REALIZATION OF A DIFFUSE SOUND FIELD WITH A PC-BASED SOUND CARD SOLUTION

Oomke Weikert

Department of Signal Processing and Communications
University of the German Federal Armed Forces Hamburg
oweikert@unibw-hamburg.de

ABSTRACT

For the quality assessment of headphones, especially the loudness measuring of headphones, a diffuse sound field is required. At this time a hardware based noise generator, one-third octave filters built up in analog mode as well as boosters are used. In this work a flexible PC-based solution with the aid of a sound card is presented. Therefore ten independent noise generators, generating Gaussian distributed white noise, are needed. The implementation using the 'Dynamic Creation of Pseudorandom Number Genrators' for 'Mersenne Twister' is described. A probability transformation to convert equal distributed numbers into Gaussian distributed ones is derived in detail. Furthermore one-third octave filters are designed and implemented according to the ANSI standard. The access to the sound card is provided using the Wave-API library under Microsoft Windows.

This work was carried out at Sennheiser electronic GmbH in Wennebostel (Germany) in the development department for cord based headphones.

1. INTRODUCTION

An ideal diffuse sound field is determined by the definition that from all space directions the sound incidence at the measuring place is equally strong and equal distributed. The average energy density is constant in a diffuse sound field. However, the average value in time of the sound intensity vector is zero.

A diffuse sound field can be realized in a room with totally reflecting walls [1]. One loudspeaker operates as a source and introduces white noise into the room. The distance from the source at which the level of the primary sound and that of the diffuse field are equal is called the "diffuse field distance". The equal statistical distribution of the direction of the sound incidence at the measuring place is provided by the reflections of the sound at the walls of the room.

A diffuse sound field can also be realized by a large number of loudspeakers in a room with non-reflecting walls [2]. Examinations in [2] with 8 loudspeakers, each of it in a corner of the room, have shown that in a spatially limited area around the center of the room the same properties as in a diffuse sound field apply. With regard to theoretical calculations in [2] the average energy density will be constant around the center of the room, if a large number of loudspeakers is used. If the signals for the loudspeakers have phase angles that are equal distributed in a sufficient time period, the average value in time of the sound intensity vector will be zero in the center of the room. The properties of a diffuse sound field are therefore given in an area around the center of the room. The signals for the loudspeakers should be independent noise sources. Otherwise the phase angles of the sources will be correlated and an interference field will appear.

A realization with 20 loudspeakers using only one noise source was made by [3]. The same noise is delivered with different delays to every loudspeaker.

Another realization of a diffuse sound field is described in the british norm BS 5108 [4] using only 4 loudspeakers. A practical realization was made by [5].

A room with non-reflecting walls as it is used for the realization of a diffuse sound field is shown in Figure 1.

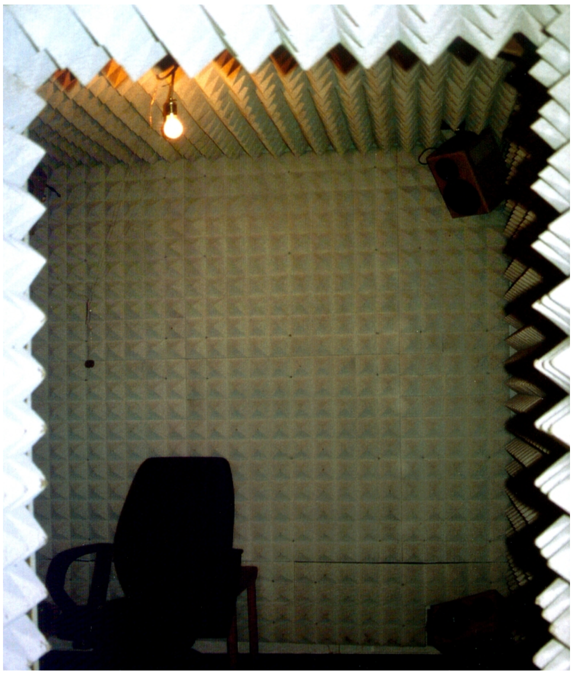

Figure 1: *Room with non-reflecting walls as it is used for the realization of a diffuse sound field.*

A diffuse sound field is used for the loudness test of stereo-headphones. A test person is located in a diffuse sound field and listens to white noise in a defined frequency range. The property of a defined frequency range of the white noise is realized with one-third octave filters. The test person is advised to adjust the volume of the stereo-headphone with the aim that the loudness of the headphone is equal to the loudness of the loudspeakers, i.e. the loudness of the diffuse sound field. This process is repeated for different frequency ranges. Because this is a subjective test, the results of several test persons are averaged. The resulting curve over the frequency is then significant for a certain headphone type.

The sound-absorbing characteristic of a headphone can also be measured with the help of a diffuse sound field. Therefore a dummy head with microphones in its synthetic ears is used. The dummy head is located in a diffuse sound field. First the sound level in the ears is measured without the headphone. Then the measurement is repeated with the headphone onto the dummy head. This measurement is performed for different frequency ranges. Both measured sound levels can be set into relation to each other. The result is a curve that shows the sound-absorbing characteristic of the headphone over the frequency.

Another application of diffuse sound fields is the measurement and calibration of microphones.

2. SYSTEM CONCEPT

The realization of a diffuse sound field is based on eight loudspeakers. Each of the eight loudspeakers and a stereo-headphone have to be supplied with independent noise signals. Therefore ten analog outputs are needed on the sound card. For a stereo microphone two analog inputs are necessary.

Ten independent noise generators are generating equal distributed white noise. A probability transformation transforms the equal distributed signals to Gaussian distributed ones. The power density spectra of the signals are not changed. With the one-third octave filters a frequency range can be chosen.

If the loudspeakers have non-flat frequency responses, a filtering with the inverse speaker impulse response should be considered. A flow chart that applies to all ten channels (eight loudspeaker channels, two headphone channels), is presented in Figure 2.

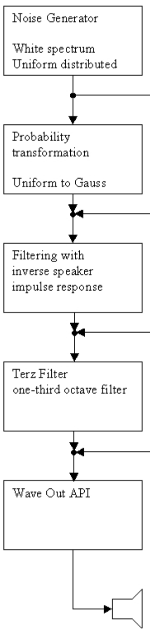

Figure 2: *Flow chart of one instance.*

3. TEN INDEPENDENT NOISE GENERATORS

For eight loudspeakers and two headphone channels ten parallel random number generators are required.

There are two essential requirements for these parallel random number generators.

- The ten random sequences produced by the random number generators must be independent from each other, so that the noise delivered to the loudspeakers is uncorrelated.
- The period length of every random number generator should be long enough that a repetition is not audible.

Matsumoto and Nishimura presented a new algorithm named "Mersenne Twister" (MT) for the generation of equal distributed pseudorandom numbers [6]. "Mersenne Twister" has a period length of $2^{19937} - 1$. The sequence is 623-distributed, i.e. 623-dimensionally equidistributed, to 32-bit accuracy. This is close to the trivial upper bound, because $2^{623*32} - 1 = 2^{19936} - 1 \leq 2^{19937} - 1$ applies. The k-distribution as an reasonable measure of randomness is described in [6].

Matsumoto and Nishimura also introduced an algorithm for the "Dynamic Creation of Pseudorandom Number Generators" [7]. Dynamic Creator is a program which

1. receives users' specification, i.e., word size, period, etc.
2. receives identity (ID) (process ID, machine ID, etc)
3. creates a (set of parameters for) MT such that
   (a) satisfying users' specification,
   (b) ID is encoded in a parameter of MT so that different ID assures essentially different pseudo random number generators (PRNG) (i.e., the characteristic polynomial of the recurring sequence is irreducible and distinct to each other).

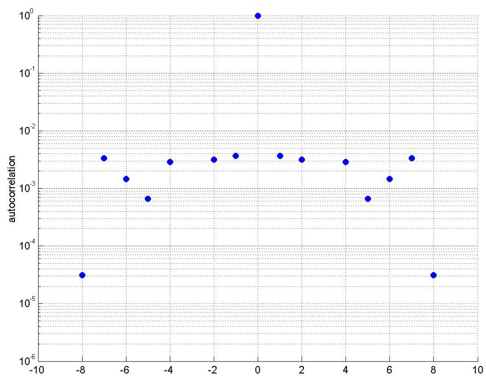

Figure 3: *Autocorrelation of one PRNG, calculated using 100000 random numbers.*

In this project a period length of 2^{512} is chosen for each of the ten parallel random number generators. To avoid on every restart of the program the time-consuming calculation of suitable parameters for the algorithm, i.e. of parameters that guarantee the independence of the ten parallel generators, the once calculated parameters are stored in a separate file. This file is loaded on every restart of the program and the stored parameters are applied to the algorithm. The generators are initialized on every restart of the program with the current system time. The autocorrelation function of one PRNG, calculated using 100000 random numbers, is shown in Figure 3.

With a sampling frequency of 44.1 kHz a period length of 2^{40} for each PRNG corresponds to a time period of a year. Due to these considerations a period length of 2^{512} for each of the ten parallel random number generators is more than sufficient.

4. PROBABILITY TRANSFORMATION

The purpose of the probability transformation is the generation of a Gaussian distributed variable Y with the probability density function

$$f_y(y) = \frac{1}{F_y(1) - F_y(-1)} \frac{1}{\sqrt{2\pi}\sigma} e^{-\frac{y^2}{(2\sigma)^2}} \text{ with } y \in [-1, +1] \tag{1}$$

from an equal distributed variable X with the probability density function

$$f_x(x) = 1 \text{ for } x \in [0, +1]. \tag{2}$$

The probability function $F_y(y)$ is calculated by

$$F_y(y) = \int_{-\infty}^{y} f_y(l)dl. \tag{3}$$

For the equal distributed variable X the probability function $F_x(x)$ follows to

$$F_x(x) = x \text{ for } x \in [0, 1]. \tag{4}$$

The probability transformation is based on the fact that the probability that y lies in the interval dy is equal to the probability that x lies in the interval dx:

$$f_y(y) \mid dy \mid \equiv f_x(x) \mid dx \mid . \tag{5}$$

For the calculation of the transformation function we use the following equation

$$F_y(y) = F_x(x) \tag{6}$$

that is evaluated to

$$y = F_y^{-1}(F_x(x)) = g(x). \tag{7}$$

This assumes that $F_y(y)$ could be represented in closed form. This is not the case for a Gaussian distribution.

A Gaussian distributed variable Y can be calculated using a two-dimensional Gaussian distribution represented in polar coordinates. For a two-dimensional Gaussian distribution $f_{y_1,y_2}(y_1, y_2)$ in cartesian coordinates y_1 and y_2 a representation in polar coordinates r_∞ and φ can be calculated. It follows that the radius r_∞ is Rayleigh distributed in the interval $[0, +\infty]$ and the phase φ is equal distributed in the interval $[0, +2\pi]$.

$$f_{r_\infty}(r_\infty) = \frac{r_\infty}{\sigma^2} e^{-\frac{r^2}{2\sigma^2}} \text{ with } r_\infty \in [0, +\infty] \tag{8}$$

$$f_\varphi(\varphi) = \frac{1}{2\pi} \text{ with } \varphi \in [0, +2\pi] \tag{9}$$

To generate a Gaussian distributed variable Y a Rayleigh distributed radius r and an equal distributed phase φ are calculated. Thereafter the polar coordinates r and φ are converted into the cartesian coordinates y_1 and y_2. Both cartesian coordinates are then Gaussian distributed. Two Transformation functions to generate a Rayleigh distributed radius r and an equal distributed phase φ from an equal distributed variable X are derived in the following.

To get a Gaussian distributed variable Y which is limited to the interval $[-1, +1]$, it is necessary that the Rayleigh distributed radius r_∞ is limited to the interval $[0, +1]$. Therefore the Rayleigh distributed radius r_1 in the interval $[0, +1]$ is normalized to one.

$$f_{r_1}(r_1) = \beta f_{r_\infty}(r_\infty) \text{ with } \beta \text{ variable} \tag{10}$$

$$\int_0^{+1} f_{r_1}(r_1)dr_1 \equiv 1 \tag{11}$$

The Rayleigh distributed radius r_1 follows to

$$f_{r_1}(r_1) = \frac{r_1}{\sigma^2} \frac{e^{-(\frac{r^2}{2\sigma^2})}}{1 - e^{-(\frac{1}{2\sigma^2})}} \text{ with } r_1 \in [0, +1]. \tag{12}$$

$$F_{r_1}(r_1) = \frac{1 - e^{-(\frac{r_1^2}{2\sigma^2})}}{1 - e^{-(\frac{1}{2\sigma^2})}} \text{ with } r_1 \in [0, +1] \tag{13}$$

Using equation (6) with the equations (4) and (13) a Rayleigh distributed variable r_1 is calculated by the transformation function

$$r_1 = \sqrt{-2\sigma^2 \log(1 - x(1 - e^{(\frac{-1}{2\sigma^2})}))} \text{ with } r_1 \in [0, +1], \tag{14}$$

and an equal distributed phase φ is evaluated by

$$\varphi = 2\pi x \text{ with } \varphi \in [0, +2\pi]. \tag{15}$$

A Gaussian distributed variable in the interval $[-1, +1]$ is then calculated by

$$y_1 = r_1 \cos(\varphi) \text{ with } y_1 \in [-1, +1] \tag{16}$$

or

$$y_2 = r_1 \sin(\varphi) \text{ with } y_2 \in [-1, +1]. \tag{17}$$

5. ONE-THIRD OCTAVE FILTERS

One-third octave filters are designed according to the ANSI S1.1-1986 standard [8]. The filters are sketched with the Butterworth algorithm as an 2Nth-order IIR filter. The default value for N is 3. The frequency response of a 6th-order one-third octave filter is shown in Figure 4.

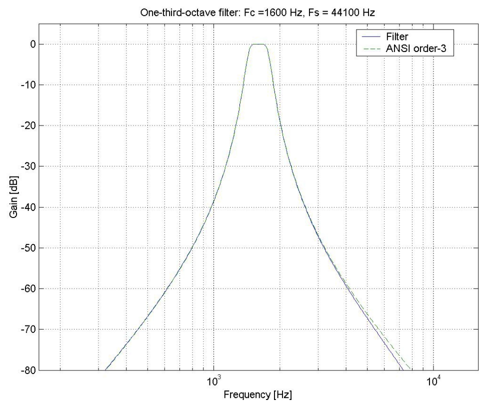

Figure 4: *6th-order one-third octave filter designed with Butterworth.*

6. THE SOUND CARD

Due to the requirements, i.e. ten analog outputs, the Terratec AudioSystem EWS88 MT was selected. It offers eight analog input and output converters that process audio signals with up to 24 bit and 96 kHz sampling frequency. Furthermore two analog output converters with up to 18 bit and 48 kHz are available. The access to the sound card is guaranteed through the Wave-API of Microsoft Visual C++.

7. 'DIFFUSE FIELD' APPLICATION

The 'diffuse field' application offers all available adjustments for the user. For every loudspeaker or headphone the center frequency of the one-third-octave filters as well as the distribution (Gaussian or equal distribution) of the white noise can be chosen separately. Furthermore the volume of each loudspeaker or headphone can be adjusted. Figure 5 shows a screenshot of the 'diffuse field' application.

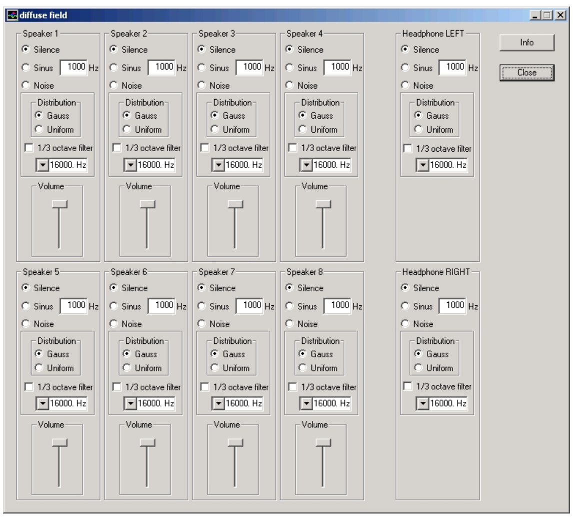

Figure 5: *The 'diffuse field' application.*

8. CONCLUSIONS

In this project it is shown how to realize a diffuse sound field with a PC-based sound card solution. Furthermore a real-time implementation demonstrates the feasibility of a software solution.

9. ACKNOWLEDGEMENTS

The author wish to thank the anonymous referees for the valuable comments.

10. REFERENCES

[1] Jens Blauert, *Spatial Hearing*, MIT Press, Cambridge, Massachusetts, 1996, Original title published under the title 'Raeumliches Hoeren' by Hirzel Verlag, Stuttgart.

[2] Harald Sander, "Erzeugung und Anwendung eines diffusen Schallfeldes in einem reflexionsarmen Raum," diploma thesis, University Hannover / Sennheiser electronic, 1984.

[3] Robinson, "The loudness of diffuse sound fields," *Acustica*, vol. 11, pp. 397, 1961.

[4] BS 5108 (british norm), "British standard method for measurement of sound attenuation of hearing protectors," 1983.

[5] Robinson, "On the measurement of realear attenuation of hearing protectors by standardized test methods," National Physical Laboratory, Acoustic Report Ac 79, February 1977, Teddington, England.

[6] Makoto Matsumoto and Takuji Nishimura, "Mersenne twister: A 623-dimensionally equidistributed uniform pseudorandom number generator," *ACM Trans. on Modeling and Computer Simulation*, vol. 8, no. 1, pp. 3–30, Januray 1998, http://www.math.keio.ac.jp/∼matumoto/emt.html.

[7] Makoto Matsumoto and Takuji Nishimura, "Dynamic creation of pseudorandom number generators," in *Monte-Carlo and Quasi-Monte Carlo Methods 1998*, pp. 56–69. Springer Verlag, ISBN 354066176X, March 2000, http://www.math.h.kyoto-u.ac.jp/∼matumoto/RAND/DC/dc.html.

[8] ANSI, "Specifications for octave-band and fractional-octave-band analog and digital filters," 1993, ANSI S1.1-1986 (ASA 65-1986).