Application of Wave Field Synthesis in the composition of electronic music

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Abstract

Wave Field Synthesis offers new possibilities for composers of electronic music to add the dimension of space to a composition. Unlike most other spatialisation techniques, Wave Field Synthesis is suitable for concert situations, where the listening area needs to be large. It is shown that an affordable system can be built to apply the technique and that software can be written which makes it possible to make compositions, not being dependent on the actual setup of the system, where it will be played. Composers who have written pieces for the system have shown that with Wave Field Synthesis one can create complex paths through space, which are perceivable from a large listening area.

1 Introduction

Spatialisation[1] has been a topic of interest in the development of electronic music since the 1950's; common techniques make use of quadraphonic or octaphonic setups and are based on providing localisation cues based on psycho-acoustics or acoustics (e.g. Chowning 1971). The technique of ambisonics has become popular since the 1990's (e.g. Malham and Matt 1995). There are also various examples where more loudspeakers are used, mostly as setups for one specific piece or location and not as a standardardized setup. A detailed historical overview of spatialisation techniques can be found in Malham & Matt (1995).

The limitation of stereo or ambisonic techniques is that it only works perfectly well for one listener, who is positioned on the so-called "sweet spot". Obviously, in common concert environments the intended effect of movement of the sound will in these cases not be heard by a majority of the listeners.

Wave field synthesis is a technique that can overcome the limitation of only working well for one "sweet spot" and can provide a good perceptual localisation in a relatively large listening area. This makes the techique ideal for concert environments.

Its increasing popularity in audio engineering shows that it is not unlikely that the technique will be available in concert halls and becomes affordable for studios in the near future (see CARROUSO[2]).

This article describes the first experiences with the application of wave field synthesis in the composition of electronic music. A short, comprehensive explanation of the technique is given, a description of the system used in the project at the TU Berlin and the interface software, followed by a description of the possibilities that were used by composers. The pieces described were presented on the Club Transmediale Festival in Berlin, on the 4th of February 2003.

2 Wave Field Synthesis

The concept of Wave Field Synthesis (WFS) is based on a principle that was thought of in the 17th century by the Dutch physicist Huygens (1690) about the propagation of waves. He stated that when you have a wavefront, you can synthesize the next wavefront by imagining on the wavefront an infinite

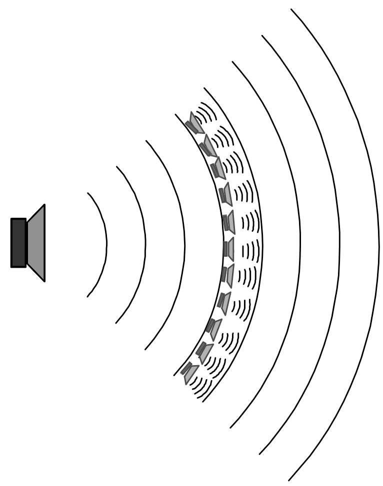

Figure 1. The Huygens' Principle

[1] In this paper no comparisons are made to headphone techniques, as these techniques are quite different from loudspeaker techniques by principle and less suitable for concert situations.

[2] CARROUSO (Creating, assessing and rendering in real-time of high-quality audio-visual environments in MPEG-4 context), http://emt.emt.iis.fhg.de/projects/carrouso/

number of small sources, whose waves will together form the next wavefront (figure 1).

Based on this principle, Berkhout (1988) introduced the wave field synthesis principle in acoustics.

By using a discrete, linear array of loudspeakers (figure 2), one can synthesize correct wavefronts in the horizontal plane (Berkhout, De Vries and Vogel 1993). For a complete mathematical treatment is referred to Berkhout (1988, 1993) and various other papers and theses from the TU Delft3.

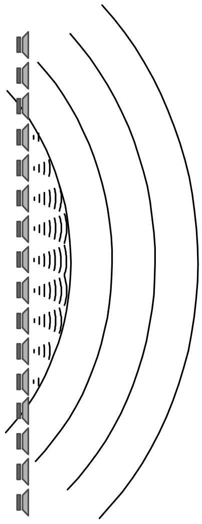

Figure 2. The Wave Field Synthesis Principle

An interesting feature is that it is also possible to synthesize a sound source in front of the speakers (Jansen 1997), which is not possible with other techniques.

Comparisons between measured wave fields and wave fields reconstructed with WFS have shown that the differences between the two are small (Bourdillat 2001); most faults in the WFS reproduction were due to reflections in the reproduction room. Perceptual experiments and practical experience have shown that with WFS one can achieve a large listening area, where the sound source is perceived correctly at the specified location (Vogel 1993, Verheijen 1998). Malham's (2001) comments that WFS cannot achieve a perfect sound image on all locations are true, but perceptually not so relevant that it makes the technique not worth considering for application in spatialisation of electronic music.

2.1 Synthesizing moving sound sources

Jansen (1997) derived mathematical formulae for synthesising moving sound sources. He took into account the Doppler effect and showed that for its application one would need to have continuously time-varying delays. He also showed that for slowly moving sources the Doppler effect is negligible and one can resort to updating locations and calculating filters for each location and changing those in time.

This approach was chosen in this project. Additionally, in order to avoid clicks in playback, an option was built in to fade between two locations to make the movement sound smoother.

3 System setup at the TU Berlin

The prototype system in Berlin was created with the specific aim to make a system for the use in electronic music (Weske 2001). The system consists of a LINUX PC, driving 24 loudspeakers with an RME Hammerfall Soundcard.

The loudspeaker signals are calculated in real time with the program BruteFIR by Torger[4]. This program is capable of making convolutions with long filters in realtime. The filter coefficients can be calculated with the interface software described in this paper.

The current system is capable of playing maximal 9 sound sources with different locations in realtime, even when the sources are moving. This is the maximum amount of sources; the exact amount of sources that can be used in a piece depend on the maximum distance range[5] of each source and the amount of reflections added. Both of these aspects influence the total filter length and the filter length determines the amount of calculation power needed. In table 1 an overview is given of the capability of the system in Berlin (running on a Dual Pentium III). The filter lengths are indicated in samples. The distances are based on the assumption that the sample frequency is 44.1 kHz. The numbers indicated in the table are the real time index calculated by BruteFIR and are a measure for the processor load; to have BruteFIR run stable while sources are moving, it is best not to let the real time index go above 0.80. It can be seen that the maximum filter length and thus the distance range, within which a source can move, can become quite large. On the other hand, the larger the filter length, the larger the I/O delay[6] will be and the time step after which one can change filter coefficients (important for the movement of sources). In some cases, using several partitions of a smaller filter length can diminish the I/O delay.

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[3] Sound Control Group, TU Delft, http://www.soundcontrol.tudelft.nl

[4] Torger, A., *BruteFIR*, http://www.ludd.luth.se/~torger/brutefir.html

[5] During calculation, the smallest delay, considering all path points and all speakers, is subtracted from all delays, so that only delay differences between speakers remain. Thus the filter lengths are based on the largest distance between points on a path.

[6] The I/O-delay is twice as large as the filter length.

Table 1. Overview of processor load (realtime index) and amount of sources per filterlength measured with BruteFIR v0.99f on a Dual Pentium III, 1004 MHz.

		Sources	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
Filt. Len.	***Dist. (m)***	***time***									
256	**1.97**	**6 ms**	0.17	0.23	0.30	0.38	0.49	0.58	0.67	0.77	0.82
512	**3.95**	**12 ms**	0.18	0.25	0.33	0.40	0.53	0.61	0.71	0.80	0.87
1024	**7.89**	**23 ms**	0.20	0.27	0.35	0.42	0.55	0.63	0.72	0.80	0.88
2048	**15.8**	**46 ms**	0.22	0.29	0.37	0.43	0.56	0.64	0.73	0.83	0.91
4096	**31.6**	**93 ms**	0.24	0.31	0.38	0.45	0.59	0.70	0.81	0.94	-
8192	**63.2**	**0.19 s**	0.27	0.36	0.45	0.55	0.72	0.85	0.96	-	-
16384	**126**	**0.37 s**	0.34	0.45	0.58	0.68	0.90	-	-	-	-
32768	**253**	**0.74 s**	0.46	0.63	0.86	-	-	-	-	-	-
65536	**505**	**1.49 s**	0.65	0.84	-	-	-	-	-	-	-
131072	**1011**	**2.97 s**	0.73	0.91	-	-	-	-	-	-	-

4 Interface software

In order to work with the system, interface software was needed to calculate the necessary filter coefficients. The aim was to create an interface that allows composers to define the movements of their sounds, independent of the system on which it eventually will be played. That is, the composer should be bothered as less as possible with the actual calculations for each loudspeaker, but instead be able to focus on defining paths through space for his sounds.

The current version of the program allows the composer to do so. The composer defines the locations and paths through space and gives the time parameters for these. The program will then calculate the necessary filters, based on the hardware setup of the system. As such, compositions can be saved and loaded on different systems, with different hardware setups, and the composition in space that the composer intended will be played back. The sound input needs to be presented at the inputs of the sound card and can come from any source (also a live source).

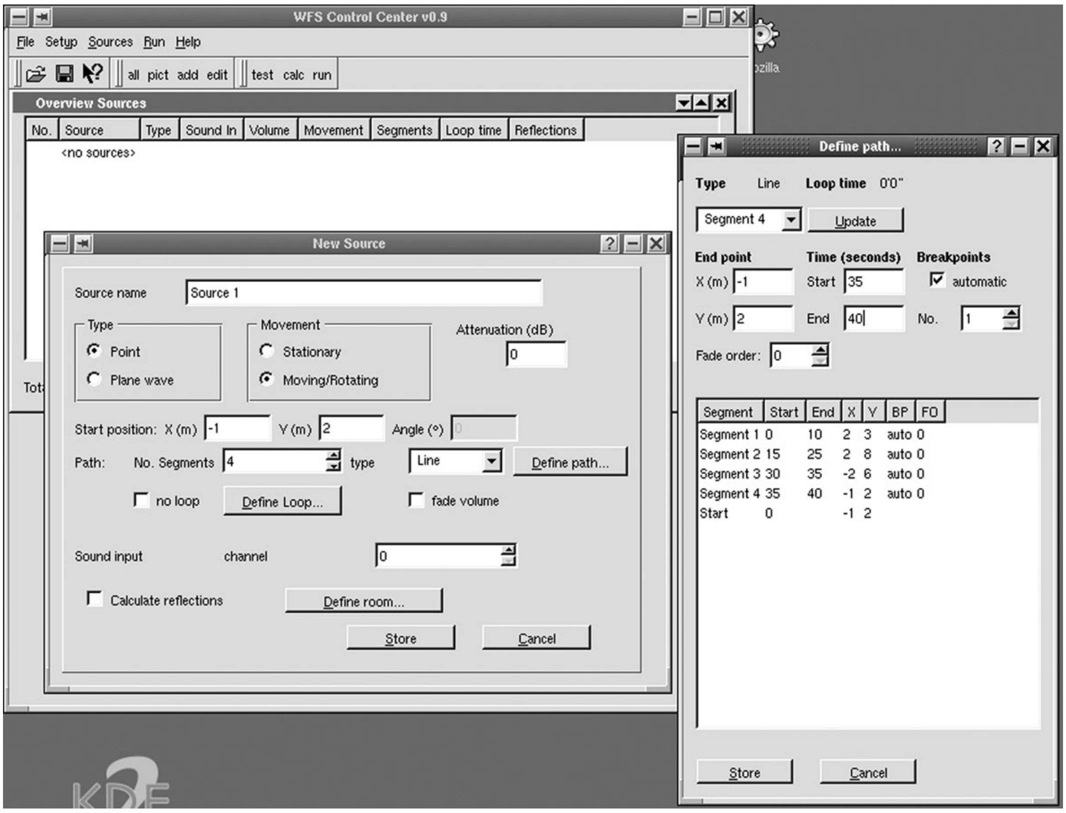

Figure 3. Screenshot of source and path definition. Graphical results of this input is shown in figure 4.

The program can also calculate room reflections, when a room is defined by the user through the position of four walls of a rectangular room, an absorption factor and the order of calculation. The calculations are done with the mirror image source model (see also Berkhout 1988).

Though with WFS one can in principle also create virtual sources in front of the loudspeaker array, this was not yet implemented in the current version. It will however be implemented in a future version.

4.1 Sound source definition

The user can define various sources, each with their own characteristics. A source in this context is the virtual source from which sound emanates in space, whose spatial parameters can be given by the user.

For each source, the user can set the type of source (a point source having a specific location or a plane wave having only a direction), whether it is moving or stationary, its location or angle, the sound input channel at which the sound will be supplied and in the case of a point source, whether reflections have to be calculated or not. If reflections have to be calculated, room characteristics can be defined (these can be different for each source). In the case of a moving source, one can define a path through space and choose to let the movement loop along the path. In figure 3 a screenshot of the source and path definition dialog is given.

After supplying all information and storing it, the user can get two overviews: a general overview in a list, with some of the most important parameters for each source, and a graphical overview showing the paths of the sources through space (figure 4); one can indicate of which sources the path is shown. It is also possible to play a movie to get an impression of the movement in realtime.

For the movement of the sounds, one can set the number of breakpoints along the path and a fade order. A breakpoint is an intermediary point on a path; movement is created by switching from one breakpoint to another. By using a fade between succesive breakpoints, the movement can become smoother and possible clicks in playback can become softer. The user can choose to let the amount of breakpoints on each segment be calculated automatically. In that case, the program uses a total of 40 breakpoints per source and divides these over the segments of the path, depending on the length of the segment and of the path and on the time interval.

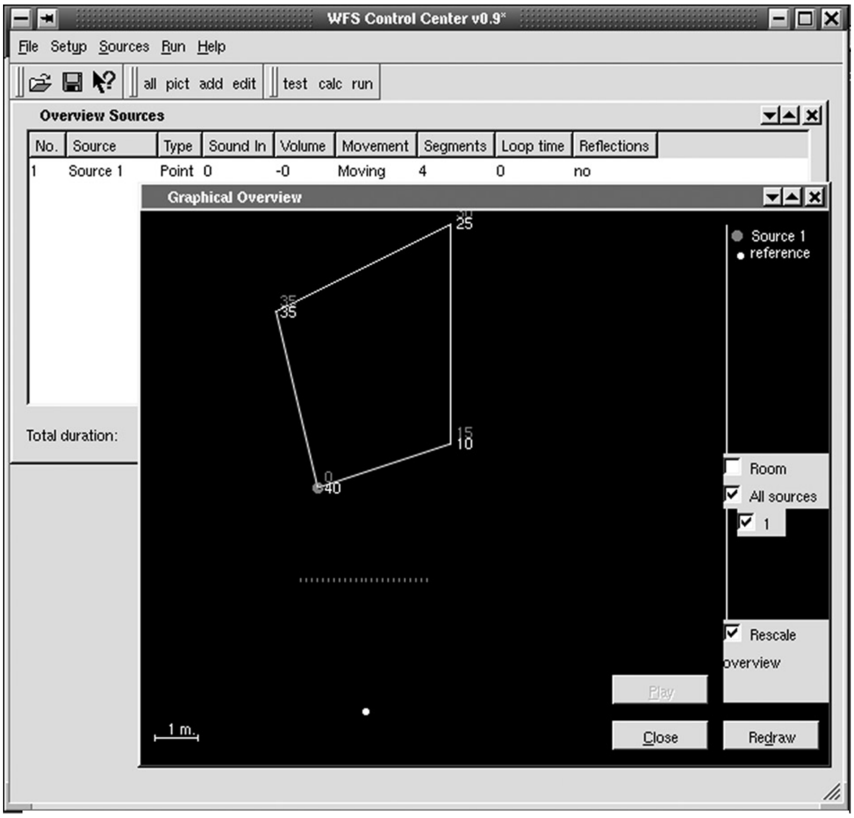

Figure 4. Screenshot of the graphical overview of the source path. The numbers at the points between segments indicate the departure (dark) and arrival (light) times. The dots in between the path and the reference point are indicating the loudspeaker array.

In practice, one needs to experiment with the optimal settings for the amount of breakpoints and the fade order in order to bring clicks to an acceptable level. Whether clicks are audible also depends on the type of sound that is moving. Sounds with a narrow frequency band, tend to create more clicks when moving than broadband signals.

In some instances one cannot get rid of the clicks altogether as BruteFIR has a minimum time after which it can update filter coefficients. The exact time depends on the filter length or block size. In the program the minimum time step was set to 200 ms. between breakpoints and to 50 ms. for a fade step.

5 Experiences with composers

For the Club Transmediale festival seven pieces for the system were prepared by seven composers: Frieder Butzmann, Boris Hegenbart, Marc Lingk, Robert Lippok, Markus Schneider, Ilka Theurich and Marije Baalman. All composers had different backgrounds; all had previously composed electronic music. I will elaborate about three works that were created.

Marc Lingk, a composer residing in Berlin, wrote a piece called Ping-Pong Ballet. The sounds for this piece were all made from ping-pong ball sounds, which were processed by various algorithms, alienating the sound from its original. Using these sounds as a basis, the inspiration for the movements was relatively easy as the ping-pong ball game provides a good basis for the distribution in space of the sounds. In this way he created various loops of movement for the various sounds as depicted in figure 5. Paths 1 & 2 are the paths of the ball bouncing on the table, 3 & 4 of the ball being hit with the bat, 5 & 6 of multiple balls bouncing on the table, 7 & 8 of balls dropping to the floor. Choosing mostly prime numbers for the loop times, the positions were constantly changing in relative distance to each other. The movement was relatively fast (loop times were between 5 and 19 seconds). In the beginning, the piece gives the impression of a ping-pong ball game, but as it progresses the sounds become more and more dense, creating a clear and vivid spatial sound image.

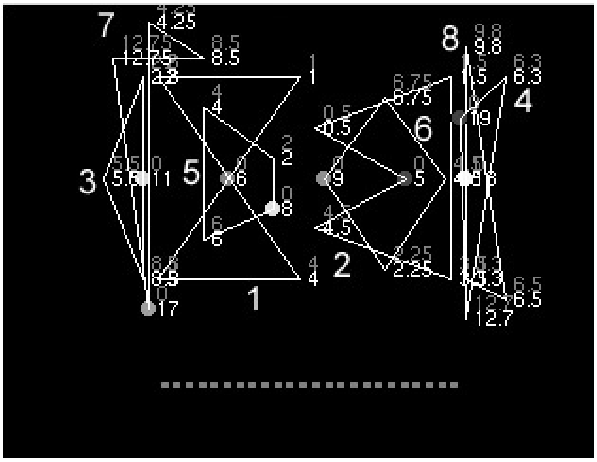

Figure 5. “Ping-Pong Ballet” movement overview. The large numbers indicate the path numbers; the small numbers are the time indications. The row in the front is the loudspeaker array.

The author, Marije Baalman, made a piece where the movements were based on a painting created before in a rather improvisational way. The different colours in the painting were mapped to different sounds and they also had different movement characteristics. One source was moving perpendicular to the array, another parallel to the array. Yet another was zigzagging to and from the array, one source was jumping from one location to another. The other two sources had other types of paths that are less easily stereotyped. The exact movement in time was made dependent on the sounds. Silences on the sound input were used to let the virtual source jump to another position for the next sound to start its path.

As the movements were relatively slow and the sound was not very dense, the movements and different positions of the sound could be heard quite clearly.

These two examples show that with WFS it is possible to create more complex paths through space than is possible with most other spatialisation techniques.

Ilka Theurich, a student of sound sculpture in Hannover, was interested most by the possibilities of including virtual rooms and reflections into the composition.

One sound was placed in a rather small room with fully reflecting walls. This resulted in a sound that was virtually at several locations (due to the mirror image source model). As the sound from the actual source location was the first sound to reach the listeners' ear, the sound would however still be located there by the listener.

Other sounds were placed in a larger room, while others were moving without being placed in a room. One of the sources was created as a plane wave, which allowed the listener to get different perspectives on the composition by moving through the listener area. The plane wave sound only had a direction and as such was always in front of one, with a specific angle, whereas the other sounds had clearly defined locations. While the listener moved, the plane wave sound would “walk along”, while the point source sounds stay fixed in their position. In this way the listener could determine his own version of the composition by chosing his own location.

The effect of the movement and reflections were the most clear for recorded sounds (having a rich spectrum), as opposed to synthetic sine-based tones.

In order to limit the CPU-load, some compromises had to be made: the total amount of reflections calculated was reduced.

During the work the idea came up to enable the room characteristics to change in time, which possibly can also provide an interesting effect. This will be implemented in a future version.

6 Concert

The concert took place at the Club Transmediale Festival on the 4th of February. This is a festival that includes electronic music both from (underground) club culture and from more academic approaches.

The hall in which the concert took place measured about 105 square meters and was relatively reverberant. The array was positioned on the stage a little bit above ear height.

The concert was preceded by a short presentation explaining the wave field synthesis technique and the software that the composers used to create the movements of their sounds.

During the concert, the biggest problem was that the system with its 24 loudspeakers could not create enough loudness for the amount of people who filled the hall (ca. 100 listeners). This had as a disadvantage that the people in the back could not perceive the music very well and were a bit loud as they started to talk. During the sound check (without the sound absorbing people in the hall) the system was loud enough for the whole hall and the effect was even clear in the back of the hall.

For the presentation of a prototype system the concert can be regarded rather as a success. The listeners who were in the front could perceive very well the movements of the sounds in the compositions. Especially when closing the eyes, some people commented that the music created a vivid visual image with its movements through space. Others were quite amazed that they could really move around the source, that is, position themselves on a different relative location to the virtual source. A sound artist, who works a lot with ambisonics, commented that especially the distance of various sources can be much better modelled with WFS than with ambisonics.

The pieces of Lingk, Lippok and Baalman were received best, as the movements of the sounds in these pieces were the clearest. This is probably due to the type of sounds that they used, which all had a broad frequency spectrum, thus enabling listeners to locate the sound more clearly.

Some listeners were disappointed, as the system was not yet a full surround system.

After the concert several other composers showed an interest in applying the system for their own work, varying from electronic music concerts, to sound installations, to a combination of electronic music with dance.

7 Conclusions

From the experiences of working with composers to create electronic music with application of wave field synthesis, we can conclude that the technique opens up new possibilities for spatialisation of electronic music. The interface proved to be easy to use, as the composers did not need to know the exact mathematical calculation and could think in familiar terms of positions in meters; the graphical feedback provided a good overview. Paths through space could be made more complex than is possible with other spatialisation systems.

Reactions from the public show that the effect reached with WFS provides a new experience for electronic music, as they can choose their own position relative to the positions of the various virtual sound sources.

For application in concert environments it became clear that a larger system is needed, both to meet the problem with loudness and to create a surround effect, where the benefits of WFS will become clearer, especially with regard to modelled reflections.

It can be expected that more work with composers will open up new possibilities of the WFS-technology for the spatialisation of electronic music. By involving composers in the development phase of the system, composers can influence the direction of development of the system and interface software.

8 Future work

The work at the Electronic Studio in Berlin will be continued to implement new options in the interface software. The possibilities for defining the path in time will be expanded, variable room dimensions will be included, as well as the possibility to synthesize sources in front of the array. Also, in order to increase realistic distance perception, high frequency decay with distance will be implemented.

The next step will then be to implement real time control over the movements, in order to allow application in live electronic music performances. Control can then be issued with commonly used protocols such as MIDI or OSC.

Plans are made to enlarge the prototype system, so that a surround effect can be created. This will allow more freedom for movement and enable to get more realistic room reflections. An interesting problem to solve will come up then: synchronising multiple computers to drive the different speaker arrays with the needed precision.

Future research will include modelling more complex source characteristics (which could be more realistically resembling acoustic instruments; thinkable is frequency dependent, directional sources) with WFS.

Application of WFS for auralisation in order to be able to listen to a composition in its performance environment in the studio is also a topic for future research.

Next to further development on the technical side, we plan to work more with composers with the system, also to see whether combinations of the WFS-system with other spatialisation techniques deliver interesting results.

9 Acknowledgements

To Club Transmediale[7] for giving the opportunity to present the work at their festival and inviting composers to work with the system.

To the Sound Control group at the University of Technology in Delft for use of their figures.

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[7] http://www.clubtransmediale.de