Microphone Array for Soloists

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Introduction

This article presents the design and the performances of a two dimensional broadband microphone array intended to record soloist singer(s) in the cathedral of Lausanne. Different array geometries are discussed with respect to the possibility to obtain a desired directivity pattern (beampattern) with the smallest number of microphones. The method used to keep the directivity roughly constant over a decade (300 - 3000 Hz) is also presented.

Specifications and methodology

A high directivity is needed in order to efficiently reject the reverberated field. For that purpose, broadside two-dimensionnal (or planar) array have been considered. The microphones are placed in a horizontal plane at about 4 m above the soloist(s). Under plane waves assumption, specifications can be set on the farfield beampattern. The position of the singers is not fixed but depends on the musical work. Consequently, it must be possible to electronically steer the main lobe without moving the microphones. For a given steering direction, the singer may not be exactly aligned with the mainlobe, which can lead to a filtering effect if the beamwidth varies too much with frequency, as it is the case with basic delay and sum beamformer. Hence, the aim of the design is to impose a constant value for the half-power beamwidth ($\Delta\theta_{3dB}$).

Constant beamwidth beamforming

Many design methods exist for broadband beamforming (see [1], [2] and [3]). A well-known method consists in separating the array in N subarrays, each of these being active only on one part of the whole frequency band of interest (hence, the whole band is split in N sub-bands). A band-pass filter placed after each sub-array allows the concerned sub-band to be selected. This method makes possible to reduce the beamwidth variation over the whole frequency band, but there might however remain some considerable variation within each sub-band.

The method used here is more performant from this point of view. We take inspiration from the nested subarray method described above, but with the following difference : If the whole frequency band is split in N sub-bands, the array includes N+1 subarrays, each of these being dimensioned to have the appropriate directivity at a single frequency, called the dimensioning frequency of the subarray. These dimensioning frequencies are the N+1 frequencies which delimit the N sub-bands. For this application, the ratio between two consecutive dimensioning frequencies has been set to r = 3. Between two dimensioning frequencies, a linear combination of the outputs of the two concerned subarrays is calculated :

$$M(\varphi,\theta,f) = H_i(f)\, M_i(\varphi,\theta,f) + H_{i+1}(f)\, M_{i+1}(\varphi,\theta,f)$$

M_i is the response of the i^{th} subarray (assumed to be known) and M is the response of the complete array. The angles φ and θ (of the spherical coordinate system) are the incident direction of the plane wave. By imposing a frequency independant value for M for two incident directions (in the direction of the maximum of the beampattern and in a direction at -3 dB below the maximum), a system of two equations with two unknowns (H_i et H_{i+1}) is obtained. When resolving this system for each frequency, one obtains the ideal amplitude response of filters H_1, H_2 and H_3. For the present application, these filters are implemented as numerical linear phase FIR filters of order 75. The resulting beamformer can be represented as shown in figure 1 (the details of each subarray is described in the next section).

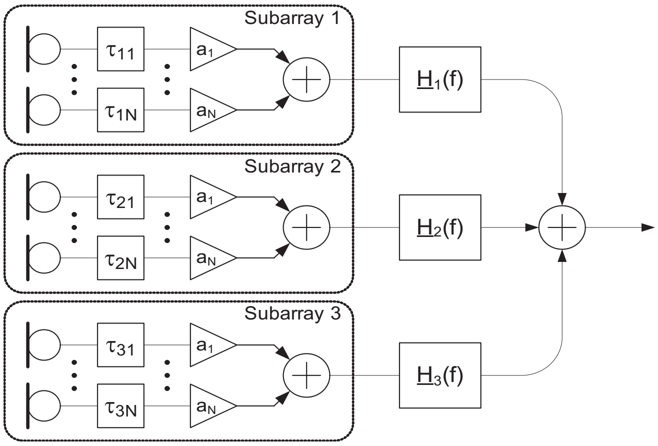

Figure 1: Beamformer structure : the complete array include 3 subarrays of type “delay and sum beamformer”. Band-pass filters placed after each subarray allows to combine subarrays outputs in an appropriate way to keep the beamwidth roughly independant with frequency.

Design of a subarray

The design of a subarray consists in choosing the positions of the microphones and in applying appropriate delays and ponderation factors on the microphone outputs, in order to obtain the desired beampattern at the dimensioning frequency of the subarray. Other subarrays are obtained by homothety of ratio r = 3 (the dimensioning frequencies are in the same ratio). Concerning the overall number of microphones, it is advantageous to choose a geometry which allows the re-use of certain microphones in several subarrays.

a) <u>Choice of the subarray geometry</u> : For linear array, it is well-known (for a given frequency) that the beamwidth is mainly fixed by the array length, and that distance between sensors is related to spatial aliasing (grating lobes). For planar array, it is less intuitive to know the influence of the position of the microphones on the beampattern, which is why the following property is useful : For each section of the beampattern (a section of the beampattern is the directivity variation with θ, for a particular value of φ), an equivalent linear array can be found, which has the same beampattern in the considered section. This property is illustrated in figure 2 in the case of an array placed in the 'xy' plane which presents a mainlobe in the 'z' direction.

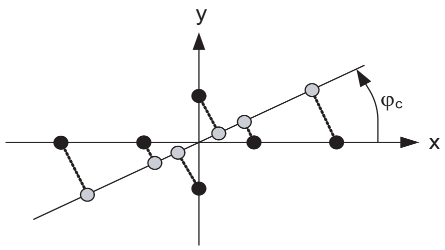

Figure 2: Projection property for planar array : the initial array is depicted in black. The equivalent array for the section of the beampattern determined by $\varphi = \varphi_c$ (in grey) is obtained by projecting the microphones on the line which determines the concerned section. It can be shown that the beampattern of these two arrays are identical in the section $\varphi = \varphi_c$.

This property allows to compare different planar geometries such as squaring of microphones, circle or cross-shape. As a result of the property mentioned above, the squaring presents a redundancy of microphones, whereas the circle appears to be the best solution with respect to the directivity we can obtain with a given number of microphones. However, the use of circles as subarrays does not allow the re-use of certain microphones in several subarrays, which leads to an excessive number of microphones. Finally, the cross-shape geometry has been chosen because it presents the best compromise between performances and number of microphones (see figure 3).

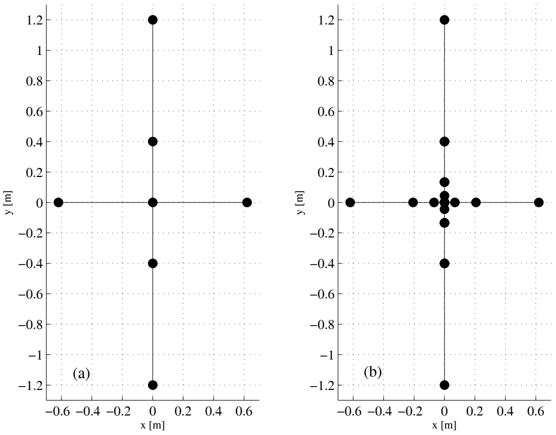

Figure 3: (a) First subarray of 7 microphones dimensioned at 310 Hz. (b) Complete array including 3 subarrays. The second and third subarrays are obtained by homothety of factor 3 and 9 respectively. Thus, this array presents frequency independant beamwidth between 310 Hz and 3x3x310 = 2790 Hz. The chosen geometry allows to use only 15 microphones instead of 3x7 = 21.

b) <u>Beamformer structure for a subarray</u> : Figure 1 shows the implementation of a subarray. Delays (τ_{ij}) allow to steer the mainlobe in the desired direction, whereas frequency independant weighting coefficients (a_i) allow to operate on the beamwidth and the sidelobe levels. These delays and coefficients must be set for each pointing direction.

Calculated performances

The calculated directivity of the designed array is presented for the case where the soloist is placed below the array (the mainlobe points in the 'z' direction) :

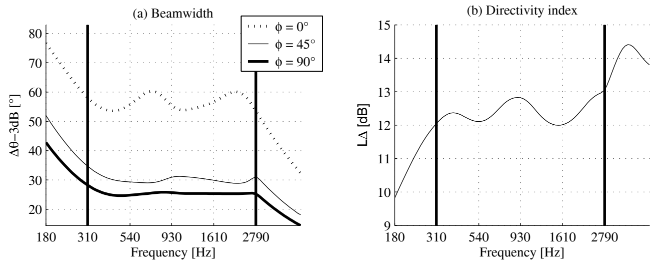

Figure 4: (a) Half-power beamwidth versus frequency for 3 sections of the beampattern. (b) Directivity index versus frequency.

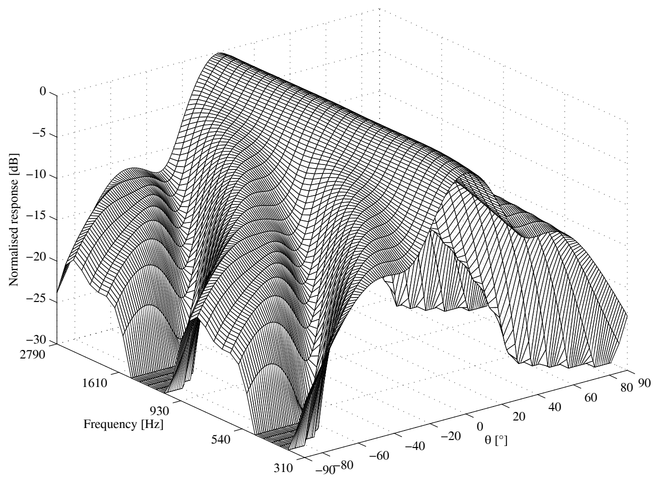

Figure 5: Frequency dependance of one particular section ($\varphi = 90°$) of the beampattern.

As desired, the beamwidth variation remains quite low in the frequency band of interest (310 – 2790 Hz). It can also be seen that the directivity index variation on this band is only 1 dB. Similar results can be obtained for other steering directions.

Conclusion

A two-dimensionnal broadband microphone array has been designed with a particular implementation of the well-known nested subarray method. This method allowed to obtain a roughly frequency independant beamwidth over a decade. For the present application, it was observed that the cross-shape geometry has presented the best compromise between performances and number of microphones.

References

[1] M. M. Goodwin and G. W. Elko, "Constant Beamwidth Beamforming", Proc. IEEE Int. Conf. Acoust. Speech Sig. Process. (ICASSP93), Vol 1, pp. 169-172 (1993).

[2] Michael Brandstein – Darren Ward, "Microphone arrays", Berlin : Springer, 2001.

[3] M. Van der Wal, E. W. Start, D. de Vries, "Design of Logarithmically Spaced Constant-Directivity Transducer Arrays", J. Audio Eng. Soc. 1996, pp 497-507.