DETECTION AND CLASSIFICATION OF THE AUDIOPHONIC SONAR SIGNAL: PERSPECTIVES OF SPACE SIMULATION UNDER HEADPHONES

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ABSTRACT

An important aspect of underwater detection is auditory surveillance, especially in the context of passive detection and classification.
To know the environment of the submarine, the operator listens to the sonar signal under headphones. It has been proved that it is useful to facilitate his perception of the sound by simulating spaciousness of the auditory images, thereby making the cues clearer and more distinct.
After a brief review of auditory localization, this paper describes the most recent techniques of space simulation under headphones. The technique based on the convolution in real time of the original monophonic signal with the tranfer functions of the ears is emphasized. Examples of applications to underwater detection are suggested.

1- INTRODUCTION

In the context of passive detection and classification, the acoustic signal received by submarine antennas is used to provide information about the environment. For the moment, signal processing techniques can tell us if there are any underwater noise sources in the environment, but nothing can replace the human operator listening to the acoustic signal: in fact the sonar operator can make use of many cues to extract from the signal technical information about noise sources and even sometimes about their identity.
The sonar operator is trained to listen to the acoustical signal through headphones. Let us recall that the acoustic signal from the sonar is monophonic, which means that the same signal goes to both ears of the operator. In the real listening situation, the operator works, sometimes for several hours in a row, in a noisy environment, scanning signals from different antennas. When a boat is detected, the operator uses the direct signal from a beam steering of a highly directive antenna, which is recorded for about 20 minutes (that means that he can play back any part of the signal to examine all the cues). He frequently needs to scan adjacent directions, in order to make auditory comparisons with the background noise.
The classical way of helping the sonar operators in detection and classification is twofold. On the one hand, they are provided with filters, to isolate the relevant frequency ranges. On the other hand, they can also change the recording speed, with or without changing the pitch, to get a better display of the temporal cues (2).
But it is also very useful to improve their global perception of the sound by simulating spaciousness in the monophonic signal, thereby making the cues clearer and more distinctive, and improving attention. This paper presents some recent developments in resarch on binaural hearing and space simulation under headphones.

2- BASIC CUES FOR SOUND LOCALIZATION

To identify the position of a sound source, the listener uses the auditory cues that are present in the acoustic field at both ears (along with visual and cognitive cues). The most important auditory cues are the level and time differences at the two ears and the spectral modifications of the sound caused by the head shadow, the reflexions from the shoulders, and the pinna.

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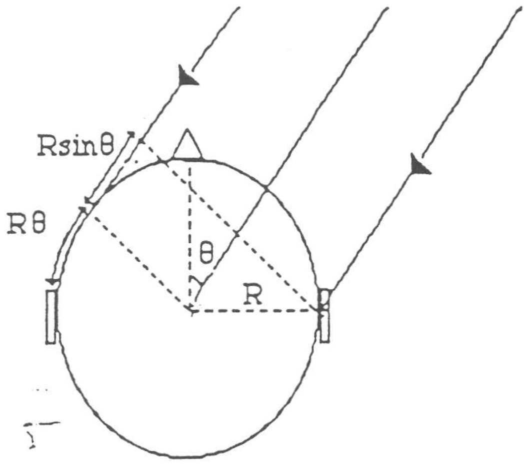

Figure 1: Difference in arrival time at the two ears for a distant sound source at an angle θ rd to the observer.
The path difference between the two ears is:
$\Delta d = R\theta + R \sin\theta$
and the time difference is:
$\Delta t = \Delta d / c$

Localization cues are still the subject of many studies: in the free field they were measured both on real subjects and dummy heads, with small microphones placed close to the eardrum. Also, modeling of the human head gives theoretical results for the level and time differences of the sound at the two ears. Extensive reviews of data and models related to binaural hearing and sound localization have been published by Blauert(1), Canévet(4), Kuhn(9), Mills(12), Moore(13) and others.

2-1 Level differences

The many results of the measurements of level differences between the sound reaching the two ears are highly variable. For example at a frequency of 4 kHz, level differences can vary from 0 dB, for the azimuths of 0° and 180°, to approximately 15 to 25 dB, for the azimuths of 90° and 270°. This variability comes from inter-individual differences.
Depending on the spectral complexity of the signal, to a first approximation this cue is used by the human auditory system for frequencies above 1500 Hz.

2-2 Time differences

In the horizontal plane, the time differences seem to be more stable and closer to the results given by the different models of the human head. The time differences between the two ears can vary from 0, for the azimuths of 0° and 180°, to approximately 0.7 ms, for the azimuths of 90° and 270°. This cue is used by the auditory system over the whole frequency range, from 20 Hz to 20 kHz. However, it can be interpreted as a phase difference (for periodic signals) only for frequencies below 1500 Hz.

2-3 Spectral modifications by the head shadow, the reflexions from the shoulders and by the pinna.

It is possible to measure the acoustic field at the two ears with small microphones placed close to the eardrum. From these measurements the transfer functions of the ears can be computed for any given position of the sound source. Usually this kind of measurement is made in an anechoic room (free-field conditions): the subject is precisely positioned relative to the sound source (a small loudspeaker) emitting a broad-band signal. The FFT of the signals from the small microphones give the transfer functions of the outer ears for the position of the source. It is clear that this kind of measurement takes into account time and level differences and spectral modifications caused by the head shadow, the reflexions from the shoulders, and by the pinna.

The main measurements of the outer ear transfer functions were made by Mehrgardt et al.(10), Morimoto et al.(11), Pösselt et al(14), Schmitz et al.(16) and Wightman et al.(19).

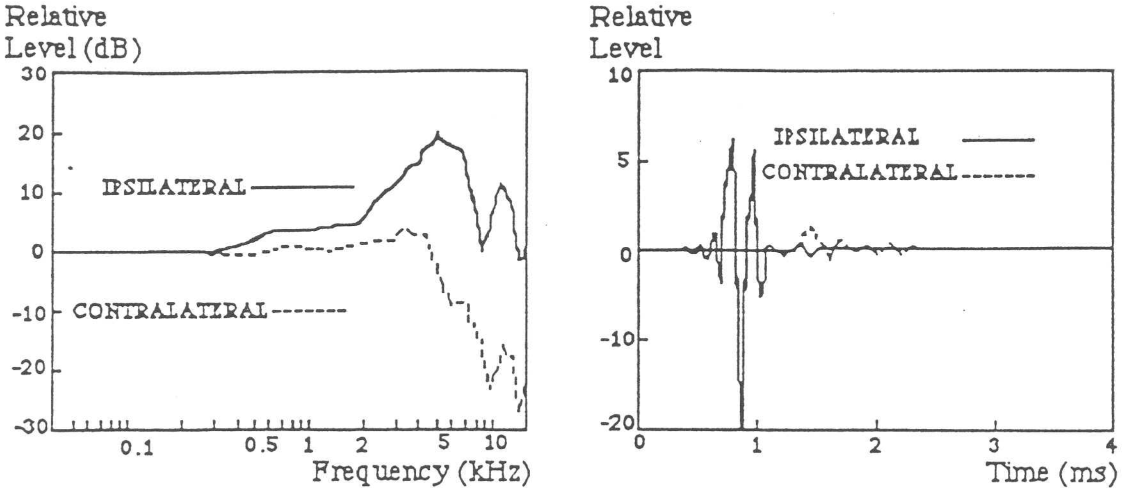

Figure 2: Modulus of the transfer functions and impulse responses of the outer ears for an azimuth of 90° in the horizontal plane measured by Ch. Pösselt.

The transfer functions of the ears are rather complex with large spectral alterations (up to 20 dB at some frequencies) mainly due to the effect of the pinna. The difference between left and right transfer functions of the ears results from the head shadow.
The impulse responses of the outer ears are not symetric, whitch means that the phase is not linear.

3- SPACE SIMULATION UNDER HEADPHONES

3-1 The effect of a pure delay

To begin with, it should be noted that it is possible to give spaciousness to a monophonic signal by introducing an artificial delay between the two ears. Here we do not refer to a natural delay of about 0.7ms or less, which only leads to a lateralisation of the auditory image within the head, but to a much longer delay, of the order of 20 to 40 ms, analogous to the late reflections in architectural acoustics (7), (15).

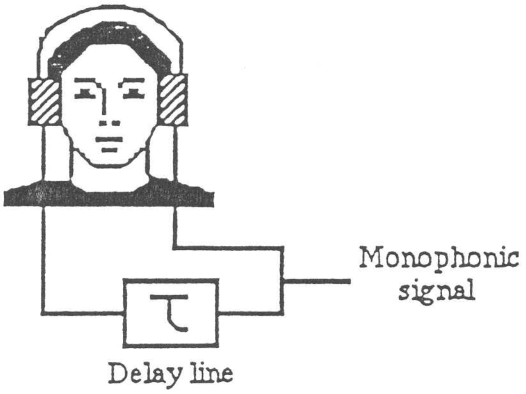

Figure 3

This type of presentation gives depth and volume to the sound. As a result, the sonar operator has the sensation of a more comfortable listening situation. Besides, he can "scan" the auditory space and recover more readily the information that is available.

3-2 The convolution with the transfer functions of the outer ears

When you listen to a signal through headphones, the sound seems to be inside the head: the action of the head, shoulders and pinna is cancelled by the headphones. A simulation of the real listening conditions under headphones is achieved by means of parallel filtering.

The simulation follows several steps. The first step is the measurement of the transfer functions of the outer ears in the free field at various spatial positions. The second step involves the measurement of headphone transfer function, with the listener wearing the headphones. Then, the parallel filters used for the simulation need to be computed: their transfer functions are given by the outer ear transfer functions divided by the headphone transfer function.

Notice that to measure the transfer functions of the outer ears in the free field, it is necessary to take into account the transfer functions of the loudspeaker, used to send a broad band signal, and that of the microphones, placed close to the listener's eardrum.

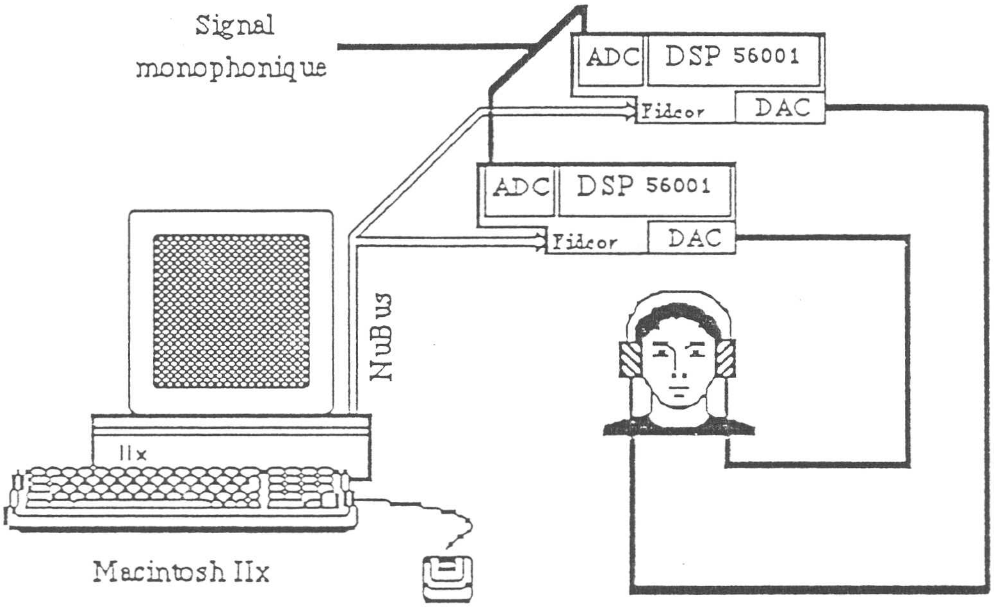

Figure 4 : Parallel filtering using two DSP boards and a Machintosh II

This parallel filtering is achieved in real time by two DSP boards and a Machintosh II (Fig. 4) using a FIR filter technique (two filters with 256 points at the sampling frequency of 44.1 kHz) with the possibility of changing in real time the direction simulated, which means changing in real time the coefficients of the filters.

Listening to this "stereophonic" signal is quite amazing : the sound seems to come from outside the head and from the direction where the transfer functions were measured. This treatment works particularly well with underwater sounds, which are, in general, rapidly fluctuating and wide-band signals with a good part of their energy at low frequencies; the global perception of the sound is improved, making the cues clearer and more distinctive.

4- SONAR APPLICATIONS OF SPACE SIMULATION UNDER HEADPHONES

As noted above, in the real listening situation the sonar operator works in a noisy environment often for several hours at a time. It is therefore especially important to provide him with a more spacious sound from which he recovers more readily the available information and from which the auditory sensation is more pleasant.

4-1 Panoramic view of the situation

When an underwater noise source is detected, the sonar operator listens to the signal received from a beam steering of a highly directive antenna pointed in one direction. He frequently needs to listen to the signal from the adjacent directions too, in order to follow the movements of the boat or to make auditory comparisons with the background noise. This is usually done by switching sequentially from one direction to the other. It would be a great improvement to give the sonar operator the possibility of having all the information simultaneously instead of sequentially. For example, he could listen at the same time to three signals, one from the direction of the target and the other two from adjacent directions. The sonar operator will then be able to scan those signals displayed as a panoramic view of the situation (2).

4-2 Simultaneous listening of different antennas

In the situation of passive watching, the sonar operator must check continuously the signals from different antennas. The principle presented above can also be applied to that case: signal from different antennas can be merged into one single holophonic auditory image in order to give the operator the possibility to scan at the same time all those signals. The auditory system can separate them out as it does in the situation of a cocktail party, and focus the attention on one particular event while ignoring the others. The main advantage of this method is to have access to the information simultaneously rather than sequentially (2).

It is obvious that the simulation of subjective auditory space under headphones will find applications in many other fields. Examples of applications are: air traffic control displays for the tower or cockpit (5), (8), acoustic navigation displays for the blind (3), monitoring of vehicules or objects during extra-vehicular activity in space (6), (17), (18), advanced teleconferencing environments, radio and Hi-Fi technology (14) and so on.

5 CONCLUSIONS

The real time implementation of space simulation under headphones, based on the convolution of the original monophonic signal with the transfer functions of the ears is achieved with the help of two DSP boards (FIR filters with 256 points at the frequency sampling of 44.1 kHz).This technique is already validated on underwater sounds and it is possible to merge different signals into one single holophonic auditory image with more DSP boards.
A number of improvements are under study, to include the simulation of distance and reverberation, in order to link the effect given by a pure delay and the effect given by the transfer functions of the ears .

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