7 \times 4\pi^2 \times 10^6}{11.62}\right)^{1/4} = 108\,\mathrm{m/s}$$

Answer ...

c) Calculate the range of critical frequencies for the panel.

$$f_c = \frac{c^2}{2\pi}\sqrt{\frac{m}{B}}$$

$$f_{c1} = \frac{343^2}{2\pi}\left(\frac{11.62}{5.27 \times 10^4}\right)^{1/2} = 278\,\mathrm{Hz}$$

$$f_{c2} = \frac{343^2}{2\pi}\left(\frac{11.62}{40.7}\right)^{1/2} = 10{,}000\,\mathrm{Hz}$$

Transmission Coefficient

Transmission Coefficient, $\tau = \dfrac{\text{transmitted energy}}{\text{incident energy}}$ ENC p359

- Transmission loss TL (in dB) is $\boxed{TL = -10\log_{10}\tau}$
  - i.e., BIG value of TL $\equiv$ small value of τ
- Several definitions for TL
  - Normal incidence TL_N
  - Field Incidence $TL_F \equiv TL$
  - Random Incidence TL_D
  - Arbitrary Incidence Angle θ, TL_θ

How is TL measured?

The simple explanation

- Measure the SPL in the
  – source room L_{p1} and
  – receiver room L_{p2}

- Calculate the Noise Reduction NR as

$$\mathrm{NR} = L_{p1} - L_{p2} = 10\log_{10}\left[\frac{\langle p_1\rangle^2}{\langle p_2\rangle^2}\right]$$

How is TL measured?

The simple explanation

- Transmission Loss TL is

$$\boxed{\mathrm{TL} = \mathrm{NR} + 10\log_{10}\left[\frac{A\,(1-\bar{\alpha})}{S\,\bar{\alpha}}\right]}$$

ENC p360

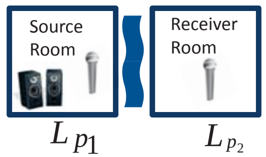
where
NR is the noise reduction
A is the area of the test panel
S is the surface area of the room
$\bar{\alpha}$ is the average receiver room absorption coefficient

Also see Australian Standard AS 1191-2002 Acoustics – Method for laboratory measurement of airborne sound insulation of building elements

Measurement of TL (1/1 or 1/3 octaves)

The derivation

- panel placed in cavity between two reverberation chambers

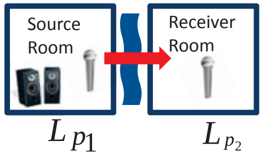


- sound introduced in source room L_{p1} measured in receiver room L_{p2}
  NR (Noise Reduction) = $L_{p_1} - L_{p_2}$ (frequency dependent)

The derivation

Measurement of TL (1/1 or 1/3 octaves)

The power transmitted by panel



$$W_t = \underbrace{\left[\frac{\langle p_i^2 \rangle A}{4\rho c}\right]}\times \tau$$

$\langle p_i^2 \rangle$ = Av. Incident pressure squared (Pa^2)
A = Area of the panel (m^2)
ρ = density air (kg/m^3)
c = speed of sound (m/s)
τ = Transmission coefficient (no units)

Fraction of power that is transmitted through the panel into the receiver room

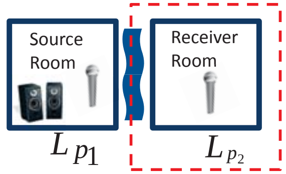
Sound power incident on panel = Diffuse field sound intensity X Area of panel

.... Now that sound power is in receiver room, calculate the sound pressure level

The derivation

Measurement of TL (1/1 or 1/3 octaves)

.... Now that sound power is in receiver room, calculate the sound pressure level.



- From the Indoor sound module, the reverberant field is due to the sound power of the "source of noise" (the sound transmitted through the panel) W_t, MINUS the sound power absorbed by the walls.

$$\langle p_r^2 \rangle = \frac{4W_t \rho c(1-\bar{\alpha})}{S\bar{\alpha}} = \left\{\left[\frac{\langle p_i^2 \rangle A}{4\rho c}\right]\times\tau\right\}\frac{4\rho c(1-\bar{\alpha})}{S\bar{\alpha}} = \frac{\langle p_i^2 \rangle A\tau(1-\bar{\alpha})}{S\bar{\alpha}}$$

The derivation

Measurement of TL (1/1 or 1/3 octaves)

$$NR \text{ (Noise Reduction)} = L_{p1} - L_{p2} = 10\log_{10}\left[\frac{\langle p_i^2\rangle}{\langle p_r^2\rangle}\right]$$

$$NR = 10\log_{10}\left[\frac{\langle p_i^2\rangle}{\left(\frac{\langle p_i^2\rangle A\,\tau\,(1-\bar{\alpha})}{S\bar{\alpha}}\right)}\right]$$

$$= 10\log_{10}\left[\frac{1}{\left(\frac{A\ (1-\bar{\alpha})}{S\bar{\alpha}}\right)}\right] + 10\log_{10}\left[\frac{1}{\tau}\right]$$

$$= -10\log_{10}\left[\frac{A\ (1-\bar{\alpha})}{S\bar{\alpha}}\right] - 10\log_{10}[\,\tau\,]$$

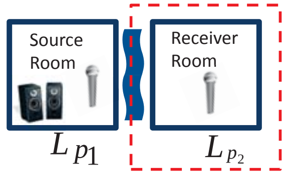
$$NR = -10\log_{10}\left[\frac{A\ (1-\bar{\alpha})}{S\bar{\alpha}}\right] + TL$$

$$\boxed{TL = NR + 10\log_{10}\left[\frac{A\ (1-\bar{\alpha})}{S\bar{\alpha}}\right]}$$

The derivation

Measurement of TL

- For testing in acoustic reverberation chambers, the absorption coeff. of the walls is small i.e. $\bar{\alpha} \approx 0$
- Therefore $(1-\bar{\alpha}) \approx 1.0$



- So we can approximate as ...

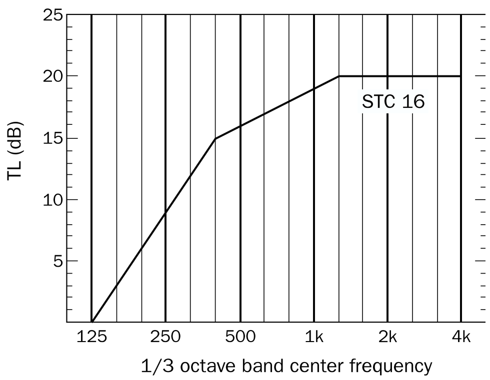
$$TL = NR + 10\log_{10}\left[\frac{A\ (1-\bar{\alpha})}{S\bar{\alpha}}\right]$$

$$\boxed{TL = NR + 10\log_{10}\left[\frac{A}{S\bar{\alpha}}\right]}$$

TL = Transmission Loss (dB)
NR = Noise Reduction (dB)
A = area of panel (m^2)
S = surface area of receiver room (m^2)
$\bar{\alpha}$ = average receiver room absorption coefficient (no units)

Sound Transmission Class (STC)

- single number descriptor of transmission loss
- horizontal segment from 1250 Hz to 4000 Hz
- middle segment increasing by 5dB from 400 Hz to 1250 Hz
- low frequency segment increasing by 15 dB from 125 Hz to 400 Hz
- STC rating determined by plotting 1/3 octave band *TL* vs frequency
- STC contour shifted vertically in 1 dB increments until
  - TL curve never more than 8dB below STC contour
  - sum of deficiencies in *TL* below STC contour over the 16 1/3-octave bands does not exceed 32 dB
- STC = value of contour at 500 Hz.



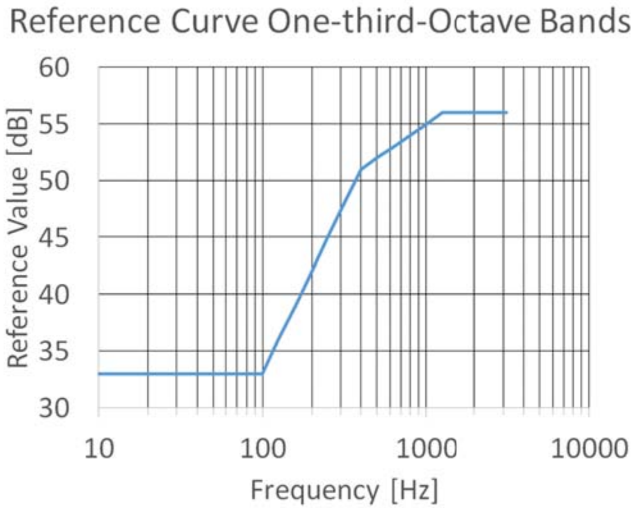
ENC p361

Weighted Sound Reduction Index, Rw

- ISO uses Weighted Sound Reduction Index, R_w instead of STC
  - ISO uses correction terms to account for different types of incident sound (text book, pp. 362, 363).
  - the 8 dB criterion (see previous page) does not apply
  - Measurements are rounded to nearest 0.1 dB when calculating deficiencies compared to 1 dB for STC
  - standard frequency range for Rw classification is 100 Hz to 3150 Hz (16 – 1/3 octave bands), compared to 125 Hz to 4000 Hz (16 – 1/3 octave bands) for STC.

See Australian Standard AS/NZS ISO 717.1:2004, (or ISO 717-1:1996) Acoustics—Rating of sound insulation in buildings and of building elements. Part 1: Airborne sound insulation

Weighted Sound Reduction Index, Rw

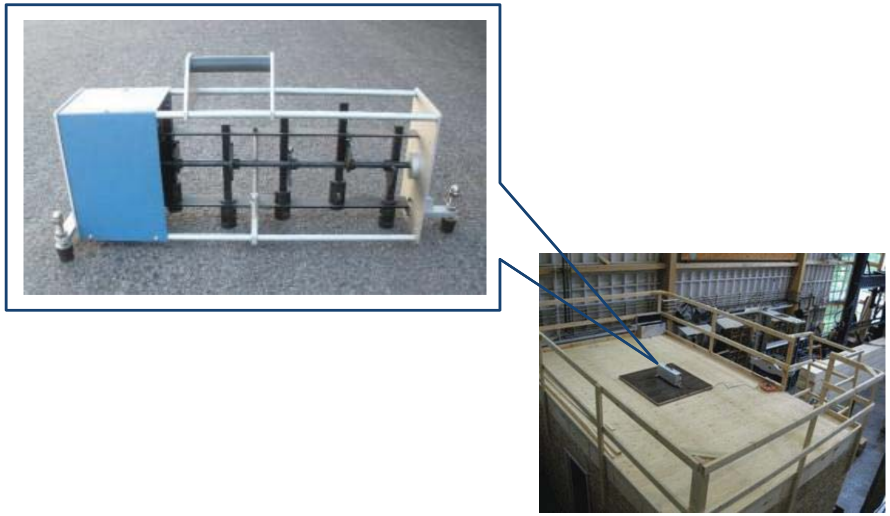


AS/NZS ISO 717.1:2004, Figure 1 – Curve of reference values for airborne sound, one-third octave bands

Impact Isolation

- Measure of the ability of a construction such as a floor or ceiling to prevent transmission of impact noise such as foot steps
  - measured using a standard tapping machine

Lab Testing Using a Tapping Machine

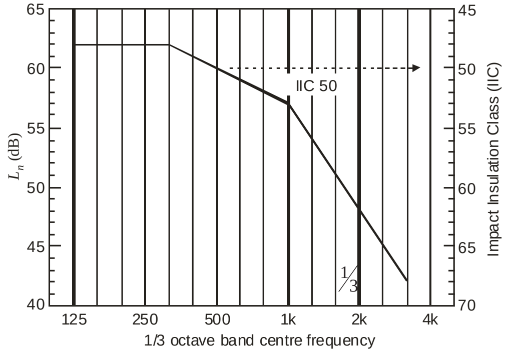


Impact Isolation

- Normalised impact sound level for each 1/3 octave measurement band is:

  $$L_n = L_p + 10\log_{10}\left(S\bar{\alpha}/10\right)$$

- Impact Insulation Class calculated from L_n
  - IIC contour shifted vertically until
    - L_n curve never more than 8dB above IIC contour
    - sum of deficiencies in L_n above IIC contour over the 16 octave bands does not exceed 32 dB
  - IIC = value of contour at 500 Hz.



ENC p364

Impact Isolation

- ISO has a similar procedure
  - terminology is $L_{n,w}$ instead of IIC
  - an additional quantity $L_{nT,w}$ is also defined and based on the measurements of T_{60} in the receiving room instead of $S\bar{\alpha}$ (see text p365)
  - 8 dB criterion is ignored in both cases
  - dB measurements are rounded to the nearest 0.1 dB when calculating deficiencies instead of 1 dB used for IIC

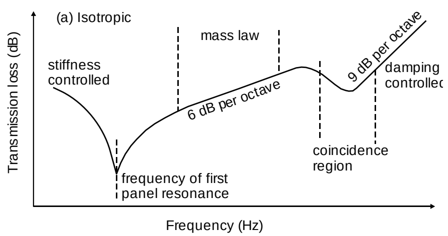
Sound Transmission Loss Estimation

- So far we've seen:
  - the general trends in sound transmission loss,
  - how to calculate some important frequencies,
  - how to measure the TL,
  - how to categorise the TL of a panel using standard curves.

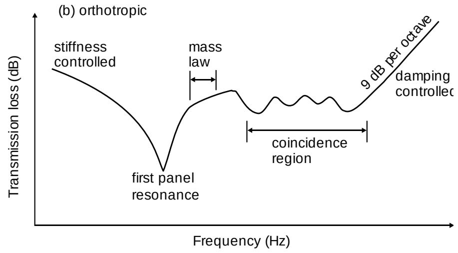
- Now we will present methods to estimate the transmission loss for:
  a) Isotropic panels – flat panels
  b) Orthotropic – corrugated panels

Sound Transmission Loss Estimation

a) Isotropic



b) Orthotropic



Method involves construction of graphs with key points (x,y == TL vs freq) for:

- First panel resonance
- Mass law range
- Coincidence region
- Damping controlled region

Panel Resonance Frequencies

simply supported isotropic rectangular panel

$$f_{i,n} = \frac{\pi}{2}\sqrt{\frac{B}{m}}\left[\frac{i^2}{a^2} + \frac{n^2}{b^2}\right] \quad \text{(Hz)}$$

$$i,\, n = 1,\, 2,\, 3, \ldots$$

or

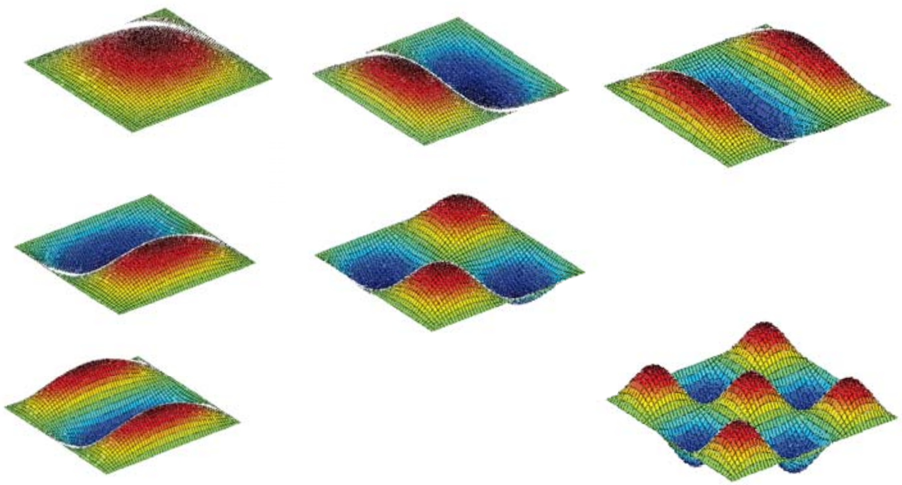
$$f_{i,n} = 0.453 c_L h \left[\frac{i^2}{a^2} + \frac{n^2}{b^2}\right]$$

$$B = \frac{EI'}{(1-\nu^2)} = \frac{Eh^3}{12(1-\nu^2)}$$

$f_{i,n}$ = resonance frequency of panel for mode i,n (Hz)
B = bending stiffness of panel ($kg.m^2.s^{-2}$)
m = mass per unit area of panel ($kg.m^{-2}$)
i, n = modal indices (1,2,3,... N)
a, b = side lengths of panel (m)
E = Young's modulus (Pa)
I' = second moment of area per unit width. (m^4)
h = plate thickness (m)
ν = Poisson's ratio (no units)

ENC p367

Mode shapes of a plate



Panel Resonance Frequencies

Process of how to calculate panel resonance frequencies

- Write out a table of i,n from 1 to N and calculate $f_{i,n}$

i	n	$f_{i,n}$ (Hz)
1	1	
2	1	
3	1	
...N	1	
1	2	
2	2	
3	2	
...N	2	

$$f_{i,n} = \frac{\pi}{2}\sqrt{\frac{B}{m}}\left[\frac{i^2}{a^2} + \frac{n^2}{b^2}\right] \quad \text{(Hz)}$$

$$B = \frac{EI'}{(1-\nu^2)} = \frac{Eh^3}{12(1-\nu^2)}$$

B = bending stiffness of panel ($kg.m^2.s^{-2}$)
E = Young's modulus (Pa)
I' = second moment of area per unit width. (m^4)
h = plate thickness (m)
ν = Poisson's ratio (no units)

- Sort them into increasing frequency

ENC p367

Panel Resonance Frequencies

simply supported orthotropic panel

$$f_{i,n} = \frac{\pi}{2\, m^{1/2}}\left(\frac{B_a i^4}{a^4} + \frac{B_b n^4}{b^4} + \frac{B_{ab} i^2 n^2}{a^2 b^2}\right)^{1/2} \quad \text{where} \quad i, n = 1, 2, 3, \ldots$$

$$B_{ab} = 0.5\left(B_a \nu + B_b \nu + G h^3 / 3\right)$$

$f_{i,n}$ = resonance frequency of panel for mode i,n (Hz)
B_{ab} = bending stiffness of panel ($kg.m^2.s^{-2}$)
m = mass per unit area of panel ($kg.m^{-2}$)
i, n = modal indices (1,2,3,... N)
a, b = side lengths of panel (m)
E = Young's modulus (Pa)
I' = second moment of area per unit width. (m^4)
h = plate thickness (m)
ν = Poisson's ratio (no units)

ENC p367

Answer ...

d) If the panel is one wall of an enclosure and has dimensions 2m × 2m, calculate its lowest resonance frequency.

Assuming that the enclosure wall edge condition is simply supported (a good approximation in practice for most enclosures), the first resonance frequency of the panel may be calculated using

$$B_{ab} = 0.5\left(B_a \nu + B_b \nu + \frac{G h^3}{3}\right)$$

$$f_{i,n} = \frac{\pi}{2m^{1/2}}\left[\frac{B_a i^4}{a^4} + \frac{B_b n^4}{b^4} + \frac{B_{ab} i^2 n^2}{a^2 b^2}\right]^{1/2}$$

with $i = n = 1$.

Answer ...

Thus ...

$$B_{ab} = 0.5\left(5.27 \times 10^4 \times 0.3 + 40.7 \times 0.3 + \frac{207 \times 10^9 \times 0.0012^3}{3 \times 2.6}\right)$$

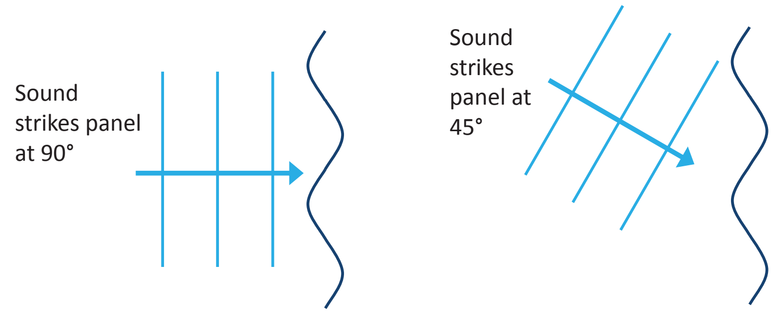
$$= 7934$$

$$f_{1,1} = \frac{\pi}{2\sqrt{11.62}}\left(\frac{5.27 \times 10^4}{2^4} + \frac{40.7}{2^4} + \frac{7934}{2^4}\right)^{1/2}$$

$$= 28.4\,\text{Hz}$$

Transmission Coefficient

- The transmission coefficient depends on the angle of incidence of the incoming sound wave.



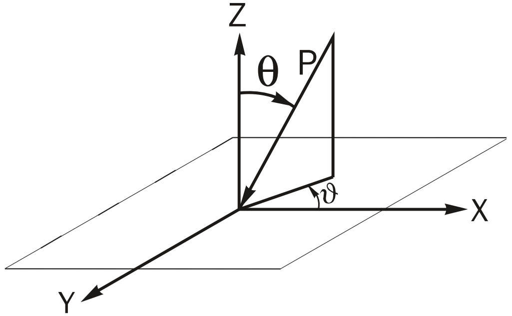
- For any angle of incidence, θ $$\tau(\theta) = \left|1 + \frac{Z\cos\theta}{2\rho c}\right|^{-2}$$

ENC p368

Transmission Coefficient

- for an *infinite* isotropic panel

$$Z = \mathrm{j}2\pi\, f\, m \left[1 - \left(\frac{f}{f_c} \right)^2 (1 + \mathrm{j}\eta) \sin^4 \theta \right]$$

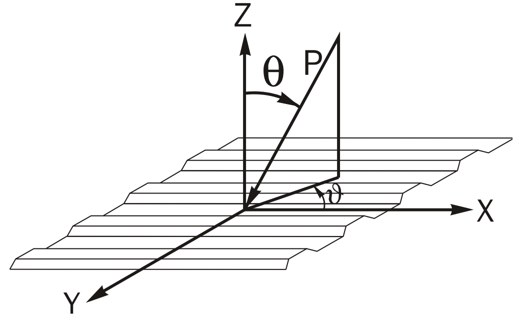


ENC p367

Transmission Coefficient

- for an *infinite* orthotropic panel

$$Z = \mathrm{j}2\pi\, f\, m \left[1 - \left[\frac{f}{f_{c_1}} \cos^2 \vartheta + \frac{f}{f_{c_2}} \sin^2 \vartheta \right]^2 \times (1 + \mathrm{j}\eta) \sin^4 \theta \right]$$



ENC p368

Transmission Coefficient

- Once we have the transmission coefficient for a particular angle, we need to calculate the overall transmission coefficient over ALL angles..... Integrate over all angles
- diffuse field transmission coefficient

$$\tau_d = \frac{1}{\pi} \int_0^{2\pi} d\vartheta \int_0^{\pi/2} \tau(\theta,\ \vartheta) \cos\theta\ \sin\theta\, d\theta$$

- for isotropic panels $\tau_d = \int_0^1 \tau(\theta) \mathrm{d}\left(\sin^2 \theta\right)$
- for orthotropic panels $\tau_d = \frac{2}{\pi} \int_0^{\pi/2} d\vartheta \int_0^1 \tau(\theta, \vartheta)\ d\left(\sin^2 \theta\right)$

ENC p369

Field Transmission Coefficient, τ_F (for panels of finite size)

- isotropic panels $\tau_F = \int_0^{\sin^2\theta_L} \tau(\theta) \mathrm{d}\left(\sin^2 \theta\right)$
- orthotropic panels $\tau_F = \frac{2}{\pi} \int_0^{\pi/2} \mathrm{d}\vartheta \int_0^{\sin^2\theta_L} \tau(\theta, \vartheta) \mathrm{d}\left(\sin^2 \theta\right)$
- limiting angle $\theta_L = \cos^{-1} \sqrt{\frac{\lambda}{2\pi\sqrt{A}}}$

ENC p369

Mass Law Transmission Loss

- for $f < f_c/2$ or $f_{c1}/2$ (mass law) $\quad Z = j2\pi f m$

$$TL_\theta = 10\log_{10}\left[1 + \left(\frac{\pi f m}{\rho c}\cos\theta\right)^2\right]$$

  – to obtain TL_N, set $\theta = 0$

- field incidence $TL = TL_N$ - 5.5 (dB); thus,

$$TL = 10\log_{10}\left[1 + (\pi f m/(\rho c))^2\right] - 5.5 \quad \text{(dB)}$$

  – assumes 1/3 octave bandwidth (to agree with measurements)
  – valid for both isotropic and orthotropic
- high frequency transmission loss $f > f_c$ (isotropic panel only)

$$TL = 20\log_{10}[\pi f m/(\rho c)] + 10\log_{10}[2\eta f/(\pi f_c)] \text{ (dB)}$$

ENC p370

Stiffness Controlled Region

- Low frequency transmission loss (stiffness controlled region)
  – below half the first resonance freq

$$TL = 20\log_{10}\left[\pi^4 B\left(\frac{1}{a^2} + \frac{1}{b^2}\right)^2\right] - 20\log_{10} f - 20\log_{10}(4\pi\rho c)$$

$$= 20\log_{10} B - 20\log_{10} f + 20\log_{10}\left(\frac{1}{a^2} + \frac{1}{b^2}\right)^2 - 20\log_{10}(\rho c) + 17.8 \text{ (dB)}$$

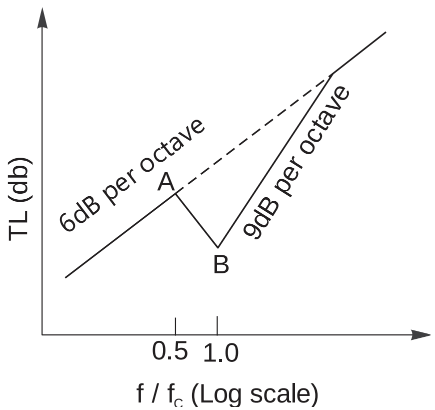
  – around the panel first resonance frequency ($0.5 f_{1,1}$ to $1.5 f_{1,1}$), if $\eta >> \rho c / 2\pi fm$

$$TL = 20\log_{10} f_{1,1} + 20\log_{10} m + 20\log_{10}\eta - 20\log_{10}(\rho c/\pi) \text{ (dB)}$$

otherwise $TL = 0$

For very thick panels the TL is limited to a maximum value
- see text, page 374

Isotropic Panel Transmission Loss



Point A: (f/f_c = 0.5)

$$TL = 20\log_{10} f_c m - 54 \quad \text{(dB)}$$

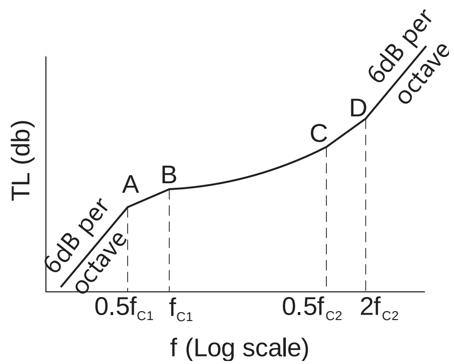
Point B: (f/f_c = 1.0)

$$TL = 20\log_{10} f_c m + 10\log\eta - 45 \text{ (dB)}$$

A more detailed, though not necessarily more accurate TL prediction scheme is offered by Davy (see text book)

See p372 in textbook

Orthotropic Panel (Damped and Undamped)



Point A:

$$TL = 20\log_{10} f_{c_1} m - 54 \quad \text{(dB)}$$

Point B:

$$TL = 10\log_{10} f_{c_1} m - 16 \quad \text{(dB)}$$

Between B and C:

$$TL = 20\log_{10} f + 10\log_{10} m - 10\log_{10} f_{c_1} - 20\log_{10}\left[\log_e(4f/f_{c_1})\right] - 13.2 \text{ dB}$$

Point D:

$$TL = 15\log_{10} f_{c_2} + 10\log_{10} m - 5\log_{10} f_{c_1} - 17 \quad \text{(dB)}$$

Orthotropic Panel *TL*

- Prediction scheme underestimates *TL* below $0.7\,f_{c_1}$
  - error gets larger as panel size gets smaller.

- Common panels exhibit dip in *TL* up to 5dB between 2000 Hz and 4000 Hz at frequency

$$f_r = \frac{c}{2L}$$

corresponding to air resonance between corrugations

L = average corrugation cavity width.

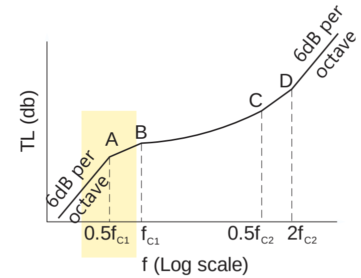
Answer ...

e) Calculate the transmission loss for the panel in octave bands from 63Hz to 8000Hz.

The sound transmission loss of the panel may be calculated using figure 8.8b in the text. Point A is at 139Hz and the corresponding TL is given by:

$$TL_A = 20\log_{10}(278 \times 11.62) - 54 = 16.2\,\text{dB}$$

A= 139 Hz, 16.2dB

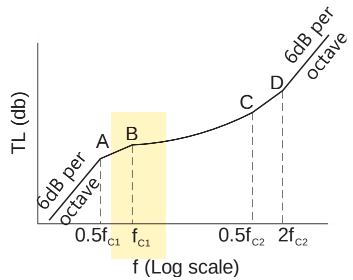


Answer ...

At point B (278Hz), the TL is:

$$\begin{aligned} TL_B &= 20\log_{10}(278) + 10\log_{10}(11.62) - 10\log_{10}(278) \\ &\quad - 20\log_{10}[\log_e(4)] - 13.2 \\ &= 19.1\,\text{dB} \end{aligned}$$

B = 278 Hz, 19.1 dB

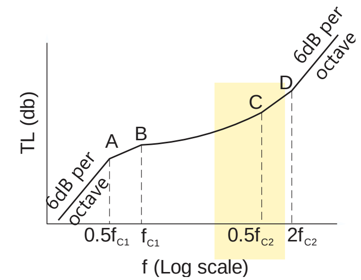


Answer ...

At point C (5,000Hz), the TL is:

$$\begin{aligned} TL_C &= 20\log_{10}(5000) + 10\log_{10}(11.62) - 10\log_{10}(278) \\ &\quad - 20\log_{10}[\log_e(20000/278)] - 13.2 \\ &= 34.4\,\text{dB} \end{aligned}$$

C = 5,000 Hz, 34.4 dB



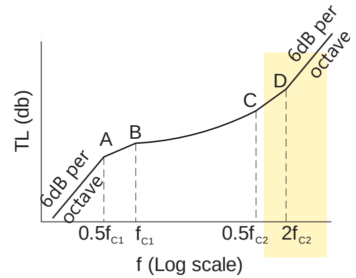
Answer …

At point D (20,000Hz), the TL is:

$$TL_D = 10\log_{10}(11.62) + 15\log_{10}(10000) - 5\log_{10}(278) - 17$$

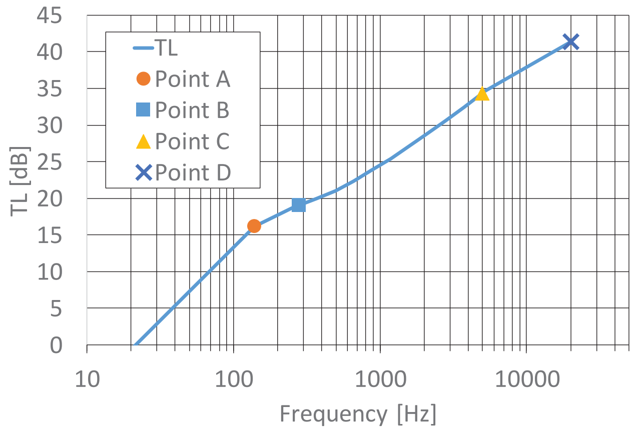
$$= 41.4\,\text{dB}$$

D = 20,000 Hz, 41.4 dB



Answer …

- These points are plotted on the following graph and interpolation is used to find the octave band TL values.



Answer …

- Strictly speaking, the curve should only be used to find 1/3 octave values and the octave band levels must then be calculated from the following equation:

$$\boxed{\mathrm{TL}_{\mathrm{oct}} = -10\log_{10}\left\{\frac{1}{3}\times\left[10^{-TL_1/10} + 10^{-TL_2/10} + 10^{-TL_3/10}\right]\right\}}$$

- However, for most practical purposes, the results obtained that way are little different to the results obtained by reading the octave band levels directly from the figure. However, in the case of isotropic panels, care should be taken to avoid errors near the dip in the curve corresponding to the critical frequency. Following the figure, the octave band results are summarised in a table.

Answer …

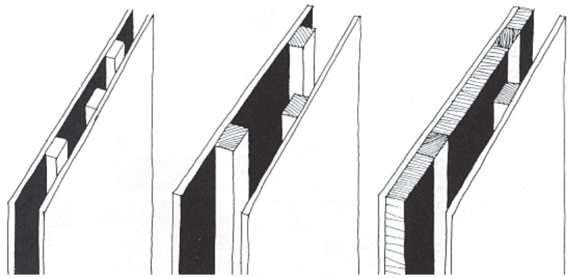
Octave band centre frequency (Hz)	Transmission Loss (dB)
63	9
125	15
250	19
500	21
1000	25
2000	29
4000	33
8000	37

Panel Loss Factor η

- see Appendix B for values
- very dependent on panel boundary conditions
- can be calculated from time for panel vibrations to decay by 60 dB after cessation of excitation (T_{60})

$$\eta = \frac{2.2}{f\,T_{60}}$$

where f is the test frequency (Hz).

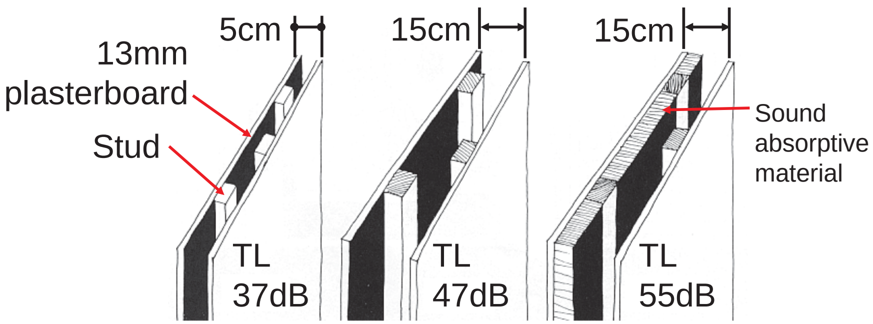


Double Walls

Light double walls provide good sound insulation

Remember this slide from the noise control section?

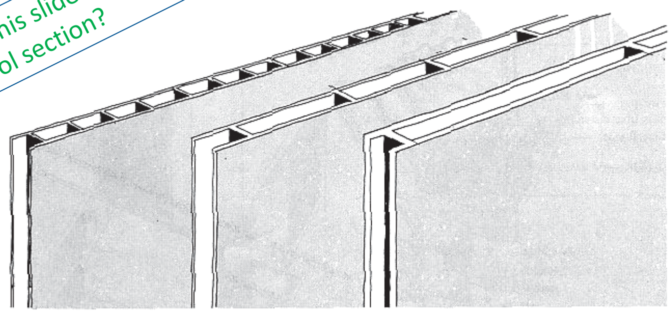
- Two lightweight walls separated by an air gap provide good transmission loss. The TL increases as the spacing increases. Double walls can provide the same TL as a single wall that is 5x to 10x as heavy.



Double walls should have few connections

- A double wall provides the best TL if each wall is heavy and has few connections between them.

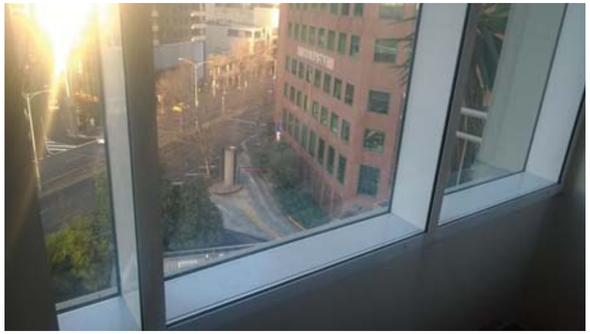
Remember this slide from the noise control section?



Double Walls

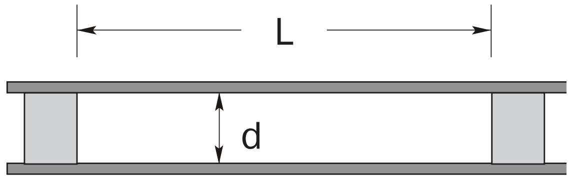
- Double glazed windows are another example.

Photo taken in apartment in Melbourne, Southbank.



Double Wall TL

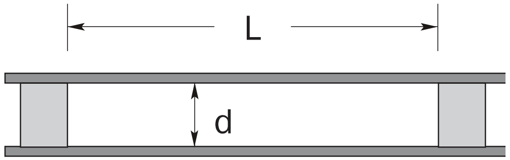
- Double wall construction is perhaps the most common building construction.
- Top down view



- Surprisingly, prediction methods are still being researched - Very complex problem.

ENC p376

Double Wall TL



- Lowest Order Acoustic Resonance $f_2 = c/2L$
- Cavity/Structural Resonance (mass-air-mass)

$$f_0 = \frac{1}{2\pi}\left(\frac{1.8\rho c^2(m_1+m_2)}{d\,m_1 m_2}\right)^{1/2} = 80\sqrt{\frac{m_1+m_2}{d\,m_1 m_2}} \quad \text{(Hz)}$$

m_1,m_2 = surface densities in kg / m^2
d = gap width in m
L = longest cavity dimension in m

- Limiting Frequency $f_\ell = c/2\pi d$

ENC p377

Double Wall TL

- Two panels completely isolated - mechanically & acoustically;

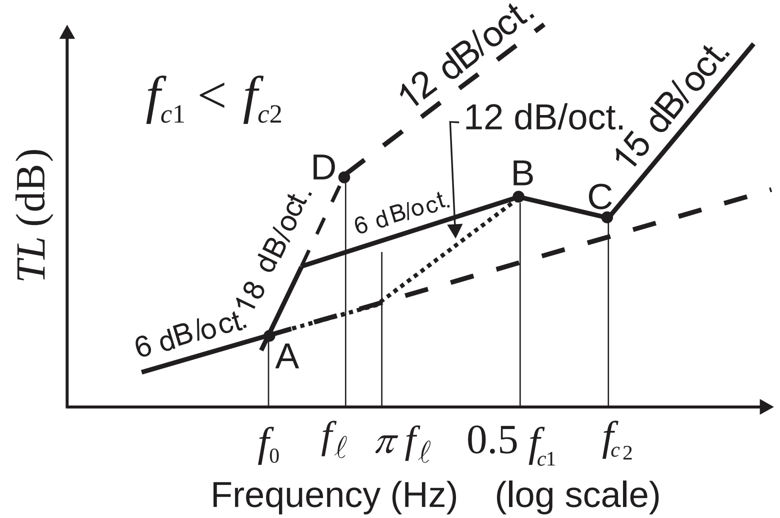
$$TL = \begin{cases} TL_M \quad (M = m_1 + m_2) & f \le f_0 \\ TL_1 + TL_2 + 20\log_{10} fd - 29 & f_0 < f < f_\ell \\ TL_1 + TL_2 + 6 & f \ge f_\ell \end{cases}$$

$$TL = 10\log_{10}\left[1 + \left(\pi f m/(\rho c)\right)^2\right] - 5.5 \quad \text{(dB)}$$

$$TL = 20\log_{10}\left[\pi f m/(\rho c)\right] + 10\log_{10}\left[2\eta f/(\pi f_c)\right] \text{ (dB)}$$

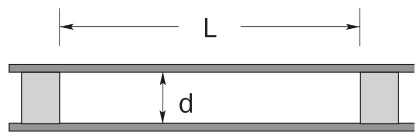
- Acoustical isolation
  Fibreglass or rockwool in cavity
- Mechanical isolation
  Difficult to achieve especially at edges
  Panels usually attached to studs - point or line support.

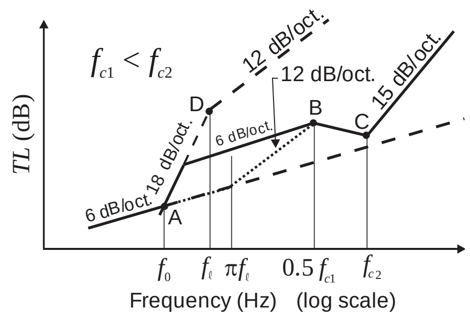
Design Chart For *TL* of a Double Panel Wall



ENC p379

Design Chart For *TL* of a Double Panel Wall





m_1 and m_2 (kg m^{-2}) are the respective panel surface densities

$d(m)$ is the spacing between panels

$b(m)$ is the spacing between line supports

$e(m)$ is the spacing of an assumed rectangular grid between point supports

c and c_L (m s^{-1}) are, respectively the speed of sound in air and in the panel material

h is the panel thickness

η_1 and η_2 are the loss factors respectively for panels 1 and 2.

Double Wall *TL*

- POINT A: $f_0 = 80\sqrt{(m_1+m_2)/dm_1m_2}$ (Hz)

  $TL_A = 20\log_{10}(m_1+m_2) + 20\log_{10} f_0 - 48$ (dB)

- POINT C:

(a) $f_{c2} \neq f_{c1}$,

$$TL_C = TL_B + 6 + 10\log_{10}\eta_2 + 20\log_{10}\left(\frac{f_{c2}}{f_{c1}}\right) \quad \text{(dB)}$$

(b) $f_{c2} = f_{c1}$,

$$TL_C = TL_B + 6 + 10\log_{10}\eta_2 + 5\log_{10}\eta_1 \quad \text{(dB)}$$

- POINT D: $f_\ell = 55/d$

ENC p379

Double Wall *TL*: Point B:

The transmission loss, TL_B, at Point B is:

– equal to TL_{B1} if no sound absorptive material is placed in the cavity between the two panels;

$$TL_{B1} = TL_A + 20\log_{10}(f_{c1}/f_0) + 20\log_{10}(f_{c1}/f_\ell) - 22 \quad \text{(dB)}$$

– otherwise $TL_B = TL_{B2}$

(a) Line-line support:

$$TL_{B_2} = 20\log_{10} m_1 + 10\log_{10} b + 20\log_{10} f_{c1} + 10\log_{10} f_{c2} + 20\log_{10}\left[1 + \frac{m_2 f_{c1}^{1/2}}{m_1 f_{c2}^{1/2}}\right] - 78 \quad \text{(dB)}$$

ENC p379

Double Wall *TL*: Point B:

(b) Line-point support:

$$TL_{B_2} = 20\log_{10} m_1 e + 20\log_{10} f_{c_1} + 20\log_{10} f_{c_2} - 99 \ \text{(dB)}$$

(c) Point-point support (see text book).

ENC p379

Double Wall *TL*

Again Davy offers a more complex (but not necessarily a more accurate) method - see text book, pages 381–383.

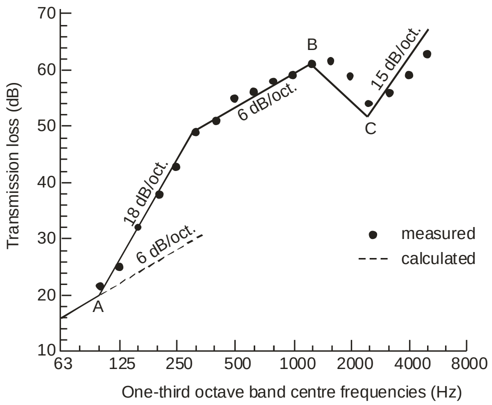
Davy does allow for the use of flexible steel studs as well whereas Sharp only considers rigid wooden studs.

Multi-leaf and composite panels are discussed in the textbook, page 386–387

Triple panel walls are discussed on page 387 in the text

measured TL values for typical wall constructions are given on pages 388–393 of the text book.

Example 8.1 Gypsum board double wall



3.0m × 2.44m × 0.1m spacing

$m_1 = m_2 = 12.67$ kg/m²

$f_{c_1} = f_{c_2} = 2500$ Hz

Porous acoustic material in cavity (50mm thick)

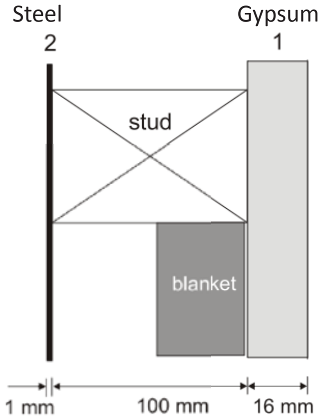
i.e. pretty accurate.

Problem … 2009 Exam Question

Q4 (a) Using the graph paper supplied on the back of this question sheet and Figure 8.9 in your text book, estimate and tabulate the sound transmission loss in dB for each 1/3 octave band from 400 Hz to 4000 Hz, for a double wall made up of one mild steel panel, 1 mm thick, and one gypsum board panel 16 mm thick, fixed to 100 mm deep studs, placed 600 mm apart, with a 50 mm thick blanket of acoustic material in the cavity. Assume that the panels are line supported. Use a loss factor of 0.1 for the gypsum board and 0.02 for the steel panel.

[25 marks]

Answer ...



$$f_c(\text{steel}) = \frac{0.55 \times 343^2}{0.001 \times \dfrac{5130}{\sqrt{1 - 0.3^2}}} = 12 \text{ kHz}$$

$$f_c(\text{gypsum}) = \frac{0.55 \times 343^2}{0.016 \times \dfrac{1670}{\sqrt{1 - 0.24^2}}} = 2.35 \text{ kHz}$$

$f_\ell = 55/0.1 = 550$ Hz

$m_{\text{steel}} = 7850 \times 0.001 = 7.85$ kg/m^2 and $m_{\text{gypsum}} = 760 \times 0.016 = 12.16$ kg/m^2

Answer ...

$$f_0 = 80\sqrt{\frac{12.16+7.85}{0.1 \times 12.16 \times 7.85}} = 116 \text{ Hz}$$

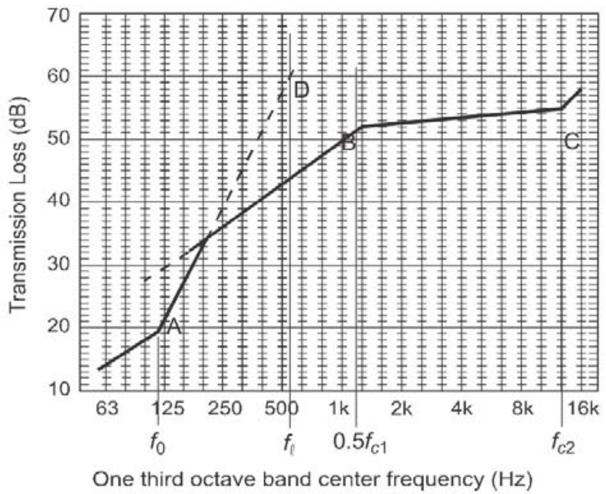
$$TL_A = 20\log_{10}(7.85 + 12.16) + 20\log_{10}115.8 - 48 = 26.02 + 41.27 - 48 = 19.3 \text{ dB}$$

$$TL_{B2} = 20\log_{10}12.16 + 10\log_{10}0.6 + 20\log_{10}(2351) + 10\log_{10}(12032) + 20\log_{10}\left[1 + \frac{7.85 \times 2351^{1/2}}{12.16 \times 12032^{1/2}}\right] - 78$$

$$21.7 - 2.2 + 67.4 + 40.8 + 2.2 - 78 = 51.9 \text{ dB at } 2351/2 = 1175 \text{ Hz}$$

$$TL_C = 51.9 + 6 + 10\log_{10}0.02 + 20\log_{10}\left[\frac{12032}{2351}\right] = 55.1 \text{ dB at } f = 12 \text{ kHz}$$

Answer ....



From the graph ...

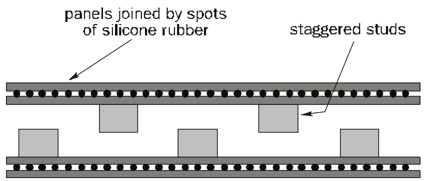
1/3 octave centre frequency (Hz)	TL
400	40.5
500	43
630	45
800	47.5
1000	49.5
1250	52
1600	52.2
2000	52.8
2500	53
3150	53.2
4000	53.5

EDGE SUPPORTED ONLY DOUBLE WALL

For this case, 4dB is added to TL_B as the area associated with the support is less than half that assumed by the design chart.

STAGGERED STUDS

Panels only connected at edges but performance not as good as edge supported with no studs. In this case add 2dB to TL_B of the design chart.



If panels are well damped, adding 4dB to TL_B will give conservative predictions.

Example Construction Techniques

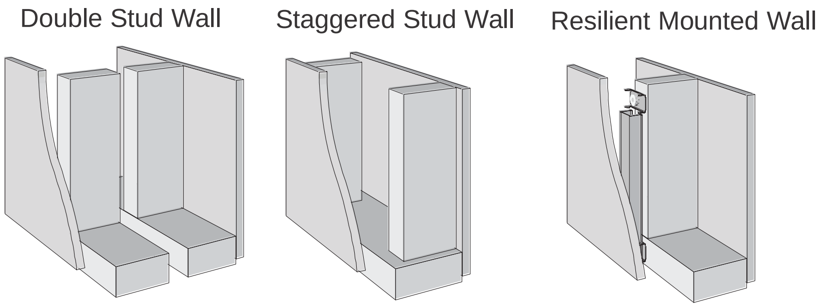


Table 8.3 Calculated transmission loss (*TL*) values (dB) for a typical blanket of porous acoustic material (medium density rockwool, 50 mm thick).

Frequency (Hz)	TL (dB)
1,000	0.5
2,000	1.5
4,000	4.0
8,000	12.0

Lining a wall with rockwool or fibreglass adds to high frequency *TL* only (assumes weak connection)

ENC p394

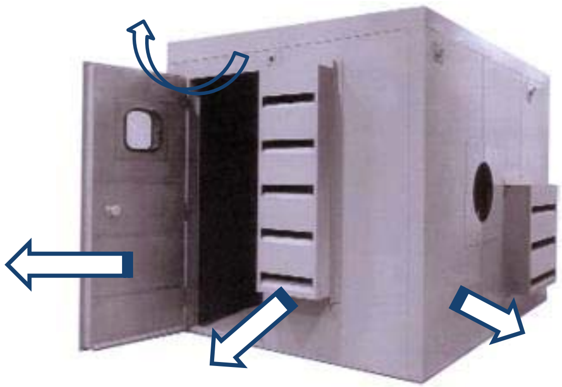
- Combining 1/3 octave band *TL* to get 1/1 octave band *TL*

Let 1/3 octave band *TL's* be TL_1, TL_2, TL_3

Then, $$TL_{oct} = -10\log_{10}\frac{1}{3}\left[10^{-TL_1/10} + 10^{-TL_2/10} + 10^{-TL_3/10}\right]$$

- Combining 1/3 octave band *SPL* to get 1/1 octave band *SPL*

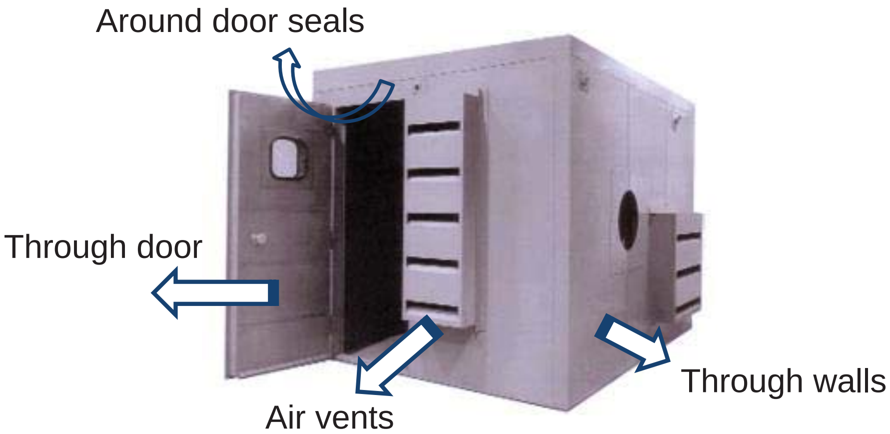
$$SPL_{oct} = 10\log_{10}\left[10^{L_{P1}/10} + 10^{L_{P2}/10} + 10^{L_{P3}/10}\right]$$



Composite Transmission Loss

Composite Transmission Loss

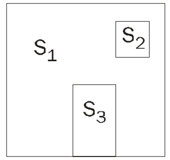
- There are multiple noise transmission paths.
- How do we combine the TLs for each path?



Composite Transmission Loss

See p395 in textbook

$$\tau = \frac{\Sigma S_i \tau_i}{\Sigma S_i}$$



WALL 1:	TL = 30 dB,	S_1 = 30m²
WINDOW 2:	TL = 20 dB,	S_2 = 1m²
DOOR 3:	TL = 15 dB,	S_3 = 2m²

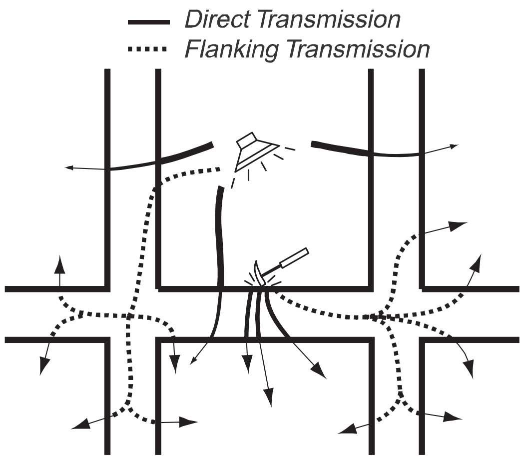
$\tau_1 = 10^{-30/10} = 0.001$
$\tau_2 = 10^{-20/10} = 0.01$
$\tau_3 = 10^{-15/10} = 0.0316$

$$\tau = \frac{0.001 \times 30 + 0.01 \times 1 + 0.0316 \times 2}{33} = 0.00313$$

$$TL = -10 \log_{10} \tau = 25 \text{ dB}$$

ENC p395

Flanking Transmission Loss



Flanking Transmission Loss

- Takes into account sound transmission paths that arise due to the way the panel is installed.

$$TL_{overall} = -10 \log_{10}\left(10^{-TL\ flank\ /10} + 10^{-TL/10}\right) \text{ dB}$$

- where

$$TL_{flank} = D_{n.f} - 10 \log_{10}\left(\frac{10}{A}\right)$$

$$D_{n.f} = L_1 - L_2 - 10 \log_{10}\left(\frac{S\bar{\alpha}}{10}\right)$$

If the field arrangement is different to the laboratory set up, then the value of $D_{n,f}$ must be adjusted as illustrated for a suspended ceiling on page 397 of the text.

ENC p397

Checkpoint ....

So far we have covered:

- How to calculate the TL of single and double wall panels.
- How to experimentally measure the TL of a panel.
- How to calculate the composite TL from multiple paths to determine the overall TL.

Now what we need to do is to be able to calculate what is the noise level OUTSIDE the enclosure
.... After all this is the whole point of installing the enclosure in the first place – to reduce the noise level outside the enclosure

Effect of Installing an Enclosure

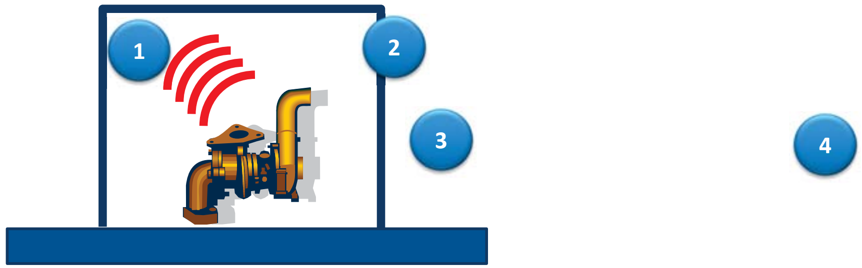
- Noise level inside enclosure will be ***louder*** than if machine sat in the open.

Effect of Installing an Enclosure

- Need to install sound absorbing material on the walls.

Prediction of Noise Level Outside Enclosure

1. Determine noise level inside enclosure
2. Determine composite transmission loss
3. Determine noise level outside enclosure
4. Determine noise level at a distance



Enclosure Noise Reduction

- sound pressure p_1 immediately outside enclosure

$$\langle p_1^2 \rangle = \tau_N \langle p_D^2 \rangle + \tau_F \langle p_R^2 \rangle / 4 = \tau_N \frac{W\rho c}{S_E} + \tau_F \frac{W(1-\bar{\alpha}_i)\rho c}{S_i \bar{\alpha}_i}$$

$$S_E \langle p_1^2 \rangle / (\rho c) = W\tau_N + W(1-\bar{\alpha}_i)[S_E / (S_i \bar{\alpha}_i)]\tau = W\tau_E$$

Taking logs of both sides

$$\boxed{L_{p1} = L_w - TL - 10\log_{10} S_E + C}$$

S_E = external surface area in m²
τ_N, τ_F = normal and field incidence transmission coeffs (no units)
L_w = source sound power in dB re 1pW
Lp = average SPL immediately outside enclosure in dB re 20micro-Pa
C = constant for enclosure internal conditions in dB

ENC p399

Enclosure Noise Reduction

As τ_F is the same as τ and

$$\tau_N = 0.316\tau \quad [TL_N = TL_F - 5]$$

$$\tau_E = \tau[0.3 + S_E(1-\bar{\alpha}_i)/(S_i\bar{\alpha}_i)]$$

$$C = 10\log_{10}\left[0.3 + \frac{S_E(1-\bar{\alpha}_i)}{(S_i\bar{\alpha}_i)}\right] \text{ (dB)}$$

ENC p399

Enclosure Noise Reduction

- sound pressure, p_2, at some distance from enclosure
  - enclosure outdoors in free space

$$L_{p2} = L_{p1} + 10\log_{10} S_E + 10\log_{10}(D_\theta / 4\pi r^2) \text{ (dB)}$$

  - enclosure indoors

$$L_{p2} = L_{p1} + 10\log_{10} S_E + 10\log_{10}\left[\frac{D_\theta}{4\pi r^2} + \frac{4(1-\bar{\alpha})}{S\bar{\alpha}}\right] \text{ (dB)}$$

- no enclosure

$$L'_{p2} = L_w + 10\log_{10}\left[\frac{D_\theta}{4\pi r^2} + \frac{4(1-\bar{\alpha})}{S\bar{\alpha}}\right] \text{ (dB)}$$

- enclosure noise reduction

$$NR = L'_{p2} - L_{p2} = L_w - L_{p1} - 10\log_{10} S_E \quad \text{(dB)}$$
$$= TL - C \quad \text{(dB)}$$

ENC p402

Enclosure Noise Reduction

- Values of constant C (dB) to account for enclosure internal acoustic conditions.

Enclosure internal acoustic conditions	Octave Band Centre Frequency (Hz) 63	125	250	500	1,000	2,000	4,000	8,000
live*	18	16	15	14	12	12	12	12
fairly live	13	12	11	12	12	12	12	12
average	13	11	9	7	5	4	3	3
dead	11	9	7	6	5	4	3	3

ENC p400

Enclosure Noise Reduction

Use the following criteria to determine the appropriate acoustical conditions inside the enclosure:

live: all enclosure surfaces and machine surfaces hard and rigid

fairly live: all surfaces generally hard but some panel construction (sheet metal or wood)

average: enclosure internal surfaces covered with sound-absorptive material, and machine surfaces hard and rigid

dead: as for "average", but machine surfaces mainly of panels.

ENC p400

Personnel Enclosures

- assume enclosure located in a reverberant field, $\langle p_1^2 \rangle$
- power flow into enclosure

$$W_i = \tau\, S_E \times \frac{\langle p_1^2 \rangle}{(4\rho c)}$$

$$L_{wi} = L_{p1} + 10\log_{10} S_E - TL - 6 \quad \text{(dB)}$$

- sound pressure level inside enclosure

$$L_{pi} = L_{wi} + 10\log_{10}\left[\frac{1}{S_E} + \frac{4(1-\bar{\alpha}_i)}{S_i\bar{\alpha}_i}\right] \quad \text{(dB)}$$

$$= L_{wi} + 10\log_{10} S_E + 6 + 10\log_{10}\left[\frac{1}{4} + \frac{S_E(1-\bar{\alpha}_i)}{S_i\bar{\alpha}_i}\right] \quad \text{(dB)}$$

ENC p402

Personnel Enclosures

- enclosure noise reduction

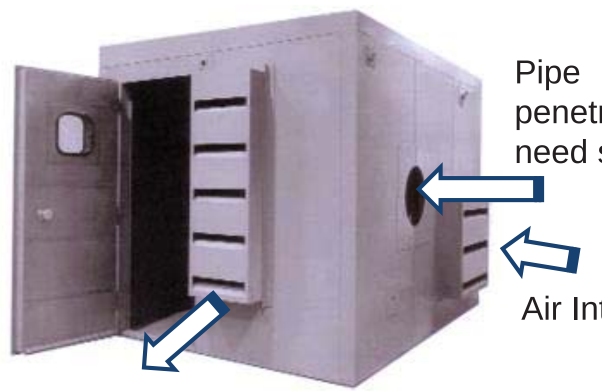
$$NR \approx L_{p1} - L_{pi} = TL - C \quad \text{(dB)}$$

ENC p402

- Enclosure performance reduction vs % internal surface covered with rockwool

% sound absorbent	10	20	30	50	70
Performance reduction (dB)	-10	-7	-5	-3	-1.5

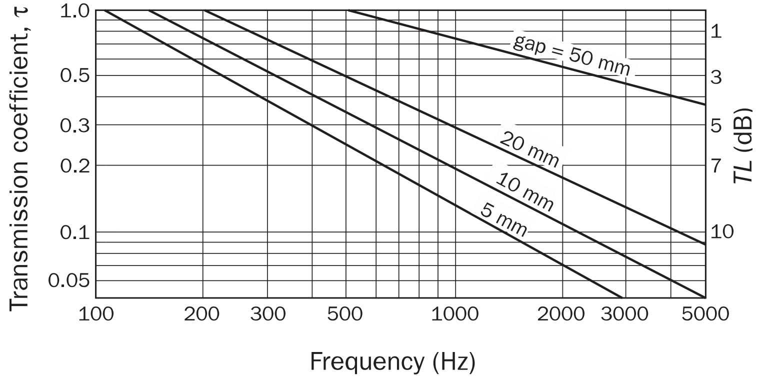
Practical Matters for Enclosures



Enclosure Leakages, Text p.406

Transmission coefficients of long, narrow cracks.

Alternatively, see text book for
Gomperts and Kihlman's (1967) expression.



ENC p406

Practical Matters for Enclosures

See p407 in textbook

- An enclosed box with equipment is going to get (very) *hot* ! Therefore, need ventilation.

$$\boxed{\rho C_P V = H / \Delta T}$$

ρ = density (air 1.21 kg /m^3)
C_p = specific heat capacity of the gas (air 1010 m^{2}s^{-2}C^{-1})
V = volume of airflow required (m^3/s)
H = heat input into the enclosure (W)
ΔT = temperature differential between external ambient and max allowable internal temp (°C)

- Vibration Isolation:
  - Classic mistakes made here.
  - Need to vibration isolate pipes from walls.
  - Need to isolate machinery from walls.

ENC p407

Practical Matters for Enclosures

- Enclosure Resonances:
  - Panel Resonances
    - vary mass or stiffness
  - Acoustic Resonances
    - add porous acoustic absorbent

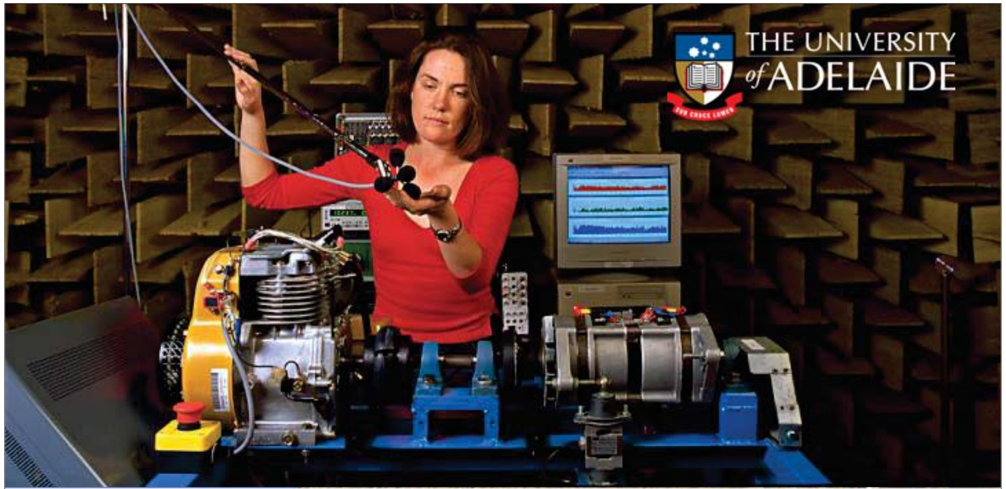
- Close Fitting Enclosures:
  - Performance degradation due to acoustic resonances ($\delta \text{TL} = 10$ dB).
  - To control low frequency sound - light and stiff panels.
  - To control high frequencies – heavy and limp panels.

Summary

- General TL features and coincidence frequency
- How TL is measured
- STC and RW ratings
- Impact Isolation Class
- How to predict TL of panels
  - (isotropic, orthotropic, double wall)
- Composite TL
- Flanking TL
- Use TL to predict the NR of an enclosure
- Enclosure leakage from a slit causing reduction of TL

Questions you should be able to answer

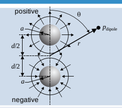
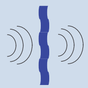
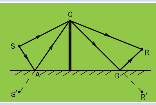
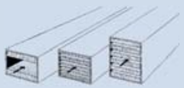
- Design a panel to provide a specified TL.
- Design an acoustic enclosure that will provide a specified NR.
- Given the TL of panels on an enclosure, the Lw of a machine inside the enclosure, what is the predicted SPL at distance *y* metres from the enclosure?
  Will it satisfy the noise ordinance requirements?
  What would be the expected SPL inside the enclosure?
- If the machine generated 5 kW of heat inside an enclosure, what rate of airflow would be required to keep the temperature inside the enclosure at *t* deg C?



Carl Howard

Engineering Acoustics
Barriers and Pipe Wrapping

Course Content – How it fits

TOPIC	
Sound Sources	
Outdoor	
Transmission Loss of Panels	
Barriers	
Mufflers	

Contents

- Design of acoustic barriers
- Pipe Wrapping

- Example of barrier design
- Example of pipe wrapping design

Learning Outcomes

Be able to:

- design an acoustic barrier
- determine the IL of a pipe wrapping

Examples of Barriers

- Famous (?) barrier on freeway into Melbourne

Examples of Barriers

- Freeways - Melbourne

Examples of Barriers

- Roads – South Rd & ANZAC Highway intersection

Examples of Barriers

From: http://www.connecteast.com.au

Noise Barriers in U.S.A.

- According to the US Federal Highway Administration, as of 2010, there were 4424 linear kilometres (!!) of noise barriers built in the U.S., at a total cost of $5.44 billion in 2010 dollars.

https://www.fhwa.dot.gov/environment/noise/noise_barriers/inventory/summary/sintro7.cfm

https://medium.com/re-form/muting-the-freeway-e18ee195bd38

Examples of Barriers

- Barrier next to train line

Examples of Barriers

- Roof top barrier

Examples of Barriers

- Indoor barrier

Examples of Barriers

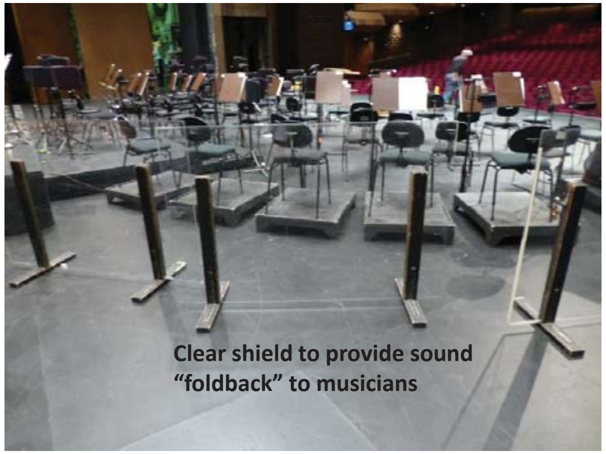
- Cube farm – Indoor barrier

Tour of Adelaide Festival Theatre

Tour of Adelaide Festival Theatre

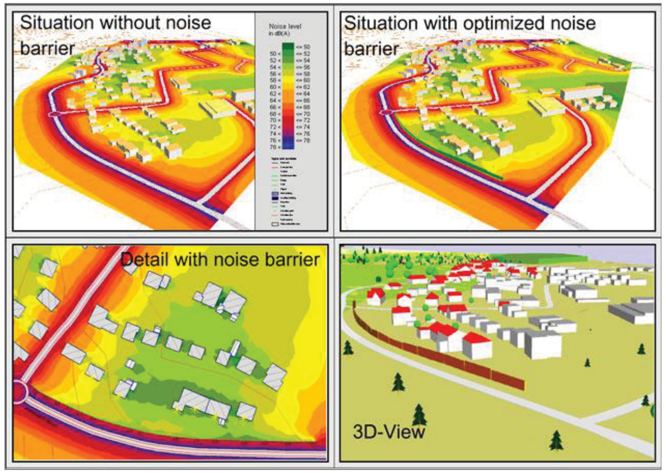
Tour of Adelaide Festival Theatre

Tour of Adelaide Festival Theatre



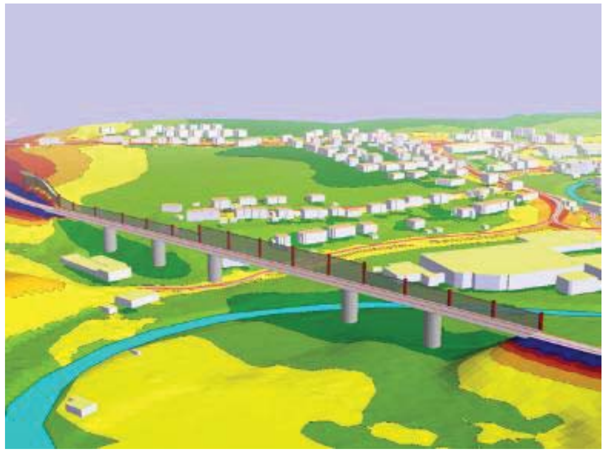
Numerical Modelling

SoundPlan software used to model effect of noise barrier



Numerical Modelling

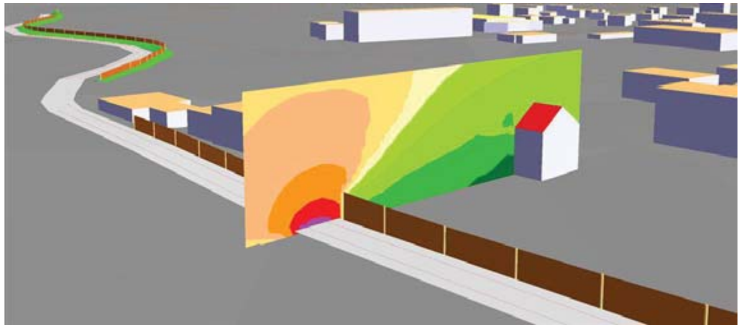
SoundPlan software used to model effect of noise barrier



Numerical Modelling

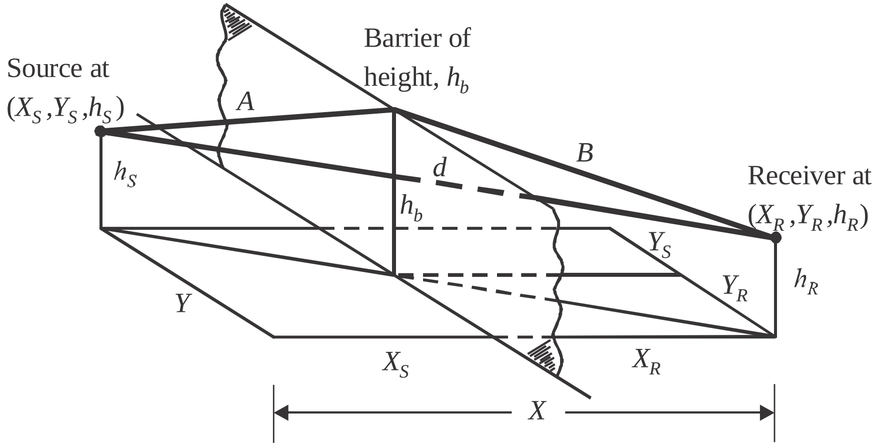
SoundPlan software used to model effect of noise barrier

Cross sectional noise control plot showing noise “shadow” behind the barrier, which reduces noise on the nearby house.



Sound Barriers

See p412 in textbook



Sound Barriers

Diffraction over a thin sheet (text, p.412)

Fresnel Number N:

$$N = (2/\lambda)\,(A + B - d)$$

$$d = [X^2 + Y^2 + (h_R - h_S)^2]^{1/2}$$

$$A = [X_S^2 + Y_S^2 + (h_b - h_S)^2]^{1/2}$$

$$B = [X_R^2 + Y_R^2 + (h_b - h_R)^2]^{1/2}$$

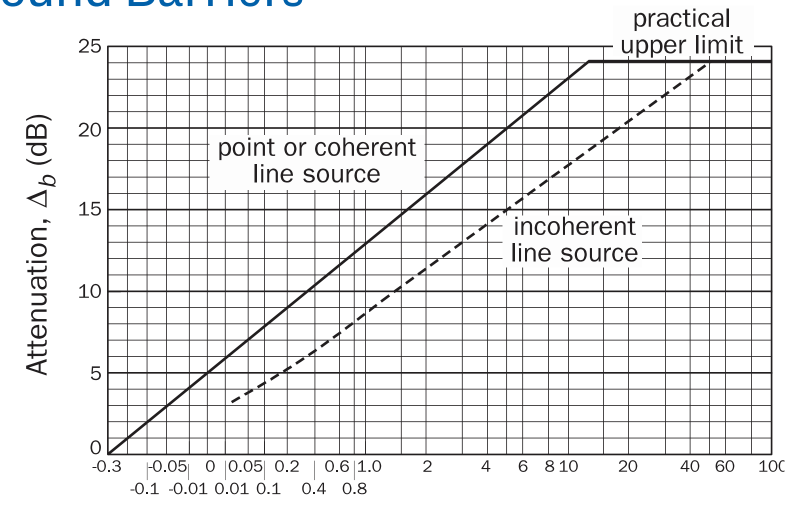
$$Y_R = Y \times \frac{X_R}{X}$$

$$Y_S = Y_R \times \frac{X_S}{X_R}$$

N is -ve for line of sight between S and R.

Sound Barriers

See p413, Fig 8.14 in textbook



Barrier noise reduction vs Fresnel number

Comments on Fresnel Graph

- Warning: the x-axis scale is irregular – it has been stretched to make the data line straight. Use with caution.
- There are 2 curves:
  - **Point** or **Coherent** line source
  - **Incoherent** line source
- Select the correct curve for your situation!

Sound Barriers

- Then use Eq. 8.104, p.414

$$\boxed{A_{bi} = \Delta_{bi} + 20\log_{10}\left[(A_i + B_i)/d_i\right] + D_{\theta R} - D_{\theta B}}$$

- to calculate the attenuation of a single sound path i due to diffraction over the barrier.
- NOTE: when paths involving *ground reflections* are considered, we use the image locations, and d_i is the effective distance between the effective position of the source and the effective position of the receiver.

Sound Barriers

- Alternative calculation to preceding graph.

$$A_{bi} = IL_s + IL_b + IL_{sb} + IL_{sp} + D_{\theta R} - D_{\theta B}$$

where

$$IL_s = 20\log_{10}\frac{\sqrt{2\pi N_i}}{\tanh\sqrt{2\pi N_i}} - 1$$

$$IL_{sb} = \left(6\tanh\sqrt{N_2} - 2 - IL_b\right)\left(1 - \tanh\sqrt{10 N_i}\right)$$

$$IL_b = 20\log_{10}\left[1 + \tanh\left(0.6\log_{10}\frac{N_2}{N_i}\right)\right]$$

ENC p414

Sound Barriers

where

N_2 =Fresnel number for a wave travelling from the image source to the receiver where the image source is generated by reflection from the barrier (not the ground).

d = straight line distance between the image source (due to reflection in the barrier) and receiver

$(A + B)$ =same as used to calculate Fresnel number for the actual source and receiver.

ENC p414

Sound Barriers

$$IL_{sp}=\begin{cases}3\text{db} & \text{for plane waves}\\ 10\log_{10}\left(1+(A+B)/d\right) & \text{for cylindrical waves}\\ 10\log_{10}\left((A+B)^2/d^2+(A+B)/d\right) & \text{for spherical waves}\end{cases}$$

ENC p414

Outdoor Barriers

- Assume <u>incoherent</u> combination of sound fields arriving from all propagation paths.

<u>Barrier source - receiver paths</u>

- No Barrier
  - 2 possible paths
- Infinite Length Barrier
  - 4 possible paths
- Finite Length Barrier
  - 8 possible paths

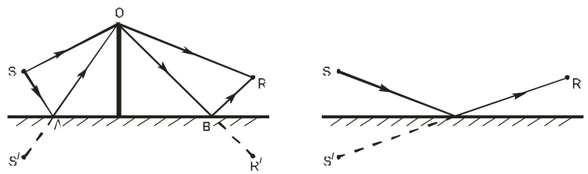
<u>Barrier lined with acoustically porous material</u>

- Some beneficial effect, dependent on:
  - barrier height
  - barrier distance from source
  - barrier distance from receiver

Sound Paths Over And Around Barriers

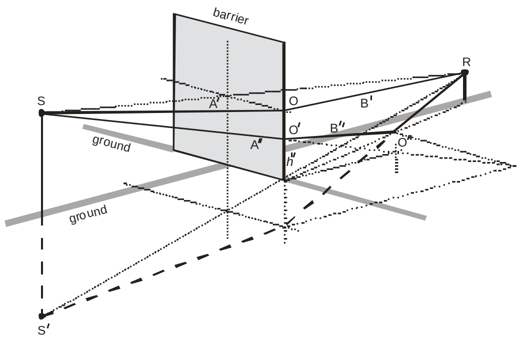
- over top

See p415 in textbook



- around sides

See p417 in textbook



h_b' = height of point O above the ground

$$h_b'=\frac{X_R h_s+X_S h_r}{X_S+X_R}$$

h_b'' = height of point O′ above the ground:

$$h_b''=\frac{X_S h_r-X_R h_s}{X_S+X_R};\quad\text{and}\quad h_b''=\frac{X_R h_s-X_S h_r}{X_S+X_R}$$

Left Equation refers to reflection on source side

$$A'=\sqrt{(h_s-h_b')^2+X_S^2+(Y_S-Y_B)^2}$$

$$B'=\sqrt{(Y_R-Y_B)^2+X_R^2+(h_b'-h_r)^2}$$

$$A''=\sqrt{(h_s-h_b'')^2+X_S^2+(Y_B-Y_S)^2}$$

$$B''=\sqrt{(Y_R-Y_B)^2+X_R^2+(h_r+h_b'')^2}$$

Barrier Insertion Loss IL

$$A_{bi} = \Delta_{bi} + 20\log_{10}\left[(A_i + B_i)/d_i\right] + D_{\theta R} - D_{\theta B}$$

Multiple Paths (over top and around sides)

$$A_b = 10\log_{10}\left[1 + 10^{-(A_{Rw}/10)}\right] - 10\log_{10}\sum_{i=1}^{n_A} 10^{-(A_{bi}+A_{Ri})/10}$$

$$A_R = -20\log_{10}\left|R_p\right|$$

ENC p418

Correction in 4th edition of textbook, on page 415, it has written:

When paths involving one or more ground reflections are considered, the straight line distance, d, used in Equation (8.101) is the same as for non-reflected waves as the Fresnel number and associated noise reduction are relative to the straight line propagation from S to R.

This is incorrect and should be deleted.

See Example 8.7, page 419-421 that show it is the straight line distance between the effective source and effective receiver.

Extract from journal paper… Maekawa, Z. (1968), "Noise Reductions by Screens", Applied Acoustics, vol 1, p157-173.

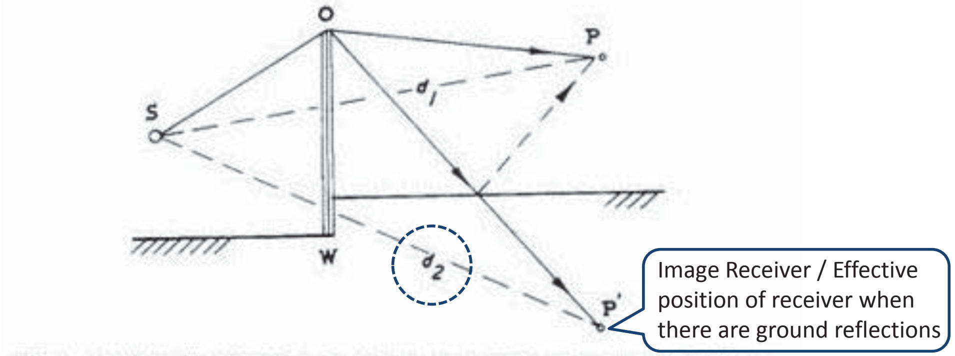


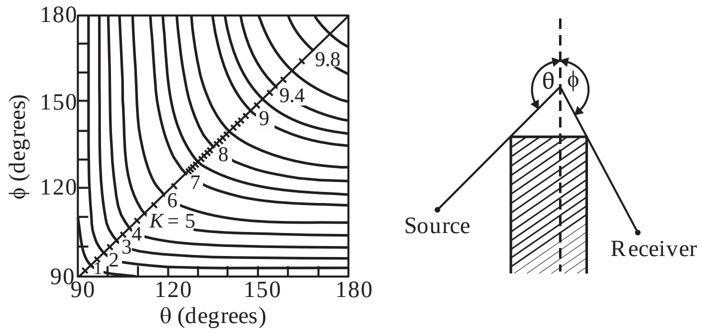
Fig. 5. Section of the long wall between a sound source S and a receiving point P.

(3) The effect of ground reflection is calculated by applying the same method for a point P′, the image of the receiving point P, assuming perfect specular reflection on the ground. The attenuation L_2 dB of the sound at P′ is obtained from Fig. 4, with the variable $N_2 = \delta_2 . 2/\lambda$, where $\delta_2 = (\overline{SO} + \overline{OP} - d_2)$.

This is the right way to solve the problem…

Thick Barriers

See p420 in textbook



Finite width barrier correction factor, K

Additional attenuation over thin barrier is

$$\Delta C = K\log_{10}(2\pi b/\lambda) \text{ for b} > \lambda/2$$

ENC p420

Sound Barriers

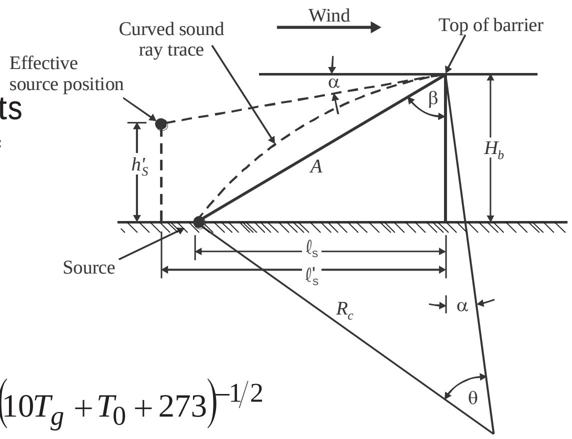
See p423 in textbook

- effects of wind and temperature gradients
  - radius of curvature of sound wave

$$r = c_0 / G_s \quad \text{(m)}$$

where the total sonic gradient, G_s is

$$G_S = 0.015U + 10.29T_g \left(10T_g + T_0 + 273\right)^{-1/2}$$



Sound Barriers

$$\ell'_s = R\theta\cos\alpha$$

$$h'_s = H_b - R\theta\sin\alpha$$

$$\alpha = \frac{1}{2}(\pi - \theta) - \beta$$

$$\beta = \cos^{-1}(H_b / A)$$

$$\theta = \pm\cos^{-1}\left[1 - \left(A^2 / 2R^2\right)\right], \quad |R| > A/2$$

ISO Approach To Barrier Insertion Loss

For path, *i*,

$$A_{bi} = 10\log_{10}\left[3 + 10N_i C_3 K_{met}\right] - A_g \quad \text{dB}$$

If ground reflected paths considered separately, replace “10" in above equation with “20" .

For multiple paths

$$A_b = -10\log_{10}\sum_{i=1}^{n_A} 10^{-(A_{bi} + A_{Ri})/10}$$

N_i is Fresnel number

Single barriers, $C_3 = 1$

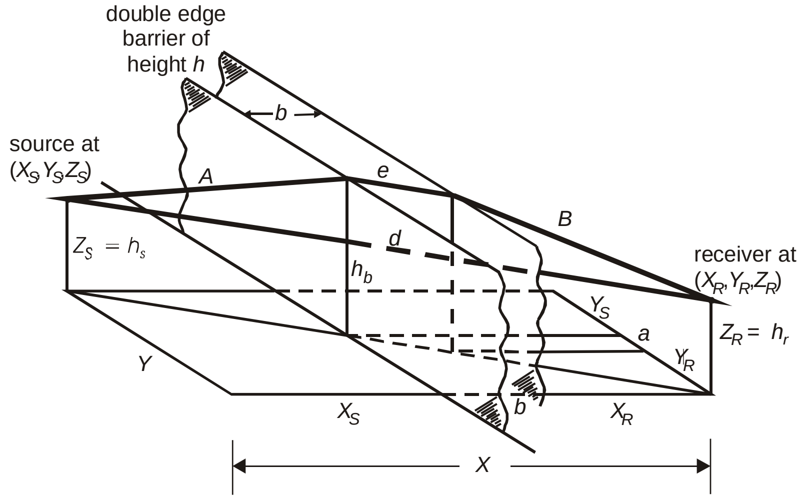
K_{met} includes effect of wind and temperature gradients

A_g excess atten due to the ground in absence of barrier

ENC p425

Double Barriers

See p426 in textbook



ISO Approach To Barrier Insertion Loss

See p425 in textbook

Double edge barriers (25 dB max possible):

$$C_3 = \left[1+(5\lambda/b)^2\right] \Big/ \left[(1/3)+(5\lambda/b)^2\right]$$
$$N = \pm\,(2/\lambda)(A+B+e-d)$$
$$e = \left(a^2+b^2\right)^{1/2}$$
$$K_{met} = \exp\left[-(1/2000)\sqrt{\frac{ABd}{2(A+B+e-d)}}\right]$$

For two barriers of different height:

$$e = \left(a^2+b^2+(h_{b1}-h_{b2})^2\right)^{1/2}$$

Indoor Barriers

See p427 in textbook

$$IL = 10\log_{10}\left[\frac{D}{4\pi r^2}+\frac{4}{S_o\alpha_o}\right] \quad \text{original room}$$
$$-10\log_{10}\left[\frac{DF}{4\pi r^2}+\frac{4K_1K_2}{S(1-K_1K_2)}\right] \quad \text{new room}$$

$$F = \sum_i \frac{1}{3+10N_i}, \quad \text{i edges}$$

$$K_1 = \frac{S}{S+S_1\overline{\alpha}_1}; \quad \text{S is open area around barrier}$$
$$K_2 = \frac{S}{S+S_2\overline{\alpha}_2}; \quad S_1\overline{\alpha}_1,\ S_2\overline{\alpha}_2 \quad \text{include barrier}$$

ASSUMPTIONS

- *TL* through barrier ignored
- L_w radiated by source not affected by barrier
- Receiver is in shadow zone of barrier
- Incoherent combination of sound from all propagation paths.

MULTIPLE BARRIERS

- See qualitative statements pp.402–403, text.

Example Barrier Design

8.22: Calculate the noise reduction (in dB(A)) due to inserting a 4m high thin barrier, 15m in length midway between a noisy refrigeration unit and a residence located 50m from the unit across an asphalt covered lot.

Assume that the acoustic centre of the refrigeration unit is 0.5m above the ground and the community location is 1.5m above ground level.

Example Barrier Design ....cont'd

Without the barrier the noise levels due to the refrigeration unit in the octave bands between 63Hz and 8kHz were measured respectively as:

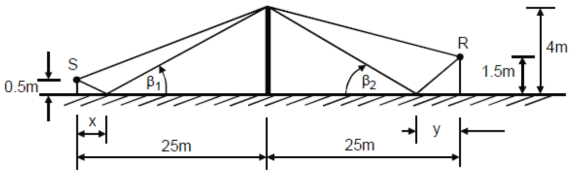
70, 75, 72, 60, 58, 56, 50, 52dB.

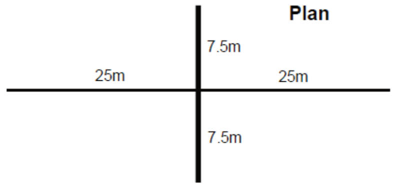
Use Fig. 5.20 in the text to calculate the ground reflection loss, assuming that no wind or temperature gradients exist and that sound from all paths combines incoherently at the receiver.

Solution

- Step 1 --- draw some diagrams of the situation

Over top (elevation)

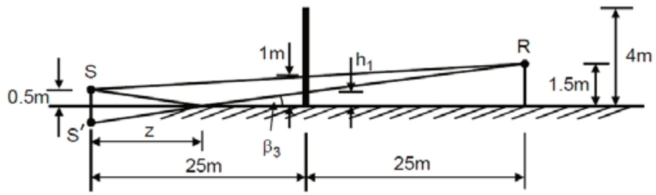


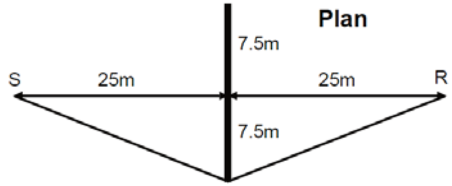


Solution

- Step 1 --- draw some diagrams of the situation

Around edge (elevation)





Solution

- Step 2: Calculate the angles

Referring to the following figures, we have for the reflection angles:

over top, source side,

$$\tan\beta_1 = \frac{4}{25-x} = \frac{0.5}{x}; \quad x = 2.78\,\mathrm{m}, \quad \beta_1 = 10.2°$$

over top, receiver side,

$$\tan\beta_2 = \frac{4}{25-y} = \frac{1.5}{y}; \quad y = 6.82\,\mathrm{m}, \quad \beta_2 = 12.4°$$

around edge, $\tan\beta_3 = \dfrac{1.5}{50-z} = \dfrac{0.5}{z}; \quad z = 12.5\,\mathrm{m}, \quad \beta_3 = 2.3°$

Solution

- Step 3: Calculate the loss from the ground reflection.

Flow resistivity of ground = $R_1 = 3 \times 10^7$ MKS rayls/m and $\rho = 1.206\mathrm{kg/m^3}$. With no barrier, $\beta_4 = \beta_3 = 2.3°$. The reflection loss, A_R is calculated using figure 5.20 in the text and the results are tabulated in the table below.

Octave band centre frequency (Hz)	$\frac{\rho f}{R_1}$	$\left(\frac{R_1}{\rho f}\right)^{1/2}$	A_{R1}	A_{R2}	A_{R3} and A_{R4}
63	2.5×10^{-6}	630	0.2	0.2	1.5
125	5.0×10^{-6}	447	0.3	0.3	2.0
250	1.0×10^{-5}	316	0.5	0.5	2.5
500	2.0×10^{-5}	223	0.7	0.7	3.5
1000	4.0×10^{-5}	158	1.0	1.0	5.0
2000	8.0×10^{-5}	112	1.5	1.5	6.0
4000	1.6×10^{-4}	79	2.3	2.3	7.0
8000	3.2×10^{-4}	56	3.0	3.0	6.5

Solution

- Step 4: Calculate angles for all paths.

Path 1 - over top of barrier with no ground reflections
$d = 50.010$m, $A = 25.244$, $B = 25.125$ and $A + B - d = 0.359$m

Path 2 - over top of barrier with ground reflection on the source side
$d = 50.040$m, $B = 25.125$
$A = (2.78^2 + 0.5^2)^{1/2} + (4^2 + 22.22^2)^{1/2} = 25.402$m and
$A + B - d = 0.487$m

Path 3 - over top of barrier with ground reflection on the receiver side
$d = 50.010$m, $A = 25.244$
$B = (6.82^2 + 1.5^2)^{1/2} + (4^2 + 18.18^2)^{1/2} = 25.598$m and
$A + B - d = 0.802$m

Path 4 - over top of barrier with ground reflection on both sides
$d = 50.010$m, $A = 25.402$, $B = 25.598$ and $A + B - d = 0.990$m

Paths 5&6 - around edges with no reflection
$B = A = (0.5^2 + 25^2 + 7.5^2)^{1/2} = 26.106$m and $A + B - d = 2.201$m

Solution

Paths 7&8 - around edges with reflection in ground
Intersection height of diffracted wave with barrier edge
$= 1.5 \times 12.5/37.5 = 0.5$m.
$A = 2(0.5^2 + 12.5^2 + (7.5/2)^2)^{1/2} = 26.120$m
$B = 2(1^2 + 25^2 + 7.5^2)^{1/2} = 26.120$m and
$A + B - d = 2.20$m

- Step 5: Now calculate the Fresnel number for each path.

Fresnel Number, $N = \dfrac{2f}{343}(A + B - d)$. Values of N for each path are tabulated below.

Solution

Octave band centre frequency (Hz)	N_1	N_2	N_3	N_4	$N_{5\&6}$	$N_{7\&8}$
63	0.13	0.18	0.30	0.36	0.80	0.80
125	0.26	0.36	0.59	0.72	1.60	1.61
250	0.52	0.71	1.17	1.44	3.21	3.21
500	1.05	1.42	2.34	2.89	6.42	6.4
1000	2.09	2.84	4.68	5.77	12.8	12.8
2000	4.19	5.68	9.35	11.5	25.7	26
4000	8.37	11.4	18.7	23.1	51.3	51
8000	16.7	22.7	37.4	46.2	103	103

The noise reductions (NR or Δ_b) corresponding to the above Fresnel Numbers are calculated using figure 8.14, p389 in the text and are tabulated below. The correction term given by equation 8.88 in the text (assuming an omnidirectional source) will be less than 0.1dB overall and will be ignored here. The numbers in brackets indicate the sum of the ground reflection losses and barrier diffraction loss. All quantities are in dB.

Octave band centre frequency (Hz)	NR_1	NR_2	NR_3	NR_4	$NR_{5\&6}$	$NR_{7\&8}$
63	8.2	8.9 (9.1)	9.9 (10.1)	10.0 (10.4)	12.3	12.3 (13.8)
125	9.5	10.1 (10.4)	11.8 (12.1)	12.0 (12.6)	15.0	15.0 (17.0)
250	10.8	12.0 (12.5)	13.8 (14.3)	14.5 (15.5)	18.1	18.1 (20.6)
500	13.0	14.8 (15.5)	17.0 (17.7)	17.7 (19.1)	21.2	21.2 (24.7)
1000	16.1	17.9 (18.9)	19.8 (20.8)	20.7 (22.7)	24	24 (29)
2000	19.2	20.9 (22.4)	23.0 (24.5)	23.8 (26.8)	24	24 (30)
4000	22.3	23.9 (26.1)	24 (26.3)	24 (28.6)	24	24 (31)
8000	24	24 (27)	24 (27)	24 (30)	24	24 (30.5)

The barrier noise reduction is given by equation 1.97 in the text and the results of the calculations are summarised in the table below. The subscript "A" refers to the condition with no barrier and the subscript "B" refers to the condition with barrier. All quantities are in dB.

Octave band centre frequency (Hz)	$10\log_{10}\sum 10^{-NR_{Ai}/10}$	$-10\log_{10}\sum 10^{-NR_{Bi}/10}$	NR	SPL at receiver
63	2.3	1.8	4.1	65.9
125	2.1	3.7	5.8	69.2
250	1.9	6.0	7.9	64.1
500	1.6	8.9	10.5	49.5
1000	1.2	12.1	13.3	44.7
2000	1.0	14.8	15.8	40.2
4000	0.8	16.7	17.5	32.5
8000	0.9	17.3	18.2	33.8

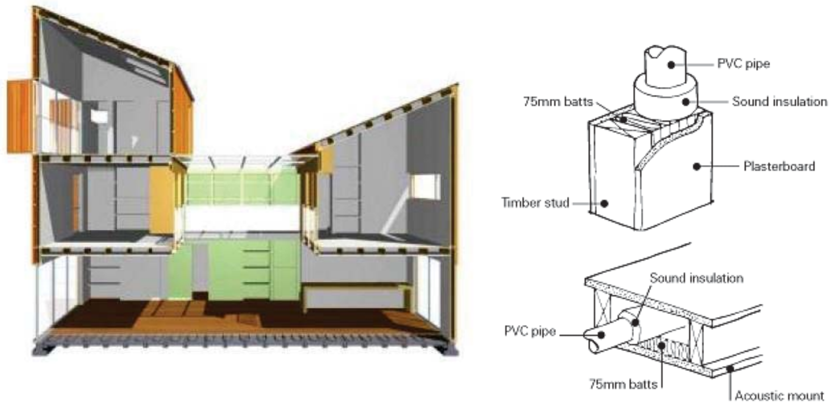
Solution

The overall A-weighted level is calculated using the octave band levels as described on pages 101 and 102 in the text. The A-weighted level with the barrier is calculated using the numbers in the last column of the preceding table and is 58.2dB(A). The A-weighted level without the barrier is calculated using the octave band levels given in the problem and is 67dB(A). Thus the noise reduction due to the barrier is 9dB(A).

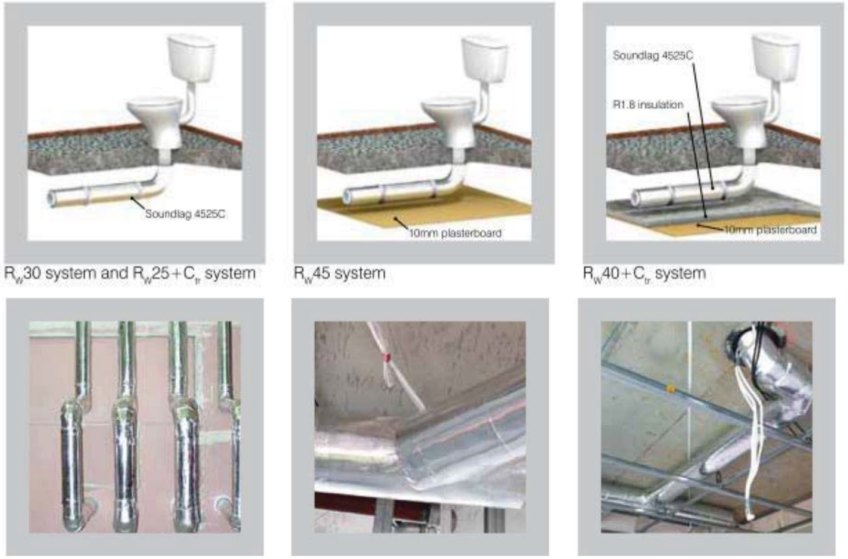
Pipe Wrappings

Pipe Wrappings

- Sound insulation for waste pipes

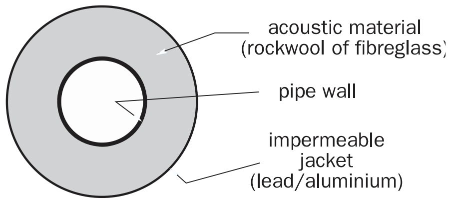


Pipe Wrappings



Pipe Lagging Insertion Loss Prediction

See p429 in textbook



- calculate f_c and f_r

  $f_c = 0.53c^2 / c_L h$ (critical frequency)

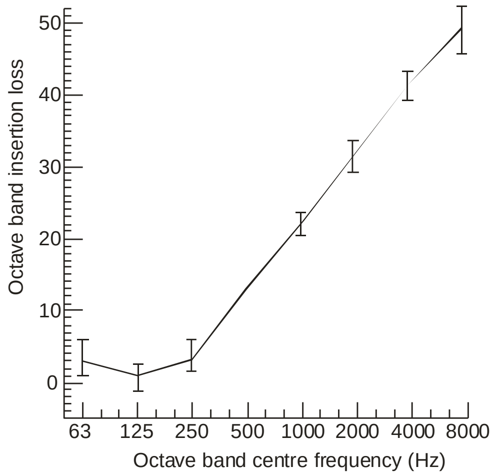
  $f_r = c_L / \pi d$ (ring frequency)

- calculate insertion loss over 3 frequency ranges
  - low: $f < f_r$
  - mid: $f_r < f < f_c$
  - high: $f > f_c$

  (see text for appropriate equations)

Pipe Lagging

Typical pipe lagging insertion loss for 50 mm glass fibre, density 70 - 90 kg/m² with a lead aluminium jacket of 6 kg/m² surface density. The I symbols represent experimental variation for 3 pipe diameters (75, 150 and 300 mm).

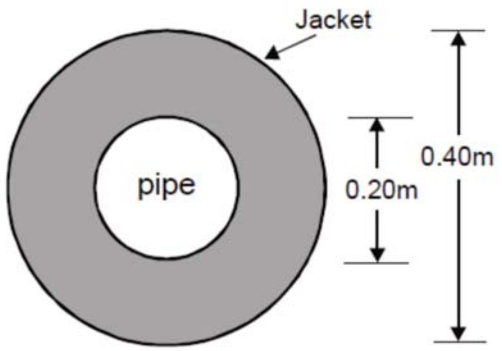


Example

- 8.28 Calculate the insertion loss in octave bands between 63Hz and 8kHz due to wrapping a 200mm diameter steel pipe with a 100mm layer of 90kgm^{-3} glass fibre covered with an aluminium jacket weighing 6kgm^{-2}.

Solution

- Follow the procedure on pages 429 and 430 in the text.
- Step 1: Draw a diagram



Solution

- Step 2:

- The thickness of the jacket is h = 6/2700 = 2.22mm and the diameter of the jacket, d = 0.2 + 2 × 0.1 + 0.0022 = 0.4022m.
- The quantity, $1000(m/d)^{1/2} = 3862$.
- The longitudinal wavespeed is:

$$c_L = \sqrt{E/[\rho_m(1-\nu^2)]} = \sqrt{71.6/[2700(1-0.34^2)]} = 5476 \text{ m/s}$$

Solution

- Calculate the ring frequency p430

The ring frequency is:

$$f_r = \frac{c_L}{\pi d} = \frac{5476}{\pi \times 0.4022} = 4{,}334\,\text{Hz}$$

- The critical frequency is obtained using the equation on p355 Eq. 8.4 as:

$$f_c = \frac{0.55\,c^2}{c_L h} = \frac{0.55 \times 343^2}{5476 \times 0.00222} = 5{,}323\,\text{Hz}$$

Solution

- For the Insertion Loss calculations, we use equation 8.136 on p430 for octave bands of 4000Hz and below and equation 8.139, p430 for the 8000Hz octave band.
- The results of the calculations are summarised in the table below at octave band centre frequencies. Of course, if three 1/3 octave bands are averaged, then the result would be slightly different.
- In addition, equation 8.144 has also been used to generate an alternative set of Insertion Loss predictions.

Solution

- With equation 8.144 p431, the Insertion Loss is given by:

$$IL = \frac{40}{1 + 0.12/0.2}\log_{10}\left(\frac{f\sqrt{6\times 0.1}}{132}\right) = 25\log_{10}(f\times 0.00587)$$

- which is valid for frequencies defined by

$$f \geq 120/\sqrt{6\times 0.1} = 155\,\text{Hz}.$$

Solution

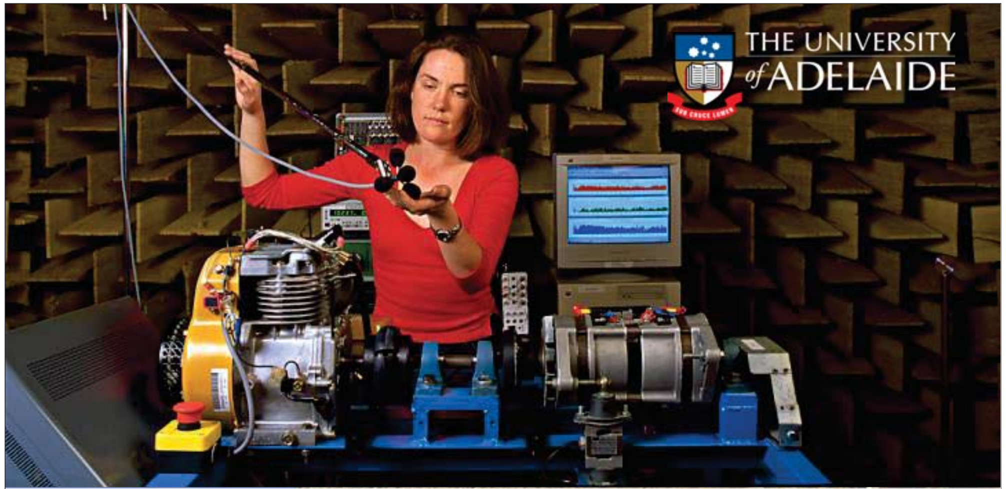
Octave band centre frequency (Hz)	f/f_r or f/f_c	C_r or C_c	X_r or X_c	Insertion Loss eqns. 8.112 - 8.119	Insertion Loss eqn.8.120
63	0.0145	0.1090	789.5	-18.8	-
125	0.0288	0.2163	1307	7.6	-
250	0.0577	0.4326	1731	17.9	4.2
500	0.115	0.8653	2120	25.5	11.7
1000	0.231	1.7305	2479	29.4	19.2
2000	0.461	3.4610	2717	19.4	26.7
4000	0.923	6.9220	1991	22.6	34.3
8000	1.503	15.849	4289	15.9	41.8

Solution

- The Insertion Loss results are somewhat different between the two methods of calculation.
- Neither prediction scheme is particularly good! However, it is supposed that the true results lie somewhere between the two.

Summary

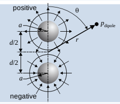
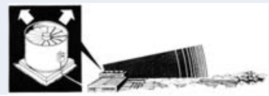
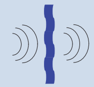
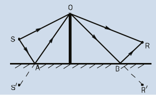
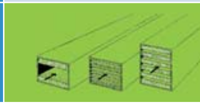
- Theory for barrier design, NR and SPL predictions.
- Example of barrier calculation.
- Theory for IL of Pipe lagging.
- Example of pipe lagging problem.



Carl Howard

Mufflers

Course Content – How it fits

TOPIC	
Sound Sources	
Outdoor	
Transmission Loss of Panels	
Barriers	
Mufflers	

Contents

- Examples of silencers
- How do they work?
- Classification: reactive vs dissipative
- Noise Reduction NR, Transmission Loss TL, Insertion Loss IL
- Acoustic impedance analysis of reactive mufflers
- Lined duct silencers
- Pressure drop calculations
- Exhaust stack directivity

Learning Outcomes

Be able to:
- Differentiate between IL, TL, NR
- Classify a muffler as reactive, dissipative, or combination
- Estimate the IL, TL, NR reactive and dissipative mufflers

References

- Engineering Noise Control (2009), Ch. 9, p432
- Munjal (2014), Acoustics of Ducts and Mufflers
- Jacobsen Moller-Juhl, Fundamentals of General Linear Acoustics, Ch. 7 Duct acoustics, p. 75
- Ver and Beranek, Noise and Vibration Control Engineering, Ch. 9, p. 279
- Physclips: http://www.animations.physics.unsw.edu.au/jw/sound-impedance-intensity.htm

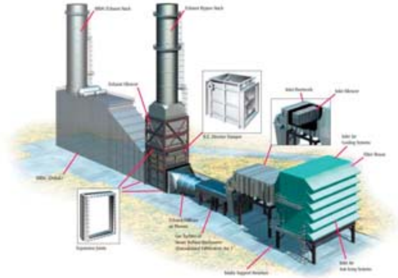
Automotive

- Mufflers

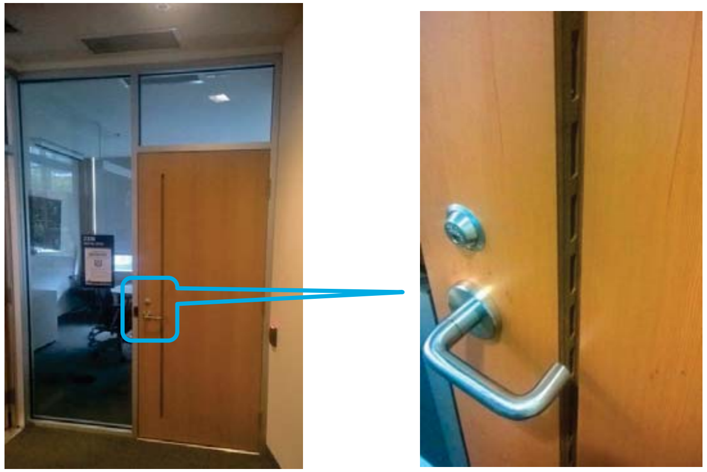
Automotive

- Intake silencers

Industrial

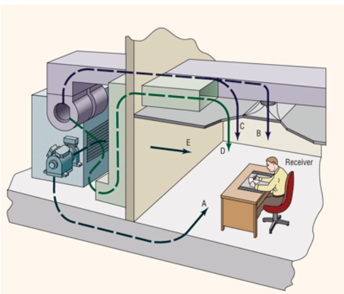


Architectural



Architectural

Air-Conditioning



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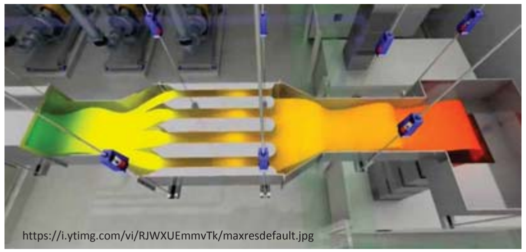

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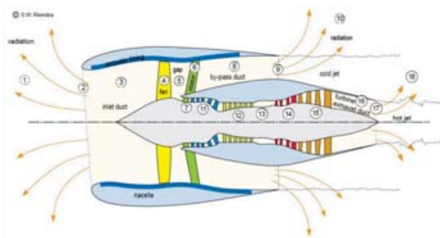
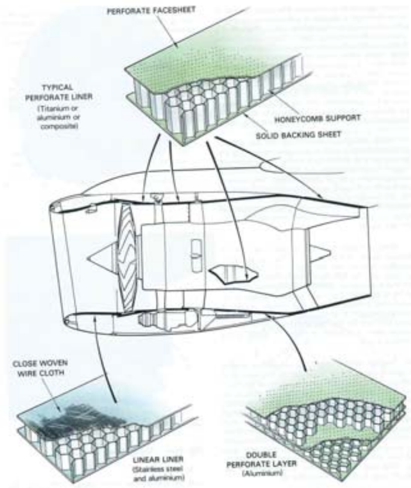
Aerospace

- Boeing 787 Dreamliner

http://www.youtube.com/watch?v=GZRKm6PG918





https://engineering.purdue.edu/~propulsi/propulsion/jets/basics/noise.html

Aerospace

- Shuttle Launch Pad - water injection
  (described in textbook, but not covered in this course)

From lecture on Noise Control

Application

- Equipment room West of The Hub

Research Facilities

- KTH University Stockholm, Sweden

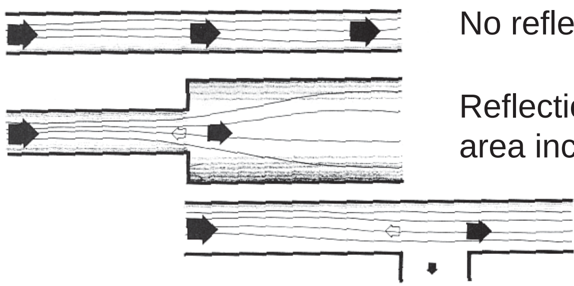
Muffler testing facility with flow

Water-cooled microphones for measuring acoustic impedance of turbo-chargers (600°C !)

From lecture on Noise Control

All duct changes reduce sound transmission

- Every change in the pathway causes some sound to be reflected back. A muffler that reflects sound energy back to the source is called a reactive muffler.One that converts sound into heat is a dissipative muffler.



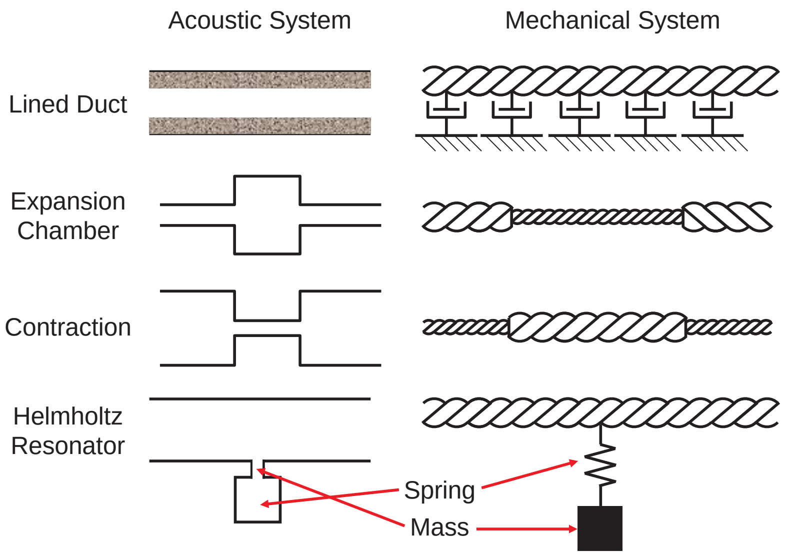
No reflections

Reflection when area increases

Reflection at branch in duct

From lecture on Noise Control

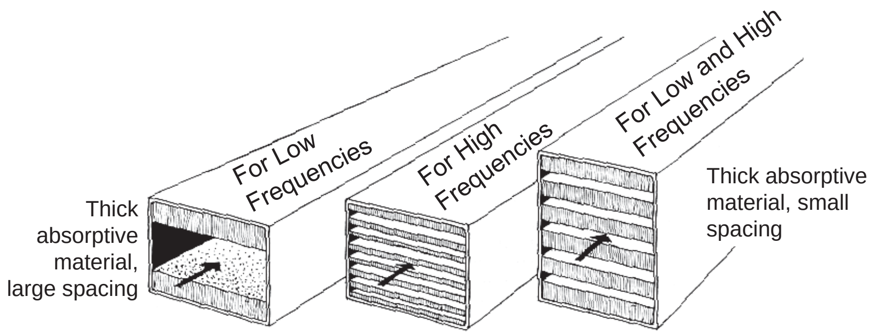
How Silencers Work



From lecture on Noise Control

Dissipative mufflers are effective over a broad range of frequencies

- The thickness and spacing of absorptive material effects the frequencies that are absorbed. The thicker the material, the lower the frequency that is absorbed. Smaller spacing is used for higher frequencies.



Classification of Muffling Devices

ENC p. 434

- REACTIVE
  - expansion chambers and small holes
  - suppress noise generation by modifying impedance seen by source
  - low frequency devices
- DISSIPATIVE
  - Uses sound absorbing material (fuzz)
  - Attenuates mid- to high- frequency noise
- Combination of dissipative and reactive

Classification of Muffling Devices

ENC p. 436

Table 9.1 Classification of muffling devices. f, c, λ and ℓ are respectively frequency, speed of sound, wavelength of sound, and critical dimension of the device

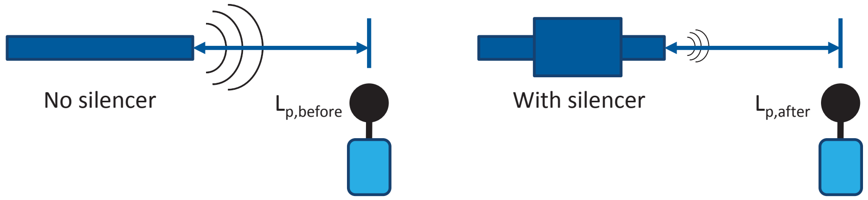
Device	Mechanism	Effective frequency range	Critical dimensions $D = f\ell/c = \ell/\lambda$ length	width	Dependence of performance on end conditions
1. Lumped element	Suppressive	Band	$D < 1/8$	$D < 1/8$	Critical
2. Side branch resonator	Suppressive	Narrow band	$D \leq 1/4$	$D < 1/8$	Critical
3. Transmission line	Suppressive	Multiple bands	$D > 1/8$	$D < 1/4$	Critical
4. Lined duct	Dissipative	Broadband	Unbounded[a]		Slightly dependent
5. Lined bend	Dissipative	Broadband	$D > 1/2$	$D > 1/2$	Not critical
6. Plenum chamber	Dissipative/ suppressive	Broadband	$D > 1$	$D > 1$	Not critical
7. Water injection	Dissipative	Broadband	Unbounded		Not critical

[a]Theoretically, D is unbounded, but a practical lower bound for D is about 1/4.

Measures of Performance

ENC p. 432

- Insertion Loss IL
  - defined as the reduction (in decibels) in sound power transmitted through a duct compared to that transmitted with no muffler in place.
  - Provided that the duct outlet remains at a fixed point in space, the insertion loss will be equal to the noise reduction that would be expected at a reference point external to the duct outlet as a result of installing the muffler.



Measures of Performance

ENC p. 432

- Insertion Loss IL
  - same as noise reduction at a location near duct exit, provided muffler does not cause exit to move.
  - defined as difference in sound pressure at a location in the duct downstream from the muffler before and after insertion of the muffler.
  - used to describe performance of reactive devices whose performance is dependent on impedances of source and duct termination.

Measures of Performance

- Insertion Loss IL

$$\boxed{\mathrm{IL} = L_{w:before} - L_{w:after} \ \text{(dB)}}$$

$L_{w:before}$= sound **power** level at distance from exit without muffler (dB re 1pW)
$L_{w:after}$ = sound **power** level at distance from exit with muffler (dB re 1pW)

$$\boxed{\mathrm{IL} = L_{p,before} - L_{p,after} = 20\log_{10}\left|\frac{p_b}{p_a}\right| \ \text{(dB)}}$$

$L_{p,before}$= sound **pressure** level at distance from exit without muffler, before installation (dB re 20µPa)
$L_{p,after}$ = sound **pressure** level at distance from exit with muffler, after installation (dB re 20µPa)

Measures of Performance

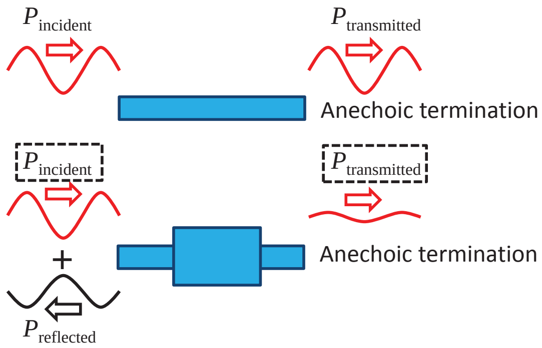
ENC p. 433

- Transmission Loss (TL)
  - Defined as the difference (in decibels) between the sound **power** *incident* at the entry to the muffler to that *transmitted* by the muffler.
  - The transmission loss is **NOT** equal to the Noise Reduction or the Insertion Loss, as the muffler can affect the sound power radiated by the sound source (and thus the sound power incident on the muffler), in addition to the transmitted sound power.
  - .... This requires a bit of explanation.....

Transmission Loss TL

- Straight duct
  - Waves unaffected
- Duct with muffler
  - Sound is reflected at impedance change

Remember this...
I will demonstrate it



- TL is defined as difference in incident and transmitted *sound power level*
  (ratio of sound power incident on muffler to that transmitted by muffler.)

Transmission Loss TL

- TRANSMISSION LOSS (*TL*)
  - TL can be used to describe *dissipative* mufflers for which IL ≈ TL provided TL>5 dB as these do not affect source radiated power.

$$\boxed{\mathrm{TL} = L_{w:incident} - L_{w:transmitted} \quad (\mathrm{dB})}$$

$L_{w:incident}$ = sound power incident on muffler (dB re 1pW)
$L_{w:transmitted}$ = sound power transmitted at exit of muffler (dB re 1pW)

Transmission Loss TL

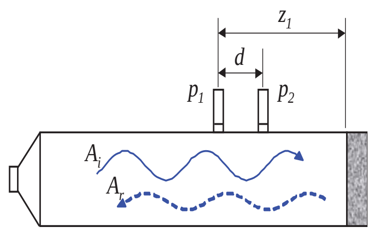
For reference only, not covered in course

- Question: How can one separate the incident and reflected wave power as a microphone will only measure the total sound pressure?
- Answer: A microphone placed upstream would measure the total pressure (incident + reflected). Hence, need a method to separate incident and reflected sound waves.... Use the 2- or 4- microphone method (Not covered in this course)

Two Microphone Method

For reference only, not covered in course

- Described in standard:
  ASTM E1050 (1998). Standard test method for impedance and absorption of acoustical materials using a tube, two microphones, and a digital frequency analysis system.
- Which comes from:
  Chung, Y. J. and D. A. Blaser (1980a). "Transfer function method of measuring in-duct acoustic properties. I. Theory". In: Journal of the Acoustical Society of America 68.3, pp. 907–913.
  — (1980b). "Transfer function method of measuring in-duct acoustic properties. II. Experiment". In: Journal of the Acoustical Society of America 68.3, pp. 914–921.

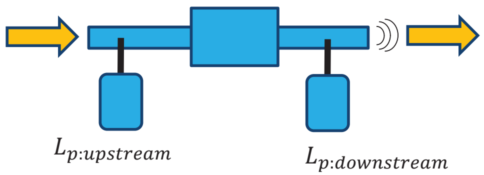


Measures of Performance

- Noise Reduction NR (or Level Difference LD)
  - Difference in SPL measured upstream and downstream of the silencer.

$$\boxed{\mathrm{NR} = \mathrm{LD} = = L_{p:upstream} - L_{p:downstream} \quad \text{(dB)}}$$

$L_{p:upstream}$ = sound pressure level upstream of muffler (dB re 20μPa)
$L_{p:downstream}$ = sound pressure level downstream of muffler (dB re 20μPa)



Noise Reduction NR

- Noise Reduction NR (or Level Difference LD)
  - The measurement of NR does not require anechoic duct terminations and does not require knowledge of the source impedance.
  - It is the easiest of the three metric to measure in practice.

Measures of Performance

- The performance of reactive devices is dependent on the impedances of the source and termination (outlet). In general, a reactive device will strongly affect the generation of sound at the source. This has the effect that the TL and IL of reactive devices may be very different.
- The performance of dissipative devices, on the other hand, by the very nature of the mode of operation, tends to be independent of the effects of source and termination impedance.
- Provided that the TL of a dissipative muffler is at least 5 dB, it may be assumed that the IL and the TL are the same.

From lecture on Fundamentals

Impedance

- MECHANICAL IMPEDANCE $\quad Z_m = \dfrac{F}{u} = \dfrac{pA}{u}$

  used to describe sound radiation from a vibrating structure

- SPECIFIC ACOUSTIC IMPEDANCE $\quad Z_s = \dfrac{p}{u}$

  used to describe propagation of sound in free space, reflection and transmission of sound at interfaces between media.

- ACOUSTIC IMPEDANCE $\quad Z_a = \dfrac{p}{uS} = \dfrac{p}{v}$

  used to describe propagation in ducts and mufflers.

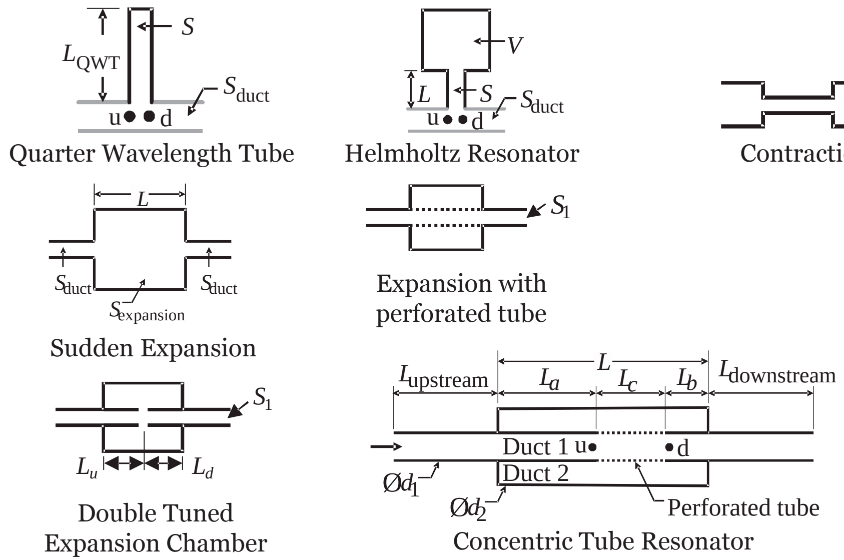
ENC p. 53

Impedance

- where

F = force (N, Units MLT^{-2})
u = acoustic particle velocity (m/s, Units LT^{-1})
p = pressure (Pa, Units $MT^{-2}L^{-1}$)
v = acoustic volume velocity (m³/s, Units L^3T^{-1})
S = area (m², Units L^2)

Reactive Muffler Examples



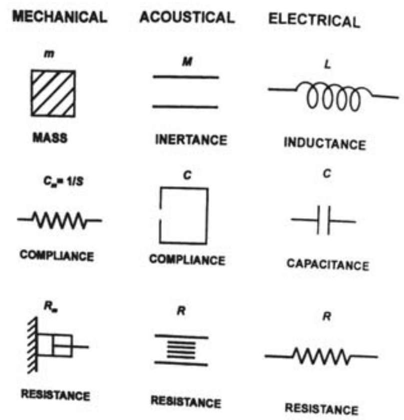
Reactive Mufflers: Analysis

- Two common methods
  - Impedance method
  - 4-pole transmission line (not covered in this course, see Munjal [2014] Acoustics of Ducts and Mufflers, ENC 5th ed)

Reactive Mufflers: Analysis

- Acoustical to Electrical Analogies

Acoustical	Electrical
Pressure	Voltage
Volume velocity	Current
Mass	Inductance
Compliance	Capacitance
Resistance	Resistance
Orifice or Narrow Duct	Inductance + Resistance
Expansion Chamber	Capacitance

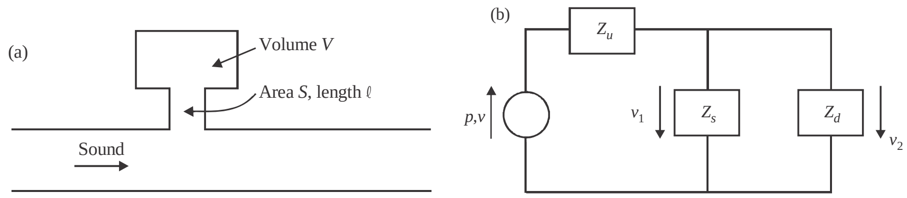


Reactive Mufflers: Analysis

- Acoustic Analogies of Kirchhoff's Laws ENC p. 447
  - Acoustic volume velocity inflow at a point must equal the outflow.
  - Acoustic pressure drops around any closed loop must add up to zero.
- Hence one can draw an "electrical" circuit diagram of a duct with a muffler.

Reactive Mufflers: Analysis

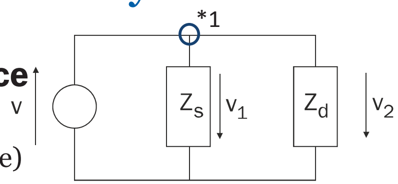
- Example of an "electrical" circuit analogy of an (side-branch) acoustic muffler. ENC Fig 9.7, p. 448



Z_u = upstream duct impedance (Pa s / m^3)
Z_d = downstream duct impedance (Pa s / m^3)
Z_s = side-branch duct impedance (Pa s / m^3)
v = acoustic volume velocity (m^3/s, Units L^3T^{-1})
p = pressure (Pa, Units $MT^{-2}L^{-1}$)

Resonator Insertion Loss Analysis

- **Constant volume velocity source**
  - Speaker with small air-tight backing
  - Reciprocating compressor (approximate)



- Using Kirchhoff's "rules" ENC p. 451
  - At point *1, Acoustic volume velocity inflow at a point must equal the outflow.

$$v = v_1 + v_2$$

  - Acoustic pressure drops around any closed loop must add up to zero.

$$v_1 Z_s = v_2 Z_d$$

$$p = v_2 Z_d + (v_1 + v_2) Z_u$$

Resonator Insertion Loss Analysis

- Insertion Loss (IL)

$$\mathrm{IL} = 10\log_{10}\left[\frac{\text{acoustic power flow through system into load before insertion of device}}{\text{acoustic power flow after insertion of device}}\right]$$

$$IL = 20\log_{10}\left|\frac{v}{v_2}\right| = 20\log_{10}\left|1 + \frac{Z_d}{Z_S}\right| \quad \text{(dB)}$$

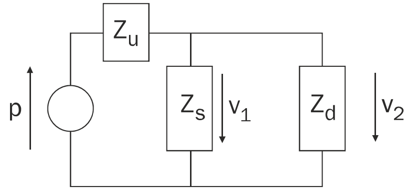
ENC p. 452

* for a constant volume velocity source

Z_d = downstream duct impedance (Pa s / m^3)
Z_s = side-branch duct impedance (Pa s / m^3)
v = acoustic volume velocity (m^3/s, Units L^3T^{-1})

Resonator Insertion Loss Analysis

- **Constant pressure source**
  - Fan, centrifugal pump



- Using Kirchhoff's "rules"

$$p = v_2 Z_d + Z_u (v_1 + v_2)$$
$$v_1 Z_s = v_2 Z_d$$
$$p = v_2 \left[Z_d + Z_u + \frac{Z_d Z_u}{Z_s} \right]$$

- Insertion Loss

$$IL = 20 \log_{10} \left| \frac{Z_s Z_d + Z_s Z_u + Z_d Z_u}{Z_s (Z_d + Z_u)} \right| = 20 \log_{10} \left| 1 + \frac{Z_d Z_u}{Z_s (Z_d + Z_u)} \right|$$

ENC p. 452
Eq. 9.57

Transmission Loss Due to Side-Branch

ENC p. 453

- TL is defined as

$$\boxed{\mathrm{TL} = 10 \log_{10} \left[\frac{\text{power flow into device}}{\text{power flow out of device}} \right] \quad \text{(dB)}}$$

- When considering only plane-waves in a duct, TL is

$$\boxed{TL = 10 \log_{10} \left| \frac{p_i}{p_t} \right|^2 \quad \text{(dB)}}$$

p_i = incident pressure (Pa, Units $MT^{-2}L^{-1}$)
p_t = transmitted pressure (Pa, Units $MT^{-2}L^{-1}$)

Transmission Loss Due to Side-Branch

- When considering only plane-waves in a duct, TL of side-branch is

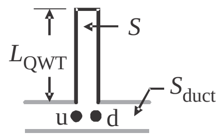
$$\boxed{TL = 20 \log_{10} \left| 1 + \frac{\rho c}{2 S_d Z_s} \right| \quad \text{(dB)}}$$

ENC p. 455

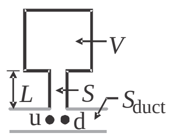
ρ = density of gas (kg / m³)
c = speed of sound (m / s)
S_d = cross-sectional area of downstream duct (m²)
Z_s = side-branch acoustic impedance (Pa s / m³)

Side-Branch Resonator

- Examples of side-branch resonator types:

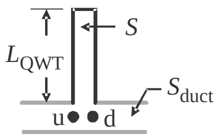


Quarter Wavelength Tube



Helmholtz Resonator

Quarter-Wavelength Tube



Description	Parameter	Value	Units
Diameter main duct	S_{duct}	0.1	m
Diameter QWT	S	0.05	m
Length QWT	L_{QWT}	1.5	m
Speed of sound	c	343.24	m/s
Density	ρ	1.2041	kg/m^3
Velocity at inlet	u_1	0.001	m/s

- Question:
  At what frequency would you expect the maximum attenuation to occur for the QUARTER WAVELENGTH tube?
  HINT: Look at lecture on fundamentals for formula.

Quarter-Wavelength Tube

- QWT (side-branch) impedance is

$$\boxed{Z_S = -\frac{j\rho c}{S}\cot k\ell_e + R_S}$$

ENC p. 447

ρ = density of gas (kg / m^3)
c = speed of sound (m / s)
k = wavenumber ($= 2\pi f / c$)
S = cross-sectional area of QWT (m^2)
ℓ_e = effective length of QWT (m)
R_S = acoustic resistance

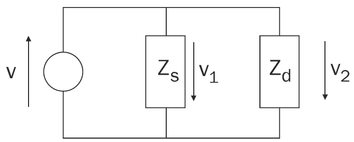
- When the effective length of the QWT is $\ell_e = \lambda/4$

  $\boxed{Z_s = 0 + R_S}$

  Huh ??? Impedance is zero ? How can that work ?

Quarter-Wavelength Tube

- Look at the circuit diagram again.

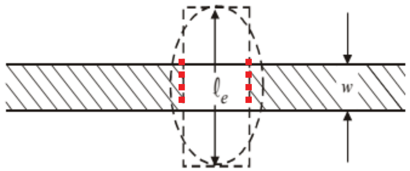


- Say for the moment that there was some impedance (e.g. electrical resistance), then some 'current' would flow down the branches v_1 and v_2.

- Current tends to flow along the path of least resistance (think of a lightning strike), so if $Z_s = 0$ then most of the 'current' would by-pass v_2, which effectively means no 'current' (sound) would go into Z_d (the downstream duct).

Acoustic Resistance

- Consider a blob of air moving through a small orifice



- As the air oscillates through the hole there will be friction with the perimeter and a complicated flow pattern.
- This is an acoustic loss mechanism and is one cause of acoustic damping.

Acoustic Resistance

- For an orifice or tube of length w

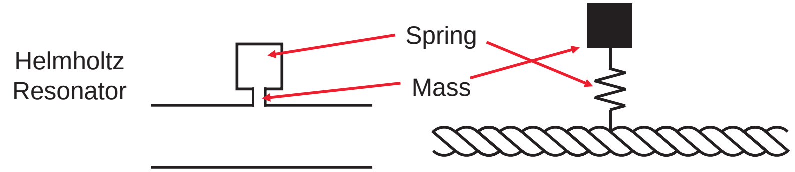
ENC p. 443

$$R_S = \frac{\rho c}{S}\left[\frac{ktDw}{2S}\left[1+(\gamma-1)\sqrt{\frac{5}{3\gamma}}\right]+0.288kt\log_{10}\left[\frac{4S}{\pi h^2}\right]+\varepsilon\frac{Sk^2}{2\pi}+M\right]$$

ρ = density of gas (kg / m^3)
w = tube length (m)
S = tube cross-sectional area (m^2)
γ = ratio specific heats for air
D = tube cross-sectional perimeter (m)
k = wavenumber ($= \omega/c = 2\pi f/c$)
M = Mach number
h = largest of t or $L/2$ (thin plate)
= largest of t or tube edge radius (thick plate or tube)
ε = 0 into chamber of dimensions < λ
= 0.5 into free space, no flange
= 1.0 into free space with flange

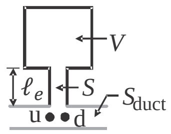
t = viscous boundary layer thickness
$= \sqrt{2\mu/(\rho\omega)}$
μ = dynamic viscosity
1.84x10^{-5} kg/ms for air at 20°C
ω = circular frequency ($= 2\pi f$)

Helmholtz Resonator



- The volume of gas acts like an acoustic ‘spring’.
- The blob of gas in the neck acts like a ‘mass’.
- One can tune the resonance frequency by adjusting the dimensions to alter the acoustic ‘stiffness’ and ‘mass’.
- Impedance is

$$Z_S = \frac{\mathrm{j}\rho\omega\ell_e}{S} - \frac{\mathrm{j}\rho c^2}{V\omega} + R_S$$



ENC p. 448

Helmholtz Resonator

- Resonance Frequency $\omega_0 = c\sqrt{S/\ell_e V}$

ENC p. 448

- A more accurate expression for a cylindrical resonator is

$$\omega_0 = c\sqrt{-\frac{3\ell_e S_V + \ell_V S}{2\ell_e^3 S_V} + \sqrt{\left(\frac{3\ell_e S_V + \ell_V S}{2\ell_e^3 S_V}\right)^2 + \frac{3S}{\ell_e^3 S_V \ell_V}}}$$

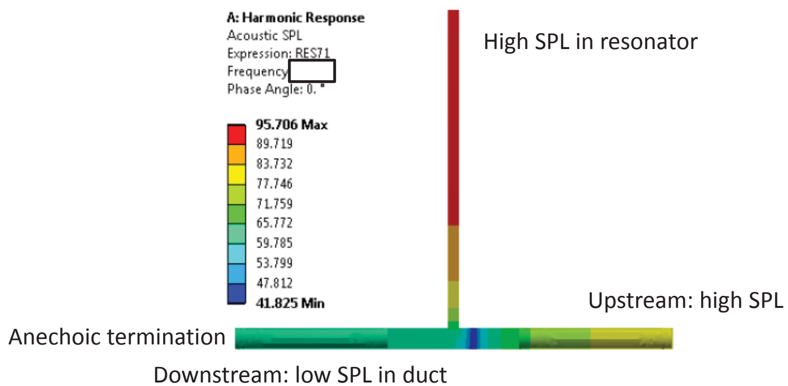
ENC p. 449

c = speed of sound (m/s)
S = neck cross-sectional area (m^2)
S_V = volume cross-sectional area (m^2)
ℓ_e = effective length of neck (m)
ℓ_V = length of volume (m)

Quarter-Wavelength Tube

- Complicated configurations can be analysed using FEA Finite Element Analysis

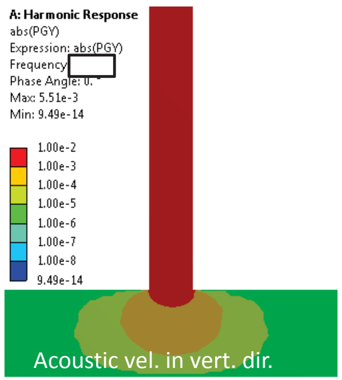
Howard and Cazzolato (2014), Acoustic Analyses Using Matlab and Ansys, p. 183

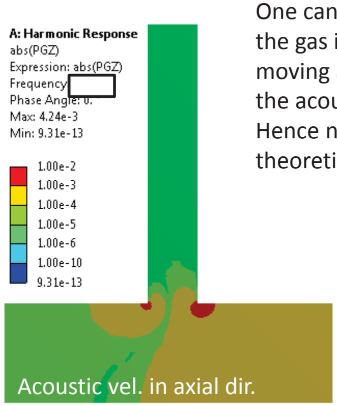


Quarter-Wavelength Tube

- Zooming in around neck of QWT and looking at acoustic particle velocity in the vertical Y and axial Z directions.

Howard and Cazzolato (2014), Acoustic Analyses Using Matlab and Ansys, p. 184





One can see that some of the gas in the main duct is moving and participating in the acoustics.
Hence need to adjust the theoretical length of QWT.

End Correction for QWT and HR

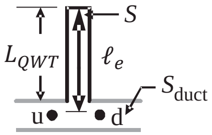
- QWT effective length $\ell_{e:QWT} = L_{QWT} + \ell_0$

$$\ell_0 = a[0.8216 - 0.0644\zeta - 0.694\zeta^2] \quad \text{for } \zeta \le 0.4$$

$$\ell_0 = a[0.9326 - 0.6196\zeta] \quad \text{for } \zeta > 0.4$$

L_{QWT} = physical length of QWT (m)
ℓ_0 = additional length (m)
a = radius of neck of QWT (m)
ζ = ratio of neck diameter to main duct diam.

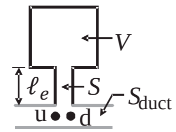
ENC p. 449



- HR effective length has two corrections:
  - neck-to-cavity:

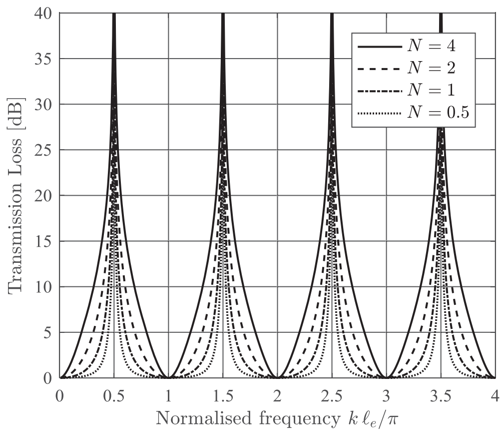
$$\ell_0 = 0.82a[1 - 1.33\zeta] \quad \text{for } \zeta < 0.4$$

ENC p. 449



  - neck-to-duct: as for QWT

Quarter-Wavelength Tube

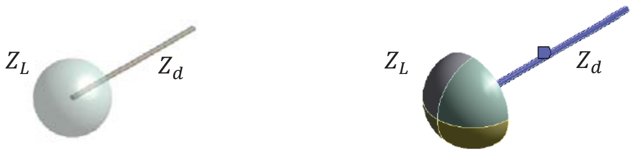


Transmission loss versus normalised frequency, $k\ell_e/\pi$, of a quarter-wavelength tube silencer for a range of area ratios of the quarter-wavelength tube to main duct N = S/S_{duct} = 0.5, 1, 2, 4.

Figure to appear in 5th Ed of ENC

Termination Impedance

- So far we have assumed that the downstream duct is anechoic. This rarely happens in practice.
- The end of the duct is often radiating into a free-field or a hemispherical free-field (baffled).



- The downstream duct impedance Z_d needs to include the termination (end) impedance, often called the load Z_L

Termination Impedance

- long pipe downstream $L_d \gg \lambda$ (no flow), is near anechoic

$$Z_d + Z_L = \frac{\rho c}{S}$$

- short pipe, length $\ell_e < \lambda/4$ (no flow)

$$Z_L = \frac{j\rho\omega\ell_e}{S} + R_S$$

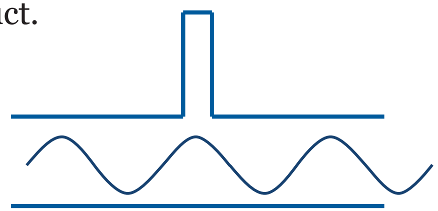
- in between long and short $L_d > \lambda/4$ (no flow)

$$Z_L = \frac{j\rho\omega\ell_e}{S}\tan(k\ell_e) + R_S$$

Optimum Location of Resonator

- Place the side-branch resonator at a pressure maximum in the duct.

ENC p. 452



Q: What would happen if the QWT were placed at a pressure node (zero) ?

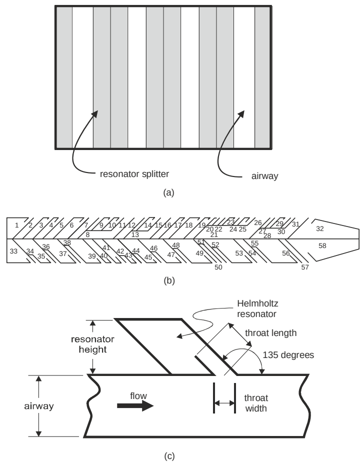
- where Z_d is maximum:
  - odd multiple of $\lambda/4$ from end of duct

$$\ell_e = \frac{2n-1}{4}\lambda$$

- minimum of 3 duct diameters from noise source

Resonator Silencers

- Multiple Helmholtz and Quarter-Wavelength resonator silencers of varying dimensions can be arranged to attenuate a range of frequencies.
- Because there is no "fluff" the acoustic performance does not degrade over time.



Exhaust stack silencer design using finite element analysis , Carl Q. Howard, Ben S. Cazzolato, Colin H. Hansen (2000), Noise Control Engineering Journal, **48** (4), p113-120.

From lecture on Noise Control

Industrial

- Reactive Silencer for Power Station

Exhaust stack silencer design using finite element analysis , Carl Q. Howard, Ben S. Cazzolato, Colin H. Hansen (2000), Noise Control Engineering Journal, **48** (4), p113-120.

Resonator Silencers

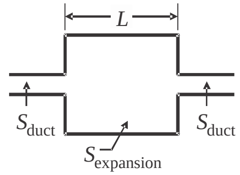
- Two types
  - no sound absorbing material
  - sound absorbing material in each resonator

- various size resonators allow a good frequency range to be covered
- construction material must be sufficiently thick to avoid component resonances and sound transmission through the walls
- Usually designed using a commercial FEA package with acoustic elements
- Each resonator element has multiple resonances, each of which can provide attenuation over a narrow frequency band about the resonance.

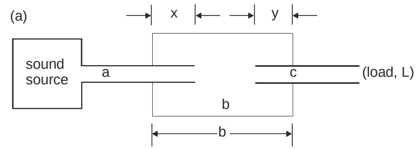
Simple Expansion Chamber Silencer

ENC p. 457

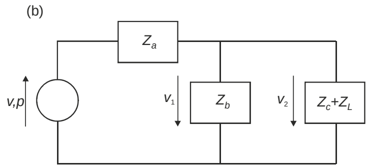
- Very common muffler is simple expansion chamber.
- Assume $L < \lambda/2$



- General case looks like... where lengths $x = y = 0$



- Equivalent electrical circuit Now include load Z_L



Simple Expansion Chamber Silencer

- constant volume velocity source

$$\boxed{IL = 20\log_{10}\left|1+\frac{Z_c+Z_L}{Z_b}\right|}$$

ENC p. 459

- constant pressure source

$$\boxed{\begin{aligned} IL &= 20\log_{10}\left|\frac{p}{V_2 Z_L}\right| \\ &= 20\log_{10}\left|1+\frac{Z_a(Z_c+Z_L+Z_b)+Z_b Z_c}{Z_b Z_L}\right| \end{aligned}}$$

ENC p. 461

Simple Expansion Chamber Silencer

- short tail pipe, length $\ell_e < \lambda/4$
  - constant volume velocity source

$$\frac{Z_c + Z_L}{Z_b} = j\frac{\omega V}{\rho c^2}\left[R_c + R_L + j\frac{\rho c}{A}\tan k\ell_e\right]$$

$$Z_b = \frac{j\rho c^2}{\omega V}$$

$$\omega_0 = c\sqrt{A/\ell_e V}$$

$$Q = \frac{\rho c}{R_A}\sqrt{\ell_e/AV}$$

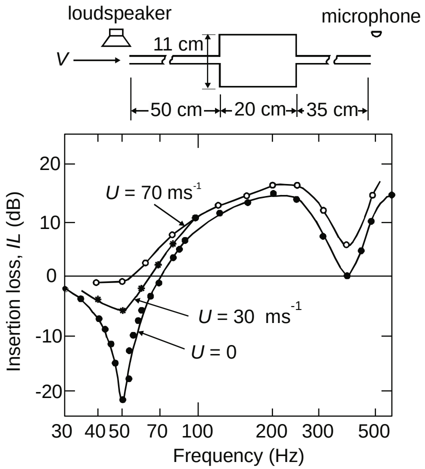
$$\frac{Z_c + Z_L}{Z_b} = -\left[\frac{\omega}{\omega_0}\right]^2 + j\frac{\omega}{\omega_0 Q}$$

$$IL = 10\log_{10}\left[\left(1-(\omega/\omega_0)^2\right)^2 + Q^{-2}(\omega/\omega_0)^2\right]$$

$$IL/_{\omega=\omega_0} = -20\log_{10} Q \quad (\text{negative})$$

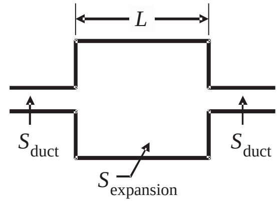
Simple Expansion Chamber Silencer

- Example of performance



Simple Expansion Chamber Silencer

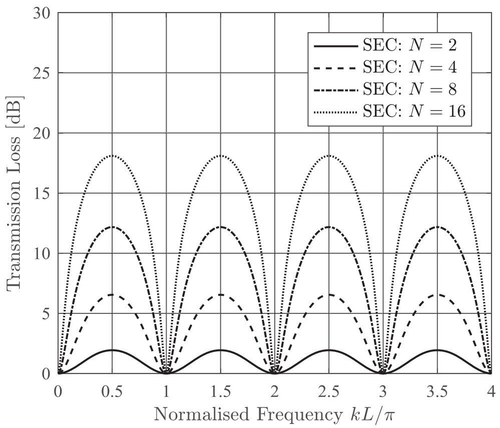
- Transmission loss
  (Assume $L < \lambda/2$)



$$\mathrm{TL} = 10 \times \log_{10}\left[1 + \frac{1}{4}\left(\frac{S_{duct}}{S_{expansion}} - \frac{S_{expansion}}{S_{duct}}\right)^2 \sin^2(kL)\right]$$

ENC p. 464

Simple Expansion Chamber Silencer



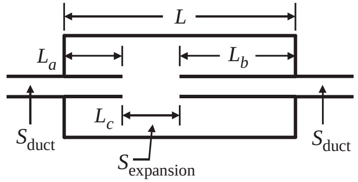
Transmission loss versus normalised frequency, kL/π, of an expansion tube silencer for a range of area ratios of the expansion chamber to main duct
N = S_{expan} / S = 2, 4, 8, 16.

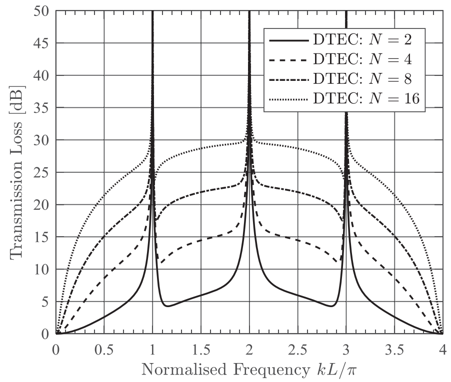
Figure to appear in 5th Ed of ENC, compare with 4th Ed ENC p 465.

Double Tuned Expansion Chamber

Not covered in this course

- There were dips at integer multiples of kL/π
- If the tubes are extended then the dips can be removed
  - Double Tuned Expansion Chamber (DTEC)

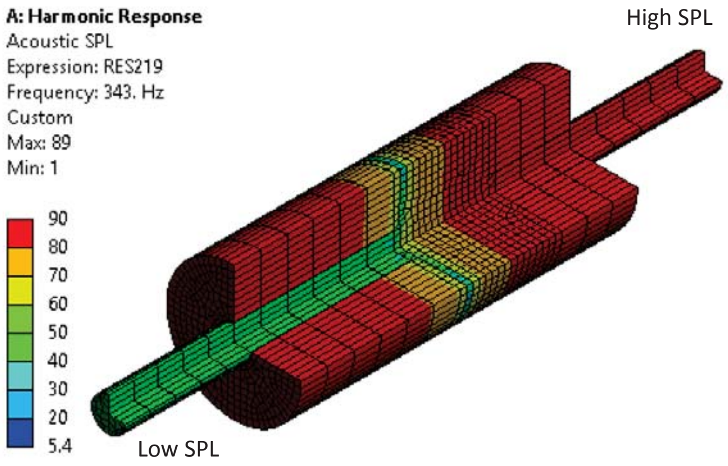




Double Tuned Expansion Chamber

Not covered in this course

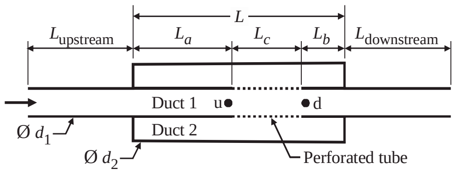
- Finite element analysis

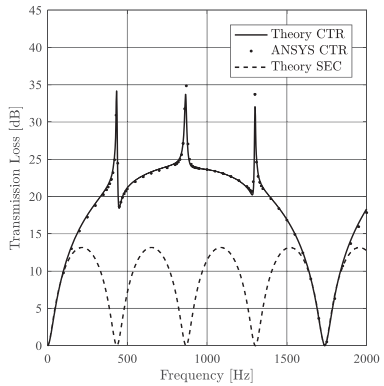


Concentric Tube Resonator

Not covered in this course

- Can be further improved by adding a perforated tube connecting upstream and downstream extension tubes





Pressure Drop Estimates

ENC p. 472

- Adding reactive muffler causes an additional back pressure:

$$\Delta P = f_m \left(\frac{LP_D}{4S} \right) \left(\frac{\rho U^2}{2} \right)$$

$$f_m = f_m' \quad \text{if } f_m' \geq 0.018$$

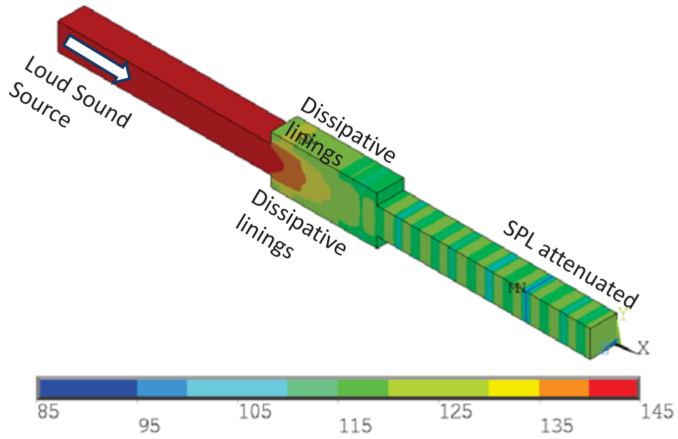
$$f_m = 0.85 f_m' + 0.0028 \quad \text{if } f_m' < 0.018$$

$$\text{where } f_m' = 0.11 \left(\frac{\epsilon P_D}{4S} + \frac{68}{\text{Re}} \right)^{0.25} \text{ for standard air}$$

$$= 0.11 \left(\frac{\epsilon P_D}{4S} + \frac{2.56 \times 10^{-4} P_D}{SU} \right)^{0.25}$$

P_D = straight pipe perimeter (m)
S = cross sectional area (m^2)
L = length (m)
U = flow speed (m/s)
ρ = gas density (kg/m^3)
ϵ = pipe roughness
1.5×10^{-4} for galvanized steel,
9×10^{-4} for fibreglass lined ducts
Re = Reynolds number of flow

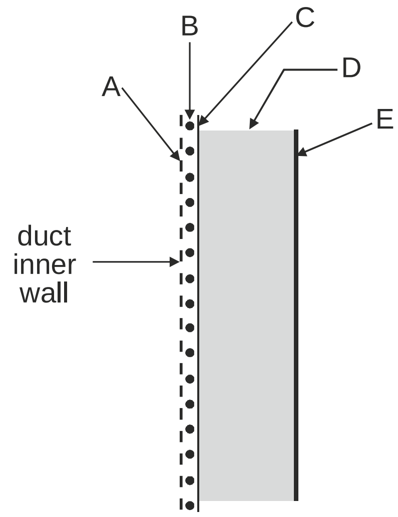
- Discontinuities $\Delta P = (1/2) \rho U^2 K$
  See text, Fig. 9.12 for values of K.



Dissipative Mufflers

Dissipative Mufflers (Lined ducts)

A =perforated sheet steel or Al
B = spacer (wire mesh)
C = plastic covering
(approx 20 μm thick)
D = rockwool or fibreglass
E = duct wall



- Extent of protection needed depends on air flow speed.
- Regularly spaced holes in perforated sheet can whistle at high flow speeds.

ENC p. 480

Duct Liner Design

σ' = mass/unit area of plastic membrane.

If a perforated sheet is added, air in holes adds more effective mass.
In this case, total effective mass σ' for use in design curves is

$$\sigma = \sigma' + 100\rho\ell_e / P$$

More accurate version including grazing flow of Mach number, M is:

$$\sigma = \sigma' + \frac{\frac{100\rho}{Pk}\tan\left(k\ell_e(1-M)\right)}{1+\frac{100\rho}{kmP}\tan\left(k\ell_e(1-M)\right)}$$

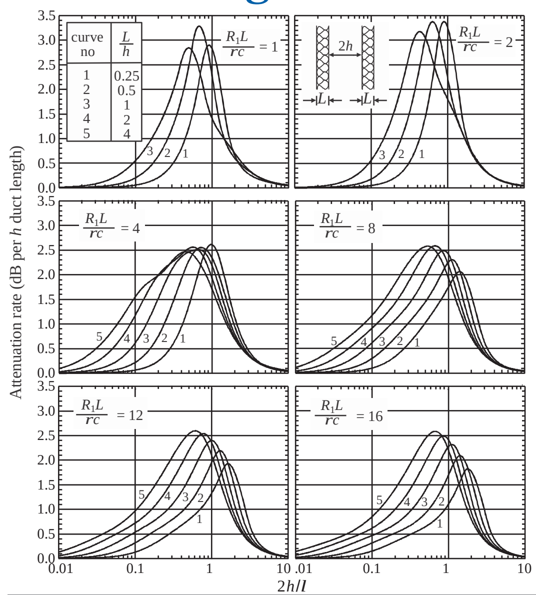
ENC p. 481

ρ = air density
ℓ_e = effective length of holes in perforated facing
k = wave number
P = percentage of open area of perforate
m = mass per unit area of panel

Design Curves

- Assume two opposite sides lined.
- For four sides lined, add results for each of the two pairs of opposite sides.
- For circular ducts, double the result for the rectangular duct with two sides lined, using the rectangular duct height equal to circular duct diameter.

Liner Design Curves



Predicted OCTAVE band attenuation rates.

Bulk reacting liner;
$\sigma/\rho h = 0.\ M = 0$

See text for additional curves

ENC p. 483

Lined Duct Design Procedure

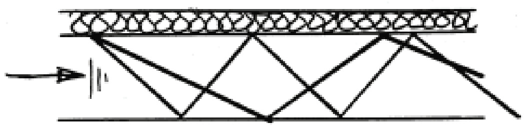
- Assumes only plane waves propagate:

$$f < f_{co} \qquad \left[f_{co} = \frac{0.5c}{d} \right]$$

f_{co} = cut-on frequency (Hz)
d = diameter of duct (m)
c = speed of sound (m/s)

  – plane wave approximation gives good results up to $f = 2f_{co}$ [i.e. $2h/\lambda \approx 1$] as plane waves carry most energy in this range.

- underestimates high frequency attenuation as higher order modes are attenuated more effectively.



Sound waves bounce into liner

Lined Duct Design Procedure

- Flow effects
  - Sound and flow in opposite directions increases attenuation and vice versa

$$D_M = D_0\left[1 - 1.5M + M^2\right]$$

ENC p. 489

D_0 = attenuation for liner without flow (dB per unit length)
D_M = attenuation for liner with plug flow of mach M (dB per unit length)
M = Mach number
c = speed of sound (m/s)

Duct Cut On Frequencies

- Higher order mode propagation

Rectangular section ducts

$$f_{co} = \frac{c}{2L_y}$$

$$f_{ny,nz} = \frac{c}{2}\sqrt{\left(\frac{n_y}{L_y}\right)^2 + \left(\frac{n_z}{L_z}\right)^2}$$

Circular section ducts,

$$f_{co} = 0.586\frac{c}{d}$$

ENC p. 490

$$f_{m,n} = \frac{c\alpha_{m,n}}{2a}$$

ENC p. 492

c = speed of sound (m/s)
L_y = largest duct cross sectional dimension (m)
d = duct diameter (m)
a = duct radius (m)
n_y, n_z = modal indices
$\alpha_{m,n}$ = coefficient from ENC Table 9.4, p. 491

Duct Cut On Frequencies

- Higher order mode propagation

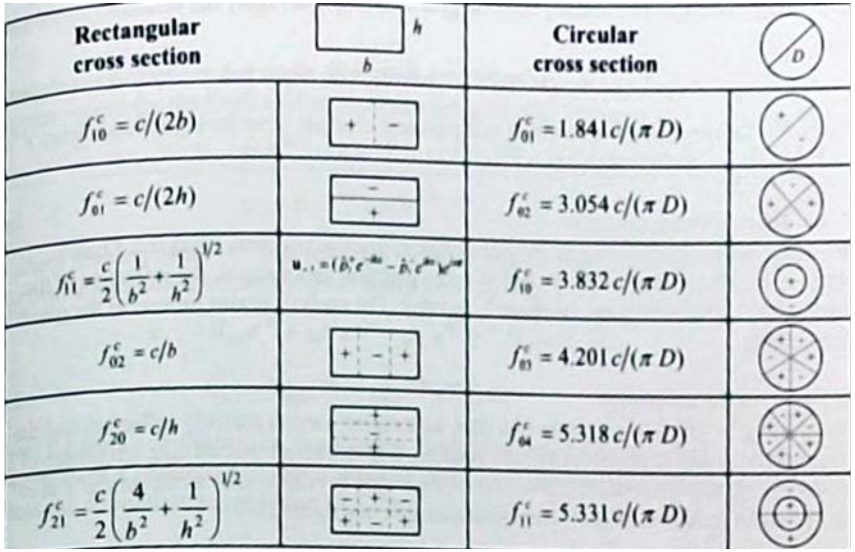
Circular section ducts,

ENC p. 491

Table 9.4 Values of the coeficient, $\alpha_{m\,n}$ for circular section ducts (after Morse and Ingard, 1968)

$m \backslash n$	0	1	2	3	4
0	0	3.83	7.02	10.17	13.32
1	1.84	5.33	8.53	11.71	14.86
2	3.05	6.71	9.97	13.17	16.35
3	4.20	8.02	11.35	14.59	17.79
4	5.32	9.28	12.68	15.96	19.2
5	6.42	10.52	13.99	17.31	20.58

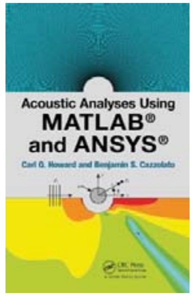
Duct Cut On Frequencies

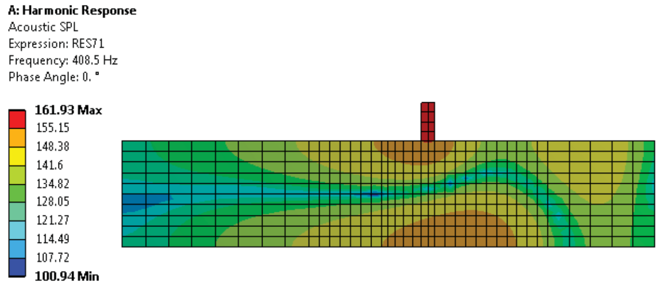


Rectangular cross section		Circular cross section	
$f_{10}^c = c/(2b)$		$f_{01}^c = 1.841c/(\pi D)$	
$f_{01}^c = c/(2h)$		$f_{02}^c = 3.054\,c/(\pi D)$	
$f_{11}^c = \frac{c}{2}\left(\frac{1}{b^2}+\frac{1}{h^2}\right)^{1/2}$	[illegible]	$f_{10}^c = 3.832\,c/(\pi D)$	
$f_{02}^c = c/b$		$f_{03}^c = 4.201\,c/(\pi D)$	
$f_{20}^c = c/h$		$f_{04}^c = 5.318\,c/(\pi D)$	
$f_{21}^c = \frac{c}{2}\left(\frac{4}{b^2}+\frac{1}{h^2}\right)^{1/2}$		$f_{11}^c = 5.331\,c/(\pi D)$	

Above Cut-On Frequency

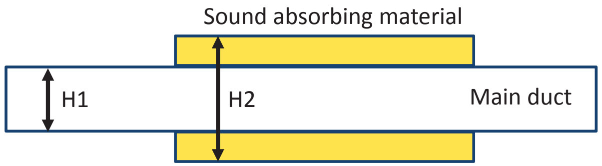
- Above cut-on frequency, cross modes can exist in the duct.
- If there is a side-branch absorber installed, acoustic waves can by-pass the absorber (i.e. silencer useless).





Cross Sectional Discontinuity

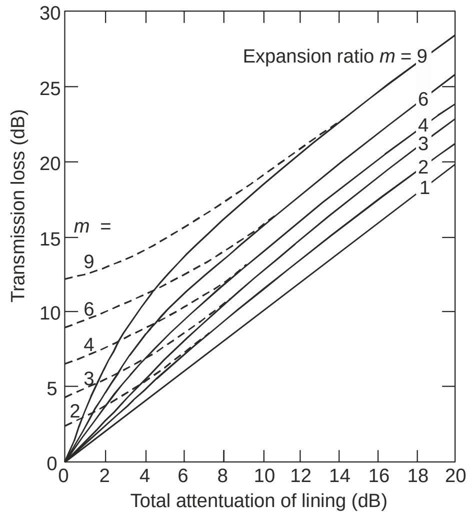
- Usually the absorptive material is placed recessed in the duct, so that the air-flow is smooth, which results in a sudden expansion in the cross-sectional area.



- The sudden expansion is beneficial and increases the transmission loss. (remember the simple expansion chamber silencer)

ENC p. 494

Cross Sectional Discontinuity



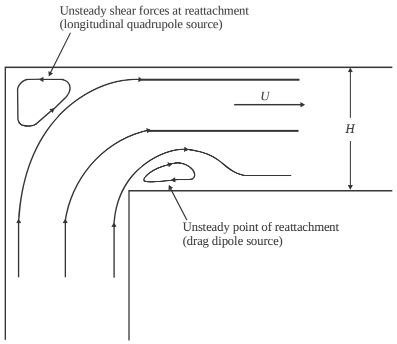
The transmission loss (TL) of a lined expansion chamber as a function of the area expansion ratio m and the total attenuation of the lining. The solid curves show TL for $kl_e = 0, \pi, \ldots .n\pi$ and the dashed curves show TL for $kl_e = \pi/2, 3\pi/2, \ldots, (2n+1)\pi/2$.

ENC p. 495

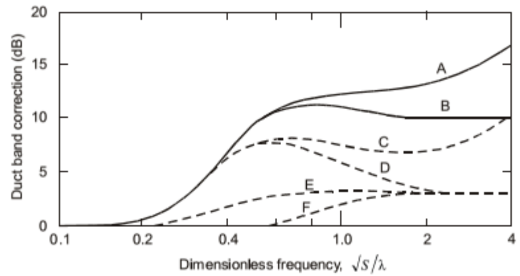
Unwanted Flow Noise at Bends

ENC p. 477

- Very common noise issue occurs at sharp bends.
- Problem can be fixed by
  - Gradual bends
  - Turning vanes



Duct Bends or Elbows



Correction for attenuation at a duct bend with no turning vanes: *A*, rectangular lined duct and axial plane wave input; *B*, rectangular lined duct and diffuse input; *C*, rectangular unlined duct and axial plane wave input; *D*, rectangular unlined duct and diffuse input; *E*, circular unlined duct 0.2 m in diameter and diffuse input; *F*, circular unlined duct 1.5 m in diameter and diffuse input. *S* is the duct cross-sectional area, and is the sound wavelength. The circular duct curves are based on data from ASHRAE-1984 Systems.

ENC p. 496

Unlined Ducts

ENC p. 497

- An unlined duct will attenuate sound and can be estimated using Table 9.5 for a rectangular duct, and Table 9.6 for a circular duct.

Table 9.5 Approximate attenuation in unlined rectangular sheet metal ducts in dB/ma. Adapted from ASHRAE (2007). If the duct is externally insulated then double these values. P = perimeter, and S = area of duct cross section.

Duct perimeter to area ratio (P/S) (m^{-1})^b	Octave band centre frequency (Hz) 63	125	250	over 250
27	0.98	0.66	0.33	0.33
13	1.15	0.66	0.33	0.20
9.8	1.31	0.66	0.33	0.16
6.6	0.82	0.66	0.33	0.10
3.3	0.49	0.33	0.23	0.07
2.2	0.33	0.33	0.16	0.07

Unlined Ducts

ENC p. 498

Approximate attenuation in unlined circular sheet metal ducts in dB/m.

Duct Diameter (mm)	Octave band centre frequency (Hz) 63	125	250	500	1000	2000	4000
$D \leq 180$	0.10	0.10	0.16	0.16	0.33	0.33	0.33
$180 < D \leq 380$	0.10	0.10	0.10	0.16	0.23	0.23	0.23
$380 < D \leq 760$	0.07	0.07	0.07	0.10	0.16	0.16	0.16
$760 < D \leq 1520$	0.03	0.03	0.03	0.07	0.07	0.07	0.07

The attenuation is unaffected by *external* insulation.

Effect of Duct End Reflection

ENC p. 498

Duct reflection loss (exit correction) (dB).

Duct diameter (mm)	Octave band centre frequency (Hz) 63	125	250	500	1000	2000
150	18(20)	13(14)	8(9)	4(5)	1(2)	0(1)
200	16(18)	11(12)	6(7)	2(3)	1(1)	0(0)
250	14(16)	9(11)	5(6)	2(2)	1(1)	0(0)
300	13(14)	8(9)	4(5)	1(2)	0(1)	0(0)
400	10(12)	6(7)	2(3)	1(1)	0(0)	0(0)
510	9(10)	5(6)	2(2)	1(1)	0(0)	0(0)
610	8(9)	4(5)	1(2)	0(1)	0(0)	0(0)
710	7(8)	3(4)	1(1)	0(0)	0(0)	0(0)
810	6(7)	2(3)	1(1)	0(0)	0(0)	0(0)
910	5(6)	2(3)	1(1)	0(0)	0(0)	0(0)
1220	4(5)	1(2)	0(1)	0(0)	0(0)	0(0)
1830	2(3)	1(1)	0(0)	0(0)	0(0)	0(0)

This table applies to ducts terminating flush with a wall or ceiling and several duct diameters from other room surfaces: if closer to other surfaces use the entry for the next larger duct. Numbers in brackets are for ducts terminated in free space or at an acoustic suspended ceiling.

Duct Breakout Noise

ENC p. 499

- Modern office buildings have air-conditioning ducts suspended between ceiling panels and the floor above, or just radiates into the open space.

It is useful to predict the noise "breaking-out" from the duct walls to know if there is going to be an acoustic problem.

Duct Breakout Noise

ENC p. 499

Noise radiated from duct walls

$$\boxed{L_{wo} = L_{wi} - TL_{OUT} + 10\log_{10}\left[\frac{P\ell}{S}\right] + C \quad \text{(dB)}}$$

L_{wo} = sound power level out
L_{wi} = sound power level in duct at entrance to duct section of interest
S = duct cross sectional area
P = duct perimeter
ℓ = duct length radiating power
C = correction factor to account for L_{wi} decreasing as sound propagates along duct

$$= 10\log_{10}\left[\frac{1-e^{-(\tau+\beta)\ell}}{(\tau+\beta)\ell}\right]$$

$\beta = \Delta/4.34$
Δ = sound attenuation (dB/m) due to internal ductwork losses (0.1 dB/m for unlined ducts)
$\tau = (P/S)10^{TL_{OUT}/10}$
TL_{out} = transmission loss of duct wall

Duct Breakout Noise

- Duct Wall Transmission Loss TL_{out} (rectangular section ducts only)
  - Calculate cross-over frequency from plane wave propagation to higher order mode propagation in duct airway

    $f_{cr} = 612/(ab)^{1/2}$ (Hz) ENC p. 500

    a, b = duct cross sect. dimensions (m).
  - Below f_{cr} (plane wave energy dominates)

    $$TL_{out} = 10\log_{10}\left[\frac{fm^2}{(a+b)}\right] - 13 \quad \text{(dB)}$$

    m = mass/unit area of duct wall (kg/m²).
  - Above f_{cr} but below ½ flat panel f_c (higher order mode energy dominates)

    $$TL_{out} = 20\log_{10}(fm) - 45 \quad \text{(dB)}$$

Duct Breakout Noise

- Duct Wall Transmission Loss TL_{out} (Cont.) (rectangular section ducts only)
  - Minimum allowed TL_{out} is $TL_{out} = 10\log_{10}(P\ell/S)$
  - Maximum allowed TL_{out} = 45 dB
  - Above ½ flat panel f_c use TL predictions for flat panel.

- Duct Wall Transmission Loss TL_{out} (other section ducts)
  - Generally much higher than rectangular ducts of same area.
  - Follow guidelines in ASHRAE SYSTEMS HANDBOOK.

Duct Breakin Sound Transmission

ENC p. 500

- Break-in Sound Transmission

$$L_{wi} = L_{wo} - TL_{in} - 3$$

L_{wo} = sound power incident on entire outside length of ductwork
L_{wi} = sound power travelling in one axial direction.

f_o = 343/2a [cut on frequency of first higher order mode]

For ($f < f_o$), TL_{in} is larger of

$$TL_{out} - 4 - 10\log_{10}(a/b) + 20\log_{10}(f/f_0)$$

or $10\log_{10}(\ell/\text{a} + \ell/\text{b})$

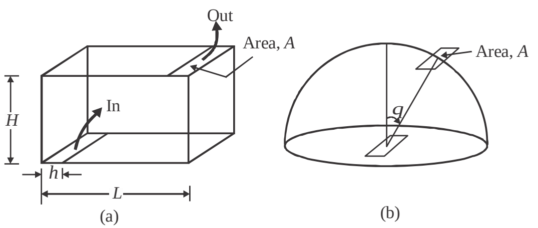
ENC p. 501

For $f > f_o$ $$TL_{in} = TL_{out} - 3$$

Lined Plenum Attenuator

ENC p. 501

- A lined plenum chamber is often used in air conditioning system as a device to smooth air-flow, and fortunately acts as a muffling device.



- It can be analysed using room acoustic theory.

Lined Plenum Attenuator

ENC p. 502

- High frequency analysis
  - Reverberant field power out $W_{RF} = W_{IN}\, A/R; \quad R = \dfrac{S\bar{\alpha}}{1-\bar{\alpha}}$
  - Direct field power out

$$W_{DF} = W_{IN}\, A\cos\theta / 2\pi r^2$$

$$TL = -10\log\frac{W_{OUT}}{W_{IN}} = -10\log_{10}\left[\frac{A}{R} + \frac{A\cos\theta}{2\pi r^2}\right]$$

Factor of 2 in above, deleted if inlet closer to edge than centre of a wall

A = area of plenum exit (m²)
R = plenum room constant (m²)
S = total wall area of plenum (m²)
$\bar{\alpha}$ = mean Sabine wall absorption coefficient
θ = angular direction
r = line of sight distance from plenum entrance to exit (m)
W_{DF} = direct field sound power (W)
W_{RF} = reverberant field sound power (W)
W_{IN} = input sound power (W)

Lined Plenum Attenuator

ENC p. 502

- Low frequency analysis - ASHRAE Method
  Treat as an expansion chamber and use reactive silencer analysis.
  For frequencies above the first mode cut on frequency, f_{co} and if plenum inlet is directly in line with the outlet.

$$TL = b\left[\frac{A}{\pi r^2} + \frac{A}{R}\right]^n$$

If inlet is closer to the centre of the wall than the corner, then π is replaced by 2π, b = 3.505 and n = -0.359.

When θ is non-zero, corrections must be added to the TL and these are listed in Table 9.8 as the numbers not in brackets.

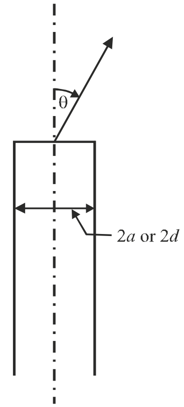
For frequencies below the duct cut on frequency: $TL = A_f S + W_e$

where the constants A_f and W_e are given in Table 9.9.

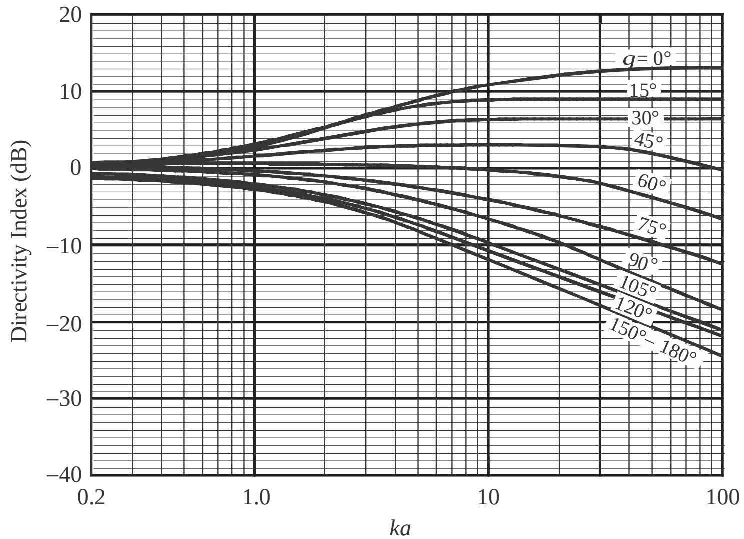
When θ is non-zero, corrections must be added to the TL and these are listed in Table 9.8 as the numbers in brackets.

Exhaust Stack Directivity

- It is advantageous to point exhaust duct upwards as sound is directed upwards.
- The directivity is a function of frequency and the radius a of a circular duct or $2d$ of rectangular duct.



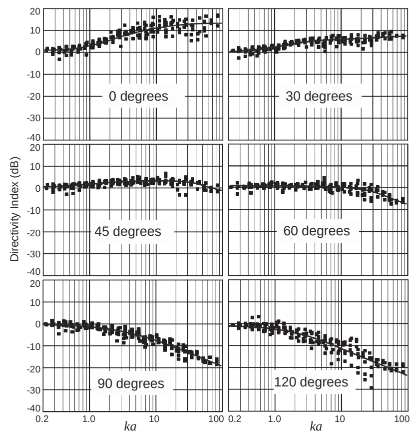
Exhaust Stack Directivity



Exhaust stack directivity index measured in the field vs ka, where a is the stack radius

p.508 Fig 9.30

Exhaust Stack Directivity



Scatter in the directivity Index data measured in the field and reported by Day (2008).

p.512 Fig 9.33

Exhaust Stack Directivity

Sound power level, L_w, radiated by duct outlet:

ENC p. 513

$$L_w = L_p - DI_\theta + 10\log_{10} 4\pi r^2 + 10\log_{10}\frac{400}{\rho c} + A_E$$

Directivity index, DI_θ, may be obtained from Figures
Excess attenuation, A_E, may be calculated as in Ch. 5.
Noise reduction due to exhaust stack addition:

$$NR = IL + A_{Es} - A_E + 20\log_{10}(r_s/r) + DI_\theta - DI_s$$

subscript, s, refers to quantities with the stack in place.
Sound pressure level, L_{ps}, at location r_s with the stack in place is given by:

$$L_{ps} = L_w - IL_s + DI_s - 10\log_{10} 4\pi r_s^2 - 10\log_{10}\frac{400}{\rho c} - A_E$$

where DI_s is the directivity of the exhaust stack in the direction of the receiver obtained from the directivity figures.

Summary

- Acoustic performance metrics
  - Noise Reduction NR, Transmission Loss TL, Insertion Loss IL
- Acoustic impedance analysis of reactive mufflers
- Lined duct silencers
- Pressure drop calculations
- Exhaust stack directivity

Questions

- Does the measurement of the Insertion Loss of a muffler also require measuring the source impedance?
- What effect does the gas temperature have on resonance frequency of a QWT?
- Design a Helmholtz resonator to attenuate sound at 100Hz.
- Design a lined duct silencer for an air-conditioning system that has a 30cm duct, input power of Lw=50 dB and require an outlet SPL of 30 dB.

Next ....

- End of course !
- Course Wrap Up
  - eSELT: what do the questions mean?
  - Revision Session Q&A
  - What is and isn't in the final exam
  - Format of the exam
  - Hints for answering multiple choice questions
- Final Exam