SWEET SPOT WIDENING FOR STEREOPHONIC SOUND REPRODUCTION

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ABSTRACT

In this paper the correction of the degradation of the stereophonic illusion due to *off-centre listening* is investigated. The main idea here is that the directivity pattern of a loudspeaker array should have a well defined shape such that a good stereo sound reproduction is achieved in a *large listening area*. Optimal digital filters are designed and applied to individual drivers of linear loudspeaker arrays in order to obtain a directivity pattern of a specific shape. This shape is adapted to the time/intensity trading mechanism of the human auditory system via psychoacoustic experiments within a wide listening area. This application is also referred to as *"Position Independent (PI) Stereo"*.

1. INTRODUCTION

The basis of stereophony is the ability to create *"phantom"* images. It is known that the brain locates a mono signal originated from a single source by comparing the differences in the arrival time and intensity of that signal at each ear. If the same mono signal is played through two loudspeakers on either side of the listener, then the sound seems to appear from midway between the two loudspeakers since the travelling time of the signal arriving at each ear is the same. This is called a *"phantom"* image [1].

Generally it is considered as a serious artefact of the traditional stereo system that the listener is aware of the stereophonic illusion only in a limited region. Optimum stereo perception only occurs if the listener is placed exactly in the median plane between the two loudspeakers. If the head is moved away from the median plane then the stereo effect deteriorates. This *off-centre listening* problem becomes even more serious when the distance between the loudspeakers is not large in comparison with the deviations from the centre position as it occurs in multimedia PC monitor and TV applications; the latter normally has a wider stereo base but in some cases a smaller stereo base is desired. Thus, if the head is moved laterally, the sound rapidly seems to come from the nearest loudspeaker only. This is mainly because of two additional effects: the intensity of the nearest loudspeaker at the listener's head is highest, and its wavefront arrives earlier (law of the first wavefront or *precedence effect*) [1] (see Figure 1).

In general, it can be stated that correct localisation within a wide listening area is beneficial for all applications where a good stereophonic sound image is required: audio, video and car stereo. The idea of achieving an increase in the listening area (*sweet spot widening*) in a stereo setup has been introduced and studied at the Philips Research Labs and it has been called *"Position Independent (PI) Stereo"* [2, 3]. The main idea is the following: if the listener moves to the left, the sound intensity from the right loudspeaker must increase, while that of the left loudspeaker must decrease in such a way that the *phantom* or *virtual image* remains in the middle. This can also be seen as a sort of *automatic balance control* depending on the position of the listener. In this paper, we will describe a robust digital implementation for the PI-stereo system so as to achieve a better performance of the system in a large listening area. Thus, we will describe a digital filtering technique which will be applied to the individual drivers of loudspeaker arrays to obtain a directivity pattern of a specific shape.

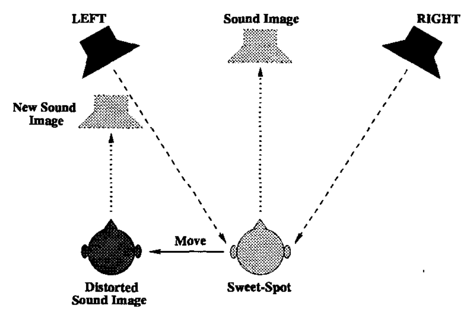

Figure 1: *Stereo setup and off-centre listening problem when the listener is not in the sweet spot position.*

2. BASIS OF POSITION INDEPENDENT STEREO

The PI-stereo system is basically composed of two loudspeaker arrays, each fitted into a single cabinet, with an optimal directivity pattern which has been designed such that a good stereo sound reproduction is achieved in a large listening area [2, 3]. A "standard" listening setup for PI-stereo for Hi-Fi audio and TV setups is shown in Figure 2.

2.1. Loudspeaker array design and frequency range splitting

It has been proved that an adjustable directivity pattern for a loudspeaker can be realised by using an array of drivers positioned at a small distance from each other [4, 5, 6]. In our case, a special design to achieve PI-stereo sound reproduction is a pair of loudspeaker cabinets equipped with a pair of drivers, which are separated at a given distance to achieve two frequency ranges (high and mid), so as to obtain a desired directivity pattern. Low frequencies, which are not important for sound localisation, can be optionally reproduced by means of a subwoofer (Figure 3).

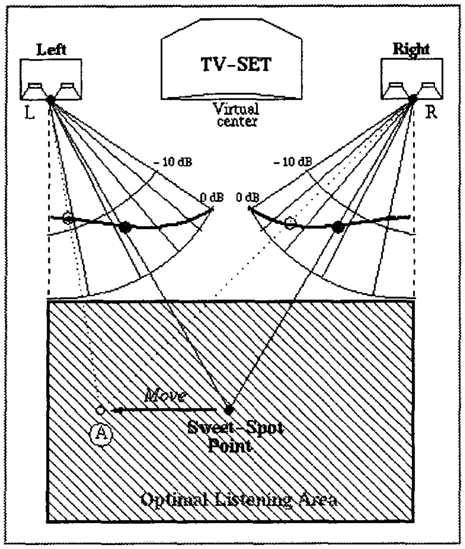

Figure 2: *Optimal listening area for PI-stereo reproduction.*

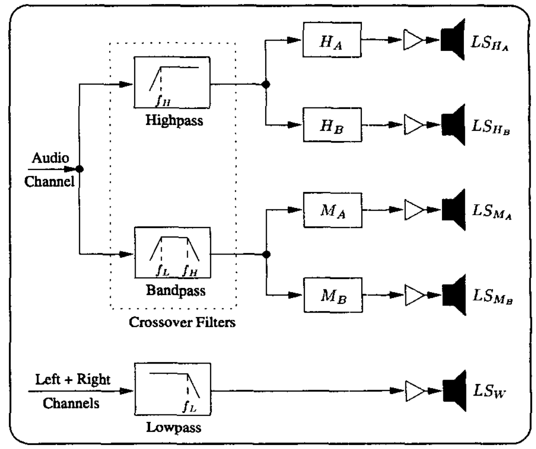

Figure 3: *A schematic block diagram for the processing of the audio channels (left and right). The frequency range splitting is achieved by means of crossover filters. Filters H_A and H_B for the high range and M_A and M_B for the mid range drive the two arrays of loudspeakers, LS_{H_A}, LS_{H_B} and LS_{M_A}, LS_{M_A}. Low frequencies of the audio channels (left and right) are reproduced by a subwoofer LS_W.*

3. OPTIMAL DIRECTIVITY PATTERN

In order to calculate the optimal directivity pattern (*target function*) for the loudspeakers, listening tests in an anechoic room were conducted [2, 7]. During these tests, the differences in intensity levels between right R and left L loudspeakers for different listening positions to obtain a central sound image were measured (Figure 4). From these experiments, which are obtained by using broadband signals, an optimal directivity pattern for the loudspeakers can be determined. Using narrowband signals, frequency dependent time/intensity trading data could be obtained, however broadband signals lead to satisfying results.

3.1. Calculation of the target function

We define here an *ad hoc* mathematical expression F^t for the sound pressure levels for the directivity patterns of the loudspeakers. This expression, which depends on two linear parameters (A and B) and one nonlinear parameter (C), is defined as follows

$$F^t = 20 \log[|A + B \sin^C \theta|] \quad (1)$$

where θ is a set of angles. In order to fit time/intensity trading data to this target function, we can use the well known *nonlinear least squares* method which gives the values for A, B and C [8]. Making the following substitution $L^t = 10^{[F^t/20]}$, the system we have is then:

$$\begin{bmatrix} 1 & sin^C\theta_1 \\ 1 & sin^C\theta_2 \\ \vdots & \vdots \\ 1 & sin^C\theta_n \end{bmatrix} \cdot \begin{bmatrix} A \\ B \end{bmatrix} = \begin{bmatrix} L_1^t \\ L_2^t \\ \vdots \\ L_n^t \end{bmatrix} \quad (2)$$

Thus, in matrix form this reduces to:

$$\mathbf{T} \cdot \underline{\mathbf{x}} = \underline{\mathbf{L}}^t \quad (3)$$

where $\mathbf{T}$ is the $[N \times 2]$ matrix of the target function to optimise, $\underline{\mathbf{x}}$ is the $[2 \times 1]$ vector of the linear parameters to find and $\underline{\mathbf{L}}^t$ is the $[N \times 1]$ vector of the experimental time/intensity trading data to fit. This results in the best estimation for the system which gives the *minimum error* in the fit to $\mathbf{T}$ by

$$\min_{x \in \mathbf{R}^2} \|\mathbf{T} \cdot \underline{\mathbf{x}} - \underline{\mathbf{L}}^t\|_2^2 \quad (4)$$

We then obtain the following target function which approximates the correct directivity pattern for the loudspeakers for the given data:

$$F^t = 20 \log[|0.19 + 1.71 \sin^3 \theta|]. \quad (5)$$

Figure 5 shows the polar plot for the optimised directivity pattern for the calculated time/intensity trading data.

4. FILTERING TECHNIQUE FOR DRIVING THE LOUDSPEAKER ARRAYS

In the next section, we focus on the optimisation problem of estimating the required FIR filter coefficients which achieve the calculated optimal target function in a wide listening area.

4.1. Optimisation problem: Digital FIR filters

We consider here the general case of having a *linear array* of N equal and equidistant omnidirectional sound sources separated by a distance d. Using the acoustic pressure equation for a simple source and given that each sound source is driven by a FIR filter $h_{n,m}$ of N number of drivers with M coefficients, the *total sound*

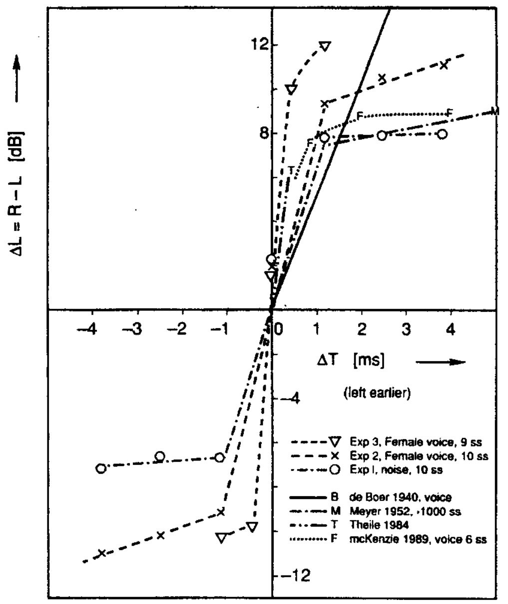

Figure 4: *Time/intensity trading results carried out in [7] in comparison with other experiments from the literature.*

pressure $P_{k,\ell}$ for a frequency ω_k and an angle θ_ℓ is given by:

$$P(\omega_k, \theta_\ell) = \sum_{n=1}^{N} \sum_{m=1}^{M} h_{n,m} e^{j\omega_k \left[\left(n - \frac{N+1}{2}\right) \frac{d}{c} \sin\theta_\ell - \frac{m-1}{f_s}\right]} \quad (6)$$

where f_s is the sampling frequency and c is the velocity of sound. Let us define for convenience $\alpha_{k,\ell,n,m}$ as:

$$\alpha_{k,\ell,n,m} = \omega_k \left(\left(n - \frac{N+1}{2}\right) \frac{d}{c} \sin\theta_\ell - \frac{m-1}{f_s}\right) \quad (7)$$

Therefore, the expression in (6) can also be written as:

$$P(\omega_k, \theta_\ell) = \sum_{n=1}^{N} \sum_{m=1}^{M} h_{n,m} \Big(\cos\alpha_{k,\ell,n,m} + j \sin\alpha_{k,\ell,n,m} \Big) \quad (8)$$

Since we want to obtain a sound pressure equivalent to that given by the optimal target function in Equation (5), we can define the *required sound pressure* $S_{k,\ell}$ for the target function as:

$$\begin{aligned} S_{k,\ell} &= F_\ell^t e^{\left[-j\omega_k \frac{M-1}{2f_s}\right]} \\ &= F_\ell^t \left(\cos\left(\omega_k \frac{M-1}{2f_s}\right) - j \sin\left(\omega_k \frac{M-1}{2f_s}\right)\right) \quad (9) \end{aligned}$$

where F_ℓ^t are the sound pressures at a angles ℓ given by the target function previously described, and the phase term corresponds to a *constant group delay* of $T(M-1)/2$. We can now formulate the *least squares optimisation* problem, so as to find the $h_{n,m}$ FIR coefficients for the different array drivers, as follows:

$$\mathbf{T} \cdot \underline{\mathbf{H}} = \underline{\mathbf{S}} \quad (10)$$

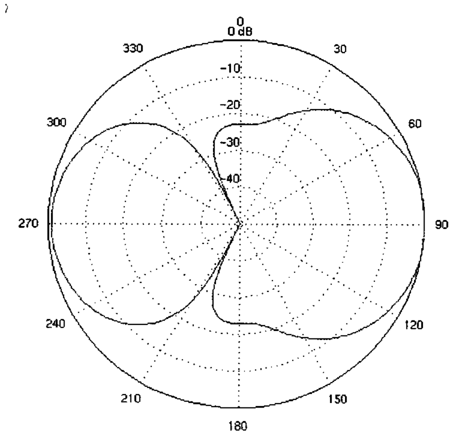

Figure 5: *Normalised polar plot of the target function in Equation (5).*

where the vector $\underline{\mathbf{H}}$ of dimensions $[NM \times 1]$ contains the filter coefficients and the matrix $\mathbf{T}$ of dimensions $[2KL \times NM]$ is defined by Equation (8) as:

$$T_{(k-1)L+\ell,(n-1)M+m} = \cos\alpha_{k,\ell,n,m} \quad (11)$$

$$T_{(K+k-1)L+\ell,(n-1)M+m} = \sin\alpha_{k,\ell,n,m} \quad (12)$$

and the vector $\underline{\mathbf{S}}$ of dimensions $[2KL \times 1]$ is defined by Equation (9) as:

$$S_{(k-1)L+\ell,1} = F_\ell^t \cos\left(\omega_k \frac{M-1}{2f_s}\right) \quad (13)$$

$$S_{(K+k-1)L+\ell,1} = -F_\ell^t \sin\left(\omega_k \frac{M-1}{2f_s}\right) \quad (14)$$

This proposed method results in the calculation of the FIR filter coefficients $h_{n,m}$ ($n = 2$ and $m = 20$) for the two sound sources or drivers (A and B). We performed some simulations and it appeared that they converged to the optimum choice of filter coefficients for each of the two frequency bands (mid and high).

5. MEASURED DIRECTIVITY PATTERNS AND LISTENING EXPERIMENTS

For the study of the directivity polar plots, the right loudspeaker response was considered, that is the main lobe is on the right-hand side and the side lobe is on the left-hand side. Another consideration here is that when listening to the PI-stereo system the loudspeaker boxes are face on, not at 30° pointing inwards to the listener as in normal stereo. To compensate for this, the frequency responses of the loudspeaker boxes were taken at 30° clockwise for the left and 30° anti-clockwise for the right box so that the frequency response in the middle of the working region was considered as opposed to the response at the edge.

Figure 6(a) shows simulations of the theoretical polar plots for the PI-stereo system for frequencies ranging between 200-1250 Hz. Figure 6(b) shows the measured polar plots in an anechoic room using the right loudspeaker box. Note here that all polar plots have been normalised at 30° which is the center of the considered working region. Theoretical and measured directivity polar plots

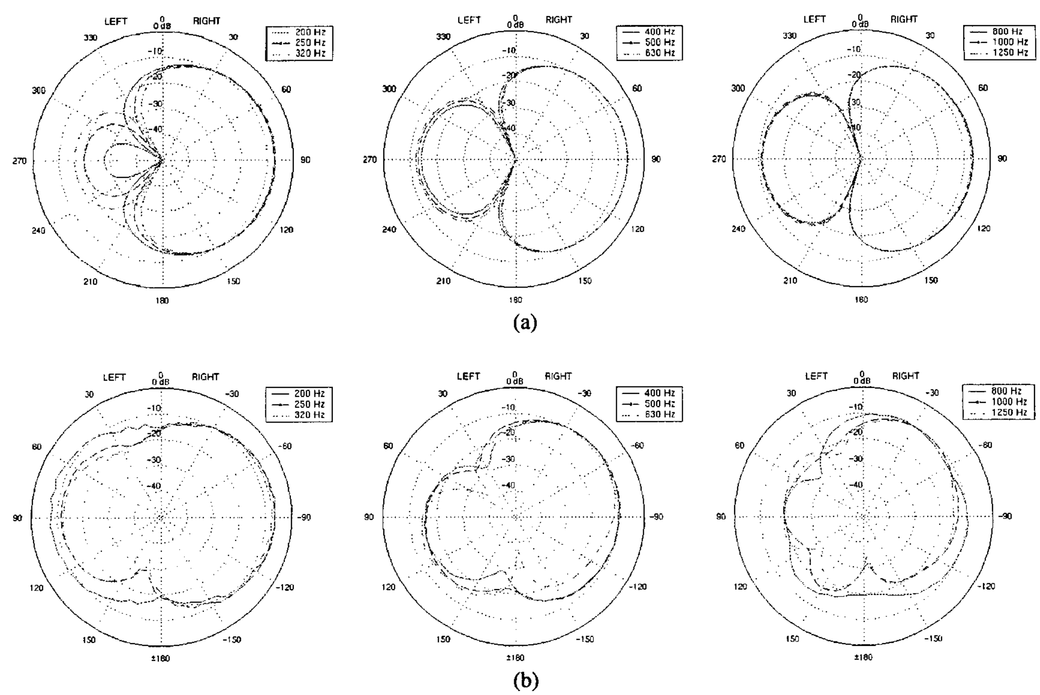

Figure 6: *Theoretical (a) and measured (b) polar plots for the PI-stereo system for frequencies ranging from 200-1250 Hz.*

for higher frequencies appeared to be very close to the ones in this figure and due to space limitations are not reproduced in this paper.

Preliminary listening experiments have shown that the PI-stereo system worked as predicted by the theoretical models. Correct central sound localisation for voices and other effects have also been demonstrated for a number of listening positions. Both normal stereo and PI-stereo were reproduced by the same loudspeaker cabinets, so there was no shift in the stereo image at the central position. The position independent stereo image for lateral positions was observed to be of better quality compared to that for normal stereo. The difference between normal stereo and PI-stereo for central listening was almost unperceivable. The stereo sound sensation, in particular the placement of central voices, was independent of the listening position within a large area. To study the influence of room boundaries, these listening tests were performed in several rooms. However, despite room reflections, the PI-stereo effect remains due to the law of the first wavefront [1].

6. CONCLUSIONS

We have described a new stereo sound reproduction system that offers a natural high quality stereophonic sound image in a large listening area. A digital filtering technique has been applied to individual drivers of linear loudspeaker arrays in order to obtain a directivity pattern having an optimal shape. This optimal shape was adapted to time/intensity trading experiments for enlarging the sweet spot area. The outcome of this work showed that optimal directivity patterns for loudspeaker arrays in stereophonic applications can be very useful for sweet spot widening.

PI-stereo can be applied to any systems where a good stereo sound reproduction in a large listening area is required, such as: TV-sets, Hi-Fi's, multimedia, home theatre, car stereo and portable audio.

7. REFERENCES

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