ON THE USE OF IRREGULARLY SPACED LOUDSPEAKER ARRAYS FOR WAVE FIELD SYNTHESIS, POTENTIAL IMPACT ON SPATIAL ALIASING FREQUENCY

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ABSTRACT

Wave Field Synthesis (WFS) is a physical based sound reproduction technique. It relies on linear arrays of regularly spaced omnidirectional loudspeakers. A fundamental limitation of WFS is that the synthesis remains correct only up to a corner frequency referred to as spatial aliasing frequency.
This paper addresses irregular spacing of loudspeaker array for WFS. Adapted driving functions are defined. New formulations of the spatial aliasing frequency are proposed. It is shown that the use of logarithmically spaced loudspeaker arrays can significantly increase the spatial aliasing frequency for non focused virtual sources.

1. INTRODUCTION

Wave Field Synthesis (WFS) is a holophonic technique that relies on the reproduction of physical properties of sound fields in an extended listening area [1]. Its original formulation relies on simplifications of the Rayleigh 1 integral. These approximations reduce the amount of required loudspeakers to a finite number of regularly spaced loudspeakers on a segment. They enable for the synthesis of the target sound field within a large portion of the horizontal plane up to a corner frequency referred to as ”spatial aliasing frequency”.
Irregular or ”random” transducer spacing is currently employed in sound reproduction [2] or sound recording [3]. However, they have not been considered in the context of Wave Field Synthesis. This paper proposes to explore the potential benefits of the use of irregularly spaced arrays for WFS. Two test geometries are considered: ”randomly spaced arrays and ”symmetrical logarithmically” spaced arrays.
First, WFS driving functions for irregularly spaced arrays are proposed and the performance of the test arrays at low frequencies are analyzed. Accurate definitions of the spatial aliasing frequency are then given for finite length arrays considering both regular and irregular spacing of the transducers. Finally, potential improvements on the value of the spatial aliasing frequency compared to regular loudspeaker spacing are studied for various types of irregularly spaced loudspeaker arrays.

2. WAVE FIELD SYNTHESIS FOR IRREGULARLY SPACED LOUDSPEAKER ARRAYS

2.1. Wave Field Synthesis for continuous loudspeaker array

WFS relies on simplifications of the Rayleigh 1 integral [4]. This surface integral defines an infinite plane of “secondary” sources that splits the space into two subspaces (cf. figure 1): a source subspace Ω_Ψ in which “primary” or virtual sources Ψ are, and a reproduction subspace Ω_R where the sound field they radiate is to be synthesized. WFS filters are derived by using the so-called stationary phase approximation as:

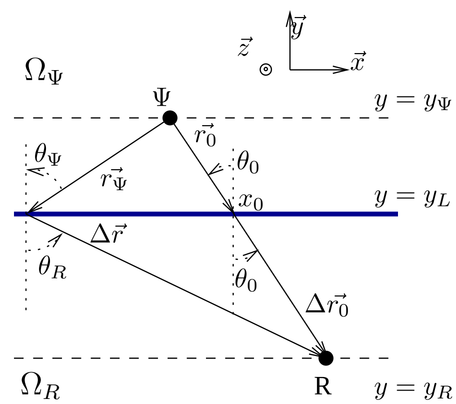

Figure 1: *Synthesis of a virtual source using WFS, source/loudspeakers geometrical description*

$$U(x_L, k) = F(k) G_\Psi(x_L) e^{-j(k\tau_\Psi(x_L)c)}, \quad (1)$$

for a given loudspeaker located at $x = x_L$ on an infinite horizontal line for the synthesis of an omnidirectional source Ψ (cf. figure 1). $F(k)$ is a filter introduced by the stationary phase approximation, which realizes a 3dB per octave attenuation and $\frac{\pi}{4}$ phase shift:

$$F(k) = \sqrt{\frac{k}{2\pi}} e^{j\frac{\pi}{4}}. \quad (2)$$

$\tau_\Psi(x_L)$ is a delay that accounts for natural propagation of the wave front from Ψ:

$$\tau_\Psi(x_L) = \frac{r_\Psi}{c}. \quad (3)$$

$G_\Psi(x_L)$ is a gain factor that stands for the natural attenuation of Ψ and compensates for level inaccuracies due to the natural attenuation characteristics of a linear array:

$$G_\Psi(x_L) = \cos(\theta_\Psi) \sqrt{\frac{|y_L - y_{R_{av}}|}{|y_{R_{av}} - y_\Psi| r_\Psi}}. \quad (4)$$

By definition, the synthesized level is thus only correct at an average listening depth $y_{R_{av}}$.

In a limit case, sources Ψ may also be located in Ω_Ψ by inverting natural propagation delays. Synthesized wave fronts are converging to the target source position and thus propagate from this position in the rest of the reproduction subspace Ω_R. Such sources are therefore referred to as focused sources.

2.2. Wave Field Synthesis for sampled loudspeaker array

We consider a finite length continuous loudspeaker array parallel to the x axis ($z = 0, y = y_L$) such that $x \in [x_A, x_B]$. Its frequency response $H_\Psi(\vec{r_R}, k)$ at position $\vec{r_R}$ for the synthesis of a virtual source Ψ using WFS filters (cf. equation 1) is given by:

$$H_\Psi(\vec{r_R}, k) = \int_{x_A}^{x_B} U(x_L, k) \frac{e^{-jk\Delta r(\vec{r_R}, x_L)}}{4\pi \Delta r(\vec{r_R}, x_L)} dx_L. \quad (5)$$

We define N sampling positions x_n, positions of the loudspeakers, and rewrite the previous equation as:

$$\begin{aligned} H_\Psi(\vec{r_R}, k) = \quad & \textstyle\sum_{n=1}^{N} \int_{x_n - \Delta x_n^-}^{x_n + \Delta x_n^+} U(x_L, k) \times \\ & \tfrac{e^{-jk\Delta r(\vec{r_R}, x_L)}}{4\pi \Delta r(\vec{r_R}, x_L)} dx_L, \end{aligned} \quad (6)$$

where Δx_n^- and Δx_n^+ determine a certain interval around x_n. The sum of these intervals spans the entire line L. Sampled driving functions $U_{samp}(x_n, k)$ may therefore be derived such that:

$$\begin{aligned} & U_{samp}(x_n, k) \frac{e^{-jk\Delta r(\vec{r_R}, x_n)}}{4\pi \Delta r(\vec{r_R}, x_n)} \simeq \\ & \int_{x_n - \Delta x_n^-}^{x_n + \Delta x_n^+} U(x_L, k) \frac{e^{-jk\Delta r(\vec{r_R}, x_L)}}{4\pi \Delta r(\vec{r_R}, x_L)} dx_L \quad \forall n \in [1, N]. \end{aligned} \quad (7)$$

The latter should remain valid at any listening position $\vec{r_R}$ in Ω_R and for a certain frequency range.
We propose here to consider simple sampled WFS filters expressed as:

$$U_{samp}(x_n, k) = F(k) \frac{|x_{n+1} - x_{n-1}|}{2} G_\Psi(x_n) e^{-j(k\tau_\Psi(x_n)c)}. \quad (8)$$

These driving functions account for the local spacing of successive loudspeakers on the array. For regularly sampled arrays, the proposed formula remains coherent with known WFS filters. Additional attenuation factors may be introduced for loudspeakers located at extremities of the loudspeaker array in order to limit diffraction effect due to finite length of the array [4].

3. WAVE FIELD SYNTHESIS AT LOW FREQUENCIES

In this part, performances of irregularly spaced loudspeaker arrays for WFS rendering at low frequencies (below 1000 Hz) are compared with those of a reference regularly spaced loudspeaker array. The analysis considers a large number of sources and listening positions. The comparison is realized using perceptually relevant criteria.

3.1. Rendering accuracy evaluation

We simulate and compare for a listening position $\vec{r_j}$ (x_j, y_j) the frequency response of the system $H_\Psi(\vec{r_j}, k)$ with an "ideal" WFS response $A_\Psi(\vec{r_j}, t)$:

$$A_\Psi(\vec{r_j}, k) = Att_\Psi^{wfs}(\vec{r_j}) e^{-jkd_\Psi^j}, \quad (9)$$

where Att_Ψ^{wfs} stands for the real attenuation of a WFS source synthesized with a linear loudspeaker array [5]:

$$Att_\Psi^{wfs}(\vec{r_j}) = \sqrt{\frac{|y_L - y_{R_{av}}|}{|y_L - y_j|}} \sqrt{\frac{|y_i - y_\Psi|}{|y_{R_{av}} - y_\Psi|}} \frac{1}{4\pi d_\Psi^j}. \quad (10)$$

A quality function $Q_\Psi(\vec{r_j}, f)$ that describes the deviation of the synthesized response from an ideal response can be defined in the frequency domain as:

$$Q_\Psi(\vec{r_j}, k) = \frac{H_\Psi(\vec{r_j}, k)}{A_\Psi(\vec{r_j}, k)} \quad (11)$$

Magnitude deviation $MAG_\Psi(\vec{r_j}, m)$ and group delay deviation $GD_\Psi(\vec{r_j}, m)$ are then calculated for $ERB_N(m)$ frequency bands [6]. They are simply obtained by averaging the corresponding quantities derived from $Q_\Psi(\vec{r_j}, k)$ in the equivalent frequency band. The calculation considers 96 ERB_N bands for the entire audible frequency range. For the low frequency evaluation, it is however limited to frequency bands having their center frequency between 100 and 1000 Hz.

3.2. Test setup

We consider a test setup of 24 loudspeakers arranged in a 3.6 m long array. This corresponds to a regular spacing of 15 cm. Two alternative loudspeaker arrays of same length are considered:

- a "randomly" spaced loudspeaker array,
- a symmetrical "logarithmically" spaced array.

The latter is defined such that loudspeaker positions x_n are obtained from:

$$\begin{aligned} x_{n+1} - x_n &= (x_n - x_{n-1}) \times a^b \text{ if } n \geq 12 \\ x_{n+1} - x_n &= (x_n - x_{n-1}) \times a^{-b} \text{ otherwise,} \end{aligned} \quad (12)$$

We define a "loudspeaker spreading coefficient" ls_{spread}. In the case of the randomly spaced array, ls_{spread}^{rand} is simply defined as the ratio between the maximum and the minimum loudspeaker spacing. In the case of "logarithmically" spaced array, ls_{spread}^{log} is defined as the ratio between the spacing of the loudspeakers at the extremities of the array and the spacing of the loudspeakers at the center of the array. a and b are then calculated considering a given value of ls_{spread} and the total length of the array.
In the following, we consider $ls_{spread}^{rand} = 2$ and $ls_{spread}^{log} = 0.5$ (smaller spacing of the loudspeakers to the sides). The corresponding loudspeaker spacings are displayed in figure 2.

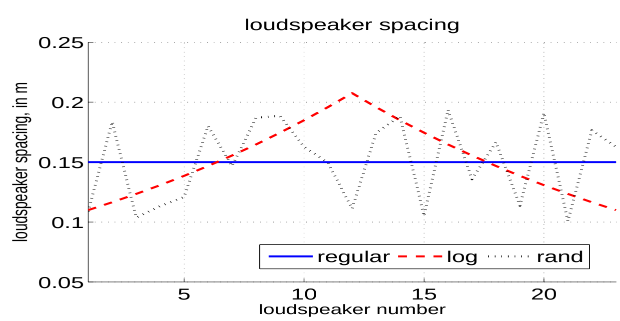

Figure 2: *loudspeaker spacing for the three array type*

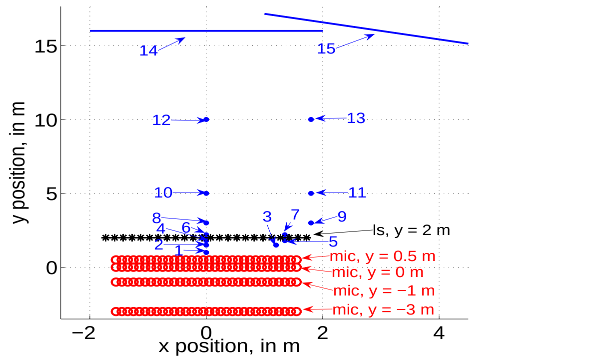

Figure 3: *Top view of loudspeakers (black *), microphones (red o), and test sources (blue dots) configuration for regularly spaced loudspeaker*

A test ensemble of 15 omnidirectional virtual sources (cf. figure 3) is composed of 5 centered and off-centered focused sources (sources 1/2/3/4/5), 8 centered and off-centered sources (sources 6/7/8/9/10/11/12), and 2 "plane waves" at 0 and 30 degrees (sources 14/15). The chosen test ensemble represents typical WFS sources reproduced by such a loudspeaker array. Figure 3 also displays measuring positions (microphone positions) at which the quality function Q_Ψ is evaluated for each source and loudspeaker array type.

3.3. Results

Tables 1 and 2 show mean values and standard deviation of MAG_{ERB} and GD_{ERB} calculated for all listening positions and virtual sources for the three loudspeaker array types between 100 and 1000 Hz.

	regular	log	rand
mean (dB)	-1.44	-1.40	-1.43
standard deviation (dB)	2.59	2.57	2.61

Table 1: *Mean value and standard deviation of MAG_{ERB} considering all microphone positions and virtual sources between 100 and 1000 Hz*

	regular	log	rand
mean (ms)	0.13	0.13	0.13
standard deviation (ms)	0.88	0.87	0.88

Table 2: *Mean value and standard deviation of GD_{ERB} considering all microphone positions and virtual sources between 100 and 1000 Hz*

The reproduction errors at low frequencies are due to known limitations of Wave Field Synthesis rendering (stationary phase approximation limitations, diffraction) that may be reduced using multichannel equalization methods such as described in [5] [7]. It can be seen that the three loudspeaker arrays show very similar performances in terms of both magnitude and group delay deviation. It can be expected that observed differences have no significant perceptual impact.

4. ALIASING FOR WAVE FIELD SYNTHESIS

The spatial sampling of the loudspeaker array limits the reconstruction possibilities of WFS at high frequencies. Contributions of individual loudspeaker do not fuse into a unique wave front as they do at low frequencies [8]. The synthesized sound field thus exhibits complex temporal and frequency characteristics [9] [8].
The spatial aliasing frequency corresponds to the corner frequency above which this phenomenon is noticeable. It is a key parameter for the analysis of the performances of a given loudspeaker array. Most available expressions of the aliasing frequency for WFS are given for infinite arrays of regularly spaced loudspeakers [9] [8]. They suggest that the aliasing frequency is independent of the listening position which is not true for finite length arrays [5].
In this section, alternative formulations of the spatial aliasing frequency are proposed that remain valid both for finite length and irregularly spaced loudspeaker arrays.

4.1. Frequency based evaluation of the aliasing frequency

4.1.1. Proposed criterion

We propose to extract the frequency response of the "aliased contributions" $H^{al}_\Psi(\vec{r_R}, k)$ from the frequency response of the considered array at position $\vec{r_R}$ for the synthesis of source Ψ using:

$$H^{al}_\Psi(\vec{r_R}, k) = H_\Psi(\vec{r_R}, k) - H^{noal}_\Psi(\vec{r_R}, k), \qquad (13)$$

where $H^{noal}_\Psi(\vec{r_R}, k)$ is the frequency response of a continuous linear array of same length for the synthesis of source Ψ. The exact response of a continuous array may be estimated as the frequency response of a regularly closely spaced (typically 1 cm) loudspeaker array. It is expected that for such an array aliasing artifacts are observed only above audible frequencies.
The aliasing frequency can thus be defined as the lower frequency for which the level of the aliased contributions exceeds a certain threshold $Tr^{simFreq}_{al}$:

$$f^{simFreq}_{al}(\vec{r_R}, \Psi) = min_f(|H^{al}_\Psi(\vec{r_R}, k)| > Tr^{simFreq}_{al}(\vec{r_R}, \Psi)) \qquad (14)$$

We propose to define this threshold as:

$$Tr^{simFreq}_{al}(\vec{r_R}, \Psi) = \frac{Att^{wfs}_\Psi(\vec{r_R})}{2}, \qquad (15)$$

which corresponds to half of the expected level at low frequencies.

4.1.2. Simulations

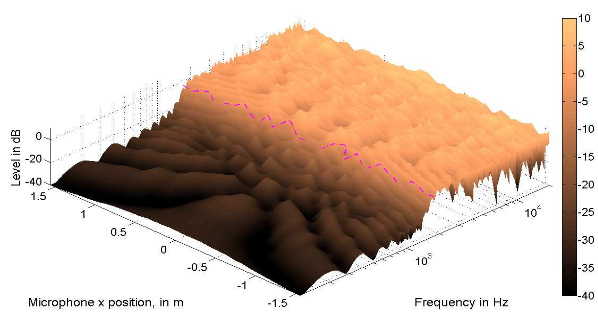

Figure 4: *Frequency responses of the aliased field H^{al}_Ψ, source 10, regularly spaced array*

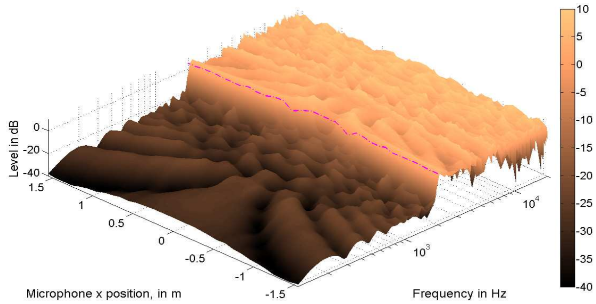

Figure 5: *Frequency responses of the aliased field H^{al}_Ψ, source 10, logarithmically spaced array*

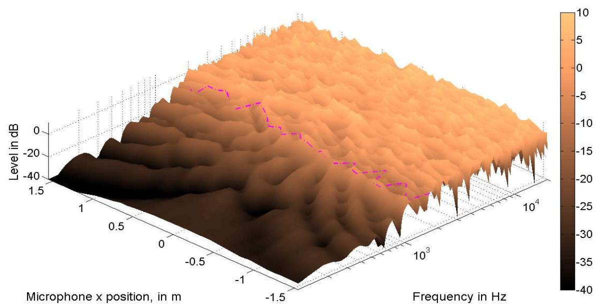

Figure 6: *Frequency responses of the aliased field H^{al}_Ψ, source 10, randomly spaced array*

$H^{al}_\Psi(\vec{r_R}, k)$ is evaluated for the three loudspeaker array types for a centered omnidirectional source located 3 m behind the loudspeaker array (source 10 in figure 3). Considered listening positions $\vec{r_R}$ are microphone positions at $y = 0\ m$ in figure 3.
Figures 4, 5, and 6 display the corresponding frequency responses. The frequency based aliasing criterion (cf. equation 15) is displayed on the figures as a magenta dashed-dotted line.
For both regularly spaced and logarithmically spaced loudspeaker arrays (cf. figures 4 and 5), a clear distinction can be observed between a low frequency response and high frequency response. At low frequencies, the level of the response is generally low (≈ -30 dB) whereas it raises quickly at higher frequencies and established a complex response with relatively high average level ($\approx 0\ dB$). The frequency based criterion establishes thus a clearly defined aliasing frequency. The same simulations were achieved considering other source/listening positions and have shown similar results.
For randomly spaced loudspeaker arrays, there is no such clear separation between low and high frequency responses. It can be seen that the "aliased field" has significant contributions ($\approx -15/-5\ dB$) from frequencies as low as 1000 Hz. The aliasing frequency is thus hardly defined for that kind of loudspeaker array. The proposed sampled version of the WFS filters (cf. equation 8) is probably not completely valid for randomly spaced loudspeaker arrays. An alternative WFS filter definition may provide increased reconstruction performances at higher frequencies but is out of the scope of this paper.

4.2. Temporal based evaluation of the aliasing frequency

The proposed frequency based criterion provides an accurate definition of the aliasing frequency. However, it requires the simulation of the aliased field response which is a computationally expensive task.
In this part, we propose a computationally efficient evaluation of the aliasing frequency which relies on sampling of the temporal response of the loudspeaker array at a given listening position.

4.2.1. Temporal response of a finite continuous array

In the following, the virtual source Ψ is located in Ω_Ψ and the 3 dB per octave filter $f(t)$ is omitted from the WFS filters to clarify the demonstration.
We define $t_\Psi(\vec{r_R}, x_L)$ as the arrival time at the listening position R of the contribution radiated by a secondary source at x_L:

$$t_\Psi(\vec{r_R}, x_L) = \frac{\Delta r}{c} + \tau_\Psi(x_L). \tag{16}$$

The impulse response h^{wfs}_Ψ of the continuous linear L for the synthesis of the source Ψ at $\vec{r_R}$ is thus expressed as:

$$h^{wfs}_\Psi(\vec{r_R}, t) = \int_{x_A}^{x_B} G_\Psi(x_L) \frac{\delta(t - t_\Psi(\vec{r_R}, x_L))}{4\pi \Delta r} dx_L, \tag{17}$$

We introduce $t^-(x_L)$ and $t^+(x_L)$,

$$\begin{array}{lcll} t^-(x_L) & = & t_\Psi(\vec{r_R}, x_L) & \forall x_L \in]x_A, \quad x_0], \\ t^+(x_L) & = & t_\Psi(\vec{r_R}, x_L) & \forall x_L \in]x_0, \quad x_B[, \end{array} \tag{18}$$

where x_0 is the intersection of L and the line joining the source Ψ and the receiving position R (cf. figure 1). Similarly, we define:

$$\begin{array}{lcll} x^-(t^-(x_L)) & = & x_L & \forall x_L \in]x_A, \quad x_0], \\ x^+(t^+(x_L)) & = & x_L & \forall x_L \in]x_0, \quad x_B[. \end{array} \tag{19}$$

Furthermore, we introduce:

$$\begin{aligned} h^-_\Psi(\vec{r_R}, t) &= \int_{x_A}^{x_0} G_\Psi(x_L) \frac{\delta(t - t^-(x_L))}{4\pi \Delta r(x_L)} dx_L, \\ h^+_\Psi(\vec{r_R}, t) &= \int_{x_0}^{x_B} G_\Psi(x_L) \frac{\delta(t - t^+(\vec{r_R}, x_L))}{4\pi \Delta r(x_L)} dx_L. \end{aligned} \tag{20}$$

By definition, the function $t^-(x_L)$ is a strictly increasing function for $x_L < x_0$ and $t^+(x_L)$ is a strictly decreasing function for $x_L > x_0$. The impulse response h^{wfs}_Ψ is thus the sum of the impulse responses of the two parts of the loudspeaker array separated by x_0 (h^-_Ψ and h^+_Ψ).
By substituting $t^-(x_L)$ and $t^+(x_L)$ to x_L into equation 20 and using the fundamental property of the direct distribution:

$$\begin{aligned} h^-_\Psi(\vec{r_R}, t) &= -Y(t - t_0) \frac{G_\Psi(x^-(t))}{4\pi \Delta r(x^-(t))} \frac{dx^-(t)}{dt} Y(t_A - t), \\ h^+_\Psi(\vec{r_R}, t) &= Y(t - t_0) \frac{G_\Psi(x^+(t))}{4\pi \Delta r(x^+(t))} \frac{dx^+(t)}{dt} Y(t_B - t), \end{aligned} \tag{21}$$

where Y is the Heavyside function.

4.2.2. *Derivation of aliasing criterion*

Let's consider an array of N ideal omnidirectional loudspeakers located at $x_n, n = [1 \dots N]$ such that $x_{n+1} > x_n$ and $x_A < x_n < x_B, i = [1 \dots N]$. We define $n_0 = min_n(x_n > x_0)$. The impulse response $h_\Psi^{samp}(\vec{r_R}, t)$ of this array for the synthesis of source Ψ can be obtained from WFS filters (cf. equation 8) as:

$$h_\Psi^{samp}(\vec{r_R}, t) = \sum_{n=1}^{n_0} \frac{|x_{n+1} - x_{n-1}|}{2} G_\Psi(x_n) \frac{\delta(t - t^-(x_n))}{4\pi\Delta r(x_n)} + \sum_{n=n_0}^{N} \frac{|x_{n+1} - x_{n-1}|}{2} G_\Psi(x_n) \frac{\delta(t - t^+(x_n))}{4\pi\Delta r(x_n)}. \quad (22)$$

Thus, it appears as the sum of time sampled versions of $h_\Psi^-(\vec{r_R}, t)$ and $h_\Psi^+(\vec{r_R}, t)$:

$$h_\Psi^{samp}(\vec{r_R}, t) = \quad (23)$$
$$h_\Psi^-(\vec{r_R}, t) \left(\sum_{n=1}^{n_0} \frac{|x_{n+1} - x_{n-1}|}{2} \frac{dt^-(x_n)}{dx^-} \delta(t - t^-(x_n)) \right) +$$
$$h_\Psi^+(\vec{r_R}, t) \left(\sum_{n=n_0+1}^{N} \frac{|x_{n+1} - x_{n-1}|}{2} \frac{dt^+(x_n)}{dx^+} \delta(t - t^+(x_n)) \right).$$

The spatial sampling of the loudspeaker array is thus equivalent to irregular time sampling of both $h_\Psi^+(\vec{r_R}, t)$ and $h_\Psi^-(\vec{r_R}, t)$. The minimum Nyquist frequency associated to each of the irregular temporal sampling therefore corresponds to the spatial aliasing frequency evaluated at R.
As for regular sampling, the Nyquist frequency is linked to the sample distribution, and especially to the time difference between successive samples. Two temporal distributions have to be considered: $t^-(x_n)$ for $n \leq n_0$ and $t^+(x_n)$ for $n > n_0$. The arrival time differences $\Delta\tau_R^\Psi(n)$ can be defined as:

$$\begin{cases} \Delta\tau_R^\Psi(n) = t^-(x_{n-1}) - t^-(x_n) & \text{for} \quad n_A < n \leq n_0 \\ \Delta\tau_R^\Psi(n) = t^+(x_{n+1}) - t^+(x_n) & \text{for} \quad n_0 < n < n_B. \end{cases} \quad (24)$$

We propose to define the spatial aliasing frequency f_{al}^{temp} derived from this analysis of the temporal response of the array as:

$$f_{al}^{temp}(\vec{r_R}, \Psi) = \frac{g_{al}}{\max_{n \in N_{sel}(\vec{r_R}, \Psi)} |\Delta\tau_R^\Psi(n)|}, \quad (25)$$

where g_{al} is a weighting factor and $N_{sel}(\Psi, \vec{r_R})$ is a subset of $n = [1 \dots N]$ defined as:

$$N_{sel}(\vec{r_R}, \Psi) = \left\{ n, \frac{G_\Psi(x_n)}{4\pi\Delta r(x_n)} > tr_{al} \cdot \max_{i=[1\dots N]} \left(\frac{G_\Psi(x_i)}{4\pi\Delta r(x_i)} \right) \right\}, \quad (26)$$

where tr_{al} is a threshold value used for the selection of loudspeakers that contribute significantly to the sound field at position R, recalling that $\frac{G_\Psi(x_i)}{4\pi\Delta r(x_i)}$ is the level of the contribution of loudspeaker i at R for the synthesis of Ψ.
Both g_{al} and tr_{al} are free parameters of the proposed calculation method. Optimization is proposed in the following.

4.2.3. *Validation*

The free parameters of the time domain method have been set so as to minimize the root mean square error of the time based estimation compared to the frequency based estimation of the aliasing frequency. Only regularly and logarithmically spaced loudspeaker arrays were considered. The obtained values are $g_{al} = 0.95$ and $tr_{al} = -13dB$.

Array type	Mean error	Standard deviation
Regular	-1.92%	7.69%
Logarithmic	1.12%	4.38%
Random	2.92%	22.58%

Table 3: *Error of aliasing frequency using time based compared to simulation based estimation for the three loudspeaker array types, considering all sources and microphone positions, cf. figure 3*

Table 3 presents mean values and standard deviation of the estimated error of the aliasing frequency using the temporal based criterion compared to the frequency based criterion. It can be seen that for finite length and/or logarithmically spaced loudspeaker arrays, the aliasing frequency can be reliably estimated using temporal based criterion which is computationally more efficient than frequency based criterion.
For the randomly spaced loudspeaker array, both criteria provide rather dissimilar results. However, for this type of array, the aliasing frequency is difficult to define (cf. section 4.1.2).

5. ALIASING FREQUENCY DEPENDENCY ON LOUDSPEAKER SPACING

In this section we compare irregularly spaced loudspeaker arrays with regularly spaced arrays in terms of obtained aliasing frequency. The test parameter is the loudspeaker spreading coefficient that determines the amount of irregularity introduced in the loudspeaker spacing.

5.1. Aliasing for randomly spaced loudspeakers

Figure 7 shows quantiles (0.1, median, 0.9) of the aliasing frequency estimated with the frequency based criterion. The analysis is performed on "random" loudspeaker spacing for different values of ls_{spread}^{rand}. For each defined loudspeaker array all sources and all microphone positions of the test setup (cf. figure 3) are considered for the evaluation. We recall that $ls_{spread}^{rand} = 1$ corresponds to a regularly spaced loudspeaker array.
It can be seen that the aliasing frequency is generally lower for randomly spaced arrays than for regularly spaced arrays. A deeper analysis considering each source and listening position separately did not show any particular improvement. One should consider however that the aliasing frequency is not properly defined for this kind of array.

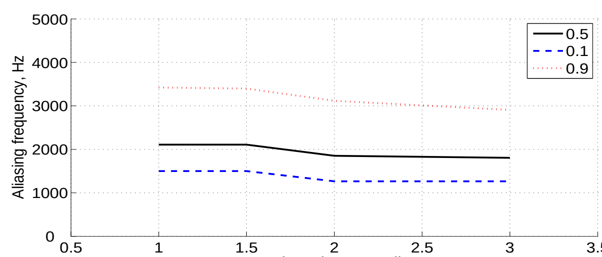

Figure 7: *Quantiles of aliasing frequency, randomly spaced arrays, all sources and microphone positions, spreading coefficient dependency*

5.2. Aliasing for logarithmic loudspeaker arrays

For logarithmically spaced loudspeaker arrays, a spreading coefficient below 1 corresponds to a larger spacing to the sides compared to the center, whereas a spreading coefficient above 1 implies a smaller spreading to the sides.

Figure 8 shows quantiles (0.1, median, 0.9) of the aliasing frequency estimated with the time based criterion for different values of ls^{log}_{spread} considering all sources and all microphone positions of the test setup (cf. figure 3). It can be seen that all spreading coefficients above 1 generally decrease the aliasing frequency, whereas spreading coefficients around 0.5 provide a slight increase of both median and 0.9 quantile.

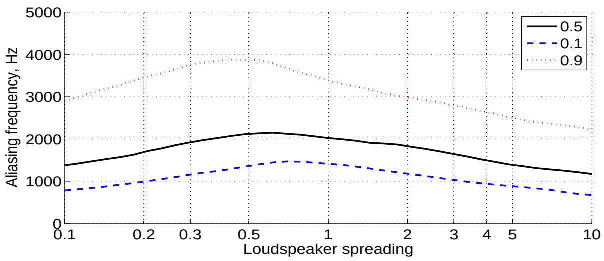

Figure 8: *Quantiles of aliasing frequency, logarithmically spaced arrays, all microphone positions, loudspeaker spreading coefficient dependency, all sources*

Figures 9 and 10 show respectively quantiles of aliasing frequency considering non-focused sources only (sources 6 to 15 in 3) and focused sources only (sources 1 to 5 in 3). This analysis shows significant increase of the aliasing frequency using a logarithmically spaced loudspeaker array for non-focused sources for a loudspeaker spreading coefficient of 0.5. Most significant increase is for the 0.9 quantile value which raises by more than 20 % compared to regularly spaced loudspeakers.

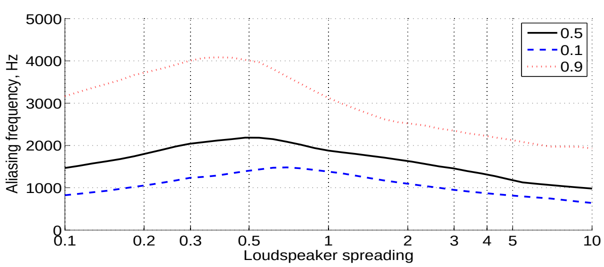

Figure 9: *Quantiles of aliasing frequency, logarithmically spaced arrays, all microphone positions, spreading coefficient dependency, non-focused sources only*

However, it can be seen from figure 10 such loudspeaker spreading coefficients lower the aliasing frequency for focused sources.

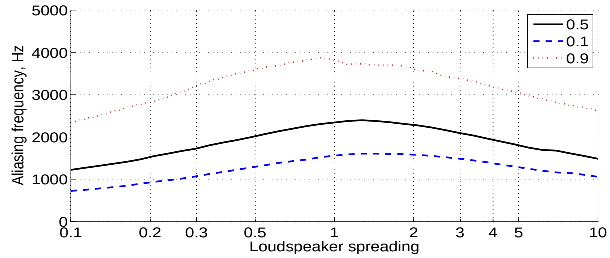

Figure 10: *Quantiles of aliasing frequency, logarithmically spaced arrays, all microphone positions, spreading coefficient dependency, focused sources only*

6. CONCLUSION

In this paper, the potential use of irregularly spaced loudspeaker arrays for WFS has been addressed. Two test arrays have been compared to a regularly spaced loudspeaker array of same length. It has been shown that the three arrays have similar performances at low frequencies. New formulations for aliasing frequency have been introduced. They provide accurate results for finite length arrays with both regular and irregular loudspeaker spacing. It has been shown however that the aliasing frequency is difficult to define for randomly spaced loudspeaker arrays. It was also shown that, for the considered loudspeaker arrays (24 channels, 3.6 m long), dual logarithmic spacing allows for a significant increase in the aliasing frequency considering non focused virtual sources. If both focused and non focused sources need to be rendered on the same array, regular spacing remains most effective.

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