VDT - Tonmeistertagung - 2012

A 3D and Multiformat Microphone Array Design for the GOArt Project - 'The Organ as a Memory Bank' -

by Michael Williams
Free Lance Sound Recording Engineer
'Sounds of Scotland' – mike@soundsscot.com

Abstract

The Organ as a Memory Bank is a research project at the Göteborg Organ Art Center (GOArt), University of Göteborg, financed by the Swedish Research Council, and carried out in collaboration with the Division of Applied Acoustics, Chalmers University of Technology, Göteborg. Organ pipes are designed and voiced with consideration for the acoustic properties of the room in which the organ is situated (from the acoustic point of view the room is certainly a "part of" the instrument). It was considered necessary to look for an optimal "neutral" recording and playback technique to archive realistic reproduction of the organ and its surrounding acoustics, and be able to compare this recording with any future modifications. The 3D/Multiformat Microphone Array Design approach was considered as a possible recording system. A trial period was arranged in August 2012 by the Göteborg Organ Center and the Acoustics Department of the University of Göteborg. This presentation will describe the proposed microphone array together with some of the observations from the various participants in this project, together with some observations concerning the psychoacoustics of height perception.

Introduction - The Organ as Memory Bank.

The Organ as a Memory Bank is a research project at the Göteborg Organ Art Center (GOArt), University of Göteborg, financed by the Swedish Research Council, and carried out in collaboration with the Division of Applied Acoustics, Chalmers University of Technology, Göteborg.

The historical organs of Sweden and Europe form a cultural landscape of great national and international significance and constitute an important map of both tangible and intangible cultural heritage. Questions of conservation of this heritage has been addressed in Sweden because of the change in relationship between the Swedish State and the Swedish Church. The organ, as well as all other inventories, were specifically mentioned when the Swedish legislation regarding Cultural Heritage was updated in 1999.

The organ can be seen as a collective memory bank of multiple kinds of handcraft, traditions of instrument building and specific developments within music performance. A typical organ is made up of many thousands of parts and is often contained in a case that can be as large as a small house. Performing a thorough technical documentation is part of the work of safeguarding the cultural heritage built into the organ and its case. Thorough documentation of the organ and its case can be a daunting task. A technical documentation only partly provides information for a better understanding of the intangible heritage that each of these objects encompasses. Photos and recordings of pipe and register sounds as well as music performances are a part of the documentation that must be preserved.

Since 1995 when the Göteborg Organ Art Center was established, documentation of the physical properties of the organ has been an important part of its research. To complement this documentation, GOArt has also developed a research model in which the building of a copy of an organ as close as possible to a well-preserved original helps verify the technical documentation, and rediscover the tacit knowledge that these objects contain, which normally is not accessible through measurement and observation alone. Through several documentation and research projects, encouraging collaboration between instrument builders, engineers, conservators and antiquarians, GOArt uses its "Organ Documentation Manual" to structure the information collected in the database.

Documentation of the sound of an organ and its habitat before and after restoration or other work is needed and has not been systematically and comprehensively done earlier. A sound recording, based on current technology developed for the commercial market, does not suffice as a technical documentation. In spite of this being current practice, and as such, well established and functional for its own needs, any commercial recording will be strongly affected by artistic judgments made by the recording engineer. It is therefore important for documentation purposes to develop a durable scientific and psycho-acoustically relevant sound documentation methodology. It is necessary to develop an optimal recording method and define the necessary equipment, analysis and evaluation methods and also to develop and define the optimal presentation methods for the sound documentation.

Organ pipes are designed and voiced with consideration for the acoustic properties of the room in which the organ is situated (from the acoustic point of view the room is certainly a "part of" the instrument). It was considered necessary to look for an optimal “neutral” recording and playback technique to archive realistic reproduction of the organ and its surrounding acoustics, and be able use this recording as a reproduction tool for comparison with any future modifications. Optimal techniques must be developed that will allow the playback formats to remain available to researchers and the general public now and as far as possible into the future. In contrast to the printed word, the sound documentation may need to be “reformatted” at various times with respect to advances in audio technology, in order to remain accessible.

The 3D/Multiformat Microphone Array Design, proposed by Michael Williams and described in AES Convention papers 7057**[1]** and 8601**[2]**, was considered as a possible recording system. A trial period was arranged in August 2012 by the Göteborg Organ Center and the Department of Applied Acoustics of the University of Göteborg represented by Professor Mendel Kleiner. Four different venues in Göteborg were proposed for the recordings - the Ôrgryte Nya Kyrka (in Figure 1)(Organist – Joel Speerstra). the Organ Hall in the Academy of Music and Drama of Göteborg (Organist – Svetla Tsvetkova), the Mariakyrkan (Organist – Per Högberg) and the Vasa Kyrkan (Organist – Per Högberg).

FIGURE 1 – THE ÔRGRYTE NYA KYRKA ORGAN

The Microphone Array

The main 1st layer of the 3D/Multiformat Microphone Array is the 7 or 8 channel **M**icrophone **A**rray **G**enerating **I**nterformat **C**ompatibility (M.A.G.I.C. Array) as described in AES preprint 7057**[1]** and 7480**[2],** plus a 2nd layer array of vertically orientated figure of eight or supercardioid microphones spaced at 52cm as described in AES preprint 8601**[3].**

The 1st layer – the M.A.G.I.C. Array

First of all it must be said that this type of array is a univalent microphone/loudspeaker system – each loudspeaker receives a signal from one and only one microphone. In addition, although the loudspeakers are positioned in the same order around the compass rose of reproduction, they do not necessarily have the same coverage segment angles as the microphones.

To understand the MAGIC Array concept it would seem appropriate to consider its development, stage by stage, starting with the front stereo pair. Most people will be familiar with the Stereophonic Recording Angle diagrams that I have published previously in AES preprints. In this case we will consider the SRA diagram for **hypo**cardioid microphones with 10db back attenuation. (a Schoeps MK21 or CCM 21 for instance)

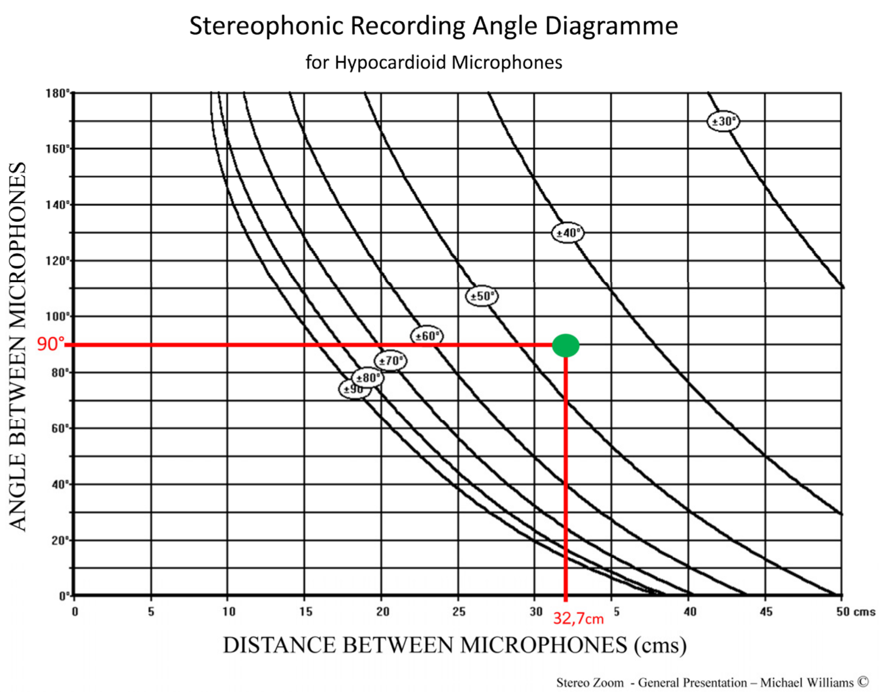

FIGURE 2 - STEREOPHONIC RECORDING ANGLE FOR HYPOCARDIOID MICROPHONES
90° / 32.7CM – SRA +/- 45°

Figure 2 show a particular solution for a SRA of +/- 45° using 90° between the axis of directivity of the microphones and 32.7cm between the capsules. This means that the **Stereophonic Recording Angle is the same as the angle between the microphones**.

The next step is to apply these same dimensions to a quadraphonic square. Figure 3 shows the layout of a typical equal segment quad array – first published at the 91st AES Convention in New York in 1991**[4].** This paper was updated to include up to eight channels in 2008[**2**] as shown in Figure 4. From this figure we can see that the distance between hypocardioid capsules for a Quad Array must be 32.7cm.

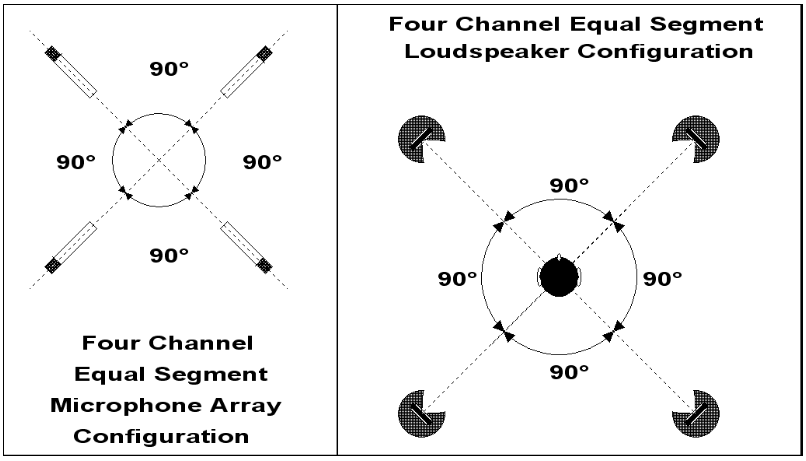

FIGURE 3 – THE QUAD SQUARE

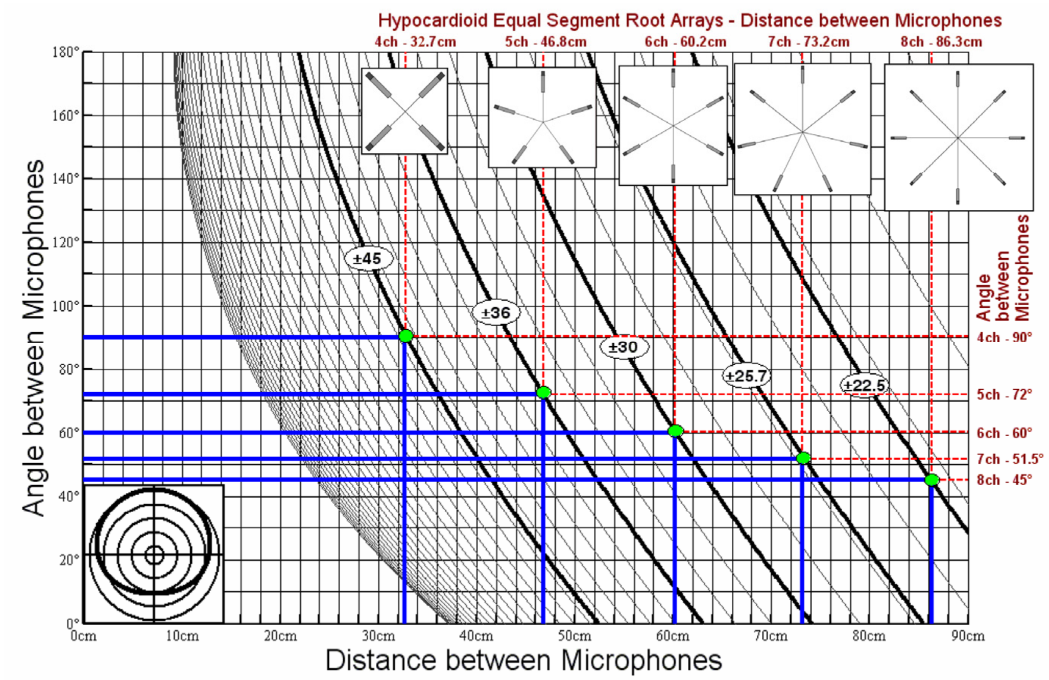

FIGURE 4 – THE SRA DIAGRAM FOR HYPOCARDIOID MICROPHONES FOR EQUAL SEGMENT MICROPHONE ARRAYS

Figure 5 shows a close-up of the Hypocardioid Quad Square used in the core position of the full array.

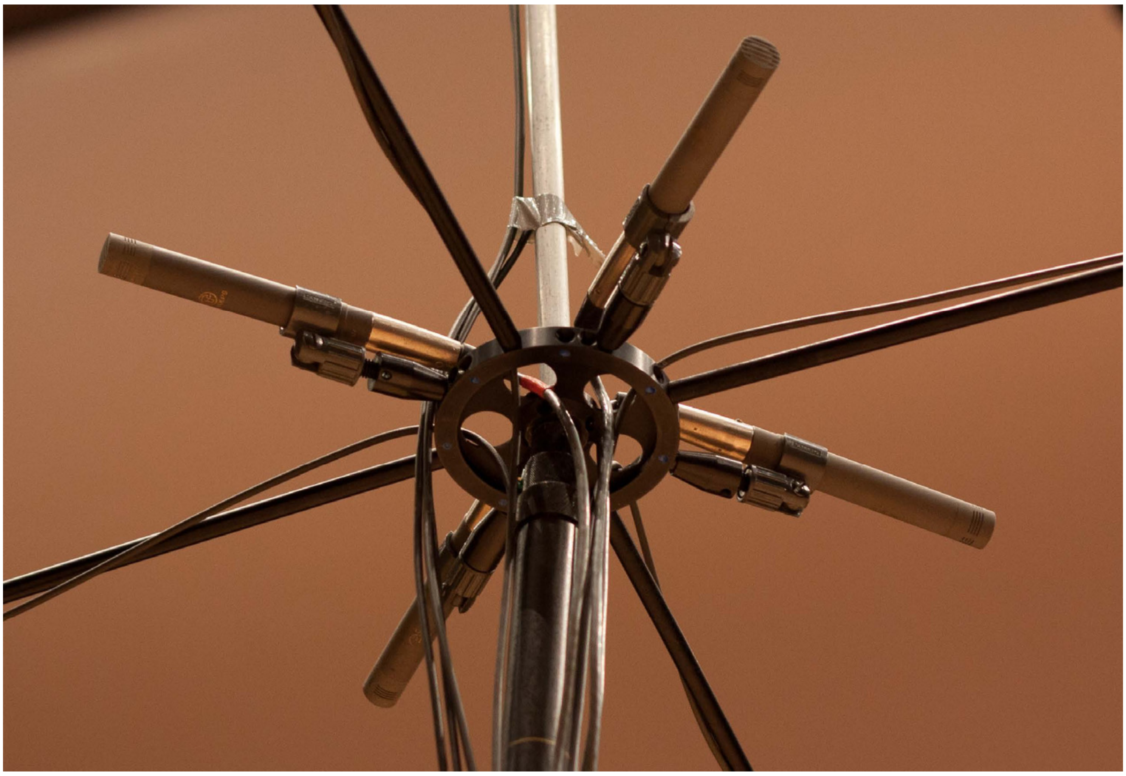

FIGURE 5 – CLOSE-UP OF CENTRAL QUAD SQUARE
USING THE 'WILLIAMS STAR' MICROPHONE ARRAY SUPPORT SYSTEM.
(PHOTO BY MICKAEL KENNESSI)

Using just the Quad Square it is also possible to create a mixdown to stereo that will fold the surround sound segments into the front stereo sound image – this is called a 'Twisted Quad' mixdown and is described in AES Preprint 6373**[5].** The following mixdown algorithm should be used – Ls is mixed with the Right channel, and Rs is mixed with Left channel. The percentage of Rs and Ls mixed with the front Left and Right channels respectively will depend on the ratio of direct to reverberant sound that is desired.

The next stage is to introduce the satellite microphones on the long arms placed on the bisector or each of the angles within the Quad Square. These microphones are placed on arms of about 1m20 long as shown in Figure 6. The theory of this configuration is explained in detail in AES preprints 7057**[1]** and 7480**[2].**

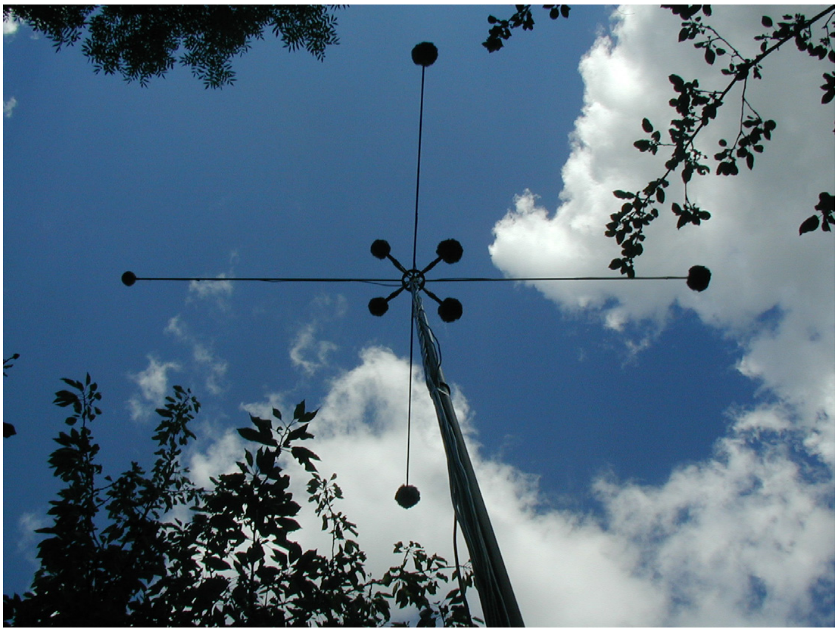

FIGURE 6 – THE MICROPHONE ARRAY GENERATING INTERFORMAT COMPATIBILITY (M.A.G.I.C.)
USING THE 'WILLIAMS STAR' MICROPHONE ARRAY SUPPORT SYSTEM

In order to simplify the explanation of the configuration it can be considered that each satellite microphone will produce two new coverage segments to the clockwise and anticlockwise sides of each satellite microphone. The sum of two adjacent segments for example between the Left and Centre(the satellite) microphones, and the Centre(the saltellite) and Right microphone will be equal to the total segment coverage between the Left and Right microphone, as shown in Figure 7

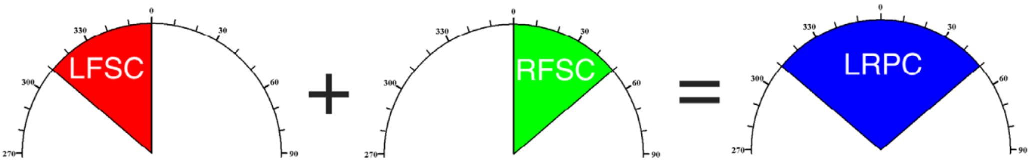

FIGURE 7 – SEGMENT COVERAGE OF THE FRONT SEGMENT

In other words the Left Front Segment Coverage must be a total of 45° coverage, and the Right Front Segment Coverage must be also 45°. The Total Coverage of the Front Segment therefore adds up to 90° (i.e. +/-45° when talking about a stereo segment). The only way to obtain such a small segment coverage (45° or +/-22.5°) for the LFSC and RFSC is to increase the distance between the Left & Centre, and Right & Centre microphones. The segment coverage in both cases will unfortunately be offset to the left and to the right. The only way to bring the segment coverage so that it corresponds to the actual angle between the microphones is to apply a 3mS delay to the Centre microphone. This is again explained in detail in the two AES preprints cited above **[1]** & **[2]**

When this construction is applied to each of the four segments, it will create an eight channel master recording array, as shown in Figure 6.

The main advantage of this MAGIC configuration is that it will be compatible with many different reproduction formats.

- Stereo – by using just the Left and Right microphones
- Or Twisted Quad stereo by mixing the Ls and Rs into the Right and Left microphone channels respectively
- Quadraphony using only the Quad Square of microphones
- 5 Channel reproduction using the Quad Square plus the Centre satellite microphone
- 7 Channel reproduction using the Quad Square plus the Centre, Left Median and Right Median satellite microphones.
- 8 channel reproduction using the Quad Square plus all the satellite microphones.

Remember, this type of microphone array is a univalent microphone/loudspeaker system – each loudspeaker receives a signal from one and only one microphone.

The system for the 1st Layer of sound recording described above uses hypocardioid microphones. This is by no means the only possibility, cardioid or supercardioid microphones can also be used as long as the distance parameter is adjusted to take this into account. It may also be interesting to use hypocardioid microphones for the Quad Square and only cardioids for the satellite microphones. This configuration using cardioids (for the satellite microphones) tends to produce a better balance in bass frequency response when using the 5 or 7 channel reproduction configurations, otherwise the bass response may become a little over-powering, especially in 7 channel reproduction.

The 2nd Layer – addition of height information

As described in AES preprint 8601**[3]** no height information can be generated by loudspeakers placed only in the horizontal plane – the complex pattern of localization cues that can be used in Binaural Technology to produce this height information is completely destroyed by the high level of acoustic crosstalk in loudspeaker reproduction.

We therefore need to resort to a 2nd layer of loudspeakers to initiate a natural perception of height. However localization information generated by this 2nd Layer must not be in conflict with the main horizontal plane localization information. Two approaches are possible

- eliminate as far as possible the interaction between the two planes of sound catchment using either Figure of Eight (or Supercardioid microphones), with the maximum attenuation angle of the directivity pattern directed towards the direct sound source
- if interaction exists, then it is necessary to construct the 2nd layer of sound catchment in such a way as to generate localization information that does not conflict with the main horizontal plane.

The first approach was the one adopted for the GOArt project in Göteborg, and also for a previous pilot recording of contempory music at the Watford Colosseum in London. However in critical listening tests conducted after this series of recordings, it was found that it was not possible to completely eliminate any interaction between the two layers of catchment.

The second layer was made up of four figure of eight microphones pointing at 90° to the horizontal plane and spaced at 52cm between each capsule, as shown in Figure 8. With these Schoeps CCM 8 Figure of Eight microphones the directivity patterns are actually pointing upwards at 90° to the body of the microphone as shown in Figure 8. They are positioned at 0°, 90°, 180° and 270° with respect to the compass rose and the 2nd layer is about 1m above the main 7 channel MAGIC array, as shown in Figure 9.

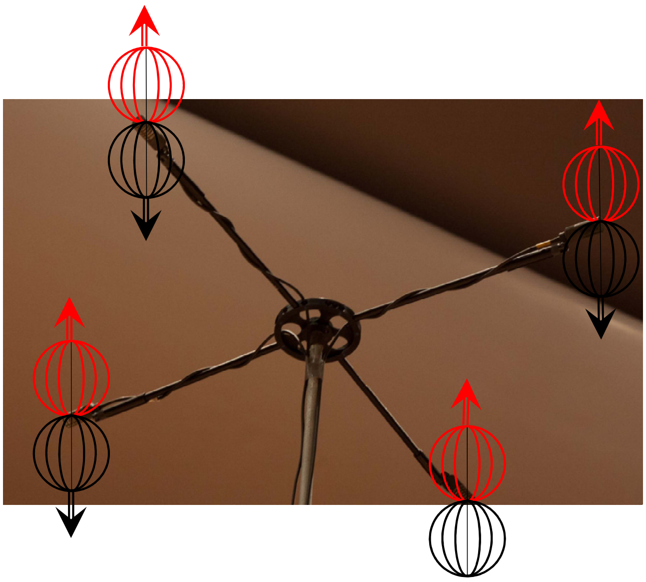

FIGURE 8 – THE SECOND LAYER – THE VERTICAL FIGURE OF EIGHT CROSS
(PHOTO BY MICKAEL KENNESSI)

FIGURE 9 – THE COMBINED 1ST AND 2ND LAYERS
- THE FULL TWELVE CHANNEL ARRAY -

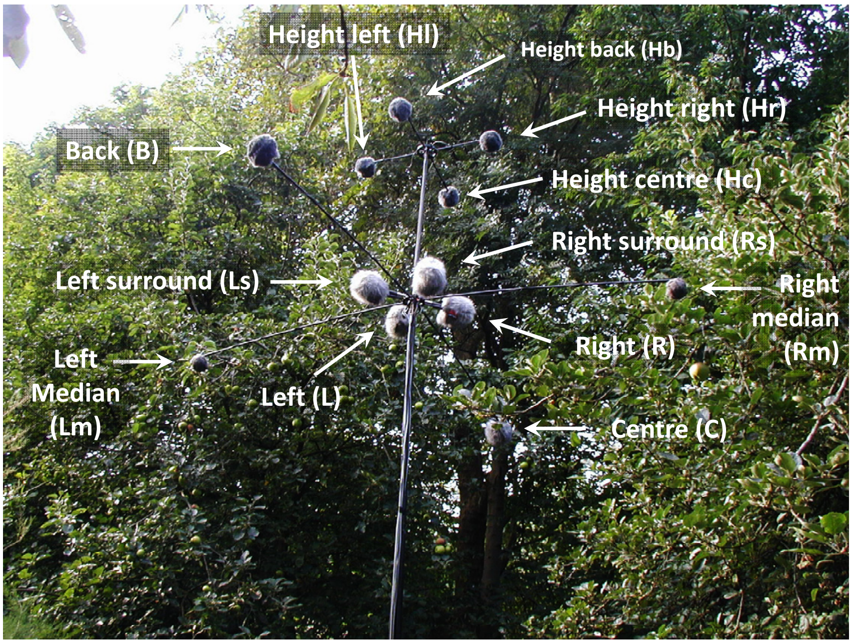

FIGURE 10 – THE COMBINED 1ST AND 2ND LAYERS
.- THE FULL TWELVE CHANNEL ARRAY WITH IDENT.

What is the reason for choosing 52cm between the 2nd layer capsules?
The Figure of Eight directivity patterns are all orientated vertically, so we can consider that this creates a 'Time-Difference-only' sound recording system. From any of the SRA diagrams (for instance in Figure 1) we can see (by extrapolation) that 52cm between the capsules (with 0° between the axes of directivity) means that we have a Stereophonic Recording Angle (SRA) of +/- 45° for each coverage segment (a total of 90° coverage for each segment). For this to be absolutely true the microphone array must be placed in plane wave propagation conditions – in other words the sound source must be relatively distant (at least five times the distance between the microphones). This indeed was the situation in all the recordings where this system was tested – the 2nd layer capitation was essentially reverberation information

We can consider that, in the full 12 channel microphone array configuration, there could be redundancy between the four satellite MAGIC array microphones and the height microphones. In fact a perfectly satisfactory 3D sound image is obtained by using only eight microphones – the central Quad Square part of the MAGIC array and the Figure of Eight Cross of the 2nd Layer. But if only eight microphone channels are recorded then this would mean that the master recording would not contain information that was compatible with the 5 channel multichannel format or the 7 channel blu-ray format. So in practice all 12 channels must be recorded for full compatibility.

The Primary Isosceles Triangle structure
The relative positions of the main MAGIC layer and the 2nd or top layer is such that each 2nd layer microphone forms an isosceles triangle with two of the microphones in the Quad Square. We will call the 2nd layer microphones Hc (Height centre at 0°), Hl (Height left at 270°), Hr (Height right at 90°) & Hb (Height back at 180°).

As can been seen in Figure 11, the primary set of isosceles triangles are formed by:

- L, R and Hc in the front (in white)
- Ls, L and Hl on the left hand side (in red)
- R, Rs and Hr on the right hand side(in green)
- Rs, Ls and Hb at the back (in black)

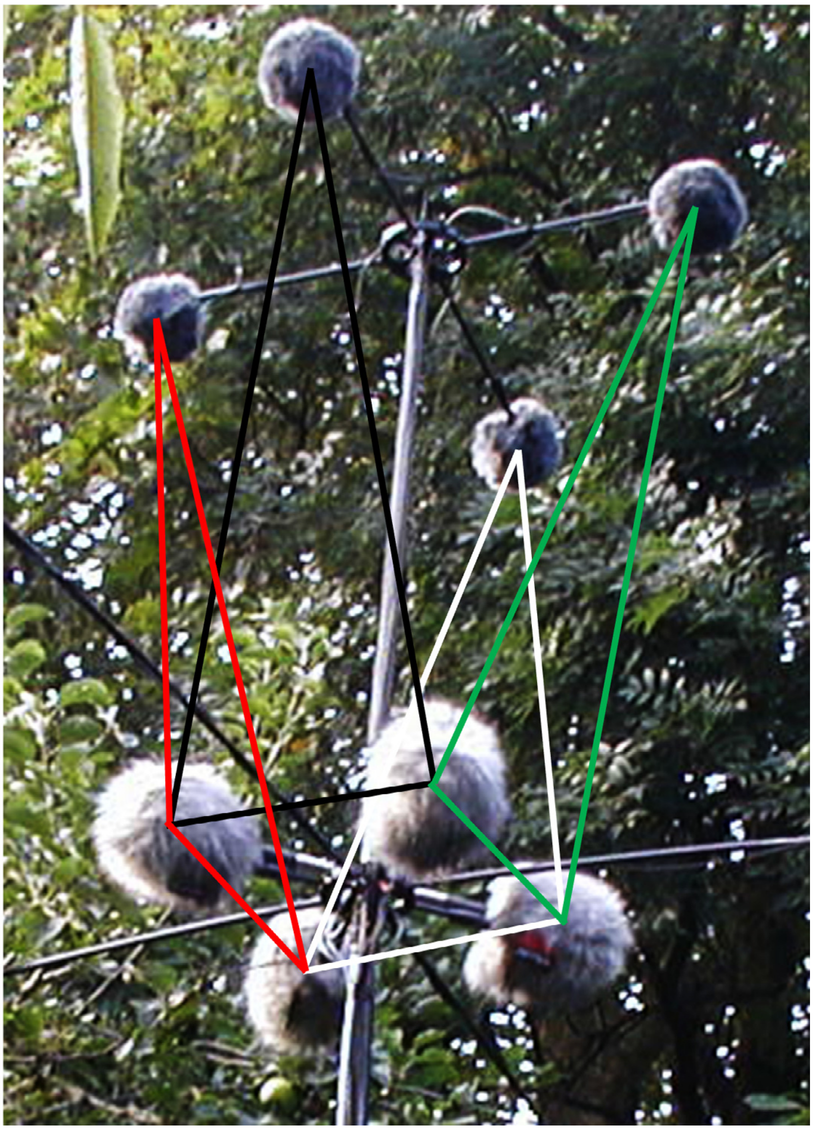

FIGURE 11 – THE PRIMARY ISOSCELES TRIANGLES
IN THE 12 CHANNEL MICROPHONE ARRAY

Figure 12 shows the Primary Isosceles Triangle Structure in the temporary listening room installed at the Applied Acoustics Deptment of Göteborg University.

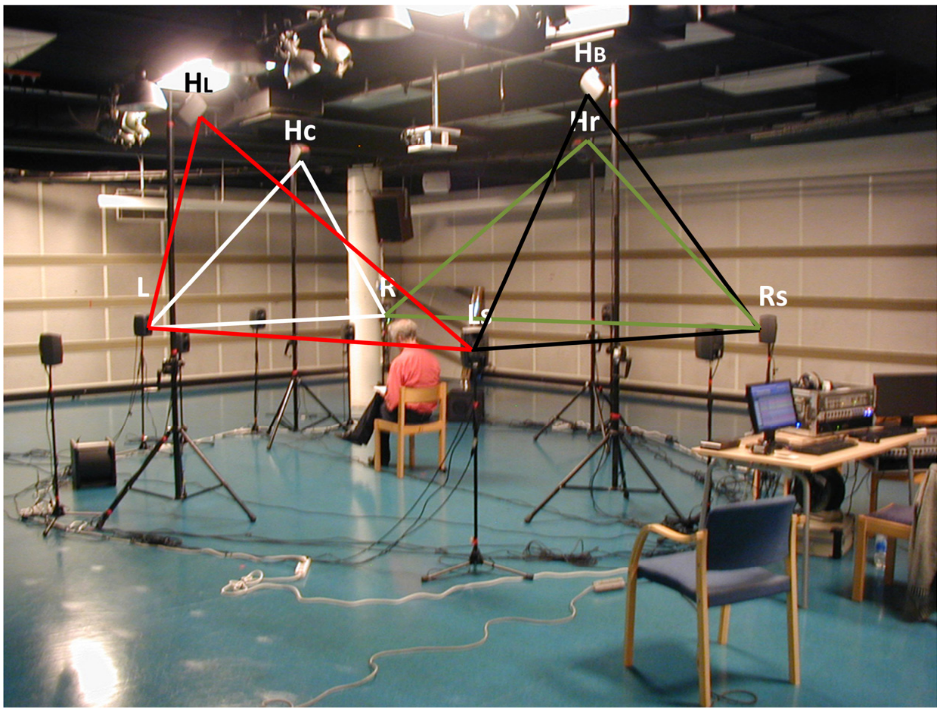

FIGURE 12 – THE PRIMARY ISOSCELES TRIANGLES
IN THE 12 CHANNEL LOUDSPEAKER CONFIGURATION

In AES Preprint 8601, I expressed the opinion that localization in the vertical plane using a standard horizontal array of loudspeakers was extremely difficult to achieve. This was based on examination of the HRTF characteristics in the vertical plane. In Binaural Technology, using headphones or earphones very carefully matched to the individual listener's HRTF, vertical localizatiion can be quite satisfactory. Acoustic crosstalk in loudspeaker listening makes this process almost impossible. However very precise crosstalk cancellation, and listener HRTF matching can produce a similar effect but it is highly listener position dependent (within a few millimeters).

The Psychoacoustics

The introduction of a 2nd layer of loudspeakers already introduces a natural vertical localization sound source. This means that we could expect virtual localization between loudspeakers to fulfill the role of creating vertical dimension localization and therefore with the complete configuration of loudspeakers, 3 dimensional space reproduction.

In the design of the 3D Multiformat Array used in this series of recordings in Göteborg, some difficulty in vertical localization was expected when loudspeakers were mounted one above the other, but it was considered probable that localization from diagonal pairs of loudspeakers would be projected only onto the horizontal plane and not the vertical plane. Careful psychoacoustic testing of vertical and diagonal localization was to produce some very interesting results. But first of all a test recording was made in the anechoic chamber in the Acoustics Department of the University of Göteborg. A series of Level Difference and Time Difference signals were generated using a standard horizontal microphone pair (25cm/90° cardioids) – the sound source moving around the microphone pair in the horizontal plane. Afterwards, in the listening tests, the two signals were routed to either just the vertical pair of loudspeakers or to just a diagonal pair of loudspeakers.

NO PRECISE LOCALISATION was experienced in the vertical plane, however very precise localization WAS observed along the diagonal plane. In the first case this confirmed what was expected (and described in AES preprint 8601**[3]**), whereas the second case was a complete surprise. In the second case localization was expected to be projected onto the horizontal plane with no vertical component, but in reality the localization followed the line between the diagonal loudspeakers and was a clear and realistic reproduction of the sound source.

These observations, concerning the localization characteristics of the loudspeaker configuration, completely justify the primary isosceles triangle structure of the experimental 3D Multiformat Microphone Array.

This means that we can expect no reliable sound source localization on loudspeakers that are situated vertically one above the other, but that loudspeakers placed in an isosceles triangle structure around the listener, as shown in Figure 12, will produce reliable virtual localization of sound images. Of course the microphone array structure must be the mirror image of the loudspeaker structure. But this does not mean that the microphones have to be in exactly the same orientation as the loudspeakers, but the general univalent triangular structure must be the same.

The 7 Channel Listening experience.

In a previous VDTonmeistertagung (2000) I made a demonstration of the compatibility of the 7 channel multiformat microphone array (M.A.G.I.C) with 4 or 5 channel systems. In that demonstration I showed how it was possible to change from 4 to 5 to 7 channels without any matrixing, by only muting the channels that were not required. In this demonstration there was little appreciable change in the total surround sound image when switching from 4 to 5 channels. However the one remark that was made, was that there was a increase in bass reproduction with the 7 channel system. This remark has been frequent in the many demonstrations that I have done of this system.

The reason seems to be that below a certain aliasing frequency the reproduction passes from individual loudspeaker spherical wave front propagation to a combined loudspeaker cylindrical propagation response. The actual energy reaching the listener is therefore greater in the cylindrical propagation situation compared with the spherical propagation situation. The aliasing frequency occurs when the half wavelength is equal to the distance between the loudspeakers. If the distance between the loudspeakers is about 1.70 metre then the aliasing frequency is about 100Hz.

In the 3D Multiformat Microphone Array the passage from a 3D eight channel system to a 3D twelve channel system (that is when the C, Lm, Rm and back loudspeakers are added to the eight channel system) the increase in bass response could explain the slight improvement, or at least preference, of listeners for the full 3D twelve channel system.

Asimple solution to this problem would be to change the directivity pattern of the C, Lm, Rm and B microphones from hypocardioid to cardioid thereby decreasing this bass frequency imbalance. The reason being that the cardioids have less bass response than hypocardioids, therefore in the extreme bass response there would be less contribution from the cardioids, compared to the hypocardioids, in cylindrical wave front propagation.

In these listening tests it was found that time alignment of the sources was extremely critical – this time alignment can either be obtained by changing the physical position of the loudspeaker (the distance from the loudspeakers to the listener must be exactly the same) or by electronic delay so that the arrival time at the listener, is exactly the same, for an identical signal from each loudspeaker.

The complete 12 channel array does have some redundancy in that some of the channel components are not necessarily needed. The Hc channel is reproduced with loudspeakers that are positioned immediately above the C channel loudspeaker. This also applies to the Hl channel with respect to the Lm channel, the Hr channel with respect to the Rm channel, and the Hb channel with respect to the B channel. It was found that the system performed well as a 3D reproduction format when only eight channels were reproduced – L, R, Ls, Rs, Hc, Hl, Hr, and Hb. That is the extremities of each isosceles triangle in the structure. Only a slight improvement or preference was observed when the C, Lm, Rm and B channels were reintroduced, but this is probably due to this artificial increase in bass response when all hypocardioid microphones are used in the 1st layer array.

This means that perfectly satisfactory reproduction of the 3D soundfield is possible with only 8 channels (or even 7 channels if we use the 7.0 Blu-ray format – but of course no back channel is available in this case).

If the listening tests are carried out in a 22.2 loudspeaker configuration then only certain loudspeakers should be used. The correspondence between the 12 channel reproduction system and specific loudspeakers in the 22.2 configuration is shown in Figure 13. Only the loudspeakers **highlighted in blue** are considered as compatible with the twelve channels of the 3D Multiformat Microphone Array.

Channel Number	Channel Name	Label	Azimuth	Elevation	Distance [m]
1	**Front Left**	**FL**	**45**	**0**	**1.55**
2	**Front Right**	**FR**	**-45**	**0**	**1.55**
3	**Front Centre**	**FC**	**0**	**0**	**1.55**
4	**Low Frequency Effect 1**	**LFE1**	**30**	**0**	**1.55**
5	**Back Left**	**BL**	**135**	**0**	**1.55**
6	**Back Right**	**BR**	**-135**	**0**	**1.55**
7	Front Left centre	FLc	30	0	1.55
8	Front Right centre	FRc	-30	0	1.55
9	**Back Centre**	**BC**	**180**	**0**	**1.55**
10	**Low Frequency Effect 2**	**LFE2**	**-30**	**0**	**1.55**
11	**Side Left**	**SiL**	**90**	**0**	**1.55**
12	**Side Right**	**SiR**	**-90**	**0**	**1.55**
13	Top Front Left	TpFL	45	45	1.55
14	Top Front Right	TpFR	-45	45	1.55
15	**Top Front Centre**	**TpFC**	**0**	**45**	**1.55**
16	Top Centre	TpC	0	90	1.55
17	Top Back Left	TpBL	135	45	1.55
18	Top Back Right	TpBR	-135	45	1.55
19	**Top Side Left**	**TpSiL**	**90**	**45**	**1.55**
20	**Top Side Right**	**TpSiR**	**-90**	**45**	**1.55**
21	**Top Back Centre**	**TpBC**	**180**	**45**	**1.55**
22	Bottom Front Centre	BtFC	0	-39	1.55
23	Bottom Front Left	BtFL	45	-39	1.55
24	Bottom Front Right	BtFR	-45	-39	1.55

FIGURE 13 – TABLE OF EQUIVALENTS BETWEEN THE 22.2 CONFIGURATION AND THE 12.0 MULTIFORMAT CONFIGURATION

The GOArt Listening Tests.
First of all it must be said that the recordings in the four churches selected were done WITHOUT the presence of the GOArt listening panel. They therefore had no knowledge of the techniques used to create the recordings. They were however asked to attend live listening session in the churches afterwards, listening to the same extracts as had been recorded. No recording equipment was present for these later sessions. They were asked to fill out a questionnaire concerning their appreciation of the sound of the organ and the church environment.

The listening panel was then invited to attend, individually, listening tests at the Applied Acoustics Dept. of Göteborg University where they were able to hear the original recordings. They were again asked to fill out a questionnaire concerning their perception of the sound of the organ and its acoustic environment.

In each church, three specific positions of the microphone array were recorded, with small variations in position according to the organ structure and acoustics of each church. There was a close correspondence between the live listening impressions and the 3D reproduction listening tests. One person in particular on the listening panel was able to locate the three positions of the recording array to within 50cm, without any previous knowledge of the actual positions used. This was indeed a remarkable performance, and confirmed that this person had exceptional 'golden ears'! His appreciation of the sound image was based also on an exceptional knowledge of the timbre of the organ (of the Ôrgate Nya Kirka), coupled with a considerable experience of the acoustics of the church – he was in fact responsible for the manufacture, tuning and voicing of the pipes in the organ! However this remarkable identification of the recording positions would not have been possible if the microphone array recording and reproduction system itself did not produce an exceptionally realistic reproduction of the organ and its surrounding acoustic environment.

The reconstruction of the North German Baroque Organ for the Orgryte Nya kyrka is described in a book entitled 'Tracing the Organ Master's Secrets' published by GOArt Publications, Göteborg University – ISBN 140468825.

Only one criticism could be made of the listening system in that the loudspeakers had a bass roll-off frequency at about 38Hz, so the real bass frequency reproduction of the organ was not totally satisfactory. lFE channels were tried but proved unacceptable.

However the mechanical assembling of the recording system was considered too long (from 2 to 3 hours) – further work has to be done in simplifying the assembly procedure. The other limiting factor was the considerable expense of a complete twelve channel microphone array and support system. Although this was considered a major difficulty for a budget that was to be allocated for this project, it was recognized that this was certainly necessary if a very high quality recording was to be achieved, both with respect to the timbre restitution and also the 3D spacial reproduction. It was also considered that the specification of the listening room would have to be upgraded if a permanent listening room was to be installed.

References

[1] **AES Preprint 7057** – 122nd AES Convention in Vienna - Magic Arrays, Multichannel Microphone Array Design applied to Multi-format Compatibility.by Michael Williams

[2] **AES Preprint 7480 –** 124th AES Convention in Amsterdam - Migration of 5.0 Multichannel Microphone Array Design to Higher Order MMAD (6.0, 7.0 & 8.0) with or without the Inter-format Compatibility Criteria by Michael Williams

[3] **AES Preprint 8601 –** 131st AES Convention in Budapest – Microphone Array Design for Localization with Elevation Cues by Michael Williams

[4] **AES Preprint 3157** – 91st AES Convention in New York – Microphone Arrays for Natural Multiphony – by Michael Williams

[5] **AES Preprint 6373** – 118th AES Convention in Barcelona - The Whys and Wherefores of Microphone Array Crosstalk in Multichannel Microphone Array Design by Michael Williams