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Circular microphone array for discrete multichannel audio recording.

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ABSTRACT

Traditional stereo microphone pair techniques for natural recording are quite capable for 2 channel stereo reproduction. However, for multichannel reproduction systems like 5.1, 7.1, ambisonics and Wave Field Synthesis compromises in terms of coverage, source localization and channel separation are unavoidable. The main reason for this is that microphones currently used only have low order directivity patterns (omni, figure-of eight, cardioid or hypercardioid) that cannot provide sufficient angular resolution to avoid unwanted cross talk between the recording channels. In this paper a discrete coincident 12 channel microphone is proposed in order to solve these problems. This microphone consists of a circular array with a radius of 1 meter using 288 microphone capsules whose output signals are combined into 24 channels using simple analog electronics. These 24 channels are captured using a multi-track computer interface and post-processed into up to 12 discrete reproduction audio channels.

INTRODUCTION

In previous AES conventions, papers were presented by the first author on the subject of circular microphone arrays. In preprint 5337 of the 110th AES convention in Amsterdam, which was published in a slightly modified form in the October 2002 issue of the AES journal, an elegant method was described to decompose the sound field measured on a circular array in terms of incoming and outgoing cylindrical harmonic solutions of the 2D wave equation [1,2]. These cylindrical harmonics were shown to be very closely related to the plane wave decomposition, a description of the sound field that proved very useful for auralization purposes. In preprint 5579 presented at the 112th AES convention in Munich the theory for circular arrays was extended to focusing and spatial filtering to control aperture [3]. Furthermore it was shown that it is possible to simulate virtual microphones with any desired directivity properties in post-processing which is not possible with currently available single microphones.

Until now the circular array was only used for impulse response measurements. A single microphone mounted by means of a rod on a slowly rotating turntable was sufficient for this purpose. In this paper, however, the focus is on building a full array, which can also be used for live audio recording purposes. A number of virtual microphones is created on the array having well controlled high order directivity patterns to deliver a large number of discrete output channels without any unnecessary crosstalk between them.

A prototype array is designed to deliver up to 12 discrete coincident channels of audio with good channel separation down to 100 Hz and a spatial aliasing frequency above 15 kHz. To achieve this a circle with a radius of 1 meter and 288 small cardioid microphones is used. The high number of microphones is not a big problem in this case; various cheap and reasonable quality capsules are available on the market. The problem, however, is the number of acquisition channels. In this paper a solution for this problem is given, reducing the number of required acquisition channels from 288 to only 24 for a 12 channel microphone array.

WAVE DECOMPOSITION

If on the circular array both pressure and normal particle velocity are recorded, the approach is to decompose the recorded sound field into cylindrical harmonics:

$$\mathcal{M}^{(1)}(k_\theta,\omega) = \frac{H'^{(2)}_{k_\theta}(kR)P(k_\theta,\omega) - H^{(2)}_{k_\theta}(kR)j\rho c V_n(k_\theta,\omega)}{H^{(1)}_{k_\theta}(kR)H'^{(2)}_{k_\theta}(kR) - H^{(2)}_{k_\theta}(kR)H'^{(1)}_{k_\theta}(kR)}$$

$$\mathcal{M}^{(2)}(k_\theta,\omega) = \frac{H'^{(1)}_{k_\theta}(kR)P(k_\theta,\omega) - H^{(1)}_{k_\theta}(kR)j\rho c V_n(k_\theta,\omega)}{H^{(2)}_{k_\theta}(kR)H'^{(1)}_{k_\theta}(kR) - H^{(1)}_{k_\theta}(kR)H'^{(2)}_{k_\theta}(kR)} \tag{1}$$

In these equations $\mathcal{M}^{(1)}$ and $\mathcal{M}^{(2)}$ are the incoming and outgoing cylindrical harmonic decompositions of the sound field, $P(k_\theta,\omega)$ and $V_n(k_\theta,\omega)$ are the spatial and temporal Fourier transforms of the pressure $p(\theta,t)$ and normal velocity $v_n(\theta,t)$ measured on a circular array with radius R, $k=\omega/c$ is the temporal wave number, $k_\theta = \ldots,-1,0,1,\ldots$ is the angular wave number and $H^{(1,2)}_{k_\theta}$ are the Hankel functions of the first and second kind [4]. See [3] for more details on the cylindrical harmonic decomposition. Since in normal recording situations all sound waves are coming from the exterior of the circle, only the incoming part $\mathcal{M}^{(1)}$ needs to be considered. The plane wave decomposition can easily be calculated from $\mathcal{M}^{(1)}$ using

$$s_\infty(\theta,\omega) = \frac{1}{2\pi}\sum_{k_\theta} j^{(1-k_\theta)}\mathcal{M}^{(1)}(k_\theta,\omega)e^{jk_\theta\theta}, \tag{2}$$

which is up to a rotation factor $j^{(1-k_\theta)}$ equal to the inverse Fourier transform of $\mathcal{M}^{(1)}$ [1,2]. Instead of using both pressure and normal particle velocity, it is sufficient to record the sound field on the array using outward pointing cardioid microphones. Using only omnidirectional or only figure-of-eight microphones is not sufficient though; not all cylindrical harmonics can be obtained from such a recording. In equation (1) it is found that the cylindrical harmonics are proportional to the 2-dimensional Fourier transform components of the recorded sound field. In figure 1 the amplitudes for the 2D Fourier transforms of a plane wave recorded on a circular array using omnis, figure-of-eights and cardioid are shown. The omni and figure-of-eight recordings from figure 1(a) and (b) both have zeros in their Fourier transform components and therefore cannot be used separately for a calculation of the non-zero cylindrical harmonic components of the plane wave shown in figure 2. However, since the zero components for the omni and the figure-of-eight microphone arrays occur at different frequency components, they can be used together for a proper cylindrical harmonic decomposition as is done in equation (1). For the cardioid case the 2D Fourier transform components are already free from zeros and can easily be used for this purpose as well. This way the number of acquisition channels can be reduced by a factor 2. Note however that it is not possible to separate between incoming and outgoing sound waves when using only one set of outward pointing cardioids. Also equation (1) and (2) cannot be used directly in this case, but the appropriate plane wave decomposition operator can easily be calculated numerically by creating an inverse for the 2D Fourier transform of a plane Dirac wave simulated on a cardioid array.

VIRTUAL MICROPHONES

In this paper the aim is to deliver 12 discrete coincident channels of audio. For that purpose 12 virtual microphones with suitable directivity characteristics can be created by combining the appropriate cylindrical harmonic components with appropriate weighting factors. The only limitation in creating vir-

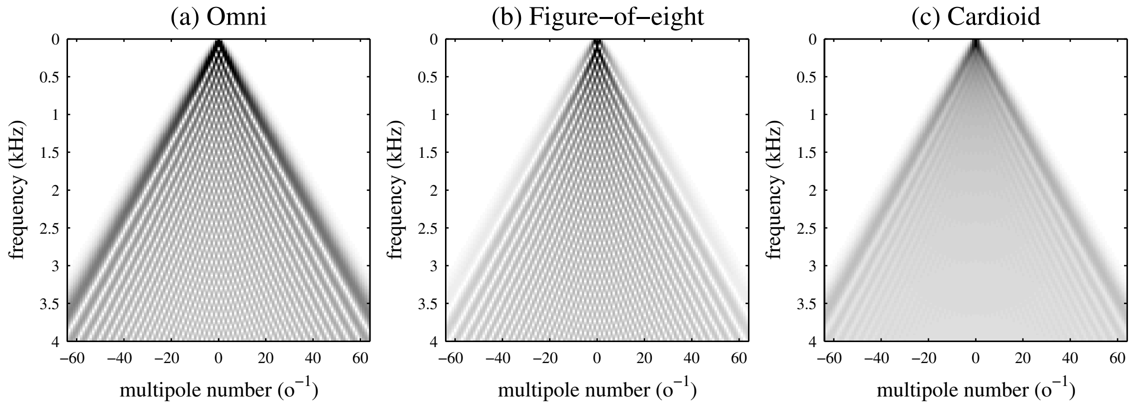

Figure 1: 2D Fourier transform of a simulated circular plane wave recording, using omni, figure-of-eight and cardioid microphones

tual microphones is that the directivity pattern of the virtual microphone cannot contain cylindrical harmonics with spatial frequencies k_θ beyond the angular resolution available for the given array size. Notice that the maximal available angular frequency component is proportional to the radius (aperture) of the circular array and the temporal frequency as can be seen in figure 1. In practice this usually means that for high temporal frequencies the created virtual microphones will match the desired directivity pattern perfectly, but below a certain temporal frequency the highest angular frequency components in the desired directivity pattern are missing. The value of this temporal cutoff frequency depends on the radius of the circular array and the sharpness of the desired directivity pattern. The sharper it is, the higher the maximal required angular frequency are and the higher the temporal cutoff frequency will be.

Virtual microphones can be created easily using the plane wave decomposition. The plane wave decomposition can be considered as a set virtual microphone responses, which have the maximal directivity sharpness that can possibly be obtained for the given microphone array aperture. Suppose $s_\infty(\theta, \omega)$ is the plane wave decomposition of a sound field and $d(\theta, \omega)$ is the possibly frequency dependent (in this paper only frequency independent directivity patterns are used though) desired directivity pattern of a virtual microphone. Then the output of the virtual microphone can be obtained from the plane wave decomposition using

$$O(w) = \sum_\theta d(\theta, \omega) \cdot s_\infty(\theta, \omega) \tag{3}$$

The actual obtained directivity pattern $d_c(\theta, \omega)$ of the virtual microphone is given by

$$D_c(k_\theta, \omega) = D(k_\theta, \omega) \cdot S_{\text{plane}}(k_\theta, \omega), \tag{4}$$

where $D_c(k_\theta, \omega)$ and $D(k_\theta, \omega)$ are the angular Fourier transform of $d_c(\theta, \omega)$ and $d(\theta, \omega)$, respectively, and $S_{\text{plane}}(k_\theta, \omega)$ is the 2D Fourier transform of the plane wave decomposition of a plane Dirac wave with angle of incidence equal to zero and arriving at the center of the circle at $t = 0$, which is limited in angular resolution by the radius of the circular array. $S_{\text{plane}}(k_\theta, \omega)$ for a circle with a 1 meter radius is shown in figure 2. The actual

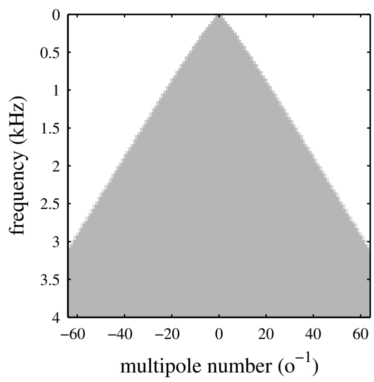

Figure 2: Cylindrical harmonics decomposition for a circular array with a 1 meter radius.

directivity pattern is thus a band limited version of the desired directivity pattern in terms of the angle. See figure 3(a) for an example of one of the desired directivity patterns that can be used for obtaining 12 discrete audio channels. The realized directivity pattern using a 288 microphone circular array with a radius of 1 meter is shown in figure 3(b). Notice that apart from the angular bandwidth limitation starts taking place at 250 Hz, also spatial aliasing artifacts occur above 15 kHz. These are determined by the distance between the microphones and can be avoided by using even more closely spaced microphones.

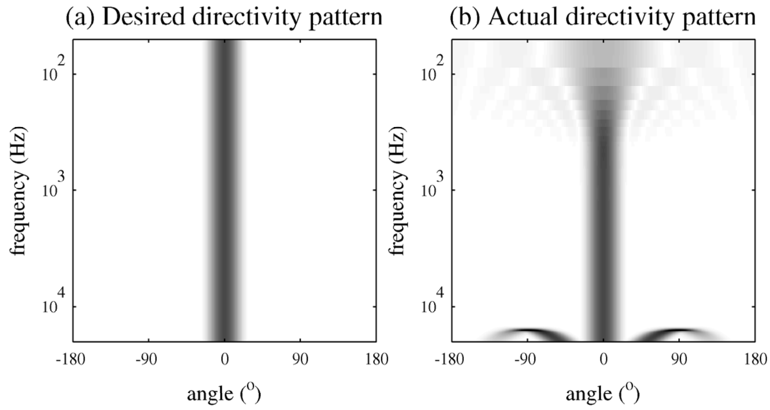

Figure 3: Directivity pattern of a virtual microphone created using a 1 meter radius array.

ANALOG PREPROCESSING

The number of microphones used in the circular recording array is not a big problem as mentioned before; various cheap and reasonable quality capsules are available on the market. The problem however, which is going to be solved next, is the number of acquisition channels. A wave decomposition has to be calculated from all the 288 microphone signals, which must be done in the digital domain using 2D Fast Fourier transform algorithms. This means that 288 microphones need to be digitally acquired and processed in real time, something which isn't feasible in practice or at least very expensive.

In order to understand how this problem can be solved easily by drastically reducing the number of acquisition channels, a closer look at the processing scheme is required. See figure 4. In figure 4(a) a circular recording of a plane wave is shown. The 2D Fourier transform for this recorded wave is then calculated and the amplitudes of these complex components are shown in figure 4(b). From this 2D Fourier transform the cylindrical harmonic components in figure 4(c) are calculated. By calculating the 2D inverse Fourier transform of this field the 288 channel plane wave decomposition from figure 4(d) can be obtained using equation (2). The last step is to combine these 288 plane wave virtual microphone signals using equation (3) to obtain a lower number of virtual microphones with the desired directivity patterns for the multichannel sound format in figure 4 (e). As can be seen in figure 4(f), which is the 2D Fourier transform of the data from figure 4(e), these final outputs only contain low cylindrical harmonic components. This means that actually a large number of angular frequency components have been acquired and processed that are not necessary at all for the final output.

This throwing away of high order angular frequency components, which was done in the last panning step of the processing, can just as well been done as a first step, since the order in which linear operations (multiplications) in the k_θ, ω take place, does not effect the end result. In figure 5 the wave field from figure 5(a) is first combined into a lower number of channels in figure 5(b) and then a 2D Fourier transform is calculated in figure 5(c), containing only the low order cylindrical harmonics in figure 5(d) needed for the final virtual microphones in figure 5(e). This approach is much more efficient in terms of processing power than the approach from figure 4 since only a limited number of 2D Fourier components need to be considered. When done correctly, it delivers the same output as the approach from figure 4 Furthermore, the panning done between figure 5(a) and (b) does not necessarily need to be done in the digital domain after acquiring the full 288 input channels; since it is a simple operation that can easily be done in the analog domain, a much lower number of acquisition channels is required. Note that this second processing approach implies that the shapes of the directivity patterns are equal for the different virtual microphones; the virtual microphones are rotated copies of each other.

If discrete microphones are used in the microphone array, the panning can be done by combining the outputs for the individual microphones using resistor ladder networks as shown in figure 6(a). The value of the resistor determines the weight of that particular microphone on the summed output.

Instead of using small discrete microphone capsules with resistor ladder networks, it is also possible to use microphone capsules that are more extended in space such that the integration of the sound field over a larger area in space is done by the microphone itself. The weighting is in this case done by varying the height of the microphone capsules. See figure 6(b) for a simple schematic example of such a microphone capsule with a triangular weighting window.

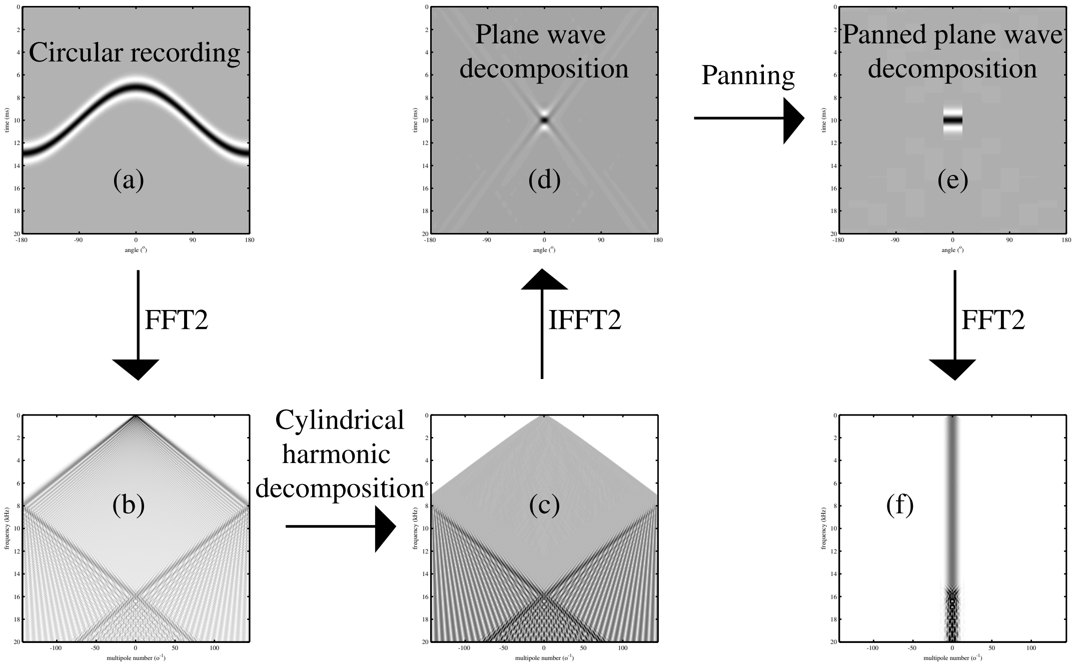

Figure 4: Circular array processing for 12 virtual microphones

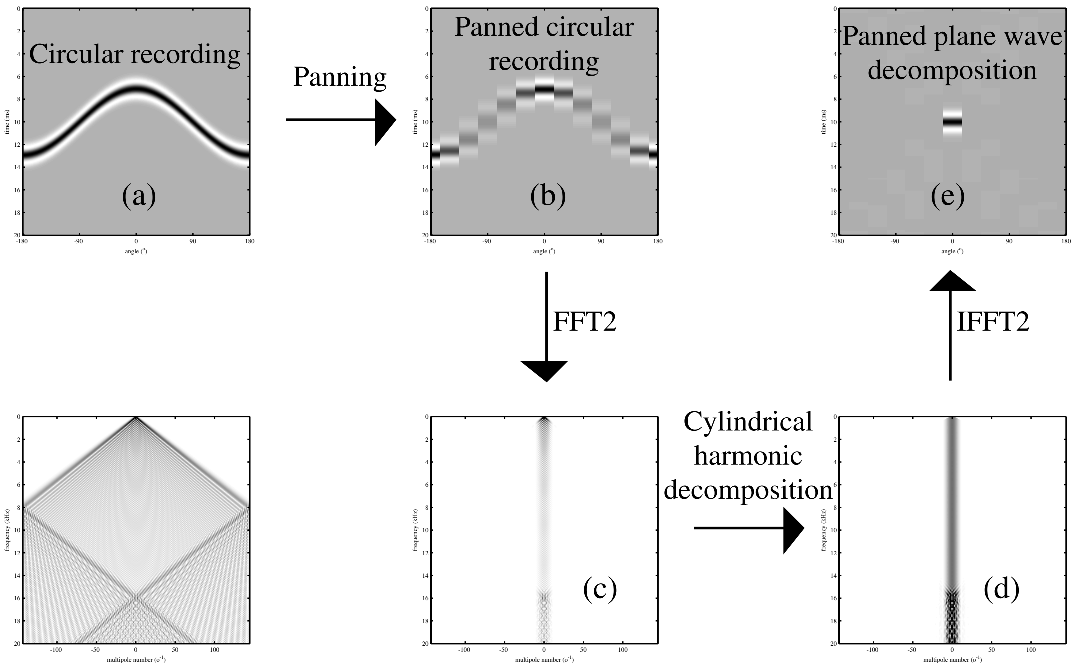

Figure 5: More efficient circular array processing for 12 virtual microphones

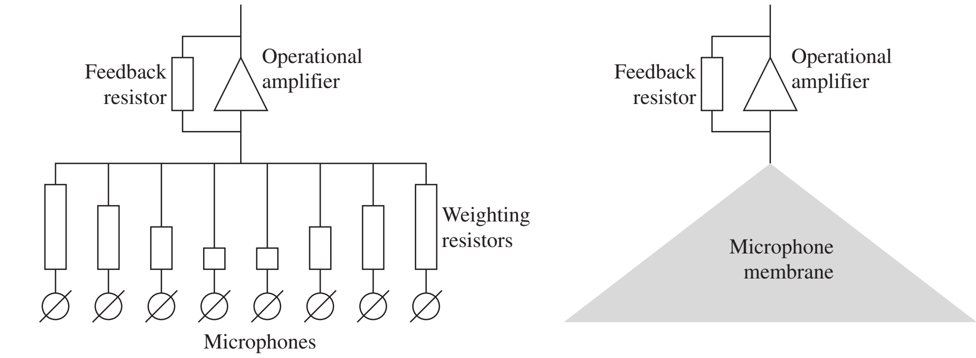

Figure 6: Panning in the analog signal domain (a) using small microphones with weighting resistor ladders, (b) using an extended microphone membrane to apply spatial weighting and integration of the sound field

The shape of the weighting window plays an important role in the performance of such an array. First of all, it determines the precise shape of the directivity patterns for the virtual microphones. Furthermore, it determines the angular frequency content of the virtual directivity patterns, which determines the number of required acquisition channels and the angular aliasing performance, as will be shown next. In figure 7 a number of weighting windows are shown: triangular, Hanning, Hamming, and Chebyshev. The sizes of the windows are chosen to obtain 12 uniformly distributed audio channels with full 360 degree coverage. Only the directivity window for the 0 degrees channel is shown in this figure. The windows for the other angles can be obtained by shifting the integration window over angles that are multiples of 30 degrees.

In figure 7 the angular fourier transforms of these windows are shown. All these windows contain approximately 24 nonzero frequency components. This implies that at least 24 samples in the angle (acquisition channels) are required for a sufficient aliasing-free wave field decomposition and virtual microphone output creation, which is twice the amount of desired output channels. If only 12 acquisition channels were used, severe angular aliasing artifacts would occur in the wave decomposition processing. Even with 24 samples the field is not entirely free from aliasing due to some frequency-lobing as can be seen from figure 7.

The effect of undersampling and frequency lobing described above can be easily demonstrated by processing a simulated plane wave on the circular array using the processing scheme from figure 5. The performance of the full 288 channel processing using the processing scheme from figure 4 is shown in figure 8(a) and 9(a) for angles of incidence of 0 and 15 degrees, respectively. This should be considered the best performance possible with the given array size. Cross talk between the 12 channels is extremely low, except for frequencies below 250 Hz, which, as discussed before, is due to the limited radius of the circular array The case of using the scheme from figure 5 using 24 triangular angles shows a relatively poor performance in figure 8(b) and figure9(b). This is due to the fact that the triangular window has quite strong side lobes in the frequency domain which cause aliasing artifacts. In figure 8(c–g) and 9(c–g) 24 uniformly distributed Hanning, Hamming, Gaussian and the product of a Gaussian and a cosine window are used, some of which are giving a much better performance due to smaller side lobes. Only the odd 12 output channels are shown in these figures since only those are used for a discrete 12 channel reproduction system. See figure 10 for a sketch of the directivity patterns of all 12 used virtual microphones. Note that the shapes are only valid between 150 and 15000 Hz and that they will be distorted by aliasing caused by the frequency lobing.

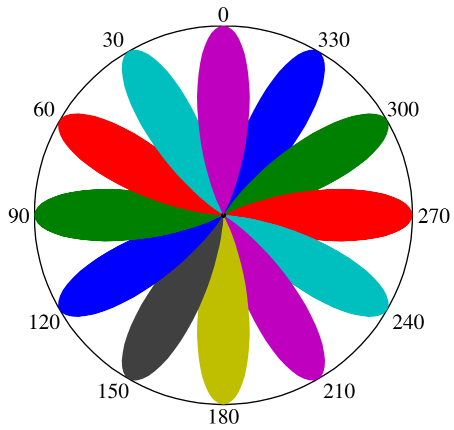

Figure 10: Virtual microphone directivity patterns

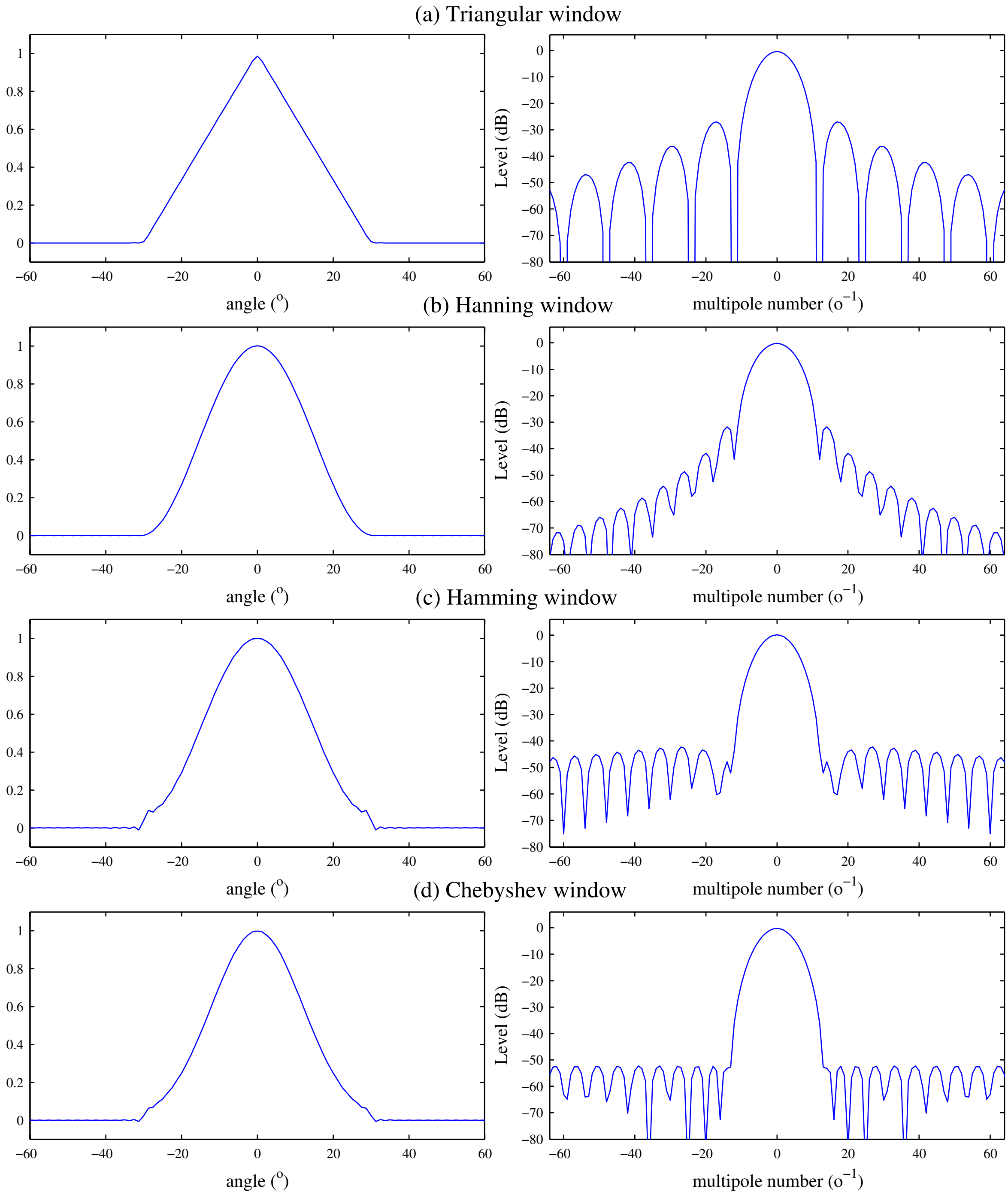

Figure 7: Panning windows and their frequency responses

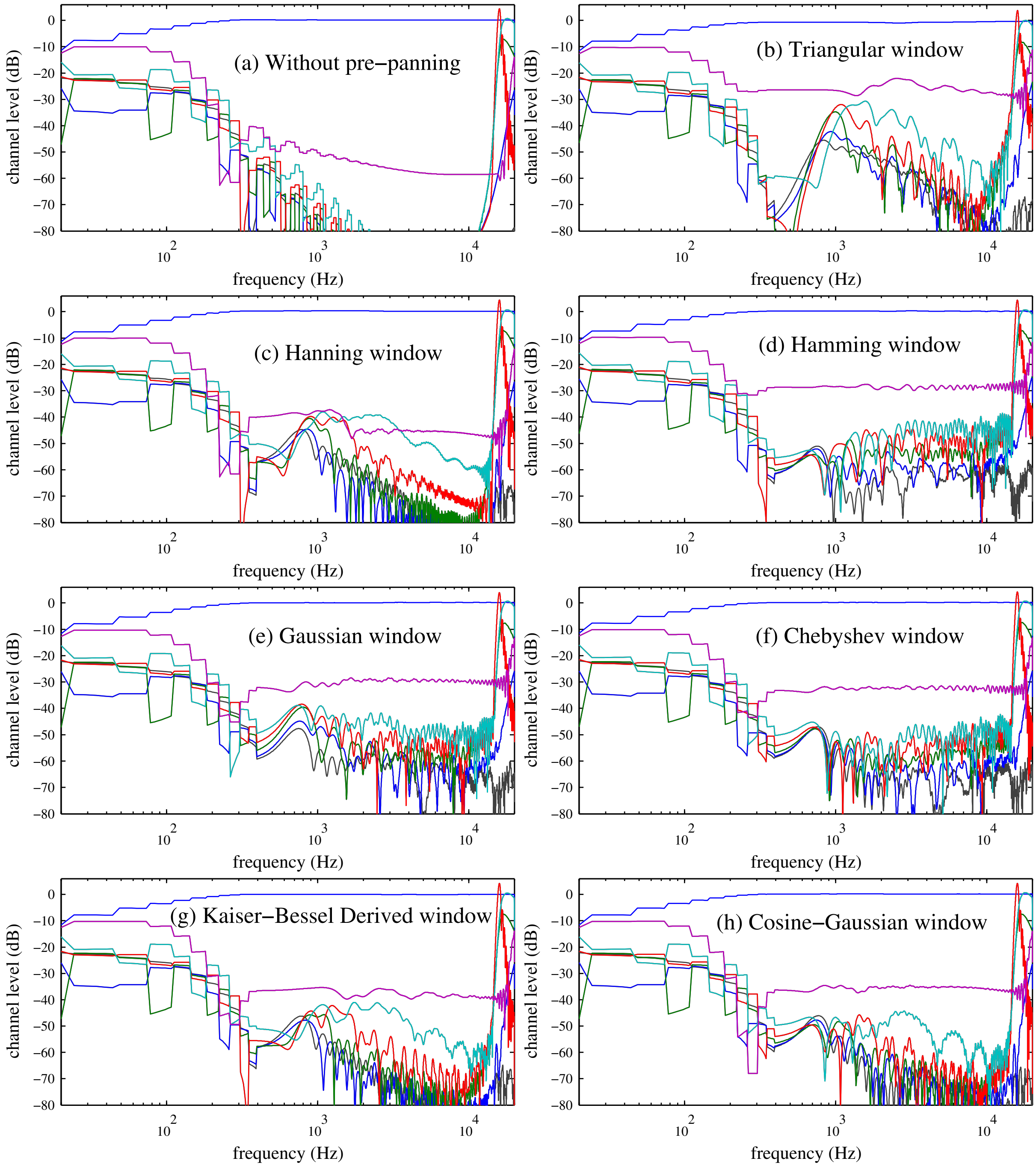

Figure 8: Aliasing performance of different panning windows recording a plane wave with 0 degree incidence, ideally being reproduced by 1 channel only

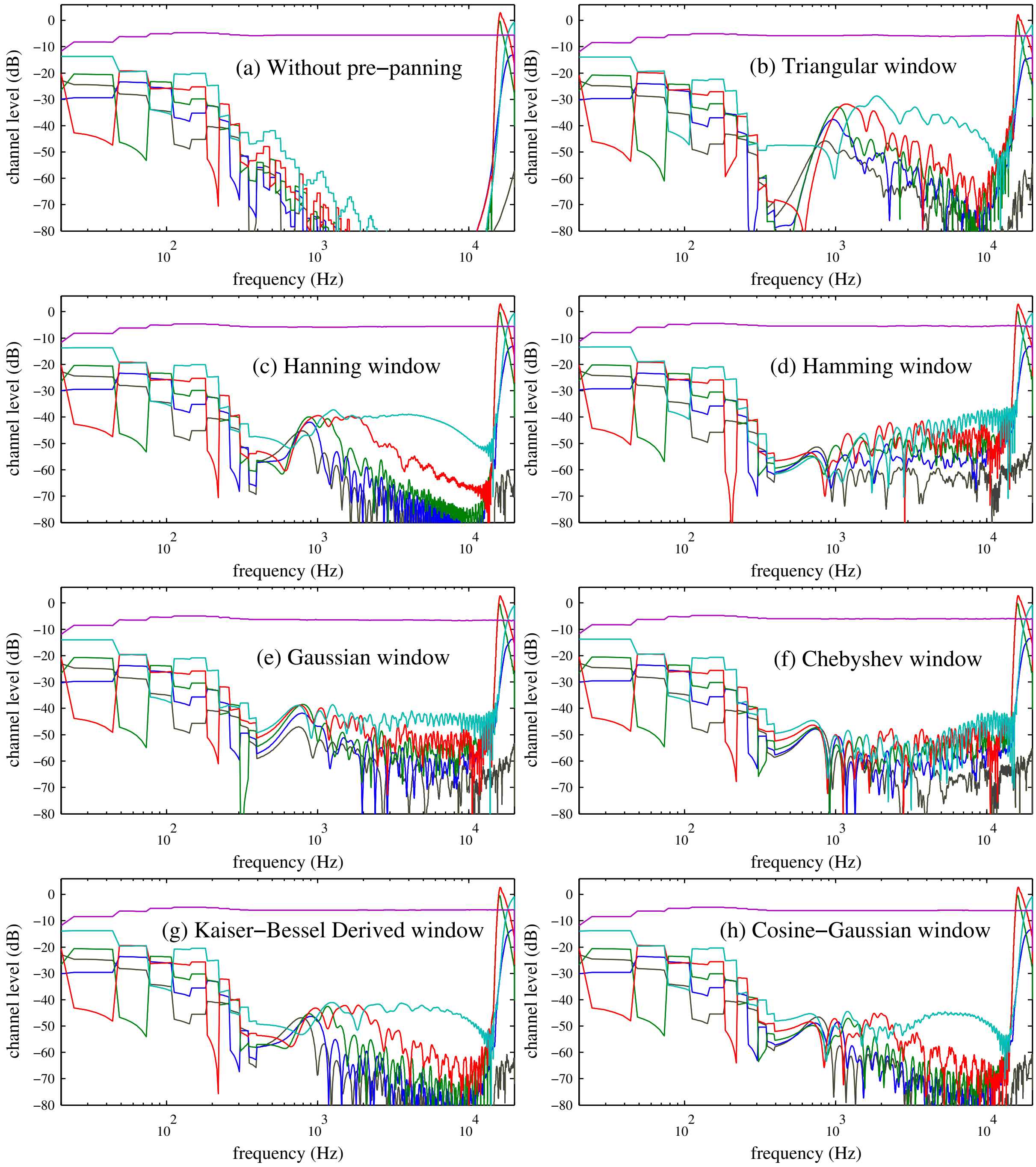

Figure 9: Aliasing performance of different panning windows recording a plane wave with 15 degree incidence, ideally being panned equally between 2 channels only

Using all 24 output channels directly for a discrete 24 channel reproduction system is probably not a good idea, since the even channels have a strong overlap in angle with the odd ones implying unnecessary crosstalk between them that results in comb filter effects between the channels during reproduction. However, as will be shown later, it is possible to create 24 better virtual microphones with less crosstalk from the original ones using an output matrix.

REAL-TIME WAVE DECOMPOSITION PROCESSING

Until now only examples where shown using simple, short and easy to process wave fields, like a simulated Dirac plane wave recording. In practice it is necessary to decompose a continuous stream of 24 channels of audio in real time. This can be done by splitting the audio input stream into time blocks, applying a FFT2-based convolution filter in the k_θ, ω-domain, transforming the data back to θ, ω-domain and using the overlap-add approach to combine the blocks into an output audio stream. This type of processing is certainly feasible in real time on a single pc nowadays. The FFT2-based convolution filter can be designed to account for both wave decomposition and compensation for the non-ideal actual performance of the cardioid microphones in terms of frequency response and frequency dependent directivity pattern.

OUTPUT MATRIX

Having a discrete 12 channel microphone array is nice, however it will not always be flexible and desirable to use the 12 fixed virtual microphones from figure 10 directly. This would imply a different array design for each different reproduction system. For example, the 12 discrete output channels on itself cannot be used directly for a 5.1 or 7.1 reproduction. However, by choosing the 24 panned acquisition channels one is not limited to the 12 virtual microphones from figure 10 but can in fact use any linear combination of all 24 virtual microphones to create new virtual microphones. This can be implemented by using a simple output matrix. The 24 original channels are routed through a matrix to obtain any desired number of output channels. Suppose that $i_1 \ldots i_{24}$ are the panned and decomposed output channels of the array and thus the input channels for the matrix and suppose $o_1 \ldots o_N$ are the N desired output channels. Then one can write

$$\begin{pmatrix} o_1 \\ \vdots \\ o_N \end{pmatrix} = \begin{pmatrix} M_{1,1} & M_{1,2} & \cdots & M_{1,24} \\ M_{2,1} & M_{2,2} & \cdots & M_{2,24} \\ \vdots & \ddots & \ddots & \vdots \\ M_{N,1} & M_{N,2} & \cdots & M_{N,24} \end{pmatrix} \begin{pmatrix} i_1 \\ \vdots \\ i_{24} \end{pmatrix} \quad (5)$$

The coefficients $M_{n,m}$ of the matrix determine the directivity patterns for the N virtual microphones. In this case a frequency independent virtual microphone is considered. If one wants to create a frequency dependent directivity pattern, the output matrix coefficients should be functions of the temporal frequency ω instead of being constants. As a first trivial example for an output matrix one could consider the full 12 channel discrete output system. The matix for this example is shown as a colormap in figure 11 . In this case the matrix is very simple: only

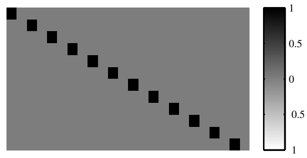

Figure 11: Matrix for 12 channel output

the 12 odd input channels are used and sent directly to the 12 outputs.

Suppose now as a second, more interesting example 5 discrete virtual microphones compatible with 5.1 reproduction angles and covering the full 360 degrees angle. If smooth transition between the left, right and surround channels is desired, the matrix could look like figure 12. This matrix was obtained

Figure 12: Matrix for 5.1 output using gradual panning between all 5 channels

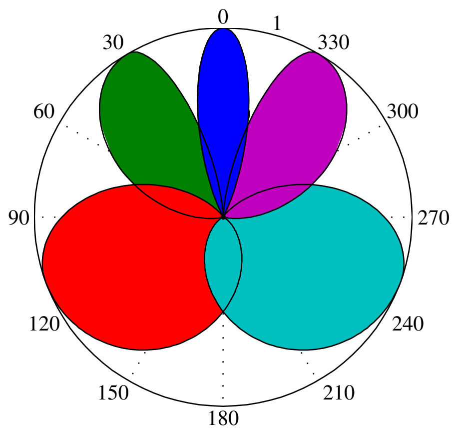

Figure 13: Desired and actual virtual microphone directivity patterns for 5.1 output using gradual panning between all 5 channels. The black lines are the desired patterns and the solid colors are the actual patterns

by creating a band-limited (24 samples long) inversion filter for the used Hanning pre-panning window from figure 7(b) in the angular frequency domain and convolving this with the 24 sample band-limited version of the desired 5.1 directivity patterns, using multiplication in the angular Fourier domain. The directivity patterns for the desired and actually created virtual microphones resulting from this matrix are shown in figure 13

If one desires a much sharper transition between the channels resulting in less overlap and correlation between the front and the rear channels, the matrix from figure 14 can be used. The

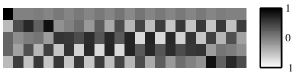

Figure 14: Matrix for 5.1 output using sharp transition between front and rear channels

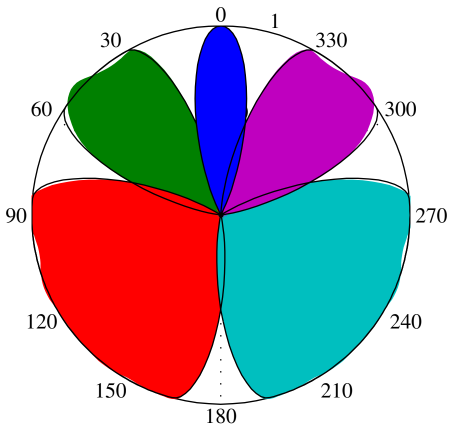

Figure 15: Desired and actual virtual microphone directivity patterns for 5.1 output using sharp transition between front and rear channels. The black lines are the desired patterns and the solid colors are the actual patterns.

directivity patterns of the resulting virtual microphones in this are shown in figure 15. Notice that the obtained virtual microphones are not perfectly equal to the desired ones, since the desired ones contain a small amount of high frequency components that cannot be resolved.

These were only a few simple examples. Notice that any directivity pattern can be created in this way as long as it is not sharper than the directivity patterns of the original 24 microphone output channels. Clearly it is not possible to achieve really sharper directivity patterns, since for that purpose higher order cylindrical harmonics, that have been filtered out by the panning step in figure 5 are required. It is possible however to create slightly more directive beams. For example, the matrix from figure16 delivers 24 discrete audio channels, but at the price of more sidelobes (crosstalk) and more noise sensitivity. Their directivity patterns are shown in figure 17

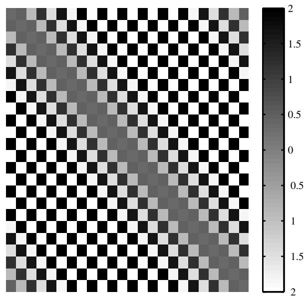

Figure 16: Matrix for 24 discrete output channels

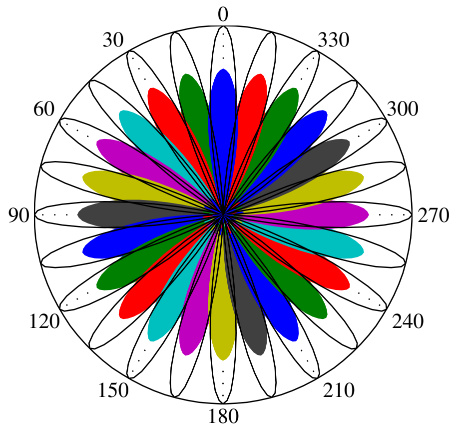

Figure 17: Desired and actual virtual microphone directivity patterns 24 discrete output channels.

The output matrix can be generalized even further by using complex matrix coefficients instead of only real ones. A complex coefficient implies, apart from the amplitude gain and sign, also a phase rotation in the audio signal. This could particularly be useful as a decorrelation/diffusion filter for the surround channels in the previous 5.1 examples, in case they are only used for the recording of ambiance (for example room effect and applause).

The matrixing is a process that can be applied and optimized to the taste of the sound engineer afterwards, if during the live recording the full 24 channels were stored. This approach is comparable B-format recording, in which case the 4 B-format signals can also be combined (matrixed) into the final outputs afterwards by using a B-format processor [5].

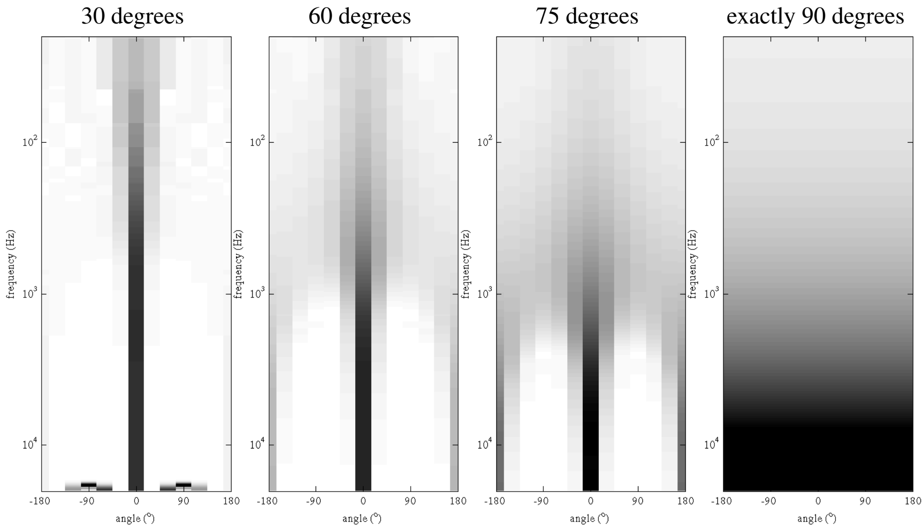

Figure 18: Frequency response of the 12 discrete channels for different elevation angles

ELEVATION ANGLES

Elevated sources cannot be expected to be properly imaged by a circular array. A spherical array, being quite unfeasible in practice currently, should ideally be used for that purpose. The use of such an array can however only be justified if the reproduction system is also capable of reproduction elevated sources, which is nog the case for most reproduction systems, including WFS. It will be shown that although the recording performance for elevated sources is not ideal, this should not be a major problem in practice. In figure 18 the content of the 12 output channels of a plane Dirac wave with various elevation angles simulated on the array system is shown. As can be concluded from this figure, an elevated source will have contributions from two angles, namely the angle of the real source and the opposite angle. An elevated source in the front will thus be reproduced as a source in front and a source in the back. Furthermore, as can also be seen from figure 18, the angular resolution for an elevated source is more limited, which affects the low temporal frequencies. The energy ratio between front and back is determined by the elevation angle. For extreme elevation angles within 1-2 degrees from the vertical axis, all output channels at all angles are affected and quite a strong high frequency amplification occurs. However, since this is only a very small fraction of the full 4π space angle where normally no direct sound sources or early reflection mirror images are present, it can be expected not to have a strong effect on the overall array performance in real-life recording situations.

NOISE PERFORMANCE OF A CIRCULAR ARRAY

Next noise sensitivity and the effect of variations between individual microphone capsules on the overall array performance are investigated. In figure 19 the noise spectrum of an individual cardioid microphone is shown. The microphones used produce a pink noise signal. The total theoretical noise spectrum for the array including processing and 12 channel reproduction at the sweet spot is shown in figure 19. The noise level is calibrated using a 1 kHz plane wave in such a way that the sound pressure level of the wave in the center of the microphone array is equal to the sound pressure level of reproduced 1 kHz tone at the sweet spot of the 12 channel reproduction system. Figure 19 shows that the spectrum of the noise after the processing has become white. The signal to noise ratio for the whole array system improves drastically for low frequencies compared to the single microphone performance, however, for frequencies above 3 kHz the noise performance deteriorates. This results in an overall noise level increase of 1–2dB(A).

Variations between individual microphone capsules, which will mainly be level variations caused by the build in microphone pre-amplifiers and weighting resistors, were also simulated. Fluctuations of 1–2 dB in microphone sensitivities were assumed. Although, as expected, the actual cross talk separation between the 12 channels becomes a bit worse in the case of these variations, the overall array performance of the array is not too sensitive to such variations and 1–2 dB variation is acceptable.

BUILDING A PROTOTYPE ARRAY

Currently a prototype circular array is being build. The array consists of 4 parts of pre-bended aluminium strips that can be jointed together easily to form a circle. Holes are drilled in the

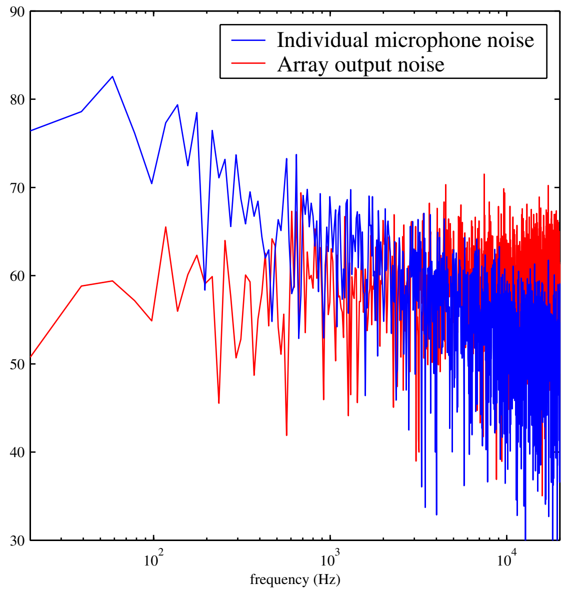

Figure 19: Noise levels from a single microphone and the complete array system

strips in which the microphone capsules are mounted. At the backside of the strips shielded multi-layer print boards are attached that contain connections to the microphones, 4×288 weighting resistors, 24 preamplifiers and some other required electronic components. The boards are constructed as thin as possible and have large holes at the positions of the microphones to avoid acoustic shielding at the backside of the cardioids as much as possible.The microphone capsules used are type EM-135 from Primomic. These relatively cheap microphones should not be expected to deliver hi-end studio quality in terms of noise and high frequency performance. However, the spatial quality of the system should be convincing enough to demonstrate the advantages of the proposed circular microphone array technology. The development and design of a microphone array using only 24 microphone capsules which are more extended in space is currently not taking place but could be a very interesting and relatively cheap high performance solution for the future.

CONCLUSIONS

In this paper a circular array has been proposed that can deliver discrete multichannel audio by simulation virtual microphones in post processing with well controlled and sharp directivity patterns that cannot be achieved with conventional microphone technology. Although such an array requires a large number of microphone capsules to avoid spatial aliasing, the number of recording channels doesn't have to be large at all since part of the processing can already be done in the analog domain using simple resistor ladder networks. Furthermore, if well shaped, in space extended microphones could be designed, such ladder networks would become unnecessary and aliasing could be completely avoided. By using a complex output matrix almost any desired virtual microphone set can be created and auditioned in real time from the original recorded microphone output channels. Since the array under consideration is circular and not spherical, elevation angles cannot be recorded and reproduced properly. In practical applications however this should not cause a big problem, even in case of strong ceiling reflections. The array system also is not very sensitive to noise and fluctuations in the microphone capsules and weighting resistors. A prototype circular array is currently being build.

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