Software for Estimation and Processing Simulation of Microphone Line Arrays

Christos Sevastiadis, Christodoulos Chamzas*, George Kalliris, George Papanikolaou

Dept. of Electrical and Computer Engineering, Aristotle University of Thessaloniki. University Campus, 640 60, Thessaloniki, Greece – csevos@otenet.gr

*Cultural and Educational Technol. Inst., P.O. Box 218, Tsimiski 58, 671 00, Xanthi, Greece

Summary: A software for estimation and processing simulation of microphone line arrays was developed, based on the fundamentals of sensor array theory and the basic concepts governing the array beamforming. This stand-alone, full adjustable simulation software can be a user-friendly tool in understanding the basic concepts of the microphone array design theory.

INTRODUCTION

In sound pick-up systems for auditoriums, teleconferencing, and multimedia workstations, the environment performs the initial transformation on speech signals. The problems of reverberation and ambient noise are greater as the distance between the speaker and the microphone is increased. A highly directive microphone near the mouth of the speaker is not a convenient solution, because the common need in these applications is transparency of communication: people when communicate via technical media would like to feel like when they communicate face to face.

An alternative practical solution is the use of microphone arrays. These systems are beamformers and they can be steered or steer dynamically to a desired talker location. This kind of microphone systems is filtering spatially the incident sound waves. This means that, waves propagated from a desired direction are allowed preferably to come out from system’s output. Consequently, microphone arrays eliminate the effect of sound reflections and noise in the desired signal by maximizing their signal-to-noise ratio.

MICROPHONE ARRAYS

The techniques used in microphone and sensor arrays can be categorized in fixed (1), (2), (3) and adaptive ones (1), (4). The fixed techniques are based on fixed filters and the most used technique is the delay–sum beamforming: the array output is the summation of the single microphone output signals. For an arbitrary microphone array with \( N \) omni-directional microphones the general form of the output signal \( H \), due to a time-harmonic plane wave with wavevector \( \mathbf{k} \), can be written as

\[
H(k, r) = \sum_{n=0}^{N-1} a_n e^{-jkr_n}
\]  

(1)
where $a_n$ is weighting for sensor $n$, and $r_n$ is the position vector of sensor $n$ with respect to some defined origin. For the case of line arrays with sum-delay fixed steering we can write

$$H(k,d) = \sum_{n=0}^{N-1} a_n e^{-j\frac{2\pi}{\lambda} x_n (\sin\theta - \sin\theta')}$$  \hspace{1cm} (2)

where $d$ is the constant distance between the microphones, $x_n$ is the position of the $n$th microphone, $\lambda$ is the wave length of the incident sound wave, $\theta$ is the incident angle of the sound wave front and $\theta'$ is the desired steering angle of the microphone array.

A typical measure used for the qualification of signal-to-noise ratio, that characterizes microphone array’s output, is the directivity index (1). Because the bandwidth of microphone arrays is inversely proportional to the frequency, the directivity index increases with frequency. However, the problem with reverberation and noise in rooms is more prominent in low frequencies. So, a uniformly spaced microphone array can not be effective for the entire sound spectrum. One solution for this problem is to use logarithmic microphone arrays: the array is split into multiple frequency bands by “nesting” the microphones harmonically (1), (2), (3). In Figure 1 we can see examples of a single and a logarithmic microphone array.

![Diagram](image)

**FIGURE 1.** Basic forms of single and logarithmic line microphone array.

**SOFTWARE DEVELOPMENT**

For the comprehension of the microphone array theory we developed a computer program under Windows 95/98 operating system. By means of this software we can design periodical, non-periodical and logarithmic line microphone arrays and study their performance. The operations of the software can be grouped in two categories.

With operations of the first category an array can be designed and evaluated by means of diagrams. Array responses and indices are calculated by hypothesizing that the sound waves have plane fronts and unity amplitude. The vertical azimuth of the incident waves supposed to be zero degrees. The microphones supposed to be omni-directional and the steering aperture supposed to be between $-\pi/2$ to $\pi/2$. In addition, the calculation results are normalized and
the spatial dimensions are disregarded. Thus, arrays consisting of subarrays with constant distance between their microphones can be created. A filter can be assigned on every subarray output, and the array can be steered to a desired direction. The software can design diagrams and forms of array directivity pattern for a desired frequency, array directivity indices for the entire frequency spectrum, microphone locations and nesting, subarray filter responses, and the array frequency response on a desired direction. There is also the option of printing the diagrams. Output examples produced by the software are shown in Figures 2 and 3.

Using the second category of operations, a designed microphone array can be simulated by employing sound recordings as incident signals. The sound recordings are processed and the output can be played. To support these operations with the software, monophonic wave files can be recorded, played stored and loaded. Each wave file is linked with an individual sound source and a desired incident direction. The output of the processed wave files is a monophonic wave file. Software dialog forms supporting these operations are shown in Figure 4.

![Image](image.png)

**FIGURE 2.** The start up screen with the array design dialog form and the directivity pattern diagram.

**CONCLUSION**

The choice of simulation versus hardware realization can offer great flexibility in the design and study of microphone arrays. This stand-alone, full-adjustable simulation software can be a user-friendly tool in understanding the basic concepts of the microphone array design theory. Thus, the development of real-time operating microphone arrays can be easier and more efficient.
FIGURE 3. Directivity index diagrams of a single array and a logarithmic array.

FIGURE 4. Sound source and output manipulation dialog forms.

REFERENCES