Acoustic current source and Transfer function of the ear

Magistère des Sciences de la Matière, 2ème Année
École normale supérieure de Lyon

Mai/Juin/Juillet 2000

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Résumé

Le laboratoire d'Acoustique Musicale de l'Université des Nouvelles Galles du Sud a mis au point depuis 1995 une methode permettant de mesurer rapidement et avec precision le spectre en impédance de divers système, allant des instruments à vent aux conduits vocaux humains. Le but de ce stage est d'adapter l'appareillage pour le rendre capable de mesurer les fonctions de transfert de l'oreille humaine. Pour ce faire, il a fallu d'abord étudier et concevoir une nouvelle source de courant acoustique, qui soit suffisamment performante pour couvrir un large domaine de fréquence (typiquement celui de l'oreille humaine). Celle-ci a put alors être utilisée pour l'étude d'un modèle de l'oreille humaine, et ainsi déduire le rôle que joue l'oreille externe dans l'audition.

Abstract

The Musical Acoustics Laboratory of the University of New South Wales has developped during the last five years an apparatus that allow the fast and precise measurement of impedance spectra for various system such as wind instruments or the human vocal tract. The aim of this traineeship is to adapt this apparatus so that it could perform measurements of the transfer function of the human ear. To do so, one first needed to study and build a source of acoustic current, efficient enough to allow measurement to a large frequency range (typically the human hearing range). It has then been used to study the properties of a model ear, and then investigate the role of the outer ear in the audition of sound.

Contents

Introduction

Measuring transfer functions is important for systems that carry information over a broad band of frequencies. It can give a lot of information on the system studied and its limitations. One particularly interesting case is the outer ear. The sound measured at the eardrum is quite different from the sound near the head. Its transfer function is responsible for an important part of our sound localisation ability (particularly in the vertical plane) and to our auditory threshold. These objective data can be useful in physiology and for medical applications, for it sometimes allows the detection and the identification of some auditory problems.

The acoustics laboratory of the University of New South Wales has developed in the last five years a high precision apparatus for measuring acoustic transfer functions. The aim of this traineeship was to try to extend the capacities of the apparatus to make it able to perform a few tests on the external ear.

The first step was to build a new source of acoustic current, which could allow measurement over a large frequency range, typically the human auditory range. Because of its technical difficulty, this part of the project took a large amount of time. It is rather difficult to build an ideal current source with a good frequency response, and it becomes even harder when its dimension are small - for a wide range of projects, a source constrained in a pipe whose diameter is less than 1 cm is needed. As a result of the work reported here, the laboratory now has a source that can be used for any project, and which allows a better precision over a considerably larger frequency range.

I The acoustic impedance spectrometer

The acoustic impedance spectrometer developped since 1995 in the musical acoustic laboratory of the UNSW is a high precision apparatus for measuring acoustic transfer fonctions. It can be used easily for real-time measurement [6], for precise studies of musical instruments [7], etc... Its efficiency, rapidity and precicion has resulted in it is being cited as a reference technique for such applications (see for example [3] p. 222-223).

The spectrometer was originally designed to measure musical wind instruments, and to characterize them through their impedance spectra. The impedance spectrum at the input of an instrument describes how the bore interacts with the excitator mechanism (reed, air jet, etc..) and its extrema (the bore resonances) determine the partials of the note played.

The impedance spectrum is a property of the instrument alone rather than the musician. Each musician has his own way of playing, controlling the sound not only through the fingering, but with the excitator and his body. The impedance spectrum is an objective measurement of the instrument, thus of interest to scientists and to manufacturers

The principle is simply to apply to apply a calibrated broad band signal at the instrument in the place of the excitator, to measure the input impedance through a microphone and to calculate its Fourier representation. Nevertheless, this simple method involves more complex technique and ingenuity.

1 Apparatus

The acoustical current source consists of an enclosed loudspeaker and an exponential horn that serves as an impedance matcher to a small pipe, whose diameter is adapted to the measurement geometry (see figure 1).

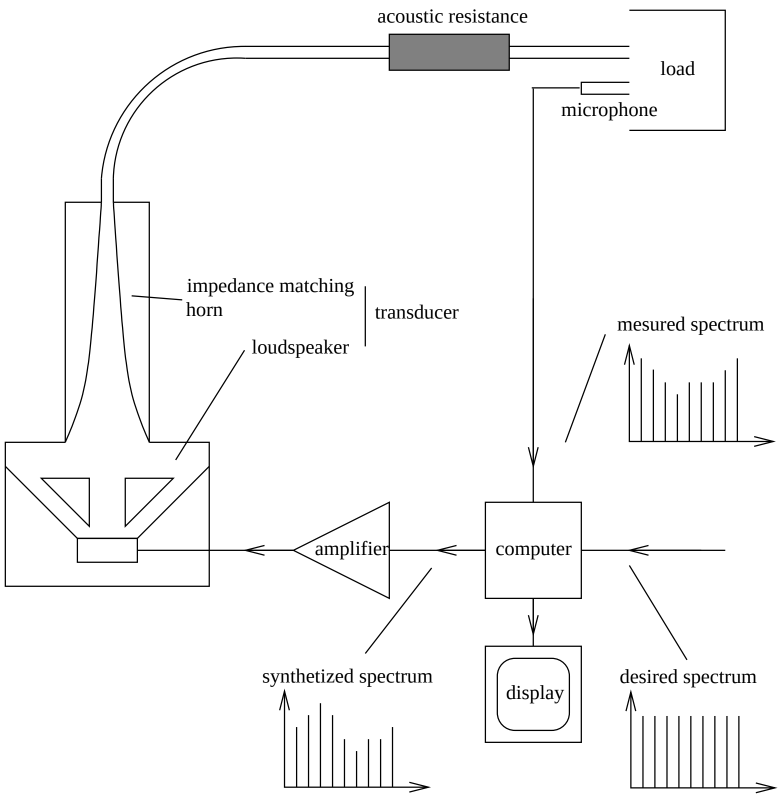

Figure 1: Schema of the spectrometer

The output is an the acoustic attenuator with a large characteristic acoustic resistance. It converts the pressure source of the loudspeaker to an acoustical current source. It is made of two coaxial cones of diameter 5 mm, separated by three wires of diameter 130 μm [5].

A current source is necessary there, for the impedance can be expressed by

$$\mathrm{Z}_{\mathrm{acous}} = \frac{p}{u_{\mathrm{ref}}}$$

where p is proportional to the pressure measured by the microphone that is plugged on its exit. A calibration procedure ensures that u_{ref} is constant (see next section). The ideal current source ensures that u_{ref} is independant of the load.

The software (written by John Smith) plays an important part in the process. Not only does it computes the impedance spectrum from the microphone signal, but, using a feedback loop and controlling the source, it compensates for the poor spectral response of the speakers and the transducer apparatus, as we shall see further.

2 Calibration

The frequency range involved in acoustics is large (in ideal conditions, ears work from 20 Hz to 20 kHz) and the dimension of the objects used is small, thus it is rather difficult to obtain a small source of acoustic current that is independant of frequency over 10 octaves. Most speakers have directional effects, diffraction effects and mechanical and acoustic resonances, so that their transfer functions greatly depend on the frequency. It get even worse when we want the source to be constrained in a pipe of small dimension (typically 8 mm for ours applications).

As we want to measure the transfer function of the load and not the speaker, we must ensure that the signal generated in front of the load is independent of frequency. That is the role of the calibration.

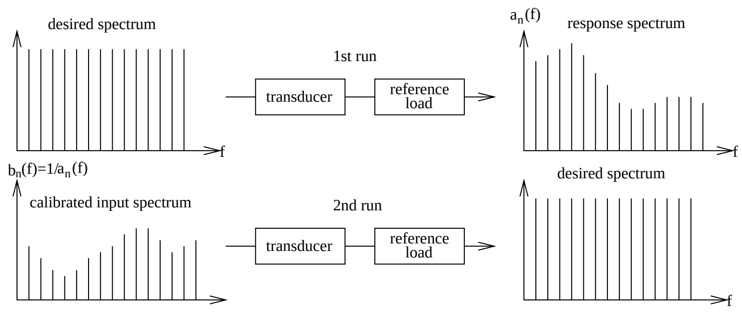

Figure 2: Schematic diagram of calibration

The apparatus is first connected to a reference load, whose spectrum is known (in the case of the wind instruments, a semi-infinite pipe is used for commodity, see [5] for more details).

The computer[1] first generates the desired spectrum, which is a sum of sinusoidal waves.

$$\mathrm{V}_1(t) = \sum_{n=n_{\min}}^{n_{\max}} a \sin\left(2\pi\, n\, \Delta f\, t + \phi_n^1\right) \tag{1}$$

were Δf is the chosen resolution in frequency and ϕ_n are the phases, selected to reduce the ratio $\max \mathrm{V}(t)/\sum a_n$ (see next section). The signal is then sent to the source and the resultant microphone signal is measured and analysed by the software. We obtain therefore the transfer function of the transducer and the part between it and the microphone (generally, the attenuator), say

$$\mathrm{V}_{\mathrm{dig1}}(t) = \sum_{n=n_{\min}}^{n_{\max}} a_n(n\,\Delta f)\, \sin\left(2\pi\, n\, \Delta f\, t + \phi'_{n1}\right)$$

Because of the problems with speakers (mentioned above) and the difficulty of producing a high value acoustic current in a delimited space, this spectrum is far from being flat ($a_n = a_n(\Delta f)$).

At the second run, the computer generates a signal for which each frequency component has an amplitude proportional to the inverse of the measured signal :

$$\mathrm{V}_2(t) = \sum_{n=n_{\min}}^{n_{\max}} b_n(n\,\Delta f)\, \sin\left(2\pi\, n\, \Delta f\, t + \phi_n^2\right) \tag{2}$$

where $$b_n \propto 1/a_n$$

Then, as the adapted spectrum cancels the frequency dependance of the transducer, we obtain the desired spectrum in acoustic current, and the signal generated by the computer at the second run (b_n) will be used for all the further measurements.

The desired spectrum can be either asignal independant of frequency (as used for this example), but also can be a spectrum in $1/f^m$; these two last cases are used when the noise at low frequency is strong.

[1]A Macintosh IIci with a 16 bit Digital/Analog Card (National Instruments NB-A2100)

3 Examples

For the flute [5], the calibration procedure involved a waveform which was a sum of frequency components equally spaced by $\Delta f \simeq 1.35$ Hz between 200 and 4000 Hz. The reference load was a semi-infinite pipe (42 m length, but the attenuation of the echo is sufficient to damp the refection under 80 dB, which makes it effectively infinite). The spectrum obtained after the calibration was independant of the frequency to ± 0.1 dB.

The performance of the whole apparatus is such that the calibration and the measurements only take a few seconds. It has been used for the RAVE (Real-time Acoustic response by Vocal tract Excitation) project, which intend to measure some important acoustic properties of the vocal tract non-invasively, in real-time, while the owner of the vocal tract is speaking (see [8]). It also has a excellent precision, and allow to measure the minima of impedance, which is crucial for instruments such as flute (other techniques do not give good precision on minima).

II Problems with the source

Even if its imperfections are digitally corrected, a few problems remain with the source, that limit its wider exploitation.

As we have said, it is difficult to have a high power source of acoustic current that is frequency independant and of small dimension. Thus, the signal obtained at the first run of the calibration often contains sharp gaps and peaks (see for example figure 3). The software can actually correct it, but due to the digital treatment, a few problems occur that harm the accuracy of the mesure.

The reasons can be inspected in three parts

- the signal/noise ratio, while measuring ;
- the analog to digital conversion ;
- the internal computation.

1 The signal/noise ratio

This problem occurs while the measurement is running, *ie* while the microphone is actually working. We simply want to avoid external noises, such as those in the building, and as we are working with very low signals, this is critical.

First, while the preparation is done during the day, most of the measurements are done during the night[2], when these external noises are minimal. Moreover, following the case, the measured instrument is placed in the anechoïc chamber or in a sound-proof box.

Second, we can increase the output level of the source, but it is desirable to remain in the linear range of the speakers.

Moreover, even if we have a signal with a correct average level, we must avoid having a generated signal with a minimum too low, as it could be surrounded by noise. This is done by carefully chosing the phases ϕ_n of each frequency component in the sum (1) in order to minimize the ratio $\max \mathrm{V}(t)/\sum a_n$. As

[2]It has been noticed on a set of measurements on the flute (1995) that they would grow better as the pitch of the fingering was higher. In fact, the measures have been done from the lowest pitch to the higher, in one night starting at 7 pm ; as the building became quiet, the signal/noise ration grew better.

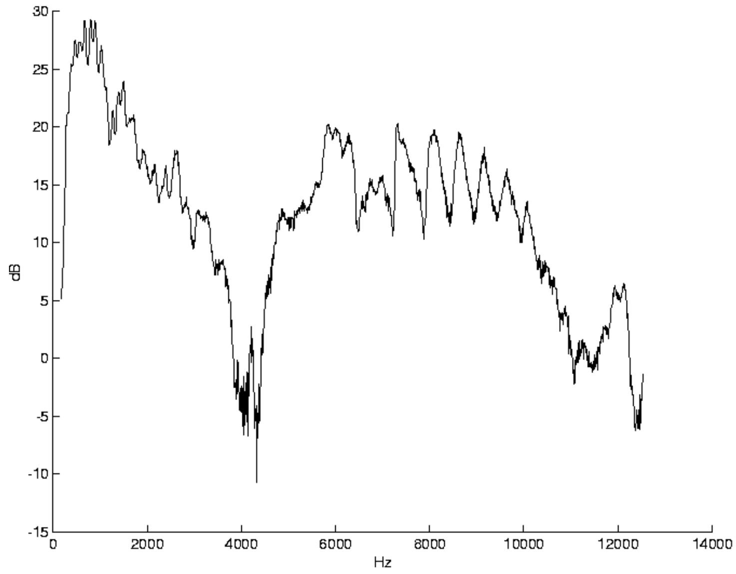

Figure 3: Spectrum given by the first step of the calibration in free field of a source

there is no general way of solving this problem, the phases are randomly choosen, until a satisfying value of this ratio is found.

Finally, there is also the problem that some noise can radiate from the bottom of the transducer (the only acceptable source must be the sound generated at the end of the exponential horn). Typical dynamic ranges are 60 dB, therefore a measure would be refused if the level of the sound due to the transducer's box radiation exceeded 60 dB below the level of the sound collected at the small pipe.

2 The analog to digital conversion

The ADC card works with 16 bits, which means that the signal will be coded over 65536 digits (± 32768). In the best case, the maximum of $\mathrm{V}(t)$ is represented internally by 32768.

The digital imprecision, caused by the acquisition is at least of 1 digit. The imprecision represented by this digit depends on the coding of the whole signal internally. This imprecision will be much more important when the max level of the input is low (see for example figure 4).

The value of the uncertainty caused by the one-digit-error during the acquisition is

$$\Delta e_{\mathrm{dig}} = \frac{\Delta \mathrm{V}}{\mathrm{V}} \approx \frac{1}{\max \mathrm{V}_{\mathrm{dig}}}$$

where $\max \mathrm{V}_{\mathrm{dig}}$ is expressed in digits.

As a consequence, we must choose carefully the gain of the amplifier behind the microphone, so that the maximum of the signal will coded as close as possible to 32768 digits (without going above).

Nevertheless, the internal uncertainty is in fact better than 1 digit. Indeed, the acquisition card has hardwar linear converters which work on 18 or 20 bits. As every acquisition is performed during more than one cycle, we get a much better precision. However, this feature is not documented and, according

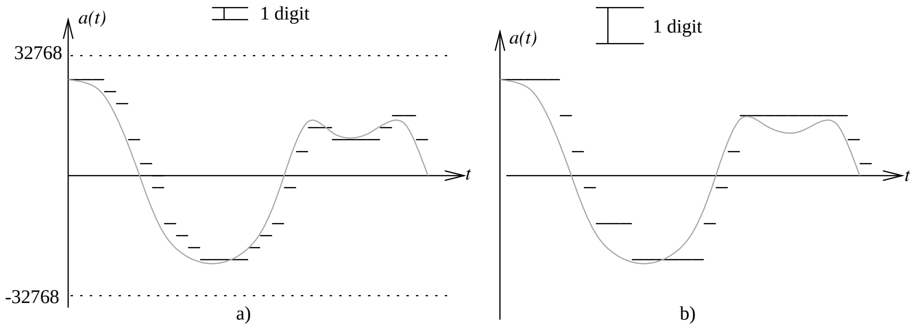

Figure 4: Example (schematic) of a digital acquisition a) of a signal with a good max input level ; b) of the same signal, but with an amplitude three times lower. Both of the frequency resolution and amplitude resolution have been exaggerated.

some caracteristic experiments by John Smith and some examples given by the constructor, a good assumption is to set the internal incertainty to 1/8 digits.

3 Loss of precision due to digitisation and improper frequency response

The first run of the calibration measures the transfer function of the source. As we already said, this spectrum obtained has gaps and peaks. There may be a big difference between the frequency component with the highest sound level and the one with the lowest. After the Fourier transformation, which gives the values of a_n and ϕ_n, there is an imprecision on everyone of these coefficients. Because the phases are chosen at random, and then the waveform has got a "random shape", it is quite hard to have a general expression of this uncertainty. We will made the assumption here that if a_n is expressed in internal digits, this uncertainty is in the order of 1 digit.

The problem is to know the consequences on the spectrum generated on the second run, given by (2). In the case where the desired spectrum is flat, each frequency has an amplitude proportional to $1/a_n$, the proportionality factor is set after the generation of the waveform, so that the maximum of $\mathrm{V}'(t)$ approcheas 32768. If a_n, expressed in digits, is known at about 1 digit (far less in fact, see previous section, but this assumption does not harm the veracity of the further results), then (using the binomial theroem)

$$\begin{aligned} b_n &\propto \frac{1}{a_n \pm 1} \\ b_n &\propto \frac{1}{a_n} \mp \frac{1}{{a_n}^2} \end{aligned}$$

Then the waveform (2) is generated and sent through the transducer. By virtue of the transfer function, the measured signal after the second run can be expressed as

$$\mathrm{V}_{\mathrm{dig2}}(t) = \sum_n c_n(n\,\Delta f)\,\sin\left(2\pi\,n\,\Delta f\,t + \phi'_{n2}\right) \tag{3}$$

where (if we neglect the error due to this second acquisition)

$$\begin{aligned} c_n &\propto a_n \times b_n \\ &\propto a_n\left(\frac{\mathrm{A}}{a_n} \pm \frac{\mathrm{A}}{{a_n}^2}\right) \\ &\propto \mathrm{A} \pm \frac{\mathrm{A}}{a_n} \end{aligned}$$

so that we have an uncertainty much greater for frequency with a low level. Thus, the signal generated on the second run does not fit exaactly the transfer function of the transducer and we won't be getting a flat spectrum after the second run.

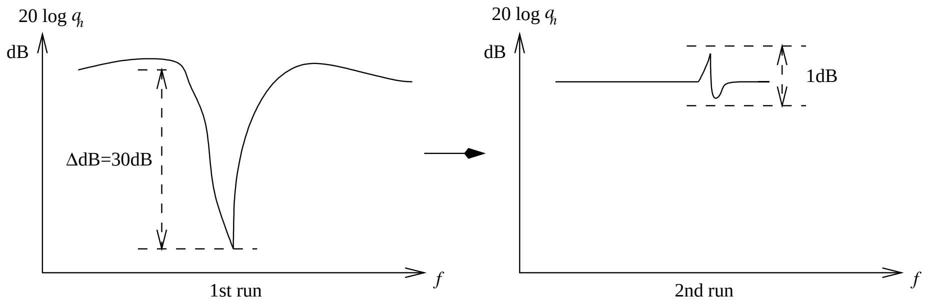

Figure 5: Example of a calibration where a gap occur. The numerical values have been taken from experiment.

This effect is important in the case of gaps. The figure 5 shows an example of a "flat" spectrum obtained if the frequency range contains a gap.

Most of the preceding transducers used had such gaps. As a consequence, the apparatus couldn't be used when a frequency range higher than 5000 Hz was used (there was a gap around this value for two of the transducers, including the one used for the flute research). Most of the precedings measurements were made over a frequency range that did not exceed 3000 or 4000 Hz. A large part of my project was to develop a source without gaps - *ie* one with a reasonably flat spectrum over a wide range of frequency. The goal was to improve the precision of results and allow the measurements in cases that require a spectrum ver a large frequency range.

III Improvement of an old transducer

1 The transducer

In order to sort out the reasons for these gaps, I have worked on an old transducer. It is composed of a loudspeaker (for frequencies below 1 kHz) embedded in 16 cm diameter cylindrical pipe. This pipe is closed above the speaker, and on the cover two auxiliary speakers (a mid-range speaker and a tweeter) are mounted, just next to the entry of the exponential horn (see figure 6).

The original transfer function is presented on the figure 7. It has been obtained after the first run of the calibration into a reference load which was open air (in the achenoic room)[3]. There are several gaps, the first and more serious being at 4000 Hz and with size at least 42 dB. Gaps of this kind are really a nuisance, because not only do they introduce very low level frequencies component but also because it is difficult to measure precisely their actual lowest level. The frequency resolution that is used ($\Delta_f \simeq 1$ Hz) does not allow a good precision on these. There are also others gaps, and one really deep around 9200 Hz but these were not so critical to the musical acoustics, which do not often go to high frequency.

[3]All the further measurements are done with this reference load.

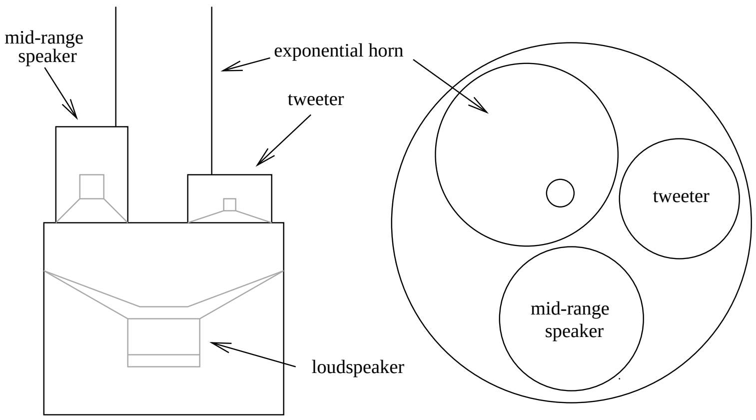

Figure 6: Original design of the old transducer, viewed a) from aside, b) from above

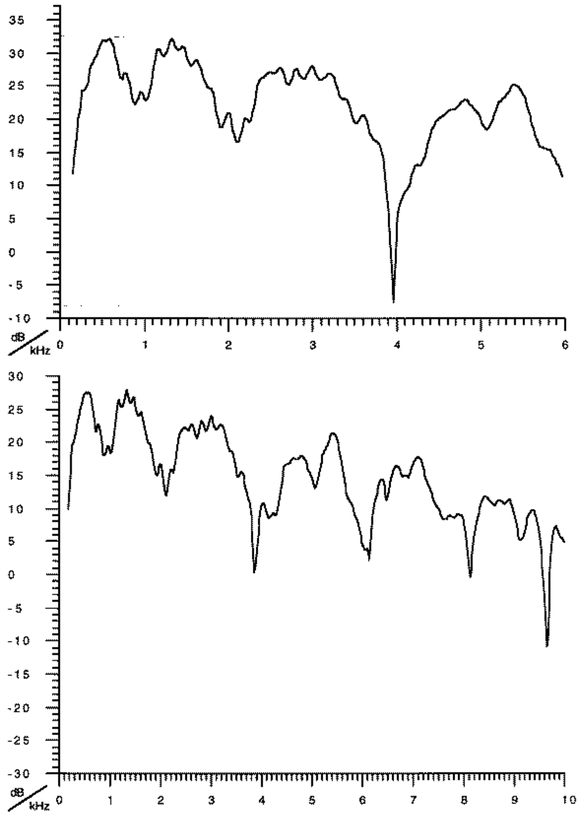

Figure 7: Spectrum of the transducer before any modification

2 Causes of irregular gain

My first job was to determine the origin of the gaps, before trying to fix them. I mainly focused on the gap at 4000 Hz, because it was obvious and sharp. I dismantled the transducer and carefully inspected each of its part, then I did some cross-experiments to investigate the predictions made. To sum up, there are in this transducer two main causes for gaps.

2.1 Resonances

Resonances occur in the bottom box and cause a low amplitude for some frequencies at the exit (see for example [9]). For a given geometry, standing waves can be generated and the corresponding half-wavelength leaks. Two of these resonances have been identified.

- The first one is caused by the whole length of the transducer ($94,3$ cm so its harmonics are separated by 180 Hz) and is responsible for the small oscillations in the begining of the spectrum. They can be reduced by damping which is achieved by adding some polyester in the bottom of the transducer and in the termination pipe.

- The second, which seemed responsible for the gap at 4000 Hz, appears to ge a resonance between the cap of the bottom box and the surface of the loudspeaker.

For the latter, it was hard to say because the loudspeaker is not flat (but the average distance corresponds well), and when it is replaced by a plastic plate, too many gaps appear to be conclusive. To confirm its source, I put an extra piece of pipe between the loudspeaker and the cap, so that the resonance should be shifted down. It worked - with an extra length of 5.6 cm a gap appeared at 2900 Hz - but the gap around 4000 Hz remained (although its shape had changed). So another cause had to be investigated.

2.2 Directional effects of the speakers

As this gap remained even when the two auxiliairy speakers were tested independantly, some directional effects were suspected. They often occur for high frequencies, and that was the case here. The spectra of the speakers were almost flat when measured in free field at their zenith, but serious leakage appeared when the microphone was moved from this position. As the entry of the exponential horn was off axis, the lobe for some of these frequencies was not be collected and missed in the output spectrum.

3 Improvment

As the causes were known, I logically worked on trying to fix the problems. The ideas were to keep the geometry of the transducer, try to break the symetry that allowed resonances and to diffuse the sound to avoid directional effect.

I have tested the effect of ten damping materials and wireframes as filters for sound, looking for the best one. We were looking for two main characteristics :

- a ΔdB as small as possible : the difference between the highest level and the lowest level should not be too important ;

- no sharp gap : to have a good precision during the acquisition, we wish to obtain the smoothest spectrum as possible.

I have retained two materials : an acoustic polyester foam and an underlay sheet ; these are actually used in sound insulation. After reconstrunction, the old transducer had these improvments :

- The main speaker has been removed and replaced by one with better specifications. It has a coaxial independant tweeter stuck on the loudspeaker that could pointed in any direction, so that the main power in high frequencies could be directed in the exponential horn.
- As a consequence, the auxiliary speakers weren't needed anymore. The auxiliary tweeter was kept to add some power in high frequency.
- A 7 mm sheet of underlay was placed above the loudspeaker but under the tweeter (the underlay appeared to be a low-pass filter, so that it can't be used directly as a filter for its absorbance in high frequency is too strong). It has a good diffusing damping characteristics by reflection and prevent the occurance of resonances.
- A small (5 mm) sheet of polyester was put above all this ; for this thickness, the polyester has only weak damping that affects high Q resonances, so it makes any gap smoother.
- I also tried a high-pass filter for the electric signal that was supposed to shift down the power of low frequencies, but it appeared that it wasn't really needed.

I also had good results by using some small balls of polystyren as a sound-diffuser, but this method hasn't been retained, for it is mechanically unstable, would require the user not to move the transducer at all (the calibration is done for a certain shape of the whole transducer apparatus, and between the calibration and the measure, this shape isn't supposed to move).

The spectrum is presented on figure 8. This upgrade had nevertheless a high price. First the average level of the output had slightly gone down due to the damping. As a second consequence, the sound insulation was poor, and the leakage noise coming from the bottom box was no longer 60 dB below the average level.

Third, the spectrum became sensitive to mechanical shocks. Indeed, the calibration is done for a given configuration of the source and the connecting pipe after it, as said before. We often have to move the source between the calibration and the measurement ; if a change occur in the configuration during this manipulation, then the measurement is flawed. This is why we have to design sources that are robust and resistant to shocks. The old transducer wasn't ; I have tried to work it out by adding a kind of cap, but the sound insulation became worse and the result wasn't convincing.

Finally, even if the spectrum was better after these fixes, the transducer was absolutely unusable. But that wasn't important, for it wasn't used anymore. By doing these improvements I nevertheless got a good understanding of the function of each part of the source and then got ready for the design of a new one.

IV Building of a new transducer

The aims were to produce a flat spectrum to the largest frequency range possible, to have a good sound insulation, a higher output level and a good resistance to shocks.

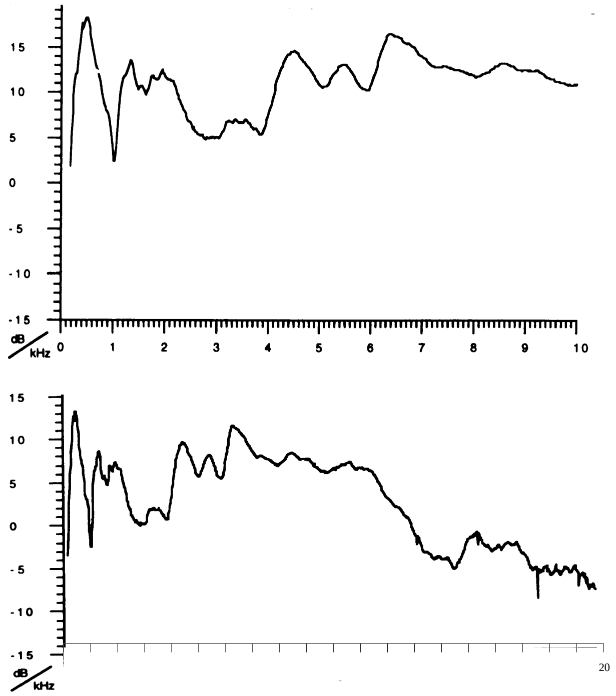

Figure 8: Spectrum of the modified transducer

1 Design

Most of the ideas from the older one have been reused, so basically the new transducer is superficially made like the others.

1.1 The conical shape

I considered the design of an exponential horn whose input diameter matches the diameter of the lower box, but it would have been impractical. For a horn with radius

$$r = r_0\, e^{-\frac{z}{l}}$$

waves do not propagate for which the wavelength $\lambda > l$. The exponential horn used so far have got a cut-off frequency of 200 Hz, which is still acceptable for musical applications, but we shouldn't go higher. Continuation of this horn could lead to horn impractically long.

A compromise was chosen. The first part of the horn is a cone, connected to an exponential horn. Thus, the same mould for the exponential shape could be reused (it is quite hard and it takes a long time to manufacture a exponential shaped-mould). The conical mould was matched to the exponential one, and the whole horn was cast in one piece of plaster.

I also had to deal with a way of protecting the plaster (quite fragile) in a plastic pipe adapted to its shape. That has been made in a simple way with several plumber pipes.

Because of the shape of the cone, there was no longer a rigid, hard wall opposed to the speaker, and the resonances that would occur in older design were no longer a problem. But the cone added its own resonances, and they were significant (harmonic spaced by 1 kHZ, 20 dB amplitude, until 20 kHz at least), even if their shape on the spectrum was smooth enough to have a good precision on them. The connection between the conical and the exponential shape as been rescaled smoother but that was not sufficient. Resonances, as many physical problems, are derived from boundary conditions, and a simple way of controlling the gaps is acting on the geometry of the problem. I added some underlay against the cone, and the resonances vanished.

1.2 The speakers and the lower box

This part of the transducer received the same treatment as for the old one. Combined with the conical shape, there was no rigid surface in the lower box, and no sharp gaps were present.

The same speaker with the concentric tweeter was used. To avoid any movement in the lower box, every piece of polyester or underlay was rigidaly held in place between the speaker and the pipe. To prevent the tweeter moving, it has been glued on its base.

2 Consideration on the microphone used

The principle of the impedance spectrometer is based on the fact that microphones are better than speakers. Even if speakers are not perfect, it is possible to measure their flaws with a microphone which frequency response is flat.

The microphone used was an omni-directional electret one. Its frequency response is given on figure 9. As we can see, the assumption that it was flat isn't correct, particularly around 16 kHz. In fact the microphone overestimates the high frequencies.

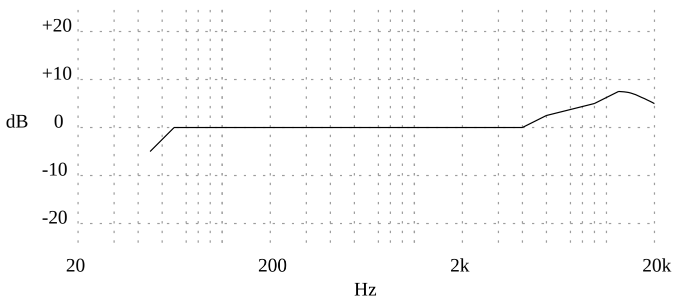

Figure 9: Frequency reponse of the microphone used.

Nevertheless, the results obtained there are not flawed. The frequency response does not contain any gap, so the spectrum of our transducer is as smooth as expected. In high frequencies, if we substract the transfer function of the microphone, we can see that the spectrum still remains under 15 dB. So the new transducer could be used with no surprises with a more adpated microphone in this frequency range.

3 Result

Due to the shape of the embedding box, the transducer is quite robust. Also as a consequence it has a very good sound insulation (nevertheless, there is some low frequency noise radiation from the lower box that necessitates the use of an isolating box in experiments on free field), and is resistant to shocks. It spectrum is given on figure 10 and shows that we achieve our goals. The ΔdB doesn't exceed 15 dB over the range of 20 kHz (the DAC card has got a

Nyquist rate of 22 kHz, moreover this is the limit of the human ears). Moreover the conical shape provides us with a much better output level than with a direct connection.

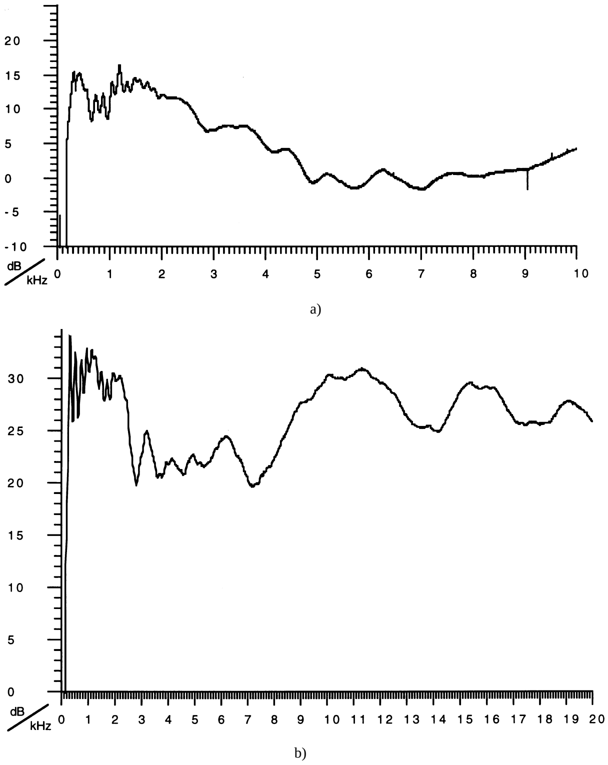

Figure 10: Spectra of the new transducer. a) the lower part of the transducer is put in a box that prevent low frequency radiation ; b) with no box

A few tests done from 0 to 10 kHz in free field show that the spectrum after calibration remains frequency independant to ± 0.2 dB. This transducer will be used for further research and publications in the laboratory.

V A few properties of the external ear

1 Design of an artificial external ear

As experiments on humans were impractical, I had to build an artificial ear for measurements. To gain some time, I have done the whole ear by myself, rather than purchasing a prothesis (for profesional use) or sending the work to a professional sculptor. From one person to another, the shape of the ear can vary considerably, but there are a few parameters that do not change. I used three different sources of anatomical sketches, and the final work reach well the requirements for a fake ear.

The auricle consists of elastic cartilage covered by skin, which adheres tightly to it. According to Békésy [11], the acoustic stiffness of bone covered with skin is such that no serious error will result if the head and meatus (apart from the eardrum) are regarded as completely rigid in respect to aerial vibration. Thus, the ear can be safely be made in modeleing plaster, which is the most convenient material.

The acoustic meatus is a kind of almost cylindrical pipe, which is about 2.5 cm long and 8 mm for the largest diameter. It has an S shape, and its radius goes shrinking to about 6 mm on the 3/4 of the section, then goes larger again until the eardrum. A mould has been made from modeling plaster, and the meatus itself has been cast on this mould in latex rubber.

The eardrum itself was not designed. It has a too complicated transfer function to be reproduced, so it has been replaced by a microphone for some measurements, and was not included in the measured gain of the external ear. For reasons of convenience, the size of the artificial ear was bigger than an average real ear by 150%. As a consequence, the frequency range of the measures done with this ear will have to be rescaled in respect to this size. It has several adavantages, such as making more convenient the design itself, and also shifting down the frequencies so that we keep in the linear range of the microphone.

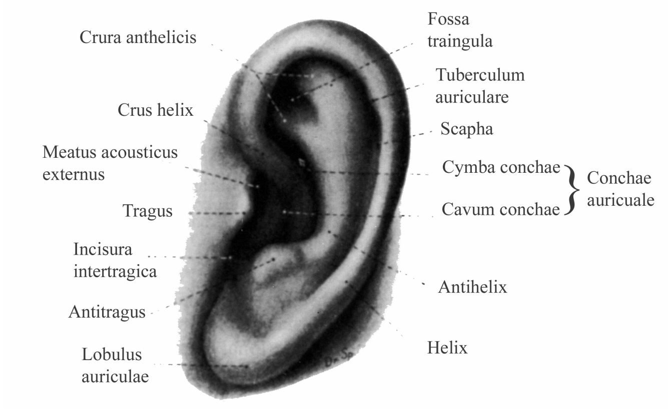

Figure 11: Left pinna (from [12]). This sketch has been used (with others) for the design of the fake ear.

2 Measurements techniques

Usually, the measurements on human ear are done in open air, in an anechoic room. The signal is a Dirac delta, supposed to contain all the frequencies of interest, and calibrated with respect to the spectrum obtained in free field. A full set of measurements using a dummy head has been done by Bill Gardner and Keith Martin in May 1994 at the MIT media lab, using this technique.[4]

It has the non-negligible advantage of avoiding any problem of reflection between the source and the microphone - as the signal is generated and measured during a period of time less than the delay the sound takes to travel and back to the source - so no such resonances can occur. But as a consequence, frequency resolution is lost.

The technique I used was quite different. The signal generated was a broad band signal, so I had to deal with the problem of the resonances between the air and the source, and this was one of the biggest problems. The calibration could not be done in free air, for it was no longer valid when the ear was put between the ear and the microphone for the measurement. So all the calibrations for measurements of the meatus have been done in presence of the ear, with the microphone just before it. A flat spectrum was then obtained at the entrance of the ear. For the measurement, the microphone was put inside the ear, and as a consequence, I was measuring the gain of the ear (or the part of interest), which is the most interesting data for audiologists. The results were very sensitive to

[4]They made the datas, results and documentation available on the web : `http://sound.media.mit.edu/KEMAR.html`.

any relative movement of the ear and the source, so as the measurement implied manipulations of the microphone, I had to ensure that they would not change the geometry of the apparatus.

3 Effect of the head

When it comes to audition, the actual shape of the head does not matter [12]. Therefore, we used for preliminary study a simple baffle plugged on a 2.5 cm cylindrical pipe closed at one end. This was supposed to give us an idea of the spectrum of the auditory meatus, the role of the lateral part of the head, and also familiarize with the forthcoming problems of measurements.

Three sets of measurements have been done : one with no baffle, a second with a small baffle, supposed to stand for the pinna, and a last one with a 9 cm radius baffle, which is the effective size of a head. The spectra are given in appendix, figures 12, 13, 14.

The formal treatment of this problem (see [4, 3]), for the pipe with no baffle let us expect a spectrum given by an ideal pipe of length

$$l = \mathrm{L} + 0.61 \times a$$

where $\mathrm{L} \simeq 25$ mm is the actual length of the pipe and $a \simeq 6$ mm its radius. The calculated frequencies[5] of the maxima are therefore 3016 Hz and 9047 Hz. The measured frequencies are slightly lower, this little error may be caused by a a bad value of the speed of sound (which is linked to the temperature).

In the case where the pipe is flanged by an infinite baffle, the spectrum is given by an ideal pipe closed at one end and whose length is given by

$$l = \mathrm{L} + 0.85 \times a$$

In fact, the baffle just needs to be rather bigger thant the pipe, and thus this expression should fit the spectrum 14. Therefore, we were expecting to have a local maximum for the frequencies 2871 Hz and 8614 Hz. As before, our measurements underestimate these values.

The head seems to have no particularly strong influence over the spectra of the external ear. It shifts down a bit the frequencies of the maxima of the auditory meatus but does not really modify its spectrum. The head, combined with the pinna, plays an important role in the localisation of the source of sound. Due to lack of time, this important aspect of auditory properties could not be started on.

4 Gain of the model external ear

I interested myself in two particularly cases : the gain of the ear under direct incidence and at an incidence of 45°. I have studied the concha and the auditory meatus separately, to be able to characterize their own role.

4.1 The auditory meatus

For these measurements, the calibration has been done by putting the microphone in the concha, just at the begining of the auditory meatus, and another microphone (same model) was put instead of the eardrum, closing rigidly the end of the meatus. The results are presented in figures 15 and 18.

As said before, the auditory meatus has a closed-pipe shape, so basically, the spectra obtained is not qualitatively different from those obtained for the preceding experiments. The main characteristics of these spectra are two strong

[5]for a temperature of 24° and a humidity of 40%

maxima around 3000 and 9000 Hz. The actual gain of the first maximum would not be so pronounced, for the eardrum which was not rigid as is the case here.

The incidence does not seem to play an important role in the gain, for both spectra are almost the same. This part of the ear does not play an important role in sound localisation, and this is the role of the pinna.

Also, the spectrum for frequencies higher than 12 kHz should be handled with care. Indeed, for these frequencies, transverse modes appear in the meatus, and the level can vary a lot radially. As our microphone has a radius of 2 mm, the spectrum may not be accurate in this frequency range.

4.2 The concha

The shape of the concha is harder to characterize, for it does not fit any simple geometry. As a first consequence, the spectrum greatly depends on the incidence. For direct incidence (figure 16), the concha acts like a small cone, which gives a maximum gain for a frequence 5000 Hz. For 45° incidence (see figure 19), it gives the same maxima, but also adds a gap around 11000 Hz, whose origin is not easy to determine. Gardner and Martin have also observed this gap, and supposed that it came probably from a concha resonance.

4.3 The whole external ear

The gain of the whole ear correspond to the sum of the gain in dB of the pinna, the concha and the meatus. Its main caracteristic is the maxima that occur between 2000 and 5000 Hz. This feature, caused by the two first resonances of the meatus and the concha, is important for the comprehension of the spoken language. There is also a second maximum, caused by the second resonance of the meatus, but weaker.

Even, as we said, if it does not correspond to the actual transfer function of the external ear (due to the fact that we do not have any eardrum), it is remarkable that it is enough to make a few assumption on it. For example, it is enough to interprete the human auditory threshold. The equal-loudness contours (obtained with some subjective tests in audiology, see figure 21) have got a strong minimum near 4000 Hz and a weaker one near 9000 Hz, which correspond to the maxima in our gain spectra.

Conclusion

Because of the technical difficulties involved in designing and building the new transducer, I had not all the time required to perform all the tests I wanted to do, particularly on the spatial localisation of sound. The measurements made on the ear were nevertheless accurate and precise, and show the capability of such an apparatus in the domain of human audition. This apparatus is also now upgraded, thanks to a new source which will allow a better precision over a large frequency range for any project using this method.

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A Spectra

1 Cylindrical pipe

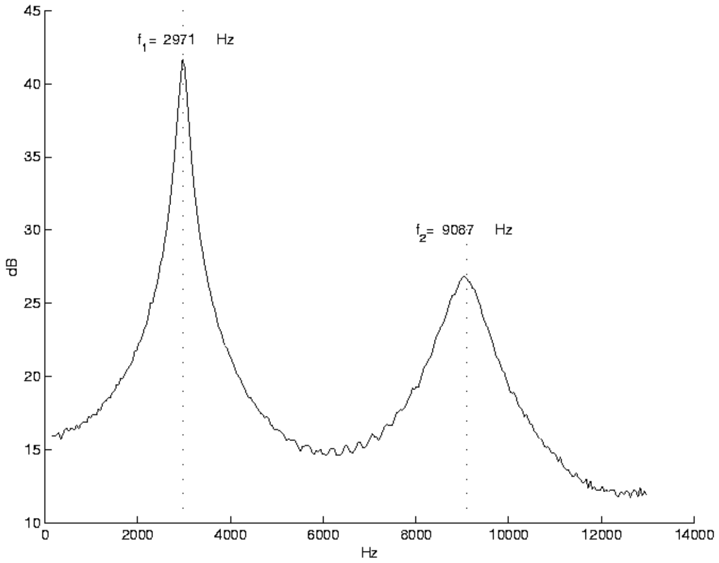

Figure 12: Gain of a 25 mm pipe closed at one end

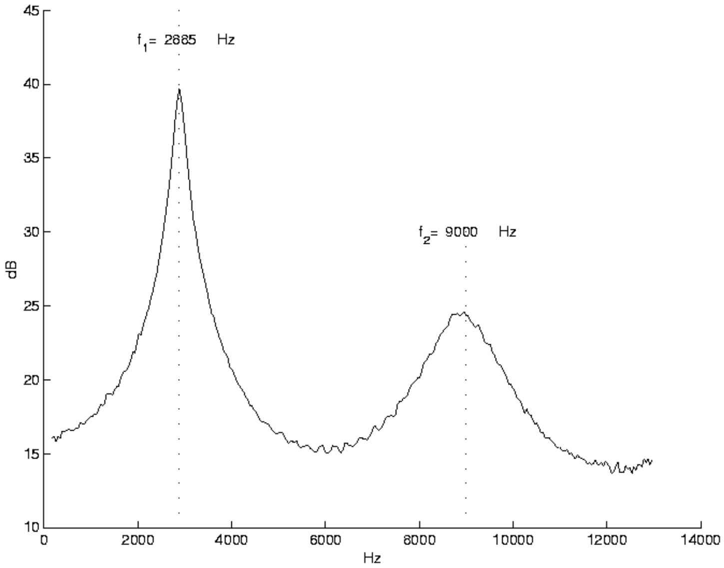

Figure 13: Gain of the same pipe, with a 3 cm radius baffle

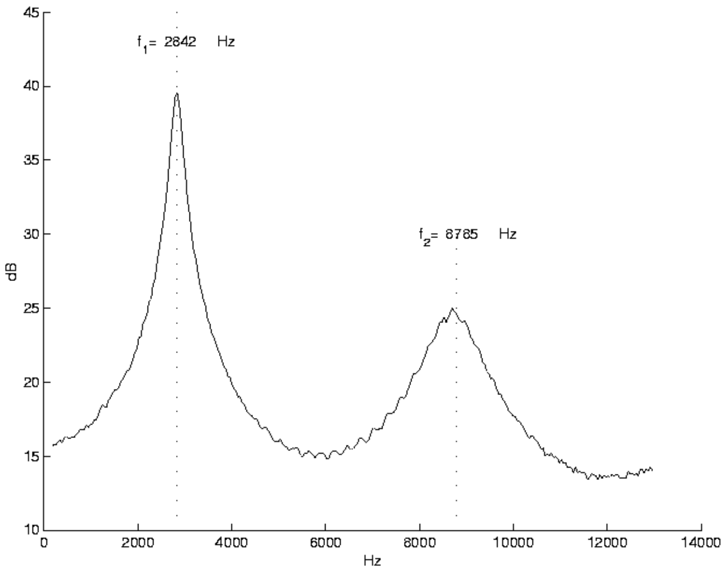

Figure 14: Gain of the same pipe, with a 9 cm radius baffle

2 Model ear, at 0° incidence

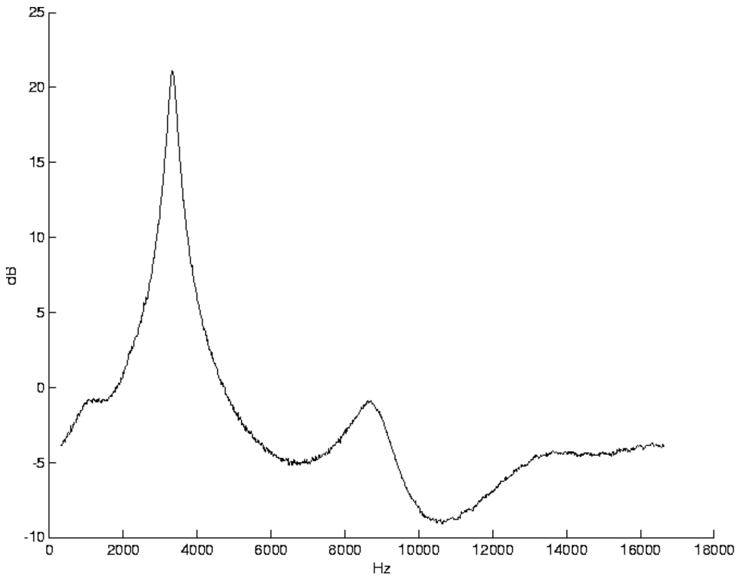

Figure 15: Gain of the auditory meatus of the model ear, at 0° incidence

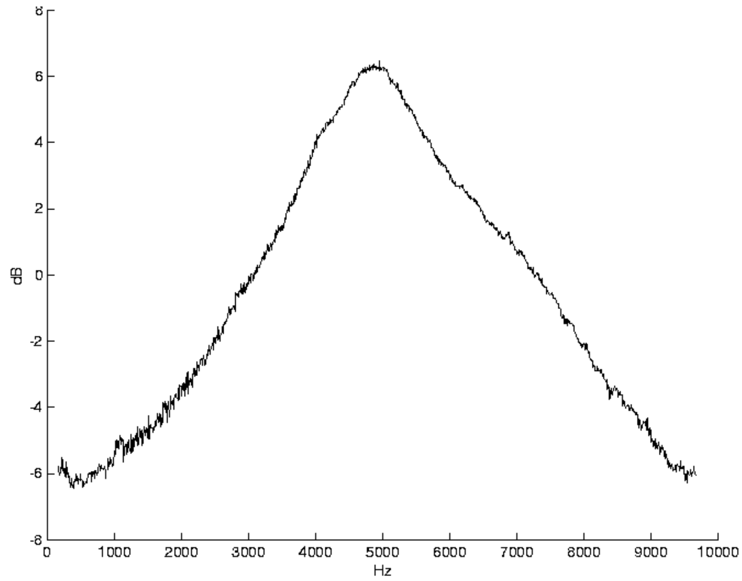

Figure 16: Gain of the concha of the model ear, at 0° incidence

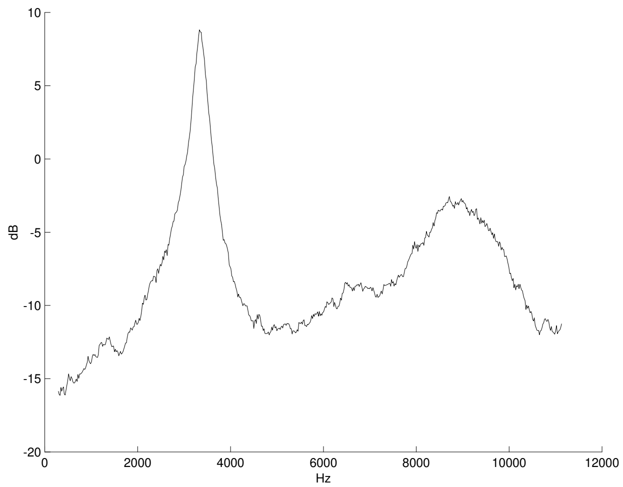

Figure 17: Total gain of the model ear, at 0° incidence

3 Model ear, at 45° incidence

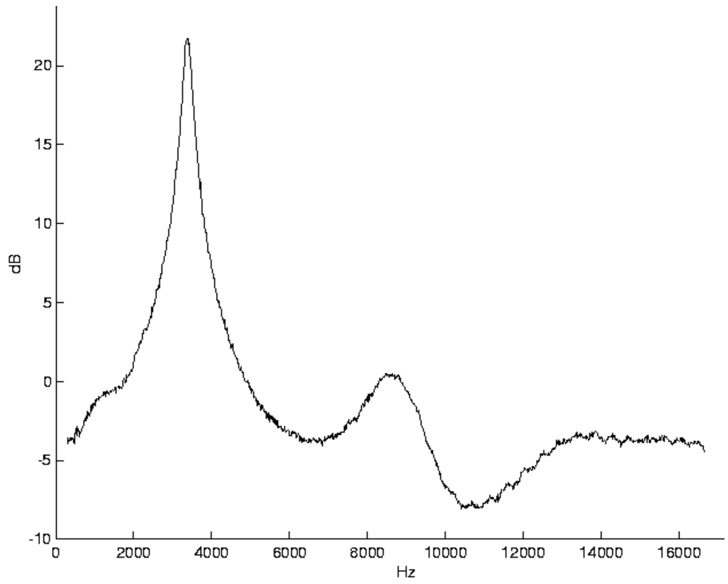

Figure 18: Gain of the auditory meatus, at 45° incidence

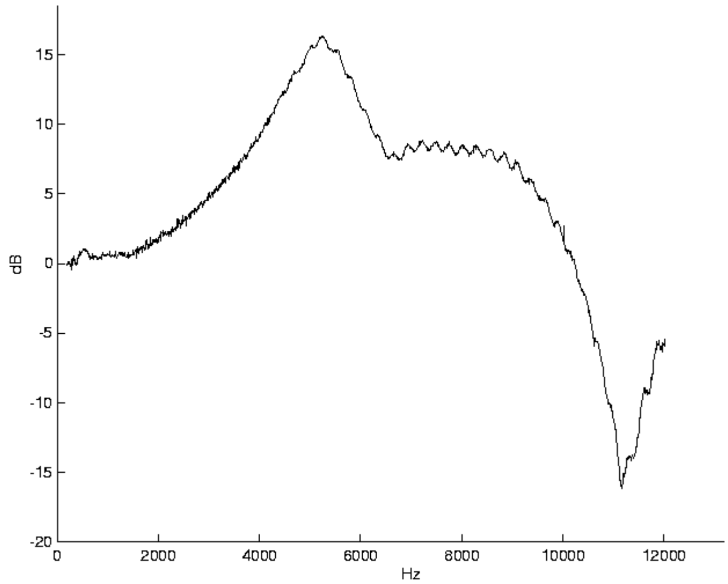

Figure 19: Gain of the concha, at 45° incidence

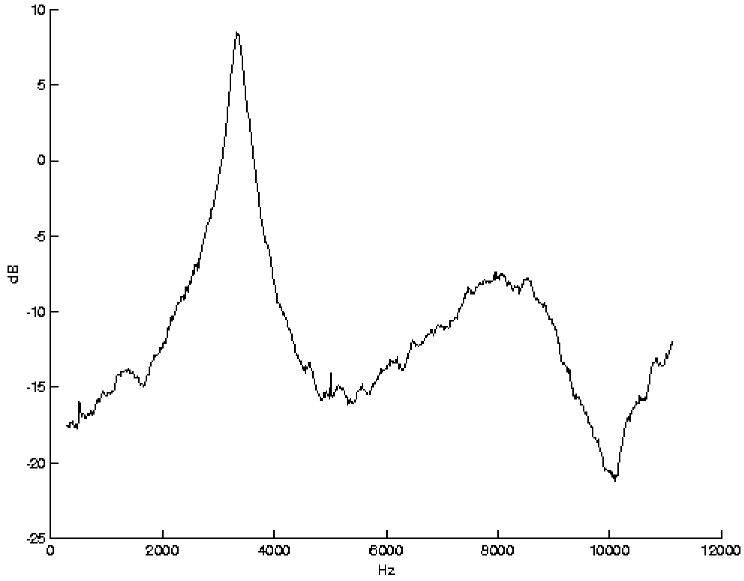

Figure 20: Total gain of the model ear, at 45° incidence

B Auxiliary data

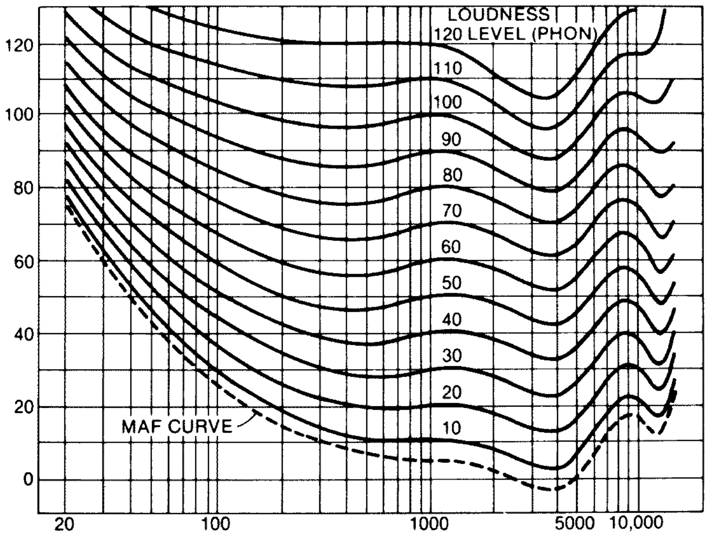

Figure 21: Equal loudness contours, from [12]

Acknowledgements

I cannot possibly end this report without trying to thank the people I have met there, for their endless support and the good time we had together. So, my most sincere thanks to

– Joe Wolfe, who gave me the chance of studying and working in Australia, in the most "user-friendly" laboratory in Australia. He was also able to cope with me as a room-mate for more than eigth days and as a student for the duration of this traineeship.

– John Tann, the very first man to trust on the Titanic, who cheered me up and helped to solve a considerable amount of problems. He is also reponsible for a large part to the good ambiance of the lab.

– John Smith, for his endless patience with me.

– Ra Inta, who was my lab-mate and the only person I have met in Australia who could not speak french.

– Janet Bell and Tom Molomby, who gave a home to a stranger and a lot of support in everything needed by a foreigner overseas.

– Little Tom Molomby, with who I learned so much.

– All the student that worked in the laboratory, with a special mention to Jane, Wendy and Manfred.