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Computer based education of acoustical measurements

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ABSTRACT OF THE MASTER'S THESIS

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The thesis deals with computer based education (or computer assisted learning) and applying it to teaching of acoustics. The main focus is in acoustical measurements and especially in loudspeaker measurements. Different methods and possibilities of using computers to aid teaching are discussed. Education programs bring several demands on computer technology and certain functionality or behaviour may be required from the education programs themselves. This may seem transparent from an outsider or student's point of view, but can create challenging problems for the author of an education program. Compared to traditional teaching methods certain advantages can be achieved with computer based education, e.g., better interactivity and student's freedom of deciding studying pace.

As a part of this thesis a program for teaching acoustical measurements named *A Simple Loudspeaker Measurement Program* has been developed. The program was written in the Lisp programming language to be used in the QuickSig digital signal processing environment. This program is described thoroughly and its properties, strengths and weaknesses are discussed along with writing such a program. This program takes advantage of so called "learning-by-doing" method and thus deviates from typical education programs. *A Simple Loudspeaker Measurement Program* has been used in teaching of the course *S-89.112 Acoustics and audio signal processing: special assignment.* Approximately 40 students have used it in the springs of 1998 and 1999. The enthusiasm and learning of the students has been compared to earlier teaching by traditional methods. Feedback from the students has been gathered and analysed.

The usage of the Lisp programming language and the advantages of the QuickSig signal processing environment in writing education programs are discussed briefly. Different kind of measurement signals and stimuli in different types of loudspeaker measurements are presented.

Keywords: computer based education, computer assisted learning, Lisp programming language, user interfaces, QuickSig, digital signal processing

TEKNILLINEN KORKEAKOULU
DIPLOMITYÖN TIIVISTELMÄ

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Diplomityö käsittelee tietokoneavusteista opetusta ja sen soveltamista akustiikassa. Pääpaino on akustisten mittausten opetuksessa ja erityisesti kaiutinmittauksissa. Työssä pohdiskellaan tietokoneiden erilaisia käyttötapoja opetuksen apuvälineenä ja tietokoneiden tuomia lisämahdollisuuksia. Opetusohjelmat asettavat tietokoneelle monia vaatimuksia ja voivat tuoda mukanaan vaikeita ongelmia. Itse opetusohjelmilta vaaditaan tietynlaista toimintaa ja käyttäytymistä, mikä voi tuntua itsestään selvältä ulkopuolisesta henkilöstä tai opiskelijasta, mutta voi tuottaa ohjelman laatijalle suuriakin vaikeuksia. Verrattuna tavanomaiseen opetukseen tietokoneavusteisella opetuksella voidaan saavuttaa tiettyjä etuja, mm. parempi vuorovaikutteisuus ja opiskelijan vapaus päättää etenemisnopeudesta.

Työn yhtenä osana on Lisp-ohjelmointikielellä QuickSig-signaalinkäsittely-ympäristöön toteutettu akustisten mittausten opetusohjelma *A Simple Loudspeaker Measurement Program.* Tämä ohjelma käydään läpi yksityiskohtaisesti ja pohditaan sen eri ominaisuuksia, vahvuuksia ja heikkouksia ja opetusohjelman ohjelmointia. Ohjelma käyttää opetuksessa niin sanottua "oppiminen tekemällä" -periaatetta ja siten poikkeaa jonkin verran tyypillisistä opetusohjelmista. Ohjelmaa on käytetty apuna kurssin *S-89.112 Akustiikan ja äänenkäsittelyn työt* opetuksessa. Sitä on käyttänyt yhteensä noin 40 opiskelijaa keväällä 1997 ja 1998. Opiskelijoiden innokkuutta ja oppimista on verrattu aikaisempaan, perinteisin menetelmin toteutettuun opetukseen. Opiskelijoilta on myös kerätty ja analysoitu palaute opetusohjelmasta ja kurssin suoritustavasta sekä sen mielekkyydestä.

Lisp-ohjelmointikielen käyttöä ja QuickSig-ympäristön etuja ja mahdollisuuksia opetusohjelmien laatimisessa on läpikäyty lyhyesti. Erilaisten mittasignaalien ja herätteiden käyttö ja merkitys erityyppisissä kaiutin-mittauksissa on esitelty.

Avainsanat: tietokoneavusteinen opetus, Lisp-ohjelmointikieli, akustiset mittaukset, käyttöliittymät, QuickSig, digitaalinen signaalinkäsittely

PREFACE

This master's thesis was carried out at the Laboratory of Acoustics and Audio Signal Processing at the Helsinki University of Technology. I want to thank all the laboratory personnel for outstanding working atmosphere and good spirit. I owe the most for Matti Karjalainen, the supervisor of this thesis, and Martti Rahkila, the instructor of this thesis. They have guided me and given invaluable information and help. I also wish to thank Toomas Altosaar for his advice in Lisp programming and the laboratory secretary Lea Söderman for keeping the laboratory in discipline and always standing on our side against the bureaucracy.

Espoo, June 14, 1999

Ila Tokola

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LIST OF ABBREVIATIONS

A/D	analogue-to-digital converter
AI	artificial intelligence
CAL	computer aided/assisted learning
CBE	computer based education
CD	compact disc
CD-R	CD-recorder, CD-recordable
CLOS	Common Lisp Object System
D/A	digital-to-analogue converter
DAT	digital audio tape
DFT	discrete Fourier transform
DSP	digital signal processing
DTFT	discrete-time Fourier transform
FFT	fast Fourier transform
GUI	graphical user interface
IC	integrated circuit
IBM	International Business Machines Corporation
IFFT	inverse FFT
LSI	large-scale integration
LTI	linear and time-invariant (system)
MCL	Macintosh Common Lisp
MLS	maximum length sequence
PC	personal computer
RAM	random access memory
RMS	root mean square
RPFS	random-phase flat spectrum
S/N	signal-to-noise ratio
SPL	sound pressure level
VLSI	very large-scale integration
WWW	world wide web

LIST OF SYMBOLS

$\delta(t)$	Dirac's delta function (ideal impulse)
$\phi(f)$	phase factor of frequency response function
$\varphi(f)$	phase shift, $\varphi(f) = \phi(f)$
ω	angular frequency $\omega = 2\pi f$
ω_0	angular frequency in discrete domain
f	frequency
f_l	low frequency limit
f_s	sampling frequency
$H(f)$	frequency response function
$H_I(f)$	imaginary part of frequency response function
$H_R(f)$	real part of frequency response function
$\lvert H(f) \rvert$	absolute value of frequency response function, i.e., gain factor
$H(z)$	z-transform of the impulse response of a discrete system
$h(n)$	impulse response of a discrete system
$h(t)$	impulse response of a system in time domain
$h(\tau)$	weighting function
I	current
L	length of a discrete time sequence
l	index variable, integer
N	sequence length
n	index variable, integer
P	power
R	resistance
T	(sampling) period
t	time
t_w	length of time window
U	voltage
$X(f)$	input signal of a system in frequency domain
$x(t)$	input signal of a system in time domain
$Y(f)$	output signal of a system in frequency domain
$y(t)$	output signal of a system in time domain
z	continuous variable in complex domain

1 INTRODUCTION

1.1 Education

Lecturers and vast auditoriums are probably the first things in mind when thinking of education. We can imagine chalkboards, slide projectors or overhead projector with an experienced lecturer teaching to younger students his or her expertise. The students are listening, some of them are making notes, and some of them trust to get copies from the others or simply relay on written books as a source for information. This is the classical situation in schools and universities everywhere.

However, not everything is best learned this way. There are many areas of knowledge that can't be learnt by merely listening lecturers. Some things may be too complex for the students to understand just by listening. They need to examine the different prospects by themselves and read the books in private and let the information absorb and settle. Some things might be so hard to understand correctly that students need to ask clarifying questions from the teacher when applying the theory in different kind of problems. This usually brings the need for smaller groups and more interplay between the teacher and the students.

There are also practical skills which can only be learnt well by trying and doing. This method is called "learning-by-doing". Sometimes there has to be an instructor, sometimes the student can practise alone. Sometimes one instructor can guide a bigger group of students, while in some cases the student may need a personal instructor. Sometimes a human teacher is required, while in some cases a machine teacher might be a better option.

There are many different types of education, many different methods of teaching and as many methods of learning. It is fair to say that life itself is some kind of education, whatever we do. And we must remember that we are all individuals, everyone has his or her best way of learning and this best way varies according to subject and situation.

1.2 Computers within education

We don't think of computers first when we think of learning and education. However, computers can be well used to aid education. There are different ways to obtain the advantage of utilising computers. We can use the graphical capabilities of computers to demonstrate, e.g., curves of mathematical functions. We can use the sound producing capabilities to provide samples from different musical instruments, we can process our information with computers in many different ways or we can make notes with computer. It is possible to prepare our teaching material, e.g., transparencies with computer or we can merely use the computer screen for reading text, like from the book. There are many cases when we are using computers in education that we don't even think the computer as a means of education.

All the ways of using computer mentioned above can be built into one program to make the studying more appealing. Then, with the help of modern fast and well-equipped computers the skilled programmer can even write code that is acting like a teacher. This is the most advanced way of using computers in education, programming them to teach humans.

There are advantages and disadvantages of using computers as a teacher and in several areas the computers can't be used, at least yet. Computers still can't match humans as a teacher, but they can be programmed to imitate a human teacher and in some areas they can even perform better than a real living teacher. In fact, sometimes computers shouldn't imitate human lecturers; it may be an advantage to act differently from the humans. In the following chapters I discuss this subject further and take a look on how computers were used in training the students to make acoustical measurements and understand them.

1.3 Goals of the loudspeaker measurement program

The course *S-89.112 Acoustics and Audio Signal Processing: special assignment*, is a self-studying course from its nature. There is a brief tutorial measurement to familiarise the students with acoustical measurements. Then the main part of the course consists of a research project, e.g., programming, literature work or making more demanding acoustical measurements. The goal of the tutorial work is to provide students the basic knowledge of making acoustical measurements and to better understand the results from measurements made by others. In short, its goal is to give basis to successfully complete the main part of the course. *The Simple Loudspeaker Measurement Program* was developed for the needs of a tutorial work.

Teaching acoustical measurements in laboratory conditions is just the kind of education that requires a single student or a very small group of students at a time and a capable instructor for the students. Students need not to learn a vast amount of difficult theory quickly or to understand complex mathematical problems, the target is to teach some simple basic rules when measuring, provide explanations for the most common phenomena, learn how to use the necessary equipment, and to point out the usual mistakes.

From the teacher's point of view it can be very frustrating to go through these straightforward routines time after time, especially if the students aren't paying any attention and are only interested in getting away as soon as possible and trusting that the teacher takes care of proper handling of equipment. Besides, teaching the same things again and again for different students is taking a lot of time away from the teacher. As the financial resources for the course are limited, it would be the best to use this money well and where it is needed, like tutoring the main project of the course better.

When the students sense the frustrated and bored instructor they automatically become less motivated. And if they know that this instructor takes care of measurements and just demonstrates how the measurements are made without letting the students measure by themselves, students become also passive. Unmotivated and passive, the learning isn't effective. Of course the above isn't the case with all the students, it's the worst case scenario.

The above situation in mind arose the question what could be done to make the introductory work more appealing for the students, provide better education and release resources for more important parts of the course while keeping the costs low. This led to the idea to try writing a computer based tutoring program to teach the students well while setting the teacher free for other studies. At the same time the suitability of computer based education in acoustics could be studied more closely. The task was not small one but some good results were achieved while some areas could still be improved.

2 COMPUTER BASED EDUCATION

2.1 The history of computer based education

Using computers as a means of education has interested programmers and teachers since the first computers were build. First programs and ideas weren't even close to today's possibilities. But the general ideas and basics were starting to develop with the first computers. A good historical perspective can be found e.g. from the book *Computer-Based Instruction, Methods and Development* (Alessi 1985).

2.1.1 Hardware

The earliest calculating machine was probably the abacus. At one time, use of the abacus was widespread around the world. All four of the basic arithmetic operations can be performed with abacus; addition and subtraction at great speed. In the Eastern world abacus is still found in many stores.

A major advance in the automation of calculation came from the invention of logarithms by John Napier in the early years of the seventeenth century. Once the table of logarithms had been initially calculated, it was simple to do even long and difficult multiplications and divisions. This invention was soon embodied into a physical device, which allowed fast calculations, even though the accuracy was not high. This was the slide rule, which was the standard method of calculation of scientists and engineers until the advent of the hand-help calculator in the early 1970s.

After Napier's invention several people designed and produced mechanical calculating machines with varying degrees of success. The best known of these was Charles Babbage who, although never successful in producing a completed version of one of his machines, was very influential in the design of later ones. The first of Babbage's calculating machines was called the difference machine. Despite the setbacks in the difference machine, Babbage was undaunted and designed the analytical engine; it can be called the prototype of the computer because not only could it perform the basic mathematical operations, but it also had separate devices for entering information, overseeing the calculations, storing numbers, and providing output—the four basic features of a computer. Babbage never built the analytical engine, although his son produced a model of it in 1910.

The next major advance came about as the result of a competition held for the 1890 census in the United States. As a result of the competition, a device build by Herman Hollerith was used to tally all the census figures. After the success in the census, Hollerith founded the Tabulating Machine Company, which later merged with other companies to become the International Business Machines Corporation, IBM.

Just before World War II, Thomas Watson, head of IBM, funded Howard Aiken, a mathematics professor at Harvard, to construct an electromechanical equivalent of Babbage's analytical machine. This machine, known as the Mark 1, cost nearly one million dollars and was unveiled in 1944.

Credit for producing the first computer, in contrast to a calculating machine, is generally accorded to John Atanasoff, a physicist at Iowa State University who, together with an assistant, Clifford Berry, produced the ABC, or Atanasoff-Berry-Computer. Based on Atanasoff's ideas, John Mauchly and Presper Eckert of the University of Pennsylvania produced one of the most influential computers—the ENIAC—or Electronic Numerical Integrator and Calculator.

The ENIAC, which was eventually completed after the end of the war, was a massive machine consisting of 18000 vacuum tubes and 1500 relays. ENIAC performed in half a minute the trajectory calculations of a missile that took the average person about 20 hours. However the limitations of the ENIAC became obvious very quickly, particularly the fact that to change its function required rewiring a substantial part of the machine, which was a slow process. The solution to this problem is usually attributed to John von Neumann. The solution was the stored-program concept, in which the sequence of instructions to be performed (called a program) could be entered in much the same way as data and stored in the computer to be used when needed.

Mauchly and Eckert continued their pioneering efforts and produced their own stored-program computer, The EDVAC, in 1952. Shortly thereafter, they formed a company and sold their first computer, UNIVAC 1. The UNIVAC 1 was the first of what have come to be called first-generation machines.

The era of the first-generation machine ended in 1959 with the replacement of vacuum tubes by transistors. Second-generation computers used transistors rather than tubes, thus increasing their reliability and computational speed, and reducing their volume, cost, and power consumption.

Third-generation computers appeared only about five years after the introduction of transistor-based computers. The invention that brought this about was the integrated circuit, or IC. Integrated circuits allow many components to exist on a single small chip. The technique was enhanced in 1970 by a process called large-scale integration (LSI), in which the density of components on a chip was increased even more. In 1975 the process was taken yet another step forward with the introduction of very large-scale integration (VLSI). One of the most important products developed from the integrated technology was the microprocessor.

2.1.2 Applications

The second-generation computers also started to incorporate advances in the way programs were written. Previously, all programs had been written in the language understood by the machine, appropriately called machine code or machine language. This comprised strings of numbers, 0's and 1's, which were virtually incomprehensible to humans. To improve programming, new languages were invented, such as assembly languages and FORTRAN. These programming languages have allowed more people to become programmers and increased the number of people using computers.

The first use of computers by educational institutions coincided approximately with the introduction of second-generation computers at the end of the 1950's. The PLATO project at the University of Illinois, which began in 1960 with the goal of designing a large computer based system for instruction, was one of the first educational projects with computers. Soon after, IBM introduced Coursewriter, a programming language designed for preparing instructional materials on IBM's large computers.

In 1972 the MITRE Corporation and Brigham Young University started development of the TICCIT (Time-shared Interactive Computer Controlled Instructional Television) system. With TICCIT, students study lessons presented on standard colour televisions and interact through modified typewriter keyboards, all of which are controlled by a microcomputer.

In the early 1970s the PLATO project introduced PLATO IV, a large time-shared instructional system. Students study on individual terminals, hundreds of which are connected to a large computer on which all lessons and student data are stored. All program execution occurs on the main computer, which communicates with the terminals by telephone line. PLATO IV allows up to 600 students to use the computer simultaneously. In addition, it allows authors to develop instructional materials at the same time that students are studying lessons.

A different approach to these university projects was taken by Minnesota Educational Computer Consortium (MECC). From its inception in 1972, MECC was oriented toward putting computers and related facilities in the public schools. But MECC, like all the projects, used large and relatively expensive computers because that was all that was available at the time.

In the mid-1970s a few small computer companies began experimenting with microcomputers. Unlike large and medium-sized computers, they could be used by only one person at a time (a distinction no longer true today), and were thus referred to as dedicated computers.

None of these early microcomputers were very successful. However, in 1977 three microcomputers were introduced that proved to be successful. Two large corporations, Radio Shack and Commodore Business Machines, introduced the TRS-80 and PET computer, respectively. And a completely unknown company—Apple Computer—introduced its own computer of the same name.

The introduction of these microcomputers ushered in the microcomputer revolution. Unlike previous microcomputers, these were complete computer systems with all necessary input, output, memory, processing, and permanent storage. They were more reliable and much easier to use than previous microcomputers. Furthermore, all three were well marketed, the unknown Apple Computer best of all.

The first microcomputers started the time of rapid development. In 1981 IBM released the PC, or Personal Computer, which became a great success. Most of modern computers are successors of the PC or its clones. Today's personal computers are based on the same concept, only the speed has increased, memory and disk space are larger and other peripheral equipment have developed a lot further. As a comparison, the first IBM PC had 16 kilobytes of memory, in 1999 a normal PC has typically 128 megabytes of memory. The difference is a huge one.

The success of personal computers has caused the developers of the CBE programs to concentrate on programs which can be used only a by one person or small group of persons simultaneously on one computer. But the personal computers are so cheap and widespread that there can be many copies of the program on different computers running simultaneously.

2.2 Advantages and demands of computers in education

According to Lifländer (Lifländer 1989) the most important task is to analyse how the student benefits if computers are to be used in education. Computers shouldn’t be used at all if they cause difficulties for the students and slow down the learning process. The student has to be regarded as the centre of the process and his or her learning the main target. The quality of teaching shouldn't be anything less than it is with human teachers. Other reasons for using computers can be found and I return to them later, but they are a question of secondary importance, the student’s benefit should always come first.

When thinking the situation from the student’s point of view, a couple of advantages can be immediately found. Usually the computer can be accessed anytime. This is not the case with lecturers who are usually very busy. The student can plan his or her own timetable and is not forced to act according to other people’s plans. The student can also pick up his or her own speed, human teachers can sometimes advance too quickly or too slowly for a certain student, whereas computers are patient and tireless and their speed can be adjusted to serve the student.

What is difference of using computers compared to books then? Things mentioned above are reachable with books as well. Here step in the talents of the programmer and the designer of the education program. Modern computers offer several possibilities beyond the capabilities of books. The most obvious ones are sound and animations. Sound can be very effective when used properly; it can greatly increase the attractiveness of the program and thicken the atmosphere (Gaver 1989, Gaver 1991). In acoustics, sound is the most important way to present certain things efficiently and understandably. For example filtering may be difficult to relate to the real world by studying plots of signals. By listening them the student can immediately understand the difference between the original and filtered signal.

Animations are very good for presenting cause and effect relationships and thus they can clarify different situations in the way that is impossible with words. Besides, well-designed animations are fun to watch and students remember easier the things the animations teach because they can relate them to something memorable. But sound and different kinds of animations are only means, which are built-in properties of modern computers. Perhaps the most important benefit with computers is interactivity. Interactivity is a fashionable word of the late nineties, but it is still the best advantage of computers over books. Interactivity means that the user can interact with the thing he or she is studying. The program reacts differently, e.g., to different answers from the student and the student can make the program act according to student's desires. It is up to the programmer how much intelligence and influence between the user and the program he or she is able to program. At it’s best, a good program can be compared to a personal human teacher.

All this requires certain properties from the computer. It must be fast enough to handle complex programs with adequate speed to keep up the interest of the user. It has to be able to handle graphics to make the program visually stimulating. And it must have good enough sound handling capabilities to reproduce all the sounds that increase the cosiness of the program. After this the responsibility of the fluent teaching is passed to the programmer of the application. Nowadays, even most of the personal computers are fast enough to fulfil the requirements from the computer and the speed and capacity of computers is growing rapidly, and the tendency for even faster and fancier computers will go on in the future.

As stated earlier, there are other advantages of using computers in education. From the teacher's point, using CBE could mean saving valuable time and resources. Many times the lecturers may be busy with other things or they may have limited amount of time for educating students. Releasing some issues for CBE the lecturer may be able to use more time with more difficult and demanding teaching issues. Also the students may complete courses faster, if they study faster alone with the computer than spending time in the lectures.

If CBE means less need for teaching resources, this could mean saving money. Computers don't cost that much anymore, while an expert lecturer can charge very expensively for his or her time. This statement is controversial though, as the development of education programs can be expensive as described in chapter 2.3.

User interface is a very important part of the computer and plays an important role in computer based education. In 1990's the most common solution is a keyboard, a mouse and colour monitor. This probably isn't the case in the future. We may think that current user interfaces we are nowadays using are ergonomic and good. Buxton (Buxton 1986) however plays with the situation when somewhere in the far future all the knowledge about our civilisation has been lost. Then, if some intelligent form of life would discover a working computer of 1980s, what would it think? Buxton's guess is that humans would be pictured as having a well-developed eye, a long right arm, a small left arm, uniform length fingers and a low-fidelity ear. And the dominating characteristics would be the prevalence of our visual system over our poorly developed manual dexterity.

Obviously these conclusions do not accurately describe humans of the twentieth century. But they would be perfectly warranted based on the available information. Today's systems have severe shortcomings when it comes to matching the physical characteristics of their operators. When compared to other human-operated machinery, e.g. automobiles or aeroplanes, today's computer systems make extremely poor use of the potential of the human's sensory and motor systems. The controls on the average user's shower are probably better human-engineered than those of the computer on which far more time is spent.

In the future, the user interfaces will probably develop more natural, easier to learn, easier to use, and less prone to error if we pay more attention to human body physics and communication. Computer based education will benefit from this undoubtedly. User interfaces mean so much for the ease of use of computers that they have to develop better. Especially the interaction by sound with the computer doesn't exist today or is single sided. We can hear the sounds from the computer, but the computer doesn't perceive our words.

2.3 Disadvantages of computer based education

Computer based education isn't merely bed of roses. It is easy to describe how the programs should behave, what kind of things the computer can take of and how capable the computer is in the paper. It is completely different thing to make this happen.

Creating a good CBE program requires a very skilled programmer or several of them. Besides knowing programming, the programmer needs to be familiar with the topic that is being programmed. This may require some groundwork from the programmer to familiarise himself or herself with the subject. Also the needs of the student need to be studied and understood.

The programs of today are vast; they have a huge amount of code. All this code needs to be created at first and then checked. This is a big task requiring a lot of work. There's still a possibility that all the errors haven't been discovered before the students are using the program. What happens if the program crashes in the hands of a student? Humans usually don't crash, but programs and computers may easily do this. Students lose their interest fast if the program crashes often.

Then we need to remember that when writing code for the experts, they know how to use it and what to do with it. They usually know also what it can't do. But the students act very differently. They want to find out. "What happens if I press this and do that? If I put this here and change that?" The nature of learning is trying out different things and to clarify how things work. This probably means doing errors as well. The CBE program has to stand up all this. It's one of the basic requirements from the educational program.

2.4 Computers in acoustics

In acoustics, the computer is inevitable instrument in any case, so it is only natural to program them to help in education. Typically acoustical and digital signal processing applications require as much number crushing power as there is available. Acoustical applications also demand that the computer must have proper sound reproduction capabilities. This means that we already have a suitable hardware basis for educational purposes, but we need suitable software.

Today it is common that computers have a sound card of proper quality installed as a "part of the package." Many motherboards even have an integrated sound card. Computers and sound cards have several kinds of connectors included. Connecting external audio hardware, like amplifiers, headphones, loudspeakers or microphones, is thus made easy. Professional quality signal processing and sound cards are also available. They can greatly increase the computational power available for acoustics and increase the sound quality needed for professional use. Though they also increase the cost of the computer hardware, this isn't too much in most cases.

What is still needed is a mass storage media for sound. High quality sound files require quite a lot of disk space. Fortunately the prices of hard disks have come down while the capacity of hard disks has increased. Digital Audio Tape recorders (DAT) can also be connected to computers and DAT recorders provide a decent storage media for sound. CD-ROMs are also included in almost every computer sold today, though they are not usually used for direct recording of measurement data, more like archiving it. These pieces of equipment make it possible to take full advantage using computers in acoustics. Different types of coding or packing methods are also reducing the need for disk space. In the future these coding methods will probably be developed further and help with the problem of audio data occupying considerable amount of disk space.

3 ACOUSTICAL MEASUREMENTS

3.1 The nature of acoustical measurements

Acoustical measurements differ in a certain way from other electrical measurements. Let's take a short example. If we want to measure the resistance of an ordinary resistor, we just take a multimeter, connect the probes and measure the resistance. This is easy. The multimeter creates a known voltage and measures the current it produces and calculates the resistance from the Ohm's law. Similarly, if we want to measure the sound isolation quality of an ordinary door, we just close the door, create a known stimulus and measure the response from the other side of the door. That's simple. Or was there something wrong?

Well, the idea is correct, but the implementation isn't. In reality we don't measure the door, we measure the whole world around the door. First, the door probably isn't the only media to carry the sound signal to the other room. Walls pass some amount of sound also. The situation is lot worse if there are any kind of holes in the walls e.g. electricity plugs. The sound propagates easily trough these small holes and radiates to the recording room. A metal heating radiator, which is connected with metal pipe to another heating radiator, which is in other room, can carry a big amount of the sound through. There may be many kinds of constructions in the whole building that absorb the sound from the stimulus and connect it directly or indirectly to the other room. In this case we are not measuring the sound isolation of the door, we are measuring the sound isolation of the whole building from room to room.

Our bad luck doesn't need to end here. There may be a jet plane just taking off outside the building. This can cause extreme distortion to our results. The worst case happens, if there's a window open in the recording room so that the outside noise can propagate freely to our recording system. While we are on the other room listening the excitation signal and deaf to outside noise, we don't even realise that we aren't measuring our known excitation at all.

And finally, the room itself can colour the measured sound or the produced stimulus. There may be many kind of attenuations or increases in the sound pressure level in different points inside the room due to echoes, reflections and diffraction.

The placing of the equipment is also very important; the sound field doesn't stay constant through time and space.

While the signals to be measured and the variables to be studied remain similar to other measurements, the media where the phenomena are happening is very different, the air around us. This means that almost everything we do can greatly interfere with our measured signals. Disturbances outside can connect to our measuring system very easily. Even passive objects can add their influences and cause unwanted changes to measured results. This makes the measurements very challenging and requires good knowledge and skills from the measurer.

3.2 Requirements for the measuring environment

The very first requirement is that the signal we want to measure can be separated from other noise. This means that the signal must be powerful enough, the noise around the signal is low enough or resides in a different area in the frequency or time domain with the signal to be measured.

A good way to get rid of interfering objects that don't belong to our system to be measured is to bring the system into a free field. A good anechoic chamber is as close to free field as we can get. A well built anechoic chamber attenuates the echoes of the signals to undisturbing level and also isolates the system from other disturbances from outside world and background noise. This way we know that we are measuring just the signal we wanted and nothing else.

Of course it is not always possible to bring the system into an anechoic chamber. This indicates that other ways have to be found to achieve reliable results. If the environment is quiet enough, but not free of objects, we can use time-windowing of signals. With a short period of measuring time all echoes from outside objects can be shaded out. The drawback is that the length of the window function sets a certain limit for the low frequency end of the measured signal in the frequency domain according to Eq. 3.1, where f_l is the lower limiting frequency and t_w is the length of the time window.

$$f_l = \frac{1}{t_w} \tag{3.1}$$

3.3 Measurement equipment

To measure a sound signal, we need a piece of equipment to convert small air pressure changes to an electrical signal: the microphone. There are several kind of microphones available and they come in different sizes, packages, types and of course, price. Different uses require different kind of microphones. Condenser microphones are almost always used for precise acoustical measurements. See the construction of a typical condenser microphone in Fig. 3.1.

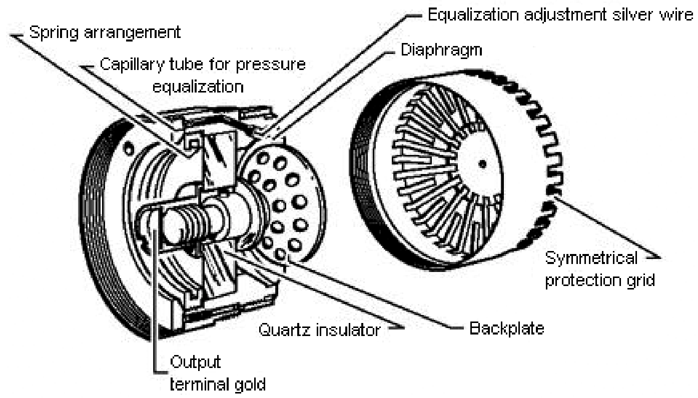

Fig. 3.1. The construction of a condenser microphone. Picture by Brüel & Kjær. (Anon. 1982).

Condenser microphones are very stable. They can keep their qualities probably for a century (Lahti 1997). They have very flat frequency response over a wide frequency range (see Fig. 3.2). They have large dynamic range and adequate sensitivity. In addition to above condenser microphones have very low internal noise. These qualities in addition to the fact that condenser microphones can be manufactured small enough not to disturb the sound field appreciably are the reasons why condenser microphone is so dominating choice.

The drawbacks of condenser microphones are the high polarisation voltage, which is typically around 200 Volts, and the price. The high polarisation voltage is a real problem in a humid environment, but certain care must be taken in other places also. The expensiveness of microphones makes a risk that careless student can drop or otherwise damage the microphone, causing loss of money. But this is a problem with all the professional measurement equipment, not only the microphones.

A microphone usually requires an amplifier to amplify the electrical signal produced by microphone. The signal is weak in amplitude, so it must be amplified before feeding it to other electrical equipment, so that it can be distinguished from the background noise. The first stage of an amplifier is located as close to the microphone as possible to minimise the stray capacity of the connection between the microphone and the amplifier and to emphasise the sensitivity of the microphone. In addition the small capacitance of the microphone requires very large impedance from the amplifier to keep the lower cut-off frequency as low as possible. The first stage of a microphone amplifier is thus designed to act as an impedance converter to provide small impedance for the cable leading to the actual microphone amplifier.

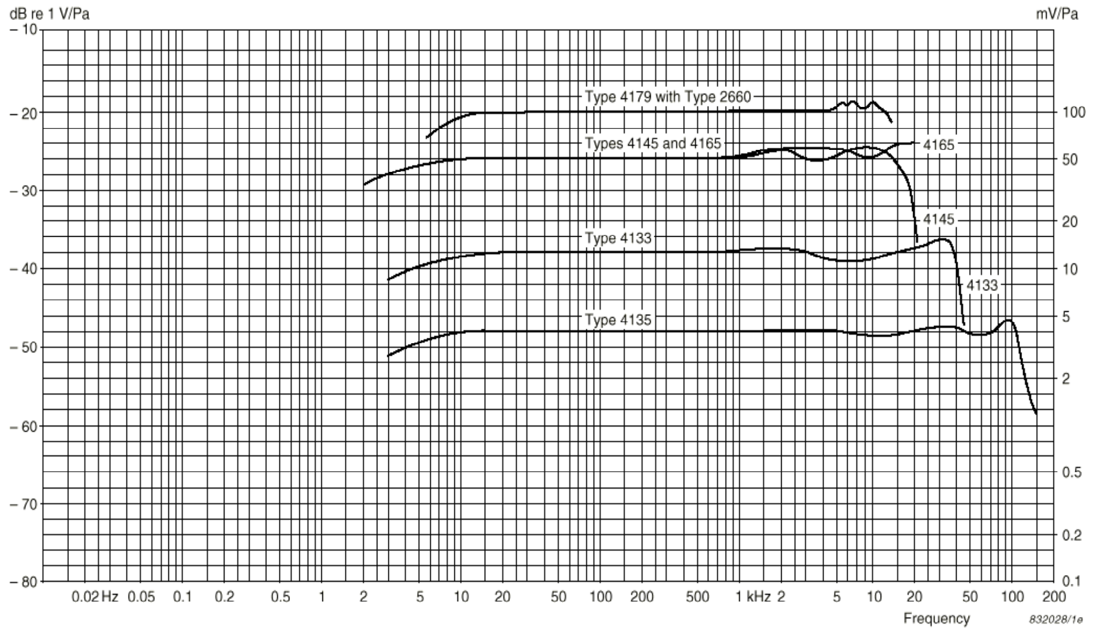

Fig 3.2 Typical zero degree incidence free-field frequency responses of the different Brüel & Kjær free field microphones. The type 4133 was used with the "Simple Loudspeaker Measurement Program." The 4133 has a very flat response from 20 Hz to 20 kHz. Picture by Brüel & Kjær. (Anon. 1982).

In addition to microphones there are other kind of sensors used in acoustics. Vibration accelerometers (Fig. 3.3) are used e.g. shock and vibration measurements and analysis, vibration monitoring, and modal and structural analysis. Many acoustical phenomena are closely related with different kind of vibration. Therefore it is sometimes more suitable to measure the system with a vibration accelerometer rather than with a microphone. On the surface of objects, or very close to surface, the microphone doesn't necessarily give reliable results. Vibration accelerometers are used also for sound intensity measurements near hard surfaces (Lahti 1997).

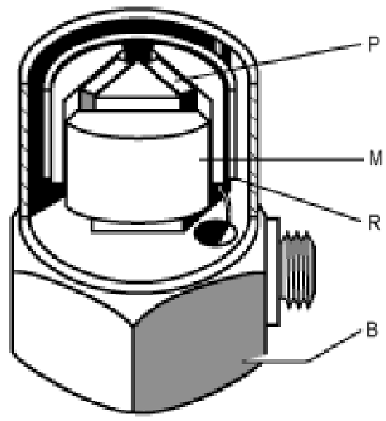

Fig. 3.3. The construction of a vibration accelerometer. M=seismic mass, P=piezoelectric element, B=base and R=clamping ring. Picture by Brüel & Kjær. (Anon. 1982).

Some surfaces are not suitable for an ordinary vibration accelerometer. An accelerometer cannot be used, for example, for measurements on delicate or hot structures, inaccessible parts, surfaces which must not be marked, very small surfaces, high voltage or radioactive surfaces and living tissue. Laser transducers are designed for non-contact measurements of angular velocity and displacement, and vibration velocity and displacement, where traditional contact methods are either difficult or impossible.

Power amplifiers are usually needed in both ends of the measurement system: in amplifying the excitation signal fed to the system and in amplifying the response from the system. Amplifiers usually handle electrical and analogue signals. Their primary function is to amplify the amplitude of the signal and act as an impedance converter to adjust the different pieces of system to act together. In some cases they have other functions like filtering out unwanted disturbances. However, the basic idea of the amplifiers is that they don't affect the signal in the frequency range we are measuring and examining in any other way than amplifying the signal amplitude.

Then we need a media to record the signal or make it visible for our eyes for analysis. Analogue devices used to handle all the audio data immediately when it arrived and present the results. This is not the case with modern digital equipment. Today everything is usually saved in the memory of the equipment in digital format. From there it is then handled in different ways, e.g., filtered, printed, presented or saved for later processing. For saving the data a tape recorder can be used, or nowadays more often a DAT-recorder (digital audio tape). With relatively short measurement data, a hard disk in a computer is an excellent media for saving signals if they can't be analysed directly from the computers memory. Hard disk is fast and signals can be easily accessed or processed when they are in digital format. All digital recording equipment requires A/D-converter (analogue-to-digital) to convert the signal from analogue format to digital. If we later want to reproduce this signal for listening we need a D/A-converter to convert it back to analogue format.

4 MEASUREMENT SYSTEM AND RELATED THEORY

4.1 Measurement system overview

The whole measurement system and work was based on one computer, see Fig. 4.1 for the block diagram of the system. The computer with its programs was the core of the system and almost all analysing and filtering was done with it. An audio amplifier was needed to amplify the excitation signal from the computer to a suitable level for a loudspeaker to reproduce the signal. There is also the loudspeaker to be measured, a microphone to measure it and a microphone amplifier to amplify the measured signal level high enough for the computer to record it. Other auxiliary equipment, like voltmeter and microphone calibrator were used, but they are not necessary for the basic operation of the measurement system, they were needed to get accurate results.

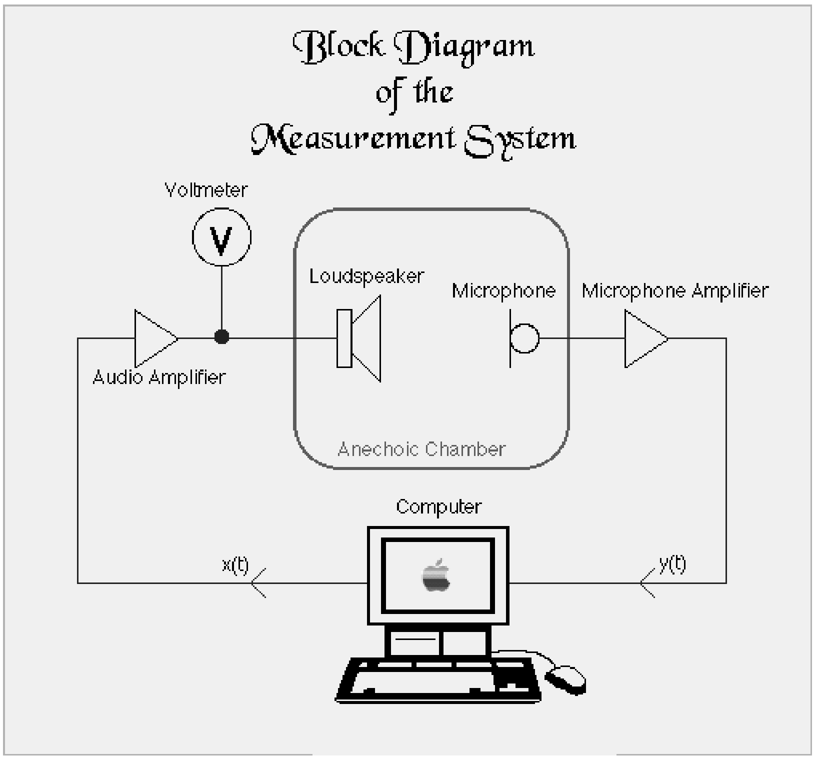

Fig. 4.1. Block diagram of the measurement system. The picture is an actual screen shot from the program. The computer creates the excitation, feeds it trough the audio amplifier to the loudspeaker, which is inside the anechoic chamber. The microphone collects the response and returns it to the computer through the microphone amplifier.

4.2 Measurement arrangements

All signals are originally generated from the computer by QuickSig; an object oriented Signal Processing Environment (the next chapter deals more with the QuickSig system). The generated excitation is then converted from digital signal to analogue one and connected to audio amplifier's input. The signal level can be adjusted from the computer and from the audio amplifier. Usually the signal is set to an appropriate level from the computer, then adjusted from the amplifier when needed. A loudspeaker to be measured is located in anechoic chamber and then connected to the audio amplifier's outputs. A voltmeter is also connected to the audio amplifier's output parallel with the loudspeaker to monitor the signal level.

The loudspeaker reproduces the electrical signal to audible sound. A microphone is located correctly in a proper place in anechoic chamber, at appropriate distance to the loudspeaker and then the microphone converts the sound signal to an electrical signal again. This signal is then connected to the input of a microphone amplifier, which is located outside the anechoic chamber. After some infrasound filtering and amplifying, the signal is connected back to the computer and converted back to digital format for analysing. Infrasound filtering made by the microphone amplifier isn't necessary, but it makes the results better by removing the infrasound disturbances, which can enter the anechoic chamber from outside world.

As can be seen in Fig. 4.1, the whole measurement system is not very complex and it is very easy to set up if some basic rules concerning acoustical measurements mentioned before are kept in mind. Ground loops in the system must be avoided. This means that all the power plugs should be connected to the same point. The condenser microphone used requires a polarisation voltage of 200 volts. Because in the anechoic chamber of Helsinki University of Technology there is a metal net acting as a floor, which is electrically floating or possibly grounded to some place, certain care must be used when wiring and connecting the condenser microphone.

4.3 The purpose of the measurements

The purpose of *Simple Loudspeaker Measurement Program* is to teach students to make reliable acoustical measurements and to make them understand acoustical measurement equipment and measurement results. This could be done by analysing previously recorded results and studying the theory of measurements and equipment. However, it can be far more educating and motivating making real measurements by their own and analysing real results. Students have the possibility to make errors and they really have to know what they are doing. This means that they have to study the subject more carefully. They have to understand what happens when they get corrupted results like it is possible in real world and distinguish unreliable results from reliable ones.

For further motivation, students have the possibility to measure their own loudspeakers. Many of the students are interested in how their own speakers perform and whether they really are as good as the manufacturer promises.

Making separate measurements each time, probably with different loudspeaker, varies the results of the measurements. So the students can't copy all the results and answers from the previous measurers, they'll have to find their own answers to their own results. Again, they have to think more about the phenomena and reasons behind them.

4.4 Theory of the measurements

There has to be a justification or a scientific basis to all used excitations and achieved results. Students learn these mathematical equations and relations between input and output signals and system in different courses and it is not the purpose of this program to teach DSP or mathematical equations and laws related to this system in depth. Only the basics are good to know. Therefore this part is discussed only briefly, better analysis can be found in the literature (see e.g. Lahti 1997, Riederer 1998).

4.4.1 Linear and time invariant (LTI) system

An acoustical system of interest here can be briefly described as a black box with an input and output, which are acoustical signals, Fig. 4.2. In the simplest model there is one input and one output and they both are sound pressures or particle velocities. In an electro-acoustical system like we have here the input is an electrical signal. If we look at the system to be measured (a loudspeaker) we have electrical input signal (voltage) and acoustical output signal (sound pressure). If the system were a microphone we would have sound pressure as an input signal and voltage as an output signal.

The following analysis can be extended to any physical system, but for simplicity we talk only of acoustical systems. An ideal acoustical system has four different characteristics: physically realisable, time invariant, stable and linear. The most important property of this kind of system is its response to the ideal impulse function, called Dirac's delta function $\delta(t)$. This response is called impulse response function and it contains the complete information from the behaviour of the system under any kind of stimulus.

If we look at the system with input signal *x(t)* causing an output signal or response *y(t)*, we can define the impulse response function as

$$h(t) = y(t) \quad \text{when} \qquad x(t) = \delta(t) \tag{4.1}$$

where *t* marks the time measured from the moment when the delta function arrives at the system.

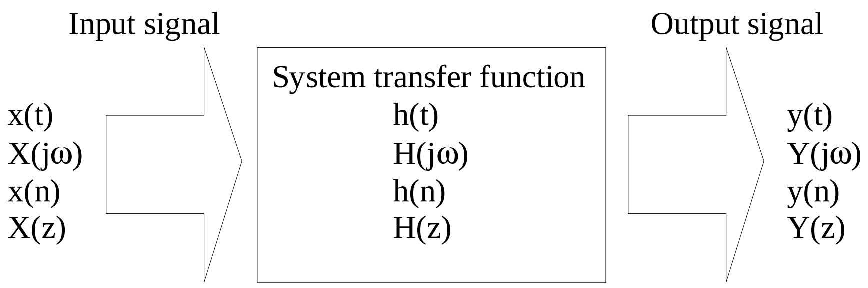

Fig. 4.2. *An LTI system. Letters 'x' and 'X' indicate input signal; 'h' and 'H' system response function and 'y' an 'Y' output signal. The capitalised letters refer to a linear transform domain. In continuous domains the signals are a continuous function of time (t) or transformed function of complex variable (jω). The discrete representation uses discrete variable n or its transformation z.*

The importance of the function *h(t)* is shown in the next equation. For any arbitrary input signal *x(t)* the response *y(t)* of the system can be resolved from integrating

$$y(t) = \int_0^{\infty} h(\tau)x(t-\tau)d\tau \tag{4.2}$$

Let's see in more detail what these four characteristics for acoustical system mean. Physically realisable means that the system can't generate the response for a particular stimulus before the stimulus has arrived to the system. This means that $h(\tau) = 0$ when $\tau < 0$. Because of this we can choose the lower limit of integration as zero instead of -∞ in Eq. 4.2.

A system is time invariant if its fundamental properties do not change with respect to time. We can say the system has constant parameters if the impulse response function is not dependent of the time when it arrives to system

$$h(t,\tau) = h(\tau) \qquad \text{for every} \qquad -\infty < t < \infty \tag{4.3}$$

A system is stable if any limited input signal creates a limited output signal. Mathematically:

$$\int_{-\infty}^{\infty} |h(\tau)| d\tau < \infty \tag{4.4}$$

Lastly, a system is linear if it is additive and homogenous. The additivity means that the output to a sum of inputs is equal to the sum of outputs produced by each input individually. The homogeneity means that the output produced by the input multiplied by a constant is equal to the output produced by the input alone multiplied by a constant.

4.4.2 Linear transform

The dynamical properties of acoustical systems are usually presented by using an integral transform of the impulse response function $h(t)$ rather than the impulse response function itself. In LTI systems the transforms $x \leftrightarrow X$, $y \leftrightarrow Y$ and $h \leftrightarrow H$ could be done by any kind of linear and generally complex-valued transform. In practice it is usually most useful and efficient to use the Fourier transform, which provides the signal presentation directly in the frequency domain. The Fourier transform of a impulse response function $h(t) = 0$, when $t < 0$ is:

$$H(f) = \int_0^{\infty} h(t) e^{-j\pi ft} dt \tag{4.5}$$

The above function $H(f)$ is called the frequency response function. It is a complex-valued function with real and imaginary parts:

$$H(f) = H_R(f) - jH_I(f)$$

$$= \int_{0}^{\infty} h(t) \cos 2\pi ft dt - \int_{0}^{\infty} h(t) \sin 2\pi ft dt \tag{4.6}$$

The polar notation is more convenient in practise:

$$H(f) = |H(f)| e^{-j\phi(f)} \tag{4.7}$$

where $|H(f)|$ is the nominal value of the frequency response function and $\phi(f)$ is the phase:

$$|H(f)| = \sqrt{H_R^2(f) + H_I^2(f)} \tag{4.8}$$

$$\phi(f) = \arctan\left(\frac{H_I(f)}{H_R(f)}\right) \tag{4.9}$$

The nominal value $|H(f)|$ is called gain factor or more familiarly the magnitude response. The phase $\phi(f)$ is called phase factor. Magnitude response is a very important function in loudspeaker measurements.

Also, if the Fourier transform $H(f)$ is known, the inverse Fourier transform returns the function $h(t)$ by:

$$h(t) = \int_{-\infty}^{\infty} H(f) e^{j2\pi ft} df \tag{4.10}$$

Fourier transform is used with the continuous variable t and referred as the (continuous) spectrum of the time domain signal. In discrete cases the transform (Eq. 4.5) is applied by the Z-transform:

$$H(z) = \sum_{0}^{\infty} h(n) z^{-n} \tag{4.11}$$

where the variable z is continuous and complex. This transform is obtained either by the discrete time Fourier transform (DTFT) for aperiodic signals:

$$H(j\omega) = \sum_{-\infty}^{\infty} h(n) e^{-j\omega nT} \tag{4.12}$$

or for periodic signals by the discrete Fourier transform (DFT):

$$\tilde{H}(l\omega_0) = \sum_{n=0}^{N-1} h(n) e^{-j\omega_0 nT} \qquad \text{where } l = 0, 1, \ldots, N\text{-}1 \tag{4.13}$$

and T is the sampling period. The discrete Fourier transform is an N-point sequence computed from an N-point time sequence. A frequency scale is provided with the sampling frequency f_s:

$$f = \frac{f_s}{N} = \frac{1}{NT} \tag{4.14}$$

This discrete Fourier transform is usually computed by means of an efficient algorithm called the fast Fourier transform (FFT). Fast Fourier transform reduces the arithmetic operations from N^2 to $N \log_2 N$. The FFT algorithm yields an identical result to direct application of the DFT, so all FFT limitations are the same as in DFT. Since the DFT and IDFT involve basically the same type of computations, same kind of algorithm exists for inverse transform, inverse FFT (IFFT).

4.4.3 Usefulness of impulse response and transforms

The response function can be interpreted physically by the following way: the sine signal arriving to ideal system produces a response in the same frequency and only in the same frequency than the arriving sine signal was. The amplitude and phase of the response are usually different

$$x(t) = \sqrt{2}X(f)\sin 2\pi ft \tag{4.15}$$

$$y(t) = \sqrt{2}Y(f)\sin(2\pi ft - \varphi(f)) \tag{4.16}$$

The ratio of the output signal and the input signal equals the system response *H(f)*. This important relation ties the input and the output and transfer function of the specified system together. The amplification of the system is achieved from

$$H(f) = \frac{Y(f)}{X(f)} \tag{4.17}$$

and the phase difference between the input and the output equals the phase response of the system

$$\phi(f) = \varphi(f) \tag{4.18}$$

The reason for using the different transforms described in chapter 3.4.2 is that we can replace difficult integrals or analytically unresolved functions by simple multiplication. Instead of convolution integral (Eq. 4.2) we can get the response of the system from comparatively easy multiplication

$$Y(f) = H(f)X(f) \tag{4.19}$$

The response in the time domain can be achieved from the inverse transform (Eq. 4.10). In many cases this can be solved even analytically. Solution can be achieved always easier by numerical inverse transform than by numerical convolution. We can use the numerical algorithms like the FFT to get the desired results.

These mathematical equations are used inside the program code in the *Simple Loudspeaker Measurement Program* to get the different results and responses needed in various measurement pages of the program.

4.5 Measurement signals

To optimise the calculations and to get the best possible results, some attention needs to be paid to the excitation signals. Different kind of signals may give different responses; some set of signals can have undesirable side effects, or may introduce some kind of unwanted phenomena. With suitable excitation signals we can get proper results much more easily and in less time. In addition some measurements may require a special kind of excitation signal to get any kind of sensible results. It might be a somewhat difficult task to measure the nonlinear distortion of the system if some kind of noise signal is used as a stimulus. On the other hand, it is easy to see the harmonic distortion produced by the system, if we use a single frequency sine signal and examine the result in frequency domain (Fig. 4.3).

Frequencywise, audio measurements can be roughly divided into two sections: narrow band and wide band. Ideally the excitation signal should have a flat spectrum and a low crest factor. The flat spectrum means that the energy of the signal is distributed evenly over different frequencies. If the excitation signal lacks energy in a certain frequency band the results in this band are probably unreliable. The crest factor is the peak amplitude of the signal divided by the root mean square (RMS) value of the signal. Low crest factor thus means that the signal is utilising the dynamical range of the equipment well and providing better signal-to-noise ratio (S/N).

The excitation signal must have high signal level to maximise the signal-to-noise ratio. Averaging is also commonly used method to get better S/N. Needless to say, all possible noise sources must be isolated as well as possible and noise levels kept as low as possible. All disturbances can create havoc in results.

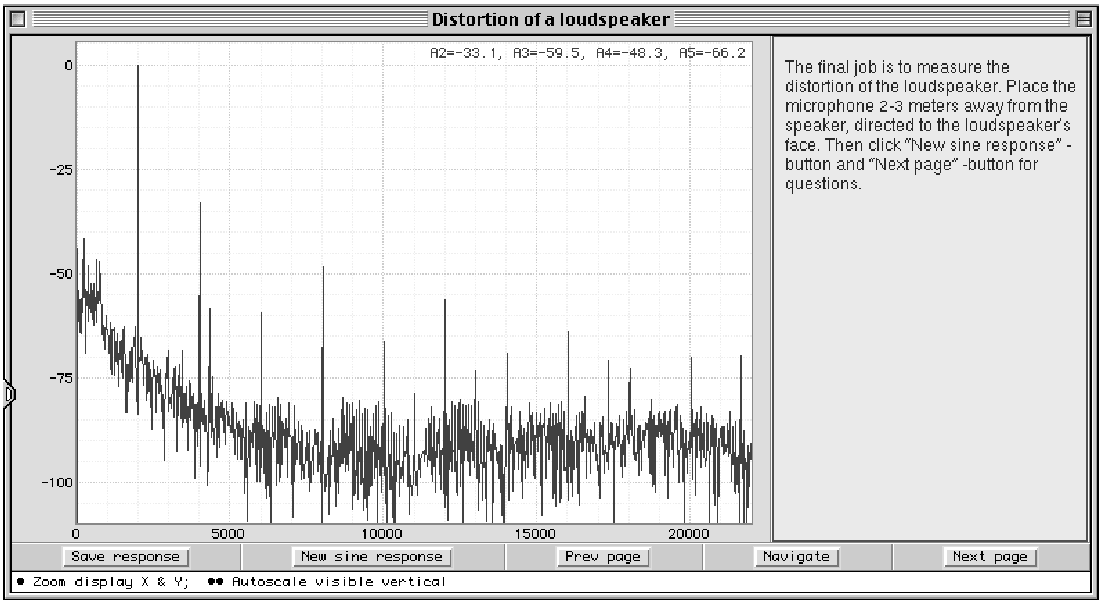

Fig 4.3 2 kHz sine signal with distortion components. It is far easier to see the distortion components from frequency domain than from the time domain. The picture is a screenshot from "A Simple Loudspeaker Measurement Program." Notice the decibel values of harmonic components on the upper right corner of the signal view.

4.5.1 Pure tone signals

Pure tone signals consist of discrete frequencies. They can be, e.g., single frequency sinusoidal or swept sinusoidal signals that have a crest factor of $\sqrt{2}$. The former ones are used in narrow band and the latter in wide band measurements.

As mentioned before a sinusoidal input signal provides a sinusoidal output in the same frequency in LTI-systems, only the amplitude and phase usually change. Therefore single frequency sine-signals are suitable for measuring system's nonlinear behaviour: the harmonic and intermodulation distortion. Sine signals, usually 1 kHz, are commonly used as calibration signals, e.g., for microphones.

Another commonly used group of pure tone signals is sweeps. A slowly swept sinusoidal is called a sweep and a fast one is called a chirp. Swept sinusoidals have been used traditionally in filter analysis techniques and in loudspeaker magnitude response measurements.

4.5.2 Impulse

The theoretical Dirac's delta function (Eq. 4.20), i.e., impulse, is approximated by a short impulse-like pulse.

$$\delta(t) = \infty,\ t = 0, \text{ otherwise } \delta(t) = 0 \text{ and } \int_{-\infty}^{\infty} \delta(t) = 1 \tag{4.20}$$

This kind of stimulation is problematic due to infinitely high crest factor. It can also be difficult to approximate. The spectrum of this kind of pulse isn't necessarily flat enough over a wide frequency scale. Audio devices, e.g., loudspeakers do not radiate enough power for high frequencies. Because none of the equipment is allowed to leave its LTI-range, the S/N with impulse excitation signal is very weak because all its energy is localised in a very brief time interval, meaning low average power.

4.5.3 Random noise

Random noise suits well for audio measurements and is widely used. A long excitation signal can be used and the response averaged, thus the excitation delivering more power to the system. Also the energy of the random noise signal is distributed more uniformly over different frequencies. The most important types of random noise are white noise and pink noise.

Ideal white noise has a flat power spectrum over all frequencies and it doesn't have any periodical components. This means an infinite RMS-value and unrealisable signal. In practise the white noise is band limited, meaning a constant power spectrum over the considered frequency range and finite power elsewhere. The pink noise has a flat power spectrum over a logarithmic frequency scale, i.e., in linear frequency scale its power spectrum is decreasing 3 dB/octave. Pink noise can be filtered from white noise.

4.5.4 Pseudo-random noise

Pseudo-random noise is the most important type of stimulus in audio measurements today. It is a periodic noise sequence, whose length is the inverse of the repetition frequency. Pseudo random noise signal can be generated and averaged to get better

S/N. Usually a flat spectrum with randomised phase is used to get low crest factor. *Simple Loudspeaker Measurement Program* uses exactly this kind of random-phase flat spectrum (RPFS) signal (fig. 4.4). It has a flat spectrum in its periodicity frame (Karjalainen 1997).

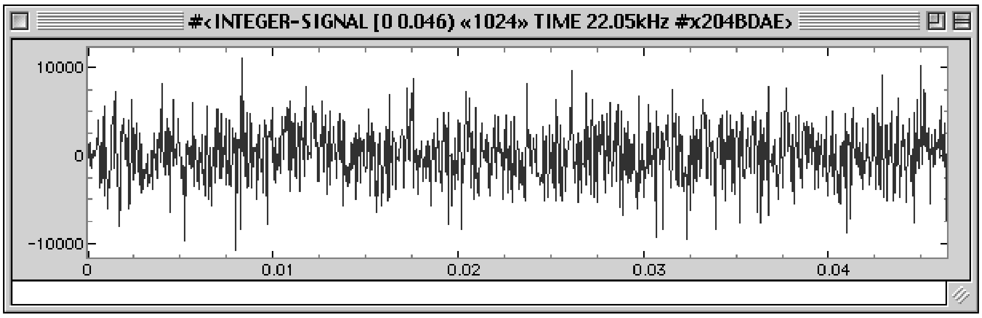

Fig. 4.4. An RPFS-signal in time domain. The picture is a screenshot from the QuickSig. RPFS-signal has a flat frequency response.

RPFS signals are not binary codes, like Golay codes (see below); instead the phase values of the signals are randomised. An RPFS signal can be easily generated with FFT/IFFT techniques, when the length L of the RPFS sequence is an exponent of 2, i.e., $L = 2^N$. Hence the impulse response of the system is also efficiently achieved by DSP techniques.

The maximum length sequence (MLS) is a periodic sequence of integers, $x'(n)$. It can be generated by primitive polynomials (Davies 1966, Schroeder 1970) or most easily by digital shift registers. The latter case leads to a binary sequence, i.e., to an impulse train consisting of discrete impulses that are two-valued integer variables. MLS signals are the most utilised ones from the pseudo-random noise signals. The discrete MLS signal has a flat spectrum and random phase that varies pseudorandomly with frequency having a uniform probability density over its range of $+\pi$ to $-\pi$. Thus the binary MLS has a minimum crest factor and the signal energy is maximal and the S/N is optimal. MLS has an important circular autocorrelation property that allows one to easily measure the periodic impulse response of a system.

Golay (complementary) pairs are binary codes that have a distinguished property, that their autocorrelation functions have complementary sidelobes (Golay 1961).

Therefore, the sum of the autocorrelation sequences is exactly zero except at he origin (Foster 1986), and the power spectrum is exactly flat. Efficient FFT algorithms can be utilised with Golay codes, because the length L of the sequence is always exponent of 2, i.e. $L = 2^N$. Otherwise the Golay codes have much the same characteristics as MLS signals.

4.6 Achieving the frequency response

The frequency response measurements of a loudspeaker can be made using pseudo-random noise as an excitation signal *x(n)*. After passing the system the excitation signal gives a response *y(n)*. The Eq. 4.10 presents the inverse Fourier transform for *H(f)* to get the impulse response *h(t)*. Same kind of inverse transforms exist for Eq. 4.11, 4.12 and 4.13 to get the desired impulse response *h(n)* for different types of signals.

$$h(n) = IFFT\{H(z)\} \tag{4.21}$$

In practise the results are nearly always calculated by means of FFT and IFFT. FFT algorithms can be found e.g. in *Digital Signal Processing – Principles, Algorithms, and Applications* (Proakis 1992).

Then, like Eq. 4.17 states for *H(f)*, similarly stands for *H(z)*:

$$H(z) = \frac{Y(z)}{X(z)} \tag{4.22}$$

Combining the results of Eq. 4.21 and 4.22, the *h(n)* can be presented as:

$$h(n) = IFFT\left\{\frac{FFT\{y(n)\}}{FFT\{x(n)\}}\right\} \tag{4.23}$$

This is the method for getting the impulse response of acoustical system, e.g., loudspeaker, with pseudo-random noise excitation signal. The actual calculating order may in reality be different, or other additional methods like filtering may be used in to get more accurate result. The Eq. 4.23 is still the basis what the *Simple Loudspeaker Measurement Program* uses to get the impulse response of a loudspeaker. In section 4.4.1 it was presented that the impulse response function *h(n)* contains the complete information from the behaviour of the system under any kind of excitation signal.

It should be noticed that the impulse response function calculated this way describes the characteristics of the whole system, including the audio amplifier, the microphone, the microphone amplifier, the anechoic chamber, all wires including connectors and the D/A- and A/D-converters of the computer itself. However, both of the amplifiers can be regarded as ideal components. They only amplify the input signal and have nearly flat magnitude response in audio frequencies. This is the case with microphone also. It only transforms the input signal from pressure to electrical and has almost flat magnitude response in audio frequencies (see Fig. 3.2). The anechoic chamber of the Helsinki University of Technology attenuates roughly speaking 20 dB every reflection in audio frequencies. Its influence on the results can thus also be left out. Normal wires and connectors can be approximated with ideal 0 ohm resistors with good accuracy. They don't have notable resistance, capacitance or inductance.

The quality of D/A- and A/D-converters can really alternate between different manufacturers and computers. The converters in this system proved to be rather good by nature. The influence of converters can be compensated by measuring the response of mere converters and then adding the inverted results with the actual measurement data. This is implemented in the QuickSig with the D/A- and A/D-converters in the computer, so they have a minimal effect on the results. Furthermore, this compensation could be used with any part of the chain to cancel the effect of that part to the results.

The Simple Loudspeaker Measurement Program is designed for teaching purposes. All distortion caused by other equipment in this system is so small that it can be safely regarded meaningless in this case. If the program is used for precision audio measurements, some further attention should be directed to all equipment and environment when examining the results.

5 THE QUICKSIG ENVIRONMENT

5.1 Platform

The Apple Macintosh Power-PC computer was selected as the platform for this program. The Macintosh has good graphics handling capabilities, good sound reproduction, is easy to use and program, and most of all, has first-rate D/A- and A/D-converters. The computer can be directly connected to the audio amplifier and the microphone amplifier can be directly connected back to the computer. No additional sound cards were required. The quality of Macintosh's own connectors was adequate for our purposes.

Two computers were tested: G3-Macintosh and 100 MHz Power Macintosh. Normal 100 MHz Power Macintosh was also powerful enough, but in some points of the program with several windows open, students would have been waiting for a couple of seconds for something to happen and that would have caused their interest towards this program to drop. G3 was really fast and students didn't need to wait something to happen. The G3-Macintosh was equipped with 64 Megabytes of RAM (random access memory), the 100 MHz Power Macintosh was equipped with 32 Megabytes of RAM. Though 32 Megabytes was enough and even less would have been sufficient, more memory made this program to run easier: there wasn't need for e.g. garbage collection so often. Because the program doesn't use hard disk very much, it doesn't make big difference how fast a hard disk is used. The measurement data requires comparatively small amount of disk space, so even a slow hard disk doesn't slow down this program very radically.

A good monitor is always recommended for CBE programs. We had a normal 17-inch colour monitor with colour graphics card. Except for the theory pages, this program doesn't require thousands of colours, so a normal graphics card is enough to view the program properly. Pictures on the theory pages look better with all the colours in use, but the program can very well be used with fewer colours. A standard keyboard is used also, though most of the things are done with the mouse. Only the answers to the questions must be typed in.

5.2 Macintosh Common Lisp

Lisp is a general-purpose language, perhaps the most general-purpose language available (Tatar 1987). Although it was originally designed with a specific problem domain (recursion theory) in mind, for twenty-five years it has evolved into a tool for solving the problems of artificial intelligence (AI). And CBE is closely enough related to artificial intelligence.

The single most important feature for the new user with Lisp is that Lisp is interactive. An expression can be typed at Lisp, and Lisp will give an immediate response. Learning to predict Lisp's behaviour at the interactive level is crucial for mastering its complexities. The first task in learning to program in Lisp is learning to predict how Lisp evaluates expressions made up of objects. Once the fundamental behaviours are understood, a programmer can write function definitions that rely on them. These definitions can be made interactively by typing them in, just like simpler statements can be typed in. Lisp programs consist of many function definitions stored in one or more files. A reference for Lisp programming can be found in *Common Lisp – The Language* (Steele 1990).

Since there has been no standard Lisp until 1984, Lisp implementations differ greatly, both in underlying concepts and in specific details. The Common Lisp specification was created by a group of Lisp programmers who wanted to create a common ground from which different implementations could diverge if necessary. The Common Lisp specification is documented in *Common Lisp – The Language* (Steele 1990). Every implementation of Common Lisp should have all the functions documented there. Some implementations may have other features as well. This is the case with Macintosh Common Lisp. It follows the syntax of Common Lisp, but it has some extra functions not included in Common Lisp.

So far there is no general standard for graphics programming in Common Lisp. This is a problem for CBE programmers and others who need to create universal code that is platform independent. Different implementations of Common Lisp use different functions for presenting graphics according to the operating system or computer type they rely on.

5.3 QuickSig

The *Simple Loudspeaker Measurement Program* was programmed in the Lisp programming language on top of the QuickSig (Karjalainen 1990). QuickSig is an object-oriented signal processing environment developed at the Helsinki University of Technology. QuickSig is also written in Lisp and takes advantage of some object-oriented Lisp extensions. It has several signal processing features ready for use without any further programming. Combined with Common Lisp Object System (CLOS) (Keene 1989). QuickSig makes a powerful environment to handle sound, signals and graphics. QuickSig was originally developed as an experimental object-oriented signal processing environment. Thus the strengths of QuickSig are it's common usability, independence of any particular application and the fact that it is a technically oriented environment. This makes QuickSig a natural choice for the basis of this program.

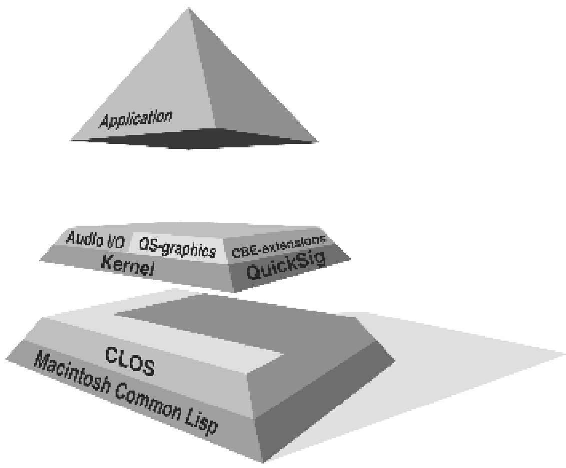

Fig 5.1. The QuickSig pyramid. QuickSig is composed of different parts on top MCL and CLOS. Applications can be build on top of QuickSig.

QuickSig consists of different kind of packages, e.g. kernel, Audio I/O and QS-Graphics (see Fig 5.1), which can be included in the program to perform all signal processing and graphics handling what is required. This means there is no need for outside signal processing, everything can be done inside the program. That makes the whole system relatively easy and minimises the different pieces of equipment needed.

The QuickSig kernel can be considered as a signal processing extension on top of the Lisp programming language. QuickSig Graphics is QuickSig's graphical user interface. It is a package, which makes it possible to present signals graphically as objects and handle them with mouse and keyboard. QuickSig Graphics has been extended to support computer-based education. Windows and views with different kind of signals, controls, text and pictures can be programmed and presented (Fig. 5.2). The behaviour of all these, as well as commands from the mouse and keyboard, are strictly under the developer's control. For example actions from mouse and keyboard can be easily restricted or released to prevent unwanted events from happening.

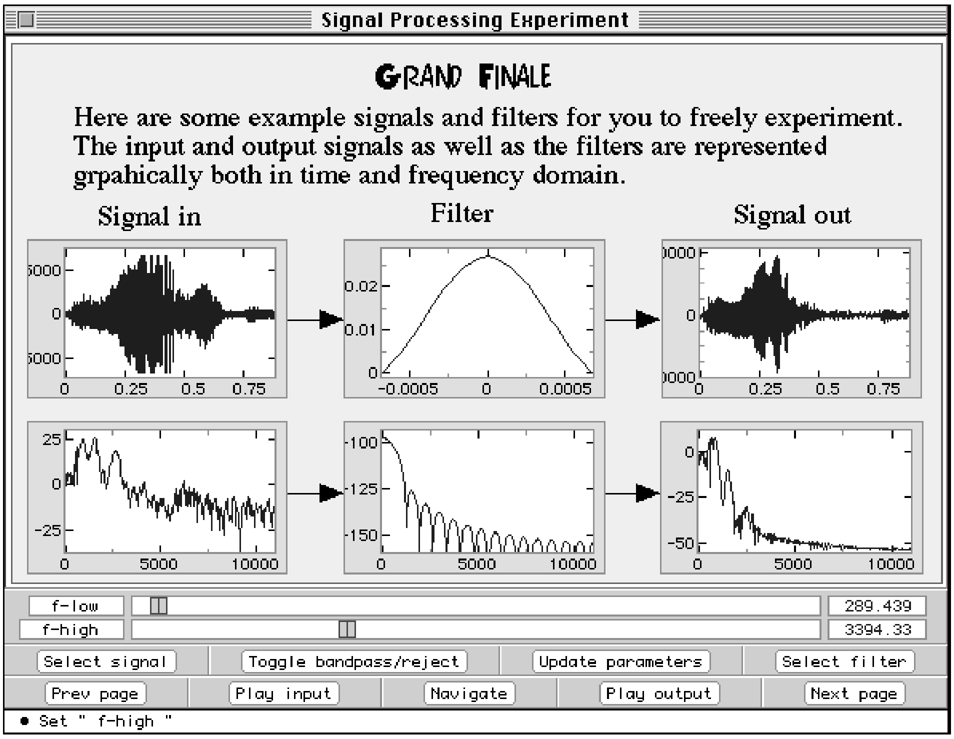

Fig. 5.2. A window from Martti Rahkila's program "Introduction to Signal Processing" (Rahkila 1996). The program is written in Lisp using QuickSig. The filter parameters can be adjusted with the mouse and the signal can be filtered. Both the original and filtered signal can be listened and examined from the screen.

QuickSig is constantly under development. This means that the documentation isn't always up to date and the QuickSig environment is not commonly available. These developer's disadvantages don't show in CBE programs developed with QuickSig. CBE programs are independent packages, which are tested and documented separately.

5.4 An example Lisp code

There are short examples from the *Simple Loudspeaker Measurement Program* in Fig. 5.3 and 5.4. Fig. 5.3 presents the definition of second measurement page and related question page is defined in Fig 5.4.

```
(defpage measure2
          (:v (:h (make-instance 'audio-group-display-base)
                  (make-pict :pict-file "pictpath:medir2.pict"
                             :pict-size #@(230 365)))
                  (controls  (save-audio-button)
                             (measure-audio-button)
                             (audio-resolution-button)
                             (prev-button 'measure1)
                             (navigate-button)
                             (button :text "Next page"
                                     :qs-help-string "• Click here for questions"
                                     :button-handler #'(lambda ()
                                                         (show-qs-page 'question2)
                                                        ))
             ))
          :title "Magnitude response 20°"
          :view-size #@(780 400)
          :view-position #@(15 40))
```

Fig. 5.3. Lisp code from the "Simple Loudspeaker Measurement Program." This code defines the second measurement page. This page is presented in Fig. 6.3.

In Fig. 5.3 *defpage* defines the name of the page. *audio-group-display-base* states that the page contains special audio-measure window. *pictpath* is a variable telling the location of the pictures in the computer's hard disk and *medir2.pict* is the picture containing the information for measurement 2 (right hand side on Fig. 6.3). *pict-size* defines the size of the *medir2.pict*. *Controls* defines the buttons located at the bottom of the window. Button actions are usually defined elsewhere, there are standard buttons and actions, but buttons can be programmed to act differently. Here the last button opens the question window of Fig. 6.4. *title* prints the window title, *view-size* orders the size of the window and *view-position* tells the location of the window on the screen.

```
(defpage question2
  (:v (make-pict :pict-file "pictpath:quest2.pict"
                 :pict-size #@(300 36))
      (editor-view :file (cbe-user-file-q2)
                   :h-scrollp nil)

      (controls (button :text "Return answer"
                        :qs-help-string "• Click here when you have answered the
question."
                        :button-handler #'(lambda () (show-qs-page 'measure3)
                                           (window-hide (find-window "Magnitude
response 20°"))
                                           (window-hide (find-window "Question
2")))))
                        )
  :title "Question 2"
  :window-class 'cbe-editor-window
  :view-position #@(15 465)
  :view-size #@(550 150)
  :close-box-p 'nil
)
```

Fig. 5.4. Lisp code from the "Simple Loudspeaker Measurement Program." This code defines the second question page. This page is presented in Fig. 6.4.

Fig 5.4 presents the Lisp code for question window 2. *editor-view* defines that the window contains an editor. *h-scrollp nil* removes the horizontal scroll bar from the editor. This time there is only one button in the window. Clicking this button opens the next measurement window, hides the previous measurement window and the question window. The class of *cbe-editor-window* is defined so that when the window is closed, the contents of editor view are saved to disk. *close-box-p 'nil* deactivates the small box from the upper left corner of the window, thus the window can only be closed by pressing the "Return answer" –button.

6 THE SIMPLE LOUDSPEAKER MEASUREMENT PROGRAM

6.1 The structure of the program

The Simple Loudspeaker Measurement Program consists of three different main parts: the introductory pages, the theory pages and the measurement pages. Introductory pages present the program and its parts shortly, show an example page of measurements and give instructions on how to set up the system and perform measurements and how to use the program. Theory pages consist of theory and pictures related to measurements and measurement equipment. Measurement pages deal with measurements, give short instructions to every measurement and present different questions for students to answer.

The Program can be fully used with the mouse. The keyboard is only needed when typing in the answers to the question the student faces. All other actions can be handled with the mouse. The program has a graphical user interface (GUI) and relies strongly in graphics, windows and pictures.

6.1.1 The introductory pages

When starting the program, the student first has to give his or her name and student number, so that different students and their answers can be later identified. The identification is based on unique student number, given to every student in the Helsinki University of Technology. After registration the program starts. The first page presents the Helsinki University of Technology and the Laboratory of Acoustics and Audio Signal Processing. Then there is a page for motivating the student and telling where the acquired skills can be later used. After this there is one instruction page to tell how to use the program, how to navigate, what kind of properties can be found, where different actions can be found etc. (see Fig. 6.1).

Navigating through the program is quite the same like navigating with a browser through the World Wide Web (WWW). There are "Next page" and "Prev page" buttons in the bottom edge of each window. With the help of these buttons all the necessary parts of the program can be completed. These buttons, however, work a bit differently than back and forward buttons in WWW browsers. The buttons have previously defined pages where they refer to and clicking a button with a mouse takes the student to the location where the button was programmed to point, not necessarily to the previous or next page in chronological order.

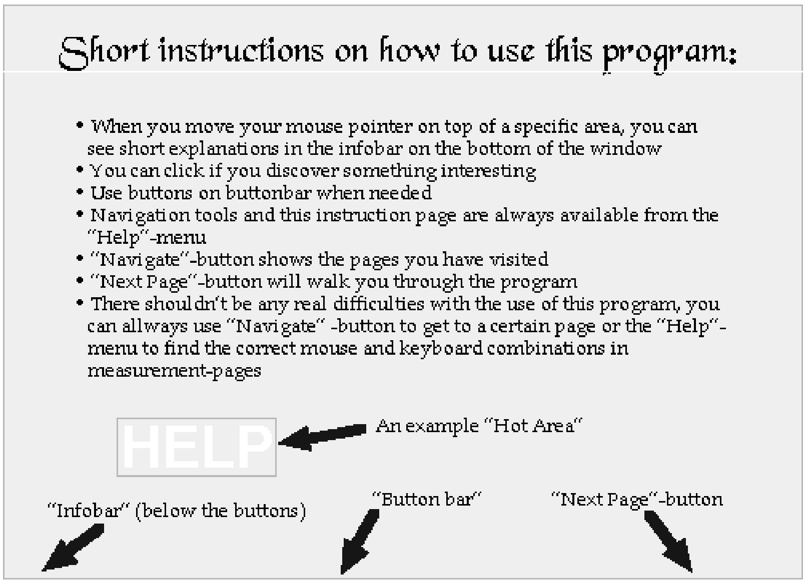

Fig. 6.1. Instruction page of "Simple Loudspeaker Measurement Program." This is an example of how the introductory pages look like.

There is also a navigation window which can be opened from the navigate button or from the pull down menu from the menu bar of the QuickSig. The navigation window lists all the pages the student has visited, and a certain page can be accessed again via the navigation window without lengthy browsing. All pages have their names shown in the upper edge of the frame surrounding the window and these names are listed in the navigation window.

There are also so called hot areas inside the windows of the program. These hot areas perform different kind of programmable actions when clicked with the mouse. One action type is underlined link. Clicking this kind of link takes the student to the page, which contains more information related to the underlined word. Moving a mouse over a hot area causes the hot area to be highlighted. A short text or instruction can be programmed to show in the instruction bar in the lower edge of the window when the mouse is on top of a hot area.

After passing the first few pages giving background information and summary of commands and action, the student gets to see the table of contents of the program. This page could be called the core page of the program; it is usually the place where the student finds himself or herself, when he or she has completed one part in the program. The theory pages can be opened here, block diagram of the system can be viewed, example measurement can be studied, and the instructions on how to set up the measurement system and the real measurements are accessed from here.

The pulldown menus contain also many useful things. Almost every informative window or part of the program can be opened from the menus. Windows can be printed from the pulldown menu and help page or complete list of commands can be opened. The navigate window can also be found through the menus.

6.1.2 The theory pages

The theory pages are divided to seven parts according to the measurement equipment and environment: the computer, the audio amplifier, the voltmeter, the loudspeaker, the microphone, the microphone amplifier and the anechoic chamber. The first part, dealing with computer, the program and environment used with measurements, describes also briefly the mathematical or digital signal processing theory related to the measurements. The second part explains the purpose of the audio amplifier. The third part describes the function of voltmeter. The fourth part, the loudspeaker, is the biggest section of the theory pages. There are so many different types of loudspeakers, so many different areas in loudspeakers that can affect the results of various measurements that this section grew big even though everything irrelevant was left out. On the other hand, the loudspeaker is probably the part that interests students the most. So there was a certain reservation for a big amount of information dealing with loudspeakers. A good reference for loudspeakers can be found in *High Performance Loudspeakers* (Colloms 1991). Also the questions the program represents in the measurement part are closely related to the loudspeaker and some background information may be needed. (See Fig. 6.2).

Closely as big part of the theory pages as the loudspeaker pages is the section of pages dealing with the microphones. This is also very important part of the theory pages. Results can be easily ruined by wrong usage of the microphone. Microphone is also very expensive piece of equipment and it is really desirable that it is left undamaged after the measurements. The theory covers different types of microphones, the technical data of the specific microphone used in measurements, the effect of different locations of the microphone to the results and the possible polarisation voltage needed with the microphone.

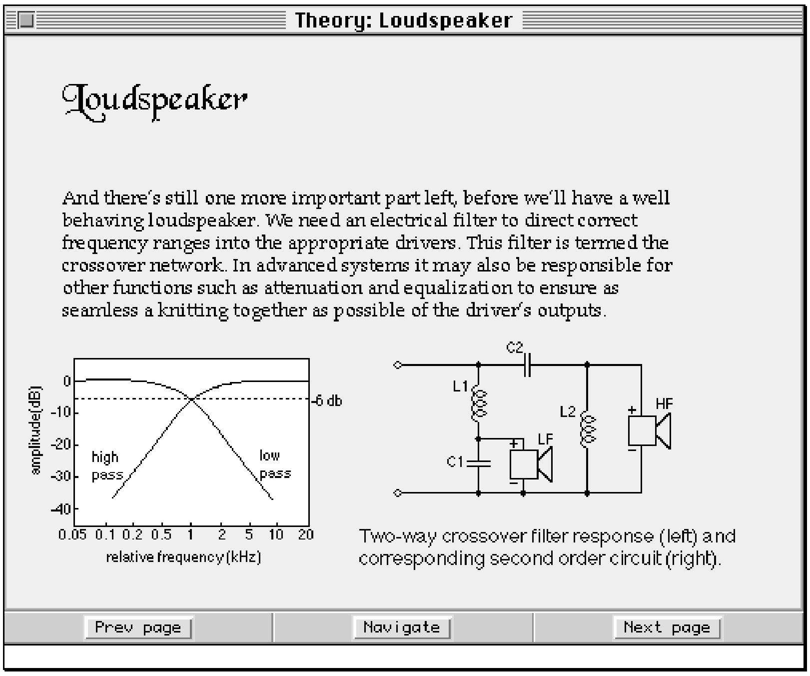

Fig 6.2. An example of theory pages. Theory pages present the computer, audio amplifier, voltmeter, loudspeaker, microphone and anechoic chamber. This page explains the function of crossover network in a loudspeaker.

The microphone amplifier pages are closely related to the microphone and present shortly the function of the amplifier and the different properties the amplifier has, like e.g. signal filtering. The final part of the theory pages deals with the anechoic chamber. It pictures the anechoic chamber of the Helsinki University of Technology, what qualities the chamber has, and how the chamber makes the measurements easier. This part describes how the environment can affect the results and how important it is to take into account the measurement environment. After this final part of the theory pages the student is returned to the introductory pages.

6.1.3 The measurement pages

The measurement pages are the most significant part of the program. See an example in Fig. 6.3. Unlike the theory pages they can't be skipped, they must be completed in order to pass the program. In the springs 1998 and 1999 there were only seven measurement pages, a lot fewer than theory pages, but the measurement pages are much harder to go through than the other pages.

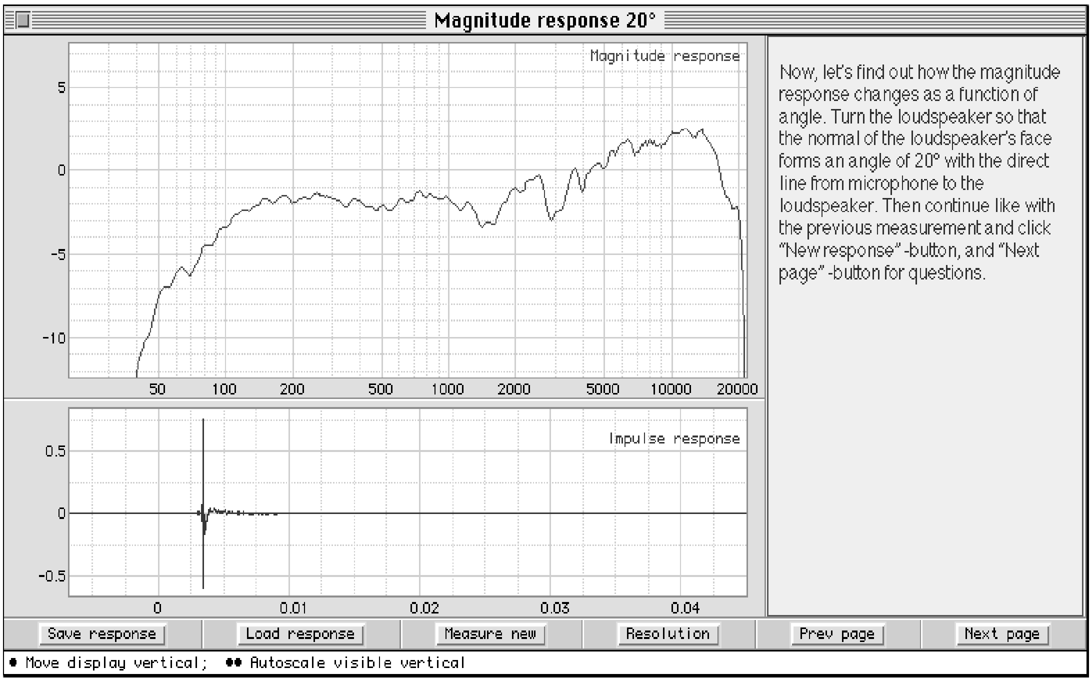

Fig 6.3 Example of measurement pages. In the upper window is the magnitude response of a loudspeaker, measured from 20° horizontal angle. In the lower window is the corresponding calculated impulse response. On the right hand side short instructions dealing with this particular measurement are presented.

During the measurement pages the student makes all the required measurements according to the instructions written on the right side of the measurement window. The student needs to check that the microphone is in the right place, in the right height, facing the right direction and all the other possible objects and equipment are in order if they are needed. Then the student needs only to click the "Measure new"-button with the mouse to get the results on the screen. It is intended that the program is very easy to use, the student needs to concentrate on the measurement itself and the acoustical phenomena, not using the program and understanding computer programming.

After the student is satisfied with the measurement results, he faces a question or several questions concerning that particular measurement. The questions can be of several possible types, but in the spring 1998 only essay type of questions were used. When the question window opens, after the student has clicked the "Next page"-button, the student faces a question with an editor view inside the window where he or she can type the answer to the question (see Fig. 6.4). The editor behaves like an emacs editor. When finished the student clicks the "Answer"-button and the program saves the answer to disk in a file named according to the student's unique student number and the question number. Then the question window and the measurement window close and a new measurement window with a new task and new instructions opens.

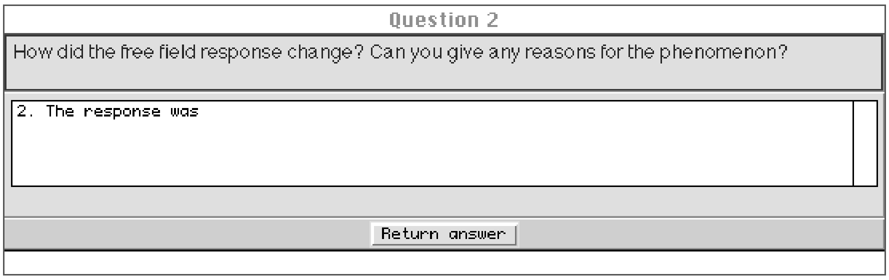

Fig 6.4. An example of question windows. The question is presented in the upper part of the window. The answer is typed in the editor view below the question. When completed, student clicks the "Return answer" –button, the answer is saved to disk and the window closes and the next measurement exercise opens. This question is related with the measurement window in Fig. 6.3.

The essay type questions require a human to check the answers afterwards, but the reason for using essay type of questions was that it is easier to control how well the student really understand what he or she is doing. The drawback is that it takes more time and teaching resources to check the answers. But in the beginning phase of the program essay type questions were useful to get a better understanding for the teacher and writer of the program on what is difficult in these measurements and how the students understand the whole picture and the program. Other kind of questions like multiple choice, right or wrong statements and numerical value questions could be added easily. The advantage of these kind of questions is that the computer can check the answers immediately, give feedback, even alter the difficulty level of the questions during the measurements. It would be also possible to make the program branch differently according the skill level of the student.

The measurement windows are customised according to the particular measurement the program presents. The first task when the student starts the measurement is to measure the free field magnitude response of the loudspeaker from zero degree angle. He then faces a question asking the frequency range of the loudspeaker according to the ISO-standard. The standard is presented in the measure-window and the question continues if there's any other way to define the frequency than the standard method.

The second window is very similar than the first one. The second window was presented in Fig. 6.3. The response is now measured from the same height, but twenty degrees from the side. The question is how the free field response is changing when compared to the first measurement and what is the reason for that. The third window carries on with the same theme asking to measure the same response from sixty degrees angle. The question is now if the response continued to change in a similar way than before and how should the response change in a well designed loudspeaker.

The fourth task differs a little from the previous ones. The response is now measured with one reflecting surface, the floor. The measurements are made in anechoic chamber and the metal net acting as a floor doesn't reflect sound like normal floor would. Therefore the student is asked to place a piece of hardboard lying on the floor exactly halfway between the microphone and the loudspeaker and then measure. The questions are what happened to the impulse response and why and how the student explains the picket fence effect on the magnitude response. In the first four measurements an RPFS excitation signal is used.

The fifth measurement is an easy computing task between other questions. There is no real measurement window and the student needs to use other equipment to help to achieve the result. The task is to measure the sensitivity of the loudspeaker from a distance of one meter with a power of one Watt fed to the loudspeaker. The nominal impedance of the speaker is given and there is a voltage meter connected to the loudspeaker terminals. The mathematics is easy, but the students have to figure out by themselves what needs to be calculated and how. The question presented is what was the voltage fed to the loudspeaker and what was the sensitivity of the loudspeaker measured from a distance of one meter. The RPFS excitation signal is filtered, its energy is mostly around 1 kHz. 1 W is quite a high continuous power for a small loudspeaker, therefore high frequencies are filtered out. Otherwise the tweeter of the speaker would be in danger of damaging.

The sixth task is to measure the distortion of the loudspeaker in 1 kHz sine signal. This task involves a real measurement window again, but differs a little from the previous measurement windows. Distortion measurement page is presented in Fig. 4.3. This time there is no impulse window and the frequency view is linear. What can't be seen, but possibly can be heard, is that the excitation is sine signal instead of RPFS-signal. Its frequency can be altered and distortion can be measured with different frequencies and sound pressure levels (SPL).

The seventh measurement window is merely a question. The used loudspeaker type with short description of the speaker is asked, so that the teacher can later verify the results. Students can also type feedback to the teacher here. The feedback is actually very much desired.

7 RESULTS AND THE USERS

7.1 Results from the teachers view

This type of education seems very suitable for this particular course. The author of this thesis was guiding the students in the spring 1997 without the *Simple Loudspeaker Measurement Program*. In the spring 1998 students using the program were followed very closely. They didn't receive any help with the measurements from the assistant. Otherwise the course and the measurements were similar, the same equipment were used, except for the computer and the program.

In 1997 some of the students were very passive, showing no interest at all for the measurements, making it very uninspiring to try to teach them. In 1998 this kind of behaviour completely disappeared. They couldn't any more afford to be passive because there was no one to help them or to make the measurements for them. They had to find out the things from the beginning and understand what they were doing. A good example is the question and measurement number five. The students needed to calculate the voltage fed to loudspeaker, when they knew that the power needed to be 1 Watt and the impedance was 8 ohms. Some elementary school students could calculate this with the help of Eq. 7.1.

$$P = UI \text{ and } U = IR \text{ yields to } U = \sqrt{PR} \tag{7.1}$$

Engineering students should be crushing this in a couple of seconds. Still, when the question was presented by human assistant a great number of the students couldn't manage to find the result even with some help and clues from the assistant. On the other hand, when the students faced this task from the program, almost all the answers were correct. One reason may be that they couldn't outline the problem merely with ears, but the pressure from human assistant was surely a factor.

With the program the students usually spent more time with the measurements trying to really figure out what was happening. Then again there were some deadlocks in 1998 which would have been quickly solved and passed with the human assistant, but maybe the students really learnt more by managing by themselves.

The students wrote longer answers in 1998 than in 1997 and the answers were better than a year earlier. It showed clearly that the answers were better formulated when there wasn't any pressure from an assistant waiting for the answer. It is easier to let the computer wait, it doesn't think that the student is stupid if the answer doesn't come immediately. This shows clearly in the answers.

All in all the students were more positive about the whole work because they were doing everything by themselves and they were thinking that the whole measurement was their own and there wasn't any outside person telling them what to do and how to do it.

The backside of the situation was that there were a couple of persons damaging the equipment by mistake. They chose too high levels for the input signal and the loudspeaker's tweeter couldn't handle the produced high frequency power. They simply didn't understand what they were doing and there was nobody to stop them.

There is always the possibility that the questions don't control the student's real level of knowledge efficiently. The student may guess the correct answer, but the human teacher always has some kind of feeling of what the student's real level of knowledge is.

7.2 Feedback from the students

The feedback received from the students was mostly positive. The students thought that choosing the computer to teach the measurements was a good decision. They liked the freedom and independence they had while doing the measurements. The program was regarded interesting and stimulating, the structure of the program was clear and easy to understand. The program worked reliably, it crashed only once and there weren't any other big oddities. Some students were already interested in loudspeakers before the introductory work so they naturally liked the measurements. Students also thought that the work gave them a good practical perspective on the theory learnt on the other courses previously.

However, some criticism was still presented. Not everyone liked the navigating system; they were used to the web browsers' navigating style and didn't like the slightly different method of the Simple Loudspeaker Measurement Program. Some misspelled words in the theory pages and instruction pages were also found and informed, luckily they were easily corrected. Some other features were also asked for. The most requested was the possibility to view the results from the previous measurements for comparing to the current results. In 1998 there wasn't any simple way to do this. The results could be saved and opened to another window or they could be printed, but it was impossible to view them simultaneously in the same window with the current results. A couple of students would have wanted to get results saved in Matlab-format for later researching.

The wiring of the system was quite a construction with different kind of unprofessional wires going hazardously to different locations. Some kind of safe and clear wiring panel with adequate cables would simplify the situation and decrease the risk of damaging equipment. It would be great to use some kind of restriction to the power level, so that too high signal levels couldn't be used. However, it is difficult to develop and take this kind of system into use, because the user controls the gain buttons of the measurement equipment. And still, even the one crash is too much, even though no information or answers was lost or damaged.

Something else than the loudspeaker to be measured was also hoped for. One student asked to see units on the graphs so he could easier realise the results. The question number four is related with the reflection from the floor measurement (see section 6.1.3) and it is: "What happened to the impulse response and why? How would you explain the "picket fence" effect of the magnitude response?" One student criticised the question; he found it difficult to understand. So there are improvements that could be done.

8 CONCLUSIONS

This master's thesis consisted of the design and programming the *Simple Loudspeaker Measurement Program* as well as holding the tutorials for the course *S-89.112 Acoustics and audio signal processing: the special assignment*. Because of the nature of Common Lisp and QuickSig, some parts of the program existed already when this thesis was started, they were only gathered and put together, possibly some modifications needed to be done. Most of pages however, didn't exist and they were programmed and created during the process. Before starting to write the code some analysing work needed to be done. For example, what was needed from the program, how could it be implemented, what kind of computer was suitable and what equipment was needed.

Some demands changed during the work. A couple of tasks proved too difficult and time consuming to implement. Some improvements to original plans were invented. One clear improvement came by accident, when it was noticed that Power Macintosh could be used rather than old Quadra Macintosh Computer. With this change some tricky parts could be left out from the code and the program wasn't dependent of one particular computer with a special signal processing card.

During the springs 1998 and 1999 it showed that the work had partly succeeded with the programming task. The program does what it is required and the results are good, but it could be more user friendly. It still crashes sometimes when the signals are zoomed and spanned. The program doesn't keep trace where it was at the time, if it crashes. Therefore the student needs to start from the beginning, if an interruption happens. The possible crashings depend partly on the existing QuickSig code, partly on the *Simple Loudspeaker Measurement Program*.

In the future the questions could be altered the way that the computer checks the answers and generates a complete measurement report, which it prints for the record. But this would be a demanding task. The program doesn't also notice if the same student has already used the program previously, therefore it is possible to write over the old answers, but this requires that the student wipes out the old answers on purpose. The navigating system of the program could be made better. At the same time it would be useful to show the student some kind of map of his or her advance.

At the moment the QuickSig Audio-response-window doesn't support multiple plots from different signals. This would be a very good improvement, making it possible to compare different measurement results, even different loudspeakers. Some kind of small audio amplifier with restricted output power would secure the loudspeakers. Now it is possible to adjust the gain of the audio amplifier too high and damage the loudspeaker.

It is obvious, that there are several things, which could be done, better or added to the *Simple Loudspeaker Measurement Program*. However, it succeeded rather well with its original purpose: to make the measurements more appealing, activate the students more, increase their knowledge level after the measurements compared to what could be achieved with traditional teaching, and remove some work from the course assistant. The thing that delighted the author most was the increased activity of the students and better learning results. And the good thing is that more properties can be added to the program, but it means taking some time and putting more work on the program.

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