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CHARACTERISTICS OF SOME MUSICAL INSTRUMENT SOURCES FOR ACOUSTICS SIMULATION AND AURALISATION

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1.INTRODUCTION.

Originally, the goal for characterising musical instrument sources, was mainly to know how to make good instruments. A field of research has devellopped, trying to qualify and quantify, what the instrument maker is able to do due to his experience. This field includes different domains.

The first one links the musical tones and the tone qualities we perceive. This field uses psychoacoustics.
Then, it is logical to try and characterise the influence of the different parts of the instrument on the desired qualities. Therefore, we need to know the way the instrument works, this means elaborate physical models , which can also allow to simulate it.

You also need to know the influence of the shape, and the assembly of the different composants on the produced tones, which constitutes another field of research.

At last, you can study the instrument intrinsequally, and try to understand the way it is going to spatially emit the sound . This field is the study of the music instruments radiation, which this paper deals with.

2.SOURCES CHARACTERISATION

Global study.

In the global study of sound sources radiation, there are two different methodologies. The first one insists on the accuracy and the reproductibility of the measurements, even if they are very far from reality [MAR42],[MEY64], and the other one uses an approach close to a sound engineer one i.e. by considerating the couple instrumentist - instrument as a source [JAN78], [GARC83].

Our method

In June 1987, was organized by P. Jouenne and C. Semidor, a measurement campaign, during a tour of the National Bordeaux Aquitaine Orchestra. The idea was to record in an anechoic room about fifty musicians , located in the « center » of a sphere of 16 microphones, in « real conditions » of perfoming.

Our method relates to the notion of acoustical flux through a fictive closed region around the source we want to study.
The solution to the propagation equation for a potentiel φ, in the case of a harmonic diverging spheric wave is :
$\varphi = \frac{A}{r}\exp(iw(t-\frac{r}{c}))$, where A stands for the strength of the source.
Whence the acoustic pressure $p = \rho\frac{\partial\varphi}{\partial t} = \rho iw\frac{A}{r}\exp(iw(t-\frac{r}{c}))$
and the normal velocity $v = -\mathrm{grad}\varphi = \frac{A}{r^2}\exp(iw(t-\frac{r}{c})) + iw\frac{A}{rc}\exp(iw(t-\frac{r}{c}))$
The mean total flux through the closed region S is :

$W = \frac{1}{T}\int_0^T\iint_S \vec{I}.d\vec{S}\,dt = \frac{1}{T}\int_0^T\iint_S \overline{p\vec{v}}.\vec{n}dS\,dt$ where $\vec{I}.d\vec{S} = \overline{p\vec{v}}.\vec{n}dS$ is the flux of the intensity I through dS.

In far field conditions, i.e. $r \gg \frac{\lambda}{2\pi}$, the expression in $\frac{1}{r^2}$ becomes negligible, p and v are in phase , and after and integration on a sphere of ray R (chosen as the closed region), we obtain
$W = \frac{4\pi R^2}{\rho c}\overline{p^2}$
Whence if L_p is the mean level on the sphere and L_w the acoustic power level of the sound source.
$L_w = L_p + 10\log(4\pi R^2)$
The expression that we established for a monopole remains accurate, for a multipole source in far field conditions ($R \gg \lambda$).
In the measurement technique we used, the reference surface is a spheric one with a spatial sampling corresponding to an isocaedron.
As we former explained, we only use 16 microphones located on a sphere . Each microphone corresponds to an elementary surface of the sphere. The total sound power level is approximated with the pression levels measured on each microphone.

$$\overline{L_P} = \log\frac{1}{4\pi r^2}\left[\sum_{\substack{i=1\\ i\neq 10}}^{16} S_i * 10^{0.1L_{Pi}} + \frac{S_i}{3}10^{0.1L_{P10}}\right]$$

L_{Pi} : level on micro n°i.

Where $S_1 = \dfrac{S}{20}$ where S is the total surface of the sphere.

Whence $L_W = \overline{L}_p + 10.\log \dfrac{4\pi r^2}{S_0}$ $\qquad S_0 = 1\ m^2$ (reference surface).

3. ANALYSIS

In a first approach, we don't want to study specifically the tone of the instrument, but try to extract from each sequence a global information (if possible) usable for simulation. Therefore we have to conceive a method to reduce the information.
The study we made consisted in a comparison between a sample and a mean value (on all samples). In this case we have to check if the mean value is characteristic of the distribution of samples. We find a good stability for high levels (forte), this is due to the fact that the reference mean favours the high levels. We obtain better results by eliminating at first the exeptionnal character samples.[COU97]
We don't have significatives results for the percussive instruments, this is certainly due to the fact that we don't treat the attacks specifically in our study.

To illustrate our results,. we created a 3D representation of the emission. We show the results we obtained for the emisision of the spoken voice (illustration 1), the rest of the database can be found in [COU97]. The evolution of the patatoid of directivity can be shared into threee zones.
Around 125 Hz, the radiation shows a symetry of revolution around a vertical axe containing the body of the speaker, the radiation is a bit more intense in front .
Around 500 Hz, the source has the same configuration but the radiation becomes more intense over the head.
The over view shows the results we would have obtained by making measurements in differents plans. In those plans the radiation seems to be omnidirectionnal, although the global radiation of the source is not. This shows the limitation of the meaurements in two plans only.
Around 1250 Hz, the radiation becomes much more complicated. We have a loss of radiation behind the talker, in an opened angle of 120° on the left and the right.
There is an important lobe down, so a reflecting ceiling will have in important part for those frequencies. At very high frequencies, the radiation is very complicated too. There is still this 120° opening, and an important lobe around the talker mouth, and down too. There is very few rear radiation.

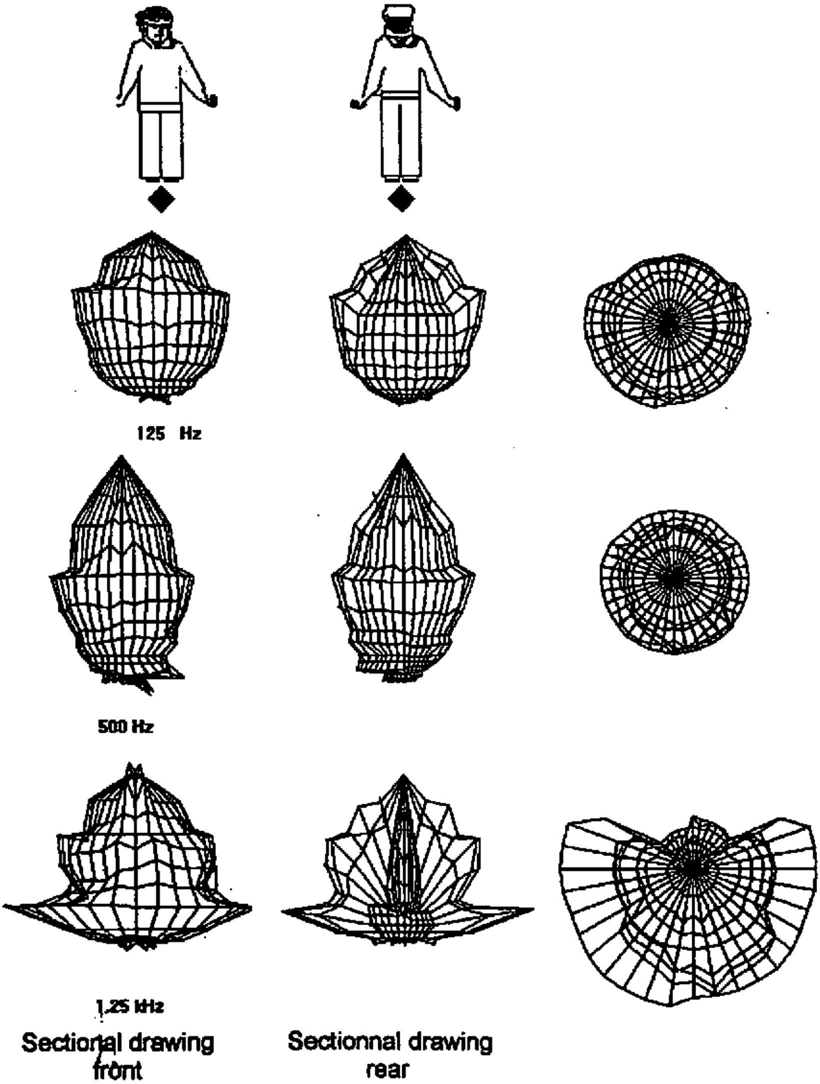

Fig.1 Spoken voice front view,rear view, top view at octave bands centered on 125, 500,1250 Hz

4.SIMULATION

The main interest of a simulation software is to be able to apprehend the effect of a modification on a project without having to materialize it.

An accurate simulation requires a good modelisation of the different parameters involved in the studied phenomenon. The knowing and modelisation of the sound sources is a main step.

The interest of our model, lies in the fact that the sound source is the instrumentist *and* his instrument, so we don't have to readapt the real position of the instrument, when being played by a musician, which is the case for the measurements in two plans.
In most softwares, you can enter the caracteritics of the source by octave bands, between 125 Hz and 4kHz, but the entry of the source is most of the time conceived for electroacoustic sound sources, and you have to give the directivity to -3 or -6 dB.

Within other softwares, it is possible to enter directivity data on a closed region (sphere or parallelepipedic surface, centered around the source). We have to transform the data of our database to adapt them to each new environnement, this means transform our data to make them fit to the kind of data accepted by the software, then we can test it on a room, and compare the results obtained for different sound sources..

Some softwares allow the emission of several sound sources at the same time, with a time delay . This allows us to share our sphere into a bigger number of elementary sources with the right amplitude and phase to reconstruct the global source.

To use this possibility, we did an antenna simulation, i.e. find the right strength and phase, with which each elementary source has to emit [COU97]. Those results can be used in software simulation, and the precision of the sound arrival in binaural simulation. Unfortunately, usually the time delay is not ajustable in fonction of the frequency.
These steps are described to show the importance of the uniformisation of the sound sources data.
We did a simulation, in a room, only by changing the source within the same group of instruments (strings), and comparing the strings to a trumpet [COU97].
As for the measurements, C80 is very sensitive to the change of sources, even for sources with a very close directivity (strings).
The other criteria change but in a smaller proportion.

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