Keynote

WAVE FIELD SYNTHESIS – A PROMISING SPATIAL AUDIO RENDERING CONCEPT

Günther Theile

Institut für Rundfunktechnik (IRT)
Munich, Germany
theile@irt.de

ABSTRACT

Modern convolution technologies offer possibilities to overcome principle shortcomings of loudspeaker stereophony by exploiting the Wave Field Synthesis (WFS) concept for rendering virtual spatial characteristics of sound events. Based on the Huygens principle loudspeaker arrays are reproducing a synthetic sound field around the listener, whereby the dry audio signal is combined with measured or modelled information about the room and the source's position to enable the accurate reproduction of the source within its acoustical environment. Not surprisingly, basic and practical constraints of WFS systems limit the rendering accurateness and the perceived spatial audio quality to a certain degree, dependent on characteristic features and technical parameters of the sound field synthesis. However, recent developments have shown already that a number of applications could be possible in the near future. An attractive example is the synthesis of WFS and stereophony offering enhanced freedom in sound design as well as improved quality and more flexibility in practical playback situations for multichannel sound mixes.

1. INTRODUCTION

Three psychoacoustic fundamentally different spatial audio imaging methods should be distinguished:

- (Multichannel) loudspeaker stereophony
- Binaural reconstruction of the ear input signals
- Syntheses of the sound field around the listener

All known spatial sound systems can be traced back to one of these methods or can contain mixed forms thereof, whereby certain advantages of the methods are being exploited, respectively its disadvantages are avoided, dependent on the intended application area.

1.1. Loudspeaker stereophony

This is in principle based on the characteristics of localization in the superimposed sound field, generated by two loudspeakers [1]. Directional imaging is done in the imaging area between two adjacent loudspeakers [2]. In the case of 3/2 stereophony, with the assistance of surround channels the imaging area between the front loudspeakers can be extended. Therefore possibilities are offered for the reproduction of early lateral sound for imaging of spatial depth as well as reverberation, in order to produce the spatial impression and the envelopment. Details are described in [3].

1.2. Binaural reconstruction of the ear input signals

The original employment of this method is the known dummy head stereophony. It is not intended to reproduce a suitable sound field at the reproduction location. Instead, the effective ear signals in the recording location are recorded with the assistance of a dummy head – and replayed in principle via headphones. Under ideal circumstances, the reproduced binaural signals are identical to the original ear signals that the listener received in the recording location. In practice it is possible to reproduce auditory events with excellent realism regarding spatial characteristics and sound color.

1.3. Synthesis of the sound field around the listener

The third approach has been pursued within the framework of the European Research Project "CARROUSO" [4]. It is based on the concept of Wave Field Synthesis (WFS, developed at the Technical University Delft, refer e.g. [5], [6]), i.e. the representation of a virtual source and a virtual room is achieved by rendering an acoustically correct sound field. The principle of WFS is based upon the assistance of loudspeaker arrays, when a complete sound field is generated in the listening zone which is identical to an appropriate real sound event (see Section 2). This acoustical counterpart to the optical holography is also described as "holophony". The binaural ear input signals that are active for the auditory event thus arise in a natural way within the sound field, contrary to dummy head stereophony.

1.4. Combining stereophony and WFS

Further developments of spatial audio systems are based on useful combinations of basic methods, using sophisticated real time convolution algorithm. This paper concentrates on overcoming certain practical drawbacks of multichannel sound on the one hand and of WFS on the other. So called "Virtual Panning Spots" (VPS) are introduced to improve the WFS rendering quality of large complex sources (e.g. "choir"), to reduce the number of WFS transmission channels and to ensure compatibility and scalability. Useful combinations of VPS and conventional or sophisticated stereophonic panning and mixing techniques will provide advanced facilities for spatial sound design. A special VPS application allows play back of conventional multichannel mixes in a virtual high quality listening room rendered by means of WFS technologies, offering full backwards compatibility with usual loudspeaker stereophony, optimum multichannel format flexibility, as well as attractive practical benefits in the home, in the cinema, or in other applications.

2. WFS PRINCIPLES AND PROPERTIES

2.1. The “Huygens” principle

“If from a point S of a homogeneous isotropic medium a spherical wave is emitted, one can imagine the procedure of the individual wave reproduction in that a particle brought into oscillation by external forces, transfers its movements to its neighboring particles. This procedure then continues symmetrically in all directions and in this way gives cause to a spherical wave...” [7].

If a sound source S (the so-called “primary source”) emits spherical wave fronts, one can imagine in accordance with the “Huygens” Principle each emitted wave front (refer to Figure 1a blue), through the addition of all participating “secondary sources” (which also emit spherical waves) on the surface O. Due to the knowledge of the wave front on surface O (Figure 1a red) the state of the oscillation can be determined at an arbitrary point P of the sound field. The wave front through point P is constructed through the summation of all participating secondary source signals.

(a)

(b)

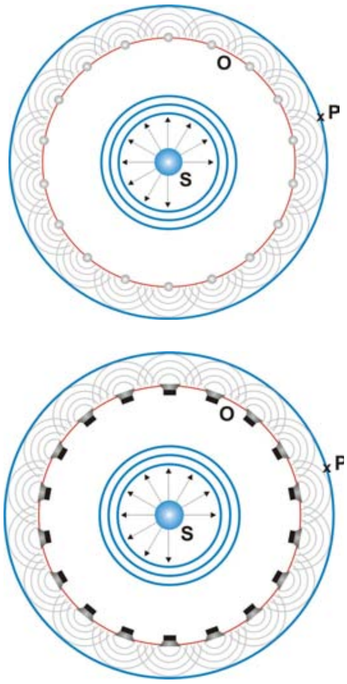

Figure 1: *The “Huygens” Principle: (a) Theoretical Model; (b) Application WFS.*

In principle, in the case of the WFS, one replaces the secondary point sources by loudspeakers and in this way again produces a spherical wave (refer to Figure 1b). The sound source S is virtual; the listener in point P receives the same wave front which is transmitted by the sound source S.

2.2. WFS – the application of the “Huygens” Principle

This applies correspondingly for a circular arrangement of the loudspeakers on a two dimensional level. Concerning a sound source S, (which emits a sine impulse and is located in an infinitely large plane without demarcation of walls), a wave front results as illustrated in Figure 2a). If one now places an array of n microphones (M) in this primary sound field and one reproduces the recorded microphone signals via an equally arranged array of n loudspeakers (L) – special equalization has to be included according to the relevant physical basics – in a reproduction room (Figure 2b), one obtains the synthesized wave front in the (red dotted) listening area. At any place in the listening area the listener perceives a virtual sound source S, as he can move around freely, whilst the virtual sound source remains correctly localized in terms of its direction (see [5] or [8]).

(a)

(b)

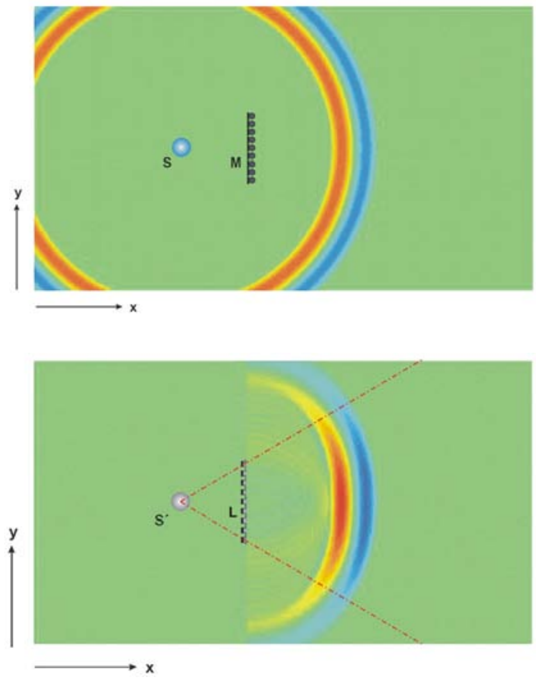

Figure 2: *Principle of WFS: (a) ideal source response and (b) typical output of a finite WFS array.*

2.3. Special Properties of WFS

2.3.1. Localization of virtual sources

Through WFS the sound engineer has a powerful tool to design a sound scene. One of the most important (with respect to conventional techniques) novel properties is its outstanding capability of providing a realistic localization of virtual sources. Typical problems and constraints of a stereophonic image vanish in a WFS sound scene.

In contrast to stereophony WFS is able to:

- produce virtual sources that are localized on the same position throughout the entire listening area, refer Figure 5: The red (dashed) and pink (dotted) arrows indicate the directions of the auditory events when the red and pink virtual point sources are reproduced.
- produce plane waves that are localized in the same di-

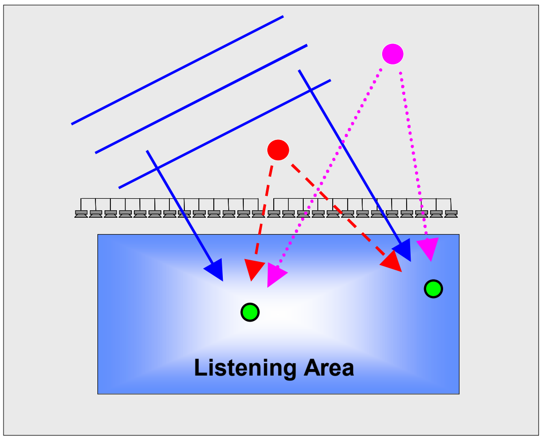

Figure 3: *WFS is capable of reproducing both the stable positions of point sources (red and pink, dashed and dotted) and the stable direction of a plane wave (blue, solid).*

rection throughout the entire listening area, refer Figure 5: The blue (solid) arrows indicate the direction of the auditory event when the blue plane wave is reproduced.

- enhance the localization of virtual sources and the sense of presence and envelopment through a realistic reproduction of the amplitude distribution of a virtual source. In other words, when the listener is approaching the location of a virtual source the amplitude increases in a realistic way. Accordingly, the amplitude of a plane wave - which can be seen as a source in infinite distance - changes least on different listener positions.

These properties enable the synthesis of complex sound scenes which can be experienced by the listener while moving around within the listening area. Figure 3 illustrates the way in which the sound image changes at different listening positions. This feature can be made use of deliberately by the sound engineer to realize new spatial sound design ideas.

Moreover, it has been shown that the enhanced resolution of the localization compared with stereophony [9] enables the listener to easily distinguish between different virtual sources, which makes the sound scene significantly more transparent.

2.3.2. Virtual Sound Sources in Front of the Loudspeaker Array ("Focused Sources")

Figure 4 shows the wave fronts of a point source behind the array (a) and in front of the array (b) in a simulation. The concave wave fronts of Figure 4a achieve the synthesis of the signal of a virtual source behind the array. WFS, however, is also capable of synthesizing a virtual source in front of the array. Therefore the WFS array emits convex wave fronts which focus in a point that will be localized as the position of the "focused source" (Figure 4b). Naturally, the localization will not be correct for listening positions between the focus point and the array because the sound emission of the virtual source occurs here reversely.

For the practical application, it is an enormous progress that virtual sound sources in the field between the listener and the loudspeakers can be created. As will be discussed in Section 4, sound engineers can be offered completely new tools for the spatial sound design.

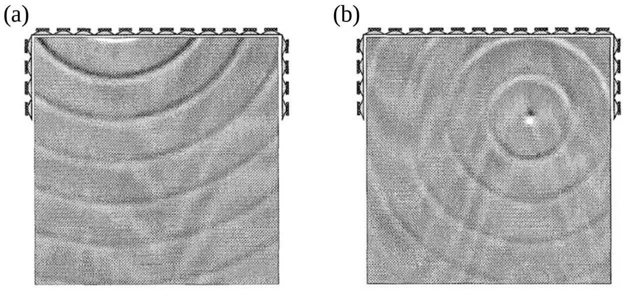

Figure 4: *Wave fronts of virtual sources [10]: (a) behind the array; (b) in front of the array.*

3. WFS PRACTICAL CONSTRAINTS

Not surprisingly, in practice it is not possible to match all theoretical requirements for a perfect result. The rendered WFS sound field differs from the desired sound field to some degree for a number of reasons (for details see [11]).

3.1. Discreteness of the array (spatial aliasing)

This effect produces spatial and spectral errors of the synthesized sound field due to the discretisation of a continuous secondary source distribution. Above the spatial aliasing frequency f_{alias} the time difference between two successive loudspeaker signals interferes at the listener's position, depending on the spatial sampling interval, i.e. the loudspeaker / microphone inter-spacing.

3.2. Reflections of the reproduction room (spatial interference)

A WFS array can not render the desired sound filed perfectly if reflections of the reproduction room produce interference in spatial perception. In particular, perception of distance, depth and spatial impression are affected, because fragile distance cues of synthesised sources can be dominated by the stronger distance cues generated by the array speakers. They interfere with the desired reflection pattern of the synthesised source. Special room compensation algorithms being under investigation [12], [13] will perhaps be able to minimize this effect.

3.3. Restriction to the horizontal plane

Theory does not restrict WFS to the horizontal plane. However, the reduction of the array dimension to the horizontal plane is the practical approach, having a number of consequences. First, virtual sources can be synthesized only within the horizontal plane. This includes virtual reflections affecting the completeness of a natural reflection pattern and thus possibly resulting in impairments of perception of distance, depth, spatial impression and envelopment.

Another aspect is related to the measuring techniques used for capturing the room response. In practice there is some mismatch with respect to elevated reflections, because the measured room response includes elevated reflections although they are reproduced only in the horizontal plane. The effects of these types of inaccurateness on spatial perception parameters are not well-known yet.

Furthermore, horizontal arrays do not generate real spherical waves, but cylindrical waves. In the case of imaging a plane wave for example there results an error with respect to the level roll-off (3dB/doubling of distance), in comparison with the ideal plane wave (no roll-off) [11], [14].

3.4. Limitation of array dimensions (diffraction)

In practical applications the loudspeaker array will have a finite length. Due to a finite array so-called diffraction waves originate from the edges of the loudspeaker array [11], [14]. These contributions appear as after-echoes (and pre-echoes respectively for focussed sources), and – depending on their level and time-offset at the receiver's location – may give rise to colouration.

However, methods to reduce these truncation effects are known, e.g. by applying a tapering window to the array signals. This means that a decreasing weight is given to the loudspeakers near the edges of the array. In this way the amount of diffraction effects can substantially be reduced at the cost of a limitation of the listening area [14].

3.5. Effects on perception

Although a number of authors have suggested methods to deal with the practical limits of rendering accurateness or to minimize their effects, there is still a lack of knowledge (some details can be found e.g. in [5], [6], [11], [12], [15]). Several effects of the constraints on specific perceptual attributes are not known yet in detail. However, this knowledge is important for further developments of WFS systems in view of future applications.

Current psychoacoustic studies are concentrating on the subjective evaluation of principle characteristics of WFS systems in comparison with stereophonic or binaural systems. They are necessary to evaluate the resulting impacts on attributes of spatial perception not only with respect to the development of WFS systems for different applications but also in view of scientific knowledge. Particular attention should be turned to the perception of direction, distance, spatial depth, spatial perspective, spatial impression, reverberance, and envelopment, as well as sound colour.

4. WFS APPLICATIONS

4.1. The European CARROUSO Project [1]

The European CARROUSO Project ("Creating, Assessing and Rendering in Real Time of High Quality Audio-Visual Environments in MPEG-4 Context") has intended to break several limitations of these current commercial systems by merging the new WFS rendering technique with the flexible new coding technology MPEG-4 standard, allowing object-oriented and interactive sound manipulation.

By means of the MPEG-4 format the signal of the source ("Gestalt") and its spatial properties are transmitted separately. For reproduction, the dry source signal is convolved with the measured or modelled set of impulse responses (containing the spatial information), and emitted by a loudspeaker array. In contrast to stereophony WFS is able to

- produce virtual stable sources localized at the same position throughout the entire listening area,
- produce virtual sources in front of the loudspeaker array ("focused sources")
- produce plane waves that are localized in the same direction throughout the entire listening area,
- enhance the sense of depth, spatial impression and envelopment through a realistic reproduction of the original room response

The key objective of the project CARROUSO was to provide a new technology that enables to transfer a sound field, generated at a certain real or virtual space, to another usually remote located space, in a bit efficient way at highest perceived quality. The principle block diagram is illustrated in Figure 5, it shows three functional components:

<u>Capturing</u>
For recording of the sound field a microphone array technology is applied. Signal processing calculates the position of the sound sources which could be fixed or moving. The microphone array as well as the video cameras is used to gather relevant information on the acoustical conditions in the recording room. Acoustic models can be obtained to parameterise the acoustic data set, thus making it suitable for transmission.

<u>Transmission</u>
Encoding of audio objects is operated by the MPEG-4 standard and encapsulated into specific data streams. For broadcasting applications the transmission adopts digital video broadcasting (DVB) streams. The coding uses a subset of MPEG-4 components (no predefined profile for the given application in the standard).

<u>Rendering</u>
In the rendering process the transmitted data are demultiplexed, decoded and processed by a compositor. It enables the reproduction of a recorded or simulated sound field via WFS loudspeaker arrays, ensuring immersive sound perception in a wide listening area. The original acoustics of the reproduction room, which may negatively influence the obtained result, is optimally corrected for.

All components were combined to a demonstrator, as a basis for validation. This was done using perceptual experiments and field tests. The results of the project are a first step towards a new quality of high quality spatial audio imaging. CARROUSO has shown the possibility to capture, transmit and render sound sources and their related acoustic environment with more realism, compared to existing stereophonic methods.

This novel spatial audio technology was developed for applications in conjunction with moving pictures, using the recently introduced MPEG-4 standard. It is considered as a major milestone for immersive audio representation at public places and in private households. Two applications have been targeted within this project. The first one concerns high quality spatial audio with associated video for broadcasting. The second application is related to cooperative and interactive work on immersive audio objects. CARROUSO is expected to contribute to information, communication and media technology.

[1] EU-Project IST–1999-20993 (Jan. 2001 – June 2003): [4].

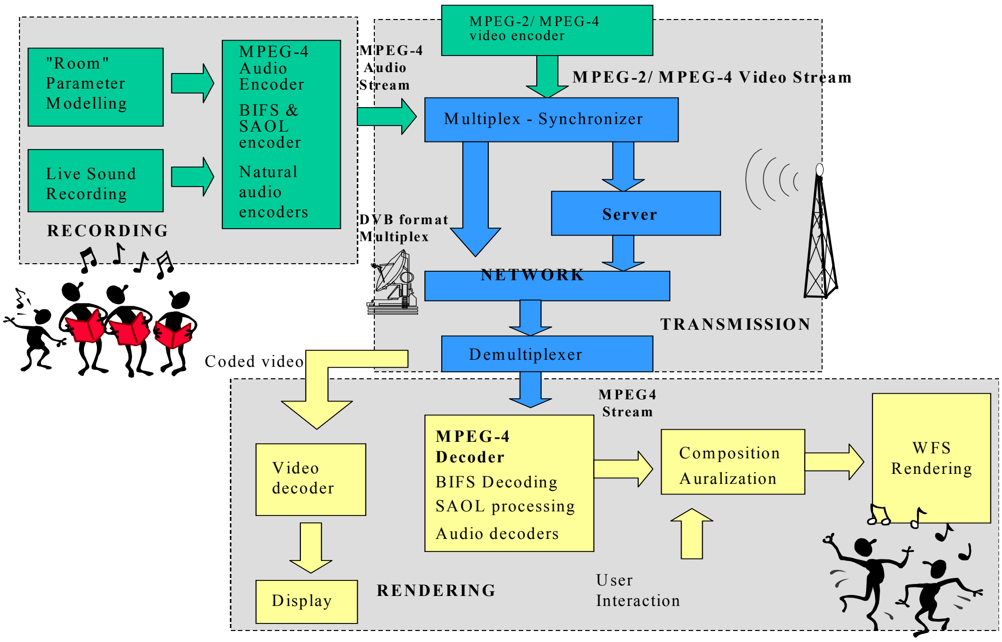

Figure 5: *Principle block diagram of the CARROUSO demonstrator.*

4.2. Synthesis of WFS and stereophony

This paper concentrates on the synthesis of WFS and stereophony. Figure 6 illustrates the basic concept by means of music recording. Step one is always the room response measurement in the music hall, done e.g. with a stepwise rotating microphone. This measured spatial information is stored in the WFS processor.

For recording of orchestra and soloist closely spaced spot microphones are used. The stereophonic orchestra mix should be composed in a way that it contains as little room information (reverb, reflections, etc.) as possible; but it should contain the adequate spatial distribution of elements. This three channel stereophonic mix signals and the soloist signal are being convolved with the appropriate spatial impulse responses. As a result, the rendered WFS sound field represents stable virtual sources located in the concert hall. Listeners within the listening area perceive a three-channel stereophonic image of the orchestra and a point source image of the soloist, whereby the reproduced characteristics of the concert hall give a new sense of realism.

On this basis apparent advantages of established conventional stereophonic recording techniques on the one hand and of WFS technologies on the other can, in principle, can be utilized through a purposeful combination.

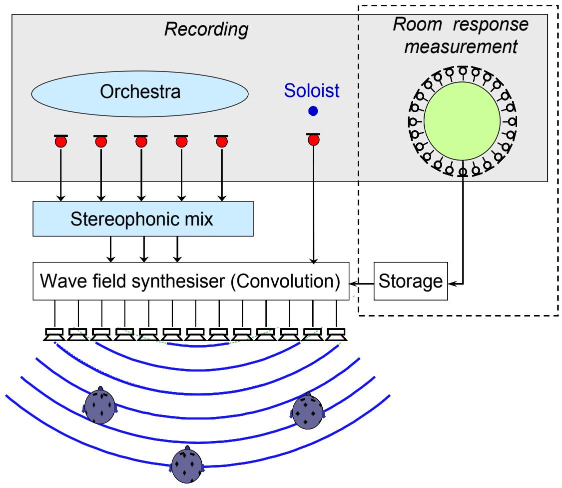

Figure 6: *WFS: Separate handling of sources and spatial information.*

4.3. Virtual Panning Spots (VPS)

The key tool is use of so-called Virtual Panning Spots (VPS) [16], virtual point sources to be applied for panning across any stereophonic imaging plane in the virtual WFS imaging area. VPS can be understood as virtual “loudspeakers” which reproduce the stereophonic sound image of a spacious sound source (e.g. a choir) in the recording room (see also [16], [17], [18]). The suitable room impulse responses have to be measured in the original room or to be created artificially in a suitable way. In the example in terms of Figure 7, the orchestra is imaged with the assistance of six VPS, which are reproduced via WFS and are relatively freely configurable with regard to localization, expansion and distance.

The sound design advantage of this concept is self-explanatory: The stereophonic recording of the orchestra according to Figure 6 produces a spacious sound image of the sources as there is an image between the VPS in accordance with the principles of phantom sources localization. The “loudspeakers” are virtual sources, generated through WFS and provided with the room characteristics of the recording room. The locations of the VPS behave direc-

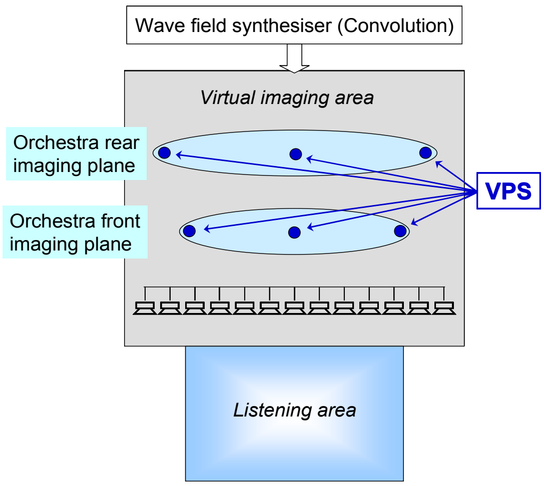

Figure 7: *Use of Virtual Panning Spots (VPS)*

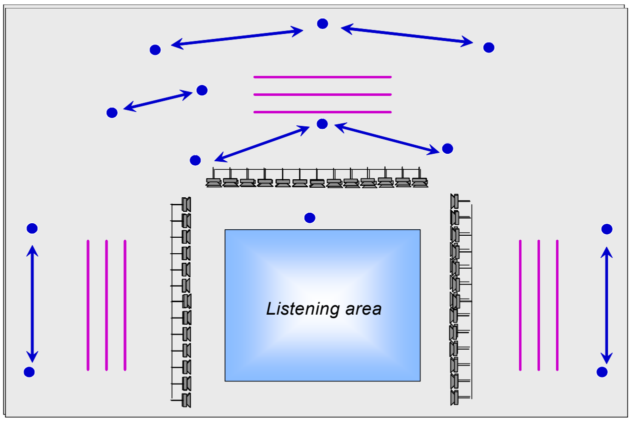

Figure 8: *Example for the configuration of the stereophonic imaging areas.*

tionally stable in the listening area. The known disadvantages of phantom source localization, especially the low directional stability can be easily avoided by employing a sufficient number of VPS.

The number of stereophonic imaging areas is in principle arbitrary. From an artistic point of view, one should orientate oneself towards the number of spacious instruments or instrument groups (in large ensembles, e.g. string groups, brass player groups, choir). The number and spatial distribution of the VPS depends on the following criteria:

- Size and shape of the homogeneous ensemble
- Circumstances of the production
- Artistic and sound balance-related intention of the sound engineer
- Available transmission capacity

Virtual Panning Spots, VPS, are selected points ("virtual loudspeakers"), which produce a stereophonic representation area. These can in principle be selected at choice in accordance with the recording situation and the desired sound image. As an example, Figure 7 shows two rear imaging planes and two front imaging planes, offering easy imaging of depth. The imaging area can be "spread out" by an arbitrary number of VPS in a random spatial expansion in accordance with the situation and intention.

The artistic arrangement of the ensemble upon the WFS transmission commences with the choice, dedication and positioning of the VPS. Three parameters should be mentioned, which lends the sound engineer to new possibilities of spatial sound design (refer to the example in Figure 8):

1. In the case that loudspeaker arrays are installed lateral to the listening area, there are in principal, no problems as far as directional stability is concerned, as a lateral stereophonic representation sector can be built up from a sufficient number of stable VPS. The same applies to the sector behind the listener.

2. A stereophonic imaging area does not only allow itself to be moved in all directions, stretched out or compressed, but can also be presented in an extensive range with different distances. The representation of depth is thus easily recognizable.

3. With certain constraints (see Section 2.3.2 and Figure 4), the VPS can be placed in a distance between the listener and the loudspeaker array and also with the stereophonic imaging field. Thus, the virtual imaging area theoretically reaches in dense closeness to the listener and allows for an expressive representation of depth.

The practical example shown in Figure 8 contains diverse new possibilities of the spatial sound design. A total of 13 VPS were employed, whereby those more remote have been arranged in terms of the 3/4 stereo ITU standard and act in the known fashion as stereophonic imaging areas (also see [3], Section 3). In the front (stage) area there are stereophonic representation areas in three different distances and also the lateral sectors have been prepared for stereophonic imaging. In addition a monophonic virtual source has been provided in the front between the listening area and the loudspeaker array, e.g. for a soloist.

The individual stereophonic presentations should contain the (dry) direct sound of spacious sound sources or ensembles, with as little room response as possible. The conventional tools for the representation of spatial depth (generation of suitable lateral reflections are superfluous within the concept of WFS and actually damaging, due to the fact that here by virtual sources (and VPS), the relevant distance information is available - in the form of room impulse responses.

In Figure 8, the parallel lines represent the plane waves for the reverberation synthesis. The reverberation is not produced with the assistance of VPS. Rather is the effective source to reproduce a plane wave infinitely far away [19].

4.3.1. Creation of VPS

The locations and number of VPSs should be found before the WFS recording starts. Suitable loudspeakers are positioned in the VPS areas of the recording room. The recording room is stimulated, the response is measured and the impulse response is calculated from this (see Figure 6). Of course, instead of a real measurement of the room impulse response, model based procedures can be employed to create artificial room impulse responses.

In practice, for certain halls, such determination of the room impulse responses does not have to be done at each recording. Rather, this once determined data is permanently available in a data base. Also impulse responses from other halls or pseudo-realistic artificial (and sometimes better) characteristics can be desirable (comparable with the input of modern reverberators).

4.3.2. *Reduction of transmission channels*

Through the utilization of the VPS concept, the number of transmission channels as well as the computing capacity can be significantly reduced. The larger an ensemble, and also the number of individual participating instruments, the larger saving in transmission channels is. In the case of a small ensemble or individual instruments, however, the VPS concept does not save transmission channels or only saves a few (see [16]).

4.3.3. *Handling aspects*

A major gain in the utilization of the VPS concept is the improved handling of the recording techniques and the improvement in the spatial quality of the WFS reproduction. The handling of the VPS concept is based on the application of stereophonic techniques, which all sound engineers are familiar with. There are not any completely new microphone techniques; the mode of operation remains the same as during conventional productions (except for the representation of distance, which is automatically given within the configuration of VPS). Also the mix to the VPS positions requires no new modus operandi - conventional panning and mixing techniques are being used. The number of microphones remains approximately the same as in the case of an appropriate stereo recording.

The recording in accordance with the VPS concept thus has the significant advantage that the sound engineer can directly employ familiar processes of work. The improvement on spatial quality by way of WFS goes hand in hand with the incorporation of stereophonic techniques. The sound engineer has clearly more influence over important quality characteristics of a recording, he has more creative room for manipulating the spatial parameters and can more strongly influence the sound color.

4.4. Virtual loudspeaker reproduction

An important application of the VPS technique is a special preset of the VPS setup on the reproduction side, which enables the reproduction of conventional multichannel recordings in a virtual listening room [16]. For this purpose, two modifications are suggested for the WFS decoder, which can be activated in the event of need for application (see Figure 9):

1. The configuration of the VPS with regard to room impulse responses and spatial arrangement is done in accordance with the preset setup of virtual loudspeakers in a virtual listening room. Arbitrary arrangements of the virtual loudspeakers can be preset and be activated dependent on the stereophonic format to be reproduced.
2. The virtual source signal is not received via the transmission channel, but from the multichannel decoder on the reproduction side (e.g. that of a DVD player).

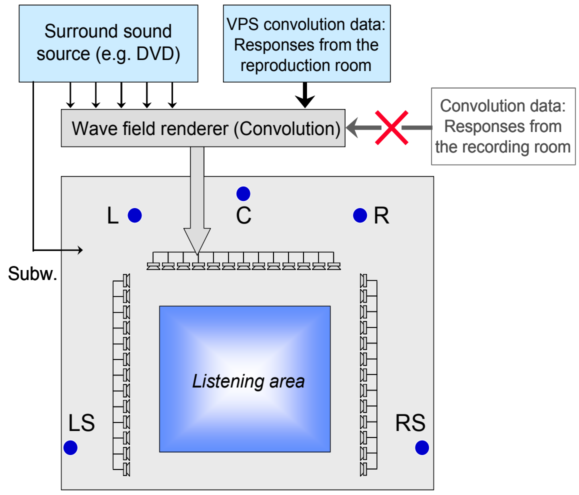

Figure 9: *VPS configuration for rendering virtual multichannel loudspeakers.*

The WPS reproduction unit operates completely detached from WFS transmission, and can principally offer three attractive advantages:

1. Diverse stereophonic multichannel formats can be easily reproduced optimal through the selection of a VPS preset, without having to appropriately adjust the loudspeaker arrangement within the living-room.
2. The virtual loudspeakers can also be placed outside the living-room, i.e. also in a confined area situation, the listening area for multi-channel stereophony is sufficiently large.
3. A future high quality WFS reproduction unit will allow for an electronic compensation of diverse defects in the reproduction room [12], especially the reduction of the effect of the early reflections and the balancing of asymmetrical arrangements of the speaker array.

From the technical and practical point of view the application of WFS for multichannel stereo reproduction could be the first step towards acceptance in the market place. In this regard, the development of the so-called MAP technology (see e.g. [20], [21]) is important. The flat panels, e.g. fed with glass fiber cables, can often be better integrated into the living-room and are more attractive than conventional loudspeakers. Thereby not only the application of virtual loudspeakers within the home is envisaged, but also the employment in cinemas, theaters as well as for high quality sound reinforcement.

5. ACKNOWLEDGMENT

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Günther Theile has obtained his Ph.D. from the Techn. University of Berlin in 1980. In his thesis he introduced a general theory on sound source localisation. Based on this, he proposed the sphere stereo microphone and the diffuse field equalization, which has become ITU-R BS. 708 for studio monitor headphones. In 1984 he invented the masking threshold based subband coding technique and started the development of the MUSICAM system, which is now ISO/IEC Standard (MPEG 1 Audio Layer II). He was engaged in investigations on dynamic range control and on multichannel sound, including multichannel source coding (MPEG 2 Audio Layer II). Under his chairmanship ITU-R TG 10-1 has produced Recommendation BS. 775, known as 5.1 surround standard.

His current activities are concentrating on further developments in the field of multichannel sound and research on spatial audio systems based on binaural techniques and wave field synthesis.

In his fields of work Dr. Theile has published about one hundred papers, and he has been granted a number of International Patents. In 1992 he received the Lothar-Cremer-Medal for his basic work on perceptual source coding. He has received the AES Fellowship and AES Board of Governors Awards. He was Chairman of the 102nd AES Convention in Munich and of the AES 19th Conference on Surround Sound. He has chaired ITU-R Working Party 10C in the period 1997-1998. Currently he is serving as Chairman of the AES South German Section, as Vice-Chairman of the AES Technical Committee on Multichannel and Binaural Audio Technologies, as Chairman of the VDT specialist group "Research & Development" and "Surround Sound Forum", as well as Chairman of the Programme Committee of the German Tonmeistertagung.