HELSINKI UNIVERSITY OF TECHNOLOGY
Department of Electrical and Communications Engineering

Benedict Slotte

ACOUSTICS SIMULATION IN MOBILE PHONE AUDIO DESIGN

Thesis for Master of Science degree submitted for approval 25.3.1999 in Espoo, Finland

Supervisor
Matti Karjalainen, Prof.

Instructor
Juha Backman, MSc.

Abstract

HELSINKI UNIVERSITY OF TECHNOLOGY

Author:	Benedict Slotte		
Name of thesis:	Acoustics simulation in mobile phone audio design		
Date:	23.03.1999	Number of pages:	103
Department:	Electrical and Communications Engineering		
Professorship:	Acoustics and audio signal processing		
Supervisor:	Matti Karjalainen, Prof.		
Instructor:	Juha Backman, MSc		

This thesis deals with the application of acoustics modelling and simulation principles and tools to acoustics design of mobile phones and accessories, especially earpieces. The acoustical/mechanical construction of a typical mobile phone and its accessories is overviewed, concentrating on the handportable earpiece. The acoustical behaviour of the earpiece in both normal use and in type approval measurements is examined in more detail, while at the same time the concept of leak tolerance is introduced. Guidelines for the design of leak tolerant earpieces are derived.

A basic toolbox for acoustics simulation of mobile phones and their accessories is built as a ”macro” collection inside the Micro-Cap V circuit simulator program. Some models are derived from results published in literature. Documentation is added to each model to describe its usage and its limitations. Basic theoretical principles behind equivalent circuits and acoustics simulation are introduced, while examining briefly the advantages and disadvantages of acoustics simulation when applied to real-world electromechanoacoustical constructions.

The created simulation tools as well as the design guidelines are put to use together in practice to design a proper mechanical/acoustical solution for the earpiece in a commercial phone. The good performance of the simulation tools is verified by evaluating possible earpiece solutions, and by comparing measurements and simulations of hardware prototypes.

Keywords: Simulation, modelling, leak tolerance, earpiece

Tiivistelmä

TEKNILLINEN KORKEAKOULU

Tekijä:	Benedict Slotte		
Työn nimi:	Akustiikkamallinnus matkapuhelinten audiosuunnittelussa		
Päivämäärä:	23.03.1999	Sivumäärä:	103
Osasto:	Sähkö- ja tietoliikennetekniikan osasto		
Professuuri:	Akustiikka ja äänenkäsittelytekniikka		
Valvoja:	Matti Karjalainen, prof.		
Ohjaaja:	Juha Backman, dipl. ins.		

Tämä diplomityö käsittelee akustiikkamallinnuksen ja -simuloinnin periaatteiden ja työkalujen käyttöä matkapuhelinten, erityisesti matkapuhelinten kuulokkeiden, akustisessa suunnittelussa. Tavanomaisen matkapuhelimen ja sen lisälaitteiden akustista/mekaanista rakennetta käsitellään lyhyesti, keskittyen ennen kaikkea puhelimen kuulokkeeseen. Kuulokkeen akustista käyttäytymistä, sekä normaalissa käyttötilanteessa että tyyppihyväksyntämittauksissa, käsitellään syvällisemmin. Samalla määritellään vuotosietoisuus-käsite, sekä johdetaan käyttökelpoisia suunnitteluohjeita vuotosietoisten kuulokkeiden suunnittelua varten.

Kokoelma perustyökaluja matkapuhelinten ja oheislaitteiden akustiikan simulointia varten laaditaan Micro-Cap V -piirisimulointiohjelmaan ”makro”-kokoelman muodossa. Muutama malli johdetaan kirjallisuudessa esitetyistä tuloksista. Jokaiseen malliin liitetään mallin käyttötapaa ja rajoituksia kuvaava teksti. Esitetään sijaiskytkentöjen ja akustiikkasimuloinnin perusperiaatteet, käsitellään lyhyesti edut ja haitat sovellettaessa simulointia käytännön sähkömekanoakustisiin rakenteisiin.

Simulointityökalut ja suunnitteluohjeet käytetään yhdessä kaupallisen matkapuhelimen kuulokkeen mekaniikka-/akustiikkaratkaisun suunnitteluun. Simulointityökalujen hyvä toimivuus varmistetaan käymällä läpi joukko mahdollisia kuulokeratkaisuja sekä vertaamalla simulointituloksia ja mittaustuloksia prototyypeistä keskenään.

Avainsanat: Simulointi, mallinnus, vuotosietoisuus, kuuloke

Preface

This master of science thesis was written mainly during my third year of employment at Nokia Mobile Phones, Salo, Finland. I started the work leading to this thesis in the summer of 1996 as an audio design engineer, acquiring experience in acoustics design of mobile phones. The thesis was finished in the spring of 1999. Without the experience that the two first years gave me I would not have had the necessary knowledge to finish a work like this.

Carl Poldy of Philips Speaker Systems, Vienna, Austria, one of my most valuable contacts with whom I have exchanged theoretical ideas, models, and experience, was the one who inspired me in the first place to choose acoustics simulation as the topic of this thesis. My instructor and colleague, Juha Backman, shares my interest in theoretical acoustics and has also given me very valuable information about literature on the subject. My superiors at the company, Arto Sailajoki, Jari Romo, Juha Raussi and Ismo Peltonen, have allowed me to spend most of the third year writing this thesis while reducing my time spent at the office, which has felt very comfortable and motivating. Credits go also to Olli Niemi who was my partner at the beginning of my employment, my supervisor Prof. Matti Karjalainen of Helsinki University of Technology, and finally to all other colleagues and people too many to mention here.

While writing this thesis I had three goals in mind:

- to create a document that could be used by other people as an educating and interesting introduction to acoustics modelling and simulation,
- to leave behind me a piece of good documentation of the development of the product whose design I was involved in, and
- to create a basic toolbox for acoustics simulation that could be used in the future.

Finishing this thesis and its accompanying simulation model collection has taken a lot of time, but I am very satisfied with the result and I hope that the whole document can be of good use to other people who want to learn more about this interesting topic.

Benedict Slotte
31.03.1999

Contents

1 Introduction 1.1

2 A typical mobile phone 2.1

2.1 The handportable and its accessories 2.1
2.1.1 The handportable 2.1
2.1.2 Common accessories 2.2
2.2 Speech transmission 2.3
2.2.1 The network 2.3
2.2.2 Speech properties and coding 2.3
2.2.3 Transducers 2.4
2.2.4 Type approval 2.4
2.3 User interface 2.5
2.3.1 Alerting 2.5
2.3.2 Keypad tones and other signals 2.5

3 Audio transducers 3.1

3.1 Principles of operation 3.1
3.1.1 Dynamic transducers 3.1
3.1.2 Electromagnetic transducers 3.2
3.1.3 Electret transducers 3.3
3.1.4 Piezoelectric transducers 3.4
3.2 Transducer choices 3.4
3.2.1 Earpieces and earphones 3.5
3.2.2 Loudspeakers 3.5
3.2.3 Microphones 3.5
3.2.4 Buzzers 3.6

4 Principles of acoustics simulation 4.1

4.1 The keys to acoustics simulation 4.1
4.2 Analogies 4.1
4.2.1 The three primary domains 4.1
4.2.2 Radiation as a fourth domain 4.3
4.2.3 Summary of analogies 4.3
4.3 Basic components and quantities 4.5
4.3.1 General impedance 4.5
4.3.2 Resistance 4.6
4.3.3 Inductance 4.7
4.3.4 Capacitance 4.7
4.3.5 Interconnecting the domains 4.8
4.4 Advantages of simulation 4.9
4.4.1 Evaluation of basic solutions 4.9
4.4.2 Transducer specification 4.10
4.4.3 Tuning and optimization 4.10
4.4.4 Verification 4.10
4.4.5 When measurement is hard or impossible 4.11
4.5 Restrictions 4.11

4.5.1 Nonlinearity 4.11
4.5.2 Model availability 4.11
4.5.3 Physical size versus wavelength 4.12
4.5.4 Transducer irregularities 4.12
4.5.5 The reality is three-dimensional 4.13
4.5.6 Unknown dimensions 4.13
4.5.7 Convergence 4.14
5 Building circuits for simulation 5.1
5.1 Starting point and methods 5.1
5.1.1 Objectives 5.1
5.1.2 Simulator features 5.2
5.2 System topology 5.3
5.2.1 Transducers 5.3
5.2.2 Measuring equipment 5.3
5.2.3 Loads 5.3
5.2.4 Driving circuitry 5.4
5.2.5 Signals 5.4
5.2.6 Selection of building blocks 5.4
5.3 Acoustics simulation as part of a development process 5.5
5.4 Implementation 5.6
5.4.1 Global definitions 5.6
5.4.2 Shape and naming conventions 5.6
5.4.3 Parallel connection 5.8
5.4.4 Impedances 5.8
5.4.5 Sources, transforming two-ports and grounds 5.9
5.4.6 Physical structures 5.10
5.4.6.1 Cavities 5.10
5.4.6.2 Holes 5.11
5.4.6.3 Ducts 5.12
5.4.6.4 Other structures 5.12
5.4.7 Air load 5.13
5.4.7.1 Air loads for specific radiator geometries 5.14
5.4.7.2 End corrections 5.14
5.4.7.3 Radiation resistances 5.15
5.4.8 Domain interfaces 5.15
5.4.9 Radiation 5.16
5.4.9.1 Spherical and plane waves 5.17
5.4.9.2 Radiation-acoustical domain interfaces 5.18
5.4.9.3 Other radiation macros 5.18
5.4.10 Transducers 5.18
5.4.10.1 Dynamic loudspeakers 5.19
5.4.10.2 Electret microphones 5.19
5.4.11 Measuring equipment 5.20
5.4.12 Probes and annotators 5.21
5.4.13 Compound macros 5.21
5.4.14 Other macros 5.22
6 Special audio problems and solutions 6.1
6.1 Mechanics and mass production 6.1
6.1.1 Saving size 6.1

6.1.2 Mechanical tolerances 6.2
6.1.3 Industrial design 6.2
6.1.4 Mass production 6.2
6.2 Acoustics and the environment 6.3
6.2.1 Environmental conditions 6.3
6.2.2 RF shielding 6.4
6.2.3 Background and handling noise 6.4
6.2.4 Wind noise 6.5
6.2.5 Echo 6.6
6.2.6 Earpiece performance 6.6
7 Design example: leak tolerant earpiece 7.1
7.1 Leak tolerance — what and why 7.1
7.1.1 Performance requirements 7.1
7.1.2 Test and measurement requirements 7.1
7.1.3 Background theory 7.2
7.1.4 Practical implementation 7.3
7.2 The transducer component 7.4
7.2.1 Dynamic earpiece capsule in three domains 7.4
7.2.2 Dynamic earpiece capsule in the acoustical domain 7.6
7.2.3 Multiple-mass equivalent circuit 7.7
7.3 Mechanical configurations 7.8
7.3.1 Lowering the output impedance 7.8
7.3.1.1 Internal structure of the earpiece capsule 7.9
7.3.1.2 Mechanical structures behind the earpiece capsule 7.11
7.3.1.3 Front cavity and holes 7.12
7.3.1.4 Leaks inside the phone 7.13
7.3.2 Stabilizing the output volume velocity 7.14
7.3.3 Combination methods and modifications 7.17
8 Case study: leak tolerant earpiece 8.1
8.1 The starting point 8.1
8.1.1 Leak tolerance and previous earpieces 8.1
8.1.2 The new transducer 8.4
8.1.3 The new phone 8.8
8.1.4 Design and production requirements 8.10
8.1.5 Solution 1 (starting point): leak in measurement 8.10
8.1.5.1 Design rules 8.10
8.1.5.2 Performance 8.13
8.1.5.3 Simulations vs measurements 8.14
8.2 Solution alternatives 8.15
8.2.1 No front leak (solution 2) 8.15
8.2.2 Internal leak into parallel cavity (solution 3) 8.17
8.2.3 Internal leak into side cavity (solution 4) 8.22
8.2.4 External leak into side cavity (solution 5) 8.24
8.3 Solution 5 — the final choice 8.25
8.3.1 Dimensioning 8.26
8.3.1.1 Front holes 8.26
8.3.1.2 External leak holes 8.26
8.3.1.3 Leaks in the back cavity 8.30
8.3.2 Final simulation model 8.30

8.3.3 The final earpiece ... 8.30

9 Conclusions ... 9.1

10 References ... 10.1

Appendix I Wavelengths and dimensions ... I.1

Appendix II Analogous quantities ... II.1

Quantities and units ... II.1
Conversion factors ... II.2

Appendix III Derivations ... III.1

Holes ... III.1
Circular hole ... III.1
Rectangular hole ... III.2
Ducts ... III.5
Circular duct ... III.5
Rectangular duct ... III.6
Air load ... III.7
Circular radiator ... III.7
Radiation of one radiator into half space ... III.7
Radiation of one radiator into any solid angle ... III.8
Mutual interaction of several radiators ... III.9
Summary ... III.10
Rectangular radiator ... III.11
One radiator ... III.11
Mutual interaction of several radiators ... III.12
Summary ... III.12

Appendix IV Micro-Cap simulation macros ... IV.1

Constant and function definitions ... IV.1
Shapes, electrical definitions and descriptions ... IV.2
Impedances ... IV.2
Sources, transforming two-ports and grounds ... IV.4
Physical structures ... IV.6
Air load ... IV.11
Domain interfaces ... IV.15
Radiation ... IV.17
Transducers ... IV.21
Measuring equipment ... IV.23
Probes and annotators ... IV.25
Compound macros ... IV.29
Miscellaneous macros ... IV.33
Macros created especially for thesis simulations ... IV.35
Unimplemented macros ... IV.36
Menu structure ... IV.36

Appendix V Models of earpiece capsule ... V.1

Appendix VI Measurement setup ... VI.1

Abbreviations

CAD	Computer aided design
DAI	Digital audio interface
DSP	Digital signal processing
DTMF	Dual tone multi-frequency
ETSI	European Telecommunications Standards Institute
FEM	Finite element method
GSM	Global System for Mobile Communications
ITU	International Telecommunication Union
LRGP	Loudness rating guard-ring position
RF	Radio frequency
SPL	Sound pressure level
R, L, C	Resistance, inductance, capacitance
cgs	Centimetres, grammes and seconds
mks	Metres, kilogrammes and seconds (= SI system)

Symbols

A	Area
B	Magnetic flux density
C	Capacitance
E	Energy
F	Force
G	Conductance
I	Impulse; intensity
K	Compressibility
L	Inductance
P	Power
R	Resistance
V	Volume
Y	Admittance
Z	Impedance

c	Speed of sound
d	Diameter
f	Frequency
h	Height
i	Current
k	Coefficient
l	Length; length in flow direction
m	Mass
n	Number
p	Pressure
q	Volume velocity; charge
t	Time; thickness
u	Voltage
v	Velocity
w	Width
x	Displacement

Φ	Magnetic flux
Ω	Solid angle
α	Voltage transformation ratio, damping coefficient; coefficient
β	Transconductance; coefficient
γ	Propagation constant; coefficient
η	Efficiency
λ	Wavelength
μ	Coefficient of viscosity
ω	Angular frequency
ρ	Density
$_a$	Acoustical
$_b$	Back
$_c$	Characteristic; voice coil
$_e$	Earpiece
$_f$	Front
$_h$	Housing
$_l$	Leak
$_m$	Mechanical; mutual; moving
$_r$	Radiation; resonance
$_s$	Specific; speaker; solid angle
$_u$	User
$_v$	Viscosity
$_x$	Additional
$_{in}$	Input
$_{\max}$	Maximum
$_{\min}$	Minimum

1 Introduction

This thesis deals with the application of well-known electronics simulation methods to acoustics design of mobile phones, which is becoming an increasingly challenging task. This is due to several different reasons. Mobile phones are rapidly becoming a part of everyday life in more and more countries. Yet there are still only a few highly penetrated areas, particularly Scandinavia, the USA and Australia, and in large parts of the world a mobile phone is a rarity. The prices of mobile phones have been going down steadily which combined with the low penetration has lead to enormous market growth. The increasing number of potential mobile phone users and the rapid technology improvement, in addition to low market prices, implies aggressive competition where manufacturers are forced to achieve better quality in all areas. Audio performance is one of these, and it has not been too great a concern until quite recently. Methods that speed up the product development process or reduce the hardware prototype need are very welcome.

Many audio signal processing methods are available in modern mobile telephony. These are used mainly to improve the quality of transmitted speech, but a part of the acoustical behaviour of the phone will always remain an outcome of its basic mechanical-acoustical design. Mechanically, mobile phones are getting smaller all the time, while their performance is improving and new solutions are invented. The shrinking dimensions add to the challenge of good acoustics design.

In view of these trends it is a good choice to start utilizing modelling or simulation methods in audio and acoustics design. Just as an electric circuit can be fine-tuned and optimized by simulation, so can the acoustical parts of a mobile phone using the methods described in this thesis. The *Micro-Cap V* computer simulation system, developed by Spectrum Software, was chosen for the simulations carried out in this thesis. The purpose of the thesis is to describe the principles behind simulation, and to apply simulation to some real acoustics problems in mobile phones.

A special collection of acoustics simulation models, tailored especially for mobile phone and accessory acoustics simulation, was created to aid in the necessary simulations. Some of these models needed additional derivations from separate results presented in acoustics literature in order to obtain unifying, general models. The concept of *leak tolerance* is defined and examined in further detail. One chapter is dedicated to the practical aspects of the development of a new acoustics solution in a Nokia phone. Also, the performance of acoustics simulation in general is discussed within the scope of this task.

2 A typical mobile phone

Although there are many manufacturers around and each of them produces their own range of mobile phones, with models aimed at different consumer groups, the principles behind the use of the phones remain the same. Mobile phones and their accessories look fairly similar or at least work very much the same way. This chapter provides some basic background information on the physical appearance, mounting and use of phones and accessories. Also, the transmission network is presented briefly, since it is easily the most significant bottleneck in terms of speech quality.

2.1 The handportable and its accessories

2.1.1 The handportable

The term *handportable* is used to denote a single package housing all the electronics and mechanical parts of a mobile phone, as opposed to a unit with a separate handset. Being the main part of a mobile phone set, the handportable contains everything that is necessary to make and receive calls. Specifically,

- a microphone close to the user's mouth,
- a small loudspeaker, earpiece, close to the user's ear,
- a buzzer or other alerting device to signal an incoming call,
- connectors for accessories and charging,
- other parts, such as an antenna, a rechargeable battery pack, a keyboard, a display, a RF power amplifier and other electronics.

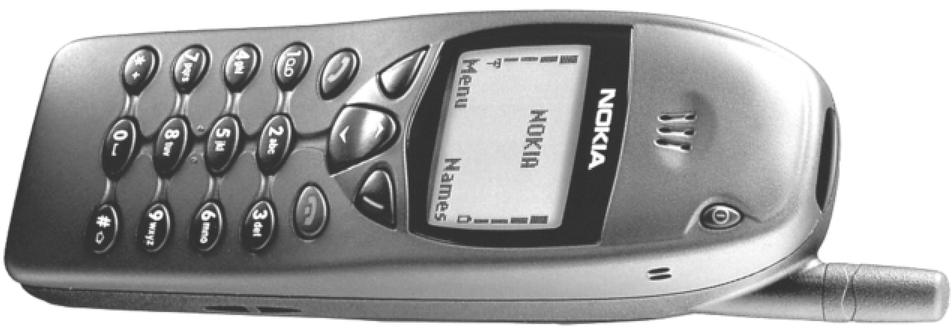


Figure 2.1: A handportable (the example shown here is the Nokia 6110, whose earpiece will be the topic of chapter 8 in this thesis).

The handportable may also be equipped with a moving lid or cover to protect the keys from being pressed by mistake. Such a lid (slide or flip) may present some additional problems for an audio designer, as will be seen later.

2.1.2 Common accessories

For the sake of both convenience and safety, a car driver should avoid using one hand to hold the phone while talking and driving at the same time. *Hands-free kits* (*car kits*) are available to most mobile phones. In some countries hands-free kits are even required by law if the phone is to be used while driving.

A hands-free kit is a combination of a loudspeaker and a microphone, usually to be mounted on the dashboard of the car. The microphone picks up the voice of the driver or other passenger, and the speech normally directed to the earpiece is amplified and sent to the loudspeaker (which is enclosed in some kind of box to provide a good enough bass response). Because both the microphone and the loudspeaker are located much farther away from the user's mouth and ears than the microphone and earpiece of the above-mentioned handportable, special care must be taken to suppress extra background noise. The loud sound from the loudspeaker is free to travel to the microphone, which, without counteractions, would lead to severe *echo* and howling. Echo is described below.

A *headset* is like a normal headphone or earphone with an added microphone, and thus the headset frees the user's hands but is not as susceptible to background noise interference and howling as a hands-free kit. This is simply because both the microphone and the earphone (or headphone) are located closer to the mouth and ear, respectively, which allows them to function more optimally.

Early mobile phones used to have separate *handsets*, but handsets are also available to small pocket-size phones to give additional flexibility. A separate handset is convenient, for example, when the phone is mounted on the dashboard with a hands-free kit. The user may then switch between hands-free and handset operating mode. From the acoustics design point of view a handset is very similar to a handportable.

Figure 2.2: Typical mobile phone accessories (from top, left to right: handset, headset, hands-free kit microphone, hands-free kit loudspeaker). All of these accessories are for Nokia phones.

2.2 Speech transmission

Most of the restrictions on speech transmission are due to the major part of the transmission path: the transmission network connecting the mobile terminal to the other party. Obtaining a good enough speech quality is a question of recognizing the restrictions on transmission quality, knowing the properties of human speech, and having a sound reproduction in the mobile phone that does not too much degrade what is left of the transmitted speech.

2.2.1 The network

The purpose of this text is not to go into any detailed descriptions of how various phone network standards differ. *Network*, in this context, means the part of the transmission path that interconnects two or more *terminals* (in this case, telephones or mobile phones) during a call. Basically, the transmission path between two cellular telephones involved in a call includes radio connections from the telephones to their base stations, and a (mostly) electrical or optical transmission path between the base stations. In spite of the general overview of typical networks presented here, there are some common properties that have impact on audio transmission quality and acoustics design.

One of the most important of these is echo, which is caused mostly by unwanted coupling from earpiece to microphone in a telephone handset. Combined with signal processing and transmission delays present in the network, the other party will hear echoes of his or her own voice while speaking on the telephone. The transmission delay is especially pronounced in modern digital networks, due to the heavy signal processing. For this reason, special echo cancellation methods are used to prevent the echoing from disturbing the conversation [1].

2.2.2 Speech properties and coding

It is important to save bandwidth by incorporating an efficient coding method in the transmission. This should happen without unnecessarily degrading the intelligibility of speech. Speech transmission therefore implies a compromise between maximizing transmission quality and minimizing transmission bandwidth. Research and experience have pointed out the well-known fact that speech may be bandpass filtered to reduce its bandwidth considerably without any corresponding loss in intelligibility [2].

Upper treble frequencies are well represented in the noise created by many consonant sounds. Yet this noise is wide-band and the upper frequencies may be filtered out without intelligibility suffering. The lowest bass frequencies are not present at all, but the higher bass frequencies are pronounced in male speech. Although low male voices have their fundamental frequency below 100 Hz, the human glottis generates a sound that is rich in harmonics in the voiced sounds. The result is that appreciable savings in bandwidth can be accomplished while maintaining an acceptably good speech quality and a negligible loss of intelligibility. Although the speech quality allowed by typical telephone networks is far from hi-fi standards, the intelligibility suffers much less than the subjective quality when speech is filtered.

2.2.3 Transducers

The *transducers* (earpiece, or loudspeaker, and microphone in this case) are the end points of the non-acoustical part of the transmission path, i.e. transducers are electroacoustical devices whose purpose is to transmit and/or receive sound waves. Modern hi-fi loudspeakers and microphones can boast technical specifications that are far beyond what is needed in a mobile phone. Since cost is a primary concern, it is not feasible to aim at hi-fi quality while the network changes the speech spectrum as much as it does by bandpass filtering the speech. In principle, transducers need only reproduce frequencies lying within the specifications of the network. It is also important to utilize transducers that are reliable and stable in their behaviour (in varying environmental conditions) and fit into the mechanics of the phone.

2.2.4 Type approval

Before making a new phone model available on the mass market, a phone manufacturer has to get the phone officially approved in special tests. This is called *type approval* and the purpose of type approval is to ensure that the phone will work properly as a part of the transmission network it is going to connect to. For this to be the case, the phone must be fully compatible with the network - mobile terminal interface. This interface is defined in the network specifications. There are special requirements on the essential audio frequency responses (an example is shown below), as well as many other other non-audio properties, e.g. RF (radio frequency) emission. The audio requirements are a main concern for the phone audio and baseband engineers.

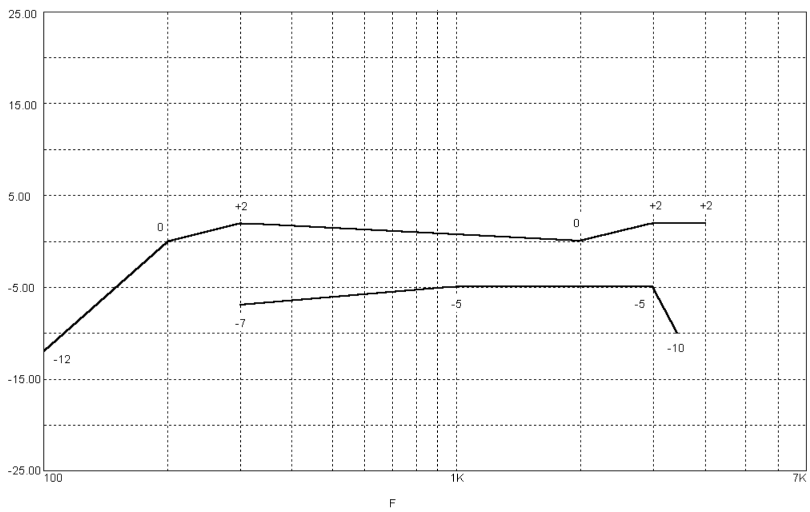


Figure 2.3: An example of a speech transmission frequency response requirement: the receiving frequency response limits specified for mobile terminals in the GSM system [3]. *The minimum allowable bandwidth is 300 Hz to 3.4 kHz. dB values are relative, not absolute.*

2.3 User interface

The *user interface* of a mobile phone includes a couple of features that belong to the domain of audio design. In addition to keys to dial numbers and the display where text is displayed, some information is also supplied to the mobile phone user as sound.

2.3.1 Alerting

The *alerting* signal, reproduced through the buzzer and indicating an incoming call, is definitely the most important of all user interface sound signals. It should be loud enough to draw attention to an incoming call even in rather noisy environments, but yet it should not be harmful to the ear if the telephone accidentally rings while held at a short distance from the ear. The shrinking dimensions of mobile phone handportables bring the buzzer closer to the earpiece, which is a potential risk.

Modern mobile phones have many different alerting signal choices. Usually most of them are musical melodies, *ringing tunes*. This places some requirements on the performance of the buzzer. It is difficult in general to combine high loudness of the alerting signal with smooth, equally loud response of several different frequencies used in ringing tunes. Typically, if a loud signal is wanted, resonance has to be taken advantage of by choosing a buzzer with some strong resonance, and a signal frequency that coincides with that resonance in such a way that the sound pressure level (SPL) is boosted.

On the other hand, such resonances also emphasize certain notes in ringing tunes and make them sound too pronounced. The ringing tunes must be chosen with care, keeping the acoustical properties of the buzzer in mind. It is also essential to know that the buzzer may have a different resonance frequency when mounted inside the phone than when in free space. The reason for this is that the acoustical *load* on the buzzer is different in both these cases. Examples of the same phenomenon will be given later for earpieces.

2.3.2 Keypad tones and other signals

Other sounds that do not need to be loud can be reproduced through the earpiece. A typical example is keypad sound indicating that a key is pressed, DTMF (dual tone multi-frequency) tones generated by the dialling keys, and warning signals indicating a low battery or lost network coverage. Such warning signals may sound in the middle of a call when the phone is held to the ear, so they should not be too loud. Reproducing DTMF tones is not a problem for a working earpiece, since the standard DTMF frequencies are chosen to lie inside the speech transmission band.

3 Audio transducers

This chapter will present the physical principles behind the transducer types encountered in mobile phones and accessories, in order to explain their properties, as well as the simulation models and equations appearing in subsequent chapters. Also, typical characteristics and factors affecting the choice of transducer type for each given application are discussed.

The word *earpiece* has been used so far to denote the small loudspeaker held against the ear in a handset or handportable. Additional mechanical structures may be built around such a loudspeaker to change its acoustical properties. For this reason the term *earpiece capsule* will be used hereafter to denote the actual transducer component, while, generally, *earpiece* will mean the whole earpiece assembly (unless the distinction is obvious from the context). The same goes for *microphone capsule* and *microphone*.

3.1 Principles of operation

3.1.1 Dynamic transducers

The dynamic transducer is the best-known and most common of all transducer types. It utilizes the force acting on a current-carrying conductor in a permanent magnetic field. This force is strongest when the direction of current flow is perpendicular to the field. The conductor is usually wound as a coil, called the *voice coil*. The voice coil is fastened to a movable diaphragm and mounted in a narrow gap with a strong magnetic field created by a permanent magnet. In a sound-emitting transducer (loudspeaker), the desired sound signal is brought to the coil as a voltage, and the resulting current sets the sound-radiating diaphragm in motion. In a sound-sensing transducer (microphone), the movement of the diaphragm is transformed into an electric signal by the voice coil. This can be expressed as:

$$\begin{aligned} F &= Bli \\ u &= Blv \end{aligned} \tag{3.1}$$

where F is the force, B is the magnetic flux density, l is the length of the conductor inside the magnetic field, u is the voltage, i is the current and v is the velocity. u and i are electrical quantities, while F and v are mechanical quantities. Eqs. (3.1) describe the transduction between electrical and mechanical quantities.

The diaphragm has a circular shape in most dynamic transducers. It is supported by a flexible suspension that allows it to move linearly within some limit. Bigger dynamic transducers (used as loudspeakers) may have a stiff cone-shaped membrane of paper or some plastic material. Small transducers (used as earpiece capsules) need not be stiff over most of their area, so typically their whole membrane is made of thin flexible material. The voice coil is, in turn, fastened to the center part of the diaphragm. The permanent magnet has a cylindrical gap for the voice coil, and the ends of the coil wire are brought out to the transducer terminals (to which the electric signal is connected).

Ready-made earpiece capsules have some kind of housing built around them. The acoustical properties of such housings vary quite a lot. Some earpiece capsules have their back and front completely open, in some cases one or two of them is covered with a layer of porous material, and in some cases the back is closed or almost closed. This has a great effect on the acoustical behaviour.

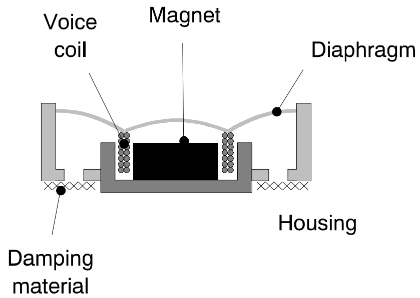


Figure 3.1: A typical dynamic earpiece capsule (or small dynamic loudspeaker).

3.1.2 Electromagnetic transducers

A magnetic field is the key to the operation of an electromagnetic transducer also, but in this transducer type the coil is part of a fixed (non-moving) electromagnet. The moving diaphragm typically is made of thin metal sheet, and mounted close to one of the electromagnet poles. Because the force between the electromagnet and the diaphragm can only be attractive, not repulsive (which would rectify the signal brought to the coil), a permanent magnet must be included to bias the transducer. In its equilibrium state, the diaphragm is partly pulled towards the permanent magnet and coil assembly by the attractive force of the permanent magnet, and the current through the coil can then modulate this force, resulting in diaphragm motion. The coil is wound around a pole piece to raise the efficiency. The diaphragm may also have a metal disc in the center.

Electromagnetic transducers can be made to have good speech reproduction properties, but in mobile phones it is much more common to employ them as buzzers for loud alerting signals only. In typical buzzers, the motion of the moving diaphragm is nonlinear. During operation the diaphragm is not kept confined to its linear region of movement, but rather driven between its two extremes of maximum displacement. This creates strong harmonics that enhance the subjective loudness of the alerting signal.

The mechanical structure of an electromagnetic transducer is simple. Typically, the assembly is as shown in the picture below. The rim of the membrane is supported by the transducer housing. It should be noted that the shape of the housing, as well as the placement of the sound port, may vary quite a lot.

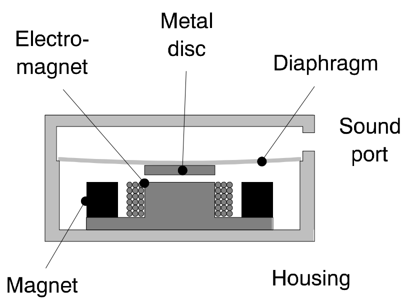


Figure 3.2: A typical electromagnetic buzzer.

3.1.3 Electret transducers

Electret transducers make use of the electrostatic force and capacitance between two charged surfaces. One of the surfaces is a rigid plate, while the other is as a movable membrane. When the distance between the plate and the membrane varies due to incoming sound pressure waves, the capacitance is modulated and an electric voltage signal is generated between the plates because of the present charge. This charge is retained through the use of a permanently charged material (*electret*) on the membrane, which is metalized to provide an electrical connection. In other transducers the rigid plate, instead of the membrane, is coated with electret material (*backplate electret*).

Electret transducers are very common as microphones. Films of polymers like polyvinylidene fluoride, polyacrylonitrile and polycarbonate are usable as electret coatings [4, p. 1897, 1935; 5, p. 6.7]. To make the electret microphone efficient, the distance between membrane and back plate should be small. However, a very narrow, closed air space also restricts the movement of the membrane by adding stiffness, which is why the back plate in typical electret transducers is perforated to open up to a bigger air cavity underneath it. However, there is also a choice of having both sides of the membrane open to the air (one of them through the back plate, of course). If this is done, the net force moving the membrane will depend not on the pressure on one side of the membrane, but rather on the difference between pressures on both sides. For this reason microphones of this kind are known as *pressure gradient microphones* as opposed to *pressure microphones* having only one side of the membrane open to the air. Pressure microphones are *omnidirectional*, i.e. their sensitivity is independent of direction. Pressure gradient microphones, on the other hand, have a well-defined axis of greatest sensitivity. This is because more or less cancellation of the pressure difference between both sides of the mebrane will result, depending on the direction of the oncoming sound waves. A small pressure gradient microphone has a ”figure of eight” directional characteristic (it is a *bidirectional* microphone).

Although pressure microphones are in effect closed at the back, a very small pressure equalization opening must still be provided to prevent fluctuations in the static atmospheric pressure (or very low frequency sounds) from displacing the membrane from its equilibrium position.

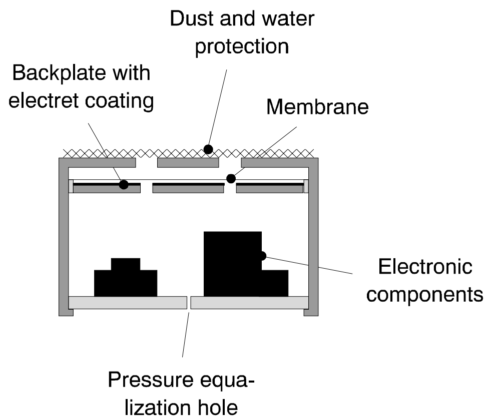


Figure 3.3: A typical electret pressure microphone. A pressure gradient microphone would have one or a few big openings instead of the very small pressure equalization hole.

3.1.4 Piezoelectric transducers

Pieces of some materials (for example, Rochelle salt, ammonium dihydrogen phosphate and lead zirconate titanate [6, p. 167; 4, p. 1896-1897; 5, p. 7.6]) have the property of changing shape proportionally to an external electric field. Vice versa, forced deformation of these leads to proportional potential differences on their surface. Materials of this kind are called *piezoelectric*. There are both crystalline and ceramic piezoelectric materials available. The relation between shape and electric field makes piezoelectric materials usable in *piezoelectric transducers*.

There is a great number of possible shapes and alignments of a piezoelectric crystal or ceramic inside a transducer, and the mechanical movement can appear in many ways: as bending, compression-expansion, shear or torque of the material. The transduction can take place either with or without a separate diaphragm. If a separate diaphragm is used, it is connected mechanically to the piezoelectric material so as to allow sound pressure acting on it to transfer to the material (and vice versa in a loudspeaker). Without a separate diaphragm, one surface of the piezoelectric material itself is responding to air pressure. In both cases two electrodes are connected to the material to provide the electrical connection. The simplest type of piezoelectric transducer consists of nothing but a disc of piezoelectric material between two plate electrodes, with one of the electrodes effectively acting as a diaphragm.

3.2 Transducer choices

The audio transducers used in handportables are earpieces, microphones and buzzers. Loudspeakers and microphones are used in hands-free kits, and headsets are combinations of a microphone and an earphone. There is no definite distinction between loudspeakers, earpieces and earphones. By *loudspeaker* one generally means a bigger sound-emitting transducer mounted far from the ears, while earpieces and earphones are in fact small loudspeakers held close to the ear. One could also distinguish these by the fact that loudspeakers radiate sound to the ears from a distance, while earpieces and earphones are held more or less tightly to the ear. A buzzer differs from a loudspeaker in that it is not meant to reproduce spectrally complex speech but simple alerting signals.

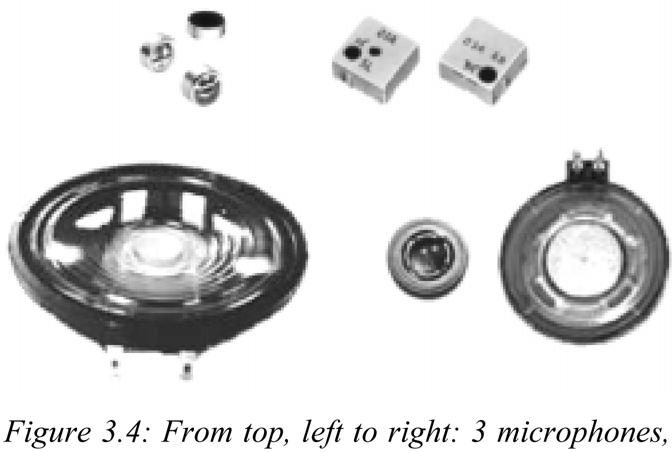


Figure 3.4: From top, left to right: 3 microphones, 2 buzzers, 1 small loudspeaker, and 2 earpiece capsules. The earpiece capsule to the left (viewed in detail in chapter 8) is used in the Nokia 6110 and has a diameter of 13 mm. Pictures by permission of Matsushita Corp., Star Micronics Co. Ltd. and Philips Speaker Systems.

3.2.1 Earpieces and earphones

Most earpiece capsules are of the dynamic type, but there are also piezoelectric and electromagnetic earpiece capsules around. Piezoelectric transducers have the added benefits of being mechanically simple and cheap. For high-quality reproduction, however, dynamic earpieces are preferred to piezoelectric ones. Their membranes can be made to have good excursion capability, which is required for good reproduction of low frequencies under varying circumstances. Also, the great number of dynamic earpiece capsules on the market make them attractive in terms of availability, although piezoelectric earpiece capsules can be made cheaper. Earpiece capsules are typically around 15 mm in diameter.

Earphones are separate transducers connected to the mobile phone with a cord. A modern headset consists of such an earphone combined with a microphone somewhere closer to the user's mouth. As with phones and other accessories, the trend is to make headset earphones small. This favours the in-ear type of earphone, which feeds sound directly into the ear canal. Being closer to the eardrum than a phone earpiece capsule, a typical earphone transfers sound to the ear more efficiently than an earpiece. The requirements on excursion capability can be relaxed, and the needed small size of the earphone transducer presents no problem as far as sound output is concerned. For these reasons, piezoelectric transducers are better suited for earphones than for earpieces. Yet most earphones are of the dynamic type because there is a vast amount of small dynamic transducers available on the market.

3.2.2 Loudspeakers

Loudspeakers are used in hands-free kits. Of such loudspeakers is required only that they be conveniently small and capable of reproducing intelligible speech. Price is another important criterion for the accessory manufacturer. Currently there are no official requirements on their frequency response, contrary to the situation in type approval of phones.

There are many small and good loudspeakers available. The consumer electronics and hi-fi industry has been involved with these for decades, and they are seen in everything from intercom equipment to small portable radios. Being capable of both low bass and high treble reproduction is a difficult task for a single transducer, as is well known from hi-fi loudspeaker technology. In this context a hands-free loudspeaker represents the midrange element in a three-way hi-fi loudspeaker, and it has to reproduce no more than the speech transmission frequency band. Typical hands-free loudspeakers have diameters of around 5 cm and are of the dynamic type. The design of the loudspeaker enclosure is essential in providing the best obtainable (or necessary) low frequency response.

3.2.3 Microphones

Electret microphones are the typical choices for mobile phones and accessories. They can be made very small, and their acoustical properties are far closer to the ideal than those of earpieces, earphones and loudspeakers. The size advantage is important in mobile phone design, but reducing microphone size also means less sensitivity. Dimensions of around 5 mm are common. Dynamic microphones are not suitable for use in mobile phones due to their larger size and interference pickup by the voice coil.

3.2.4 Buzzers

In mobile phones, buzzers are not used for speech reproduction. Their only purpose is to emit loud enough alerting and warning signals. For this reason it does not matter how much they distort the incoming (electric) signal as long as the resultant sound is, subjectively, loud enough and not too unpleasant at the intended frequencies.

There is a minority of piezoelectric buzzers available, but most buzzers are of the electromagnetic type. They are nonlinear in behaviour, for reasons stated above. Consequently, if a sine signal is fed to such a buzzer, it will be distorted and harmonics will appear in the sound spectrum. The nonlinearity is an advantage here, since one or several of the harmonics may be boosted by an acoustic Helmholtz resonator (corresponding to a series RLC electrical resonator, as will be seen later). This raises the SPL and makes the alerting sound more penetrating to the ear. Buzzers are typically fed with rectangular pulse signals, using narrower pulses when lower sound output level is wanted. Thus the spectral composition of the same ringing sound signal will vary with the level setting.

4 Principles of acoustics simulation

This chapter deals with the basic theoretical principles that allow one to apply tools and methods for electric circuit analysis to acoustics (and mechanics, which belongs to the same field since physical transducers with moving parts are employed). How these methods are practically implemented will be viewed in the next chapter.

4.1 The keys to acoustics simulation

Computer programs for electric circuit simulation are widely used by circuit designers. These programs, hereafter called *simulators*, are typically able to run at least transient, AC and DC analyses on circuits. They are then able to output results to a file, a printer or the screen. Circuits are described using simple text statements obeying a specific syntax, or drawn on the screen using a graphical editor. The best-known circuit simulator is *SPICE* (in various appearances). In some more advanced systems (such as *APLAC*) the circuit description approaches a programming language, which allows the designer to include optimization routines or other special functions [7].

It is very desirable to take advantage of the efficient electric circuit simulation tools that are available and apply them to acoustics design. This is quite simple in principle, because the equations governing voltage and current in basic lumped and linear electric circuits are also found in the fields of mechanics and acoustics. For example, an equation that relates current to voltage in a circuit element in an electric circuit might, in mechanics, have the same form except for voltage and current being exchanged with different physical quantities. The key point here is that this allows one to create *analogies* and use electric circuit theorems, analysis, and simulation tools on, in this particular case, mechanics and acoustics. Before running a simulation, a corresponding electric circuit (*equivalent circuit*) must be created and presented to the simulator. Simulation results (voltages and currents) can then be interpreted as corresponding acoustical quantities.

4.2 Analogies

4.2.1 The three primary domains

The basic physical laws governing charge, voltage, current and other electrical quantities are transformable to other *domains* (such as mechanics and acoustics) by exchanging quantities according to certain analogies. Such analogies may be formed in very many ways. The equations in the other domains are simply required to have the same form and relations as in the electrical domain, except for the exchanged quantities. Because electric circuit simulators are to be utilized in acoustics (and mechanics) simulation, there are a couple of additional requirements that the analogies should meet in order to be practically useful in this case:

- equivalent circuits should be easy to create, and
- the simulation software should be able to output the necessary information.

Simulators output node voltages and branch currents. In addition to these, charge stored in capacitors and magnetic fluxes in inductors are directly available in typical simulators. This can be taken advantage of by choosing a suitable set of quantities. The first requirement means that it should be as straightforward as possible to sketch an equivalent circuit when the mechanoacoustical topology of the system is known.

In this particular context, acoustics and mechanics laws are to be transformed to the electrical domain. Acoustics literature defines two electrical analogies for both mechanical and acoustical systems. The two analogies are related by the swapping of quantities. Both will be presented here, since in most cases one of them is easier to use in acoustics, while the other is easier to use in mechanics [8]. Corresponding analogies for rotational motion are also available, but since all cases viewed here will involve only rectilineal motion, these analogies will be left out. A starting point is formed by letting quantities correspond to each other as follows:

Table 4.1: Basic analogous quantities in the electrical, mechanical and acoustical domains (impedance analogy).

Electrical domain	Mechanical domain	Acoustical domain
Voltage	Force	Pressure
Current	Velocity	Flow rate

Flow rate, as mentioned above, is called *volume velocity* in acoustics. Both pressure and flow rate are defined as in fluid physics. The analogy defined above is called *impedance analogy* or *direct analogy*. Another well-known analogy (called *admittance analogy*, *mobility analogy*, or *inverse analogy*) is formed by exchanging the roles of voltage and current [6, p. 52]. In this analogy the voltage corresponds to velocity and flow rate, and current corresponds to force and pressure:

Table 4.2: Alternative analogous quantities in the electrical, mechanical and acoustical domains (admittance analogy).

Electrical domain	Mechanical domain	Acoustical domain
Current	Force	Pressure
Voltage	Velocity	Flow rate

The impedance analogy is the more intuitive choice for acoustics. This is because air flow routes appear as current branches, acoustic impedances translate into electric impedances, and the topology of the equivalent circuit thus resembles that of the original acoustical system. The situation is different in mechanics. Some like to use the impedance analogy in mechanics too to avoid confusion, but the topologies of the physical system and the equivalent circuit do not resemble each other as in the acoustical domain. Hence the choice to use the admittance analogy for mechanics in this thesis, although the familiar impedance analogy is used in acoustics. The price paid is that the nature of quantities in the mechanical domain is less intuitive. A voltage does not correspond to a force, but rather to a velocity, and components in the mechanical domain are admittances, not impedances.

4.2.2 Radiation as a fourth domain

The three domains described above are the ones commonly used in simulation of electroacoustic transducers. Volume velocity is a convenient quantity in acoustic circuits having confined spaces with given cross-sectional areas and volumes, but it is not very useful when describing sound radiated into free air. Such a sound field is better described by pressure and particle velocity instead of pressure and volume velocity. *Particle velocity* is the net velocity of the air molecules caused by the propagating sound waves (i.e. neglecting thermal motion). The corresponding volume velocity is obtained by multiplying particle velocity by area. For these obvious reasons, it was decided to employ a fourth domain in this thesis (and in the collection of simulation tools to be developed) to describe radiation.

The *radiation domain* (as it will be called hereafter) can be seen as a modification of the acoustical domain described previously:

Table 4.3: Relation of radiation domain to common acoustical domain.

Acoustical domain	Radiation domain
p (pressure)	p (pressure)
q (volume velocity)	v (particle velocity)

Of course, sound radiation is an acoustical phenomenon just as air flow and air compression in confined spaces, and in that respect the radiation domain is also ”acoustical”. However, in order to distinguish this domain (as defined here) from the acoustical domain (as defined above and in common acoustics literature), it was given the special name ”radiation domain”. Impedances have a special meaning in the radiation domain (see below), but most other quantities and equivalences in the radiation domain will not be examined any further because the radiation domain will not be used this thesis.

4.2.3 Summary of analogies

All the needed quanitities can be derived from the few ones chosen above, adding also the time t which is identical in all domains. This is straightforward and the details will not be presented here due to lack of space. The table below shows all the useful quantities, their units (in the SI system) and their definitions (boxed equations).

Table 4.4: Summary of analogies used in this thesis. Some quantities that are not often used, and/or do not have any established name, are left out.

Electrical	Mechanical (admittance analogy)	Acoustical (impedance analogy)	Radiation
Voltage u volt (V)	Velocity v m/s	Pressure p pascal (Pa)	Pressure p pascal (Pa)
Current i ampere (A)	Force F newton (N)	Volume velocity q m^3/s	Particle velocity v m/s
Charge q coulomb (C) $dq = idt$	Impulse I Ns $\boxed{dI = Fdt}$	Volume V m^3 $dV = qdt$	Particle displacement x m $dx = vdt$

Magnetic flux Φ weber (Wb) $d\Phi = udt$	Displacement x meter (m) $dx = vdt$		
Impedance Z ohm (Ω) $\boxed{Z = \frac{u}{i}}$	Mechanical admittance Y_m m/Ns $Y_m = \frac{v}{F}$	Acoustic impedance Z_a Pas/m^3 $\boxed{Z_a = \frac{p}{q}}$	Specific acoustic impedance Z_{as} Pas/m $\boxed{Z_{as} = \frac{p}{v}}$
Admittance Y mho $Y = \frac{i}{u}$	Mechanical impedance Z_m Ns/m $\boxed{Z_m = \frac{F}{v}}$	Acoustic admittance Y_a m^3/Pas $Y_a = \frac{q}{p}$	
Resistance R ohm (Ω) $\boxed{R = \frac{u}{i}}$	Mechanical conductance (responsiveness) G_m m/Ns $G_m = \frac{v}{F}$	Acoustic resistance R_a Pas/m^3 $\boxed{R_a = \frac{p}{q}}$	
Conductance G mho $G = \frac{i}{u}$	Mechanical resistance R_m Ns/m $\boxed{G_m = \frac{F}{v}}$	Acoustic conductance G_a m^3/Pas $G_a = \frac{q}{p}$	
Inductance L henry (H) $u = L\frac{di}{dt}$ $\boxed{L = \frac{d\Phi}{di}}$	Mechanical capacitance (compliance) C_m m/N $v = C_m\frac{dF}{dt}$ $\boxed{C_m = \frac{dx}{dF}}$	Acoustic inductance (inertance) L_a kg/m^4 $p = L_a\frac{dq}{dt}$ $\boxed{L_a = \frac{pdt}{dq}}$	
Capacitance C farad (F) $i = C\frac{du}{dt}$ $\boxed{C = \frac{dq}{du}}$	Mechanical inductance L_m (mass m) kilogram (kg) $F = L_m\frac{dv}{dt}$ $\boxed{L_m = m = \frac{dI}{dv}}$	Acoustic capacitance C_a m^3/Pa $q = C_a\frac{dp}{dt}$ $\boxed{C_a = \frac{dV}{dp}}$	
Power P watt (W)	Power P watt (W)	Power P watt (W)	Intensity I W/m^2

Energy E joule (J) $\boxed{dE = Pdt}$	Energy E joule (J) $\boxed{dE = Pdt}$	Energy E joule (J) $\boxed{dE = Pdt}$	

Many new quantities are seen in this table. The electrical domain has the familiar quantities impedance, resistance, inductance and capacitance, as well as admittance and conductance (reciprocals of impedance and resistance). The corresponding quantities in the mechanical and acoustical domain are called *mechanical* and *acoustical* impedances, resistances etc. Impedance in the radiation domain is the ratio of pressure to particle velocity, which is a quantity commonly known as the *specific acoustic impedance*. Also new is *impulse*, a mechanical domain quantity which equals the product of force and time. It should be noted that mechanical conductance is also known as *responsiveness*, mechanical capacitance as *compliance*, acoustic inductance as *inertance*, and mechanical inductance is identical to mass.

4.3 Basic components and quantities

This section provides a more detailed look at the definitions of the various impedances and the ways that they are disguised in real mechanics and acoustics. Such understanding is essential in order to be able to extract components for the equivalent circuits from real-world acoustical systems.

The *SI* (*mks*) system is used throughout in this thesis, although formerly the *cgs* system (centimetres, grammes and seconds instead of metres, kilogrammes and seconds) was used almost exclusively in acoustics. The interrelations between these systems are tabulated in appendix II.

4.3.1 General impedance

Mechanical impedance relates the force acting on a rigid object to its velocity (with the force always parallel to the velocity, i.e. by definition a one-dimensional case), just as electrical impedance relates the voltage across a lumped electrical device to its current. The definition is seen in table 4.4. The mechanical impedance is a complex quantity, and just like an electric impedance it can be split up into a mechanical resistance and/or a mechanical reactance. Mechanical impedance has the unit Ns/m, or kg/s, which is also called *mechanical ohm*. *Acoustic impedance*, in the same manner, relates the pressure on a surface to its volume velocity. If nonuniform, one must integrate infinitesimal surface elements. It should be noted that the volume velocity can be viewed either as the flow rate of medium through the surface (which may be hypothetical), or as the time derivative of the volume of space that some real moving surface traverses [6, p. 11]; in both cases the volume velocity is the time derivative of the volume. This is important, but it will not be explicitly stated hereafter. The unit of acoustic impedance is Pas/m^3, or kg/(m^{4}s), which is also called *acoustical ohm*. *Specific acoustic impedance* (unit: Pas/m or kg/(m^{2}s)), which is the radiation domain equivalent of the acoustic impedance defined above, is defined for a propagating acoustic wave as the ratio of pressure to particle velocity.

From the definitions it can be deduced that the acoustic and mechanical impedances for a given case are related as follows:

$$Z_a = \frac{Z_m}{A^2} \tag{4.1}$$

where A is the surface area.

Resistances, inductances and capacitances will be viewed next. There are practically no ideal resistances, inductances or capacitances in the real world, however. A good example is a rigid body having a certain moving mass (mechanical inductance). This body will usually encounter friction of some kind as it moves, which means that its mechanical impedance will not be purely inductive but also resistive. This resistive part is hard to get rid of completely because the body has to be supported in some way. A certain impedance describing the relationship between force and velocity, or pressure and volume velocity, can be found for some frequency. In most cases impedances will also be more or less frequency-dependent or nonlinear. This puts restrictions on the use of simple lumped and linear impedances in simulation. On the other hand, the Micro-Cap V program can handle frequency dependence better than many other typical simulation tools, which allows one to take frequency dependence into account in a frequency sweep. Being used to calculate frequency responses, this will be the most used feature of the simulator.

4.3.2 Resistance

A pure *mechanical resistance* is formed by an ideal viscous force opposing the movement of a rigid, massless object. Because the viscous force is assumed to be ideal, the resultant velocity is proportional to the force acting on the object. Mechanical resistance has the unit Ns/m (or kg/s). Any force behaving as described appears as a mechanical resistance, regardless of its origin (viscosity or some other phenomenon). Non-viscous or nonlinear forces might be linear enough in some amplitude and frequency range, such that they can be modelled approximately by linear mechanical resistances. The most typical mechanical resistances appearing in the context of electroacoustics are:

- mechanical damping (for example, to reduce vibration or smooth out a mechanical resonance), and
- suspension losses (example: the suspension of a loudspeaker membrane).

A pure *acoustic resistance* (unit: Pas/m^3 or kg/(m^{4}s)) can be viewed as an ideal viscous force opposing the flow of a massless medium, whose pressure is *p*, through a hypothetical surface. From eq. (4.1) , for some given case,

$$R_a = \frac{R_m}{A^2} \tag{4.2}$$

Acoustic resistance is typically due to the viscosity of the medium (air) itself. This viscosity is especially pronounced in:

- small holes,
- narrow slits, and
- layers of cloth.

There are always viscosity losses present when air flows through mechanical structures. To model this important behaviour, the corresponding acoustic resistance must be taken into account. Of course, if some other acoustic impedance predominates, one may discard the viscous resistance unless too much accuracy is lost.

4.3.3 Inductance

It was already stated that *mechanical inductance* (unit: kg) is identical to mass, i.e. a pure mechanical inductance would be formed by a rigid body whose movement is opposed by no other forces except the inertia of its own mass. In real life, any moving object will have a mechanical inductance in its mechanical equivalent circuit because massless objects do not exist except at elementary particle level. The mechanical inductance of some moving part may still be left out if it is not relevant in the given case, e.g. if some other mechanical impedance predominates. Moving mechanical parts are found in all electroacoustic transducers used in mobile phones.

Acoustic inductances (unit: kg/m^4) are not as easy to visualize as mechanical ones. The definition derived from the analogies and shown in table 4.4 is not very easy to understand either. But comparing it to the definition of electric inductance one sees that acoustic inductance is the property of the medium that opposes changes in the volume velocity (just as an inductance opposes changes in the current). This property is also called *inertance*. Acoustic inductances appear because the medium has moving mass. This is also seen in the relation between acoustic inductance and mechanical inductance (mass):

$$L_a = \frac{L_m}{A^2} = \frac{m}{A^2} \tag{4.3}$$

where A is the surface area (over which the pressure acts) and m is the moving mass. A pure acoustic inductance would be formed by a volume of medium moving without compression and viscous losses. Typical cases where acoustic inductances may be large are:

- in small openings,
- in long tubes or ducts.

The smaller the area, the bigger the acoustic inductance, because the area is in the denominator in eq. (4.3).

4.3.4 Capacitance

A pure *mechanical capacitance* (unit: m/N or s^2/kg) is formed by a force that is proportional to displacement, as is understood looking at the definition in table 4.4. This is the kind of behaviour shown by an ideal spring, and the mechanical capacitance is simply the reciprocal of the *spring constant* (ratio of force to displacement from equilibrium). Typical mechanical capacitances are:

- suspensions (around a loudspeaker cone, for example), and
- flexible membranes.

Analogously, an *acoustic capacitance* (unit: m^3/Pa or m^{4}s^2/kg) is formed by a pressure that is proportional to a change of volume. If a pressure acts to change some closed volume, it will be opposed by a pressure proportional to the change of volume. By definition of *compressibility* of an ideal fluid, the acoustic capacitance for some volume of fluid is equal to the product of compressibility and equilibrium volume,

$$C_a = KV \tag{4.4}$$

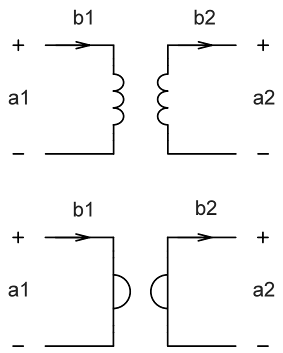
where K is the compressibility. In other words, acoustic capacitance and volume are proportional for a given compressible fluid. In this particular case, the fluid is the air that propagates sound. Real air is not an ideal fluid, hence a real-world acoustic capacitance will practically always have losses. Finally, from eq. (4.1),

$$C_a = A^2 C_m \tag{4.5}$$

Acoustic capacitance appears whenever the flow of medium is opposed by a pressure proportional to the change of volume. A cavity of air is by far the most typical case, with a small cavity appearing as a small acoustic capacitance, and vice versa.

4.3.5 Interconnecting the domains

So far, four domains have been described. All electroacoustics that will be viewed in this context includes some or all of these domains. Direct combination of domains is not possible in principle, since their units differ and they represent different systems (the acoustical and radiation domains are closely related, however). It is logical to have all electric circuitry in the electrical domain, and all moving parts in the mechanical domain. It is obvious that the border between these domains goes through the transducer, because the transducer transforms signals in one domain to signals in the other. As will be seen in later examples, this transformation can be implemented by interconnecting the domains with either transformers or *gyrators* [6, pp. 71-76; 8]:



$$a_2 = \alpha a_1 \quad b_2 = \frac{b_1}{\alpha}$$

$(\alpha = \text{"turns ratio", i.e. voltage transformation ratio})$

$$b_2 = \beta a_1 \quad a_2 = \frac{b_1}{\beta}$$

$(\beta = \text{transconductance})$

Figure 4.1: Ideal (top) transformer and (bottom) gyrator and their characteristic equations. The quantities a ("voltage") and b ("current") may represent any analogous quantities as described above.

The border between the mechanical and acoustical domain is easiest to visualize as a border between mechanical movement and air flow, i.e. where the medium enters the stage. A good example is the surface of a loudspeaker membrane. Using the above relations, it can be seen that an interface between the mechanical and acoustical domain must have the properties of a gyrator. The fourth (radiation) domain can be thought to have its border between an acoustic circuit (consisting of structures altering the flow of air) and the free air that sound radiates into. An example of this might be the bass reflex port opening in a bass reflex loudspeaker cabinet, with an acoustic circuit describing the port and the interior of the cabinet on one side, and the radiation into free air on the other side.

A special complication arises from the exchange of only one quantity in going from the acoustical domain to the radiation domain and vice versa. Using the relations defined above, it can be seen that in this case the domain interface has to be a transformer with different transformation ratios for voltage and current: the voltage transformation ratio must be unity (pressure on both sides) while the current transformation ratio must depend on the radiating area (volume velocity on one side and particle velocity on the other side, see table 4.3).

After creating an equivalent circuit, the domain borders can be moved or even eliminated by using circuit theorems to *reduce* the net on one side of a transformer or gyrator to the other side. This lowers the number of components in the circuit that is to be simulated. At the same time, border elimination provides additional insight into how an impedance in one domain is seen by some other domain. A well-known example is the change in the electric input impedance of a dynamic transducer when measured first with its membrane fixed (mechanical movement eliminated) and then free. The expressions for the reduced component values tend to be more complicated than the original expressions. When creating the equivalent circuit, it is good to do it the simple and intuitive way (with all domains there) to avoid errors. The additional number of components does not mean much to the fast simulation software of today, since even the complete circuits will be small compared to what modern simulators can handle.

4.4 Advantages of simulation

Before going into more detail, some advantages of simulation are presented in a general manner. Many of the statements here will be exemplified in later chapters where practical cases are viewed. The same goes for the disadvantages and restrictions mentioned at the end of this chapter.

4.4.1 Evaluation of basic solutions

The procedure of changing parameters and routing connections is fast and simple in simulation, especially if a graphical schematics editor is available. Changing component values represents changing electric, mechanical and acoustic impedances, and changing connections means creating new flow paths for current, i.e. air in the acoustical domain. All this is especially good at an early stage in the development of a product, when no or very little hardware is available. Simulation is then useful when evaluating which hardware solution to use. There are often several mechanical design and transducer choices.

It is clear that some part of the system must be specified before starting to search for the best solution. Trying different mechanical solutions needs a model of the transducer, and vice versa. The task becomes difficult if there are too many unspecified parameters. Price, mechanical complexity (which directly affects the production process), as well as water and dust resistance, are other important factors. In any case, it is a great advantage that hardware is not necessarily needed at all for such early stage tasks as:

- evaluating the acoustical performance of various phone mechanical design options,
- finding the best mounting option of the transducer component inside the phone,
- checking whether or not some intended transducer can be protected by a dust and water shield membrane without performance suffering, and
- evaluating transducer components of various brands in the intended phone mechanics.

4.4.2 Transducer specification

If there is a choice of having a custom transducer component made to order, simulation is easily the only way that optimal parameters for the transducer can be found (except for more tedious equation solving or hand calculation, of course). Upper and/or lower limits for such parameters as efficiency, membrane area, excursion capability etc. may be found by simulation when the desired performance characteristics (and phone mechanics restrictions) are known. The transducer can be a modification of an off-the-shelf component by the same manufacturer, in which case it helps a lot if a simulation model is available for that component. Examples of simulation tasks are:

- creating an initial transducer model (unless one is available),
- specifying required output capability of a transducer component when the mechanics design, mounting, input signal amplitude etc. are known,
- testing possible improvements to an available transducer component, and
- specifying maximum allowable tolerances of given transducer parameters.

4.4.3 Tuning and optimization

One or more design parameters can be tuned to the best values (optimized) using repeated *parameter stepping*, a simulator feature involving sequential simulation runs with a given series of values for one or more parameters [9, p. 151]. This is a very common application of simulation. Optimization can be applied also to first-cut estimates or hand calculations. Doing this with real hardware is far more expensive and time-consuming, because several samples (with varying properties) of some mechanical part or transducer have to be built. If this operation later turns out to be unnecessary, even more time and resources have been wasted than would have been the case using simulation.

4.4.4 Verification

As soon as measurable hardware is at hand, one should start by verifying measurement results against simulation results. Then the equivalent circuit models can be fine-tuned to include possible irregularities that are found (for example, a small air leak in a cavity that was assumed to be air-tight). Sometimes accidental errors in measurements can be found by running simulations of the same measurements. For example, if a frequency response measurement result looks surprising due to improper assembly of mechanical parts, the location and size of the error (usually leakage) can be tracked down by trying leaks in the simulated equivalent circuit of the measurement. Measurements and simulation can complement each other, but one must take care to be sure which result is the more reliable one when they differ appreciably. If this happens, the best thing to do is to work backward and remove parts from the assembly.

All decent simulators have some kind of *Monte Carlo* or *worst case analysis* option to verify the effect of the inevitable tolerances. If some particular solution appears to be very sensitive to tolerances or small air leaks, that solution can be either modified or discarded to give way to another better solution. All this is cheap and quick using simulation, but changing the acoustics or mechanics solution at a later stage in the product development process is costly and time-consuming.

4.4.5 When measurement is hard or impossible

The small dimensions of mobile phones and their transducers complicate measurements. To examine, say, the excursion of a microphone membrane, the sound pressure inside a small cavity, or the forces acting on the moving transducer parts, one needs to have specialized equipment at hand. In some cases such measurements are even impossible, leaving the engineer with other methods such as estimation, hand calculation or simulation. Of these three, simulation is the best choice in terms of simplicity and accuracy, provided of course that good models are available.

4.5 Restrictions

4.5.1 Nonlinearity

Typical simulators are not capable of simulating nonlinear behaviour in the frequency domain. This is because an operating point analysis is made first, and then the circuit is linearized around that operating point [9, p. 92]. In this process all nonlinearity is lost, no matter how well modelled. The most important nonlinearities in real-world electroacoustics are:

- the nonlinear compression and flow of air,
- nonlinear behaviour of materials (suspensions, membranes), and
- nonlinearity resulting from the construction of transducers (nonuniform magnetic field around the voice coil, etc.)

Of course, nonlinearity can be neglected if small-signal behaviour only is simulated. Using simulators that properly support nonlinearity, distortion effects can be examined if all the relevant nonlinearities are built into the equivalent circuit. Usually this means that some components have mathematical expressions instead of constant values. The Micro-Cap V version 2 simulator cannot simulate nonlinearity in the frequency domain, but fortunately it can handle frequency-dependent component values, which is a feature that will be put to use in this thesis.

4.5.2 Model availability

Modelling transducers in detail can be a difficult task that needs accurate information from the manufacturer. The situation is optimal if the manufacturer is capable of providing a complete simulation model (equivalent circuit), and quite good if values for membrane mass, stiffness, internal dimensions etc. are available. Unfortunately, however, too many manufacturers are not able to provide enough of the information needed for acoustics simulation. In this case a model must be developed from scratch, which may be time-consuming due to the measurements and verifications required. A compromise between model accuracy and modelling effort can be sought, but in this case it is important to keep remembering that the accuracy is limited.

In principle a transducer, for which no model is available, can be modelled as a two-port with relevant parameters such as input and output impedance (with one of these being electric and the other one acoustic). Circuit synthesis methods can be applied, or a mathematical expression (or even raw measurement data) modelling the measured behaviour can be included in the equivalent circuit — provided that the simulator supports such models. Unknown electric impedances can be measured using some of the more advanced multimeters or analyzers on the market, but no commercial multimeters exist for acoustic impedances. Thus acoustic impedances have to be measured using pressure or flow generators and probes, or, more commonly, terminating a transmission line with the impedance to be measured and calculating its value with the aid of a Smith diagram or signal analyzer.

A model obtained directly from results of measurements is probably more accurate than a lumped circuit model tuned by hand with values taken from manufacturer specifications. The disadvantage is that the picture of several internal performance parameters is lost, and one single impedance cannot be isolated. For example, it may not be possible to simulate what happens when some internal mechanical dimension changes due to tolerances.

4.5.3 Physical size versus wavelength

The modelling of mechanical structures using simple equivalent circuits with lumped components is restricted by one of the important differences between electric circuits and acoustics at audio frequencies. In electric circuits, the wavelengths are far greater than the component dimensions even at the highest audio frequencies. Such is not necessarily the case for mechanical dimensions encountered in mobile phones. Generally, to permit the use of lumped components in equivalent circuits, the physical dimensions of the mechanical structures that are to be modelled should be considerably smaller than the sound wavelength (at all frequencies used). If they are, use of lumped acoustic impedances is justified.

A quick calculation gives a wavelength for sound in room-tempered air of less than 10 cm at the highest speech transmission frequencies (around 4 kHz). Thus only small mechanical structures with dimensions less than 1 cm can be modelled by lumped components in all cases, with only low loss of accuracy at these frequencies. If more accuracy can be sacrificed or if only middle and lower frequencies are interesting, it is fully acceptable to relax this restriction. This is also the case if there is no appreciable interference between signals with different phases. Requiring that the maximum phase difference be, say, $\pi/6$ (30°), which corresponds to a twelfth of the wavelength, gives the maximum dimensions 7 mm at 4 kHz and 3 cm at 1 kHz. A table of these can be found in appendix I.

To simulate larger structures with phase differences, one needs to employ transmission line models. Such models are available in all good electric circuit simulators, including Micro-Cap [10, p. 370]. These models are made for simulation of single-mode electric transmission lines, though, which restricts their use in acoustics simulation: only one mode can be simulated. Viewed in a general way, this means that although the length (in the sound propagation direction) is allowed to be comparable to or greater than the wavelength, the other dimensions (width, height) must not be. In other words, phase differences may exist along one axis only.

4.5.4 Transducer irregularities

Earpiece capsules (and loudspeakers), feeding the network consisting of acoustic impedances of the surrounding mechanics and air, must be modelled accurately enough to get reliable results. The small microphones used in mobile phones have few irregularities and are typically easy to model. Earpiece transducers, on the other hand, may have several resonances inside their intended frequency range. These not only shape the frequency response of the transducer by itself, but may also interact with acoustic impedances in the surrounding mechanics.

Breakup modes of transducer membranes are well-known sources of frequency response irregularities. When these occur, there are points on the membrane surface moving with different, and even opposite, phases [6, p. 199]. This can happen far below the frequencies where the membrane is no longer negligibly small compared to the wavelength. Breakup modes are not modelled by lumped equivalent circuits. However, by dividing the membrane into several smaller parts (surface elements), and creating interconnected lumped circuit models for each part, at least some of the breakup modes can be made to appear in a simulation. The same principle of dividing is employed in so-called *finite element method* (*FEM*) *analysis*.

An accurate model may need to include irregularities that lie outside the intended frequency range of the transducer. This is because a resonance somewhere above or below the limits may introduce a peak or dip that is broad enough to be visible inside the intended frequency range. Even if done the opposite way (a self-made equivalent circuit that is used to specify transducer properties to the manufacturer), there will be compromises in the performance. In all cases, the transducer model should be verified by measurements and information exchange with the manufacturer.

4.5.5 The reality is three-dimensional

Air flow and sound radiation are three-dimensional physical processes. This is not taken into account when lumped components and one-dimensional transmission lines are used. In most simple situations this does not matter. However, if more than one dimension of a cavity is comparable to or larger than the wavelength, it is clear that the problem changes from one to two or three dimensions, and special care must be taken to simulate such cases.

There are other important situations where three-dimensionality comes into the picture even when all dimensions are small. One of these is the case of breakup modes discussed above. Another example is the modulation of air flow by some moving mechanical part. More specifically, one can think of a typical case with a transducer mounted very close to a rigid surface, such that there is only a narrow air gap between the membrane and the surface. When the membrane moves, the width of the air gap changes. This change may be appreciable for earpiece capsules and loudspeakers, but not for microphones since their excursions are a lot smaller. A narrow air gap acts like an air flow resistance, i.e. an acoustic resistance. When the gap width changes due to membrane vibrations, the acoustic resistance gets modulated by the membrane movement. The consequence is distortion [11]. This phenomenon passes by completely unnoticed in a simple simulation, since the usual procedure is to model each part (membrane, air gap) separately.

4.5.6 Unknown dimensions

Reliable simulation is not possible if there are unknown mechanical dimensions or transducer properties. In such a case one can sweep the unknown quantity over some range of possible values to see how it affects the result. If the result is insensitive enough to the changed quantity, then the situation is quite safe. If, on the contrary, there is a lot of deviation, it is probable that the measurement results of real hardware will not look like the simulation. Unknown dimensions appear in equivalent circuits as unknown impedances. Taking into account the small physical dimensions encountered in mobile phones, it can be hard to measure these accurately and reliably.

4.5.7 Convergence

Dimensions found in mobile phone acoustics are small compared to some fundamental SI units. The question is about millimetres and grammes instead of metres and kilogrammes. In many cases, the value of some impedance in an equivalent circuit is obtained by squaring and dividing values that lead to impractically small or large results. A good example is eq. (4.1), where an area appears squared. Expressed in metres, this will be a very small value. Then, when there are component values with both very small and very large magnitudes mixed up in the same circuit, it may happen that the iterative simulation algorithms encounter convergence problems. In such a case no reliable solution is found and the simulation halts with an error message.

5 Building circuits for simulation

According to the preceding chapter on principles of simulation, the system to simulate is input as an equivalent circuit to the simulator. In this case one can do very well using just the basic components available in most circuit simulators. However, most mechanical structures and acoustical phenomena appear as combinations of several components, whose values are related to given physical dimensions according to mathematical formulae. For the sake of simplicity it is reasonable to build complete subcircuits (called *macros* in Micro-Cap) of various mechanical and acoustical structures and phenomena. These macros are like ”black boxes” that can take physical properties (dimensions, material constants etc.) as input parameters and calculate corresponding internal component values automatically. Full descriptions of all macros created for this thesis are not included in the text. The complete macro schematics and names, as well as the menu structure, can be found in appendix IV.

From now on (unless stated otherwise), the word *object* will mean a physical structure or phenomenon that is to be simulated using a macro. A *system* is a complete assembly, usually including components in all domains: electrical, mechanical, acoustical (and radiation). Micro-Cap *components* are any building blocks (either ready-made or macros) combined to form a *schematic*. A *shape* is the graphical appearance of a Micro-Cap component in a schematic. A *circuit* can be either electric, mechanical, acoustic, or a combination of these. *Parameters* are data supplied to macros, and *pins* are the connection points of macros.

5.1 Starting point and methods

5.1.1 Objectives

Knowing the goal of creating a handy and versatile acoustics simulation package (in this case especially for mobile phones and their accessories), a starting point is established by realizing that such a simulation package should:

- be flexible, yet easy, to use,
- be able to simulate relevant systems with proper accuracy,
- accept measurable physical properties (dimensions etc.) of objects as input parameters,
- allow the user to define custom components, and
- be able to output necessary information about the behaviour of the system.

These objectives were kept in mind when creating the Micro-Cap acoustics macros for this thesis. In addition to what is listed above, care was taken also to make the acoustics macros suitable for documentation purposes (circuit schematics).

A flexible acoustics simulation package can be obtained by identifying basic objects appearing in electroacoustical systems, and implementing these as building blocks that can be combined by the user. An equivalent circuit (of varying complexity and accuracy) can then be built into each block, and the block can be attached to a suitable graphical shape and name. This was the principle behind the creation of the Micro-Cap acoustics macros for this thesis. Names and shapes were chosen to depict the purpose of each macro in a systematic and intuitive way, trying to keep in mind the vocabulary and symbols used in literature.

The following steps can be identified in creating Micro-Cap V macros for acoustics simulation, as soon as the area of application (in this case, mobile phones) has been specified:

- identification of domains (electrical, mechanical, acoustical, radiation) and their extent,
- identification and selection of needed (and simulable) building blocks,
- identification of relevant input parameters for each building block,
- creation of the equivalent circuit for each building block,
- implementation of each building block as a Micro-Cap V macro,
- documentation of each macro (parameters, usage, limitations)
- choice of a suitable graphical shape and name for each macro,
- placement of each macro in a hierarchical menu in the Micro-Cap V user interface, and
- verification of the macros (accuracy, reliability etc.).

Domains were identified in the preceding chapter, as well as needed quantities. Choosing the right building blocks means to identify what kind of objects (mechanical structures, electrical connections, physical phenomena such as air flow, radiation etc.) are likely to appear in the systems that are to be simulated. The building blocks should allow flexible interconnection without losing too much accuracy. Input parameters should be kept general but still simple enough to find and supply to the building blocks, since properties of (for example) transducers can be specified by manufacturers in many ways — if specified at all. In the latter case they should preferably be measurable.

The weaknesses of simple linear (and, for the most part, lumped) circuit acoustics simulation discussed in the preceding chapter will become apparent in the long run. However, in this case one has to be satisfied with less accurate results in many cases, since the algorithms of circuit simulators like Micro-Cap do not even allow any general nonlinear and three-dimensional analysis. If the analysis method is simplified, then so is also the way of inputting information to the simulator — no extensive data on shapes and material properties has to be supplied, but just a few parameters describing the simplified model will do.

5.1.2 Simulator features

In Micro-Cap V, a macro consists of an equivalent circuit stored as a normal circuit schematic with a given filename. A graphical shape is linked to the file. Input parameters are specified by a special text statement (.PARAMETERS) placed somewhere on the macro schematic. The macro can also have named input/output pins [9, p. 178].

Micro-Cap has a great number of ready-made shapes for electrical components. It is fully possible to do without any additional macros and shapes, especially for an acoustics engineer with equivalent circuit experience. Choosing this approach will make acoustics simulation more low-level and concrete, but harder to perceive for anyone who is not aware of the physical details of the system that the equivalent circuit describes. The higher-level approach chosen here (involving tailor-made graphical shapes and macros with observable physical properties as parameters) is clearly better for documentation purposes. It also helps the designer to remember mechanical solutions, it is simple and understandable for anyone who knows the shapes and parameters, and (above all) it liberates the engineer from calculating component values. The drawback is that the picture of magnitudes and proportions of component values is easily lost at the same time. In many cases a compromise is best, i.e. the user can still go on using separate resistors, inductors etc., or mixing them with macros.

Micro-Cap allows one shape and name to be attached to one macro. If one wants different shapes for similar objects (having the same equivalent circuit), a separate macro must be made for each object. There are four domains involved. For clarity and simplicity it was decided to use different descriptive shapes for macros in each domain, although many macros have similar equivalent circuit representations.

5.2 System topology

5.2.1 Transducers

A basic equivalent circuit of a transducer is formed by including its electric, mechanical and acoustic impedances as circuit elements in the corresponding domains. Electric impedances are due to the electric circuitry in the transducer. Mechanical impedances originate from the moving parts. Acoustic impedances result from various acoustic loads (for example, cavities and air flow routes inside the transducer). Concrete examples of mechanical and acoustic impedances were given in the preceding chapter.

5.2.2 Measuring equipment

All acoustics design involves measurements to verify and optimize operation. Typical loudspeaker frequency response measurements employ just a single free field microphone to measure the SPL on-axis. The situation is different in the case of earpieces. Because earpieces are not intended to radiate sound into free air, it makes no sense to measure them that way. Special *ear simulators* (*artificial ears* or *couplers*) have been developed to provide more or less realistic acoustical models of the human ear, with microphones to measure the resulting responses. These couplers are easy to simulate using equivalent circuits that manufacturers supply in their data sheets [12, 13].

Standard *artificial mouths* are used as radiation sources to approximate the radiation from a typical human mouth. These have certain directional characteristics, and their frequency responses are calibrated before each measurement to cancel individual variations that are due to nonidealities in their internal loudspeakers. For these reasons it does not make any real sense to simulate artificial mouths in this context.

5.2.3 Loads

Any object altering the ratio of pressure to flow rate has the effect of an acoustic impedance. The transducer is a part of the same acoustic circuit as any acoustic load connected to it. In the same manner as electric signals are shaped and altered using electric circuits, so may also the acoustical response of a transducer be altered by including various acoustic circuits. Typical examples of acoustic loads are

- enclosed volumes of air (air cavities),

- layers of textile, cloth or other damping material,
- openings (holes, slits) and
- ducts.

The sound to and from the transducer may be forced through one or more of these mechanical structures in order to alter the resultant frequency response.

Because there is also sound radiation from a transducer, some of the incoming energy has to be transferred to the surrounding medium (air) as well (alternatively, transferred from the medium to the transducer in the case of a microphone). The surrounding air can be seen as a load in which the radiated acoustic power is dissipated. For obvious reasons this air load is known as the *radiation impedance*. The radiation impedance may be split up in a resistance and a reactance. The power loss in the radiation resistance represents the acoustic power radiated by the transducer, and the reactance accounts for reactive, stored, energy returned back to the transducer [6, p. 117]. This situation may be compared to an electronics case: the resistive part of an impedance connected to an electric circuit transfers effective power (typically, heat) to the surrounding medium, in addition to loading the circuit.

Although radiation impedance may be neglected in many cases, it is very useful when calculating the radiated acoustic power. For example, the effective power is simply the power dissipated in the resistance modelling the resistive part of the radiation impedance. "Measuring" the acoustic output power this way is easy in a circuit simulator, while the other well-known way of integrating the radiated sound intensity over a 4π solid angle in three-dimensional space would be much more elaborate in a general case.

5.2.4 Driving circuitry

The electronics connected to the electrical inputs of the transducer are hereafter referred to as *driving circuitry*. Good examples are the preamplifier of a microphone or the power amplifier of a hands-free loudspeaker. Any other electrical components, such as filters and crossover networks, belong to this group also. The driving circuitry is, of course, included as it is in the simulation.

5.2.5 Signals

Input signals in the various domains are included in the equivalent circuit as corresponding analogous sources. For example, an electric signal to the driving circuitry appears as it is. Sound pressure acting on a microphone is simulated as a voltage source connected to the input of the microphone equivalent circuit. To output a force as a result of a simulation, the simulator is told to output the corresponding current in the mechanical part of the circuit. It is important to keep in mind that there are no nonlinear effects limiting signals in simulation, as there are in the real world. For this reason the physical validity of the obtained result may have to be verified separately.

5.2.6 Selection of building blocks

A rough picture of what is needed can be obtained from the above outline of the system to simulate:

- various impedances (mechanical, acoustic),
- various sources (force, velocity, pressure, volume velocity),
- inter-domain transformers and gyrators, models of transduction,
- models of mechanical structures (suspensions, cavities, holes, ducts etc.),
- models of sound radiation and air load,
- general transducer models (earpiece capsules, loudspeakers, microphones),
- models of relevant measuring equipment (ear simulators etc.), and
- other necessary macros to support the above.

The word *model* above is used to denote an equivalent circuit model (implemented as a macro) that models the item in question. *Modelling* means creating a model of the item.

5.3 Acoustics simulation as part of a development process

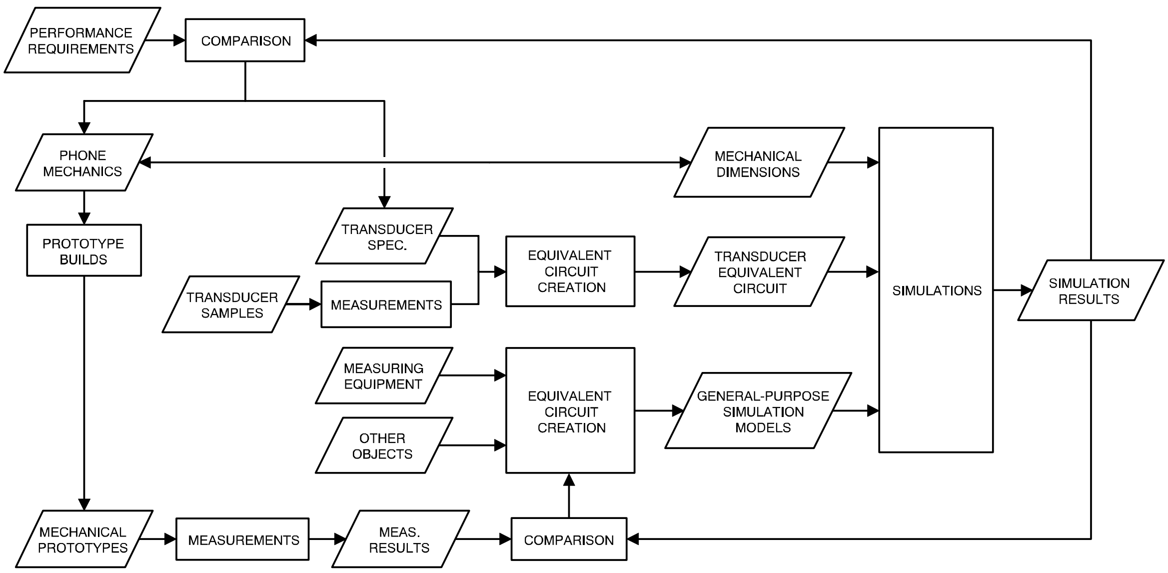


Figure 5.1: An example of a typical acoustics simulation procedure.

Acoustics simulation also should be viewed in its proper context as in fig. 5.1. This is an example of how acoustics simulation can be used as a part of a development process in which a system (transducer and surrounding mechanics) is tuned to match performance requirements. This is shown as a feedback path at the top of the diagram. Another possible feedback path is shown at the bottom. Such comparison of measured and simulated results is essential when verifying the accuracy of the simulations, just as mentioned in sec. 4.4.4. At the same time the simulation models themselves can be continuously improved, e.g. phenomena whose effects are hard to derive theoretically could be approximated (based on empirical results) in the models. Of course, this requires great care and good understanding of acoustics.

In the above diagram ”performance requirements” may mean e.g. requirements on the frequency response of an earpiece. ”Phone mechanics” is the intended mechanics of the phone (e.g. in the form of a CAD file), and such data is used to build *prototypes* (real hardware for testing purposes). A subset of the dimensions contained in the CAD file is used as parameters for the simulations (”mechanical dimensions”, e.g. the diameter and length of a sound outlet in the phone cover). Transducer *samples* obtained from the transducer manufacturer are used in these preliminary prototypes to allow measurements of performance. The transducer specification, as well as data from transducer sample measurements, can be used to form an equivalent circuit of the transducer (unless one is available from the manufacturer).

The bottom branch leading to the ”simulations” box (”measuring equipment”/ ”other objects” to ”general-purpose simulation models”) is the part of the development work that will be overviewed in this chapter. The ”general-purpose simulation models” collection is the result of this work (appendix IV). The middle branch (”transducer spec.”/ ”transducer samples” to ”transducer equivalent circuit”) was handled by the component manufacturer in the practical case viewed in this thesis (chapter 8), and the results (”transducer equivalent circuit”) are shown in appendix V. Most other parts of this diagram also are exemplified in the case study in chapter 8. Finally it should be noted that all parts of the flowchart will probably not be active at the same time; some subprocedures are most essential at earlier stages of a project while others are more important later. The ”general-purpose simulation models” branch can be a once-and-for-all procedure that does not have to be repeated once the basic models are finished.

5.4 Implementation

In this section, the selection, parameter identification, modelling, shape selection and macro implementation will be briefly outlined for each macro (all macros are divided into 11 categories for clarity). Appendix IV contains the full descriptions of all the created macros, in the same order as below. As mentioned, the macros are only outlined here. Lack of space precludes detailed descriptions, and more information must be sought in appendix IV, as well as in appendix III containing derivations of some of the models.

5.4.1 Global definitions

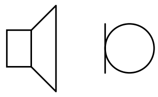
Micro-Cap needs to know values of relevant physical constants such as air density, speed of sound etc. These are defined in a separate text file (ACOUDEF.TXT). Another text file (TUBEDEF.TXT) includes definitions of mathematical functions that are used by some of the macros.

5.4.2 Shape and naming conventions

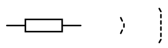
There are some general symbols that are used in several macro shapes to further describe the properties of the objects that the macros are modelling:

E M A R
1 2

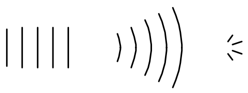
Letters denote domains. Numbered ports on a two-port means that the component in question is nonreciprocal.



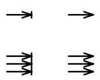
Loudspeakers and microphones are depicted using standard symbols.



General impedances are depicted by the ”box” resistor symbol (pure resistances use the ”zig-zag” symbol). Radiation impedances are shown as wave fronts (dashed lines, to distinguish them from actual radiation that uses solid lines). Different symbols are used for spherical and plane waves.

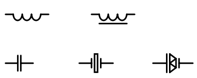


Radiation is shown as wave fronts (spherical or plane waves). Zero sound pressure (typically: opening to static atmospheric pressure in a circuit) is shown as flow lines of pressure escaping into free air.

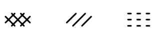


A single arrow, resembling a vector, means a mechanical quantity (force, velocity) or electrical current. Triple arrows, resembling flow lines of a fluid, indicate an acoustical quantity (pressure, volume velocity). The cross-line is used in the force and pressure shapes to show that they are acting on a body or surface.

These shapes symbolize displacement (mechanical domain) and volume (acoustical domain).



Coils are used to symbolize dynamic transduction (coil with core = electromagnetic transducer). Standard capacitor, crystal and electret shapes depict electrostatic transduction (crystal = piezoelectric transducer, electret = electret transducer).



Patterns used to represent (from left to right): rigid physical structures in the mechanical domain, rigid physical structures in the acoustical domain, damping material.

Micro-Cap V stores macros as separate files, and the operating system restricts file names to a maximum of 8 characters plus extension (.CIR). For this reason a number of abbreviations were created:

Z	Impedance (general)
R	Resistance
L	Inductance (incl. mass = mechanical inductance)
C	Capacitance (incl. compliance = mechanical capacitance)
Prb	Probe
Cir	Circular
Rct	Rectangular
Pln	Plane
Sph	Spherical
AirL, A	Air load

EndC, E	End correction
Rad	Radiation
Disp	Displacement
Vel	Velocity
Vol	Volume
VolVel, VolV	Volume velocity
Dyn	Dynamic
Es	Electrostatic
Em	Electromagnetic
Elct	Electret
Pz	Piezoelectric
Int	Interface
Mic	Microphone
Spkr	Loudspeaker
B	Baffle

5.4.3 Parallel connection

In practical designs, several equally dimensioned mechanical structures are often seen ”in parallel”. The most typical case is a mobile phone or handset cover, with several small holes as sound outlets for the earpiece capsule underneath it. An array of identical holes in parallel, with equal pressure and volume velocity in each hole at every instant, can be modelled as one single impedance equal to the impedance of one hole divided by the total number of holes. This principle of viewing similar parallel structures or phenomena as one single entity is used in several other macros (e.g. air load, end corrections). All of these have the number of objects (no) as a parameter.

5.4.4 Impedances

The built-in impedance components (resistors, inductors, capacitors) may as well be used directly without any need for macros. Micro-Cap allows the creation of components that are defined as built-in components, but have custom shapes attached to them. However, a problem is encountered with some of the components due to the use of the inverse (admittance) analogy in mechanics: mechanical impedances will show up in the circuit as their duals. This means that a mechanical resistance shows as a reciprocal-valued resistor, a mechanical inductance (mass) shows as a capacitor, and a mechanical capacitance (compliance) shows as an inductor. While the two latter cases can be handled without macros, it was still decided to use macros to implement all the impedances in the mechanical and acoustical domain for the sake of consistence.

The definitions of the impedance macros are self-explanatory. Shapes were chosen to depict the typical appearances of the impedances in a clear and intuitive way. Similar symbols also have been used in the literature, e.g. [6, p. 53-55; 14, p. 83].

Figure 5.2: Impedance shapes (top row: acoustical impedance, resistance, inductance and capacitance, bottom row: the same components in the mechanical domain).

The mechanical inductance (mass) macro Lm is the only one with just one pin instead of two. The macro is internally grounded, and the ungrounded terminal is connected to the circuit. This is because the velocity (voltage) of the mass is, by definition, referenced to the inertial frame (zero potential = ground). Also, a mass is rigid and no velocity (voltage) differences can exist, implying again that a mass has one single node in the circuit.

5.4.5 Sources, transforming two-ports and grounds

It is clear from the preceding chapter that the ideal voltage and current sources of the Micro-Cap simulator are directly usable as force, velocity, pressure and volume velocity sources (in corresponding domains). Micro-Cap (like most other simulators) defines several types of voltage and current sources, ranging from independent sources to piecewise linear, Laplace and file sources [10]. In order not to have too many graphical shapes for all possible sources, a choice was made to use just four shapes (force, velocity, pressure and volume velocity sources). These sources were instead made voltage-controlled, i. e. connectable to other Micro Cap voltage sources. This allows any of the many available Micro-Cap voltage sources to be used to generate the desired force, velocity, pressure or volume velocity.

In addition to simple sources, the domain borders need transforming two-ports (ideal transformers and gyrators), implemented here as two combined controlled voltage or current sources. See sec. 4.3.5.

Every circuit needs a ground. Ground means zero potential, which is directly interpretable as zero velocity in the mechanical domain, and zero sound pressure in the acoustical domain. In other words, ground means an immovable point (inertial reference frame) in the mechanical domain, and equilibrium (atmospheric) pressure in the acoustical domain. For this reason, special shapes were created for mechanical and acoustic grounds.

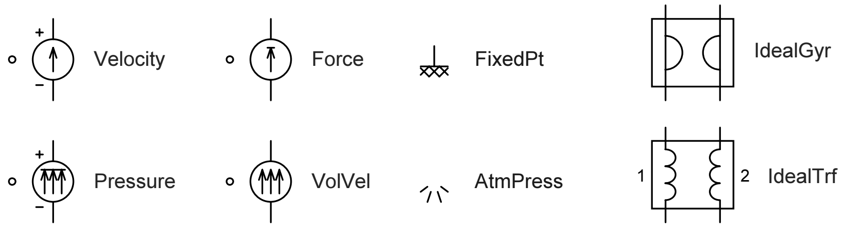


Figure 5.3: Velocity, force, pressure and volume velocity source, immovable point and atmospheric pressure, ideal gyrator and transformer.

The transformer has its ports numbered, while the gyrator has not. This is to show that the transformer is nonreciprocal, as opposed to the gyrator. The boxes around these are meant to show that the components are ideal, and to further distinguish them from the built-in ones. It should be noted that the AtmPress shape does not symbolize sound radiation (see below), although the shape could lead to confusion. However, no better shape was fancied, and radiation shapes were made very different from this one to prevent confusion.

Plus and minus signs indicate the polarity of the velocity and pressure sources, just as in the Micro-Cap built-in voltage sources. In the force and volume velocity sources, the arrows indicate the flow direction of the corresponding currents through the sources, just as in the built-in current sources.

5.4.6 Physical structures

This category of macros is very important, but also troublesome since it is impossible to model all kinds of existing mechanical structures using a limited set of models. Reasons for this can be found in the fact that one-dimensional lumped circuits cannot, in general, model three-dimensional distributed phenomena accurately throughout the audio frequency band. As a result, some basic structures have to be identified, and it helps if even these can be modelled with good accuracy.

Rigid mechanical structures create flow paths for the air in an acoustic circuit. Examples of such structures are cavities and tubes of varying length. The word *hole* will, hereafter, mean a flow path with a uniform cross section (circular, rectangular etc.) bounded by rigid walls, and short (in the flow direction) compared to the sound wavelength. A *duct* is defined as otherwise equivalent to a hole, but not acoustically short, i.e. possibly having a length comparable to or greater than the wavelength. In other words, a short tube will be called a hole, while a long tube will be called a duct. Although ”hole” is perhaps not the best word for a short tubular connection, it is used here because it associates to phone and accessory mechanics, where small holes are extremely common as sound outlets. The openings at the ends of a hole or duct will be called *apertures* (or simply *openings*). In other words, an aperture is the hypothetical surface that, together with the walls, bounds the volume of air inside the hole or duct. There are also non-rigid mechanical structures encountered in acoustic circuits, such as more or less elastic membranes (used commonly as dust and water shields).

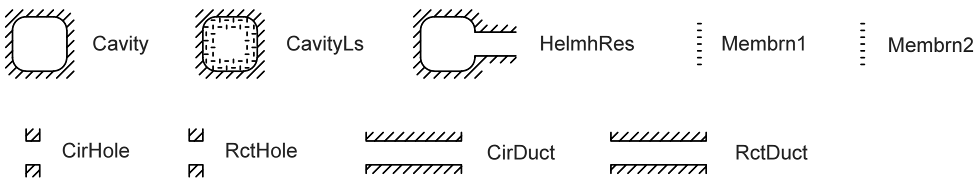


Figure 5.4: Cavity (loss-free and lossy), Helmholtz resonator, membranes, circular and rectangular hole, circular and rectangular duct.

5.4.6.1 Cavities

In a *cavity*, the predominant physical phenomenon is compression and decompression of an enclosed volume of air. When restricting the attention to small enough pressure amplitudes, so that air compression can be considered linear, the spring-like action of the compressed air will appear as a constant acoustic capacitance. In modelling an air cavity this way, it is assumed that its dimensions are small compared to the wavelength, but not small compared to the *thermal boundary layer* thickness. (If the cavity is small compared to the thermal boundary layer thickness, then conditions will be isothermal and the apparent capacitance will be up to 40 % higher [15].) The static atmospheric pressure is taken as the reference, which means that a capacitance due to a cavity always appears grounded. All entrances to a single cavity are, in effect, connected to the same node in the equivalent circuit. This is because the pressure has to be equal at each entrance, which, in turn, is because the dimensions of the cavity are assumed to be clearly smaller than the wavelength, implying that no appreciable phase differences can exist.

The one and only parameter supplied to the Cavity macro is the volume of the cavity. This volume defines the corresponding capacitance according to eq. (4.4) in the preceding chapter, with the compressibility of air as the proportionality coefficient. Most real cavities have losses, and a lossy cavity model (CavityLs) was also included in the macro collection. This macro differs from the loss-free cavity in that it has two additional resistors, one to account for leaks and the other to account for losses in, for example, filling material that may be used to damp the cavity. Filling material is common in loudspeaker boxes (hands-free loudspeakers). Unfortunately, it is hard to create an accurate and versatile model of a filled cavity. The physical properties of the fibres in the filling material are seldom known, there are many parameters, and measuring them may not be worth the effort. A simple approach was chosen here, with just two time constants (one for the filling and one for the leaks) as additional parameters characterizing the losses. A fourth provided parameter is a coefficient to characterize the apparent volume increase (about 40 % maximum, as stated above) that is due to the thermodynamical properties of the filling material. It may still be difficult to know what values to use, but at least there are fewer parameters than would be needed in a more general case.

5.4.6.2 Holes

A hole contains a volume of air, bounded by the rigid inner walls of the hole and the apertures at the ends. When sound enters such a hole, the air volume will vibrate with negligible compression. This is because the hole is, by definition above, much shorter in the flow direction than the wavelength, i.e. no appreciable phase differences can appear inside the hole. However, there is an inertance (acoustic inductance) due to the moving mass of air, and viscosity losses appear as an acoustic resistance. A hole thus is represented by a series RL (resistance-inductance) branch in an equivalent circuit, with two nodes (the ends of the hole) connecting to the outside acoustic circuit.

For physical reasons, the flow profile of the air inside a hole shows a frequency dependence. At low frequencies (assuming a circular cross-section), the flow profile is parabolic with zero flow velocity on the wall and maximum on the center axis. As the frequency rises, the *viscous boundary layer* of air (whose movement is restricted by the viscosity) narrows, and a greater portion of the air moves with about the same velocity as the air on the center axis. The resistance is constant below a certain threshold frequency, and rises proportionally to the square root of the frequency above it. The inductance shows a transition (4:3 in circular holes, 6:5 in very wide rectangular holes) around the same threshold frequency [6, pp. 135-137; 15; 16, p. 496].

Holes may have various cross-sectional shapes. Calculating explicit RL component values for any arbitrary shape and frequency is impossible. It is more realistic to model just two basic cases (circular and rectangular cross-section, CirHole and RctHole). Real-world holes may then be approximated by one of these two. A circular hole is a fair approximation of all holes that are elliptical, square-shaped etc, as long as the cross-sectional area is equal. There are also cases when the ratio between height and width of the hole is far from 1:1. For these cases a model of a rectangular hole was created, with height and width as parameters instead of the diameter. The length in the flow direction (from aperture to aperture) is needed as a parameter by both hole macros.

Although accurate formulae are available for circular holes and infinitely wide slits, these involve complex Bessel and tangent functions that the Micro-Cap simulator cannot handle. For this reason polynomial approximations were created using numerical optimization. Also, reasonable modifications to models available in acoustics literature were made to allow arbitrary rectangular hole aspect ratios (not just ratios far from unity, as most published models assume).

5.4.6.3 Ducts

Ducts are modelled by the CirDuct and RctDuct macros, both of which need the same dimension parameters as the corresponding hole macros described above. It has already been mentioned that a tube not short compared to the wavelength must be treated as a transmission line. The frequency dependence of the acoustic impedance of a hole (short tube) will show up in ducts (long tubes) too, since a long tube can be viewed as a series connection of many shorter tubes. All this means that a realistic duct model should consist of a transmission line with frequency-dependent distributed impedances: series resistance accounting for viscosity, series inductance accounting for moving air mass, shunt capacitance accounting for air compression, and shunt resistance accounting for thermal diffusion losses [15].

Micro-Cap can handle frequency-independent transmission line parameters only, and so a good duct model is out of reach. Also, no more than two or three (depending on line type) of the four distributed impedances may have nonzero values [10, p. 372]. The RLC transmission line type was chosen, and the shunt resistance had to be left out. This introduces a downward error in the real part of the propagation constant (i.e. damping). To counteract this, the distributed series resistance can be raised to provide a correct damping, but then, of course, the series resistance will be too high. Both these conflicting errors are in the order of 40 %. A compromise is inevitable. An extra parameter was added to give the user some control over how the duct should be simulated. Finally, even the frequency of interest has to be supplied as a parameter to allow Micro-Cap to calculate distributed impedance values at one single frequency, and use the same static values over the whole simulated frequency range (with more or less inaccurate results as a consequence).

5.4.6.4 Other structures

The Helmholtz resonator (typically, a closed cavity connected to a tube) appears more or less disguised in many acoustic circuits. It is easy to simulate a Helmholtz resonator by using the cavity and hole macros together, but in some cases one wants to try out a Helmholtz resonator with given acoustical properties without caring much about the dimensions. The HelmhRes macro is created with these cases in mind, and it takes the acoustical properties (resonance frequency and quality factor) as parameters. One more parameter is needed in order to determine a unique equivalent circuit, thus the cavity volume is given as a third parameter. Air volume is a very limited resource in small phones and accessories, and restrictions on cavity volumes are easily known, so the choice of volume as the third parameter is motivated.

The final structure modelled here is the membrane. Membranes used in acoustic circuits are usually porous, which means that they let more or less of the sound through. Typical examples in mobile phones are dust and water shield membranes attached to sound outlets. Such a membrane has a flow resistance (due to the porosity). Also, unless the membrane is rigid, there is a moving mass, and a compliance associated with the stiffness. Finally, the mechanical losses in the membrane show up as a resistance. In many typical cases the flow resistance will be the only relevant parameter, while the mechanical behaviour of the membrane can be neglected. The compliance cannot be accurately modelled using Micro-Cap, since in real-world cases it might be strongly nonlinear at high volume velocity amplitudes, and Micro-Cap cannot simulate nonlinear behaviour in the frequency domain. This nonlinearity is due to the fact that (as is usually the case) a non-elastic membrane fastened by its edges, without tension, will allow movement up to some displacement limit, but strongly resist movement beyond this limit.

There are two membrane models in the macro collection. One (Membrn1) is simple and models just the flow resistance, accepting the specific acoustic resistance as a parameter. The other membrane model (Membrn2) is more complex and models also the effects of the finite stiffness of the membrane. The three mechanical parameters (resistance, moving mass, compliance) of the mounted membrane are given separately to the macro, as well as a few parameters describing the geometry of the pores. The area and the number of parallel membranes are given as parameters to both membrane macros.

5.4.7 Air load

The concept of radiation impedance was introduced in the beginning of this chapter. It is clear that radiation impedance will show up whenever sound is radiated by some source. Now it is important to note that there is no difference between an aperture (as defined above) and a real mechanical surface of the same shape, at the same location, and moving with the same velocity as the air particles in the aperture, as far as radiation is concerned. This means that the same radiation impedance will be seen as a load by both these sources, as can be understood from the definition of acoustical impedance. From now on, a source of radiation (of any type) is called a *radiator*. This means that both apertures and moving surfaces may act as radiators in acoustic circuits, and macros modelling radiation impedance must be connectable to these.

Dimensions of radiators dealt with in this thesis (especially phone cover sound outlet holes) are small compared to the sound wavelength, implying omnidirectional sound radiation. This simplifies the radiation models, and the radiation resistance becomes effectively independent of the geometrical properties of the radiator. Although proportional to the square of the frequency (see appendix III), radiation resistances will tend to be small compared to other (typically, viscous) resistances, except perhaps at the highest frequencies. Viscous resistance appears not only inside a tube, but also at the surrounding surfaces outside the tube close to the apertures [17]. This resistance must be included in simulations of sound radiation to maintain accuracy. Flow lines of vibrating air entering a space through a smaller opening spread out gradually, not abruptly. This means that there is a kind of smooth transition region in which some of the free air outside, and close to, the opening can be thought to vibrate together with the air in the opening. The situation is the same for any radiator. Hence the viscous losses outside the radiator. An additional consequence of this is that a certain mass of *attached air* has to be included as an acoustic inductance. This inductance forms the radiation inductance, the reactive part of the radiation impedance.

Now it is obvious that a radiator will see two loads: an impedance due to the radiation (with a resistive part accounting for radiated power and an inductive part accounting for the attached air mass), and a resistance due to viscous losses in the immediate vicinity of the radiator. The combination of these loads will hereafter be called *air load*, to distinguish it from the radiation impedance (which, as defined here, does not include the viscous losses). In other words, air load is the sum of radiation impedance and viscous loss resistance (two acoustic resistances and one acoustic inductance all in all).

Air load could be readily included in other macros. However, this would lead to an inconvenient number of combinations: it is not desirable or even possible to build one general model, with convenient input parameters, for all kinds of air load. Also, there are many possible radiators: tubes of various cross-section, loudspeakers, or other vibrating surfaces. For this reason it was decided to create separate macros modelling nothing but the air load.

These can, in turn, be connected to any macro modelling a radiator of some kind (e.g. a hole or a loudspeaker).

EndCCir EndCRct

AirLCir AirLCirB AirLRct AirLRctB

SphRadR PlnRadR

Figure 5.5: End correction for circular and rectangular radiator (top row), air load on circular and rectangular radiator, any solid radiation angle, or equipped with finite baffle (middle row), radiation resistance of radiator radiating spherical or plane waves (bottom row).

5.4.7.1 Air loads for specific radiator geometries

Full air load models were created specifically for circular and rectangular radiators (AirLCir, AirLCirB, AirLRct, AirLRctB). Details of the derivations and principles behind the models can be found in appendix III. The four full air load models are capable of modelling the air load on one or more equivalent radiators, lying in the same plane, more or less separated from each other. The wall surrounding the radiator(s) may be plane, it may form a conical horn (infinite extent assumed) with an arbitrary solid opening angle, or there may be no wall at all. Two of the macros (AirLCirB and AirLRctB) are specifically for cases when the surrounding wall (baffle) is plane and of finite extent, and the resultant transition in the air load is to be modelled. Of course, all real-world baffles are of finite extent anyway, but in many cases it is accurate enough and more convenient to assume infinite baffle dimensions.

All four air load macros are simply series connections of the three components mentioned above: radiation resistance, radiation inductance (= attached air inertance) and viscous drag resistance. The differences lie only in how the macros calculate component values from given physical dimensions, and how these values depend on frequency. To be able to calculate the air load, the macros need the dimensions of the radiator(s) that they are connected to. Micro-Cap macros cannot read the parameter values of other adjacent macros in the schematic, so the dimensions have to be given as parameters the usual way. This means diameter for circular radiators, and width and height for rectangular radiators. AirLCir and AirLRct need two more parameters to describe the environment surrounding the radiator(s): a coefficient describing what portion of full space the radiation is into, and a coefficient characterizing the strength of the mutual coupling. Mutual coupling appears when two or more radiators are located close to one another, and it has the effect of an added *mutual radiation impedance*. AirLCirB and AirLRctB take the baffle perimeter as input instead of the solid radiation angle coefficient just mentioned.

5.4.7.2 End corrections

There are cases when the air load models described above are not necessary, or not even the best choice. Good examples are irregular geometries that are hard to specify using the needed parameters, cases when less accuracy is enough, or cases when some complicated frequency dependence appears that cannot be modelled using the previous four air load macros. Also, for compatibility with common acoustics models, it is good to include a couple of more straightforward air load macros.

In tubes, the effect of the viscosity and inertance parts of the air load is simply to increase the total acoustic impedance above that associated with the air inside. For this reason the viscosity and inertance can be included in a hole or duct model as what is known as an *end correction*. An and correction is an imagined length extension, usually expressed as some fraction of the tube diameter (or the smaller of either the width or the height in a rectangular tube). To make calculation simple, it is usually assumed that the end correction is frequency-independent (this is in fact true at low frequencies). An inductance end correction of 0.42 times the diameter is usually assumed for a circular radiator when it is surrounded by an infinite baffle, and 0.30 times the diameter is the value used when there is no baffle. The resistance end correction is either neglected, or set equal to the inductance end correction when it is desired to model the viscosity effects [17, 24]. A resistance end correction is in effect a model of the viscous losses only (the radiation resistance is not constant but varies considerably with frequency, thus it cannot be modelled using a frequency-independent end correction). At low frequencies, for small radiators, the radiation resistance is outweighed by the viscous resistance, and the use of a simple end correction model without radiation resistance is motivated.

A couple of general end correction macros were created. These macros (EndCCir and EndCRct, for circular and rectangular radiators, respectively) need end correction coefficients as parameters (in addition to the surface dimensions of the radiator they are to be connected to). Coefficients are given separately for the resistive (viscous) and inductive (attached air) parts, for the sake of versatility. The macros even can be given frequency-dependent end correction coefficients to model more complicated air loads.

5.4.7.3 Radiation resistances

Radiation resistances were included as separate macros (SphRadR, PlnRadR) to allow assembly of custom air load models by combining them with custom end corrections. This way all three parts of the air load can be modelled, just as the full air load macros do. Also, it is simple to calculate (simulate) the acoustic output power of a radiator when the radiation resistance is directly accessible in the schematic.

The radiation resistance of a radiator radiating spherical waves is independent of radiator dimensions (as long as the radiator is small compared to the wavelength), and no dimensions are needed as parameters [18, p. 165]. However, in some cases, such as when sound radiation is into a narrow tube or cavity or the frequency is very high, the waves are not spherical but plane. The radiation resistance is then also different from the spherical wave case, and a separate macro is needed. Now the radiation resistance depends on radiator area, and the area has to be given as a parameter.

5.4.8 Domain interfaces

Having four different domains means that at least three types of interfaces are needed. Now one of these, the acoustical-radiation domain interface, will be given special treatment here. This is because of its special three-dimensional transmission line nature, making the macros more complicated than those for the two other interfaces. Also, the acoustical-radiation domain interface can be seen as belonging to the group of macros modelling radiation as a phenomenon, so it will be viewed separately in the next section on radiation macros.

The two other interfaces are very straightforward, however. They are simply transforming quantities by multiplying or dividing them with some characteristic constant, possibly changing their nature from a voltage to a current or vice versa. In a dynamic transducer, the electrical domain current is transformed into a mechanical domain force, with a proportionality coefficient equal to the transducer force factor. At the same time, the mechanical domain velocity is transformed into an electrical domain voltage, with the same proportionality coefficient. This was expressed above by eqs. (3.1).

The situation is analogous in an electrostatic transducer, but then it is the voltage that is transformed into a mechanical domain force, and the mechanical domain velocity that is transformed into an electrical domain current. Now the proportionality constant is dependent on the membrane to back plate distance, membrane area, and polarization voltage. Thus there are two basic kinds of electrical-mechanical domain interfaces. The dynamic type has the properties of a transformer, while the electrostatic type acts as a gyrator (recalling again that the inverse, admittance, analogy has been chosen for the mechanical domain).

Going from the mechanical to the acoustical domain involves transforming a mechanical domain velocity into an acoustical domain volume velocity (which means multiplying the velocity by area). An acoustical domain pressure transforms into a mechanical domain force, multiplying again by the same area. This is equivalent to saying that a mechanical-acoustical domain interface is a gyrator with the area of the connecting surface as the proportionality constant.

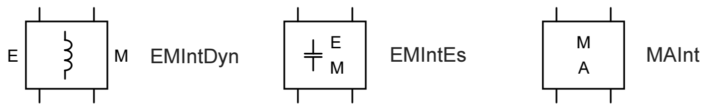


Figure 5.6: Electrical-mechanical domain interface for dynamic transduction, electrical-mechanical domain interface for electrostatic transduction, and mechanical-acoustical domain interface.

The three interfaces mentioned are very simple as macros, containing just the needed ideal transformer or gyrator and the one necessary parameter. For the electrical-mechanical domain interface of the dynamic type (EMIntDyn) it is the transducer force factor. The electrostatic type (EMIntEs) needs a similar constant that can be calculated from the transducer dimensions and the permanent charge or polarization voltage. Finally, the MAInt macro needs the effective area of the surface (real or hypothetical) connecting the mechanical and acoustical domains. The effective area of a moving rigid surface is the area of the projection of the moving surface on a plane perpendicular to the direction of movement. If the moving surface is instead a flexible membrane fastened by its edges, the effective area will be smaller than the total moving area.

It should be noted especially that modelling complete dynamic or electrostatic transducers generally requires additional components in both the electrical and the mechanical domain, such as voice coil resistance and membrane to back plate capacitance. Electrostatic transduction will not be simulated in this thesis.

5.4.9 Radiation

Radiation models are made necessary by the fact that the free air is an essential part of many of the systems to be simulated. Free-field radiation is not involved when simulating the sound pressure created by an earpiece held close to the ear, but it comes into play e.g. when checking the performance of a hands-free loudspeaker. Furthermore, radiation into free air cannot be described using simple lumped circuits in the acoustical domain. The reason is that distances involved are typically long enough to make the finite propagation velocity and wavelength of sound visible in the results. This is no problem if just the free-field sound pressure of a loudspeaker is simulated. However, it will lead to wrong results if some kind of feedback (with a microphone picking up the sound at a distance and feeding it back to the sound source) is used, since then the loop delay is an essential parameter.

A sound source whose dimensions are negligible compared to the wavelength, and whose points radiate in phase, will emit spherical waves just as an ideal point source. This requirement is usually fulfilled for all small sound sources encountered in mobile phones and accessories, except perhaps at the highest frequencies. It is therefore natural to have this type of radiation modelled by a few macros. The other basic case that comes to mind is plane wave radiation. Plane waves are commonly used as approximations of spherical waves when the sound source is distant (in the *far field*). Also, waves propagating inside a narrow duct are usually treated as plane waves.

According to the description in the preceding chapter, radiation is simulated using briefly a fourth ”radiation” domain, with volume velocity exchanged by particle velocity. The propagation delay suggests a transmission-line analogy. However, this would be possible only for plane waves in a duct. Spherical waves spread out, which makes the specific acoustic impedance distance-dependent, which in turn would require a nonuniform transmission line. It has already been pointed out that this lies beyond the capabilities of the Micro-Cap simulator. The result is that the radiation macros have to be made made nonreciprocal.

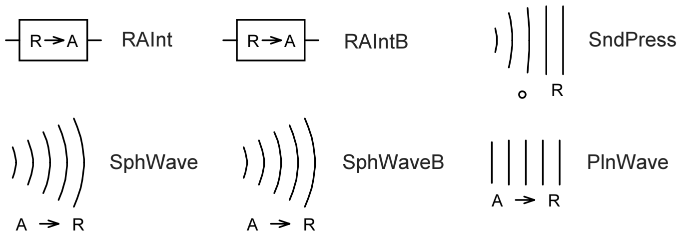


Figure 5.7: Radiation-acoustical domain interface (general and with finite baffle), sound pressure source, spherical wave radiation (general and with finite baffle), and plane wave radiation.

5.4.9.1 Spherical and plane waves

The air load macros described in sec. 5.4.7 were left ungrounded. It was also stated that this was done in order to let them be connected to radiation macros. This convenient solution, separating air load and radiation, was chosen in order to make the macros as flexible as possible. If actual radiation is irrelevant, an air load macro followed by ”acoustical ground” (atmospheric pressure AtmPress) is enough. If the radiated waves have to be simulated, then the air load macro is further connected to a radiation macro, and this radiation macro uses the resultant volume velocity of the radiator as its input in order to be able to determine the radiation. The chosen approach means that the air load macros account for all the load, and the radiation macros do not load the circuit but model the radiated waves only.

The specific acoustic impedance of the waves is modelled as an internal impedance. Two extra outputs are provided. One of these ”copies” the input volume velocity, to allow it to be used as input to another radiation macro. This provides a convenient way of simulating pressures at several distances from one single radiator. The meaning of the other output will be discussed below.

Radiation macros need the distance to the observation point as a parameter. Furthermore, the space that propagates the radiation must be described (resultant sound pressure will depend on the solid angle of radiation). SphWave and SphWaveB need the solid angle (given as a coefficient as in the AirLCir and AirLRct macros). PlnWave needs the area of the wave front as a parameter in order to be able to calculate the resultant specific acoustic impedance. SphWaveB simulates the effect of a finite baffle around the radiator, and thus the baffle perimeter is also given as a parameter. Parallel arrays of all these macros are made possible by having a ”number” parameter.

The input of the radiation macros belongs to the acoustical domain, but the output is essentially ”radiation domain”, i.e. input current is analogous to volume velocity, while output current is analogous to particle velocity. Hence the ”R → A” in the shapes, to show that all these radiation macros are in fact also domain interfaces at the same time. The arrow emphasizes that the macros are nonreciprocal, i.e. they cannot, for example, model what happens when waves reflect back and interact with the radiator. Usually this interaction is negligible in free air anyway, except for very close distances.

5.4.9.2 Radiation-acoustical domain interfaces

Another interface from the radiation domain to the common acoustical domain should be provided in order to allow simulation of complete feedback systems. Such an interface has to convert oncoming radiation, described as pressure and particle velocity, back to the pressure and volume velocity domain. At first, this would seem modellable by a simple transforming two-port that multiplies the velocity (current) by the area of the surface subject to radiation, while not changing the magnitude of the pressure (voltage).

However, difficulties arise since it must also be known whether the surface subject to radiation is situated in free space or surrounded by some reflecting baffle. A baffle doubles the pressure due to reflections. This is why radiation macros are implemented with two pressure outputs: one for free-field radiation and another that can be further connected to acoustic circuits (which must be done through the radiation-acoustical domain interfaces RAInt or RAIntB). The ”load” output of the radiation macros doubles the pressure to prepare to take possible reflections into account. The RAInt or RAIntB macros, in turn, are implemented in such a way that the correct pressures and particle or volume velocities will result, depending on whether or not there is a reflecting baffle around the surface hit by the radiated waves. RAInt is told about the existence or nonexistence of a baffle (assumed infinite) by a single parameter. RAIntB simulates (very roughly) the frequency dependence due to a baffle with finite dimensions, and it is supplied with the baffle perimeter as a parameter.

5.4.9.3 Other radiation macros

It seems unnecessary to have a dedicated sound pressure macro when an ideal pressure source could do the job, and this is also true as long as the inherent characteristic impedance of air does not affect the results. However, there may appear situations when the particle velocity is a quantity of interest, e.g. when dealing with microphones and wind shields. For these cases a special sound pressure macro, voltage-controlled just as the other ideal sources, was created. Moreover, it has the added benefit of double outputs, for both free field and loads, as the radiation macros.

5.4.10 Transducers

The most commonly modelled electroacoustic transducers are the dynamic loudspeakers. They have mechanical and electrical properties that are relatively easy to measure and specify. As a result of the common modelling of these transducers, many manufacturers have also become quite good at supplying necessary information in their product data sheets. Unfortunately, the situation is not as good for electromagnetic or piezoelectric loudspeakers. Microphones are usually much more ideal in their behaviour than loudspeakers (which is partly because of their smaller size). Dynamic microphones are an exception to this rule, but such microphones are not used in mobile phones and accessories anyway, and they will not be considered here. As a consequence of all this, it seems that one can manage well enough modelling just two transducer types: dynamic loudspeakers and electret microphones. As mentioned before, buzzers are very nonlinear in behaviour and trying to approximate their behaviour using a linear circuit simulator is probably not worth the effort.

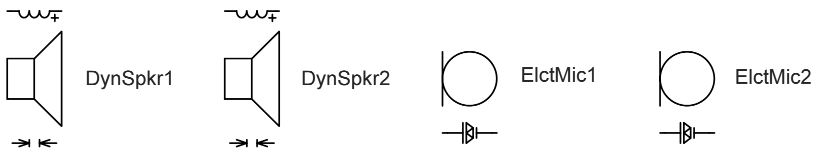


Figure 5.8: Dynamic loudspeakers, electret microphones (omnidirectional and pressure gradient type).

5.4.10.1 Dynamic loudspeakers

The provided dynamic loudspeaker macros DynSpkr1 and DynSpkr2 are nearly equivalent, except for the omission of voice coil inductance in DynSpkr2, and the different set of parameters. The former macro accepts directly the electrical, mechanical and acoustical properties of the loudspeaker, while the latter macro uses *Thiele-Small parameters* as described in [19]. Thiele-Small parameters are the ones usually supplied by manufacturers. Giving all the needed electrical, mechanical and acoustical component values directly might be more useful if simulation is to be run at a lower, more manufacturer-oriented, level. This could be the case if a custom component is under development and a change in some isolated property (e.g. membrane stiffness) is to be simulated.

Both loudspeaker macros model loudspeakers using one single resonant circuit in the mechanical domain. This may not be accurate enough at high frequencies, or when a greater part of the loudspeaker membrane is nonrigid (which is usually the case in the smallest dynamic loudspeakers). An example is the 13 mm earpiece capsule that plays a leading role in the case study later in this thesis. The employed model of this component, adapted from an equivalent circuit supplied by the manufacturer (Philips Speaker Systems Ltd, Austria), can be found in appendix V.

5.4.10.2 Electret microphones

Data on the electroacoustical properties of small electret microphones is usually very sparse. From a modelling point of view, an electret microphone can be seen basically as a simple combination of a very high acoustic input impedance (due to the small area and high tension of the membrane), a pressure-controlled voltage source (the built-in preamplifier), and an electric output impedance (which is also due to the preamplifier). The basic approach chosen in both microphone macros (ElctMic1 and ElctMic2) assumes a negligibly high acoustic input impedance. The sensitivity (given in data sheets) is accepted as a parameter together with the parallel RC output impedance. Manufacturer data sheets may give these for a single frequency (usually 1 kHz), but the macro assumes that there is no frequency dependence.

The two macros differ in the number of connections to the membrane. A conventional pressure microphone (represented by ElctMic1) has one side of its membrane open to the air, while the other side is enclosed in a small air cavity. A pressure gradient microphone (ElctMic2) has both sides of the membrane open to the air. The latter macro defines sensitivity as the sensitivity to pressure difference (front vs back of membrane). Although this is not the way that pressure gradient microphone manufacturers usually specify sensitivity, it was chosen here because it is convenient and requires fewer parameters.

5.4.11 Measuring equipment

Three standard ear simulators (artificial ears) were included in this macro collection. The equivalent circuits for these were obtained from technical data for the corresponding equipment from Brüel & Kjær, and this equipment follows the ITU-T P.57 standard. There are inherent inaccuracies in these circuits, especially at the highest frequencies. Nevertheless, it is extremely important to have macros modelling the ear simulators, in order to be able to compare measured and simulated results in a reliable way. Of course, equally important information about the acoustical performance of an earpiece can then also be obtained using pure simulation with no hardware at hand.

The three modelled types are the sealed ITU-T P.57 type 1 (Ear1), and type 3.2 low-leak (Ear32Lo) and high-leak (Ear32Hi) ears. A special leak adapter (”German leak ring”, manufactured by Systel Elektronik, Germany) was also included. This leak adapter connects to the Brüel & Kjær 4185 (type 1) ear simulator, and allows measurement of what happens when an earpiece is not well sealed to the user’s ear. The type 3.2 low-leak ear simulator has a leak of roughly the same size and kind. The type 3.2 high-leak ear has a big leak and corresponds to cases when the handset or handportable in question is held loosely against the user’s ear. Running the same simulation or measurement with different ear simulators as loads will demonstrate the *leak tolerance* of the earpiece. This is one of the central topics in this thesis, and it is deferred to the later chapters (6-8) dealing with it.

None of the ear simulator macros needs any parameters. The macros simply accept a connection to the acoustic circuit at their input, and their output gives the pressure that would be sensed by the microphones inside the ear simulators. The leak openings are available as separate pins in Ear32Lo, Ear32Hi and GermLk, for special simulation cases.

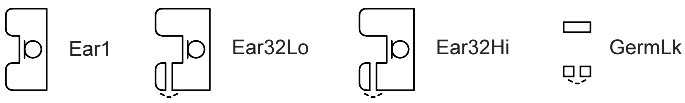


Figure 5.9: ITU-T type 1, type 3.2 low-leak, type 3.2 high-leak ear simulators, leak adapter for the type 1 ear simulator.

It should be noted that the German leak ring is not a standardized leak adapter (in the sense that it is not a part of type approval specifications). For example, GSM type approval did not allow any leaks at all in the past, but the situation changed during the writing of this thesis. Nowadays GSM type approval allows type 3.2 ear simulators. But even before that the German leak adapter was a useful tool in measurements of leak tolerance and hence it is included in the model collection developed in this thesis. Another very important fact is that the type 3.2 ear simulators are always compensated (using a standard equalization curve as described in [13]) in type approval measurements. A model of this equalizer is not included in this thesis.

5.4.12 Probes and annotators

Most of these macros are not much more than a convenient addition to the graphical capabilities of Micro-Cap. For example, a velocity probe in a schematic can be used both to annotate a given velocity, pressure or other quantity (which is good for documentation purposes) and to reference the given point in the circuit in a very convenient way when running a simulation. The probe macros sense the magnitude and phase of the given quantity, and output the result as a voltage without interfering with the circuit in any way. This makes them useful in building controlled sources (the force, velocity, pressure and volume velocity sources viewed above were made voltage controlled). Moreover, mistakes resulting from the user forgetting which quantities are voltages and which ones are currents (or integrated/differentiated voltages or currents) can be prevented. Some of the probe macros have built-in derivators/integrators.

Three of the probe/annotator macros have built-in voltage or current sources to allow them to act as simple impedance probes. Checking an impedance at some point can be very useful during a simulation (to verify that the equivalent circuit makes sense, to optimize some part of it, to regain a picture of the magnitudes of resultant acoustic impedances, etc.). Of course, the probe macros do not model anything in the sense that most other macros do.

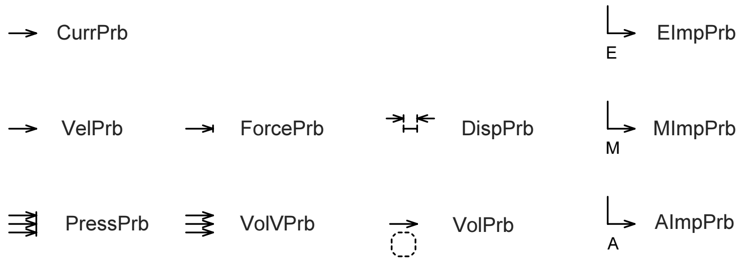


Figure 5.10: Probes/annotators for: current, electric impedance (top row), velocity, force, displacement, mechanical impedance (middle row), pressure, volume velocity, volume, and acoustic impedance (bottom row).

5.4.13 Compound macros

The group of compound macros consists of often used combinations of macros, e.g. the obvious combination of an air load and a radiation macro. The shapes of these macros are also simple combinations of the other shapes, and they show clearly why and how the macro shapes were designed to fit together. The only advantages provided by the compound macros is that some common parameters have to be given only once, and drawing schematics becomes a little quicker. This group could be extended by adding other common macro combinations.

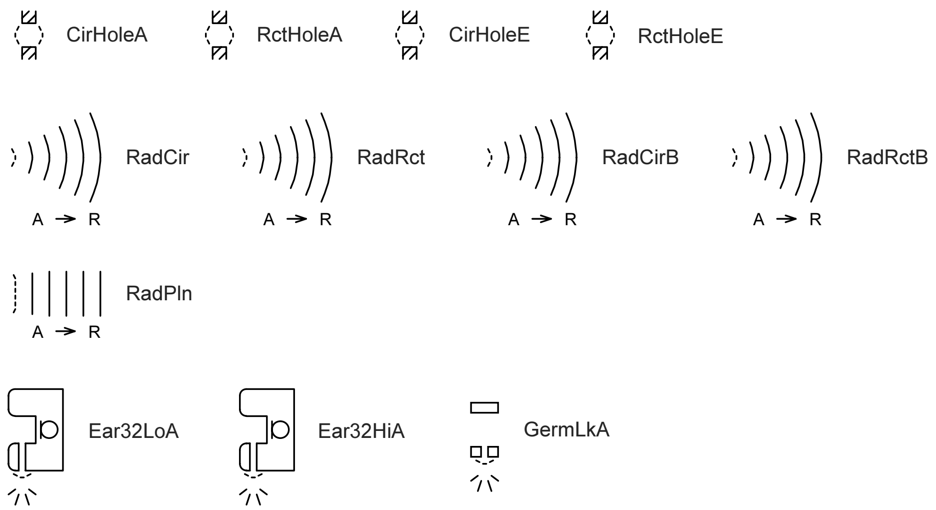


Figure 5.11: Holes with air loads and end corrections (top row), air loads and radiation of sources with various shapes (middle and bottom row).

5.4.14 Other macros

The rest of the macros (two combined derivators/integrators and an IEC 711 coupler model) are used internally by some of the macros viewed above. However, the DispVel and VolVolV macros have some value of their own, too. Although their equivalent circuits are identical, they have different shapes and parameter names. Examples of their use can be found in some of the probe and annotator macros (see appendix IV).

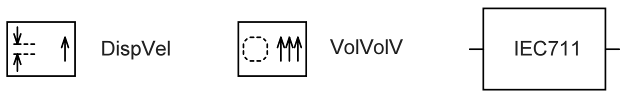


Figure 5.12: Displacement-velocity and volume- volume velocity converter, IEC 711 model.

6 Special audio problems and solutions

Some important and common problems influencing acoustics design will be briefly overviewed in this chapter, both as such and from a simulation viewpoint. These facts may sometimes even be forgotten, and the associated troubles caused by not taking them into account may not show up at all in early prototypes and computer simulation. A few of the points mentioned here will be exemplified in chapter 8 which describes a commercial Nokia phone.

6.1 Mechanics and mass production

6.1.1 Saving size

The constant desire to reduce size appears in many direct and indirect ways as a restriction on audio performance of mobile phones. Facts that have to be taken into account include:

- Reducing the effective area of a transducer lowers its sensitivity analogously (if other properties are kept unchanged).
- Reduced dimensions usually lead to narrower tolerance limits that must be taken care of in specifications and in the production process.
- Transducers have to be mounted closer to each other in a small handportable, which brings the buzzer closer to the ear, and the earpiece closer to the microphone. The former is a potential product liability risk, the latter can lead to more severe echoing.
- Making the handportable shorter brings the microphone farther away from the mouth (unless a flipping or sliding cover is used). Again, this is likely to make the echo relatively louder.

A small earpiece capsule or buzzer has a smaller membrane area to move air, and the loss in area must be compensated for by an increase in displacement to keep the volume velocity unchanged. Thus if the size of a given earpiece capsule or buzzer is to be reduced, one cannot expect the output to be at the same level unless the membrane can handle the needed increase in displacement. This problem is most pronounced at low frequencies (i.e. around 300 Hz for an earpiece capsule), where the displacement has a maximum.

The consequences of tolerances and more pronounced echo are dealt with below. The fourth and last statement above is a natural consequence of the behaviour of the typical phone user: keeping the earpiece against the ear prevents the microphone from being close to the mouth at the same time. This is no problem if the handportable is extended during a call, by flipping up or sliding out an extra cover, and the microphone is mounted on that cover. However, long distance to the mouth, perhaps combined with low microphone sensitivity, requires more electric gain in the microphone preamplifier. Noise and interference from the RF parts of the phone are then also amplified. A lot of care must be taken in the circuit design and layout as a whole to prevent this interference from disturbing the audio performance. It always helps if microphone amplifier gains can be kept as low as possible.

6.1.2 Mechanical tolerances

Although mobile phone and accessory audio specification limits are not as tight as those required for measuring or hi-fi equipment, a badly planned mechanical construction can easily cause some of the units in a production series to be out of specifications. Tolerances in the mounting of parts inside the phone housing can be surprisingly high, especially if the construction is such that several separate tolerances accumulate at points that are critical from an acoustics viewpoint. Transducer tolerances add to these mechanical tolerances.

Mechanical dimensions are convenient as input parameters to simulation macros. Fortunately, this makes simulation of tolerance effects easy, provided of course that the tolerances themselves are known. Simulation of mechanical tolerances can, in fact, be seen as one of the most important and valuable tasks in the simulation of mobile phone acoustics. The reliability of tolerance simulation can be good, since one is focusing on relative variations in well-known properties. Care must be taken not to forget to add the tolerances of the transducer component itself, at least if the component is a loudspeaker. Microphones tend to be more ideal in their behaviour.

Monte Carlo analysis, available in all decent simulators, can do the job. This analysis principle involves repeated simulation runs with random changes to all parameters subject to tolerances specified by the user. Micro-Cap V has built-in Monte Carlo analysis support for many component properties, but unfortunately such is not the case for macro parameters. This makes parameter stepping the method of choice when using Micro-Cap macros.

It helps in the design and implementation process if simulation can be used to locate and eliminate critical spots in the mechanical assembly or transducer specifications. Simulated tolerance effects are important indicators of successfulness in the evaluation of possible acoustical solutions. The tolerances shall, of course, be related to the type approval limits in order to prevent costly changes at the late type approval stage.

6.1.3 Industrial design

Industrial designers deal with the shapes of phone covers, keys, display lenses, and other characteristics visible on the outside of the phone. Sound outlets for the earpiece are a part of these characteristics too. In many cases the shape, number and location of the front holes has to be a suitable compromise between good industrial design and good acoustics design. It is worthwhile to establish a dialogue between these two at an early stage in the product development phase to prevent unnecessary or bad compromises. Environmental conditions and dust and water shielding (dealt with below) must also not be forgotten.

6.1.4 Mass production

The fact that mobile phones are mass produced in huge numbers complicates matters. The time needed for each step in the assembly process must be as short as possible (which is usually equivalent to requiring that the mechanical construction of the phone be as simple as possible). The actual mounting method of parts matters, i.e. which part is inserted where and in what order. For example: some clearance must be left around components that are to be mounted in crowded spaces (due to mounting tolerances), and parts such as rubber seals must be dimensioned and mountable in such a way that they are suited for rapid mass production assembly.

In order to keep the part count and cost to a minimum, acoustical solutions involving several special parts (e.g. cavities or tubes that must be separately assembled) are undesirable. Unfortunately, this also puts the acoustical designer in a situation where he or she has less control over how things develop during the product development phase. Some change in internal component layout made by printed circuit board designers may require immediate evaluation and feedback from an acoustics designer (for example if a big newly added component would accidentally be blocking some sound opening, or changing a critical internal air volume).

An interesting restriction arises from the fact that phone covers are cast in moulds. Although covers are made of plastic material and the moulds are metal, there will be some gradual wearing of the moulds. This may be a significant effect in the buzzer or earpiece sound openings. Narrow holes in the cover mean thin rods in the mould, and as these rods wear out, the front holes will become slightly smaller. The mould could even break in the long run if too narrow openings were specified and the cover is thick. Another important matter is the tolerances in the making of the mould itself. The finished mould may not have the exact dimensions specified in the mechanics CAD file, which is why solutions requiring dimensioning to an accuracy of just a few hundredths of a millimeter are better avoided.

6.2 Acoustics and the environment

6.2.1 Environmental conditions

Mobile phones, and the components contained within them, have to pass certain environmental tests to assure that they withstand usage in various situations without malfunctioning. Important parameters in such tests are humidity, temperature and vibration. How a phone behaves in a humid or wet environment is more or less influenced by the sizes and locations of openings in the covers, e.g. buzzer and earpiece sound outlets. From a water resistance point of view, several small openings are better than one big opening of the same total area. This is, to put it simply, the principle employed in porous dust and water shield membranes. A porous membrane is in effect a collection of a great number of very small openings, which prevents water entry (up to a certain maximum pressure) but lets sound pass more or less unattenuated. Dust and water shield membranes always have some specific acoustic resistance, and the choice of shield membrane is usually a best compromise between the resistance and the water entry pressure.

Shielding of the phone buzzer is especially problematic because a typical buzzer relies on acoustic resonance (a tuned Helmholtz resonator) to obtain the highest possible sound pressure level. If the resonance is sharp, i.e. has a high Q value (which is desirable), then it is also very sensitive to any acoustic resistances such as dust and water shield membranes.

Fortunately, the acoustic resistances of dust and water shield membranes are among the easiest to measure (at least at steady flow, zero frequency). The situation is more complicated at arbitrary frequencies, because the membrane can move slightly along with the oscillating air flow. Of course, this behaviour is strongly dependent on how the membrane is mounted. As a first rough approximation, a dust and water shield membrane may be represented in a simulation as a pure acoustic resistance equal to the specific acoustic resistance of the membrane multiplied by its flow-through area.

6.2.2 RF shielding

The presence of strong electromagnetic fields (radiated by the antenna) very close to sensitive baseband electronics has some impact on phone audio design as well. Proper shielding at the high frequencies used requires that there be no big openings in the conducting shields around sensitive electronic components and circuits. Here it is not the area that matters, but the longest dimension. Acoustic impedance, on the other hand, is most dependent on the cross-sectional area. So if an air flow route has to be provided through an RF shield, it is important to know that, ideally,

- openings should be circular in shape (a circle has the greatest possible ratio between area and longest dimension), and
- their diameter must be kept below some minimum.

If a big air flow route is needed, this opening may have to be split up into several small (e.g. circular or square-shaped) openings instead of one big opening, otherwise the maximum allowable dimension limit would be exceeded.

A flipping or sliding cover to protect the keypad is a potential source of RF shielding problems if the microphone is placed in the far end of the cover and has thus to be connected to the baseband electronics using a pair of long leads. The leads act as an unintentional antenna that picks up the RF field, which may cause harmful interference in the baseband audio circuitry.

6.2.3 Background and handling noise

Background noise is any sound, present on the location where the phone is used, that is in excess of the speech of the parties engaged in a call. The problems caused by background noise are missed calls (alerting not heard) and reduced or lost speech intelligibility (due to excessive background noise pickup of the microphone during conversation over the phone). Background noise pickup is further exaggerated by the compression techniques used in telephony to reduce voice sound level variations and to improve the signal to noise ratio. DSP algorithms are used to suppress picked-up background noise in digital telephony. A possible acoustical solution to the same problem is to use a pressure gradient microphone (mentioned briefly in chapter 3).

A pressure gradient microphone has the great advantage of being able to attenuate background noise in relation to speech emitted by the user's mouth. One of the reasons for this behaviour is the fact that sound coming from directions perpendicular to the axis of greatest sensitivity are strongly attenuated The other effect that reduces background noise pickup is the attenuation of spherical sound waves with distance. This means that the pressure amplitude difference between both sides of the membrane, for a given sound incidence angle, will be proportionally greater when the sound source (mouth) is closer to the microphone. A distant source of background noise will produce a proportionally smaller pressure amplitude difference, since the relative difference in path length from the source to the front and back of the membrane will be a lot smaller. Pressure gradient microphones are therefore also known as *noise-cancelling microphones*. For low-frequency sounds, the phase difference between pressures on both sides of the membrane will always be very small, resulting in almost complete cancellation when the sound source is distant. Such is not the case for sources close to the microphone, however, since then the 1/distance attenuation of spherical waves will predominate and create a considerable pressure amplitude difference. Thus if a pressure gradient microphone is equalized to have a flat response to (spherical wave) sources at a given distance, the net effect is that low frequencies will be boosted if the source moves closer to the microphone, and damped if it moves farther away. This is known as the *proximity effect*.

The situation of background noise preventing the phone user from hearing the alerting of an incoming call can only be improved by raising the sound level and/or spectral composition of the alerting signal, or by employing other kinds of alerting (vibration, flashing light, etc.). The latter method involves improvements in the phone user interface. The former method is about using a buzzer capable of creating a higher maximum sound level, which (usually) means a bigger buzzer component. It is impossible to use smaller and smaller buzzer components without having to sacrifice sound level, unless the maximum displacement of the buzzer membrane is enhanced proportionally (to keep the resulting volume velocity constant).

Simulating the background noise itself is typically not a relevant task. However, simulation is useful to some extent when optimizing the performance of the phone buzzer. The typical procedure is to enhance one of the harmonics produced by the alerting signal by including a tuned Helmholtz resonator in front of the buzzer. As the buzzer itself relies on pronounced nonlinearity (with the moving diaphragm or plate oscillating beyond its linear movement limits), it is not realistically simulable in the frequency domain using common linear circuit simulators. Fortunately, thanks to the large acoustic output impedance of a typical buzzer, the Helmholtz resonator can be tuned (at least to a first approximation) as a separate entity using simulation. If needed, measurements can aid in the final tuning.

There is one more, perhaps less obvious, source of interference: handling noise, i.e. strong pickup of noises caused by touching and scratching the phone during a call. Handling noises can probably be counteracted to some extent by improving the damping of mechanical vibrations around the microphone. The roughness and thickness of the phone surface, as well as possible air cavities beneath the surface acting as sound boards, play a role too. It should also be remembered that pressure gradient (noise cancelling) microphones tend to be more sensitive to handling noises than pressure (omnidirectional) microphones (the reasons for this will not be treated here).

6.2.4 Wind noise

Noise created by turbulence around the microphone can interfere more seriously with uplink speech than does loud background noise. Vortices created by wind turbulence are easily located closer to the microphone than the user's mouth. The situation can be even worse if a pressure gradient microphone is used. Then, the low-frequency turbulence noise pickup gets further enhanced by the close distance between the vortices and the microphone (recalling the proximity effect described above), especially because of the *quadrupole* nature of vortices as sound sources. Sound waves originating from a quadrupole source show an 1/distance^3 attenuation characteristic at small distances. This causes a greater pressure amplitude difference between front and back of a pressure gradient microphone membrane than would a point source (emitting spherical waves with an attenuation characteristic of 1/distance) at the same location.

Filtering can only lead to limited improvement. The wind noise picked up by the microphone will usually have quite a broad spectrum with a peak at the low frequencies and extending into the speech transmission band (e.g. [20]). Wind screen techniques used to protect bigger microphones rely on providing a partly isolated space of air around the microphone. Such space is harder to provide around a mobile phone microphone without making the phone bigger. It is good to try to find a location on the phone cover where the microphone can have some protection against turbulence. The wind noise problem will not be examined any further in this thesis.

6.2.5 Echo

The problem of echoing was briefly introduced in section 2.2.1. Echo cancellation is mainly a task for the phone software, but some acoustics guidelines can help to reduce the load on the echo cancellation algorithms. The main concern is the sound coupling from the earpiece to the microphone. This sound can couple through the air and through the mechanical structures of the phone. No experimental or theoretical study of the mechanical coupling is known to the writer. Coupling through the air is more straightforward, and this kind of coupling could even be analyzed using simulation. Such a simulation would involve the recognization of a typical leak between the phone cover and the user's ear, and some approximate representation of the air path from the earpiece to the microphone. Free field conditions could probably be used as a rough first approximation, although the presence of the head will have some effect.

6.2.6 Earpiece performance

Of the three mobile phone audio transducers, the earpiece is special in that it has to operate into a widely varying acoustic load. The microphone and the buzzer are practically always surrounded by free air, while the earpiece sees the acoustic input impedance of the user's ear as a load. Now, however, the ear impedance alone is not enough to provide a realistic picture of the situation. There will almost always be some leak present between the phone cover and the user's ear, and this leak appears as an impedance in parallel with the impedance formed by the ear. Since the ear impedance is high compared to the radiation impedance into free air, a varying leak will show as a considerable variation in the load impedance seen by the earpiece.

The leak between the phone cover and the user's ear depends on the shape of the ear as well as on the relative position of the phone. Ear shapes are individual, and the position of the phone may change a lot even during a single call. The user might press the phone firmly against the ear, hold it loosely (lightly touching the pinna), or anything between these two extremes. This can have a great effect on the subjective sound quality of the earpiece. Depending on the acoustic design, not only the perceived loudness but also the frequency balance may vary more or less depending on how the phone is held against the ear. Typically, bass frequencies are most affected by a varying leak. If the earpiece frequency response is shaped to be flat or nearly flat into a sealed ear, the bass frequencies will usually drop when a leak is present. Thus a normal user may perceive the sound as being unpleasantly thin or weak, even if measurement results into a sealed artificial ear indicate a well balanced frequency response.

To counteract the effects of the varying acoustic load, some important facts must be recognized, and according design rules established. This way an earpiece can be made more *leak tolerant*. Leak tolerance is a major topic in this thesis, and the following two chapters are dedicated to this particular area of acoustics design.

7 Design example: leak tolerant earpiece

The concept of leak tolerance will be viewed in a general manner in this chapter. It was mentioned in chapter 5 that measuring, or simulating, the frequency response of an earpiece with different artificial ear impedances as loads will provide a picture of the leak tolerance of the earpiece in question. What leak tolerance is, and why it is important, will be the first topic of this chapter. The end of the chapter is dedicated to design methods for good leak tolerance. It is then logical to go on to a practical example in the next chapter, in order to exemplify and illustrate the statements made here.

7.1 Leak tolerance — what and why

7.1.1 Performance requirements

It has been mentioned that normal use of a phone involves keeping it against the ear in a variety of positions, with a varying pressure. The earpiece itself should be able to provide a sufficient amount of sound pressure to the ear in all normal situations. Ideally, there should be no need to press the phone tightly against the ear in order to get a good speech reproduction. Having to keep pressing the phone tightly against the ear during a call creates discomfort, and relieving the pressure introduces a leak that will cause the low frequencies to drop if the earpiece in question is not leak tolerant (the reason for this will be examined later). Missing low frequencies combined with background noise can also lead to reduced or lost speech intelligibility. The possible remedy — equalizing the response to be flat enough when a leak is present — would then lead to too loud bass frequencies if the phone were pressed tightly against the ear. This is very important for product liability reasons: the phone user might be tempted to press the phone tightly against his or her ear if the environment is noisy, but then the downlink speech might also create dangerously high sound pressure levels in the ear. A phone manufacturer is not willing to take this risk.

7.1.2 Test and measurement requirements

In order to formulate an unambiguous requirement on some audio frequency response (e.g. *receiving frequency response*, shown in fig. 2.3), the full signal path has to be clearly defined. Many type approval audio tests include the phone transducers, and thus these transducers have to be connected to the test equipment in a predefined way. This is done using standard artificial mouths and ears. In addition, the mounting of these to the phone under test is accurately defined in type approval test specifications. For obvious reasons, the tests are defined so as to be accurate and simple enough to run.

In some cases test requirements and performance requirements are contradictory. A very important example is the phone earpiece performance, described in the above section. Type approval tests usually specify (for simplicity, unambiguity, and obviously to some extent even historical reasons) that the earpiece be sealed to an artificial ear during measurement. This removes the additional leak that is almost always present in a normal usage situation. In other words, the type approval measurement will be into an acoustic load that differs quite a lot from the load seen by the earpiece under most normal circumstances. Unless the earpiece is made leak tolerant, this will lead to inevitable tradeoffs in the acoustics design: sound pressure level into a sealed artificial ear must be kept below specified limits, but then the output into a normal (leaking) ear will be too low, i.e. the speech will sound too weak and thin.

7.1.3 Background theory

A clear insight into what the concept of leak tolerance is about can be gained by creating an equivalent circuit of a conventional phone earpiece held against the user's ear.

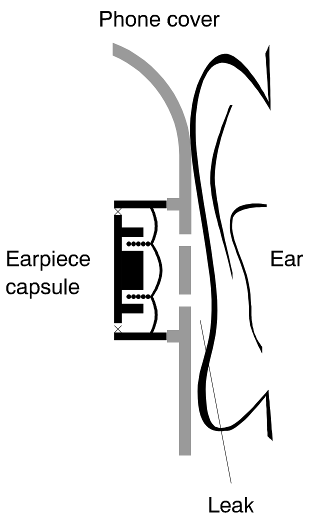


Figure 7.1: Cross-sectional view of conventional phone earpiece held against the user's ear (internal parts of the phone, other than the earpiece capsule, are not shown).

The visualization of the circuit begins by recognizing that the earpiece can be simply described as a pressure source with a given internal series impedance (by applying Thevenin's theorem in the acoustical domain, just as in electric circuit theory). This source is then connected to an acoustic impedance representing the ear. Finally, the leak must be included. Remembering that air flow is equivalent to current in the acoustical domain, it is easy to understand that the leak must be an acoustic impedance in parallel with the ear: the current (volume velocity) supplied by the earpiece is divided between the ear and the leak. The result is:

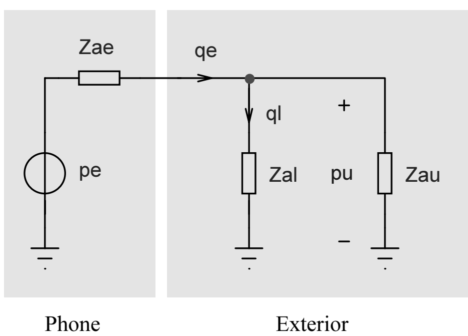


Figure 7.2: Simple equivalent circuit of earpiece held against ear.

In the above picture, and from now on, the following subscripts are used: e = earpiece, l = leak, u = user's ear. The pressure into the user's ear is denoted here by p_u. The ear input impedance Z_{au} is frequency-dependent and varies to some extent between individuals, but for simplicity it is considered invariant here. A non-leaking artificial ear such as the ITU-T P.57 type 1 ear [12] is an approximation of a typical human ear with impedance Z_{au}. The internal impedance Z_{ae} depends on the properties and mounting of the earpiece capsule inside the phone, i.e. it is constant for any given earpiece capsule and phone mechanics. To sum it up: at a given frequency, the circuit consists of two constant impedances and one that shows a wide variation (the leak impedance Z_{al}). This impedance is very high (in theory, infinite) when the phone is pressed firmly against the ear, while it is very low when there is a big leak between the ear and the phone cover.

Assuming a constant electric input signal to the earpiece capsule, the source pressure p_e will also be constant. However, the resultant volume velocity q_e will vary due to the presence of the varying leak. As a result, there will be a varying pressure drop across Z_{ae}, which will in turn lead to a varying p_u delivered to the user's ear. Expressed mathematically, the variation of p_u is

$$\Delta p_u = \Delta(q_e Z_{ae}) \tag{7.1}$$

It is seen now that to minimize this variation (which is equivalent to making the earpiece leak tolerant), one has to:

- minimize the internal impedance Z_{ae}, and/or
- stabilize the output volume velocity q_e.

According to this it is possible to state a more stringent definition of leak tolerance: *the leak tolerance of an earpiece is the relative sensitivity of the sound pressure delivered to the user's ear to changes in the total load impedance formed by the user's ear and any leaks between the user's ear and the earpiece.*

It is also important to note that, being a more or less narrow constriction, Z_{al} has the nature of a RL series impedance. Thus the leak will be more open at low frequencies, and low frequencies can be expected to drop more than high frequencies when the leak is present (as was briefly stated in sec. 7.1.1). Therefore it is usually good to focus on the low frequency behaviour when designing an earpiece for good leak tolerance.

7.1.4 Practical implementation

The internal acoustic impedance (or, equivalently, acoustic output impedance) Z_e of the earpiece can be altered by changing the properties and/or mounting of the earpiece capsule. Z_{ae} is the acoustic impedance that would be seen looking into the earpiece from the outside of the phone. Making this impedance small means making the membrane of the earpiece capsule as mobile as possible (recalling the definitions of acoustic and mechanical impedances in chapter 4). Another possible way is to add air flow routes (leaks) in parallel with the transducer. These solutions will all be viewed in more detail below.

Stabilizing the volume velocity q_e could mean either making Z_{ae} very large (which would contradict the above and waste most of the earpiece output pressure and is therefore worthless as a design method), or stabilizing the leak impedance Z_{al}. The simplest (and, usually, most elegant) way to accomplish this is to add a deliberate parallel leak that cannot be blocked by the user's ear. In other words, this added parallel leak is to be implemented as topologically equivalent to Z_{al} but a part of the earpiece itself — not belonging to the exterior of the phone as Z_{al} does.

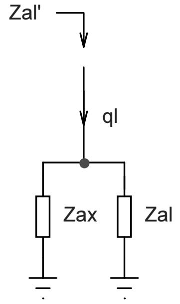


Figure 7.3: Adding a deliberate leak.

The added leak Z_{ax} must be chosen small enough in comparison with Z_{au} for good stabilization. Instead of varying between very small and very large (theoretically: infinite) values, the total leak impedance Z_{al}' will now have an upper bound equal to Z_{ax}. Z_{ax} must not be chosen too small, however, because that could make the leak draw too much volume velocity, and as a result the pressure drop across Z_{ae} would be excessive.

In theory (as can be seen from eq. (7.1)), good performance could be achieved also by making q_e as small as possible (which would mean eliminating all leaks). This solution is not relevant here, of course, because the very idea of leak tolerance is about allowing a varying amount of volume velocity to escape through the leak between the phone cover and the user's ear. Even if this leak were eliminated, the ear itself would still draw some volume velocity since the acoustic input impedance of a human ear is finite (see fig. 8.2 for an approximation by the ITU-T P.57 type 1 ear simulation model). Thus one has to be satisfied with merely stabilizing, not eliminating, q_e.

7.2 The transducer component

7.2.1 Dynamic earpiece capsule in three domains

The typical earpiece capsule is a small dynamic transducer. Even in e.g. an electromagnetic transducer the mechanical construction remains the same in principle: there is a movable membrane acting as a sound source. This membrane always has a given mechanical impedance (or acoustic impedance, if rather viewed in the acoustical domain). In its simplest form, the mechanical impedance of the membrane can be represented by a parallel RLC circuit. The inductance and capacitance are the compliance C_{mm} and moving mass L_{mm} of the membrane, respectively, and the resistance is the reciprocal value of the mechanical resistance R_{mm}. This representation of the membrane is understood by recalling that the inverse (admittance) analogy is used in the mechanical domain. Other necessary components in the equivalent circuit are the voice coil resistance and the inter-domain transformers and gyrators. The (effective) area of the membrane A_m appears as the transconductance of the gyrator, and the reciprocal of the force factor *Bl* appears as the transformation ratio of the transformer. It is assumed from now on that the voice coil inductance is small enough to be safely disregarded.

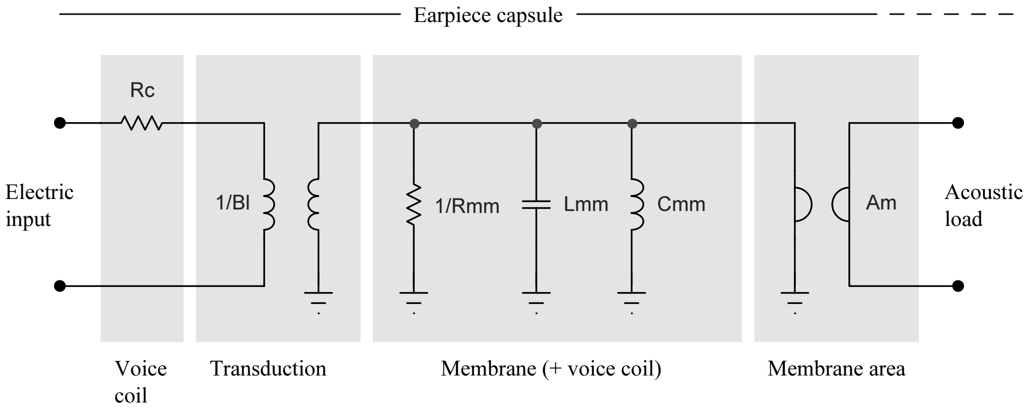


Figure 7.4: Equivalent circuit of the transduction mechanism and membrane of a typical dynamic earpiece capsule. The transformer and the gyrator are ideal.

The extent of the earpiece capsule is marked in the picture. It should be noted that there is always some kind of acoustic load that loads the moving membrane. The most important of these is the housing of the earpiece capsule itself. In other words, a complete model of an earpiece capsule will need to include a few components representing the housing in addition to the electrical and mechanical domain components shown above. Subscripts used, in addition to the ones already introduced, are: c = voice coil, m = moving parts (membrane and voice coil together).

The housing, supporting the membrane and containing the magnet and the electric connectors, may vary in construction. However, a general model of the housing can be visualized by noting that there is always a small space of air behind the diaphragm. This space may be either fully enclosed by the housing (closed, air-tight back), or completely free (open back), or anything in between. It is common to have some kind of small perforation or porous membrane on the back of the housing, which functions as a finite acoustic resistance (this is denoted by small crosses in fig. 7.1). This resistance can be small or large, and the extreme limits correspond to a fully closed or fully open back. Any openings in the back show up as a series RL impedance, and the space behind the diaphragm appears as a grounded capacitor. The housing is denoted by the subscript h.

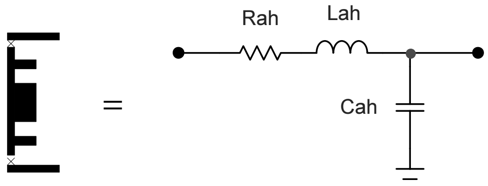


Figure 7.5: Acoustic impedance of a typical earpiece capsule housing (the membrane is left out in the picture).

Now the full equivalent circuit can be created:

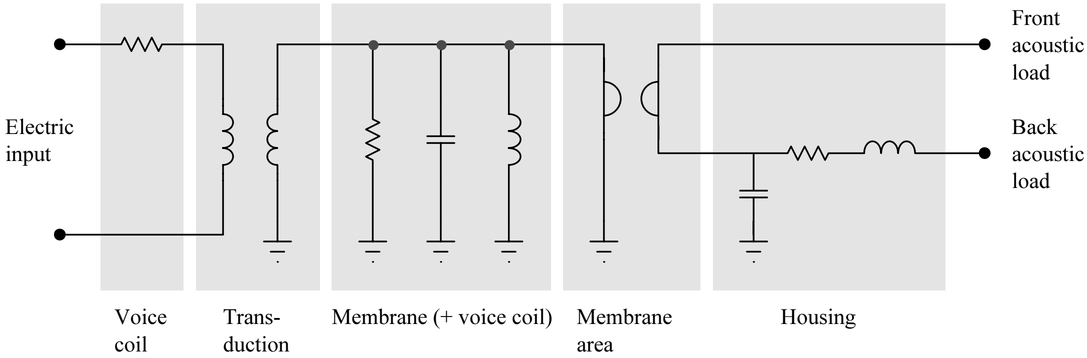


Figure 7.6: Full three-domain equivalent circuit of a typical earpiece capsule.

A general dynamic earpiece capsule model has four terminals (two electric input terminals, as well as front and back acoustic terminals). Of course, if the back of the membrane is completely closed (air-tight), then in effect the acoustic resistance in the ”housing” part of the circuit is infinite and there is only a front acoustic terminal.

7.2.2 Dynamic earpiece capsule in the acoustical domain

To illustrate the concept of leak tolerance, it is useful to restrict the attention to the acoustical domain only. First is is assumed that the above earpiece capsule is connected to an amplifier (with zero or negligible output impedance) by its electric input terminals, and the acoustic terminals are left open-circuited. (If the amplifier does not have a negligible output impedance, then this impedance can be included in the voice coil impedance.) As a result, a given pressure will appear between the front and back acoustic terminals. Also, there is a certain impedance associated with the acoustic output terminals. This impedance can be modelled by first reducing the electric circuit to the mechanical domain. The result of this operation is a resistance in parallel with the membrane mechanical resistance. This means also that the mechanical resistance (damping) of the membrane is dependent on the resistance in the electric circuit, a phenomenon known as *electromagnetic damping*. The smaller the electric circuit resistance, the higher the damping.

The same procedure can then be applied to reduce the total mechanical impedance (including electromagnetic damping) to the acoustical domain. This will result in a series RLC circuit. To sum it up, the whole equivalent circuit to the left of the housing in fig. 7.6, including the amplifier not shown in the picture, can be represented simply by a pressure source and an acoustic impedance (series RLC circuit) according to Thevenin’s theorem.

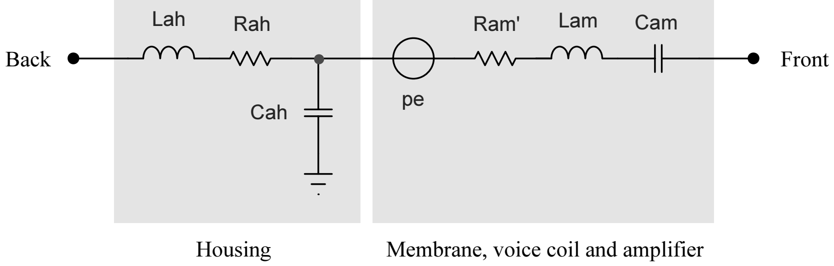


Figure 7.7: Acoustical domain equivalent circuit of dynamic earpiece capsule fed by an amplifier.

Using figs. 7.4-7.6 and basic electric circuit theorems, it can be shown that:

$$p_e = \frac{Bl}{R_c A_m} u_{in} \tag{7.2}$$

$$R_{am}' = \frac{R_{mm}}{A_m^2} + \frac{(Bl)^2}{R_c A_m^2} \tag{7.3}$$

$$L_{am} = \frac{L_{mm}}{A_m^2} \tag{7.4}$$

$$C_{am} = C_{mm} A_m^2 \tag{7.5}$$

where u_{in} is the input voltage brought to the electric connectors by the amplifier. The second term in eq. (7.3) represents the electromagnetic damping, and this term goes to zero if the amplifier is disconnected (because the total resistance in the amplifier - voice coil loop, normally equal to R_c, will then be infinite).

It is important to note that the circuits above assume that the membrane is air-tight and that no air can flow past the membrane inside the earpiece capsule. If this is not the case, then an extra RL series circuit (representing the flow path) will appear in parallel with the membrane RLC impedance branch. There exist such earpiece capsules having porous membranes or deliberate leaks in parallel with the membrane. The extra RL branch has frequency dependent component values in this simplified circuit if the membrane is porous [21].

7.2.3 Multiple-mass equivalent circuit

The circuits created above are usually good estimates at low and midrange frequencies, because at those frequencies it is justified to treat the entire moving membrane as one single entity. However, typical small dynamic loudspeakers have a membrane that consists of two separate parts: a circular center part inside the perimeter of the voice coil, and an annular suspension outside the perimeter of the voice coil.

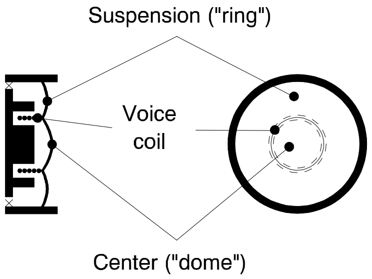


Figure 7.8: A typical earpiece capsule membrane divided by the voice coil into a circular, dome-shaped center part and an annular suspension.

The dome and the voice coil can be safely regarded as one single rigid entity at lower frequencies, while the center of the dome may no longer move in phase with the voice coil at the highest frequencies. This *decoupling* of the dome is due to its nonzero compliance combined with the small mass of the dome material (including attached air), forming a separate mass-spring system coupled to the voice coil mass. Of course, decoupling would not happen if the dome were made rigid. If necessary, even the ring and the voice coil can be separated and modelled as masses coupled by (lossy) springs, leading to a total of three coupled masses in the model of the voice coil and membrane assembly. Also, as can be seen in the above picture, there are two separate cavities under the dome and the ring, and these cavities are coupled to each other and to the back of the earpiece capsule by narrow air flow paths. An accurate model for a wide frequency range may need to include these separate acoustic impedances, instead of one single RLC circuit modelling the housing [21].

The multiple-mass approach has been used in the model of the Philips Speaker Systems 13 mm earpiece capsule (see appendix V) that will be the subject of the case study in the next chapter. The model has been recreated (from an original supplied by Philips Speaker Systems) using the developed Micro-Cap macro collection, so most components do not look like normal resistors, inductors, capacitors, transformers and gyrators in the schematic although they are still there hidden beneath newly created symbols. There are three masses modelling the membrane and voice coil, with one lossy spring (compliance) modelling the glue between the voice coil and the membrane, and another lossy spring modelling the dome compliance. The ring (suspension) is modelled as a loss-free mass-spring system. Of course, the ring has losses too, but they were obviously considered irrelevant and were therefore left out. It should be noted that there are two separate mechanical-acoustical domain interfaces (i.e. gyrators); one for the ring and one for the dome. The separate acoustic impedance components model the housing.

7.3 Mechanical configurations

Basic methods of achieving leak tolerance are overviewed next. There is a great number of possible mechanical solutions, especially since there are significant differences between various brands and types of earpiece capsules. However, all the solutions work in ways that reduce the values of certain impedances in the equivalent circuit of figs. 7.2 and 7.7. The solutions can be reduced to a few basic cases.

7.3.1 Lowering the output impedance

First the attention is focused on the acoustic output impedance Z_{ae} of the earpiece. Until now attention has been paid only to the impedances of the earpiece capsule itself. As defined earlier in this thesis, the word ”earpiece” means not only the earpiece capsule component but the whole part of the phone feeding speech into the user’s ear (see fig. 7.1). In other words, the earpiece includes the whole acoustic circuit formed by the earpiece capsule and the mechanics surrounding it inside the phone. From this definition it is clear that there exist additional acoustic impedances connected to the ”front” and ”back” terminals of the equivalent circuit shown in fig. 7.7. The front impedance is due to the sound outlets and the small air cavity between the phone cover and the sound outlets (fig. 7.1). Of course, the mechanical structure at the front of the earpiece capsule may vary, but the series connection of a cavity and a few holes in the cover is by far the most common construction.

The back of the earpiece capsule may be closed, and if this is the case it does not matter what is behind the capsule inside the phone. Of course, if the back of the capsule is instead open or only partly closed, anything behind the capsule will be seen as an additional acoustic impedance in an equivalent circuit of the earpiece. The mechanics behind the earpiece capsule may vary quite a lot between phone brands and types. Usually the back pressure just propagates along some route into the irregular space between the phone covers and the printed circuit board and electronic components. This space is in practice never air-tight, so a fraction of the volume velocity sourced by the back of the earpiece capsule will escape into free air through narrow leaks that are always present in a typical mobile phone. This would suggest a model consisting of a (grounded) capacitance for the cavity, combined with RL series impedances for the air flow paths from the capsule to the cavity, and from the cavity to the outside air (as well as the radiation impedance of the leak or leaks in the phone cover). The reality may be more complicated, with standing waves inside the phone at higher frequencies, and multiple resonances due to the various air spaces and air flow routes. For now, however, it is enough to model the effect of any mechanical structures behind the back of the earpiece capsule as just one single impedance.

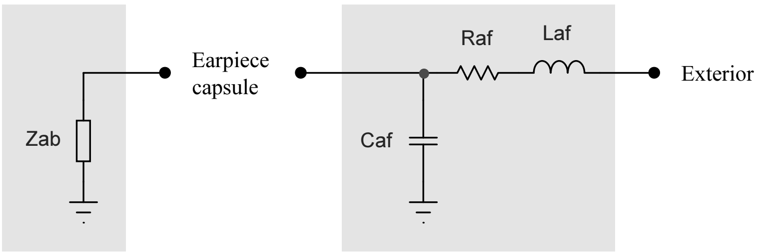


Figure 7.9: Equivalent circuit of typical mechanical structures in front of and behind the earpiece capsule.

The back impedance Z_{ab}, the front impedance formed by R_{af}, L_{af} and C_{af}, together with the equivalent circuit of the earpiece capsule in fig. 7.7, form the equivalent circuit of the whole earpiece. This is also the circuit shown at the left in fig. 7.2 as a simple Thevenin source.

If the earpiece is viewed as such a pressure source with a given internal series impedance Z_{ae}, then it is clear that the ideal solution would be to eliminate the internal impedance completely or almost completely. This way the source (earpiece) would be capable of delivering its pressure to any load without appreciable losses. Such a solution is impossible for a number of reasons:

- the membrane will always have a nonzero mechanical (and acoustic) impedance,
- the voice coil (usually a lot heavier than the membrane) cannot be eliminated,
- the openings in the phone cover protecting the earpiece capsule (fig. 7.1) will always have a nonzero acoustic impedance,
- the air gap between the magnet and the voice coil must be kept narrow to maintain efficiency (which raises the acoustic impedance of the air flow path through the air gap unless a separate flow route is provided),
- the structures behind the membrane (e.g. the magnet) cannot be eliminated,
- the area of the membrane must be kept small enough to keep earpiece capsule dimensions within acceptable limits for use in small mobile phones.

The acoustic output impedance of the earpiece cannot be eliminated, but ways can be found to minimize it.

7.3.1.1 Internal structure of the earpiece capsule

It is easy to start by considering the housing behind the membrane. In the extreme case of a fully sealed (air-tight) back, only C_{ah} exists and the R_{ah} - L_{ah} branch (see fig. 7.7) is removed. C_{ah} is seen in series with the membrane impedance. If C_{ah} is small (i.e. the cavity under the membrane is small), then the impedance due to the housing will be high, especially at low frequencies. This is not desirable. So it is clear that the air cavity must be either big and/or opened up to the surrounding air (by including the R_{ah} - L_{ah} flow path) in order to make the desired acoustic output impedance reduction possible. Since size is a primary concern in mobile phones, the latter way is better. A simplified equivalent circuit of such a housing (valid at all but the highest frequencies) can be formed by removing the C_{ah} branch.

A typical earpiece capsule may have a resonance frequency in free air (hereafter denoted by f_r) of roughly 500 Hz. This is a direct consequence of the resonance of the series connection of the five acoustic components R_{ah}, L_{ah}, R_{am}', L_{am} and C_{am} (C_{ah} can be seen as an open circuit at low frequencies). After minimizing the impedance of the first two components (housing), further improvement is possible by lowering the membrane impedance. The total earpiece capsule impedance (denoted here by the subscript *s*) will be

$$\begin{aligned} Z_{as}\big|_{\omega<<\omega_r} &= \frac{1}{j\omega C_{as}} \\ Z_{as}\big|_{\omega=\omega_r} &= R_{as} \\ Z_{as}\big|_{\omega>>\omega_r} &= j\omega L_{as} \end{aligned} \tag{7.6}$$

where

$$\begin{aligned} R_{as} &= R_{am}' + R_{ah} \\ L_{as} &= L_{am} + L_{ah} \\ C_{as} &= C_{am} \end{aligned} \tag{7.7}$$

i.e. only one of three terms predominates at the resonance frequency and at frequencies far above or below. If all three terms are minimized, then Z_{as} will also be minimized over a wide frequency range. Minimizing all three terms means implementing the earpiece capsule with minimized acoustic resistance, minimized acoustic inductance, and maximized acoustic capacitance.

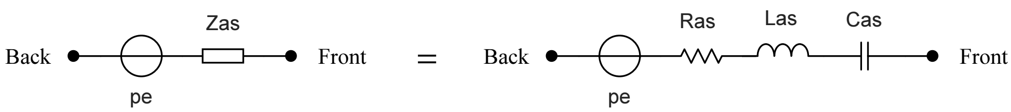


Figure 7.10: Simplified equivalent circuit of earpiece capsule fed by an amplifier.

It can be seen from eq. (4.1) that increasing membrane area will reduce the acoustic impedance proportionally to the square of the area increase (if the mechanical impedance is kept constant). Increasing the area is usually not feasible in mobile phones, however, because component size tends to be the most important criterion of all as far as mounting inside the phone is concerned.

The voice coil is responsible for most of the moving mass, but making the voice coil lighter means reducing either the length or the diameter of the coil wire. The former method leads to lower efficiency (the force factor *Bl* is reduced, unless the magnetic field is strengthened). The latter method increases the voice coil resistance, which leads to lower damping and efficiency (more power is lost in the voice coil). Increasing the compliance of the membrane, e.g. by making it thinner or using a more compliant material, does not harm the performance of the capsule itself, but the membrane may lose mechanical strength and be more sensitive to various environmental conditions that mobile phone components have to withstand. To summarize the effects on efficiency when various parameters are changed,

$$\eta\big|_{f>>f_r} \propto \frac{(Bl)^2 A_m{}^2}{R_c L_{mm}{}^2} \tag{7.8}$$

as can be readily seen from eq. (11) of [19], substituting eq. (4.1). The housing is not included in eq. (7.8), but it is easily accommodated by using instead of L_{mm} the combined (mechanical) inductance $L_{ms} = L_{mm} + L_{mh}$ (membrane and housing together). The fact that cost and efficiency (in addition to size) are primary concerns for the phone and earpiece manufacturer contradicts many of the design requirements to achieve a low acoustic output impedance.

The earpiece capsule structure depicted in fig. 7.8 is very simple. Other slightly more complex structures are common. For example, there may be additional air flow paths through the magnet (or through the pole piece inside the voice coil), and there may be additional leaks at the edges of (or even through) the membrane. Last but not least, it is important to remember that the frequency response of the earpiece capsule is no less important than good leak tolerance. The changes for better leak tolerance suggested here will all alter the frequency response in some way, and it is the purpose of good design by the manufacturer to maintain a proper frequency response.

7.3.1.2 Mechanical structures behind the earpiece capsule

It is perhaps not worthwhile to try to build a universal model of the mechanical structures usually encountered behind the earpiece capsule. In conventional cases these structures are not even specified, but the capsule is rather just fitted inside the phone and possible irregularities in the frequency response are smoothed out by equalization. Typical capsules not designed for leak tolerance may have a high R_{ah} and L_{ah}, which has the benefit of making the earpiece frequency response less sensitive to what is behind the capsule. However, if the capsule acoustic impedance Z_{as} is minimized, then any additional series impedance formed by the mechanics behind the earpiece capsule will need to be given a thought. The impedance seen by the back of the earpiece capsule will be called Z_{ab} as above in fig. 7.9.

It is easy to outline some general design rules. An air-tight cavity behind the capsule would show as an acoustic capacitance C_{ab}, proportional to the volume of the cavity. So in order to aim at a low total impedance

$$Z_{ae} = Z_{ab} + Z_{as} + Z_{af} \tag{7.9}$$

the cavity volume should be made big enough, i.e. C_{ab} should be greater than C_{as}. This is not a universal truth, however: even a small cavity may be ”cancelled” by the inductive series impedance L_{ms}, because at the corresponding resonance frequency only the resistive part of the impedance shows (as is the case for series RLC circuits in general). The drawback, of course, is that Z_{ae} can be reduced this way in a narrow frequency range only. If the back of the capsule is not enclosed in an air-tight cavity but rather left open to the irregular space of air inside the phone, then some space should be provided around the openings at the back of the capsule. If this is not done, e.g. if the capsule is mounted with its back against a printed circuit board or some big electronic component, then the flow of air will be constricted and Z_{ab} will consist of a high RL series impedance. Usually the reality lies somewhere between these two cases, and all three components (R, L and C) are found in Z_{ab}.

The obvious drawback to making the back of the earpiece capsule more open is that the rest of the phone mechanics will have a greater effect on the frequency response and the leak tolerance. Preferably some acoustically well behaved (or at least predictable) structure should be specified behind the capsule inside the phone. Unfortunately this is not always possible for e.g. space saving, cost saving or production assembly reasons. Moreover, reliable simulation of the irregular space and leaks inside a mobile phone is very difficult or impossible using the methods described in this thesis. The effects on frequency response cannot in general be simulated or predicted at an early stage, but one is rather forced to wait for good prototype parts to measure the performance in real mechanics. The equivalent circuit can then be adjusted for a better match to the measured results.

Another obvious problem is that of mechanical assembly tolerances. Acoustically critical tolerances will be visible as instabilities in the frequency response. If the space inside the phone accidentally creates one or more resonances together with the leaks in the phone covers, then mechanical tolerances might shift these resonances up or down along the frequency axis. Pronounced peaks and dips will appear if the resonances are sharp as well, and the frequency response will vary between phone units (the equalizing filters are not individually fine-tuned in each phone unit because that would be a bottleneck in the production line).

Another interesting and less obvious way of getting rid of the unpredictability of the acoustical behaviour of the structures behind the earpiece capsule is to connect the back to the free air outside the phone. Holes (or a porous dust and water shield membrane) in the back cover must still be used to protect the capsule, but at least it will be easier to specify and predict the earpiece performance, and Z_{ab} can be made low. This kind of assembly should be quite easy to implement in a thin phone, provided of course that the electrical connections to the capsule can still be made. A drawback is that it may be easier for other people to overhear what is said on the phone.

7.3.1.3 Front cavity and holes

The front holes and front cavity form together an equivalent circuit similar to that of the housing. The cavity is usually bigger than the internal cavity inside the earpiece (although it has to be made small if space is a major concern inside the phone). Moreover, a Helmholtz resonator is formed in front of the earpiece capsule by the cavity and the holes. This Helmholtz resonator will attenuate the earpiece output above its resonance frequency formed by L_{af} and C_{af}. Thus it is important to choose cavity and hole dimensions so as to tune the Helmholtz resonance above the upper frequency limit of the speech transmission frequency range. If wanted, the highest frequencies may be boosted by tuning the resonance just slightly above the upper frequency limit (some margin must be left for tolerances).

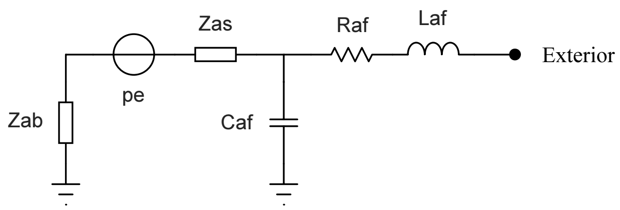


Figure 7.11: Simplified equivalent circuit of front cavity and holes together with other parts of the earpiece.

The fact that the earpiece capsule has a finite output impedance complicates the situation. The exact location of the front cavity Helmholtz resonance will not in general be that predicted by L_{af} and C_{af} alone. If, for some reason, the front cavity is made very small, then the impedance of the C_{af} branch will be considerably higher than the impedance of the Z_{ab} - Z_{as} branch even at the highest frequencies, and the Helmholtz resonance cannot be tuned low enough to boost the highest speech frequencies as described above. If the cavity is made big, then the actual Helmholtz resonance will be slightly higher than that predicted by L_{af} and C_{af} due to the parallel inductive impedance of the Z_{ab} - Z_{as} branch.

A design rule can be derived for the front cavity as follows: it can be assumed that Z_{as} and Z_{ab} form an inductive impedance (hereafter called L_{abs}) at the high frequencies now considered. This inductance appears in parallel with the acoustic capacitance of the front cavity. These two will have a parallel resonance at a certain frequency. Above this frequency the capacitive impedance of C_{af} will be lower than the inductive impedance of L_{abs}, i.e. the impedance formed by everything behind the front holes will be capacitive. Below the parallel resonance frequency, L_{abs} will have the lower impedance and the impedance formed behind the front holes will thus be inductive. Since Helmholtz resonance in the front cavity is not possible without a capacitive impedance behind the front holes, it can be concluded that in order to make high frequency boost by utilization of the Helmholtz resonance possible, the front cavity must be made big enough to put the parallel resonance frequency (of L_{abs} and C_{af}) below the upper limit of the speech transmission frequency range.

7.3.1.4 Leaks inside the phone

The above analysis has concentrated on the path from the back of the earpiece capsule to the front holes. At low frequencies, this path is a simple series connection of impedances, and the total impedance can be no lower than the highest impedance along the path.

It is however possible to open up ”parallel” air flow paths in a variety of ways by having controlled leaks at certain locations. One obvious location is the front cavity. If a leak is introduced in the front cavity, and this leak is left open e.g. to the air between the phone covers, then the equivalent circuit will have an extra branch in parallel with C_{af} and the Z_{ab} - Z_{as} branch.

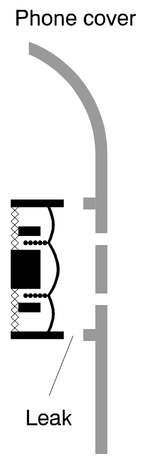


Figure 7.12: Adding an extra leak to the front cavity (other parts inside the phone, apart from the earpiece, are not shown in the picture).

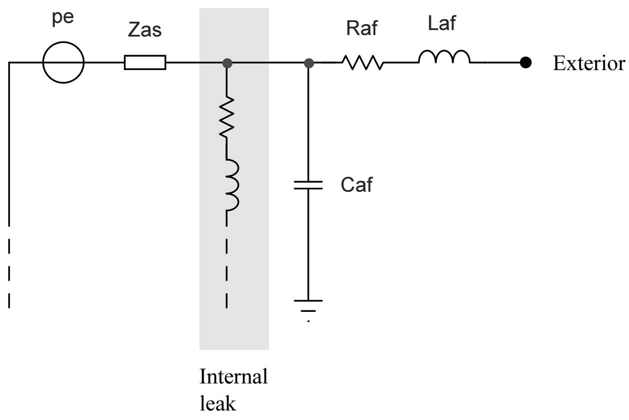


Figure 7.13: The added internal leak appears as a parallel branch in the equivalent circuit.

The leak (hereafter called *internal leak* because it is in the front cavity located inside the phone) is not connected anywhere in the above picture. This is to emphasize that there are very many possible ways to route the leak, e.g. to outside air, to a cavity behind the earpiece, or to the irregular space of air between the phone covers. In other words, a part of Z_{ab} may or may not be common to the two branches shown with dashed lines. The main point to note here is that the earpiece capsule itself is bypassed in some way by the leak, and as a result the total acoustic impedance of the earpiece is lowered. In other words: what was achieved above by altering the construction of the capsule itself can also be achieved using leaks that bypass the capsule. This can be understood by thinking about the Thevenin equivalent which has an internal impedance Z_{ae} equal to the impedance seen looking into the front holes. The internal leak will usually raise the Helmholtz resonance frequency of the front cavity (because the total inductive acoustic impedance seen by the cavity will be lower), which may be a disadvantage in some cases.

7.3.2 Stabilizing the output volume velocity

It was already stated that the leak between the phone cover and the user's ear is best stabilized by adding another constant parallel leak as in fig. 7.3. Dimensioning of this leak is a compromise between leak tolerance, maximum sound pressure into the ear, and mechanical realizability. A very big leak yields a very good leak tolerance, but the maximum available sound pressure level may be too low (remembering that the earpiece capsule and/or driving circuits will have some limit above which they will distort the speech).

A leak route going from the earpiece area to the outside air must, of course, start close enough to the sound output holes. This is also to guarantee that the earpiece will be measured correctly using some standard artificial ear (e.g. the ITU-T type 1 sealed coupler with a circular area of 25 mm diameter to connect to the earpiece that is to be measured). Also, the leak must be routed some way past the ear and along the phone cover, which may require a flow path that is quite long in the air flow direction. A long flow path means a higher resistance and inductance (i.e. a less efficient leak), which may have to be counteracted by using a large cross-sectional area instead.

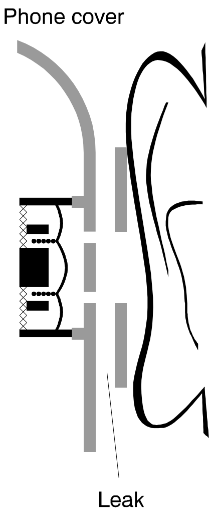


Figure 7.14: Adding an external leak to the earpiece.

An obvious disadvantage of this solution is that it may require quite a lot of extra space (unless some suitable mechanical structure is already available in the phone cover). The equivalent circuit is obvious from fig. 7.14:

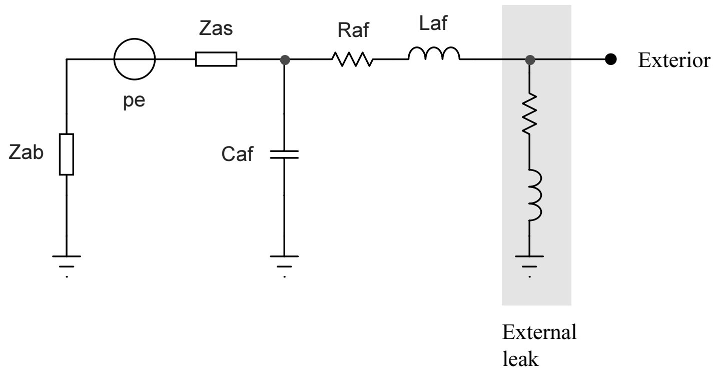


Figure 7.15: An external leak to the outside air (the radiation impedance of the leak is not explicitly shown, but it is relatively small and it has the same RL nature).

It has been pointed out already that there is a leaking air space between the phone covers and the electronics parts. This provides an interesting solution in which the external leak is routed through the phone cover instead of between the cover and the user's ear (both this kind of leak and the one depicted in fig. 7.14 will be called *external leaks* because they are routed from outside the front cavity).

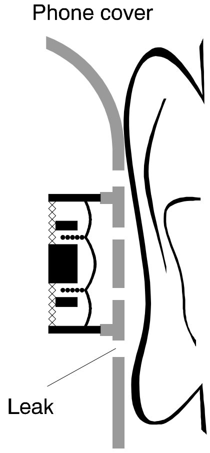


Figure 7.16: A leak through the phone cover (from the outside air to the inside of the phone).

The equivalent circuit will have an RL series impedance branch in parallel with the front holes and the earpiece capsule:

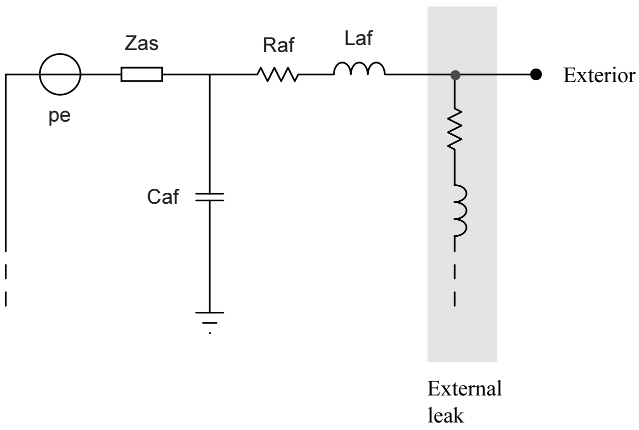


Figure 7.17: The equivalent circuit of fig. 7.16.

Dashed lines are used again in the above circuit tho show that the external leak route may end in various ways. The most typical case, of course, would be simply a grounded capacitor in parallel with an RL series impedance, representing the space and leaks between the phone covers and electronics. It is important to note the similarity of the equivalent circuits in figs. 7.15 and 7.17. This means that a leak through the phone cover can work much the same way as a leak to the outside air, provided that the impedance from the leak holes to ground is not too big. An excellent example of this will be viewed in the next chapter.

The great advantage of a leak through the cover compared to a leak to the outside air is that the leak route itself can be made a lot shorter in the air flow direction, which makes the leak more efficient (the impedance is lowered). Secondly, at best no extra space is required in the phone mechanics (except for some space beneath the front cover for the air to leak into, but such space is usually available anyway unless parts are very tightly fitted inside the covers). Thirdly, the front cavity and its holes can be dimensioned independently to tune the Helmholtz resonance if needed. The important disadvantage is that the leak is not necessarily a simple RL series impedance because it ends in some cavity inside the phone. This cavity and its leaks may even form resonances inside the speech transmission frequency range, and these resonances may be hard to predict (simulate) unless all leaks are well known.

7.3.3 Combination methods and modifications

The basic configurations viewed above are in no way the only working ones, but practically any kind of mechanical construction can be reduced to a combination of one option from each of the following three groups:

1. Back of earpiece capsule
- closed (a),
- enclosed in a cavity (b),
- open to air inside phone (c), or
- open to outside air (d),

2. Front of earpiece capsule
- sealed to phone cover (a),
- leaking to air inside phone (b), or
- leaking to outside air (c),

3. Region around front holes
- sealed (a),
- leaking to outside air (b), or
- leaking to air inside phone (c).

Thus a conventional earpiece would correspond to (1a) or (1c) together with (2a) and (3a). An earpiece with a good leak tolerance could have e.g. (1d), (2a) and (3c), or (1c), (2b) and (3a). Of the above, (1c), (1d), (2b) or (2c) all implement the first option described in sec. 7.1.4 (reducing Z_{ae}), while (3b) or (3c) implement the second option (stabilizing q_e). (In fact (3b) and (3c) could also be seen as implementations of the first option; it is simply a question of whether the leak routes are viewed as parts of Z_{ae} or not.) As stated in sec. 7.1.4, both options do not necessarily have to be implemented but one of them may be enough. For example, a capsule with a high internal acoustic impedance can still perform very well if (3b) or (3c) is used. Also it should be noted that more complicated structures are possible inside the phone: for example, an internal leak from the front cavity could be connected to another cavity, which is further connected to a cavity behind the capsule. One such construction will be exemplified in the following chapter.

The presence of leaks (either wanted or unwanted) in cavities or between the phone covers can be seen as additional important characteristics of the above options. For example, at least in a limited frequency range, (1b) will approach (1c) if there are big leaks in the cavity, and (1c) will approach (1d) if there are big leaks in or between the phone covers.

8 Case study: leak tolerant earpiece

This chapter is a study of the design and implementation of a leak tolerant earpiece in a Nokia GSM mobile phone known officially as the Nokia 6110. The history behind the choice of the final acoustical solution for the earpiece is important and it contains several good examples of problems to be solved. Mechanical design, industrial design, measurement and production requirements are described to the extent that the compromises made can be understood.

Both simulation and measurement results are used in this chapter. Also, the resistor, inductor, capacitor, voltage source, current source, transformer and gyrator representation used in the previous chapter is substituted by actual Micro-Cap circuits (containing macros defined in appendix IV). This is done to display the properties and power of the developed simulation system. Of course, the construction of the circuits in terms of basic components (resistors, inductors, capacitors etc.) can always be recalled by referring to the macro definitions in appendix IV.

8.1 The starting point

8.1.1 Leak tolerance and previous earpieces

No concentrated efforts were made in Nokia Mobile Phones before the 6110 to assure that the earpieces used in commercial products were leak tolerant. Instead, mechanical solutions were developed together with the earpiece capsule manufacturer who usually had a simple set of design rules for the front cavity and front holes. Neither the earpiece capsules nor the mechanics around them were designed especially for leak tolerance, but of course some amount of leak tolerance was still achievable depending on the properties of the earpiece capsules themselves.

The earpiece used in the Nokia 1610, another very successful Nokia phone, can be used as an example of an earpiece not designed for leak tolerance. In order to see things in their right proportions, measurement results of the 1610 should be compared to 6110 results presented later.

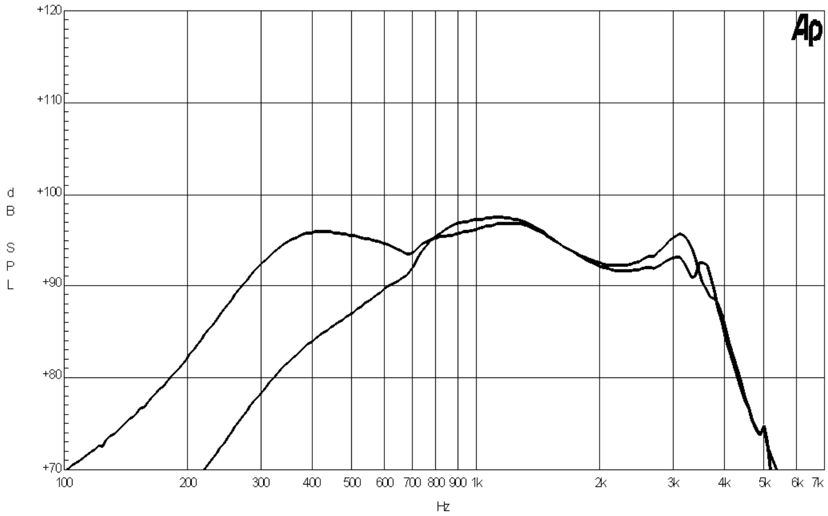


Figure 8.1: Measured frequency response of Philips Speaker Systems WD00909 earpiece capsule in Nokia 1610 mechanics, using ITU-T P.57 type 1 coupler both sealed and with leak ring ("German leak", see sec. 5.4.11).

As described in section 5.4.11, the amount of leak tolerance of a given earpiece is indicated by the relative difference between the two measured (or simulated) frequency responses using the type 1 coupler with and without the German leak ring. Of course, any kind of realistic varying acoustic load could be used, but the treatment in this chapter will concentrate on the type 1 coupler and leak ring because they were the ones used during the development of the 6110 earpiece solution. See appendix VI. From now on, frequency responses with and without the German leak ring (hereafter also called simply *leak ring* or *German leak*) will be used to display the leak tolerance of each particular solution.

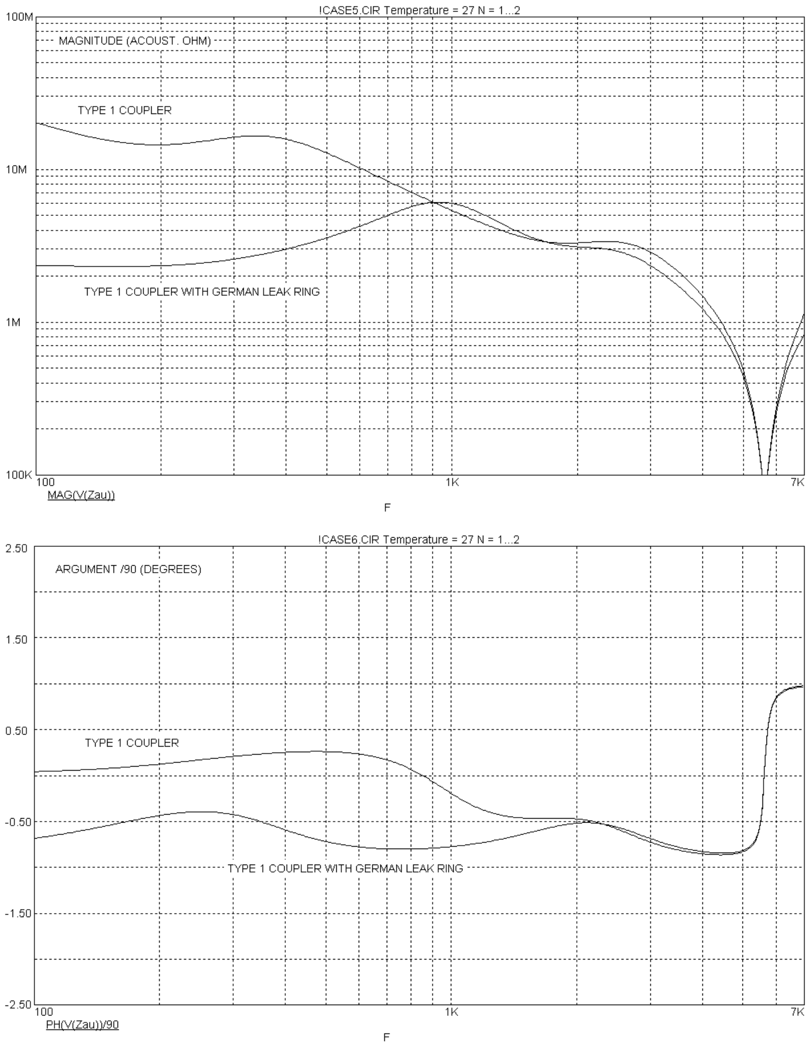


Figure 8.2: Simulated acoustic impedance (top: magnitude, bottom: argument) of type 1 coupler with and without German leak ring.

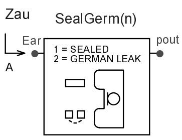


Figure 8.3: An acoustic impedance probe is used to obtain the results of fig. 8.2. The SealGerm *macro to the right was created to simplify leak tolerance analysis (see the end of appendix IV).*

It is very informative to note that the type 1 coupler (which should resemble a typical human ear with good enough accuracy) has a high capacitive impedance below 850 Hz when sealed, but this impedance drops dramatically and becomes inductive when the German leak ring is added. The difference above 850 Hz is not very significant. Since the German leak ring is intended to correspond to a typical smallish leak between the phone cover and the user's ear, it is easy to understand the statement made earlier that the bass frequencies are the ones suffering most when the earpiece leaks. There are other coupler types (e.g. the type 3.2 high-leak coupler) modelling typical big leaks, but these will not be used in this thesis. The type 1 coupler simulation model has an impedance that deviates 1 dB (at most) from the measured result in the frequency range used for speech transmission (according to graphs in [12]). The accuracy of the German leak model is not known to the writer.

8.1.2 The new transducer

The earpiece capsule tends to be one of the biggest components in a mobile phone, so quite a lot of space could be saved by trying to miniaturize it. This was done in the 6110 phone, which was to use a new 13 mm diameter earpiece capsule from Philips Speaker Systems (PSS), shown below (a photograph can be found in chapter 3, fig. 3.4). The capsule was co-developed by PSS and Nokia Mobile Phones (NMP) for an intended new acoustical solution that is viewed in more detail below.

The 13 mm capsule, in addition to being small, was also to have a lower acoustic impedance than typical competitors (i.e. a low Z_{as}). This was achieved by making the mechanical construction of the capsule very open to the air flow.

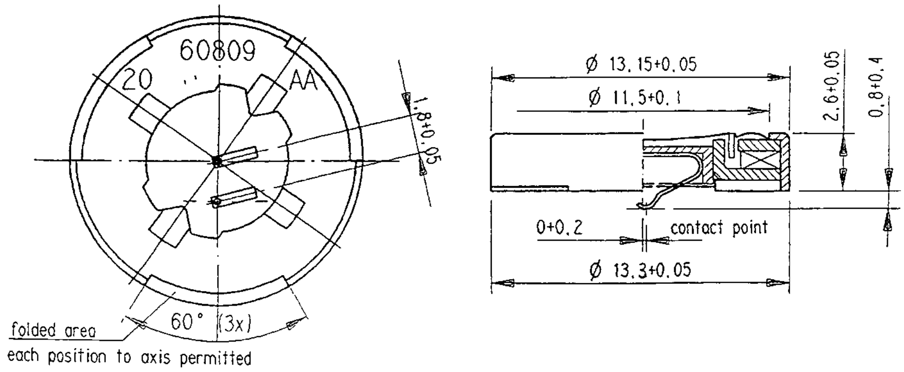


Figure 8.4: Bottom view (left) and side view/ cross-sectional view (right) of new PSS 13 mm leak tolerant earpiece capsule, type WD00518 (from [22]*).*

The openness of the capsule is not seen in the cross-sectional drawing. However, there are four quite big openings (hereafter called the *core plate holes*) that can be seen in the bottom view drawing. These openings are directly connected to the small air cavity formed under the center ("dome") part of the membrane. When the capsule is placed between two parallel plane surfaces (with compressed spring contacts), two additional small cavities are formed: one under the capsule body, and one above the membrane. This is why the PSS capsule models shown in appendix V have these cavities included as well. Of course, at least the front of the capsule is normally connected directly to a bigger cavity and the cavity between the membrane and the plane is too small to be significant. However, it is very convenient to have simple reference planes for the capsule simulation model that comply with the mechanical mounting of the capsule component.

The three capsule models (of varying complexity and accuracy) shown in appendix V should be understandable by referring to fig. 8.4 and chapter 7. Hereafter only one single shape will be seen in simulation schematics, though, because the capsule model will be hidden beneath a macro (unless mentioned otherwise, the most accurate three-mass model is the one used, and the macro containing this model is called PSS13_1).

Important data describing the earpiece capsule are:

- nominal impedance: 32 Ω,
- voice coil resistance (R_c): 29 Ω ± 10 %,
- sensitivity: 103 ± 3 dB (1 mW, 1 kHz, type 1 coupler),
- maximum linear displacement ($x_{m\max}$): ± 0.1 mm
- effective area (A_m): 0.7 cm^2.

It is useful to find out the values of the components shown in fig. 7.7. A couple of these are seen directly in the capsule models. Other values are easily calculated from the models using eqs. (7.2)-(7.5) and the formula for the acoustic capacitance of a cavity (found in the definition of the Cavity macro). The results are, for the membrane:

$$\begin{aligned} R_{am}' &= 1.5\ \text{M}\Omega \\ L_{am} &= 5.1\ \text{kH} \\ C_{am} &= 29\ \text{pF} \end{aligned} \tag{8.1}$$

and for the output open-circuit pressure:

$$p_e = 230 u_{in} \tag{8.2}$$

and, finally, for the housing:

$$\begin{aligned} R_{ah} &= 500\ \text{k}\Omega \\ L_{ah} &= 300\ \text{H} \\ C_{ah} &= 490\ \text{fF} \end{aligned} \tag{8.3}$$

Using the L_{ah} and C_{ah} values now obtained, it can be calculated that the housing will become capacitive at a frequency as high as 13 kHz, far above the speech transmission frequency range. Thus, below 13 kHz, the housing can be seen as a pure RL series circuit. To put it another way: the housing remains acoustically ”open” to the outside air at all frequencies of interest and the capacitance of the housing can be left out, as already assumed in chapter 7. Now if the other components are added in series, the values for the whole capsule are obtained:

$$\begin{aligned} R_{as} &= 2\ \text{M}\Omega \\ L_{as} &= 5.4\ \text{kH} \\ C_{as} &= 29\ \text{pF} \end{aligned} \tag{8.4}$$

Some very important things can be seen: L_{am} is considerably bigger than L_{ah}, i.e. the moving mass (membrane and voice coil) forms the major part of the total acoustic inductance L_{as}, and it would not help much to open up the core plate holes more to reduce L_{ah}. The typical main resonance frequency (400 Hz) of the capsule can be readily calculated from L_{as} and C_{as}.

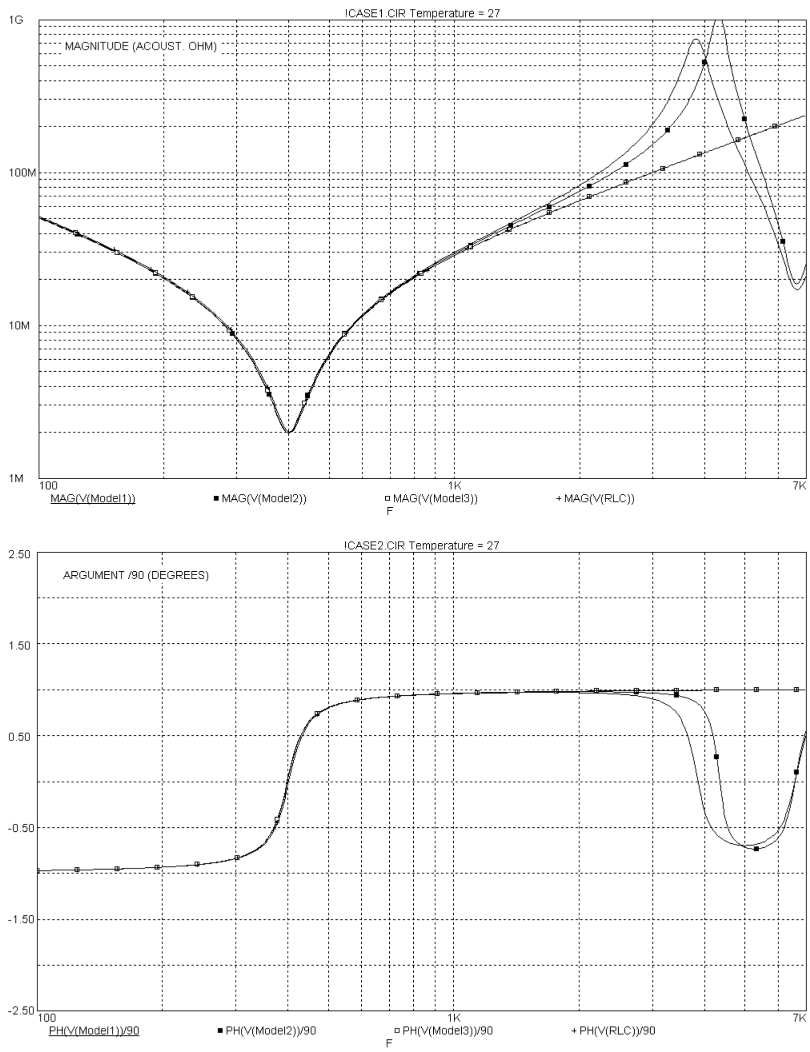


Figure 8.5: Simulated acoustic impedance Z_{as} (with the amplifier connected) of the three earpiece capsule models (top: magnitude, bottom: argument).

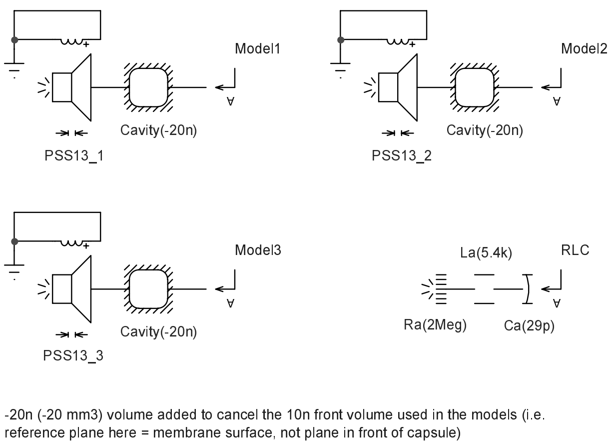


Figure 8.6: Schematic to obtain the simulation results of fig. 8.5. The voice coil is short-circuited to simulate the situation with a zero-impedance amplifier connected to the earpiece capsule. Acoustic impedance probes are used to probe the impedances.

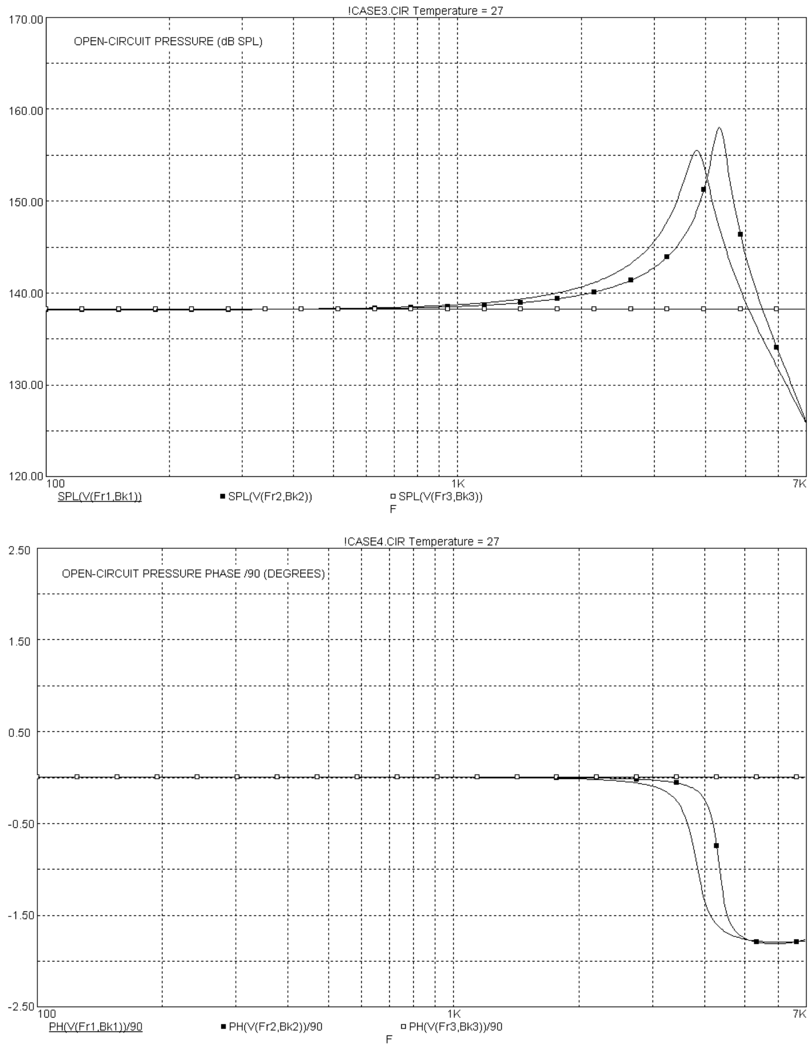


Figure 8.7: Simulated open-circuit pressure level (top) and phase (bottom) of the three earpiece capsule models.

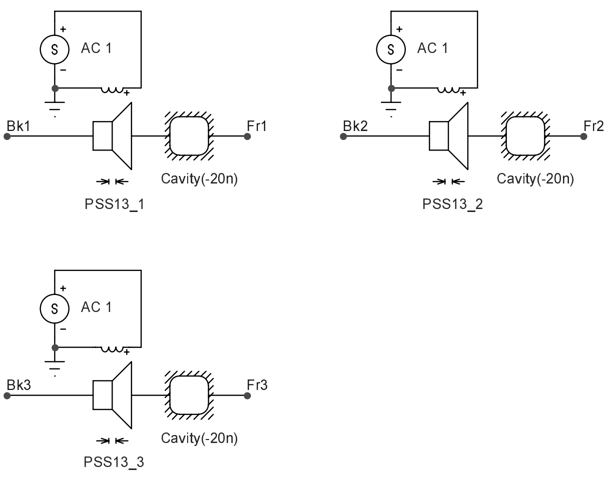


Figure 8.8: Schematic to obtain the simulation results of fig. 8.7.

The main resonance is also seen as zero acoustic reactance in fig. 8.5 (right). Models 1 and 2 have irregularities at high frequencies because they simulate the effects of dome decoupling.

Perhaps surprisingly, the models in appendix V have no resistance at all in their mechanical domains. This is not to say that the mechanical impedance of the membrane (with no amplifier connected) is zero, but the resistance of the whole capsule (membrane and housing) is included as a resistance in the acoustical domains instead. This was an approach chosen, for simplicity, by the engineers at PSS who developed the model. The resistance might just as well be split up into a mechanical membrane resistance and an acoustic housing resistance if wanted. The mechanical resistance would then be very small.

8.1.3 The new phone

All the intended features of the new phone will not be described here, but it should be self-evident by now that good audio performance of the earpiece was one of the important targets. The reasons for aiming at a good leak tolerance have already been mentioned.

It was specified at an early stage that the phone was to have two printed circuit boards (PCBs) separated by a plastic frame. The upper PCB (i.e. the one closer to the phone front cover, hereafter called the *A-cover*) was to have user interface (UI) related functions and connections on it, while the lower PCB was to be loaded mostly by baseband and RF components. Hence the names that are to be used from now on: *UI PCB* and *Engine PCB*. The earpiece capsule, naturally, was to reside on the UI PCB. The back of the capsule was to be open (otherwise Z_{ab}, and hence Z_{ae}, would be too large at low frequencies), so big holes through the UI PCB were specified behind the capsule. The size (diameter 1.3 mm) and number (6 pieces) of these holes was chosen to keep their acoustic impedance low enough compared to other impedances appearing in series with them.

Moreover, the two PCBs separated by a frame provided a convenient way of arranging a cavity behind the capsule (more specifically, behind the holes in the UI PCB, but the impedance of these holes is not very significant). The exact size of this cavity, which is hereafter to be called the *back cavity*, was not known in advance. Also, it would inevitably differ between phone models (other models belonging to the same family of products as the 6110 but with slightly different components and mechanics, e.g. the 5110). A volume of 4 cm^3 was estimated, and the earpiece capsule was to function correctly in a back cavity volume of this size. What this meant was that C_{am} (the acoustic capacitance of the membrane) was to be lower than C_{ab} (the series acoustic capacitance of the back cavity, see sec. 7.3.1.2). This is easily verified using the C_{am} value calculated above and the formula for the acoustic capacitance of a cavity. It was also known from the beginning that the back cavity would not be completely sealed; the PCBs (of multi-layer construction, approx. 0.8 mm thick) were to have a great number of small through holes in them. This is not a bad thing — on the contrary the holes would have a beneficial damping effect on the main resonance as will be seen later. It was estimated that the through hole diameter would be 0.2 mm and the total number of holes would be around 100. For simplicity, it is assumed in the following simulations that all the through holes are connected to free air (although this was not going to be the case in the real phone mechanics finalized later).

In terms of the classes of topologies defined at the end of the previous chapter, this solution would correspond to (1b), (2a), (3b) (with a leak in the back cavity).

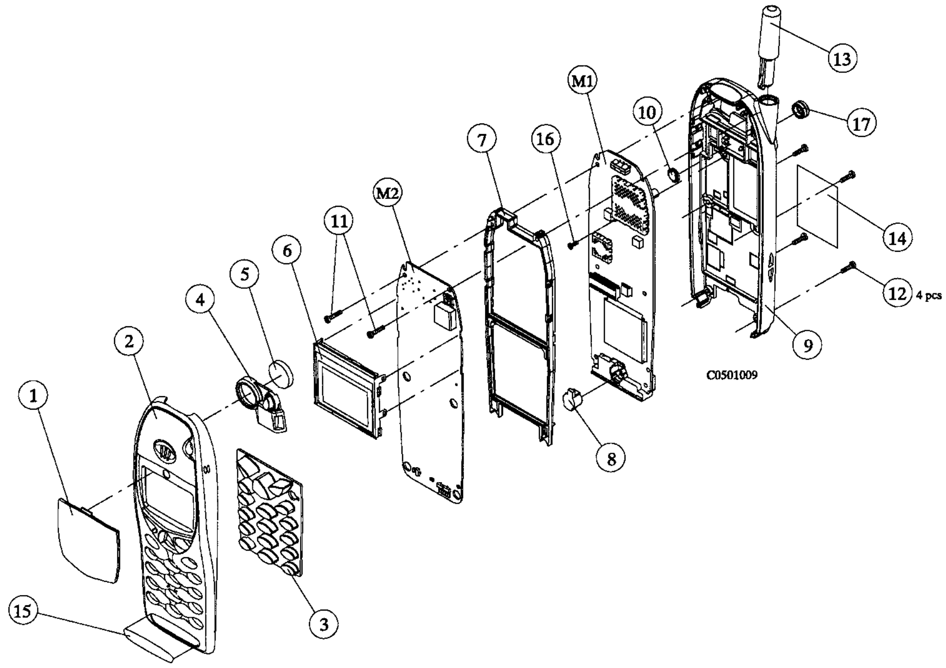


Figure 8.9: Exploded view of the Nokia 6110 phone. (2) A-cover (front cover), (4) sealing gasket for earpiece capsule, (5) earpiece capsule, (M2) UI module, (7) frame, (M1) RF & baseband (Engine) module, (9) B-cover (back cover). Other parts: (1) display lens, (3) keypad, (6) display, (8) rubber boot containing microphone, (13) antenna.

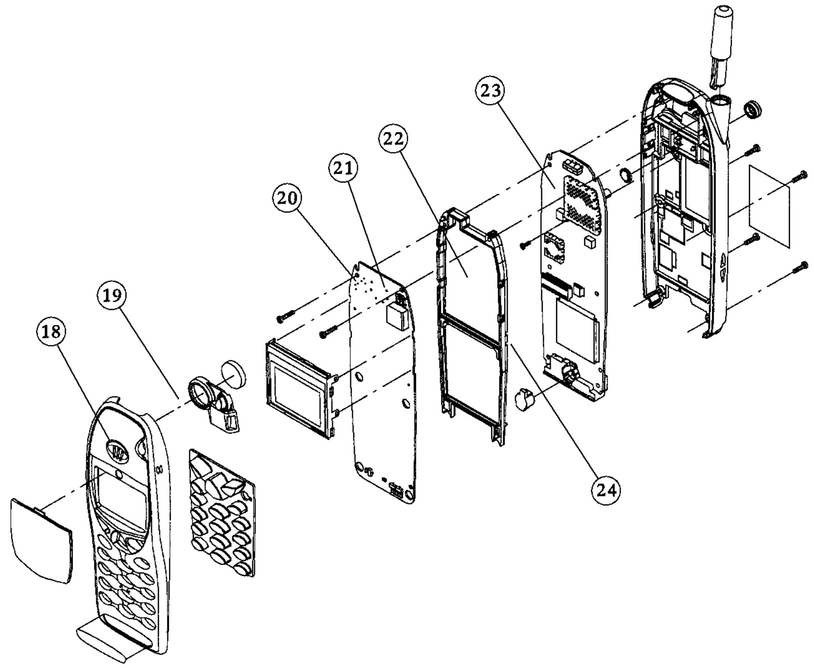


Figure 8.10: Some important acoustical parts. (18) front holes, (19) front cavity bounded by A-cover, gasket and capsule, (21) holes in UI PCB behind capsule, (22) back cavity bounded by UI module, frame and RF & baseband module, (23) RF & baseband (Engine) PCB with through holes.

There are a few more important parts not mentioned in figs. 8.9-8.10, but these will be treated later together with the acoustical solutions making use of them.

8.1.4 Design and production requirements

A fact that will become obvious in this chapter is that mechanical and acoustical design can be surprisingly closely related. This is true especially when many mechanical structures around the earpiece capsule are a part of the earpiece acoustical circuit. An earpiece with a closed back, sealed against the phone front cover, would have only the front cavity and front holes common with the phone mechanics. The intended solution for the Nokia 6110 phone extends this to include the back cavity as well. This way the locations and sizes of components on the Engine PCB can have some effect on the earpiece frequency response and leak tolerance in critical cases. It is not desirable to have very high components behind the holes in the UI PCB, since then the holes may be acoustically blocked (leading to a high Z_{ab}).

Since the front covers were to be cast in moulds, it was decided at an early stage not to use front holes with cross-sectional dimensions smaller than 0.7 mm in the 6110 phone (for reasons mentioned in sec. 6.1.4). Maximum hole sizes are restricted by dust and water shielding requirements. Dust and water shield membrane was not intended to be a part of the 6110 phone, so sufficient water resistance of acoustical openings was to be verified separately.

Although the back cavity was specified to have a great number of small openings (through holes) in it, it was known also that the PCB manufacturing process could block the through holes by solder resist, so one could not rely on the holes being open in each phone unit. This had to be taken into account in the acoustics design. Checking the effect of blocked holes is very easy in a simulation, but more difficult in a measurement made with real phone parts.

8.1.5 Solution 1 (starting point): leak in measurement

The development of the earpiece capsule was at first targeted at the solution involving a back cavity (as described above) behind the earpiece capsule. In type approval requirements at the time, earpiece measurements were made using a sealed (type 1) coupler. This would correspond to an infinite Z_{al} (see sec. 7.1.3). The situation could be improved by adding a stabilizing leak Z_{ax} according to the principles described in section 7.1.4: a special carefully specified, well behaved leak could be used between the A-cover and the measurement coupler during type approval measurements. Without a leak the low frequencies would be very loud, which would require heavy damping in the DSP equalization filters to fit the earpiece frequency response (strictly speaking: receiving frequency response, as described in GSM specifications) within type approval limits. Then bass response would suffer too much in normal use of the phone because of the inevitable leak that would be present between the phone and the user's ear. By using the additional purpose-built leak, the measurement would take account of the leaks in normal use and less damping of bass frequencies would be needed to pass type approval limits. This would appear to the user as a fuller and more pleasant sound (compared to conventional earpieces) because of the good bass reproduction.

8.1.5.1 Design rules

The best possible measurement leak would, naturally, be dimensioned according to PSS design rules for the capsule. This would mean providing a flat A-cover surface with a broad and shallow groove in it. When pressed against e.g. the Brüel & Kjær coupler (see appendix VI) having its seal and collar ring removed, this would form a narrow slit-shaped leak bounded by the groove and the edge around the microphone in the middle of the acoustic coupler, as indicated in the following drawing of the measurement setup used by PSS:

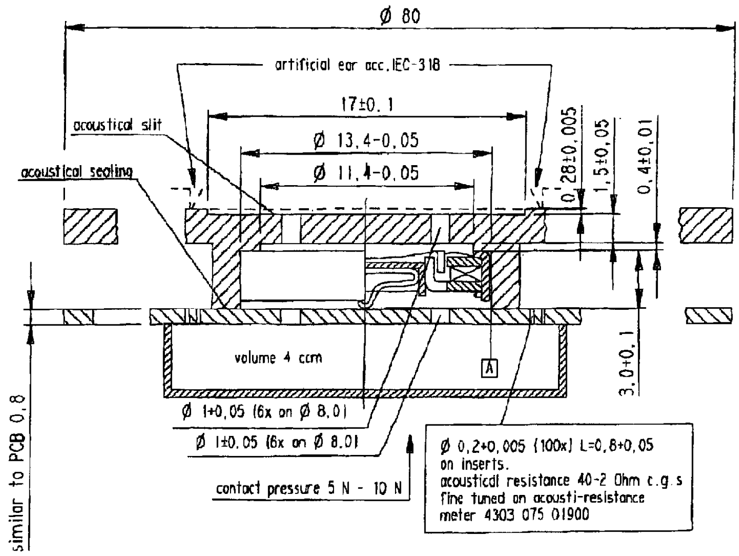


Figure 8.11: PSS design rules to get a good frequency response when measuring the 13 mm earpiece capsule (from [23]*). The parts below the capsule correspond to intended phone mechanics.*

The following leak dimensions would give good results according to measurements:

- effective width 18.6 mm (the width of the groove is 17 mm, but the coupler has a circular edge so the effective width is greater),
- height 0.28 mm (= depth of groove),
- length approx. 0.3 mm (the air is simply flowing past the edge of the coupler; the length in the air flow direction is thus not clearly defined).

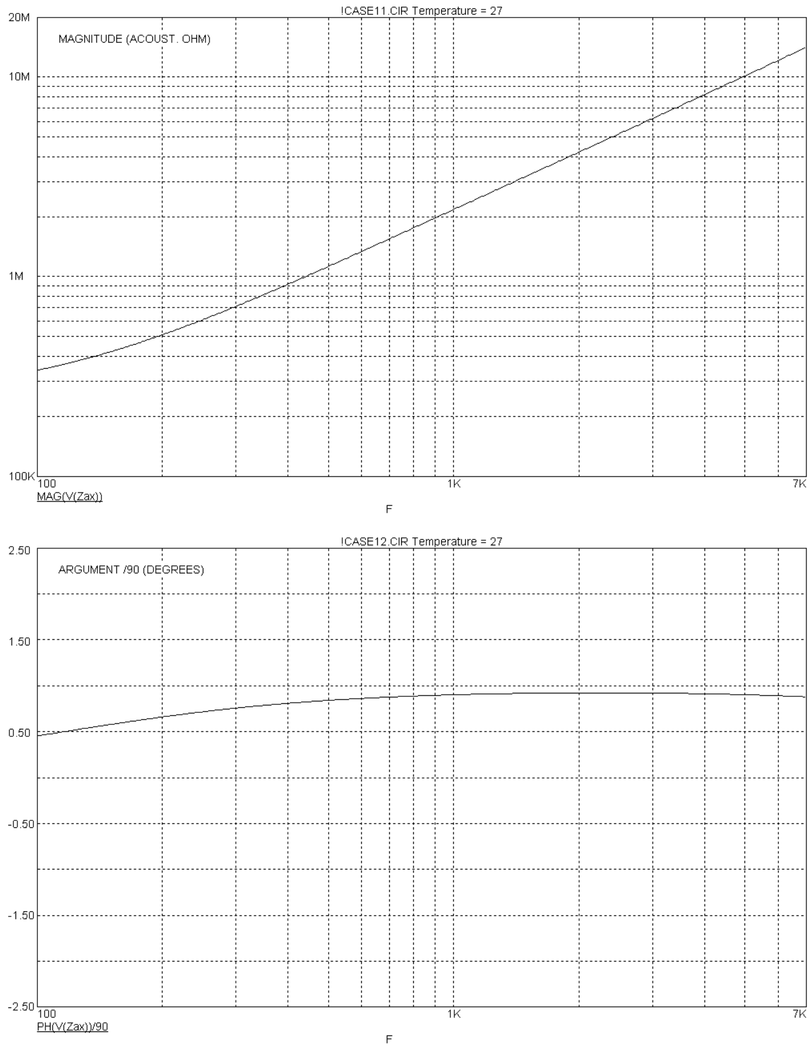


Figure 8.12: Simulated acoustic impedance of the leak formed by the groove and the type 1 coupler.

Other relevant dimensions are seen in the drawing (fig. 8.11). It is not necessary to ensure that the front cover and front holes of the phone be dimensioned exactly as in the drawing, however. The total front volume (between capsule membrane and A-cover) would be only about 0.06 cm^3 (0.04 cm^3 in the cover + 0.02 cm^3 because the membrane is slightly concave) if built this way. If this volume were to be enlarged, then a change would be seen in the frequency response, but this change could be compensated by changing also the dimensions and/or number of the front holes. An example of such tuning will be given later. Also the leak could be realized as some kind of separate adapter instead of a groove in the phone cover itself.

8.1.5.2 Performance

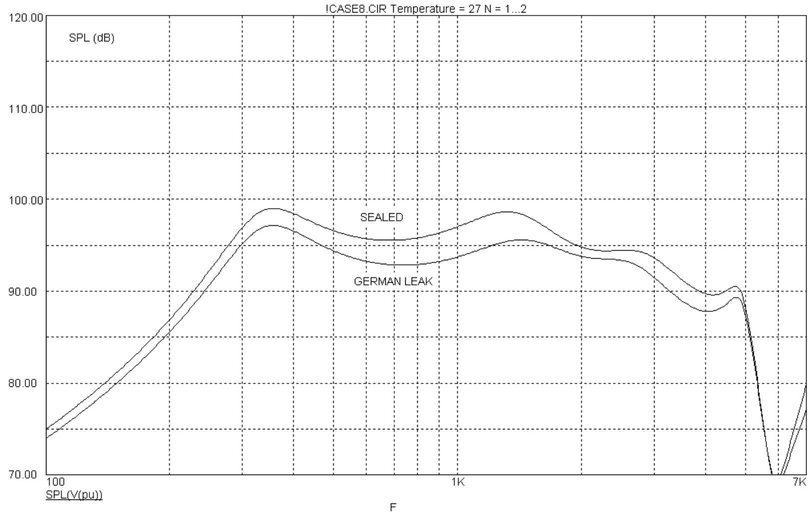


Figure 8.13: Simulated sound pressure level into type 1 coupler (sealed and with German leak ring) of solution 1. The input voltage amplitude is 100 mV (peak).

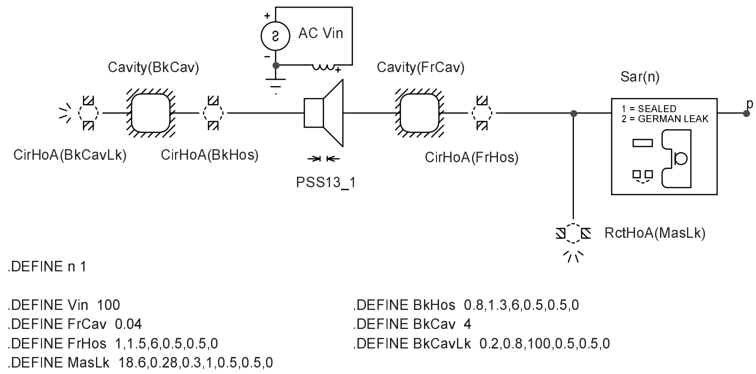


Figure 8.14: Schematic to obtain the result shown in fig. 8.13. BkHoles *are the six holes in the UI PCB under the capsule. The six front holes have a diameter of 1 mm (cover thickness 1.5 mm = hole length).*

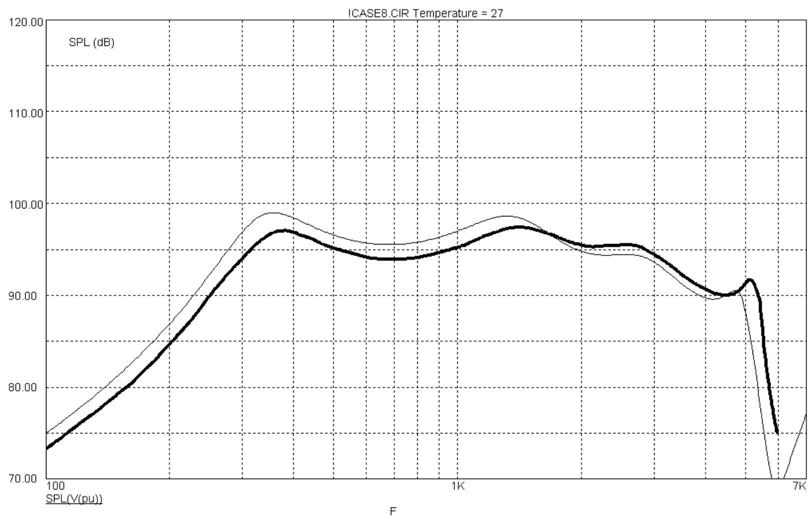


Figure 8.15: Comparison of measured (thick curve) and simulated (thin curve) frequency response of solution 1 (type 1 coupler without German leak, i.e. identical to "sealed" in fig. 8.13). The measurement result was obtained from PSS.

The simulated frequency response of solution 1 is well behaved: it would even pass GSM type approval with no equalization at all. The leak tolerance is also good (German leak ring vs no German leak ring). This is the case thanks to the additional stabilizing leak formed by the groove. The significant difference appearing if the groove is removed can be seen if the same simulation is run without the branch containing the RctHoleA(MeasLk) macro, and the groove is blocked by sealing material in the measurement:

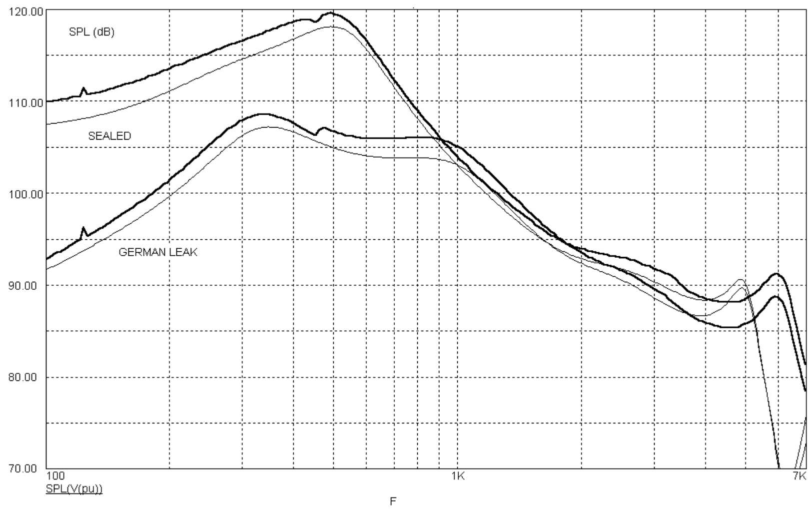


Figure 8.16: Simulations (thin curves) and measurements (thick curves) of solution 1 without the stabilizing leak (type 1 coupler, sealed and with German leak). The input voltage is the same as above.

The dramatic improvement is obvious in figs. 8.13 and 8.16. The level difference between low and high frequencies is, at most, close to 30 dB without the groove leak, while it is only a few dB with it (the two curves marked ”sealed” should be compared). The leak tolerance (sealed coupler vs German leak) improves from 8-13 dB without the groove to less than 5 dB with it. This is because the parallel impedance introduced by the groove leak is clearly smaller than the impedance of the measurement coupler with or without German leak (except at around 2 kHz and above, as seen by comparing figs. 8.12 and 8.2), so the stabilizing effect mentioned in sec. 7.1.4 is very efficient.

Unfortunately, in spite of its advantages, this straightforward solution based on optimized design rules is not a good choice in a commercial phone for the following reasons:

- Type approval requirements vary between systems (GSM, AMPS, PDC etc.). In some cases type approval measurement specifications explicitly require that the coupler be sealed (no extra leak groove or adapter etc. allowed).
- It may not be possible to shape the phone front cover to obtain the desired leak in a measurement.
- There would be a possibility of too high sound levels getting into the user’s ear (a product liability risk) — the pinna is quite soft and so it could block a groove in the phone cover.

8.1.5.3 Simulations vs measurements

The measurements match the simulations quite well except for the differences at the highest frequencies (the peak is due to the Helmholtz resonance of the front cavity). A probable reason for this is the geometry of the prototype: a front cavity of only 0.06 cm^3 means a distance of only about 0.5 mm from the earpiece capsule membrane to the A-cover. This changes the flow of air close to the front holes, so the Micro-Cap hole models based on simple ideal geometries may not be accurate enough. Also there are some tolerances in the earpiece capsule itself that affect these high frequencies.

The advantages of the macros and shapes developed for the Micro-Cap acoustics simulation system are becoming obvious looking at fig. 8.14: the actual physical topology of the solution (i.e. how cavities, holes and other things are interconnected) can be understood simply by looking at the schematic. For this reason only a couple of mechanics drawings are included. It is believed that, in the rest of the cases viewed here, all relevant information can be obtained from figs. 8.9, 8.10 and the simulation schematics themselves. Physical dimensions are seen as parameters to macros. It should be noted that the SI system is used, and thus for example a volume in cm^3 will have the postfix u (= µ in Micro-Cap) and a length in mm will have the postfix m.

The values given (both above and in most of the remaining simulations) to certain advanced parameters, such as space coefficients required by the hole and air load macros, are just rough estimates based on the mechanical construction (mutual coupling is simply ignored by setting the mutual coupling coefficients to zero). The reality is more complicated than the models, and acoustic impedance measurements would have to be made to find out the most realistic parameter values. Such analysis has not been carried out in this thesis.

8.2 Solution alternatives

In addition to the design rule based solution (solution 1), a few other solutions were considered in more or less detail to develop a final solution for the Nokia 6110 phone and its similar successors. Most of these had their own clear advantages and disadvantages, and the final solution had to be chosen with all the restrictions mentioned above and in chapter 6 (assembly requirements, environmental conditions, tolerances, industrial design, cost) in mind. The main concerns were how to route the leak that would have been handled by the leak in solution 1, and (which is just as important as good leak tolerance) how to obtain a stable frequency response, preferably equalizable without too complicated filters.

8.2.1 No front leak (solution 2)

One straightforward way of solving the front leak (groove or adapter) problem is to simply leave it out and design the equalizing filters according to the ”sealed” response shown in fig. 8.16. The leak tolerance is not good at the low frequencies, but on the other hand the solution has the advantage of being mechanically simple. Although tilted, the frequency response curve is quite smooth and should be easily equalizable except for the peak at 500 Hz.

Before going on to other possible solutions, it is informative to check the magnitudes of various impedances along the path from the front holes to the back cavity, and compare these with the impedances of the front leak and the ear:

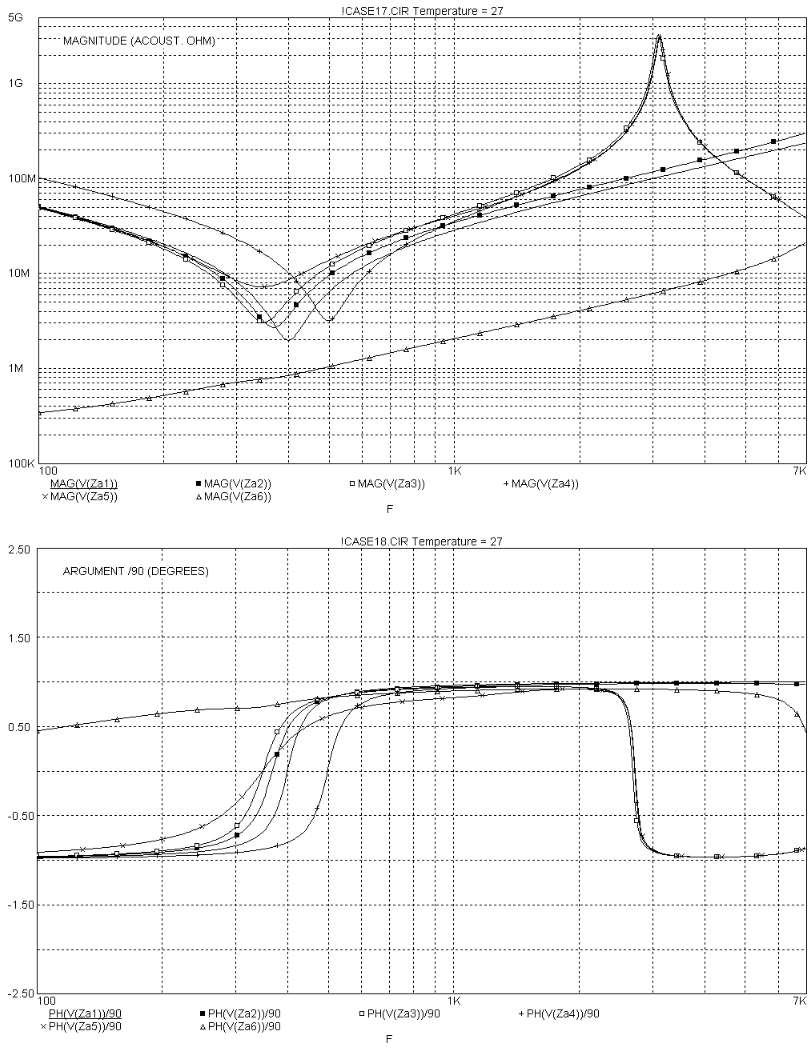


Figure 8.17: Simulated acoustic impedance (magnitude and argument) of: capsule only (Za1), UI PCB holes added (Za2), front cavity and front holes added (Za3), closed back cavity added (Za4), through holes added to back cavity (Za5), front leak added (Za6).

It is seen again in fig. 8.17 that the impedance of the front leak is much smaller than the impedance of the capsule and the structures connected in series with it. Consequently the front leak (which appears in parallel with the other impedances) plays a key role in minimizing the total impedance of the whole earpiece and it would be a pity to have to get rid of it completely. Other important conclusions from fig. 8.17 are:

- The holes behind the capsule in the UI PCB lower the main resonance slightly.
- The holes behind the capsule do not raise the total impedance appreciably (i.e. they are open enough).
- The back cavity leak (holes through PCB) have a significant effect — without them the main resonance is shifted up to 500 Hz, and with them the resonance remains unchanged in frequency but has a lower Q value (which is good for equalization).
- The front holes and front cavity do not raise the total impedance very much at low and midrange frequencies (there is a resonance at the highest frequencies).

The one-mass model, PSS13_3, was used in this simulation to disregard the effects of dome decoupling (for the sake of clarity). Solution 2 could be described as (1b), (2a), (3a) using the options defined at the end of the previous chapter.

8.2.2 Internal leak into parallel cavity (solution 3)

This solution is a good example of a solution that looks quite good in simulations, but does not work in reality unless dimensions of cavities and leaks can be realized as they were in the simulation. In other words, the solution is sensitive to certain changes in the mechanics.

Solution 3 differs from the above solutions in that a third cavity (hereafter called *parallel cavity* because it is, topologically, in parallel with the capsule) is added and connected between the back cavity and the front cavity. The volume available for this parallel cavity was estimated to be 3 cm^3, and no leaks were assumed. Realizable openings connecting the parallel cavity to the back cavity and front cavity were dimensioned manually (using simulation) until an acceptable frequency response was found. The results are:

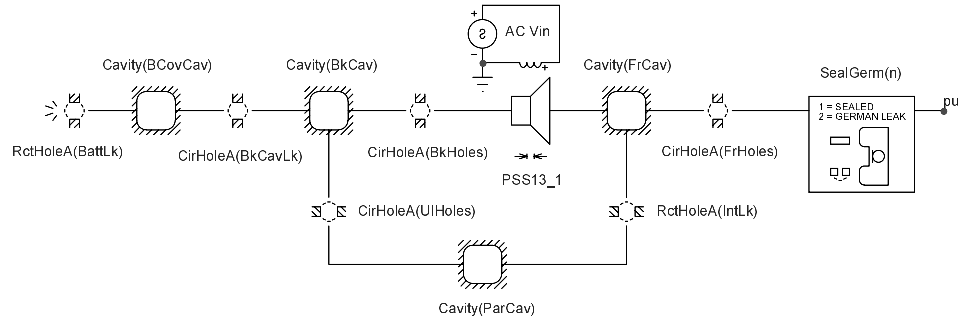
- 1 slit (hereafter called the *internal leak*) of width 20 mm, height 0.5 mm, and length 0.8 mm in flow direction (front cavity to parallel cavity), and
- 12 holes of diameter 1 mm (parallel cavity to back cavity).

The slit would be formed by opening up the cylindrical flange on the inner side of the A-cover (planned to seal the capsule to the A-cover, see also fig. 8.25 below). The 12 holes would be easily realizable in the UI PCB (provided of course that the PCB is not fully crowded with components). Hence the name, *UI PCB holes*, used for these holes from now on (the holes behind the capsule will be called *back holes* although they also reside in the same PCB). The UI PCB holes can also be seen as no. (20) in fig. 8.10. The number of holes shown in this picture is less than 12 because the final solution did not need all holes to be there.

Some assumed dimensions and leaks were also updated while this solution was sought, because the A-cover shape had been developed further and more information was available:

- front cavity 0.3 cm^3 (not 0.04 cm^3 + 0.02 cm^3 as above),
- extra cavity formed between B-cover and Engine PCB (volume 5 cm^3),
- slit-shaped leak between B-cover and phone battery; width 40 mm, height 0.2 mm and length (in flow direction) 7 mm.

It was assumed first (for simplicity) that all the through holes in the Engine PCB would lead to the cavity in the B-cover (hereafter called *B-cover cavity*). The leak route between the B-cover and the battery was thought to be the most significant leak out of the B-cover cavity (in spite of other smaller leaks that may be present), and thus it was used to represent the total leak in the simulations. As a final change, the front holes were shortened from 1.5 mm to 0.8 mm, which improved high frequency response a bit. The schematic to simulate this solution is as follows:




```
.DEFINE n 1

.DEFINE Vin  100m
.DEFINE FrCav  0.3u
.DEFINE FrHoles  1m,0.8m,6,0.5,1,0
.DEFINE IntLk  20m,0.5m,0.8m,1,0.5,0.5,0
.DEFINE ParCav  3u

.DEFINE BkHoles  0.8m,1.3m,6,0.5,0.5,0
.DEFINE BkCav  4u
.DEFINE BkCavLk  0.2m,0.8m,100,0.5,0.5,0
.DEFINE UIHoles  1m,0.8m,12,0.5,0.5,0
.DEFINE BCovCav  5u
.DEFINE BattLk  40m,0.2m,7m,1,0.5,0.5,0
```

Figure 8.18: Schematic to simulate solution 3.

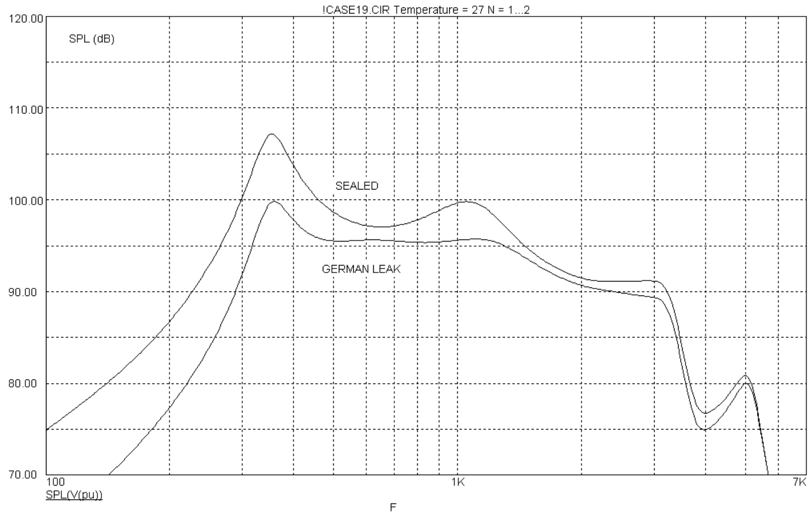


Figure 8.19: Simulated frequency response (and leak tolerance) of solution 3.

The leak tolerance of solution 3 is well behaved: the difference between sealed and German leak curves decreases smoothly from low to high frequencies. Even the frequency response itself is good, provided that the quite strong bass resonance does not shift along the frequency axis due to tolerances.

Solution 3 looks quite complicated. It was not built from scratch but rather found as a result of simulations using mechanical structures available at the time. Further analysis reveals that the underlying principle is fairly simple, and two of the four cavities could be omitted. One is the B-cover cavity and the other is the parallel cavity (the fact that the air flow route around the earpiece capsule goes through this cavity is not essential, it is the route itself that matters). Using the classification introduced at the end of the previous chapter, solution 3 could be described as a modification of (1c), (2b), (3a). All this will be briefly analyzed in the following simulations.

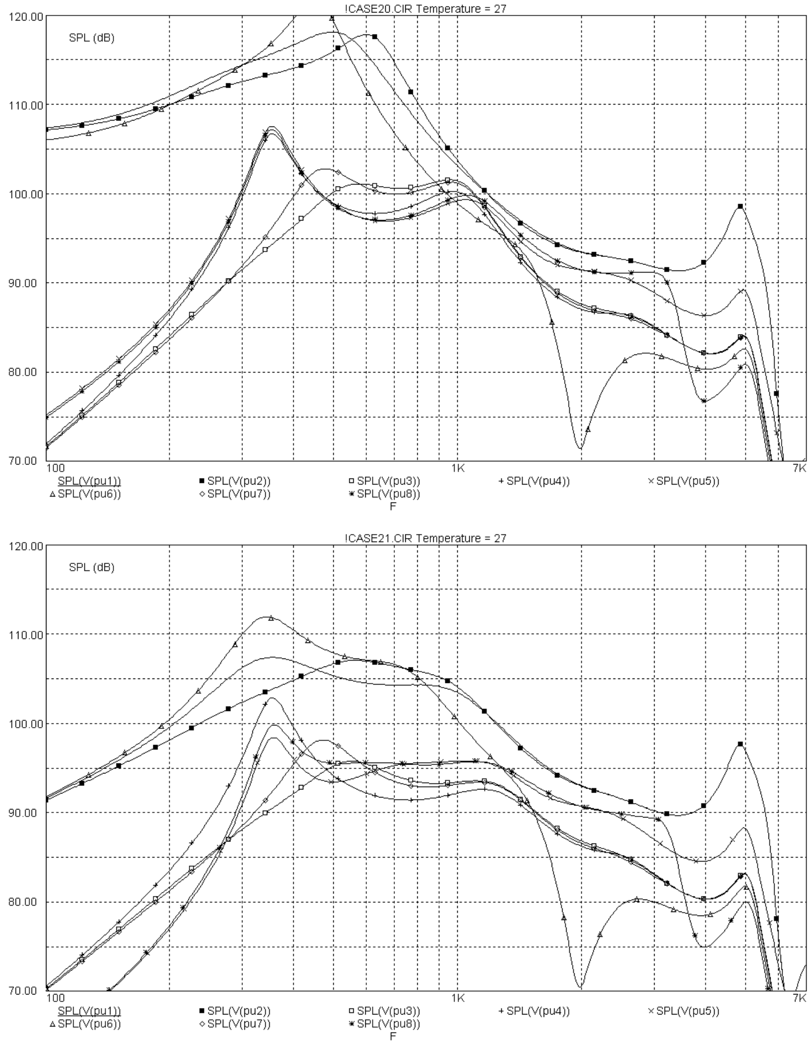


Figure 8.20: Frequency responses (top: sealed, bottom: German leak) of capsule with back holes, back cavity, through holes, front cavity and front holes (pu1), B-cover cavity and battery leak added (pu2), internal leak added (pu3), UI PCB holes added (pu4), internal leak and UI PCB holes connected together (pu5), parallel cavity connected to internal leak only (pu6), parallel cavity connected to UI PCB holes only (pu7), parallel cavity connected between UI PCB holes and internal leak (pu8).

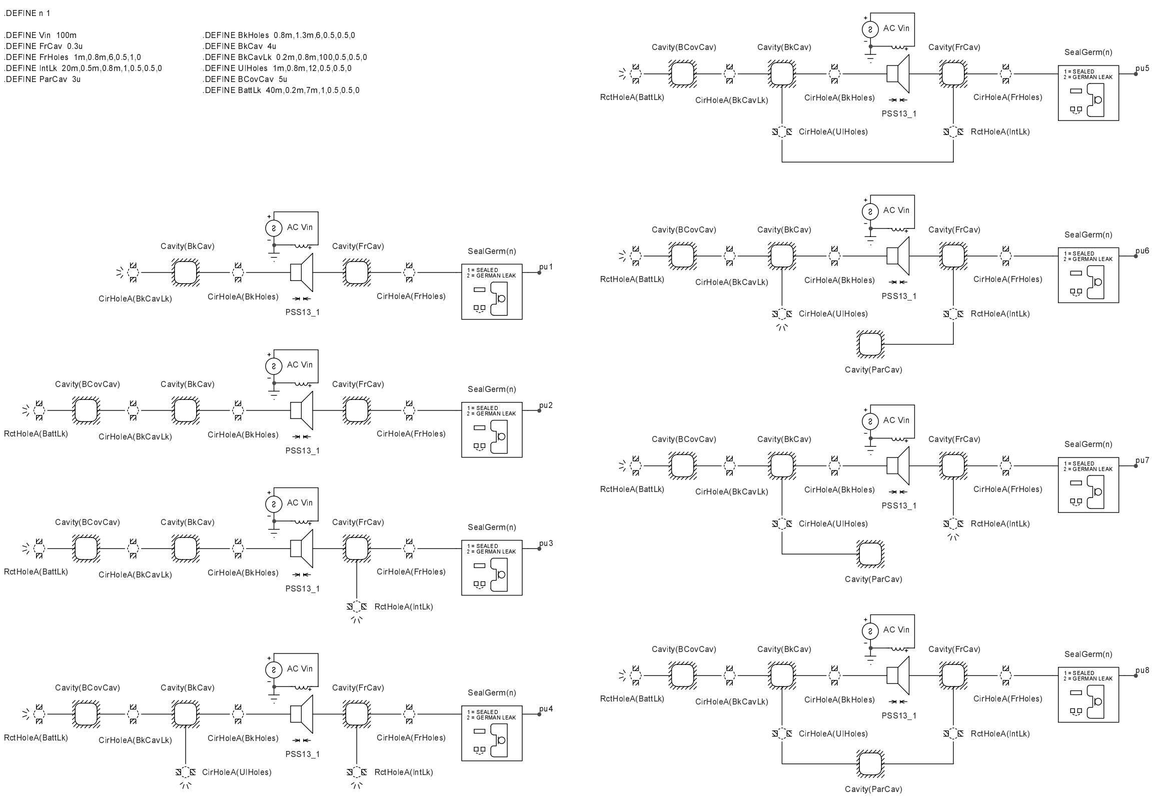


Figure 8.21: Schematic to obtain the results in fig. 8.20.

Comparison of the first two partial results (pu1, pu2) indicates that addition of the B-cover cavity and battery leak in series with the back cavity and through holes does not improve the frequency response in any way. Adding an internal front leak to free air (pu3), however, changes the behaviour dramatically, and the total earpiece output impedance is considerably reduced just as in solution 1. When the UI PCB holes are also opened to free air (pu4), the capsule membrane can move more freely at low frequencies. Its movement is not opposed by any acoustic compliance, ”air spring”, neither that of the back cavity nor that of the fairly tight volume formed between the phone cover and the type 1 coupler. This is a general fact: if the earpiece capsule has to be enclosed in a cavity smaller than a given limit (roughly 4 cm^3 for the capsule in question), then the main resonance seen at 400 Hz in free air will rise. A way to counteract this is to open up a big enough leak in the cavity — this bypasses the compliance of the enclosed air and lets the membrane move more freely. The best example so far is the leak formed by the through holes in the back cavity, as seen in fig. 8.17 (Za4 vs Za5).

Now, connecting the UI PCB holes to the internal leak (pu5, pu8) changes the topology of the solution to a kind of bypass or short-circuit solution (the front and the back of the speaker are connected together through a couple of cavities and holes). Strictly speaking, the ”pu5” simulation is not realistic: the UI PCB holes and the internal leak slit must necessarily have some air space between them because they are separated by several millimetres inside the phone. The result is still informative and therefore included here. This illustrates another strength of simulation: it is possible to try out solutions that have a theoretical/analytical interest although they may not be physically realizable in the phone mechanics in question. Also interesting is the ”pu6” case, which is identical to the complete solution apart from the disconnection of the parallel cavity from the back cavity. This forms a Helmholtz resonator (parallel cavity + internal leak) that absorbs most of the sound at 2 kHz, which is seen as a sharp dip in fig. 8.20.

As a result of closing the path through the parallel cavity, the high frequencies are boosted, which is good because they were unnecessarily weak in simulations pu3 and pu4. The reason obviously is that the impedance seen looking from inside the front cavity into the internal leak route around the capsule has risen at those frequencies.

Further analysis of this solution shows that the amount of leak from the parallel cavity to the back cavity is an important and critical parameter. Omitting this leak (pu6) destroys the high frequency response as was already pointed out. If the UI PCB is crowded and, say, only half the space is available for leak holes (i.e. 6 holes instead of the intended 12), the high frequencies will drop very quickly above 2.5 kHz. This is illustrated below:

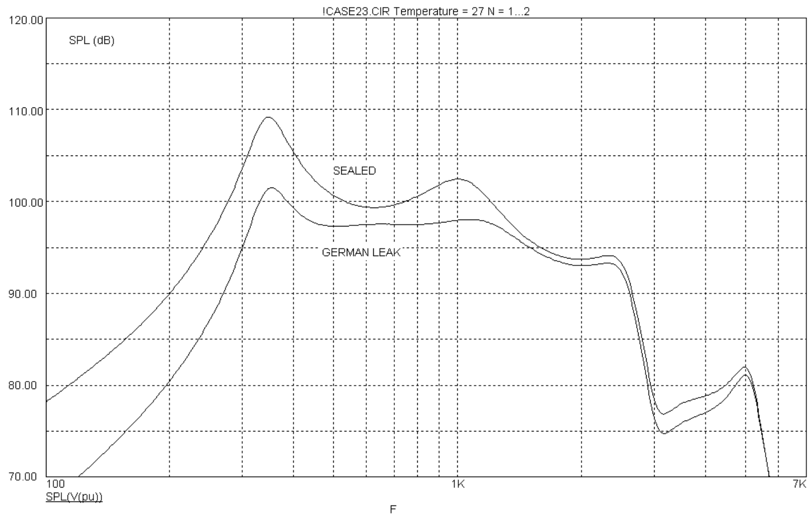
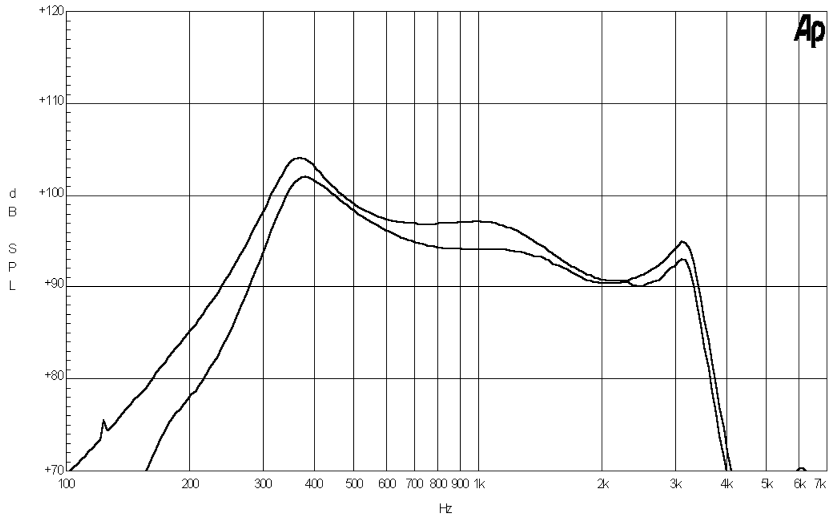


Figure 8.22: Simulation of solution 3 as in fig. 8.19, but with only 6 instead of 12 holes in the UI PCB.

A response like this would be next to impossible to equalize to fit within the GSM specifications for receiving frequency response (in a reliable way, remembering the inevitable tolerances).

Another problem is the fact that solution 3 assumes a quite small and well sealed parallel cavity (except for the connections to back and front cavity, of course). If the parallel cavity is clearly bigger than 3 cm^3 and other dimensions are kept unchanged, then high frequency response will also suffer. The same thing happens if the parallel cavity has big leaks, e.g. to free air. It turned out to be impossible to guarantee a sealed parallel cavity because of leaks between phone covers (if the whole space of air between the covers and the PCBs were to be used as a parallel cavity). This space of air is quite big (roughly 8.5 cm^3). Looking at figs. 8.9 and 8.10, it is easy to understand that creating a separate cavity structure beside the earpiece capsule would also be difficult (the UI PCB would already be crowded with components such as the buzzer and its driving circuitry).



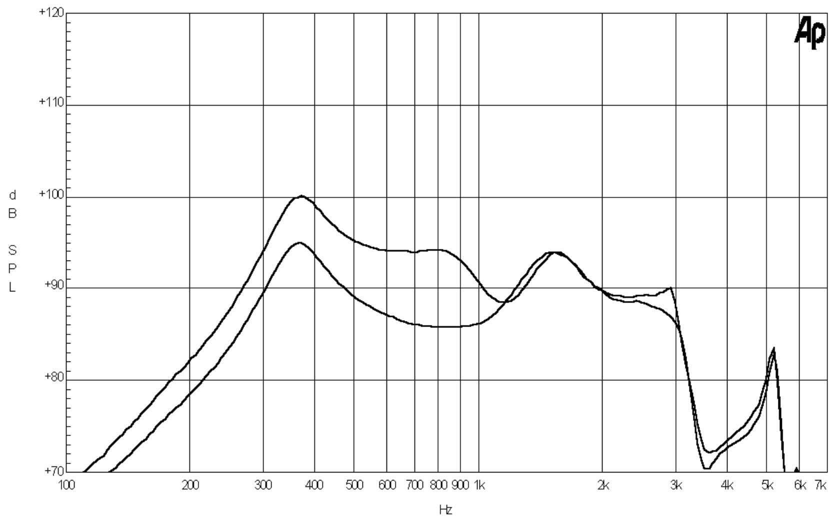


Figure 8.23: Measurements on a mechanical prototype showing ideal situation (top) and what happens when the parallel cavity is enlarged to 8 cm^3 and a big leak is added to it (bottom).

Figs. 8.23 show that the solution is able to work well if the parallel cavity is realized as in the simulations, but enlarging it and introducing a big leak reduces the cutoff frequency to less than 3 kHz. For this reason a simpler and less critical solution than solution 3 was sought, especially because it could not be known at the time exactly how big the cavities and leaks were going to be in the final product.

8.2.3 Internal leak into side cavity (solution 4)

It has been pointed out that the parallel cavity of solution 3 was not feasible to implement mechanically. The fact that the whole space of air between the covers is big (8.5 cm^3) and has leaks makes it appear acoustically as a big grounded capacitance (cavity volume) shunted by a small RL series impedance (leaks, e.g. joint between covers). If the cavity is big enough and/or the leaks between the covers are big enough, then the resulting impedance will be low. Acoustically, any leak routed to this cavity will approach a leak routed to free air (at least in a limited frequency range), provided that the cavity impedance is small compared to the impedance of the leak opening leading to the cavity. Of course, there will probably be a Helmholtz resonance of the cavity together with the leaks between the covers. But leaks in a phone tend to be narrow slit-like openings, which also means that their ratio of resistance to inductance is clearly higher than that of circular holes, so the Helmholtz resonance would be well damped anyway and most likely not cause problems. (More generally: the ratio of resistance to inductance of a constriction, e.g. a leak hole, depends on the ratio of perimeter to cross-sectional area. If the perimeter/area ratio is big, then the resistance is big compared to the inductance, and vice versa. The two extremes are a slit with a very high width/height ratio (high R/L) and a circular hole (low R/L). This can also be verified using the formulae found in the macros for rectangular and circular holes in appendix IV.)

The space of air between the covers and the PCBs will hereafter be called the *side cavity* because it formed around, not in front of or behind, the capsule. According to the above assumptions the side cavity has a fairly low impedance compared to other acoustic impedances around the earpiece capsule, so it seems that it could be used as a kind of substitute for free air.

All this suggests a simple solution with just an internal leak between the earpiece capsule and the A-cover (with the back cavity behind the capsule as described above). An illustration of what happens if the same internal leak as in solution 3 is used can be found in fig. 8.20, simulation ”pu4”. The leak tolerance is good, but the high frequencies are more damped than in solution 3. If the internal leak slit is narrowed below 0.5 mm, then low frequencies are boosted and the leak tolerance is reduced. However, the frequency response can be made to fall quite smoothly at a rate close enough to –6 dB/octave, so a first-order equalizing filter damping bass frequencies and boosting high frequencies could actually make the response look very good. If, on the contrary, a bigger slit is used, then the leak tolerance is improved but the overall output level is reduced (because most of the sound pressure is wasted through the internal leak). Reducing the overall level is undesirable: the earpiece capsule membrane has to move more to generate a given pressure inside the user’s ear, and more distortion will appear as a result.

Some updates to the mechanics were evident after the abandoning of solution 3. The Nokia 6110 back cavity volume was to become bigger than expected, about 7.5 cm^3 instead of 4 cm^3 (the exact space taken up by electronic components could not be known in advance). 7 holes were created in the UI PCB, although they are not needed as much as in solution 3. However, having these holes is a useful way of assuring that there will be a stable leak in the back cavity. This way tolerance effects can be reduced.

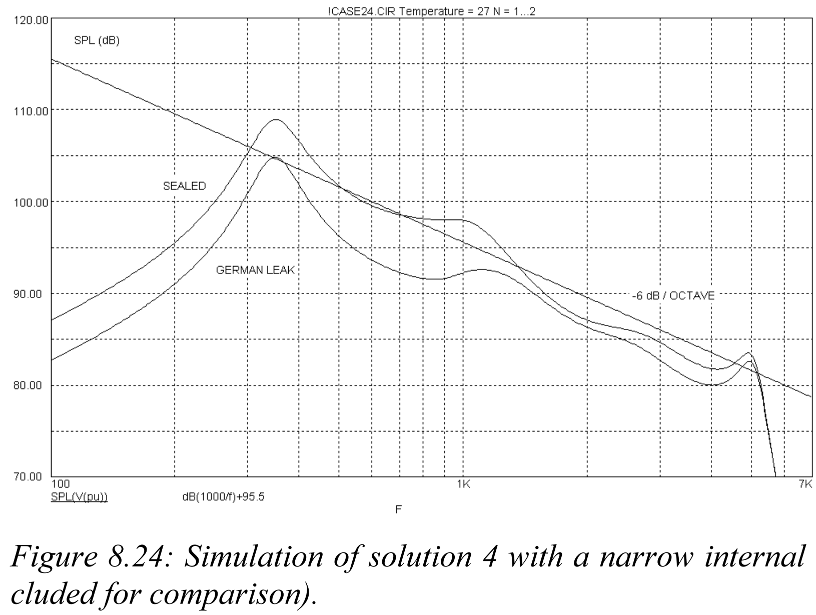


Figure 8.24: Simulation of solution 4 with a narrow internal leak (–6 dB/octave slope included for comparison).

The internal leak slit used in the above simulation had the following dimensions: height 0.12 mm, width 40 mm (extended around the whole perimeter of the capsule) and flow direction length 0.4 mm. A problem is obvious: the height is very small and thus sensitive to tolerances. The smoothness of the frequency response of solution 4 is a result of the resistive, damping nature of the internal leak (recalling again that a slit has a high R/L ratio), so the slit height has to be small. Alternatively a number of narrow vertical slits (instead of one horizontal) could be created between the capsule and the A-cover, but these would be very difficult to implement in reality because of mould limitations. A separate precision manufactured part could be used instead, but that would increase cost and complicate the assembly.

The side cavity was omitted and replaced by ground (free air) in the simulation of solution 4. This rough approximation will be analyzed in more detail later. Accurate pre-production simulations of this solution (or any other solution employing the side cavity) are made difficult by the fact that the effective size of all the leaks to outside air present in final mass-production mechanics are not known in advance. It would of course be possible to create dedicated leaks between the covers for acoustical purposes, and then one would know that the final leak will be greater than or equal to the total dedicated leak. Without such a dedicated leak, the simplest compromise in simulations is to replace the side cavity with either free air or a fully sealed cavity (of the assumed size). Then some slit could be added to the circuit and its dimensions stepped between probable values.

8.2.4 External leak into side cavity (solution 5)

According to current assumptions, the side cavity can perform well as a termination of a front leak, because in this respect its behaviour is close enough to that of free air. The front leak was internal in solution 4. The original solution, solution 1, had an external front leak to free air. Putting this all together reveals that another interesting solution (hereafter called solution 5) is possible: an external leak into the side cavity, through the A-cover, could be created outside the perimeter of the earpiece capsule. This option was also briefly mentioned in sec. 7.3.2. If a leak equivalent to the leak in the leak adapter used in solution 1 could be created in the A-cover, then it would in principle be possible to return to the good performance of solution 1 (provided that the side cavity is open enough). If such a solution were to be simulated with the side cavity approximated by free air, then one would arrive at a simulation schematic having the same topology as the one shown in fig. 8.14 for solution 1.

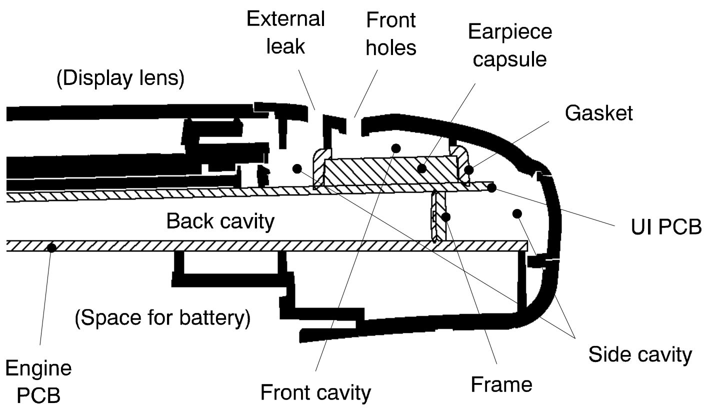


Figure 8.25: Cross-section (from a CAD file) through the earpiece of the Nokia 6110, showing the final mechanical construction. The back holes and UI PCB holes are not visible in this drawing.

A leak through the A-cover, as narrow as the one formed by the leak adapter of solution 1, is very desirable but unfortunately not a good option for the phone mechanics. This is partly due to tolerances (recalling that the leak has the shape of a narrow slit with a smaller dimension of only 0.28 mm), industrial design reasons, and mould technology. But the leak used in solution 1 gives some rough estimate of what kind of cross-sectional area would be required: width 18.6 mm and height 0.28 mm gives an area of roughly 5 mm^2. Unfortunately, this area would need to be further increased to keep the impedance equally low, because the leak in solution 1 had a very short physical length in the flow direction, while the external leak of solution 5 needs to go through the A-cover (which is about 1.5 mm thick).

Good mechanical prototypes were available at the time when this solution was being finalized, so measurements were used to find out possible sizes for the external leak and front holes. The front holes could be dimensioned optimally, but the leak holes were faced by conflicting requirements: water and dust shielding (requiring small holes), good leak tolerance (requiring a big total leak area), and mould technology and industrial design (requiring a few big holes instead of many very small holes). It is a purpose of the next section to explain the compromises made.

8.3 Solution 5 — the final choice

A measurement of the frequency response and leak tolerance of the finalized 6110 phone is given first for later reference:

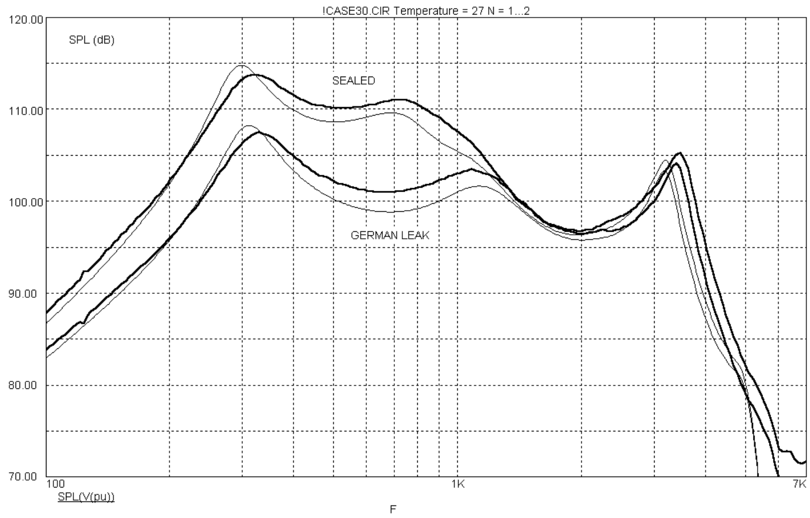


Figure 8.26: Frequency responses (unequalized, i.e. no DSP correction) of final 6110 phone (thin curves: simulated, thick curves: measured).

8.3.1 Dimensioning

8.3.1.1 Front holes

When the topology is chosen, the next step is to tune relevant dimensions. In this process it is good to start with the most straightforward parts of the design. Such parts are, in this particular case, the front cavity and front holes, forming together a Helmholtz resonance as described in sec. 7.3.1.3. Various industrial design options were considered (involving from 3 to 5 front holes). When the number of holes is not known in advance, a good result can be obtained simply by assuming some number and tuning the resonance to the wanted frequency by changing the diameter and/or length (in the flow direction) of the holes in the simulation, and recording the total area of the holes that gave the best results. The total area of the holes is the most critical parameter, thus if the number of front holes has to be changed one should try to keep the total area constant. Final tuning should preferably be done by measurements to avoid simulation inaccuracies. The front cavity in the Nokia 6110 was to be 0.274 cm^3 in the final mechanics, and three front holes were tuned to place the Helmholtz resonance at roughly 3.4 kHz. This way the resonance could be taken advantage of as a booster of the high frequencies. Being quite sharp, the resonance could not be placed inside the GSM speech transmission frequency range, since that would have made suitable DSP equalization of the receiving frequency response hard or impossible. The response drops rapidly above the resonance, hence the choice to have the resonance approximately at the upper limit of the required frequency range (3.4 kHz).

Industrial design provided three front holes, and as a result of tuning using both measurements and simulation each hole was chosen to have an area of 0.89 mm^2. The length of each hole in the flow direction was 1.6 mm. This choice placed the Helmholtz resonance at 3.3 kHz according to verification measurements (fig. 8.26). Although the front holes are not circular but elliptic in the 6110, simulations still show quite good correspondence with the measurements.

8.3.1.2 External leak holes

The final industrial design choice allocated 3 holes for external leak through the A-cover. Having one wide slit as in solution 1 was not desirable for mechanical and industrial design reasons, as stated above. So the goal was to find out a sufficient hole area to fulfil conflicting requirements. The final decision to go for an area of 1.05 mm^3 per hole (3.16 mm^2 all in all) was based on results from measurements on prototypes that were available. Reducing the length of the leak holes from 1.4 mm (thickness of A-cover where the leak holes are) to less than 1 mm would improve the performance, but at the expense of reduced water resistance. Consequently the leak holes were not shortened.

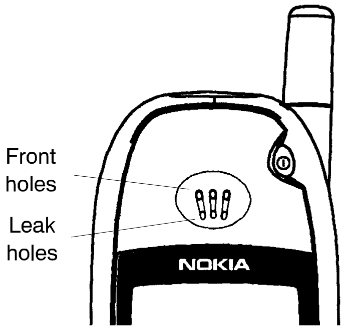


Figure 8.27: The Nokia 6110 earpiece viewed from the outside (seen also in fig. 2.1).

The disadvantage of having a smaller leak area (and holes that are quite long in the flow direction) is that the leak tolerance is worse than that obtained in the idealized solution 1. The advantage is that the sensitivity (of the earpiece as a whole) is higher than in solution 1, i.e. less power is required to reach a given sound pressure level in the user's ear. Further analysis of solution 5 shows that low frequency output is higher because of the higher impedance in the leak, while high frequency output is higher because of the front cavity Helmholtz resonance. The 2 kHz region is the only region having almost as low output as solution 1.

Good prototypes were available during the tuning of solution 5, so it was possible to verify that a leak through the A-cover really worked, although it did not lead directly to the outside air but to the side cavity inside the phone. Simulation of the side cavity is not immediately possible since the leaks and losses in this irregular space of air are very hard to estimate without special acoustic impedance measurements. It has already been pointed out that the side cavity would probably appear as a capacitance shunted by an RL series circuit (sec. 8.2.3). The capacitance can be calculated from the side cavity volume (8.5 cm^3), but the resistance and the inductance are unknown. An unconventional but very simple measurement was set up to allow some rough estimation of these components: the type 1 coupler was moved to cover the leak holes only, while the front holes were allowed to radiate sound into free air. In other words, the transmission of sound from the earpiece capsule through the back cavity, UI PCB holes, side cavity and leak holes was measured. In this chain everything can be simulated with reasonable accuracy except for the unknown side cavity resistance and inductance. Thus it seems logical that the two unknown components could be estimated by trying to match a simulated response (through the same chain) to the measured result. The matching would be done manually by trying different resistance and inductance values. The outcome of this modelling was a success, illustrating in a great way the power of simulation when combined with measurements:

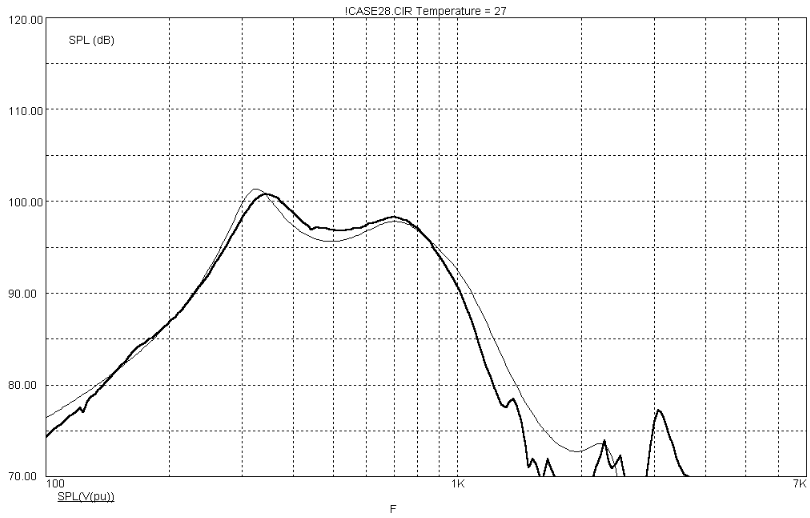


Figure 8.28: Sound output through leak holes only (using a type 1 coupler without a German leak ring). The thick curve is the measurement result to which the simulated result (thin curve) was matched.

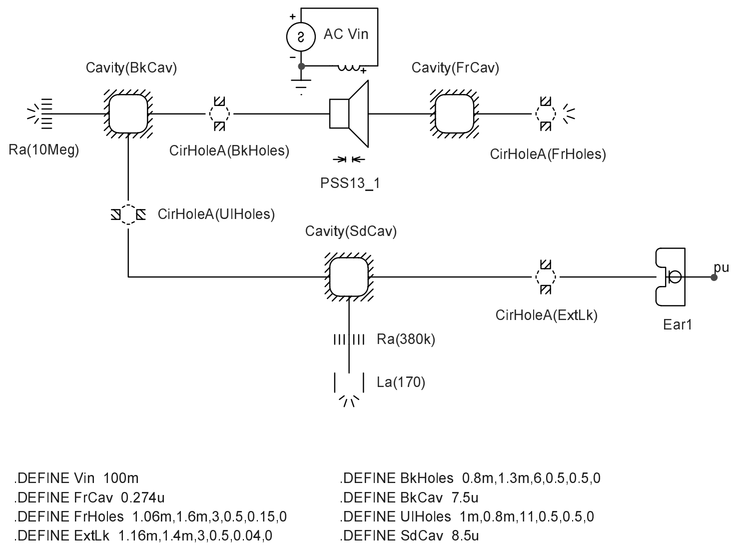


Figure 8.29: The schematic used in the simulation of fig. 8.28.

As seen in fig. 8.28, the side cavity was found to behave as if it had a shunt acoustic RL impedance corresponding approximately to a resistance of 380 kΩ and an inductance of 170 H. This impedance could be translated to a corresponding mechanical leak opening: for example, a slit of width 100 mm, height 0.2 mm and length 1.4 mm would have the same impedance at low frequencies. This is very reasonable and close to the leak one would expect between the phone covers. One more pure acoustic resistance is seen in the schematic: a ”grounded” resistance of 10 MΩ, effectively shunting the back cavity capacitance. This resistance was added to take account of losses that turned out to be higher in reality than in the simulated back cavity. This is not surprising since the back cavity walls are not perfectly rigid as assumed in the Cavity macro. The 10 MΩ resistance was not found as a result of careful measurements but it is just a rough estimate that improved the correspondence between certain measurements and simulations with blocked PCB holes (results of these will not be analyzed here). The space coefficients passed to the CirHoleA(FrHoles) (front holes) and CirHoleA(ExtLk) (external leak) macros are quite small; this is to take account of the fact that the holes are quite close to each other and the solid angle should be divided between the holes. Some fine-tuning of the space coefficients was done also to get a better correspondence between measured and simulated results.

Now the assumption that the back cavity impedance is low enough to work as a good termination of the external leak can be verified:

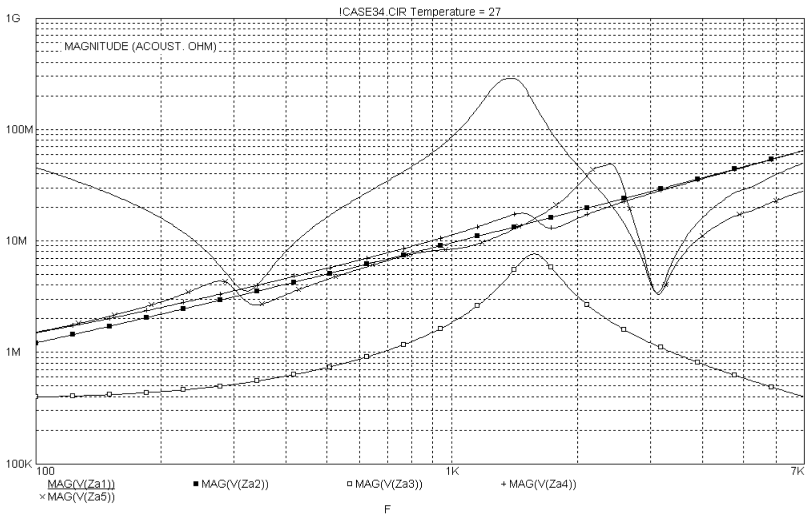


Figure 8.30: Simulated acoustic impedances (magnitude) of: earpiece without the external leak (Za1), external leak holes only (Za2), side cavity only (Za3), external leak connected to side cavity (Za4), earpiece with the external leak (Za5).

```
.DEFINE FrCav 0.274u                          .DEFINE BkHoles 0.8m,1.3m,6,0.5,0.5,0
.DEFINE FrHoles 1.06m,1.6m,3,0.5,0.15,0       .DEFINE BkCav 7.5u
.DEFINE ExtLk 1.16m,1.4m,3,0.5,0.04,0         .DEFINE UlHoles 1m,0.8m,11,0.5,0.5,0
                                              .DEFINE SdCav 8.5u
```

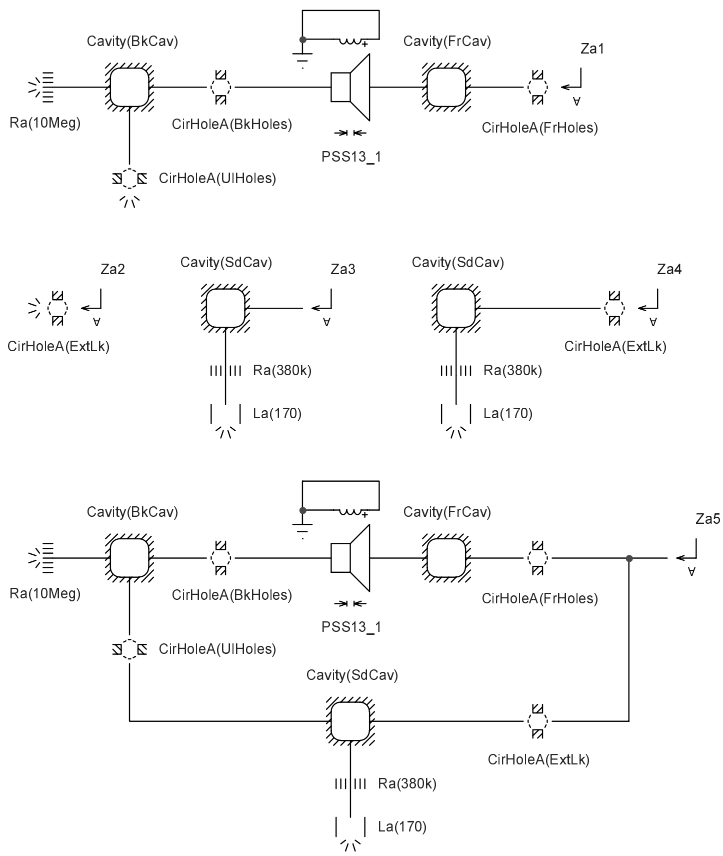


Figure 8.31: Schematic used in the above simulation.

As seen in the above simulation result the side cavity impedance (Za3) remains clearly lower than the impedance of the leak holes (Za2) at all frequencies of interest, so according to the simulations the leak behaves much the same way as it would do if routed to free air. The back cavity has some effect, however, so it is viewed next.

8.3.1.3 Leaks in the back cavity

As stated above, 7 holes were created in the UI PCB to maintain a leak in the back cavity even when the through holes are blocked by solder resist (which happens most of the time according to visual inspection of prototypes). Without the UI PCB holes the sharpness of the resonance seen at 300 Hz (fig. 8.26) would vary depending on the number of open through holes. During the fine-tuning measurements it turned out that a slightly smoother response could be obtained by adding 4 more holes. This is seen in fig. 8.26 as a smooth peak at 700 Hz; with 7 holes instead of 11 this peak would be sharper. There was no space for the 4 holes in the UI PCB, so they were added instead to the plastic frame between the PCBs (fig. 8.10, no. (24)).

The back cavity has roughly the following dimensions: width 37 mm, height 55 mm and depth 4 mm. One might expect that if the back cavity were accidentally very well sealed in the ”lower” end (no through holes open), then there might perhaps appear irregularities in the frequency response because of standing waves inside the back cavity. Two of the dimensions just mentioned are no longer very small compared to the wavelength at the highest speech transmission frequencies, so the back cavity could accidentally act as a quarter-wave resonator at those frequencies. Standing waves were found out to cause visible changes in the frequency response in an early prototype (which had a harder frame and whose back cavity was unrealistically well sealed in the ”lower” end). However this problem has never reappeared in later prototypes or mass-produced units, indicating that there are high enough losses in the back cavity to prevent standing waves from destroying the performance.

8.3.2 Final simulation model

The final simulation model of the Nokia 6110 earpiece is seen in fig. 8.31 (the lowest circuit to which the ”Za5” probe is connected). The many small PCB through holes are left out because they tend to be blocked in most units as was already mentioned.

This model was also tried with blocked leaks between phone covers. Blu-Tack™ was used as sealing material in a corresponding measurement, and the RL series branch accounting for the leaks was removed in the simulation. For some reason the model failed to predict the frequency response in real mechanics in this particular case, indicating that the derived 6110 earpiece model is not failsafe. Separate modelling would probably have to be done to get a better correspondence between measurements and simulation even when the leaks in the side cavity are removed. It is hard to eliminate all leaks in the side cavity in a real measurement, and even when they are eliminated there will be other losses e.g. due to the not fully rigid mechanical structures inside the phone. This may be another reason why the model does not yield realistic results when leaks between phone covers are eliminated.

8.3.3 The final earpiece

The receiving frequency response of the final earpiece was tuned to specifications using separate DSP equalization emphasizing the 2 kHz region and slightly damping the bass region. A measurement of the response, done through the whole transmission path using artificial speech, is shown below. It should be noted that this measurement does not correspond to type approval specifications, in which a *DAI* (*digital audio interface*) is used to bypass the speech coding. The response thus measured would be very similar to the one shown here, however.

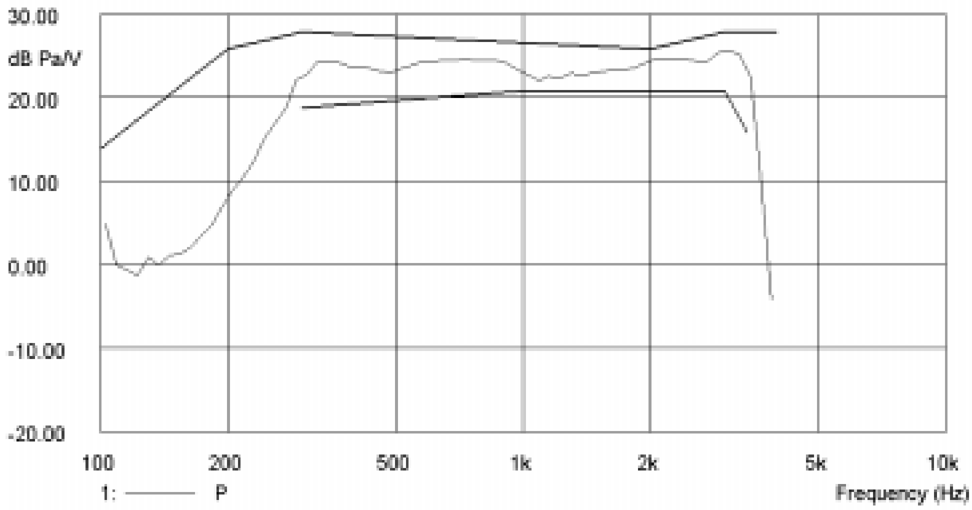


Figure 8.32: Receiving frequency response of a Nokia 6110 phone measured using artificial speech through the entire transmission path (not through a DAI as in type approval measurements).

As seen above, it was no problem to get a good-looking frequency response using DSP equalization. The phone passed type approval as intended.

Now, at the end of this analysis, the performance of the 6110 earpiece can be compared to that of the Nokia 1610 earpiece whose response and leak tolerance was shown at the beginning of this chapter. This provides a picture of the improvement. Comparing figs. 8.1 and 8.26 at e.g. 300 Hz, one can see that the difference between sealed and leaking (German leak) conditions appears as a 14 dB drop in the 1610, while the drop in the 6110 is just 7 dB. The net result is a $14-7=7$ dB louder output of this frequency compared to high frequencies (which are not sensitive to leaks of this size according to the same measurement results), i.e. a significant improvement in the bass response as experienced by the phone user. It should be remembered again that the shapes of the ”sealed” responses do not matter in this leak tolerance comparison because they are flattened anyway using DSP equalization; it is only the relative difference between sealed and leaking conditions that matters.

Some fine-tuning of the DSP equalization was done to get a good-sounding response within the type approval limits. All in all, the earpiece has proven to be a successful design according to opinions stated by end users and testers in magazine reviews. The same basic solution has been used in Nokia 6110 successors belonging to the same generation of products, e.g. Nokia 5110.

9 Conclusions

This thesis has described the development of a new acoustical solution for a leak tolerant earpiece in a Nokia mobile phone as an example of an interesting and challenging design task. A useful set of simulation models was developed to aid in this work. Relevant background theory has been introduced, and documentation provided to make it possible to understand the usage and limitations of the simulation models. A few important models were derived from results in the literature that have not yet been presented in a more general way — namely, the impedances of very small circular and rectangular openings and radiators, and especially the frequency dependence of these. A couple of special choices were made to make the simulation tools more flexible, e.g. the inclusion of a fourth domain in addition to the usual three, and the choice to use different analogies in the mechanical domain than in the other three domains. The complete model collection has capabilities that actually go quite far beyond what was needed in the design described in this thesis.

The practical aspects of mobile phone audio/acoustics design have been overviewed to provide a picture of the advantages and disadvantages of simulation when applied to such design, and to explain choices made in the Nokia product whose earpiece solution has been analyzed. Simulation proved to be powerful in the design process, and the designed earpiece itself proved to be a clear improvement compared to those used in earlier phone generations.

It is worth noting that the situation has changed in a very favourable way during the writing of this thesis, at least as far as GSM type approval is concerned. When previously GSM type approval measurements of receiving frequency response had to be done using a sealed coupler, they can now (while this thesis is being finished) be done using standard leaking couplers as well. This reduces the reasons to aim at good leak tolerance from three (1: more realistic type approval measurements, 2: product liability, 3: the sound depends less on how the phone is held against the ear) to only two (no 1 is no longer as relevant as before). However, this does not mean that leak tolerance is less important than before. Leak tolerance has, in effect, simply become something that should be implemented as a good feature that makes the use of any phone more comfortable, instead of a feature that makes type approval measurements more realistic and allows one to use a more favourable DSP equalization. The ever-increasing demands of end users can only make good leak tolerance even more desirable in future products.

10 References

1 Tuttlebee, W. H. W. Cordless telecommunications in Europe. London, UK, 1990. Springer-Verlag.

2 Chari, N. C., Herman, G., Danhauer, J. L. Perception of one-third octave band filtered speech. J. Acoust. Soc. Am., vol. 61, 1977. Ref. in [40].

3 Digital cellular telecommunications system (phase 2+), mobile station (MS) conformance specification (GSM 11.10-1 version 5.7.0), chapter 30.3. Receiving sensitivity/frequency response. 1998. ETSI.

4 Crocker, M. J. (editor). Encyclopedia of acoustics, vol. 4. New York, USA, 1997. John Wiley & Sons, Inc.

5 Benson, K. B. (editor). Audio engineering handbook. USA, 1988. McGraw-Hill.

6 Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill.

7 Aplac 7.0 user's manual. 1996. Karisto Oy.

8 Bauer, B. B. Equivalent circuit analysis of mechano-acoustic structures. J. Audio Eng. Soc., vol. 24, no. 8, 1976.

9 Micro-Cap V user's guide. 1997. Spectrum Software.

10 Micro-Cap V reference manual. 1997. Spectrum Software.

11 Backman, J. Distortion from boundary layers (draft). 1997.

12 Product data: ear simulator for telephonometry - type 4185. Brüel & Kjær, Nærum, Denmark, 1997.

13 Product data: wideband ear simulator for telephonometry - type 4195. Brüel & Kjær, Nærum, Denmark, 1997.

14 Olson, H. F. Acoustical engineering (1991 edition, orig. D. van Nostrand Co. 1957). Philadelphia, PA, USA, 1991. Professional Audio Journals Inc.

15 Nolle, A. W. Small-signal impedance of short tubes. J. Acoust. Soc. Am., vol. 25, no. 1, 1953.

16 Borwick, J. (editor). Loudspeaker and headphone handbook. 2nd edition. Oxford, UK, 1994. Focal Press.

17 Ingard, K. U. On the theory and design of acoustic resonators. J. Acoust. Soc. Am., vol. 25, no. 6, 1953.

18 Kinsler, L. E., Frey, A. R. Fundamentals of acoustics. Monterey, Calif., USA, 1982. John Wiley & Sons, Inc.

19 Small, R. H. Direct-radiator loudspeaker system analysis. J. Audio Eng. Soc., vol. 20, no. 2, 1972.

20 Wüttke, J. Microphones and wind. J. Audio Eng. Soc., vol. 40, no. 10, 1992.

21 Poldy, C. A. Electrical analogs for membranes with application to earphones. J. Audio Eng. Soc., vol. 31, no. 11, 1983.

22 Data sheet no. 590-1 for earpiece capsule type WD00518. Philips Speaker Systems, Vienna, Austria.

23 Data sheet no. 590-5.1 for earpiece capsule type WD00518. Philips Speaker Systems, Vienna, Austria.

24 Stinson, M. R., Shaw, E. A. G. Acoustic impedance of small, circular orifices in thin plates. J. Acoust. Soc. Am., vol. 77, no. 6, 1985.

25 Crandall, I. B. Theory of vibrating systems and sound. New York, USA, 1926. Van Nostrand. Ref. in [24].

26 Ingard, K. U. Notes on sound absorption technology. USA, 1994. Noise Control Foundation.

27 Pritchard, R. L. Mutual acoustic impedance between radiators in an infinite rigid plane. J. Acoust. Soc. Am., vol. 32, 1976. Ref. in [28].

28 Jacobsen, O. Some aspects of the self and mutual radiation impedance concept with respect to loudspeakers. J. Audio Eng. Soc., vol. 24, no. 3, 1976.

29 Morse, P. M., Ingard, K. U. Theoretical acoustics. Princeton, New Jersey, USA, 1990. Princeton University Press.

30 Bank, G., Wright, J. R. Radiation impedance calculations for a rectangular piston. J. Audio Eng. Soc., vol. 38, no. 5, 1990.

31 Kreyszig, E. Advanced engineering mathematics. 1988. John Wiley & Sons, Inc.

32 Leach, W. M., Jr. On the electroacoustic-analogous circuit for a plane wave incident on the diaphragm of a free-field pressure microphone. J. Audio Eng. Soc., vol. 38, no. 7/8, 1990.

33 Bradbury, L. J. S. The use of fibrous materials in loudspeaker enclosures. J. Audio Eng. Soc., vol. 24, no. 4, 1976.

34 Leach, W. M. Electroacoustic-analogous circuit models for filled enclosures. J. Audio Eng. Soc., vol. 37, no. 7/8, 1989.

35 Morse, P. M. Vibration and sound (2nd edition). New York, USA, 1948. McGraw-Hill. Ref. in [6].

36 Suzuki, H. Mutual radiation impedance of a double-disk source and its effect on the radiated power. J. Audio Eng. Soc., vol. 34, no. 10, 1986.

37 Toivanen, J. Teknillinen akustiikka. Espoo, Finland, 1991. Otatieto Oy.

38 Thurston, G. B., Martin, C. E. Periodic fluid flow through circular orifices. J. Acoust. Soc. Am., vol. 25, 1953.

39 Thurston, G. B. Periodic fluid flow through circular tubes. J. Acoust. Soc. Am., vol. 24, 1952. Ref. in [24].

40 Rossing, T. D. The science of sound (2nd edition). USA, 1990. Addison-Wesley.

Appendix I Wavelengths and dimensions

Table I.1: Approximate physical dimension corresponding to a phase shift of π/6 as a function of sound frequency (for sound waves in air at room temperature). This is a kind of rough estimate of the greatest dimension allowed in lumped modelling if accurate results are wanted.

Frequency	Dimension (approx.)
100 Hz	30 cm
150 Hz	20 cm
250 Hz	10 cm
400 Hz	7 cm
650 Hz	4.5 cm
1 kHz	3 cm
1.5 kHz	2 cm
2.5 kHz	1 cm
4 kHz	7 mm

Table I.2: Approximate frequency corresponding to a wavelength of half the perimeter of a radiator as a function of the perimeter. This is a rough estimate of the highest frequency below which a radiator can be considered omnidirectional.

Perimeter	Frequency (approx.)
3.5 cm	20 kHz
6 cm	11 kHz
10 cm	7 kHz
20 cm	3.5 kHz
35 cm	2 kHz
60 cm	1 kHz

Appendix II Analogous quantities

Quantities and units

Table II.1: Summary of analogous quantities and units in the SI and cgs systems. Some less common and/or unnamed quantities are left out.

Electrical	Mechanical domain		Acoustical domain		Radiation domain	
	SI units	Cgs units	SI units	Cgs units	SI units	Cgs units
Voltage u	Velocity v		Pressure p		Pressure p	
volt (V)	m/s	cm/s	pascal (Pa)	microbar (μbar)	pascal (Pa)	microbar (μbar)
Current i	Force F		Volume velocity q		Particle velocity v	
ampere (A)	newton (N)	dyne (dyn)	m^3/s	cm^3/s	m/s	cm/s
Charge q	Impulse I		Volume V		Particle displacement x	
coulomb (C)	Ns	dyns	m^3	cm^3	m	cm
Magnetic flux Φ	Displacement x					
weber (Wb)	meter (m)	centimeter (cm)				
Impedance Z	Mechanical admittance Y_m		Acoustic impedance Z_a		Specific acoustic impedance Z_{as}	
ohm (Ω)	m/Ns	cm/dyns	Pas/m^3	$\mu bars/cm^3$	Pas/m	μbars/cm
Admittance Y	Mechanical impedance Z_m		Acoustic admittance Z_a			
mho	Ns/m	dyns/cm	m^3/Pas	$cm^3/\mu bars$		
Resistance R	Mechanical conductance (responsiveness) G_m		Acoustic resistance R_a			
ohm (Ω)	m/Ns	cm/dyns	Pas/m^3	$\mu bars/cm^3$		
Conductance G	Mechanical resistance R_m		Acoustic conductance G_a			
mho	Ns/m	dyns/cm	m^3/Pas	$cm^3/\mu bars$		
Inductance L	Mechanical capacitance (compliance) C_m		Acoustic inductance (inertance) L_a			
henry (H)	m/N	cm/dyn	kg/m^4	g/cm^4		
Capacitance C	Mechanical inductance L_m (mass m)		Acoustic capacitance C_a			
farad (F)	kilogram	gram (g)	m^3/Pa	$cm^3/\mu bar$		

	(kg)					
Energy E joule (J)	Energy E joule (J)	erg (erg)	Energy E joule (J)	erg (erg)		
Power P watt (W)	Power P watt (W)	erg/s	Power P watt (W)	erg/s	Intensity I W/m^2	erg/m^{2}s

Conversion factors

Table II.2: Conversion factors for SI and cgs units and various quantities. To convert a value in one system to the other, the value has to be scaled by the factor in the corresponding column.

Quantity	SI → cgs	Cgs → SI
Force	10^5	10^{-5}
Velocity	100	0.01
Displacement	100	0.01
Impulse	10^5	10^{-5}
Mechanical impedance	10^3	10^{-3}
Mechanical admittance	10^{-3}	10^3
Mechanical resistance	10^3	10^{-3}
Mechanical conductance (responsiveness)	10^{-3}	10^3
Mechanical inductance (mass)	10^3	10^{-3}
Mechanical capacitance (compliance)	10^{-3}	10^3
Pressure	10	0.1
Volume velocity	10^6	10^{-6}
Volume	10^6	10^{-6}
Acoustic impedance	10^{-5}	10^5
Acoustic resistance	10^{-5}	10^5
Acoustic inductance (inertance)	10^{-5}	10^5
Acoustic capacitance	10^5	10^{-5}
Power	10^7	10^{-7}
Energy	10^7	10^{-7}

Appendix III Derivations

Holes

Circular hole

The relevant parameters of one hole are: d (diameter) and l (physical length in flow direction). By assumption, $d, l << \lambda$. From [24, 25], the acoustic impedance can be written as

$$Z_a = -j\frac{J_0(k_c d)}{J_2(k_c d)} \cdot \frac{4\rho_0 \omega l}{\pi d^2} \tag{III.1}$$

with the coefficient

$$k_c = \frac{1-j}{t_v} \tag{III.2}$$

where t_v is the viscous boundary layer thickness,

$$t_v = \sqrt{\frac{\mu}{\rho_0 \pi f}} \tag{III.3}$$

and J_0 and J_2 are the Bessel functions of the first kind of order 0 and 2. As the Micro Cap simulator cannot handle Bessel functions, the first factor in eq. (III.1) must be expressed using other mathematical functions available in the simulator. The acoustic impedance is complex with a positive imaginary part, i.e. it corresponds to an RL series impedance. There are limiting values for both the resistance and the inductance in the extreme low and high frequency regions, and between these there are transition regions:

$$\begin{aligned} R_a\big|_{f\to 0} &= \frac{128\mu l}{\pi d^4} \\ R_a\big|_{f\to\infty} &= \frac{16l}{\pi d^3}\sqrt{\mu\rho_0\pi f} \end{aligned} \tag{III.4}$$

and

$$\begin{aligned} L_a\big|_{f\to 0} &= \frac{16\rho_0 l}{3\pi d^2} \\ L_a\big|_{f\to\infty} &= \frac{4\rho_0 l}{\pi d^2} \end{aligned} \tag{III.5}$$

It can also be observed that in many cases the transition region, in which the viscous boundary layer thickness and the hole diameter are comparable, will tend to occur inside the typical audio transmission frequency range of mobile phones. For this reason modelling of the transition is important. Polynomial approximations of the first factor in eq. (III.1) were found using optimization algorithms of the *APLAC* circuit simulator (developed by Helsinki University of Technology, Nokia Research Center and Rolf Nevanlinna Institute) together with Mathcad for Windows:

$$\operatorname{Re}\left(-j\frac{J_0(x)}{J_2(x)}\right) \approx \frac{1}{x^2}\sqrt[4]{4096+45x^3+4x^4} \tag{III.6}$$

and

$$\operatorname{Im}\left(-j\frac{J_0(x)}{J_2(x)}\right) \approx \frac{332-13x+7x^2+2x^3}{249-9x+4x^2+2x^3} \tag{III.7}$$

with

$$x = |k_c d| \tag{III.8}$$

and accurate to within about ±2 % (real part) and ±0.6 % (imaginary part), approaching correct values asymptotically in the low and high frequency limits.

Assuming now that there is an array of identical holes in parallel (with *n* as the number of holes), it is possible to write the final expressions for the acoustic resistance and inductance of the holes using eqs. (III.1)-(III.3) and (III.6)-(III.8):

$$R_a = \frac{8f\rho_0 l\sqrt[4]{4096+45x^3+4x^4}}{nd^2x^2} \tag{III.9}$$

and

$$L_a = \left(\frac{332-13x+7x^2+2x^3}{249-9x+4x^2+2x^3}\right)\frac{4\rho_0 l}{n\pi d^2} \tag{III.10}$$

where

$$x = \frac{d}{\sqrt{2}\,t_v} \tag{III.11}$$

Rectangular hole

The case of a rectangular hole is quite similar to the circular hole, except for the diameter being exchanged by the parameters *w* (width) and *h* (height). Again, it is assumed that $w,h,l \ll \lambda$. An expression for the impedance of an infinitely wide channel, consisting of two parallel planes, one above the other with a spacing *h*, is available in the literature [26, p. 2-19]:

$$AZ_a = -\frac{j\omega\rho_0 l}{1-\dfrac{\tan k_r h}{k_r h}} \tag{III.12}$$

Appendix IV Micro-Cap simulation macros

Constant and function definitions

```
; CONSTDEF.TXT: DEFINITIONS OF CONSTANTS
; (assuming a static pressure of 1 atm and a temperature of 20 C)

.DEFINE p0 101k;     Static pressure of 1 atmosphere
.DEFINE c0 343;      Sound propagation velocity
.DEFINE rho0 1.2;    Air density
.DEFINE mu 1.56E-5;  Coefficient of viscosity
.DEFINE Zcair 410;   Characteristic acoustic impedance of air
.DEFINE Pair 0.77;   Prandtl number of air

.DEFINE Kair (1/(rho0*c0**2)); Air compressibility

; REFERENCES:
; Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 10, 11, 22, 135, 137.
; Ingard, K. U. Notes on sound absorption technology. USA, 1994. Noise Control Foundation. p. 2-6.
```

```
; TUBEDEF.TXT: FUNCTIONS FOR TUBE AND AIR LOAD MODELS

; Polynomial approximations found by numerical optimization.

; Boundary layer thicknesses
.DEFINE tv (SQRT(mu/(rho0*pi*f))); Viscous boundary layer
.DEFINE th tv/SQRT(Pair);          Thermal boundary layer

; Functions for dimensions vs boundary layers
.DEFINE dr(x) (x/(2*tv))
.DEFINE dc(x) (x/(SQRT(2)*tv))

; Functions for circular tubes
.DEFINE RTrC(x) ((4096+45*x**3+4*x**4)**0.25/x**2);                 Resistance transition
.DEFINE LTrC(x) ((332-13*x+7*x**2+2*x**3)/(249-9*x+4*x**2+2*x**3)); Inductance transition
.DEFINE RCircHole(d,l,n) (8*f*rho0*(l)/((n)*(d)**2)*RTrC(dc(d)));   Resistance
.DEFINE LCircHole(d,l,n) (4*rho0*(l)/((n)*pi*(d)**2)*LTrC(dc(d)));  Inductance

; Functions for rectangular tubes
.DEFINE RTrR(x) (((405+23*x**3+5*x**4)/80)**0.25/x**2);                                  Resistance transition
.DEFINE LTrR(x) ((204+17*x+12*x**2+8*x**3)/(170+15*x+8*x**2+8*x**3));                    Inductance transition
.DEFINE RRectHole(w,h,l,n) (2*pi*rho0*(l)*f/((n)*(w)*(h))*(RTrR(dr(w))+RTrR(dr(h)))); Resistance
.DEFINE LRectHole(w,h,l,n) (rho0*(l)/((n)*(w)*(h))*LTrR(dr(w))*LTrR(dr(h)));          Inductance

; REFERENCES:
; Stinson, M. R., Shaw, E. A. G. Acoustic impedance of small, circular orifices in thin plates. J. Acoust. Soc. Am., vol.
; 77, no. 6, 1985.
; Ingard, K. U. Notes on sound absorption technology. USA, 1994. Noise Control Foundation. pp. 2-19- 2-27.
; Kreyszig, E. Advanced engineering mathematics. 1988. John Wiley & Sons, Inc. p. 208.
```

```
; DEF.MC5: USER DEFINITIONS

.DEFINE SPL(p) (DB((MAG(p)/SQRT(2))/20u)); Sound pressure level (amplitude as argument)
```

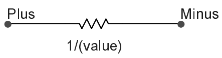
Shapes, electrical definitions and descriptions

Impedances

Figure IV.1: Ideal impedance components (top row: acoustical impedance, resistance, inductance and capacitance, bottom row: the same components in the mechanical domain). All of these have two pins, except the mechanical inductance (mass) with only one pin (the other pin is internally grounded).

Figure IV.2: Some of the above components with their pins shown (the mechanical inductance, not shown here, has its pin at the center of the filled circle).

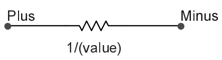
ZM.CIR: MECHANICAL IMPEDANCE



.PARAMETERS(value)

```
PARAMETER: value = value of mechanical impedance.

USAGE:
This macro implements a general mechanical impedance.

PRINCIPLE:
The reciprocal of the given impedance value is taken to provide the needed admittance value for the admittance analogy.
```

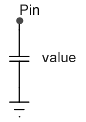
RM.CIR: MECHANICAL RESISTANCE



.PARAMETERS(value)

```
PARAMETER: value = value of mechanical resistance.

USAGE:
This macro implements a general mechanical resistance.

PRINCIPLE:
The reciprocal of the given resistance value is taken to provide the needed conductance value for the admittance analogy.
```

LM.CIR: MECHANICAL INDUCTANCE (MASS)



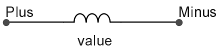
.PARAMETERS(value)

PARAMETER: value = value of mechanical inductance.

USAGE:
This macro implements a general mechanical inductance (mass).

PRINCIPLE:
A mechanical inductance appears as a capacitance in the admittance analogy. One terminal is grounded because the inertial frame (ground, zero velocity) is the reference.

CM.CIR: MECHANICAL CAPACITANCE (COMPLIANCE)



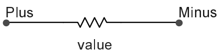
.PARAMETERS(value)

PARAMETER: value = value of mechanical capacitance.

USAGE:
This macro implements a general mechanical capacitance (compliance).

PRINCIPLE:
A mechanical capacitance appears as an inductance in the admittance analogy.

ZA.CIR: ACOUSTIC IMPEDANCE

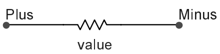


.PARAMETERS(value)

PARAMETER: value = value of acoustic impedance.

USAGE:
This macro implements a general acoustic impedance.

RA.CIR: ACOUSTIC RESISTANCE

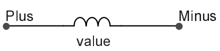


.PARAMETERS(value)

PARAMETER: value = value of acoustic resistance.

USAGE:
This macro implements a general acoustic resistance.

LA.CIR: ACOUSTIC INDUCTANCE (INERTANCE)

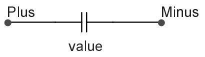


.PARAMETERS(value)

PARAMETER: value = value of acoustic inductance.

USAGE:
This macro implements a general acoustic inductance (inertance).

CA.CIR: ACOUSTIC CAPACITANCE



.PARAMETERS(value)

```
PARAMETER: value = value of acoustic capacitance.

USAGE:
This is a general acoustic capacitance, i.e. not specifically for a cavity, in which case one terminal is grounded.
```

Sources, transforming two-ports and grounds

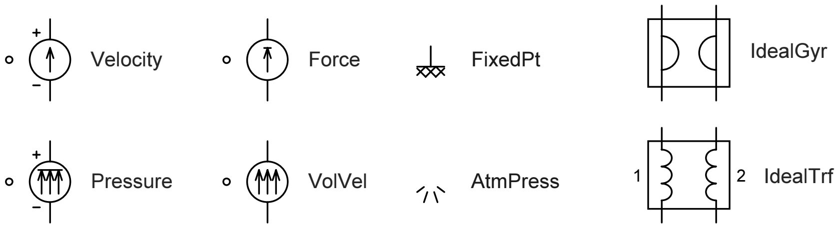


Figure IV.3: Velocity, force, pressure and volume velocity source, immovable point ("mechanical ground"), atmospheric pressure ("acoustic ground"), ideal gyrator and transformer.

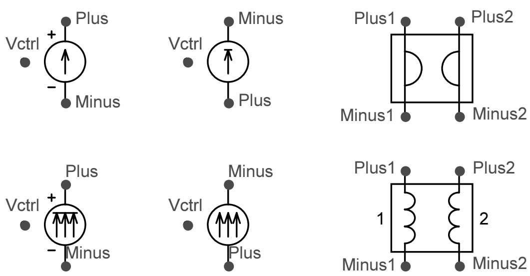
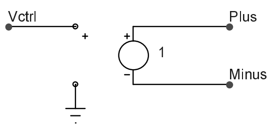


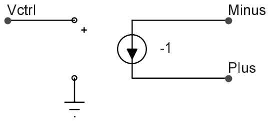
Figure IV.4: The above components with their pins shown (the two grounds have just one pin).

VELOCITY.CIR: VELOCITY SOURCE (VOLTAGE-CONTROLLED)




```
PARAMETERS: -

PINS:       Vctrl:       voltage control input,
            Plus, Minus: output of velocity.

USAGE:
This macro can be controlled by any available Micro Cap voltage source, or circuit voltage, to produce a velocity in the
mechanical domain. The velocity is output as a voltage in the mechanical domain (admittance analogy).
```

FORCE.CIR: FORCE SOURCE (VOLTAGE-CONTROLLED)

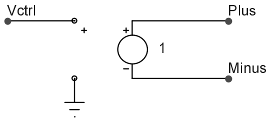


PARAMETERS: -

PINS: Vctrl: voltage control input,
Plus, Minus: output of force (a positive force is across the load from Minus to Plus).

USAGE:
This macro can be controlled by any available Micro Cap voltage source, or circuit voltage, to produce a force in the mechanical domain. The force is output as a current in the mechanical domain (admittance analogy).

PRESSURE.CIR: PRESSURE SOURCE (VOLTAGE-CONTROLLED)

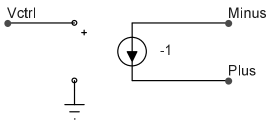


PARAMETERS: -

PINS: Vctrl: voltage control input,
Plus, Minus: output of pressure.

USAGE:
This macro can be controlled by any available Micro Cap voltage source, or circuit voltage, to produce a pressure in the acoustical domain. The pressure is output as a voltage in the acoustical domain (impedance analogy).

VOLVEL.CIR: VOLUME VELOCITY SOURCE (VOLTAGE-CONTROLLED)

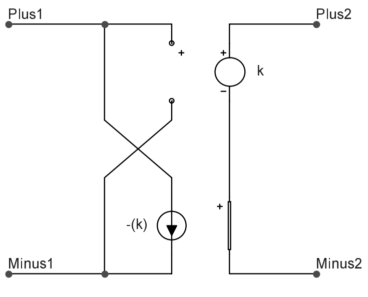


PARAMETERS: -

PINS: Vctrl: voltage control input,
Plus, Minus: output of volume velocity (a positive volume velocity is through the load from Minus to Plus).

USAGE:
This macro can be controlled by any available Micro Cap voltage source, or circuit voltage, to produce a volume velocity in the acoustical domain. The volume velocity is output as a current in the acoustical domain (impedance analogy).

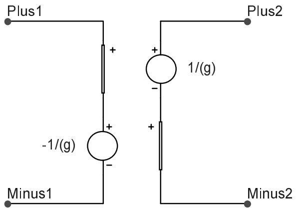
IDEALTRF.CIR: IDEAL TRANSFORMER



.PARAMETERS(k)

PARAMETER: k = voltage ratio ("turns ratio") secondary/primary.

IDEALGYR.CIR: GYRATOR



.PARAMETERS(g)

```
PARAMETER: g = transconductance.
```

Physical structures

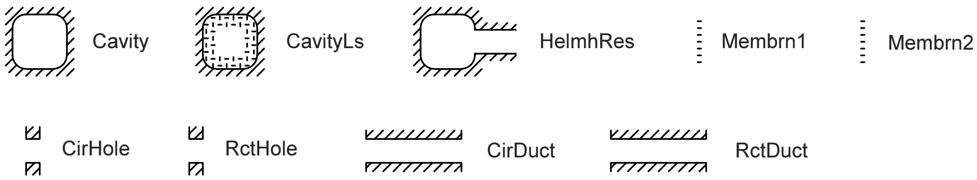


Figure IV.5: Loss-free and damped air cavity, Helmholtz resonator, membranes, circular and rectangular hole, circular and rectangular duct.

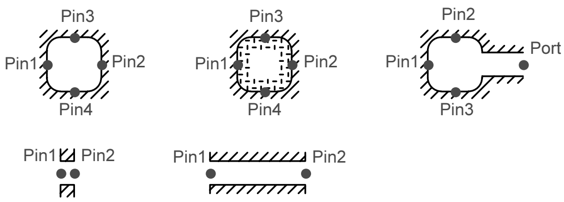
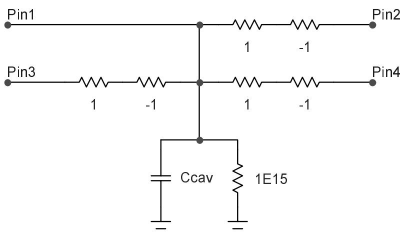


Figure IV.6: The above components with their pins shown. The 4 pins of the cavity components are in fact one and the same - 4 separate pins is just a feature to make schematics look better in the common case of many entrances to one single cavity. The same goes for the 3 leftmost pins of the Helmholtz resonator.

CAVITY.CIR: AIR CAVITY (<< WAVELENGTH)



.PARAMETERS(vol)
.INCLUDE CONSTDEF.TXT
.DEFINE Ccav Kair*(vol)

```
PARAMETER: vol = cavity air volume.
PINS:      Pin1...Pin4: entrance(s) to cavity.
```

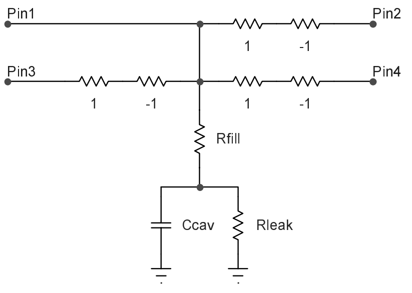
USAGE:
This macro can be used to model a lossless cavity whose dimensions are considerably smaller than the wavelength.

LIMITATIONS:
No losses are included. If the macro is used to model a cavity not smaller than the wavelength, accuracy is lost because reflections and standing waves inside the cavity are neglected. Adiabatic conditions are assumed (no appreciable heat diffusion, i.e. thermal boundary layer thickness considerably smaller than cavity dimensions).

PRINCIPLE:
The compression of air in the cavity appears as an acoustic capacitance. One pin is grounded because static atmospheric pressure is the reference. The opposite resistor pairs are used merely to prevent an error resulting from shorting different pins together. The 4 pins (instead of 1 single pin) are just a convenient feature when drawing schematics.

REFERENCE:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. p. 129.

CAVITYLS.CIR: AIR CAVITY (<< WAVELENGTH) WITH LOSSES



.PARAMETERS(vol, volcf, Tfill, Tleak)
.INCLUDE CONSTDEF.TXT
.DEFINE Ccav0 (Kair*(vol))
.DEFINE Ccav (1+0.4*(volcf))*Ccav0
.DEFINE Rfill (Tfill)/Ccav0
.DEFINE Rleak (Tleak)/Ccav0

```
PARAMETERS: vol   = cavity air volume,
            volcf = volume increase factor (0<=volcf<=1),
            Tfill = time constant considering filling only,
            Tleak = time constant considering leaks only.

PINS:       Pin1...Pin4: entrance(s) to cavity.
```

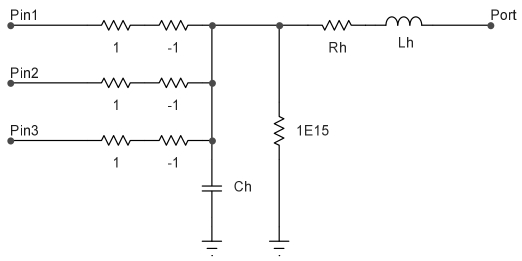
USAGE:
This macro can be used to model a lossy cavity (e.g. leaking, filled with damping material, or both) whose dimensions are considerably smaller than the wavelength. The volume increase factor (volcf) is normalized: volcf=0 means no apparent volume increase, volcf=1 means maximum volume increase (about 40%, corresponding to a fully isothermal process, in practical cases volcf<1). The two time constants must be supplied as estimates characterizing the losses. If losses are due to leaks only (no filling in cavity), then volcf=0, Tfill is negligible (zero), and Tleak depends on the size of the leaks. Bigger leaks mean a smaller Tleak, and vice versa. If there are no leaks and the cavity is filled with damping material, then Tleak is very large (infinite), while Tfill and volcf depend on the properties and placement of the filling. Tfill depends also on the areas of the entrances to the cavity. More filling and/or smaller entrances lead to a larger Tfill, especially if the filling obstructs air flow. In addition, more filling leads to a larger volcf (up to some limit determined by filling location and material properties). Also, volcf rises at very low frequencies.

LIMITATIONS:
If the macro is used to model a cavity not smaller than the wavelength, accuracy is lost because possible reflections and standing waves inside the cavity are neglected. There is no simple way of calculating the volcf, Tfill and Tleak parameters for any general case. Instead, these parameters are better determined by experiments. Tfill and Tleak cannot be given zero (or infinite) values, but very small or very large values can be used instead. The model assumes that the leaks and filling can be approximated by purely resistive components, which is not true in all cases.

PRINCIPLE:
It is assumed that the losses in the cavity can be characterized accurately enough using two resistances, one in series and one in parallel with the acoustic capacitance formed by the cavity. The apparent increase in volume, due to thermodynamic effects of the filling, is modelled as a corresponding increase in the acoustic capacitance value. The series resistance models the losses due to the filling, and the parallel resistance represents the leaks. For more details, see the description of the Cavity macro.

REFERENCES:
Leach, W. M. Electroacoustic-analogous circuit models for filled enclosures. J. Audio Eng. Soc., vol. 37, no. 7/8, 1989.
Bradbury, L. J. S. The use of fibrous materials in loudspeaker enclosures. J. Audio Eng. Soc., vol. 24, no. 4, 1976.
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 217-220.

HELMHRES.CIR: HELMHOLTZ RESONATOR




```
.PARAMETERS(vol, fres, Q)
.INCLUDE CONSTDEF.TXT
.DEFINE Ch  Kair*(vol)
.DEFINE Lh  1/(4*pi**2*(fres)**2*Ch)
.DEFINE Rh  2*pi*(fres)*Lh/(Q)
```

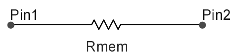
```
PARAMETERS: vol  = volume of cavity,
            fres = resonance frequency,
            Q    = Q value (quality factor).

PINS:       Pin1...Pin3: entrances to the resonator cavity,
            Port:        resonator port.
```

USAGE:
This macro is usable when modelling a Helmholtz resonator with given resonance frequency and quality factor. The volume needs to be given (to determine unique component values). The specified resonance frequency and quality factor describe the properties of the resonator with all its entrances (except the port) blocked.

LIMITATIONS:
It is assumed that all resonator dimensions are considerably smaller than the wavelength.

PRINCIPLE:
Lumped components are used to model the cavity and the port. The opposite resistor pairs are used merely to prevent an error that would result from shorting different pins together. The 3 pins (instead of 1 single pin) are just a convenient feature when drawing schematics.

MEMBRN1.CIR: RIGID PERMEABLE MEMBRANE(S)




```
.PARAMETERS(Rspec, area, no)
.DEFINE Rmem  (Rspec)/((area)*(no))
```

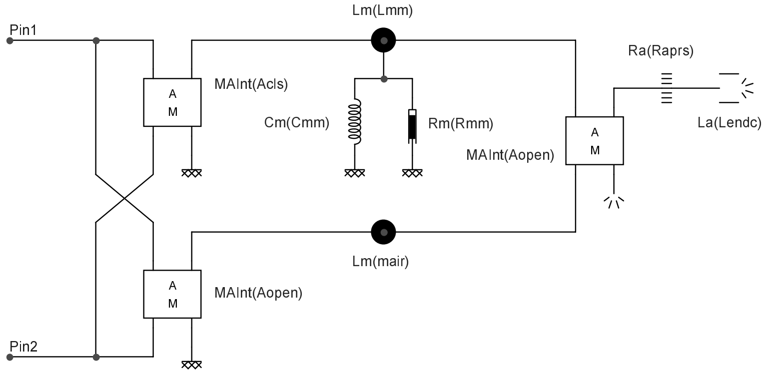
```
PARAMETERS: Rspec = flow resistance per unit area,
            area  = area of membrane(s),
            no    = number of membranes.
```

USAGE:
This macro models a permeable membrane (or an array of similar membranes in parallel). A typical example is a dust and water shield membrane to protect sound openings. The specific flow resistance of the membrane is passed as a parameter.

LIMITATIONS:
The model assumes that the membrane is completely rigid. Any acoustic inductance associated with the membrane is neglected.

PRINCIPLE:
The permeable membrane appears as a single resistance corresponding to the acoustic resistance of the membrane material.

MEMBRN2.CIR: PERMEABLE MEMBRANE(S)



.PARAMETERS(Rspec, Rmmem, Lmmem, Cmmem, area, prcopen, thick, endc, no)
.INCLUDE CONSTDEF.TXT
.DEFINE Acls (area)*(1-(prcopen)/100)*(no)
.DEFINE Aopen ((area)*(prcopen)/100*(no))
.DEFINE Rmm (Rmmem)/(no)
.DEFINE Lmm (Lmmem)/(no)
.DEFINE Cmm (Cmmem)*(no)
.DEFINE mair Aopen*(thick)*rho0
.DEFINE Raprs Rspec/((area)*(no))
.DEFINE Lendc rho0*(endc)/Aopen

```
PARAMETERS: Rspec   = specific acoustic resistance,
            Rmmem   = mechanical resistance of assembled membrane(s),
            Lmmem   = moving mass of assembled membrane(s),
            Cmmem   = compliance of assembled membrane(s),
            area    = area (open + closed),
            prcopen = percent open area,
            thick   = thickness (length of pores in flow direction),
            endc    = end correction (length in flow direction),
            no      = number of parallel membrane(s).
```

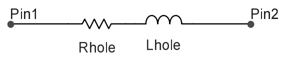
USAGE:
This macro models a permeable membrane (or an array of similar membranes in parallel). A typical example is a dust and water shield membrane to protect sound openings. Known material parameters are passed to the macro: the specific acoustic resistance has the unit of resistance times area, thick and prcopen should together define the total volume of air contained in the pores. The end correction is given as a length (in the flow direction), and it applies to each pore. Rmmem, Lmmem, Cmmem and area describe the properties of the membrane when mounted in the assembly.

LIMITATIONS:
The model assumes that the membrane is fully flexible. This is typically not the case, a nonlinear compliance would be more realistic. Any frequency dependence of the acoustic impedance of the pores is neglected.

PRINCIPLE:
The permeable membrane appears as a parallel connection of the solid parts of the membrane and the air in the pores. The relative difference in velocity between these two equals the velocity of the air in relation to the pore walls, and this relative movement is affected by the acoustic resistance (and an added acoustic inductance due to the end correction) of the pores.

REFERENCE:
Poldy, C. A. Electrical analogs for membranes with application to earphones. J. Audio Eng. Soc., vol. 31, no. 11, 1983.

CIRHOLE.CIR: CIRCULAR HOLE(S) (<< WAVELENGTH)



.PARAMETERS(diam, length, no)
.INCLUDE CONSTDEF.TXT
.INCLUDE TUBEDEF.TXT
.DEFINE Rhole RCircHole(diam,length,no)
.DEFINE Lhole LCircHole(diam,length,no)

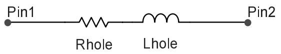
```
PARAMETERS: diam   = diameter of hole(s),
            length = length in flow direction (excl. end corrections),
            no     = number of parallel holes.
```

USAGE:
This macro models the acoustic impedance of a circular hole (or tube) (or several holes/tubes in parallel). Only the internal part is modelled (no end corrections are added automatically).

LIMITATIONS:
The model is applicable to holes or tubes with circular (or almost circular) cross-section, with a length in the flow direction that is considerably smaller than the wavelength.

PRINCIPLE:
The impedance is calculated directly, calling predefined impedance formulae that include the frequency dependence.

RCTHOLE.CIR: RECTANGULAR HOLE(S) (<< WAVELENGTH)




```
.PARAMETERS(width, height, length, no)
.INCLUDE CONSTDEF.TXT
.INCLUDE TUBEDEF.TXT
.DEFINE Rhole  RRectHole(width,height,length,no)
.DEFINE Lhole  LRectHole(width,height,length,no)
```

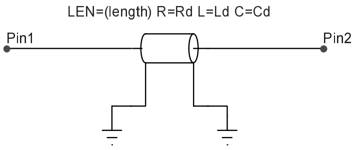
```
PARAMETERS: width  = width of hole(s),
            height = height of hole(s) (>, = or < width),
            length = length in flow direction (excl. end corrections),
            no     = number of parallel holes.
```

USAGE:
This macro models the acoustic impedance of a rectangular hole (or tube) (or several holes/tubes in parallel). Only the internal part is modelled (no end corrections are added automatically). A narrow slit can also be modelled using this same macro.

LIMITATIONS:
The length in the flow direction must be considerably smaller than the wavelength. Even if some other dimension is greater than the wavelength and oblique propagation modes could appear, these modes are not modelled by the macro.

PRINCIPLE:
The impedance is calculated directly, calling predefined impedance formulae that include the frequency dependence.

CIRDUCT.CIR: CIRCULAR DUCT(S)




```
.PARAMETERS(diam, length, no, freq, compcf)
.INCLUDE CONSTDEF.TXT
.DEFINE kRd  (16/11)**(compcf)                              = shunt resistance compensation
.DEFINE Rd  16*SQRT(mu*rho0*pi*(freq))/(pi*(diam)**3*(no))*kRd   = distributed resistance
.DEFINE Ld  4*rho0/(pi*(diam)**2*(no))                       = distributed inductance
.DEFINE Cd  Kair*pi*(diam)**2*(no)/4                          = distributed capacitance
```

```
PARAMETERS: diam   = diameter of duct(s),
            length = length of duct(s),
            no     = number of ducts,
            freq   = desired frequency of best accuracy,
            compcf = shunt resistance compensation coefficient (0<=compcf<=1).
```

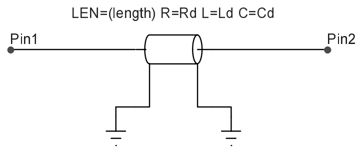
USAGE:
This macro is an approximate high-frequency model of a circular duct (or several such ducts in parallel) usable in a limi-
ted frequency range. The length in the flow direction need not be smaller than the wavelength. Waves propagating in the duct are assumed to be plane. Only the internal part is modelled (no end corrections are added automatically). The freq parameter should be given a value that lies somewhere in the middle of the simulation frequency range. compcf=0 gives the most accurate viscous resistance, but the damping (real part of propagation coefficient) will be 31% too low. compcf=1 gives the most accurate damping coefficient, but then the viscous resistance will be 46% too high. A value of compcf=0.5 is a useful compromise in many cases (damping 18% too low, viscous resistance 20% too high).

LIMITATIONS:
Frequency dependence of the distributed acoustic impedances is not modelled. Instead, the desired frequency of best accuracy is supplied as a parameter, and results are accurate in a limited range around this frequency. The macro is usable at high frequencies only, when the viscous boundary layer is significantly smaller than the duct diameter (the CirHole macro can be used instead at lower frequencies, provided that the length is considerably smaller than the wavelength). Losses due to thermal diffusion are not modelled, and as a result the viscous resistance and the damping cannot both be accurately modelled at the same time. Only one (axial) propagation mode is modelled.

PRINCIPLE:
A transmission line is used to model the duct. Micro-Cap supports three distributed impedances at most, thus the shunt resistance (due to thermal diffusion) must be left out. An optional compensating change is made to the series resistance instead.

REFERENCES:
Ingard, K. U. Notes on sound absorption technology. USA, 1994. Noise Control Foundation. pp. 2-19- 2-27, 7-1- 7-3.
Nolle, A. W. Small-signal impedance of short tubes. J. Acoust. Soc. Am., vol. 25, no. 1, 1953.

RCTDUCT.CIR: RECTANGULAR DUCT(S)




```
.PARAMETERS(width, height, length, no, freq, compcf)
.INCLUDE CONSTDEF.TXT
.DEFINE kRd  (16/11)**(compcf)                                                      = shunt res. compensation
.DEFINE Rd  2*((width)+(height))*SQRT(mu*rho0*pi*(freq))/(((width)*(height))**2*(no))*kRd   = distributed resistance
.DEFINE Ld  rho0/((width)*(height)*(no))                                            = distributed inductance
.DEFINE Cd  Kair*(width)*(height)*(no)                                              = distributed capacitance
```

```
PARAMETERS: width  = width of duct(s),
            height = height of duct(s),
            length = length of duct(s),
            no     = number of ducts,
            freq   = desired frequency of best accuracy,
            compcf = shunt resistance compensation coefficient (0<=compcf<=1).

USAGE:
This macro is an approximate high-frequency model of a rectangular duct (or several such ducts in parallel) usable in a
limited frequency range. The length in the flow direction need not be smaller than the wavelength. Waves propagating in
the duct are assumed to be plane. Only the internal part is modelled (no end corrections are added automatically). The
freq parameter should be given a value that lies somewhere in the middle of the simulation frequency range. compcf=0 gi-
ves the most accurate viscous resistance, but the damping (real part of propagation coefficient) will be 31% too low.
compcf=1 gives the most accurate damping coefficient, but then the viscous resistance will be 46% too high. A value of
compcf=0.5 is a useful compromise in many cases (damping 18% too low, viscous resistance 20% too high).

LIMITATIONS:
Frequency dependence of the distributed acoustic impedances is not modelled. Instead, the desired frequency of best accu-
racy is supplied as a parameter, and results are accurate in a limited range around this frequency. The macro is usable
at high frequencies only, when the viscous boundary layer is significantly smaller than the duct diameter (the CirHole
macro can be used instead at lower frequencies, provided that the length is considerably smaller than the wavelength).
Losses due to thermal diffusion are not modelled, and as a result the viscous resistance and the damping cannot both be
accurately modelled at the same time. Only one (axial) propagation mode is modelled.

PRINCIPLE:
A transmission line is used to model the duct. Micro-Cap supports three distributed impedances at most, thus the shunt
resistance (due to thermal diffusion) must be left out. An optional compensating change is made to the series resistance
instead.

REFERENCES:
Ingard, K. U. Notes on sound absorption technology. USA, 1994. Noise Control Foundation. pp. 2-19- 2-27, 7-1- 7-3.
Nolle, A. W. Small-signal impedance of short tubes. J. Acoust. Soc. Am., vol. 25, no. 1, 1953.
```

Air load

Figure IV.7: General end correction for circular and rectangular radiators (top row), air load on circular and rectangular radiators, any solid radiation angle, or equipped with finite baffle (middle row), radiation resistance of radiator radiating spherical or plane waves (bottom row).

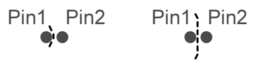
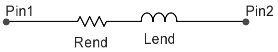


Figure IV.8: The above components with their pins shown (all the air load macros modelling spherical waves (wavelength >> dimensions) have the same shape and pin positioning).

ENDCCIR.CIR: GENERAL END CORRECTION FOR CIRCULAR RADIATOR(S)




```
.PARAMETERS(diam, Rcorr, Lcorr, no)
.INCLUDE CONSTDEF.TXT
.DEFINE Rend  128*mu*(Rcorr)/(pi*(diam)**3*(no))
.DEFINE Lend  16*rho0*(Lcorr)/(3*pi*(diam)*(no))
```

```
PARAMETERS: diam  = diameter of radiator(s),
            Rcorr = end correction (in proportion to diameter) for resistance,
            Lcorr = end correction (in proportion to diameter) for inductance,
            no    = number of radiators.
```

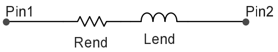
USAGE:
This macro is a general form of end correction macro, allowing the user to select any suitable end correction for the resistive and inductive parts separately. The macro can be used whenever end corrections need to be modelled, in which case it is connected in series with the hole or duct opening(s) (or radiating surface(s)). As such, the macro is meant primarily to model attached air mass and viscous losses (not radiation resistance, which is strongly dependent on frequency). End corrections are given as factors, in proportion to the diameter of the radiator(s).

LIMITATIONS:
No separate term for the radiation resistance of a small source is included, and mutual interaction is not automatically taken into account. Any frequency dependence is neglected (unless Rcorr or Lcorr are given frequency-dependent values).

PRINCIPLE:
The actual impedances are calculated using the low-frequency impedance formula for a short circular tube. The length of this tube is the given end correction (separate lengths given for resistance and inductance).

REFERENCES:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. p. 135.

ENDCRCT.CIR: GENERAL END CORRECTION FOR RECTANGULAR RADIATOR(S)




```
.PARAMETERS(width, height, Rcorr, Lcorr, no)
.INCLUDE CONSTDEF.TXT
.DEFINE w  MAX(width,height)
.DEFINE h  MIN(width,height)
.DEFINE Rend  12*mu*(Rcorr)/(w*h**2*(no))
.DEFINE Lend  6*rho0*(Lcorr)/(5*w*(no))
```

```
PARAMETERS: width  = width of radiator(s),
            heigth = height of radiator(s),
            Rcorr  = end correction (in proportion to smaller dimension) for resistance,
            Lcorr  = end correction (in proportion to smaller dimension) for inductance,
            no     = number of radiators.
```

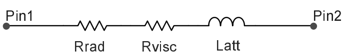
USAGE:
This macro is a general form of end correction macro, allowing the user to select any suitable end correction for the resistive and inductive parts separately. The macro can be used whenever end corrections need to be modelled, in which case it is connected in series with the hole or duct opening(s) (or radiating surface(s)). As such, the macro is meant primarily to model attached air mass and viscous losses (not radiation resistance, which is strongly dependent on frequency). End corrections are given as factors, in proportion to the smaller dimension of the radiator(s).

LIMITATIONS:
No separate term for the radiation resistance of a small source is included, and mutual interaction is not automatically taken into account. Any frequency dependence is neglected (unless Rcorr or Lcorr are given frequency-dependent values).

PRINCIPLE:
The actual impedances are calculated using the low-frequency impedance formula for a short rectangular tube (assuming no viscous drag at the two more widely separated walls). The length of this tube is the given end correction (separate lengths given for resistance and inductance).

REFERENCE:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 135-136.

AIRLCIR.CIR: AIR LOAD ON CIRCULAR RADIATOR(S) (<< WAVELENGTH)




```
.PARAMETERS(diam, no, spacecf, mutualcf)
.INCLUDE CONSTDEF.TXT
.INCLUDE TUBEDEF.TXT
.DEFINE Rrad  pi*rho0*f**2/((spacecf)*c0)                         = resistance due to pure radiation
.DEFINE lxr  0.42*(diam)/SQRT(2*(spacecf))                        = resistance end correction (viscosity)
.DEFINE lxl  (0.42+0.126*(mutualcf)*((no)-1))*(diam)/SQRT(2*(spacecf))   = inductance end correction (attached air)
.DEFINE Rvisc  RCircHole(diam,lxr,no)
.DEFINE Latt  LCircHole(diam,lxl,no)
```

```
PARAMETERS: diam     = diameter of radiator(s),
            no       = number of radiators,
            spacecf  = portion of full space that the radiation is into (0 < spacecf <= 1, 1 = full space, 0.5 = half
                       space, etc.),
```

mutualcf = mutual coupling coefficient describing average strength of mutual coupling (0 <= mutualcf <= 1).

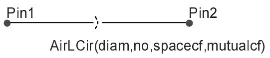
USAGE:
This macro models the air load on one single circular radiator, or an array of similar radiators lying in the same plane. The macro should be used whenever air load effects need to be taken into account, in which case it is connected in series with the opening or vibrating surface that radiates sound. The radiator (or radiator array) can radiate into any solid angle (as specified by spacecf). For most cases with radiation into free air, spacecf=1 is appropriate (unless there is a large baffle). The mutual coupling coefficient (mutualcf) can be set to zero if the strength of the mutual coupling is not known. mutualcf describes the average coupling (averaged over all pairs of radiators), and it is normalized so that mutualcf=1 means the strongest possible interaction (which appears only when there are 2 or 3 radiators touching each other). In all other possible cases, mutualcf<1 and decreases with increasing distances between radiators. If good accuracy is wanted, mutualcf must be calculated separately from the geometry of the radiator array. If the volume velocity output by the radiator is not needed any further, the macro should be followed by a ground connection (i.e. AtmPress). If the sound pressure at some distance from the radiator is wanted, the macro should be followed by a suitable radiation macro. If radiation is into a narrow tube (not spherical waves), other air load macros should be used.

LIMITATIONS:
Viscous losses are modelled assuming a radiating opening, i.e. not a piston. The overall dimensions of the radiator or radiator array must be smaller than the wavelength in order to maintain accuracy. The model treats spacecf as describing an infinite space, e.g. ideal full space, an infinite baffle (spacecf=0.5) or an infinite conical horn (spacecf<0.5).

PRINCIPLE:
The macro works by assembling the total air load using three parts: the radiation resistance of a spherical radiator, the attached air mass, and the viscous losses, forming together a RL impedance. The attached air mass and the viscous losses are modelled by calling the predefined formula for the impedance of a short circular tube. This tube is the end correction of the radiator, and end correction lengths are calculated separately for the resistive and inductive terms. Although one terminal of the radiation impedance is normally grounded in an equivalent circuit (since nothing follows it), both terminals are connectable here to allow the resulting volume velocity to be used directly as input to radiation macros.

REFERENCES:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 116-128.
Kinsler, L. E., Frey, A. R. Fundamentals of acoustics. Monterey, Calif., USA, 1982. John Wiley & Sons, Inc. pp. 191-193.
Stinson, M. R., Shaw, E. A. G. Acoustic impedance of small, circular orifices in thin plates. J. Acoust. Soc. Am., vol. 77, no. 6, 1985.
Jacobsen, O. Some aspects of the self and mutual radiation impedance concept with respect to loudspeakers. J. Audio Eng. Soc., vol. 24, no. 3, 1976.
Olson, H. F. Acoustical engineering (1991 edition, orig. D. van Nostrand Co. 1957). Philadelphia, PA, USA, 1991. Professional Audio Journals Inc. pp. 102-103.

AIRLCIRB.CIR: AIR LOAD ON CIRCULAR RADIATOR(S) (<< WAVELENGTH) SURROUNDED BY FINITE BAFFLE




```
.PARAMETERS(diam, no, bperim, mutualcf)
.INCLUDE CONSTDEF.TXT
.DEFINE fc  (2.7*c0/(bperim))                  = transition frequency (halfway between full- and half-space radiation)
.DEFINE spacecf  (1+0.5*(f/fc)**4)/(1+(f/fc)**4)   = transition from spacecf=1 to spacecf=0.5
```

```
PARAMETERS: diam     = diameter of radiator(s),
            no       = number of radiators,
            bperim   = perimeter of baffle,
            mutualcf = mutual coupling coefficient describing average strength of mutual coupling (0 <= mutualcf <= 1).
```

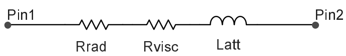
USAGE:
This macro is similar to the AirLCir macro, except that it models one or more radiators surrounded by a finite baffle, whose perimeter is specified. For more details, see the description of the AirLCir macro.

LIMITATIONS:
In modelling the baffle, accuracy is reasonable in vicinity of the transition frequency from full- to half-space radiation, as long as the baffle shape does not depart too much from a circle or square (i.e. the aspect ratio must not be large). Also, the radiator(s) must be located close enough to the center of the baffle. For other limitations, see the description of the AirLCir macro.

PRINCIPLE:
The transition from full- to half-space radiation (as frequency increases) is modelled here using a frequency-dependent value for spacecf, which is passed to the AirLCir macro. The function describing this change was obtained by manually fitting its graph to a curve derived from the frequency dependence of the radiation impedance of a circular piston in the end of a long tube.

REFERENCE:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 116-128.

AIRLRCT.CIR: AIR LOAD ON RECTANGULAR RADIATOR(S) (<< WAVELENGTH)




```
.PARAMETERS(width, height, no, spacecf, mutualcf)
.INCLUDE CONSTDEF.TXT
.INCLUDE TUBEDEF.TXT
.DEFINE h  MIN(width,height)                                          = smaller dimension
.DEFINE w  MAX(width,height)                                          = larger dimension
.DEFINE Rrad  pi*rho0*f**2/((spacecf)*c0)                             = resistance due to pure radiation
.DEFINE lxr  0.46*h/SQRT(2*(spacecf))                                 = resistance end correction (viscosity)
.DEFINE lxl  (1+(mutualcf)*((no)-1))*(0.305*LN(w/h)+0.42)*h/SQRT(2*(spacecf))   = inductance end correction (attached air)
.DEFINE Rvisc  RRectHole(w,h,lxr,no)
.DEFINE Latt  LRectHole(w,h,lxl,no)
```

```
PARAMETERS: width    = width of radiator(s),
            height   = height of radiator(s) (>, = or < width),
            no       = number of radiators,
```

```
           spacecf  = portion of full space that the radiation is into (0 < spacecf <= 1, 1 = full space, 0.5 = half
                      space, etc.),
           mutualcf = mutual coupling coefficient describing average strength of mutual coupling (0 <= mutualcf < 1).
```

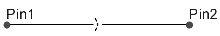
USAGE:
This macro models the air load on one single rectangular radiator, or an array of similar radiators lying in the same plane. The macro should be used whenever air load effects need to be taken into account, in which case it is connected in series with the opening or vibrating surface that radiates sound. The radiator (or radiator array) can radiate into any solid angle (as specified by spacecf). For most cases with radiation into free air, spacecf=1 is appropriate (unless there is a large baffle). The mutual coupling coefficient (mutualcf) can be set to zero if the strength of the mutual coupling is not known. mutualcf describes the average coupling (averaged over all pairs of radiators), and it is normalized so that mutualcf=1 means the strongest possible interaction (which appears, theoretically, only when there are 2 radiators with infinite aspect ratio touching each other along the longer sides). Thus, in all practical cases, mutualcf <1 and decreases with increasing distances between radiators. If good accuracy is wanted, mutualcf must be calculated separately from the geometry of the radiator array. If the volume velocity output by the radiator is not needed any further, the macro should be followed by a ground connection (i.e. AtmPress). If the sound pressure at some distance from the radiator is wanted, the macro should be followed by a suitable radiation macro. If radiation is into a narrow tube (not spherical waves), other macros should be used.

LIMITATIONS:
Viscous losses are modelled assuming an end correction (for the resistance) equal to the inductance end correction for a radiator with 1:1 aspect ratio. The accuracy of this assumption has not been verified. The overall dimensions of the radiator or radiator array must be clearly smaller than the wavelength in order to maintain accuracy. The model treats spacecf as describing an infinite space, e.g. ideal full space, an infinite baffle (spacecf=0.5) or an infinite conical horn (spacecf<0.5).

PRINCIPLE:
The macro works by assembling the total air load using three parts: the radiation resistance of a spherical radiator (this approximation is valid since by definition the dimensions are considerably smaller than the wavelength), the attached air mass, and the viscous losses, forming together a RL impedance. The attached air mass and the viscous losses are modelled by calling the predefined formula for the impedance of a short rectangular tube. This tube is the end correction of the radiator, and end correction lengths are calculated separately for the resistive and inductive terms. Although one terminal of the radiation impedance is normally grounded in an equivalent circuit (since nothing follows it), both terminals are connectable here to allow the resulting volume velocity to be used directly as input to radiation macros.

REFERENCES:
Kinsler, L. E., Frey, A. R. Fundamentals of acoustics. Monterey, Calif., USA, 1982. John Wiley & Sons, Inc. pp. 192-193.
Stinson, M. R., Shaw, E. A. G. Acoustic impedance of small, circular orifices in thin plates. J. Acoust. Soc. Am., vol. 77, no. 6, 1985.
Olson, H. F. Acoustical engineering (1991 edition, orig. D. van Nostrand Co. 1957). Philadelphia, PA, USA, 1991. Professional Audio Journals Inc. pp. 102-103.
Bank, G., Wright, J. R. Radiation impedance calculations for a rectangular piston. J. Audio Eng. Soc., vol. 38, no. 5, 1990.

AIRLRCTB.CIR: AIR LOAD ON RECTANGULAR RADIATOR(S) (<< WAVELENGTH) SURROUNDED BY FINITE BAFFLE



AirLRct(width,height,no,spacecf,mutualcf)

```
.PARAMETERS(width, height, no, bperim, mutualcf)
.INCLUDE CONSTDEF.TXT
.DEFINE fc  (2.7*c0/(bperim))                   = transition frequency (halfway between full- and half-space radiation)
.DEFINE spacecf  (1+0.5*(f/fc)**4)/(1+(f/fc)**4)  = transition from spacecf=1 to spacecf=0.5
```

```
PARAMETERS: width    = width of radiator(s),
            height   = height of radiator(s) (>, = or < width),
            no       = number of radiators,
            bperim   = perimeter of baffle,
            mutualcf = mutual coupling coefficient describing average strength of mutual coupling (0 <= mutualcf < 1).
```

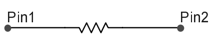
USAGE:
This macro is similar to the AirLRct macro, except that it models one or more radiators surrounded by a finite baffle, whose perimeter is specified. For more details, see the description of the AirLRct macro.

LIMITATIONS:
In modelling the baffle, accuracy is reasonable in vicinity of the transition frequency from full- to half-space radiation, as long as the baffle shape does not depart too much from a circle or square (i.e. the aspect ratio must not be large). Also, the radiator(s) must be located close enough to the center of the baffle. For other limitations, see the description of the AirLRct macro.

PRINCIPLE:
The transition from full- to half-space radiation (as frequency increases) is modelled here using a frequency-dependent value for spacecf, which is passed to the AirLRct macro. The function describing this change was obtained by manually fitting its graph to a curve derived from the frequency dependence of the radiation impedance of a circular piston in the end of a long tube.

REFERENCE:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 116-128.

SPHRADR.CIR: RADIATION RESISTANCE (SPHERICAL WAVE)



Rrad

```
.PARAMETERS(no, spacecf)
.INCLUDE CONSTDEF.TXT
.DEFINE Rrad  pi*rho0*f**2/((spacecf)*c0)
```

```
PARAMETERS: no      = number of radiators,
            spacecf = portion of full space that the radiation is into (0 < spacecf <= 1).
```

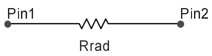
USAGE:
This macro models the pure radiation resistance of a radiator (or array of similar radiators) radiating spherical waves (i.e. only the resistance representing radiated sound power, excluding reactances and possible viscous losses). As such,

the macro can be seen as a subset of the AirLCir and AirLRct macros. To form a more general air load model than can be built using the AirLCir and AirLRct macros, use instead this macro in series with an EndCCir or EndCRct macro (with any desired frequency dependence in the parameters).

LIMITATIONS:
The model assumes that the dimensions of the radiator(s) are considerably smaller than the wavelength (or, if this is not the case, that the radiator surface is spherical). The model treats spacecf as describing an infinite space, e.g. ideal full space, an infinite baffle (spacecf=0.5) or an infinite conical horn (spacecf<0.5).

REFERENCE:
Kinsler, L. E., Frey, A. R. Fundamentals of acoustics. Monterey, Calif., USA, 1982. John Wiley & Sons, Inc. pp. 191-193.

PLNRADR.CIR: RADIATION RESISTANCE (PLANE WAVE)




```
.PARAMETERS(area, no)
.INCLUDE CONSTDEF.TXT
.DEFINE Rrad  rho0*c0/((area)*(no))
```

```
PARAMETERS: area = area of each wave front (and radiator),
            no   = number of wave fronts (and radiators),
```

USAGE:
This macro models the pure radiation impedance of a radiator (or array of similar radiators) radiating plane waves (i.e. only the actual radiation impedance, excluding possible viscous losses). As such, the macro can be seen as a high-frequency limiting case of the AirLRct and AirLCir macros (again, excluding possible viscous losses). It should be used instead of the AirLCir, AirLRct or SphRadR macros when radiation is into a space where only plane waves propagate, or when one or more radiators with plane surfaces radiate sound whose wavelength is considerably smaller than the dimensions of the radiator(s).

LIMITATIONS:
Possible reflections altering the load impedance are not taken into account, i.e. the macro assumes that the space that sound is radiated into is infinite (or terminated with a matching impedance). Any possible oblique propagation modes are also not modelled.

REFERENCE:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. p. 35.

Domain interfaces

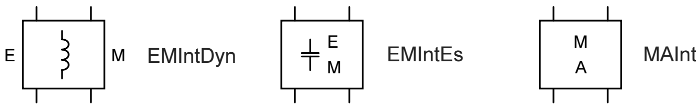


Figure IV.9: Electrical-mechanical domain interface for dynamic transduction, electrical-mechanical interface for electrostatic transduction, and mechanical-acoustical domain interface. Note: interfaces between acoustical domain and radiation are treated separately below.

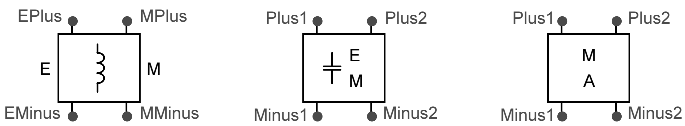
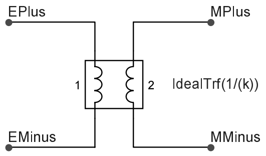


Figure IV.10: The above components with their pins shown.

EMINTDYN.CIR: ELECTRICAL-MECHANICAL DOMAIN
INTERFACE FOR DYNAMIC TRANSDUCER



.PARAMETERS(k)

PARAMETER: k = force factor (force/current).

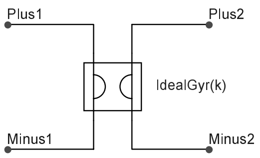
PINS: EPlus, EMinus: electrical domain pins,
MPlus, MMinus: mechanical domain pins.

USAGE:
This macro is an electrical-mechanical domain interface, modelling the kind of transduction appearing in dynamic transducers (output force proportional to input current, output voltage proportional to input velocity). The interface macro is ideal and does not include any other characteristics such as e.g. coil impedances (these must be added separately).

PRINCIPLE:
An electrical-mechanical domain interface of the dynamic type appears as an ideal transformer because of the change to admittance analogy in going from the electrical to the mechanical side. On the mechanical side, voltage represents velocity and current represents force.

REFERENCE:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 70-71.

EMINTES.CIR: ELECTRICAL-MECHANICAL DOMAIN
INTERFACE FOR ELECTROSTATIC TRANSDUCER



.PARAMETERS(k)

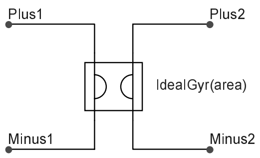
PARAMETER: k = force factor (force/voltage).

USAGE:
This macro is an electrical-mechanical domain interface, modelling the kind of transduction appearing in electrostatic transducers (output force proportional to input voltage, output current proportional to input velocity). The interface macro is ideal and does not include any other characteristics such as e.g. membrane-to-plate capacitance (these must be added separately). The macro can be connected either way (port 1 electrical, port 2 mechanical, or vice versa).

PRINCIPLE:
An electrical-mechanical domain interface of the electrostatic type appears as an ideal gyrator because of the change to admittance analogy in going from the electrical to the mechanical side. On the mechanical side, voltage represents velocity and current represents force.

REFERENCE:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 71-75.

MAINT.CIR: MECHANICAL-ACOUSTICAL DOMAIN INTERFACE



.PARAMETERS(area)

PARAMETER: area = (effective) area of connecting surface.

USAGE:
If a mechanical circuit is to be connected to an acoustical circuit, this macro should be used as the interface. It represents a (hypothetical) surface that is common to both circuits. The most typical case is the area of loudspeaker membrane. The macro can be connected either way (port 1 acoustical, port 2 mechanical, or vice versa).

PRINCIPLE:
A mechanical-acoustical domain interface appears as a gyrator (not a transformer) because of the change from admittance analogy to impedance analogy in going from the mechanical to the acoustical side. On the mechanical side, voltage represents velocity and current represents force. On the acoustical side, voltage represents pressure and current represents volume velocity.

REFERENCE:

Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. p. 75.

Radiation

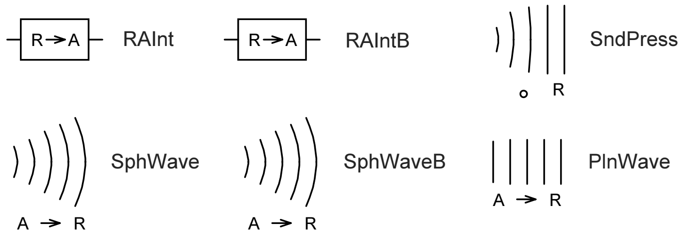


Figure IV.11: Radiation-acoustical domain interface (general and with finite baffle), sound pressure source, spherical wave (general and with finite baffle), plane wave.

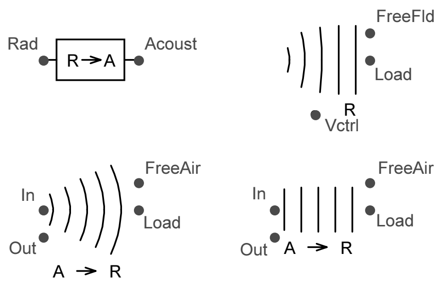
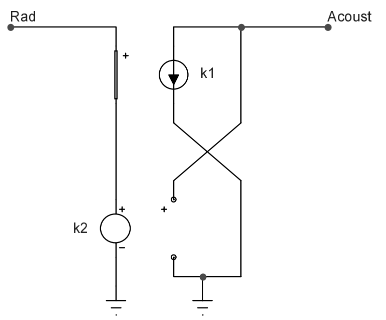


Figure IV.12: The above components with their pins shown.

RAINT.CIR: PRESSURE/ VELOCITY - PRESSURE/ VOLUME VELOCITY INTERFACE



.PARAMETERS(area, no, baffle)
.DEFINE k1 -(area)*(no)/(2-(baffle))
.DEFINE k2 2-(baffle)

```
PARAMETERS: area   = area of connecting surface(s) subject to sound radiation,
            no     = number of surfaces,
            baffle = 1 or 0 (infinite baffle or no baffle around surface(s)).

PINS:       Rad:    input/output of pressure (voltage) and particle velocity (current),
            Acoust: input/output of pressure (voltage) and volume velocity (current).
```

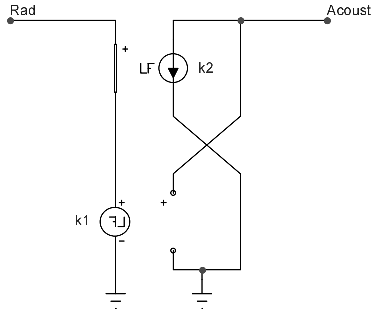
USAGE:
This macro should be connected between an acoustic circuit and the output of a radiation macro, to model the effect of sound radiation incident on the circuit (through a hole, a movable surface etc., or an array of such entrances). Because radiation macros output pressure and particle velocity (not volume velocity), this interface is needed to obtain the pressure and volume velocity input for the acoustic circuit that follows. If the entrance subject to sound radiation resides in an infinite baffle, in the wall of a cavity etc., baffle=1 should be used. If conditions correspond to a free field (e.g. a small microphone measuring a loudspeaker), then baffle=0.

LIMITATIONS:
It is assumed that the dimensions of the surface(s) connecting to the following acoustic circuit are considerably smaller than the wavelength. If a baffle is present (baffle = 1), it is assumed that the baffle is considerably larger than the wavelength.

PRINCIPLE:
The transformation between pressure/ velocity ("radiation domain") and pressure/ volume velocity (normal acoustical domain) is realized as a transformer with unequal transformation ratios for voltage and current. The pressure remains the same on both sides, but the particle velocity must be transformed to a volume velocity, and vice versa. If there is a baffle, the pressure at the receiving surface doubles due to reflections from the baffle.

REFERENCE:
Leach, W. M., Jr. On the electroacoustic-analogous circuit for a plane wave incident on the diaphragm of a free-field pressure microphone. J. Audio Eng. Soc., vol. 38, no. 7/8, 1990.

RAINTB.CIR: PRESSURE/ VELOCITY - PRESSURE/ VOLUME VELOCITY INTERFACE WITH BAFFLE



.PARAMETERS(area, no, bperim)
.INCLUDE CONSTDEF.TXT
.DEFINE sc (2*pi*2.7*c0/(bperim))
.DEFINE k1 ((2+(s/sc)**4)/(1+(s/sc)**4))
.DEFINE k2 -(area)*(no)/k1

```
PARAMETERS: area   = area of connecting surface(s) in the baffle subject to sound radiation,
            no     = number of surfaces in the baffle,
            bperim = perimeter of baffle.

PINS:       Rad:    input/output of pressure (voltage) and particle velocity (current),
            Acoust: input/output of pressure (voltage) and volume velocity (current).
```

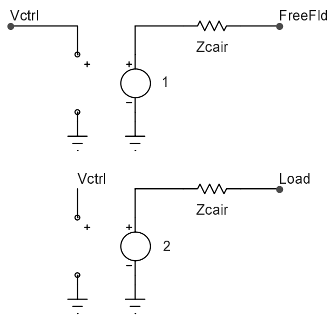
USAGE:
This macro is similar to the RAInt macro, except that it models one or more entrances surrounded by a finite baffle, whose perimeter is specified. For more details, see the description of the RAInt macro.

LIMITATIONS:
In modelling the baffle, accuracy is reasonable in vicinity of the transition frequency from full- to half-space radiation, as long as the baffle shape does not depart too much from a circle or square (i.e. the aspect ratio must not be large). Also, the connection(s) to the loading acoustic circuit must be located close enough to the center of the baffle. For other limitations, see the description of the RAInt macro.

PRINCIPLE:
The gradual pressure rise due to the surrounding baffle (above a certain threshold frequency) is modelled as a gradual change in the gains of the controlled sources. The function describing this change was obtained by manually fitting its graph to a curve derived from the frequency dependence of the radiation impedance of a circular piston in the end of a long tube.

REFERENCES:
Leach, W. M., Jr. On the electroacoustic-analogous circuit for a plane wave incident on the diaphragm of a free-field pressure microphone. J. Audio Eng. Soc., vol. 38, no. 7/8, 1990.
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 116-128.

SNDPRESS.CIR: SOUND PRESSURE SOURCE (PLANE WAVE, VOLTAGE-CONTROLLED)



.INCLUDE CONSTDEF.TXT

```
PARAMETERS: -

PINS:       Vctrl:   voltage control input,
            FreeFld: output of free-field sound pressure (with no obstacle present),
            Load:    pin that can be connected to other acoustical loads that receive the sound waves.
```

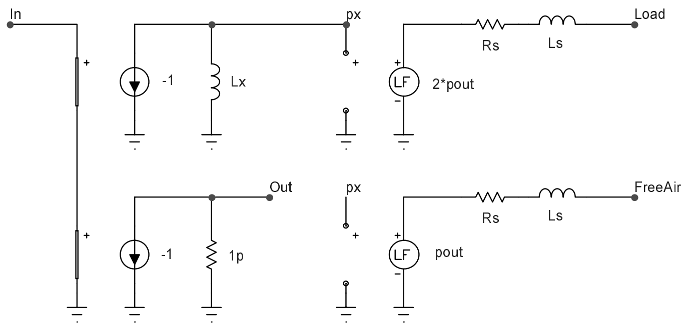
USAGE:
This macro can be controlled by any available Micro Cap voltage source, or circuit voltage, to produce a sound pressure in the acoustical domain. What distinguishes this macro from a general pressure source is the specific acoustic impedance (of a plane wave) that is included at the output. Also, this macro is convenient when simulating the interaction of a given free-field sound pressure with an acoustical circuit. Then, that circuit should be connected to the Load pin via a separate RAInt macro (FreeFld always gives the sound pressure with no obstacle, and this is also the pressure specified by the voltage brought to the Vctrl pin). The free-field particle velocity is available as the short-circuit current of the FreeFld pin.

LIMITATION:
The model assumes that the waves are plane.

PRINCIPLE:
On the right side of this circuit, current corresponds not to volume velocity, but to particle velocity. The sound pressure to the Load pin is doubled to account for effects of various loads that can be connected to this output. Impedances shown are the specific acoustic impedance of a plane wave.

REFERENCES:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. p. 35.
Olson, H. F. Acoustical engineering (1991 edition, orig. D. van Nostrand Co. 1957). Philadelphia, PA, USA, 1991. Professional Audio Journals Inc. p. 101.
Leach, W. M., Jr. On the electroacoustic-analogous circuit for a plane wave incident on the diaphragm of a free-field pressure microphone. J. Audio Eng. Soc., vol. 38, no. 7/8, 1990.

SPHWAVE.CIR: SPHERICAL WAVE RADIATION



.PARAMETERS(dist, spacecf)
.INCLUDE CONSTDEF.TXT
.DEFINE Lx rho0/(4*pi*(spacecf))
.DEFINE pout EXP(-s*(dist)/c0)/(dist) = propagation delay and attenuation
.DEFINE kd (2*pi*f*(dist)/c0)
.DEFINE Rs rho0*c0*kd**2/(1+kd**2) = specific acoustic resistance of wave
.DEFINE Ls rho0*(dist)/(1+kd**2) = specific acoustic inductance of wave

PARAMETERS: dist = distance (from radiator surface to observation point),
spacecf = portion of full space that the radiation is into (0 < spacecf <= 1, 1 = full space, 0.5 = half space, etc.),

PINS:
In: input pin for the volume velocity causing the radiation,
Out: output pin copying the input volume velocity,
FreeAir: pin for sensing of free-field sound pressure at distance specified by dist,
Load: pin that can be connected to other acoustical loads that receive the radiated sound waves.

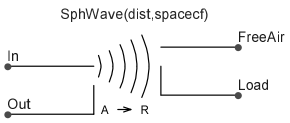
USAGE:
Normally, this macro should be connected in series with an air load macro (with the volume velocity passing through the air load macro used directly as input to this SphWave macro). The output will be the sound pressure (appropriately delayed and attenuated) at the given distance. Radiation can be into any solid angle (as specified by spacecf). For most cases with radiation into free air, spacecf=1 is appropriate (unless there is a large baffle). The sound pressure in free air (no obstacles) should be sensed at the FreeAir pin. Also, the particle velocity is available as the short-circuit current at the same pin. If there is an obstacle of some kind (e.g. a microphone, a wall etc.), the corresponding acoustic circuit should be connected to the Load pin via a separate RAInt macro (FreeAir always gives the sound pressure with no obstacle). The only purpose of the Out pin is to provide a simple means of connecting several SphWave macros to one single volume velocity source (to simulate pressures at several distances from the same source).

LIMITATIONS:
Reflections (from a possible obstacle back to the radiator) are taken into account at the Load pin, but not at the other end (although the reflected waves arrive there delayed and attenuated). At very close distances (near field), the simulated sound field is really accurate only for radiators with a spherical surface (not just any radiators small compared to the wavelength). The model treats spacecf as describing an infinite space, e.g. ideal full space, an infinite baffle (spacecf=0.5) or an infinite conical horn (spacecf<0.5).

PRINCIPLE:
On the right side of this circuit, current corresponds not to volume velocity, but to particle velocity. The sound pressure to the Load pin is doubled to account for effects of various loads that can be connected to this output. Impedances shown are the specific acoustic impedance of the spherical wave.

REFERENCES:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. p. 36.
Morse, P. M. Vibration and sound (2nd edition). New York, USA, 1948. McGraw-Hill. (Ref. in Beranek)
Kinsler, L. E., Frey, A. R. Fundamentals of acoustics. Monterey, Calif., USA, 1982. John Wiley & Sons, Inc. pp.163-173.
Olson, H. F. Acoustical engineering (1991 edition, orig. D. van Nostrand Co. 1957). Philadelphia, PA, USA, 1991. Professional Audio Journals Inc. pp. 102-103.

SPHWAVEB.CIR: SPHERICAL WAVE RADIATION (RADIATOR SURROUNDED BY FINITE BAFFLE)




```
.PARAMETERS(dist, bperim)
.INCLUDE CONSTDEF.TXT
.DEFINE fc  (2.7*c0/(bperim))                    = transition frequency (halfway between full- and half-space radiation)
.DEFINE spacecf  (1+0.5*(f/fc)**4)/(1+(f/fc)**4)  = transition from spacecf=1 to spacecf=0.5
```

```
PARAMETERS: dist   = distance (from radiator surface to observation point),
            bperim = perimeter of baffle,

PINS:       In:      input pin for the volume velocity causing the radiation,
            Out:     output pin copying the input volume velocity,
            FreeAir: pin for sensing of free-field sound pressure at distance specified by dist,
            Load:    pin that can be connected to other acoustical loads that receive the radiated sound waves.
```

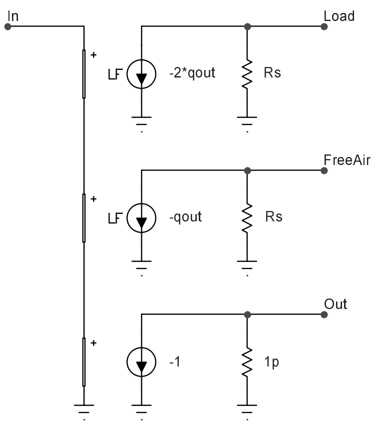
USAGE:
This macro is similar to the SphWave macro, except that it models radiation emitted by one or more radiators surrounded by a finite baffle, whose perimeter is specified. For more details, see the description of the SphWave macro.

LIMITATIONS:
In modelling the baffle, accuracy is reasonable in vicinity of the transition frequency from full- to half-space radiation, as long as the baffle shape does not depart too much from a circle or square (i.e. the aspect ratio must not be large). Also, the radiator(s) must be located close enough to the center of the baffle. For other limitations, see the description of the SphWave macro.

PRINCIPLE:
The transition from full- to half-space radiation (as frequency increases) is modelled here using a frequency-dependent value for spacecf, which is passed to the SphWave macro. The function describing this change was obtained by manually fitting its graph to a curve derived from the frequency dependence of the radiation impedance of a circular piston in the end of a long tube.

REFERENCE:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 116-128.

PLNWAVE.CIR: PLANE WAVE RADIATION




```
.PARAMETERS(area, no, dist)
.INCLUDE CONSTDEF.TXT
.DEFINE qout  EXP(-s*(dist)/c0)        = propagation delay
.DEFINE Rs  rho0*c0/((area)*(no))      = specific acoustic resistance of wave
```

```
PARAMETER: area = area of wave front (and radiator),
           no   = number of wave fronts (and radiators),
           dist = distance from radiator(s) to observation point.

PINS:      In:      input pin for the volume velocity causing the radiation,
           Out:     output pin copying the input volume velocity,
           FreeAir: pin for sensing of free-field sound pressure at distance specified by dist,
           Load:    pin that can be connected to other acoustical loads that receive the radiated sound waves.
```

USAGE:
Normally, this macro should be connected in series with an air load macro (with the volume velocity passing through the air load macro used directly as input to this PlnWave macro). The output will be the sound pressure (appropriately delayed) at the given distance. The sound pressure in free air (no obstacles) should be sensed at the FreeAir pin. Also, the particle velocity is available as the short-circuit current at the same pin. If there is an obstacle of some kind (e.g. a microphone, a wall etc.), the corresponding acoustic circuit should be connected to the Load pin via a separate RAInt macro (FreeAir always gives the sound pressure with no obstacle). The only purpose of the Out pin is to provide a simple means of connecting several PlnWave macros to one single volume velocity source (to simulate pressures at several distances from the same source).

LIMITATION:
Reflections (from a possible obstacle back to the radiator) are taken into account at the Load pin, but not at the other end (although the reflected waves arrive there delayed and attenuated).

PRINCIPLE:
On the right side of this circuit, current corresponds not to volume velocity, but to particle velocity. The sound

pressure to the Load pin is doubled to account for effects of various loads that can be connected to this output. Impedances shown are the specific acoustic impedances of the plane wave.

REFERENCES:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. p. 35.
Olson, H. F. Acoustical engineering (1991 edition, orig. D. van Nostrand Co. 1957). Philadelphia, PA, USA, 1991. Professional Audio Journals Inc. p. 101.
Leach, W. M., Jr. On the electroacoustic-analogous circuit for a plane wave incident on the diaphragm of a free-field pressure microphone. J. Audio Eng. Soc., vol. 38, no. 7/8, 1990.

Transducers

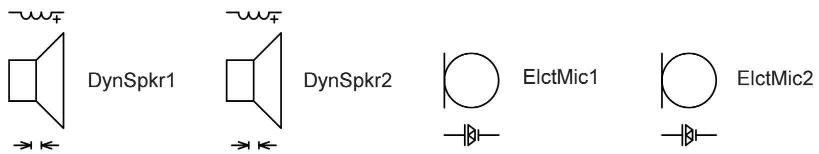


Figure IV.13: Dynamic speakers and electret microphones.

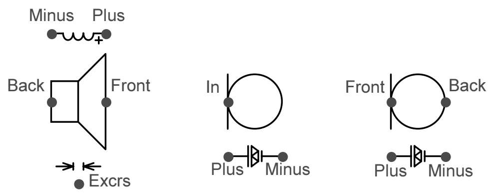
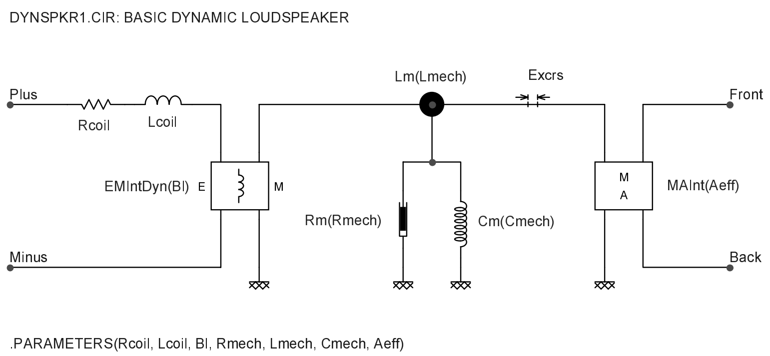


Figure IV.14: The above components with their pins shown.



PARAMETERS: Rcoil = voice coil resistance,
Lcoil = voice coil inductance,
Bl = force factor,
Rmech = mechanical resistance of suspension,
Lmech = mechanical inductance (moving mass of loudspeaker),
Cmech = mechanical capacitance (compliance) of suspension,
Aeff = effective area of diaphragm,

PINS: Plus: "+" pole of voice coil,
Minus: "-" pole of voice coil,
Front: acoustic output, front of diaphragm,
Back: acoustic output, back of diaphragm,
Excrs: diaphragm excursion (output as a voltage).

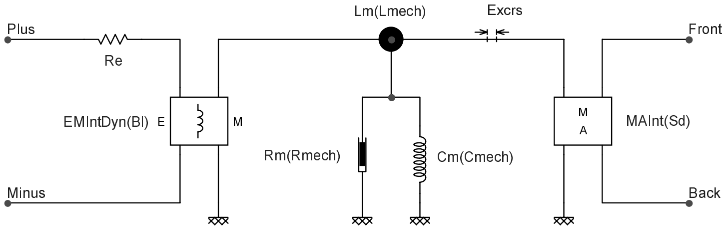
USAGE:
This macro models the behaviour of a typical dynamic loudspeaker at low frequencies. Acoustical loads (such as radiation impedance) can be connected to both the front and the back of the diaphragm.

LIMITATIONS:
Phenomena entering at higher frequencies (breakup resonances etc.) are not modelled. Thus, there is an upper limiting frequency (above which the model is no longer accurate) that depends on the properties of the loudspeaker that is modelled.

PRINCIPLE:
The loudspeaker is modelled using a three-domain equivalent circuit with the membrane and its suspension modelled as a mechanical resonator with given component values. The effective area of the diaphragm forms the interface between the mechanical and acoustical circuits, and the force factor is seen as the interface between the electrical and mechanical circuits.

REFERENCE:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 70-81.

DYNSPKR2.CIR: BASIC DYNAMIC LOUDSPEAKER (WITH THIELE-SMALL PARAMETERS)




```
.PARAMETERS(Re, fs, Vas, Qms, Qes, Sd)
.INCLUDE CONSTDEF.TXT
.DEFINE Bl  SQRT(rho0*c0**2*(Sd)**2*(Re)/(2*pi*(fs)*(Vas)*(Qes)))
.DEFINE Rmech  rho0*c0**2*(Sd)**2/(2*pi*(fs)*(Vas)*(Qms))
.DEFINE Lmech  rho0*c0**2*(Sd)**2/((2*pi*(fs))**2*(Vas))
.DEFINE Cmech  (Vas)/(rho0*c0**2*(Sd)**2)
```

```
PARAMETERS: Re  = voice coil resistance,
            fs  = free-air resonance frequency,
            Vas = air volume equivalent of suspension compliance,
            Qms = Q value at resonance considering non-electrical resonances only,
            Qes = Q value at resonance considering electrical resonances only,
            Sd  = effective area of diaphragm.

PINS:       Plus:  "+" pole of voice coil,
            Minus: "-" pole of voice coil,
            Front: acoustic output, front of diaphragm,
            Back:  acoustic output, back of diaphragm,
            Excrs: diaphragm excursion (output as a voltage).
```

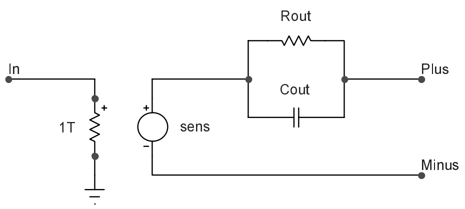
USAGE:
This macro models the behaviour of a typical dynamic loudspeaker at low frequencies. Acoustical loads (such as radiation impedance) can be connected to both the front and the back of the diaphragm.

LIMITATIONS:
Phenomena entering at higher frequencies (breakup resonances etc.) are not modelled. Thus, there is an upper limiting frequency (above which the model is no longer accurate) that depends on the properties of the loudspeaker that is modelled. The voice coil inductance is not included in the model.

PRINCIPLE:
The loudspeaker is modelled using a three-domain equivalent circuit with the membrane and its suspension modelled as a mechanical resonator with component values calculated from given Thiele-Small parameters. The effective area of the diaphragm forms the interface between the mechanical and acoustical circuits, and the force factor is seen as the interface between the electrical and mechanical circuits.

REFERENCES:
Beranek, L. L. Acoustics. New York, USA, 1954. McGraw-Hill. pp. 70-81.
Small, R. H. Direct-radiator loudspeaker system analysis. J. Audio Eng. Soc., vol. 20, no. 2, 1972.

ELCTMIC1.CIR: BASIC ELECTRET MICROPHONE




```
.PARAMETERS(sens, Rout, Cout)
```

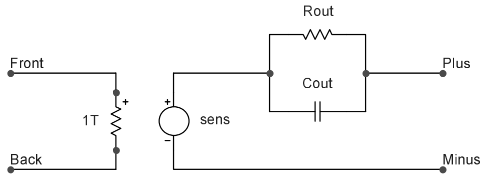
```
PARAMETERS: sens = sensitivity (V/Pa),
            Rout = output resistance,
            Cout = output capacitance.

PINS:       In:    acoustic (pressure) input,
            Plus:  "+" electric output (positive at positive pressures),
            Minus: "-" electric output.
```

USAGE:
The macro is usable as a simple model of a typical electret microphone.

PRINCIPLE:
The microphone is modelled as a simple VCVS with the given output impedance. An input impedance of 1 Tohm is included to prevent divergence (the acoustic input impedance is, in effect, assumed high enough to be negligible).

REFERENCES:
Various data sheets from microphone manufacturers.

ELCTMIC2.CIR: BASIC ELECTRET MICROPHONE (PRESSURE GRADIENT TYPE)



.PARAMETERS(sens, Rout, Cout)

```
PARAMETERS: sens = sensitivity (V/Pa),
            Rout = output resistance,
            Cout = output capacitance.

PINS:       Front: acoustic (pressure) input (front of diaphragm),
            Back:  acoustic input (back of diaphragm),
            Plus:  "+" electric output (positive at positive pressures, front higher than back),
            Minus: "-" electric output.
```

USAGE:
This macro is usable as a simple model of a typical electret microphone of the pressure gradient type. The sensitivity is expressed as output voltage with a given pressure difference between front and back pressure. This differs from the way sensitivity is given in typical data sheets, so some calculation might be needed to get the correct sensitivity value for the macro.

PRINCIPLE:
The microphone is modelled as a simple VCVS with the given output impedance. An input impedance of 1 Tohm is included to prevent divergence (the acoustic input impedance is, in effect, assumed high enough to be negligible).

REFERENCES:
Various data sheets from microphone manufacturers.

Measuring equipment

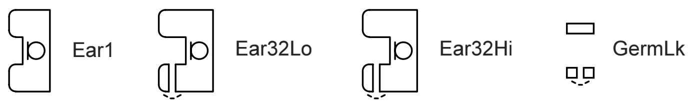


Figure IV.15: ITU-T P.57 type 3.1 sealed coupler (ear simulator), type 3.2 low-leak and high-leak coupler, leak ring ("German leak") for the sealed coupler.

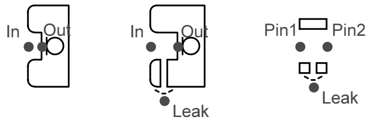
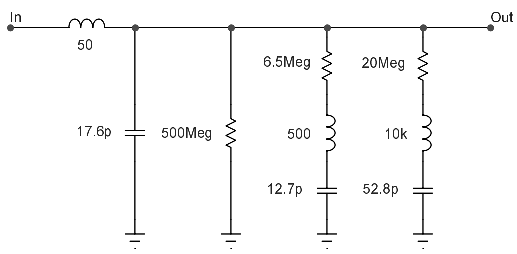


Figure IV.16: The above components with their pins shown.

EAR1.CIR: ITU-T P.57 TYPE 1 EAR SIMULATOR

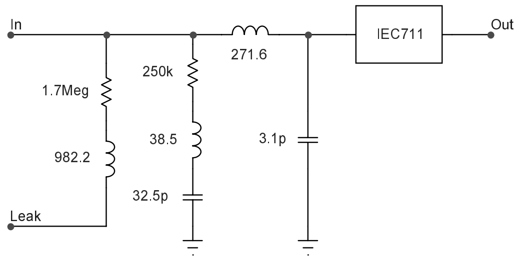



```
PARAMETERS: -

PINS:       In:  pressure input,
            Out: internal pressure measured by microphone.
```

USAGE:
This macro models the ITU-T P.57 type 1 ear simulator. The pressure measured by the internal microphone can be sensed at the Out pin.

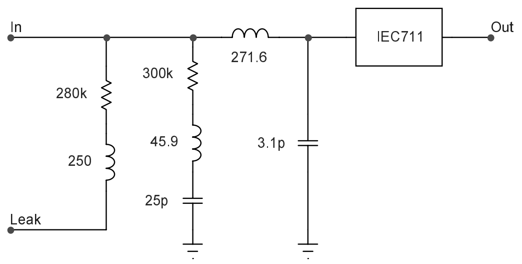
```
LIMITATION:
The macro should not be used above 8 kHz.

REFERENCE:
Product data: ear simulator for telephonometry - type 4185. Brüel & Kjær, Nærum, Denmark, 1997.
```

EAR32LO.CIR: ITU-T P.57 TYPE 3.2 LOW-LEAK EAR SIMULATOR



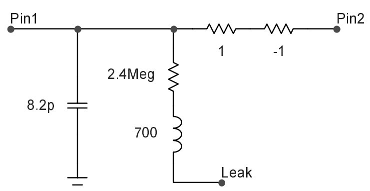

```
PARAMETERS: -

PINS:       In:   pressure input,
            Out:  internal pressure measured by microphone,
            Leak: leak openings.

USAGE:
This macro models the ITU-T P.57 type 3.2 low-leak ear simulator. The pressure measured by the internal microphone can
be sensed at the Out pin. In normal usage, the Leak pin is to be connected to atmospheric pressure (i.e. followed by an
AtmPress macro).

LIMITATIONS:
The accuracy of the macro is best at low and midrange frequencies. Above 2.5 kHz, there are local deviations of up to
+2 and -6 dB (simulated vs real). The macro should not be used above 8 kHz.

REFERENCE:
Product data: wideband ear simulator for telephonometry - type 4195. Brüel & Kjær, Nærum, Denmark, 1997.
```

EAR32HI.CIR: ITU-T P.57 TYPE 3.2 HIGH-LEAK EAR SIMULATOR




```
PARAMETERS: -

PINS:       In:   pressure input,
            Out:  internal pressure measured by microphone,
            Leak: leak openings.

USAGE:
This macro models the ITU-T P.57 type 3.2 high-leak ear simulator. The pressure measured by the internal microphone can
be sensed at the Out pin. In normal usage, the Leak pin is to be connected to atmospheric pressure (i.e. followed by an
AtmPress macro).

LIMITATIONS:
The accuracy of the macro is best at low and midrange frequencies. Above 3.5 kHz, there are local deviations of up to
-6 dB (simulated vs real). The macro should not be used above 8 kHz.

REFERENCE:
Product data: wideband ear simulator for telephonometry - type 4195. Brüel & Kjær, Nærum, Denmark, 1997.
```

GERMLK.CIR: GERMAN LEAK RING




```
PARAMETERS: -
```

PINS: Pin1, Pin2: connection points for Ear1 ear simulator and acoustic circuit,
Leak: leak openings.

USAGE:
This macro can be used in series with the Ear1 ear simulator macro, to model a special leak adapter manufactured by Systel Elektronik GmbH, Germany. In normal usage, the Leak pin is to be connected to atmospheric pressure (i.e. followed by an AtmPress macro).

LIMITATIONS:
The model does not have any frequency dependent impedances in it, which leads to lower accuracy.

PRINCIPLE:
The model was created by measuring mechanical dimensions. The opposite resistor pair is used merely to prevent an error that would result from shorting two different pins together.

REFERENCE:
Model created by C. Poldy, Philips Speaker Systems, Austria.

Probes and annotators

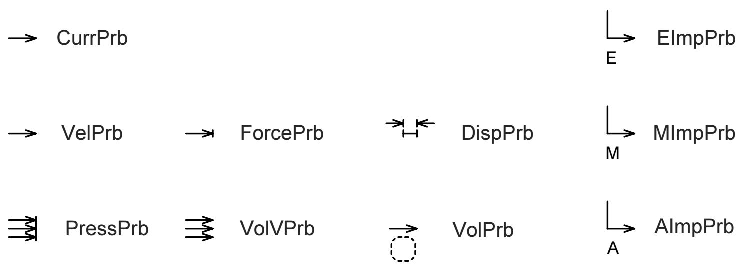


Figure IV.17: Current and electric impedance probe (top row), velocity, force, displacement and mechanical impedance probe (middle row), pressure, volume velocity, volume and acoustic impedance probe (bottom row).

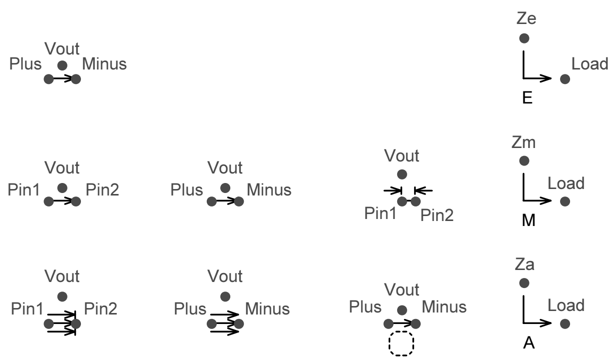
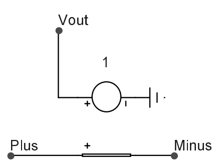


Figure IV.18: The above components with their pins shown.

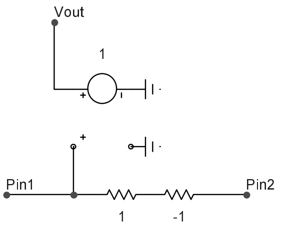
CURRPRB.CIR: CURRENT PROBE



PARAMETERS: -

USAGE:
The current probe is used for easy annotation and analysis display of currents. The probe is inserted into the desired branch, and named by putting grid text on pin Vout, whose voltage will be equal in magnitude to the current passing through the probe. This macro is just a convenient feature when annotating currents in a circuit, or implementing current-controlled sources.

VELPRB.CIR: VELOCITY PROBE



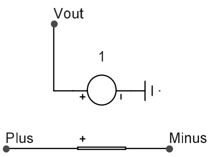
PARAMETERS: -

USAGE:
The velocity probe is used for easy annotation and analysis display of velocities in a mechanical circuit. The probe is inserted at the desired node, and named by putting grid text on pin Vout, whose voltage will be equal in magnitude to the velocity of the probed point. This macro is just a convenient feature when annotating velocities in a mechanical circuit, or implementing velocity-controlled sources.

LIMITATION:
The macro can be used in mechanical circuits only. Trying to probe velocities in an acoustic circuit gives wrong results.

PRINCIPLE:
The opposite resistor pair is used merely to prevent an error resulting from shorting two different pins together. Velocity corresponds to voltage in the admittance analogy used in the mechanical domain.

FORCEPRB.CIR: FORCE PROBE



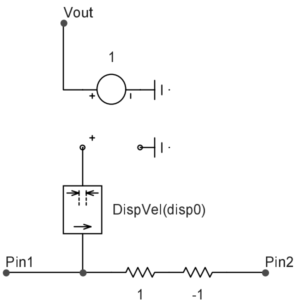
PARAMETERS: -

USAGE:
The force probe is used for easy annotation and analysis display of forces in a mechanical circuit. The probe is inserted into the desired branch, and named by putting grid text on pin Vout, whose voltage will be equal in magnitude to the force "passing" through the probe. This macro is just a convenient feature when annotating forces in a circuit, or implementing force-controlled sources.

LIMITATION:
The macro can be used in mechanical circuits only. Trying to probe a force acting on something in an acoustic circuit gives wrong results.

PRINCIPLE:
Force corresponds to current in the admittance analogy used in the mechanical domain.

DISPPRB.CIR: DISPLACEMENT PROBE



.PARAMETERS(disp0)

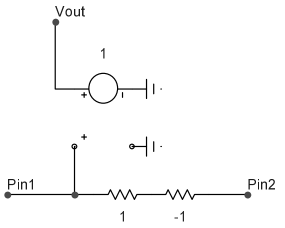
PARAMETER: disp0 = initial displacement.

USAGE:
The displacement probe is used for easy annotation and analysis display of displacements in a mechanical circuit. The probe is inserted at the desired node, and named by putting grid text on pin Vout, whose voltage will be equal in magnitude to the displacement of the probed point. This macro is just a convenient feature when annotating displacements in a mechanical circuit, or implementing displacement-controlled sources.

LIMITATION:
The macro can be used in mechanical circuits only. Trying to probe the displacement of something in an acoustic circuit gives wrong results.

PRINCIPLE:
Integration of the velocity is carried out to get the displacement. The opposite resistor pair is used merely to prevent an error resulting from shorting two different pins together.

PRESSPRB.CIR: PRESSURE PROBE



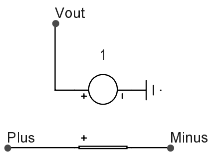
PARAMETERS: -

USAGE:
The pressure probe is used for easy annotation and analysis display of pressures in an acoustic circuit. The probe is inserted at the desired node, and named by putting grid text on pin Vout, whose voltage will be equal in magnitude to the pressure at the probed point. This macro is just a convenient feature when annotating pressures in a mechanical circuit, or implementing pressure-controlled sources.

LIMITATION:
The macro can be used in acoustic circuits only. Trying to probe pressures in a mechanical circuit gives wrong results.

PRINCIPLE:
The opposite resistor pair is used merely to prevent an error resulting from shorting two different pins together. Pressure corresponds to voltage in the impedance analogy used in the acoustical domain.

VOLVPRB.CIR: VOLUME VELOCITY PROBE



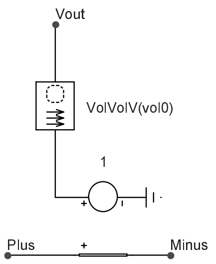
PARAMETERS: -

USAGE:
The volume velocity probe is used for easy annotation and analysis display of volume velocities in an acoustic circuit. The probe is inserted into the desired branch, and named by putting grid text on pin Vout, whose voltage will be equal in magnitude to the volume velocity flowing through the probe. This macro is just a convenient feature when annotating volume velocities in a circuit, or implementing volume velocity -controlled sources.

LIMITATION:
The macro can be used in acoustic circuits only.

PRINCIPLE:
Volume velocity corresponds to current in the impedance analogy used in the acoustical domain.

VOLPRB.CIR: VOLUME PROBE



.PARAMETERS(vol0)

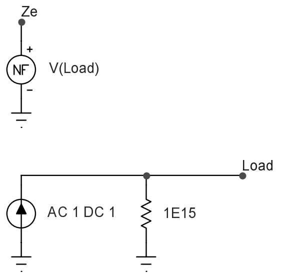
PARAMETER: vol0 = initial volume.

USAGE:
The volume probe is used for easy annotation and analysis display of volumes in an acoustic circuit. The probe is inserted into the desired branch, and named by putting grid text on pin Vout, whose voltage will be equal in magnitude to the volume flowing through the probe. This macro is just a convenient feature when annotating volumes in an acoustic circuit, or implementing volume-controlled sources.

LIMITATION:
The macro can be used in acoustic circuits only.

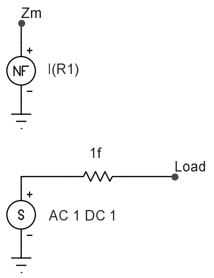
PRINCIPLE:
Integration of the volume velocity is carried out to get the volume. The opposite resistor pair is used merely to prevent an error resulting from shorting two different pins together.

EIMPPRB.CIR: IMPEDANCE PROBE FOR ELECTRIC IMPEDANCE



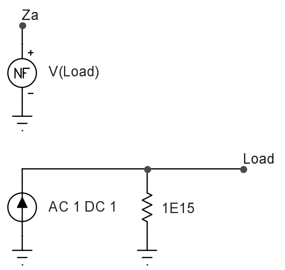

```
PARAMETERS: -

PINS:       Load: this pin is loaded with the impedance to be measured,
            Ze:   outputs the impedance as a (complex) voltage.

USAGE:
This macro is suitable for measuring impedances (to "ground") at various points in a circuit. Also, the impedance can be
annotated by putting grid text on node Ze.

LIMITATIONS:
An internal shunt resistance of 1E15 ohm is included to prevent divergence.
```

MIMPPRB.CIR: IMPEDANCE PROBE FOR MECHANICAL IMPEDANCE




```
PARAMETERS: -

PINS:       Load: this pin is loaded with the impedance to be measured,
            Zm:   outputs the impedance as a (complex) voltage.

USAGE:
This macro is suitable for measuring impedances (to "ground") at various points in a circuit. Also, the impedance can be
annotated by putting grid text on node Zm.

LIMITATIONS:
An internal series resistance of 1 fohm is included to prevent divergence.
```

AIMPPRB.CIR: IMPEDANCE PROBE FOR ACOUSTIC IMPEDANCE




```
PARAMETERS: -

PINS:       Load: this pin is loaded with the impedance to be measured,
            Za:   outputs the impedance as a (complex) voltage.

USAGE:
This macro is suitable for measuring impedances (to "ground") at various points in a circuit. Also, the impedance can be
annotated by putting grid text on node Za.

LIMITATIONS:
An internal shunt resistance of 1E15 ohm is included to prevent divergence.
```

Compound macros

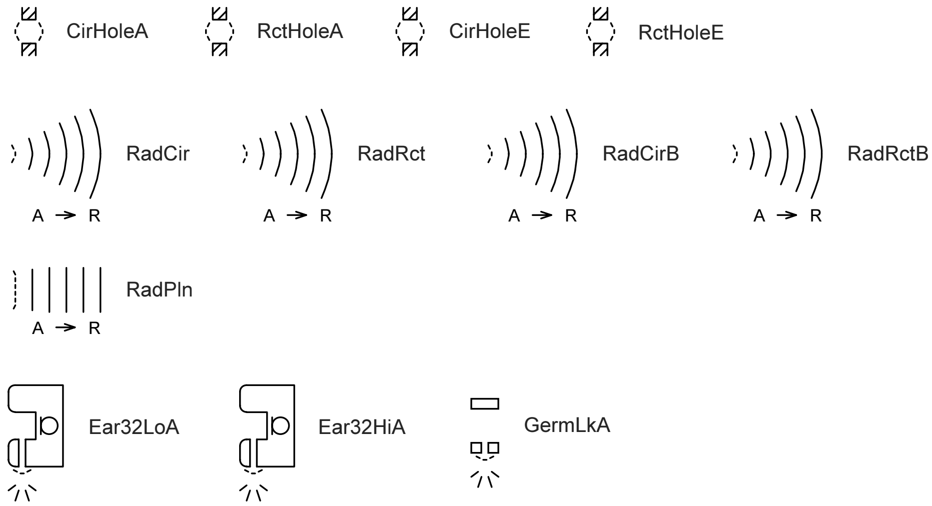


Figure IV.19: General Helmholtz resonator, circular and rectangular hole with air load and general end correction (1st row), circular and rectangular radiators, general and with finite baffle (2nd row), plane wave radiator (3rd row), low- and high-leak ear simulators and leak adapter, each with no separate leak pin (4th row).

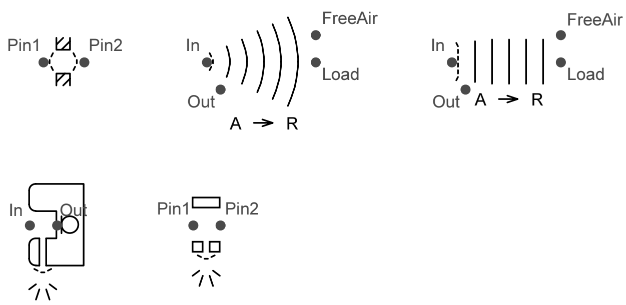
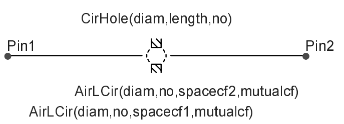


Figure IV.20: The above components with their pins shown.

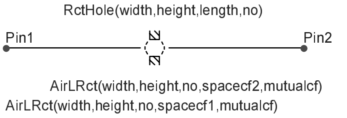
CIRHOLEA.CIR: CIRCULAR HOLE(S) WITH AIR LOAD



.PARAMETERS(diam, length, no, spacecf1, spacecf2, mutualcf)

```
PARAMETERS: diam     = diameter of hole(s),
            length   = length in flow direction (excl. end corrections),
            no       = number of parallel holes,
            spacecf1 = portion of full space that the radiation is into (one side),
            spacecf2 = portion of full space that the radiation is into (other side),
            mutualcf = mutual coupling coefficient.

USAGE:
This macro is a ready-made combination of a circular hole and its two end corrections. For more instructions, see the
corresponding macros.

LIMITATIONS:
See the CirHole and AirLCir macros.
```

RCTHOLEA.CIR: RECTANGULAR HOLE(S) WITH AIR LOAD



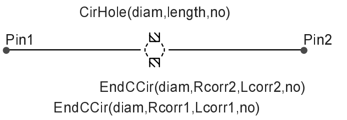
.PARAMETERS(width, height, length, no, spacecf1, spacecf2, mutualcf)

```
PARAMETERS: width    = width of hole(s),
            height   = height of hole(s),
            length   = length in flow direction (excl. end corrections),
            no       = number of parallel holes,
            spacecf1 = portion of full space that the radiation is into (one side),
            spacecf2 = portion of full space that the radiation is into (other side),
            mutualcf = mutual coupling coefficient.
```

USAGE:
This macro is a ready-made combination of a rectangular hole and its two end corrections. For more instructions, see the corresponding macros.

LIMITATIONS:
See the RctHole and AirLRct macros.

CIRHOLEE.CIR: CIRCULAR HOLE(S) WITH END CORRECTION



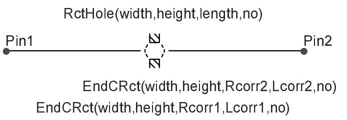
.PARAMETERS(diam, length, Rcorr1, Lcorr1, Rcorr2, Lcorr2, no)

```
PARAMETERS: diam     = diameter of hole(s),
            length   = length in flow direction (excl. end corrections),
            Rcorr1   = end correction for resistance (one side),
            Lcorr1   = end correction for inductance (one side),
            Rcorr2   = end correction for resistance (other side),
            Lcorr2   = end correction for inductance (other side),
            no       = number of parallel holes,
```

USAGE:
This macro is a ready-made combination of a circular hole and its two end corrections. For more instructions, see the corresponding macros.

LIMITATIONS:
See the CirHole and EndCCir macros.

RCTHOLEE.CIR: RECTANGULAR HOLE(S) WITH END CORRECTION



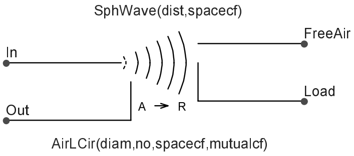
.PARAMETERS(width, height, length, Rcorr1, Lcorr1, Rcorr2, Lcorr2, no)

```
PARAMETERS: width    = width of hole(s),
            height   = height of hole(s),
            length   = length in flow direction (excl. end corrections),
            Rcorr1   = end correction for resistance (one side),
            Lcorr1   = end correction for inductance (one side),
            Rcorr2   = end correction for resistance (other side),
            Lcorr2   = end correction for inductance (other side),
            no       = number of parallel holes.
```

USAGE:
This macro is a ready-made combination of a rectangular hole and its two end corrections. For more instructions, see the corresponding macros.

LIMITATIONS:
See the RctHole and EndCRct macros.

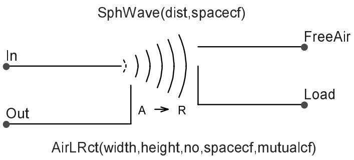
RADCIR.CIR: RADIATION FROM CIRCULAR RADIATOR(S)
(<< WAVELENGTH)



.PARAMETERS(diam, no, spacecf, mutualcf, dist)

```
PARAMETERS: diam     = diameter of radiator(s),
            no       = number of parallel radiators,
            spacecf  = portion of full space that the radiation is into,
            mutualcf = mutual coupling coefficient,
            dist     = distance from radiator(s) to observation point.

PINS:       In:      pin to be connected to the acoustic circuit,
            Out:     output pin copying the input volume velocity,
            FreeAir: pin for sensing of free-field sound pressure at distance specified by dist,
            Load:    pin that can be connected to other acoustical loads that receive the radiated sound waves.

USAGE:
This macro is a ready-made combination of air load and radiation (spherical waves) from a circular radiator. For more
information, see the corresponding macros.

LIMITATIONS:
See the AirLCir and SphWave macros.
```

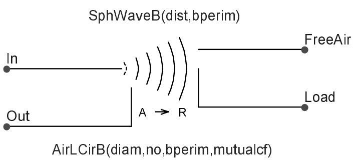
RADRCT.CIR: RADIATION FROM RECTANGULAR RADIATOR(S)
(<< WAVELENGTH)



.PARAMETERS(width, height, no, spacecf, mutualcf, dist)

```
PARAMETERS: width    = width of radiator(s),
            height   = height of radiator(s),
            no       = number of parallel radiators,
            spacecf  = portion of full space that the radiation is into,
            mutualcf = mutual coupling coefficient,
            dist     = distance from radiator(s) to observation point.

PINS:       In:      pin to be connected to the acoustic circuit,
            Out:     output pin copying the input volume velocity,
            FreeAir: pin for sensing of free-field sound pressure at distance specified by dist,
            Load:    pin that can be connected to other acoustical loads that receive the radiated sound waves.

USAGE:
This macro is a ready-made combination of air load and radiation (spherical waves) from a rectangular radiator. For more
information, see the corresponding macros.

LIMITATIONS:
See the AirLRct and SphWave macros.
```

RADCIRB.CIR: RADIATION FROM CIRCULAR RADIATOR(S)
(<< WAVELENGTH) IN FINITE BAFFLE



.PARAMETERS(diam, no, bperim, mutualcf, dist)

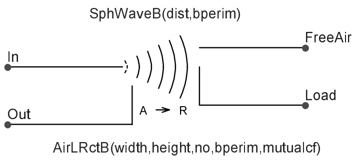
```
PARAMETERS: diam     = diameter of radiator(s),
            no       = number of parallel radiators,
            bperim   = perimeter of baffle,
            mutualcf = portion of full space that the radiation is into,
            dist     = distance from radiator(s) to observation point.

PINS:       In:      pin to be connected to the acoustic circuit,
            Out:     output pin copying the input volume velocity,
```

FreeAir: pin for sensing of free-field sound pressure at distance specified by dist,
Load: pin that can be connected to other acoustical loads that receive the radiated sound waves.

USAGE:
This macro is a ready-made combination of air load and radiation (spherical waves) from a circular radiator in a finite baffle. For more information, see the corresponding macros.

LIMITATIONS:
See the AirLCirB and SphWaveB macros.

RADRCTB.CIR: RADIATION FROM RECTANGULAR RADIATOR(S) (<< WAVELENGTH) IN FINITE BAFFLE



.PARAMETERS(width, height, no, bperim, mutualcf, dist)

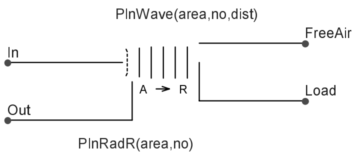
PARAMETERS: width = width of radiator(s),
height = height of radiator(s),
no = number of parallel radiators,
bperim = perimeter of baffle,
mutualcf = portion of full space that the radiation is into,
dist = distance from radiator(s) to observation point.

PINS: In: pin to be connected to the acoustic circuit,
Out: output pin copying the input volume velocity,
FreeAir: pin for sensing of free-field sound pressure at distance specified by dist,
Load: pin that can be connected to other acoustical loads that receive the radiated sound waves.

USAGE:
This macro is a ready-made combination of air load and radiation (spherical waves) from a rectangular radiator in a finite
baffle. For more information, see the corresponding macros.

LIMITATIONS:
See the AirLRctB and SphWaveB macros.

RADPLN.CIR: RADIATION OF PLANE WAVES BY RADIATOR(S)



.PARAMETERS(area, no, dist)

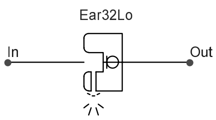
PARAMETERS: area = area of radiator(s),
no = number of parallel radiators,
dist = distance from radiator(s) to observation point.

PINS: In: pin to be connected to the acoustic circuit,
Out: output pin copying the input volume velocity,
FreeAir: pin for sensing of free-field sound pressure at distance specified by dist,
Load: pin that can be connected to other acoustical loads that receive the radiated sound waves.

USAGE:
This macro is a ready-made combination of air load and radiation (plane waves) from a plane radiator of arbitrary shape. It can also be seen as the high-frequency equivalent to the RadCir and RadRct macros. For more information, see the PlnRadR and PlnWave macros.

LIMITATIONS:
See the PlnRadR and PlnWave macros.

EAR32LOA.CIR: ITU-T P.57 TYPE 3.2 LOW-LEAK EAR SIMULATOR (WITHOUT SEPARATE LEAK PIN)



PARAMETERS: -

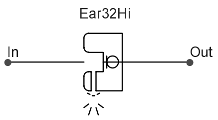
PINS: In: pressure input,

Out: internal pressure measured by microphone.

USAGE:
This macro is equivalent to an Ear32Lo macro with the leak connected to atmospheric pressure. For more information, see the Ear32Lo macro.

LIMITATIONS:
See the Ear32Lo macro.

EAR32HIA.CIR: ITU-T P.57 TYPE 3.2 HIGH-LEAK EAR SIMULATOR (WITHOUT SEPARATE LEAK PIN)



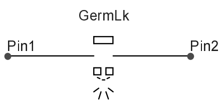
PARAMETERS: -

PINS: In: pressure input,
Out: internal pressure measured by microphone.

USAGE:
This macro is equivalent to an Ear32Hi macro with the leak connected to atmospheric pressure. For more information, see the Ear32Hi macro.

LIMITATIONS:
See the Ear32Hi macro.

GERMLKA.CIR: GERMAN LEAK RING (WITHOUT SEPARATE LEAK PIN)



PARAMETERS: -

PINS: Pin1, Pin2: connection points for Ear1 ear simulator and acoustic circuit.

USAGE:
This macro is equivalent to a GermLk macro with the leak connected to atmospheric pressure. For more information, see the GermLk macro.

LIMITATIONS:
See the GermLk macro.

Miscellaneous macros

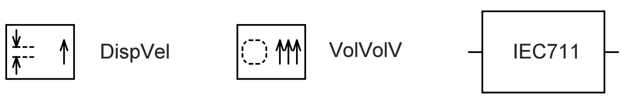


Figure IV.21: Derivators/integrators for conversion between displacement and velocity, volume and volume velocity, IEC 711 coupler (used internally by Ear32Lo and Ear32Hi macros).

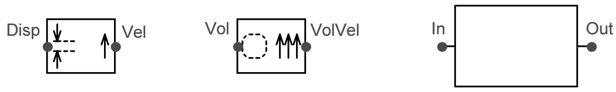
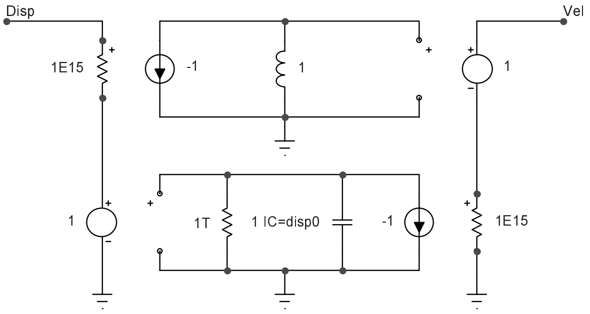


Figure IV.22: The above components with their pins shown.

DISPVEL.CIR: DISPLACEMENT-VELOCITY CONVERTER



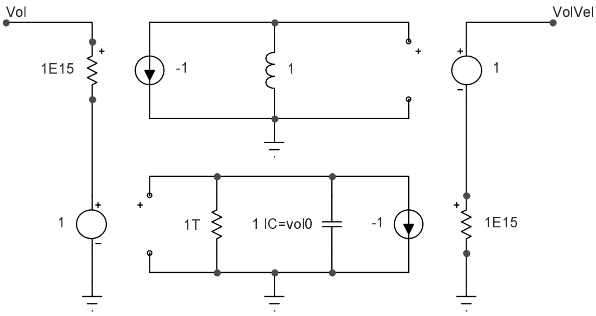
.PARAMETERS(disp0)

PARAMETER: disp0 = initial displacement.

PINS: Disp: displacement input/output,
Vel: velocity input/output.

USAGE:
This macro is an integrator/derivator used to convert between displacement and velocity (both given as voltages) in the mechanical domain.

VOLVOLV.CIR: VOLUME - VOLUME VELOCITY CONVERTER



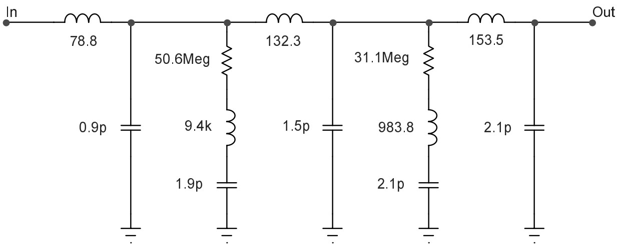
.PARAMETERS(vol0)

PARAMETER: vol0 = initial volume.

PINS: Vol: volume input/output,
VolVel: volume velocity input/output.

USAGE:
This macro is an integrator/derivator used to convert between volume and volume velocity (both given as voltages) in the acoustical domain.

IEC711.CIR: IEC711 COUPLER (USED BY ITU-T P.57 LOW-LEAK AND HIGH-LEAK EAR SIMULATOR MACROS)



PARAMETERS: -

PINS: In: pressure input,
Out: internal pressure measured by microphone.

USAGE:
This macro models the IEC711 coupler, which is attached to ITU-T P.57 low and high leak ear simulators. It is used by the macros modelling those ear simulators.

REFERENCE:

Product data: wideband ear simulator for telephonometry - type 4195. Brüel & Kjær, Nærum, Denmark, 1997.

Macros created especially for thesis simulations

These macros are used in thesis simulations just to make schematics more compact and easy to read. They are not seen as belonging to the real acoustics macro collection and are thus not included in the menu structure (see below). Definitions (schematics) are included here without further explanation because the macros are self-explanatory.

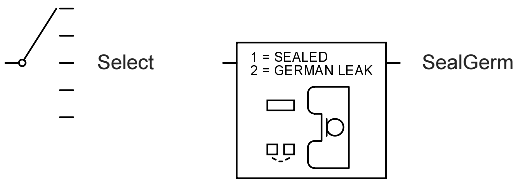


Figure IV.23: Selector (5-position switch), type 1 coupler with and without German leak ring built into one single macro.

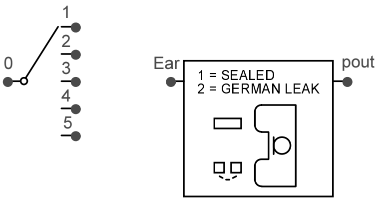
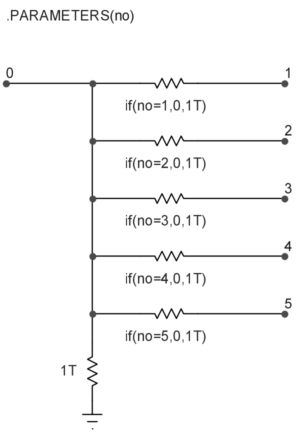
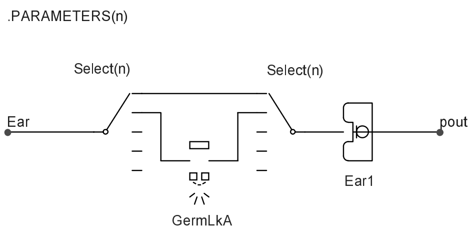


Figure IV.24: The above macros with their pins shown.



SELECT.CIR: SELECTOR

Parameter:
value (1,2,...,5 = number of pin, other values: no connection).



SEALGERM.CIR: TYPE 1 COUPLER SEALED OR WITH GERMAN LEAK RING

Unimplemented macros

The following macros were seen as equally important building blocks, but they were not implemented in this thesis due to lack of space and time, and also because they were not actually needed in any simulations. However, if the above collection of macros were to be extended, these unimplemented macros should be taken into consideration as important candidates.

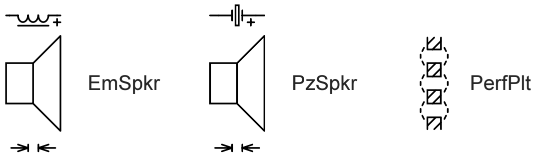


Figure IV.25: Electromagnetic and piezoelectric loudspeaker, perforated plate, circular and ractangular duct with losses.

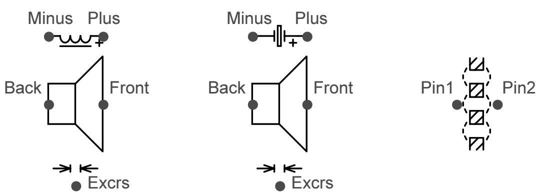


Figure IV.26: The above components with their pins shown.

Other unimplemented macros (whose shapes have yet to be designed):

- mechanical suspension (combination of compliance, resistance and possibly mass),
- equalizer for type 3.2 low-leak and high-leak ear simulators,
- lever (transformer in the mechanical domain).

Menu structure

The following menu structure (containing all the useful macros) was created inside the Micro Cap ”Component” menu (groups in regular print, macro names in bold print):

Acoustics

- Electrical domain
  - Probes
    - **CurrPrb** (current probe)
    - **EImpPrb** (electric impedance probe)
- Mechanical domain
  - Ideal components
    - **Zm** (mechanical impedance)
    - **Rm** (mechanical resistance)
    - **Lm** (mechanical inductance, mass)
    - **Cm** (mechanical capacitance, compliance)
    - **Force** (force source)
    - **Velocity** (velocity source)
    - **FixedPt** (immovable point, mechanical ground)
  - Probes
    - **ForcePrb** (force probe)
    - **VelPrb** (velocity probe)
    - **DispPrb** (displacement probe)
    - **MImpPrb** (mechanical impedance probe)
  - Miscellaneous
    - **DispVel** (displacement-velocity converter)
- Acoustical domain
  - Ideal components
    - **Za** (acoustic impedance)
    - **Ra** (acoustic resistance)
    - **La** (acoustic inductance, inertance)
    - **Ca** (acoustic capacitance)
    - **Pressure** (pressure source)
    - **VolVel** (volume velocity source)
    - **AtmPress** (atmospheric pressure, acoustical ground)
  - Sound
    - **SndPress** (sound pressure source)
  - Structures
    - Compound
      - **CirHoleA** (AirLCir + CirHole + AirLCir)
      - **RctHoleA** (AirLRct + RctHole + AirLRct)
      - **CirHoleE** (EndCCir + CirHole + EndCCir)
      - **RctHoleE** (EndCRct + RctHole + EndCRct)
    - Cavities
      - **Cavity** (air cavity)
      - **CavityLs** (air cavity with losses)
    - Tubes
      - **CirHole** (circular hole(s))
      - **RctHole** (rectangular hole(s))
      - **CirDuct** (circular duct(s))
      - **RctDuct** (rectangular duct(s))
    - Membranes
      - **Membrn1** (rigid permeable membrane(s))
      - **Membrn2** (permeable membrane(s))
    - **HelmhRes** (Helmholtz resonator)
  - Air load
    - General

- **EndCCir** (end correction for circular radiator(s))
- **EndCRct** (end correction for rectangular radiator(s))

Plane
- **PlnRadR** (rad. resist. for radiator(s) radiating plane waves)

Spherical
- **AirLCir** (air load on circular radiator(s))
- **AirLCirB** (air load on circ. radiator(s) with finite baffle)
- **AirLRct** (air load on rectangular radiator(s))
- **AirLRctB** (air load on rect. radiator(s) with finite baffle)
- **SphRadR** (rad. resist. for radiator(s) radiating sph. waves)

Probes
- **PressPrb** (pressure probe)
- **VolVPrb** (volume velocity probe)
- **VolPrb** (volume probe)
- **AImpPrb** (acoustic impedance probe)

Miscellaneous
- **VolVolV** (volume - volume velocity converter)

Domain interfaces

General
- **IdealTrf** (ideal transformer)
- **IdealGyr** (ideal gyrator)

Electrical-mechanical
- **EMIntDyn** (elec.-mech. dom. interf. for dynamic transduction)
- **EMIntEs** (elec.-mech. dom. interf. for electrostatic transduction)

Mechanical-acoustical
- **MAInt** (mechanical-acoustical domain interface)

Radiation

Acoustical-radiation

Plane
- **PlnWave** (plane wave radiation)

Spherical
- **SphWave** (spherical wave radiation)
- **SphWaveB** (spherical wave rad., with finite baffle)

Radiation-acoustical
- **RAInt** (radiation-acoustical domain interface)
- **RAIntB** (rad.-acoust. domain interface, with finite baffle)

Compound
- **RadCir** (AirLCir + SphWave)
- **RadCirB** (AirLCirB + SphWaveB)
- **RadRct** (AirLRct + SphWave)
- **RadRctB** (AirLRctB + SphWaveB)
- **RadPln** (PlnRadR + PlnWave)

Transducers
- **DynSpkr1** (basic dynamic loudspeaker)
- **DynSpkr2** (basic dynamic loudspeaker with Thiele-Small parameters)
- **ElctMic1** (basic electret microphone)
- **ElctMic2** (basic electret microphone, pressure-gradient type)

Measuring equipment

Compound
- **Ear32LoA** (ITU-T P.57 type 3.2 low-leak ear sim., without leak pin)
- **Ear32HiA** (ITU-T P.57 type 3.2 high-leak ear sim., without leak pin)

GermLkA (leak adapter, ”German leak ring”, without leak pin)

Ear1 (ITU-T P.57 type 1 ear simulator)

Ear32Lo (ITU-T P.57 type 3.2 low-leak ear simulator)

Ear32Hi (ITU-T P.57 type 3.2 high-leak ear simulator)

GermLk (leak adapter, ”German leak ring”)

Appendix V Models of earpiece capsule

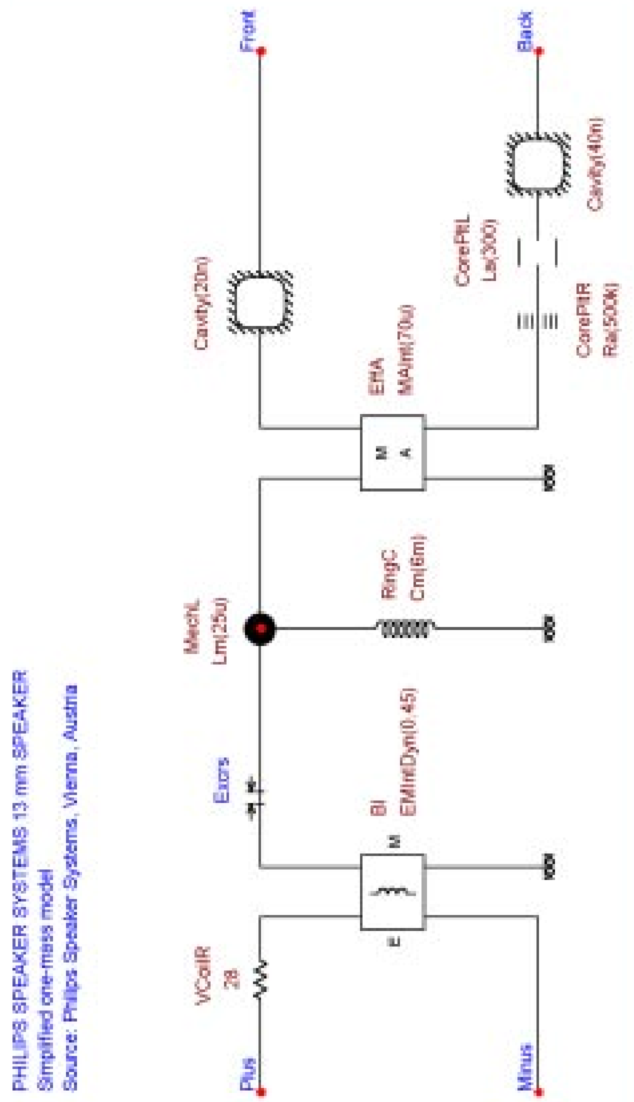


Figure V.1: One-mass model of the Philips Speaker Systems (PSS) 13 mm leak-tolerant earpiece capsule. All mechanical structures (e.g. the small air cavity under the membrane) are not modelled.

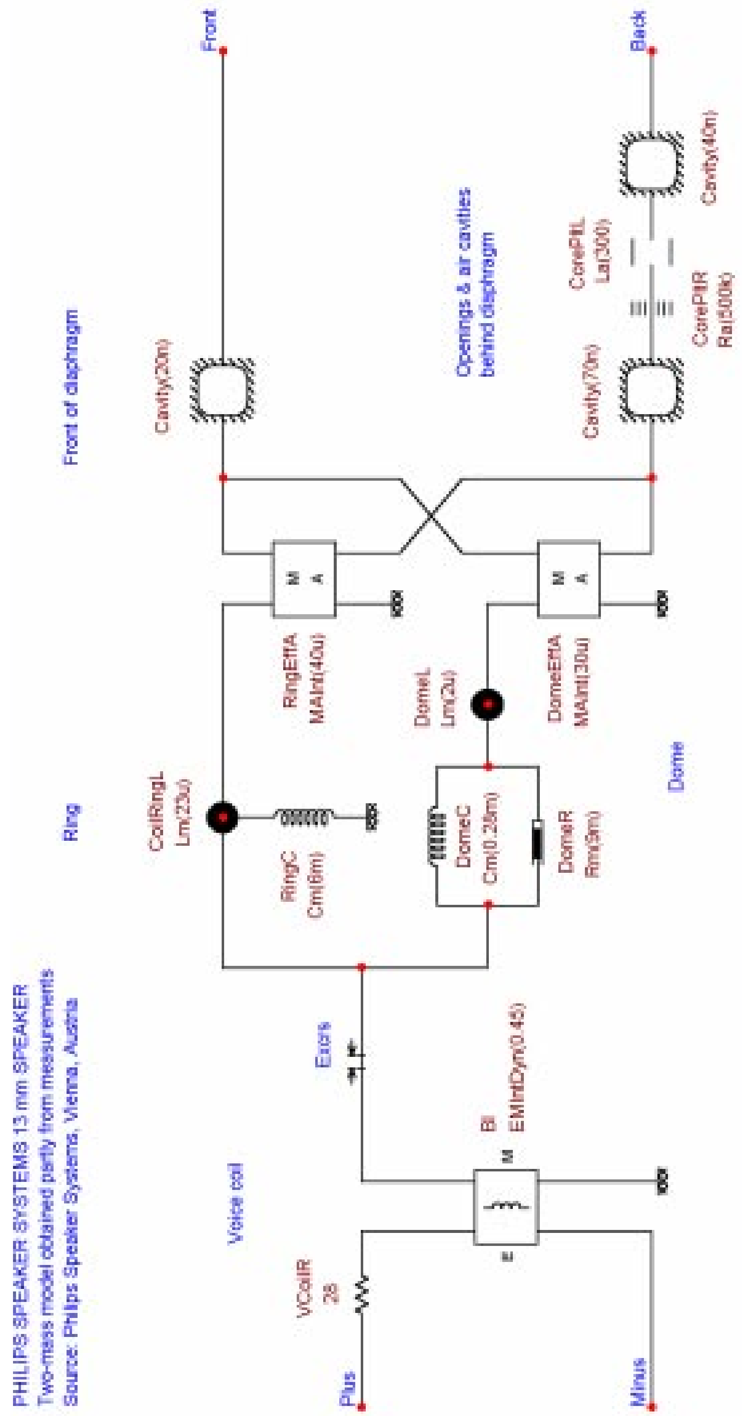


Figure V.2: Two-mass version of the above model, modelling also the decoupling of the dome (center part of the membrane). The model includes the small air cavity under the membrane left out in the simplified one-mass model.

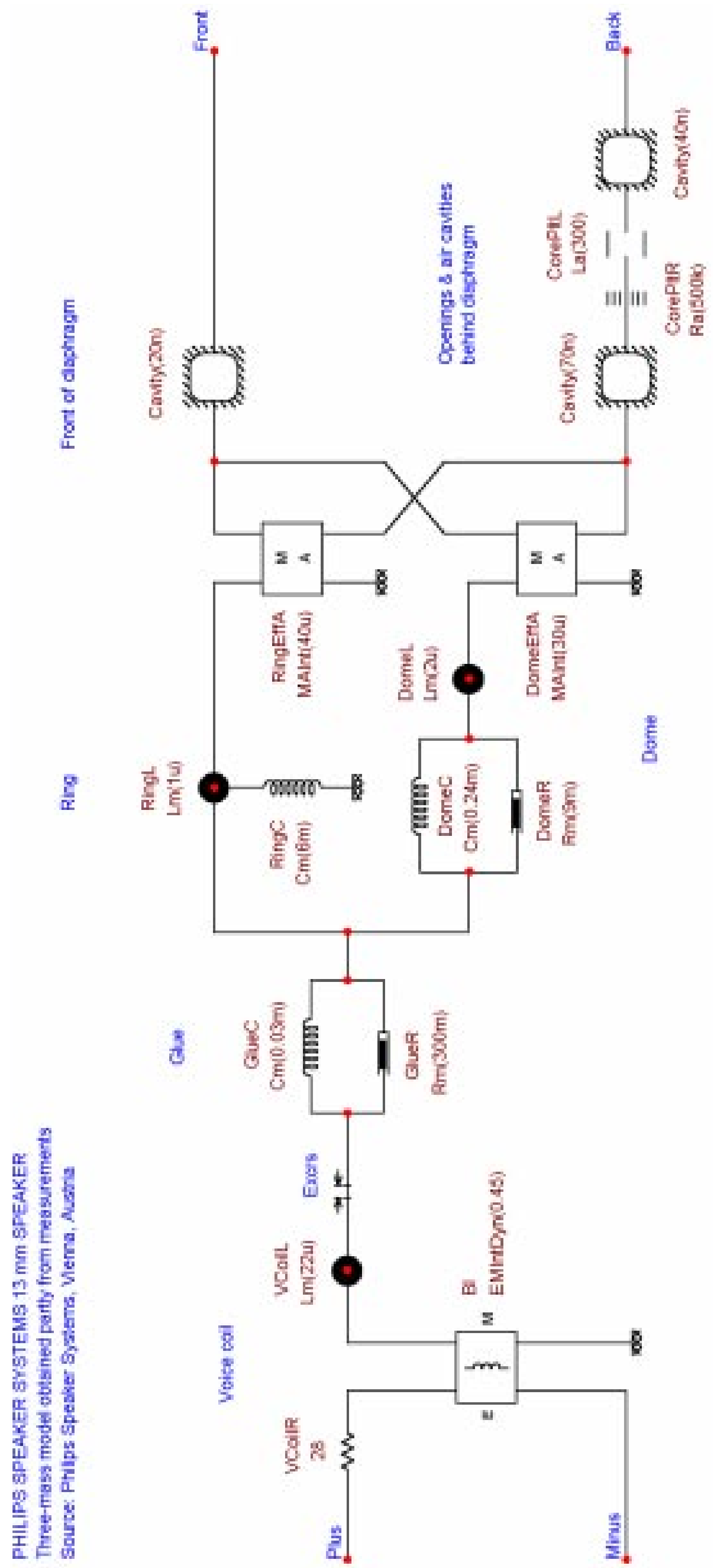


Figure V.3: Three-mass version of the above model, separating the voice coil and the membrane into two masses interconnected by the glue between them.

Appendix VI Measurement setup

The earpiece measurements contained in this thesis were made using a Brüel & Kjær 4185 acoustic coupler with a type 4134 ½ ” microphone and a 2669 preamplifier.

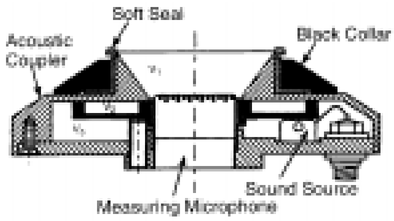


Figure VI.1: Brüel & Kjær 4185 acoustic coupler, corresponding to ITU-T P.57 specified type 1 coupler (the soft seal/ black collar part is removable) (from [17]*). The internal sound source is just for verifying that the coupler is well sealed before a measurement begins.*

Although the coupler is equipped with a fairly good seal, it was necessary to use added sealing material (Blu-Tack™) to prevent any leaks between the coupler and the phone cover during ”sealed” measurements, because of the curved surface of the Nokia 6110 front cover. Moreover, the joint between the cover and the display lens in the phone was sealed separately to eliminate any leaks other than those deliberately built into the earpiece. In ”German leak” measurements the soft seal/ black collar part was removed and the German leak ring inserted in its place. Blu-Tack™ was used in German leak measurements also to eliminate leaks between the phone cover and the rubber seal of the leak ring.

A sinusoidal voltage of 100 mV peak amplitude was brought directly to the earpiece capsule terminals in frequency response measurements. Unless mentioned otherwise, all speech processing and/or filtering circuitry in the phone was bypassed.

Frequency response measurements were made using an Audio Precision System Two 2322 analyzer (as combined generator and analyzer). The sinusoidal driving signal from the Audio Precision analyzer was routed to the earpiece terminals through a Rotel RB-991 power amplifier, with feedback to the Audio Precision analyzer to maintain the desired level of the driving signal. The microphone was connected to the Audio Precision analyzer via a Brüel & Kjær type 2610 measuring amplifier. The microphone, coupler and phone under test were located in an anechoic chamber during all measurements.

A separate receiving frequency response measurement result through the whole transmission path was included also to show the final response. This measurement used a Hewlett-Packard 8922 GSM MS test set, Brüel & Kjær 2144 dual channel real time frequency analyzer, Brüel & Kjær WQ 1105 power amplifier, Brüel & Kjær 4227 artificial mouth, Brüel & Kjær 6712 GSM test software, and a Brüel & Kjær 4905 test head. A call was established to the GSM MS test set, the phone was mounted in the LRGP (loudness rating guard-ring position) in the test head together with the artificial mouth, and the receiving frequency response was measured by the software through the frequency analyzer using standard artificial speech. The test head, artificial mouth and phone under test were located in an anechoic chamber during the measurement.