2aEA3. Small directional microelectromechanical systems (MEMS) microphone arrays

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Directional microphone arrays that are physically small compared to the acoustic wavelength are of great interest for hand-held communication devices. Spatially directive microphones can reduce the impact of background acoustic noise without adding distortion to the signal. This talk will present some design topologies and requirements as well as a new physical design for a MEMS velocity sensing microphone that could enable directional microphone responses while being small in size.
INTRODUCTION

Small directional microphones are becoming important in communication devices that need to reduce background noise in acoustic fields in order to improve communication quality and speech recognition performance. As communication devices have become smaller, the need for small directional microphones has become more important. However, small directional microphones are inherently sensitive to wind noise and wind-induced noise. The wind noise problem has been well known in the hearing aid industry, especially since the introduction of directionality in hearing aids. One technique that has proven effective in combating wind noise has been to realize a directional microphone by using multiple omnidirectional microphones with an adaptive beamforming algorithm. By allowing the beamformer to adaptively alter its beampattern as a function of time, wind and sensitivity to microphone self-noise can be significantly reduced.

To attain directivity with closely spaced omnidirectional microphones it is necessary to have the beamformer respond to the spatial derivatives of the sound field. These types of microphone arrays are therefore referred to as differential microphone arrays. One requisite for a microphone to respond to the spatial pressure differential is the implicit constraint that the microphone size is smaller than the acoustic wavelength. Differential microphone arrays can be seen directly analogous to finite-difference estimators of continuous spatial field derivatives along the direction of the microphone elements. Differential microphones also share strong similarities to superdirectional arrays used in electromagnetic antenna design. The well-known problems with implementation of superdirectional arrays are the same as those encountered in the realization of differential microphone arrays. It has been found that a practical limit for differential microphones using currently available microphones is at third-order.

Commensurate with the realization of a differential microphone is the inherent requirement that the phase and magnitude matching be controlled and reproducible. In the past the strict requirements on magnitude and phase matching has made it expensive to realize differential microphone arrays due to the cost of matching in production or matching the microphones during manufacturing of the product. Fortunately newer semiconductor production methods that have been introduced with MEMS technology have resulted in inexpensive microphones that have usable magnitude and phase matching tolerances. With the growing trend toward using MEMS-based microphones in consumer devices, the ability to realize differential microphones in consumer devices has therefore greatly improved.

The paper discusses the implementation of an adaptive dual microphone first-order differential microphone array as well as a novel way to construct a velocity sensing MEMS element that utilizes the physics of the boundary layer near a surface to potentially offer some protection from wind and fluid flow.

FIRST ORDER DIFFERENTIAL MICROPHONES

Fig. 1 illustrates a first-order differential microphone having two closely spaced pressure (i.e., omnidirectional) microphones spaced at a distance $d$ apart, with a plane wave $s(t)$ of amplitude $S_0$ and wavenumber $k$ incident at an angle $\theta$ from the axis of the two microphones.

The output $m_i(t)$ of each microphone spaced at distance $d$ for a time-harmonic plane wave of amplitude $S_0$ and frequency $\omega$ incident from angle $\theta$ can be written according to the expressions of Equation (1) as follows:

$$m_1(t) = S_0 e^{j\omega t - jkd \cos(\theta) / 2}$$
$$m_2(t) = S_0 e^{j\omega t + jkd \cos(\theta) / 2}$$

Equation (1)

The output $E(\theta, t)$ of a weighted addition of the two microphones can be written according to Equation (2) as follows:

$$E(\theta, t) = w_1 m_1(t) + w_2 m_2(t)$$
$$= S_0 e^{j\omega t \left[(w_1 + w_2) + (w_1 - w_2) jkd \cos(\theta) / 2 + h.o.t.\right]}$$

Equation (2)
FIGURE 1. Schematic of dual microphone beamformer where \( W_1 \) and \( W_2 \) are weighting values (possibly complex and a function of frequency) applied to the first and second microphone signals, respectively.

If \( kd \ll \pi \), then the higher-order terms ("h.o.t." in Equation (2)) can be neglected. If \( W_1 = -W_2 \), then we have the pressure difference between two closely spaced microphones. This specific case results in a dipole directivity pattern \( \cos(\theta) \) as can easily be seen in Equation (2). However, any first-order differential microphone pattern can be written as the sum of a zero-order (omnidirectional) term and a first-order dipole term \( \cos(\theta) \). A first-order differential microphone implies that \( W_1 = -W_2 \). Thus, a first-order differential microphone has a normalized directional pattern \( E \) that can be written according to Equation (3) as follows:

\[
E(\theta) = \alpha \pm (1-\alpha)\cos(\theta)
\]

where typically \( 0 \leq \alpha \leq 1 \), such that the response is normalized to have a maximum value of 1 at \( \theta = 0 \degree \), and for generality, the \( \pm \) indicates that the pattern can be defined as having a maximum either at \( \theta = 0 \degree \) or \( \theta = \pi \degree \). One implicit property of Equation (3) is that, for \( 0 \leq \alpha \leq 1 \), there is a maximum at \( \theta = 0 \degree \) and a minimum at an angle between \( \pi/2 \) and \( \pi \). For values of \( 0.5 < \alpha \leq 1 \), the response has a minimum at \( \pi \degree \), although there is no zero in the response. A microphone with this type of directivity is typically called a "sub-cardioid" microphone. Fig. 2 shows some typical first-order beampatterns.

FIGURE 2. Some typical beampatterns for a first-order differential microphone (a) dipole, (b) cardioid, (c) hypercardioid, (d) supercardioid.
For values of $0 \leq \alpha \leq 0.5$, there is a null at

$$\theta_{null} = \cos^{-1} \frac{\alpha}{\alpha - 1}. \quad (4)$$

Fig. 2(b) shows a directional response corresponding to $\alpha = 0.5$ which is the cardioid pattern. The concentric rings in the polar plots of Fig. 2 are 10dB apart. One issue in forming a first-order differential microphone using two omnidirectional microphone elements can be seen by examining Eqn. 2. Since the frequency and phase responses of the omnidirectional and the dipole term are different one or both terms have to be filtered to allow for the correct combination to realize the first-order beamformer. A computationally simple and elegant way to deal with the equalization problem to form a general first-order differential microphone is to form a scalar combination of forward-facing and backward-facing cardioid signals. These signals can be obtained by using both solutions in Equation (3) and setting $\alpha = 0.5$. The sum of these two cardioid signals is omnidirectional (since the $\cos(\theta)$ terms subtract out), and the difference is a dipole pattern (since the constant term $\alpha$ subtracts out).

Fig. 3 shows a first-order differential beamformer constructed by the combination of two omnidirectional microphones to form back-to-back cardioid microphones. The back-to-back cardioid signals can be obtained by a simple modification of the differential combination of the omnidirectional microphones as shown in Fig 3. [1]. Cardioid signals can be formed from two omnidirectional microphones by including a delay ($T$) before the subtraction (which is equal to the propagation time ($d/c$) between microphones for sounds impinging along the microphone pair axis).

A practical way to realize the back-to-back cardioid arrangement shown in Fig. 3 is to carefully choose the spacing between the microphones and the sampling rate of the A/D converter to be equal to some integer multiple of the required delay. By choosing the sampling rate in this way, the cardioid signals can be made simply by combining input signals that are offset by an integer number of samples. This approach removes the additional computational cost of interpolation filtering to obtain the required delay, although it is relatively simple to compute the interpolation if the sampling rate cannot be easily set to be equal to the propagation time of sound between the two sensors for on-axis propagation.

By combining the microphone signals defined in Equation (1) with the delay and subtraction as shown in Fig. 3, a forward-facing cardioid microphone signal can be written according to Equation (5) as follows:

$$C_{F}(kd, \theta) = -2jS_{a} \sin(kd[1 + \cos \theta]/2). \quad (5)$$

Similarly, the backward-facing cardioid microphone signal can similarly be written according to Equation (6) as follows:

$$C_{B}(kd, \theta) = -2jS_{a} \sin(kd[1 - \cos \theta]/2). \quad (6)$$

If both the forward-facing and backward-facing cardioids are averaged together, then the resulting output is given according to Equation (7) as follows:

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**FIGURE 3.** Schematic of back-to-back cardioid first-order differential beamformer. Filters $h_{12}$ and $h_{21}$ are the required delay filters and $h_{1eq}$ and $h_{2eq}$ and $h_{L}$ are equalization filters.

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The microphone signals are sampled at the frequency represented in Figure 1 and 3 arrays due to the small size of the microphone array. Beamforming practically realizable.

Combinations of closely spaced pressure microphones already have SNR design limitations, one can conclude that directional microphones based on a differential design is a severe, if not insurmountable, technical challenge. With the knowledge that small MEMS have an SNR that is more than 55 dB below the SNR at 343 kHz where the SNR will be 3 dB higher than the SNR for a single microphone. At 1 kHz, the output will be exceedingly low (the signal will be orders of magnitude below the acoustical noise).

The sensitivity to noise is also indicative of robustness issue in the differential beamformer requiring good matching of microphone elements and thus the differencing used in differential microphones does not commensurately reduce the independent noise components as it does to acoustic signals. In fact, the noise components add and further reduce the SNR of the differential MEMS microphone array. As a result, the SNR will be exceedingly low (the signal will be orders of magnitude below the noise) for a MEMS pressure-differential microphone over the desired audio frequency of human hearing. The sensitivity to noise is also indicative of robustness issue in the differential beamformer requiring good matching of both amplitude and phase.

A main result from the above development is that the frequency response of the differential microphone has a high-pass response at 6 dB per octave. In a realization of a differential microphone as the difference between closely spaced pressure microphones, the distance between the elements sets the overall sensitivity. This overall sensitivity is proportional to $kd$. It is therefore evident that for very small arrays (relative to the acoustic wavelength), the output of a differential microphone system can be extremely small, especially at lower frequencies. It is well known that the physics of small MEMS microphone leads to overall SNR limitation due to thermal self-noise. Both electronic and physical thermal noise is independent between the microphone elements and thus the differencing used in differential microphones does not commensurately reduce the independent noise components as it does to acoustic signals. This is due to the fact that the noise components add and further reduce the SNR of the differential MEMS microphone array. As a result, the SNR will be exceedingly low (the signal will be orders of magnitude below the noise) for a MEMS pressure-differential microphone over the desired audio frequency of human hearing. The sensitivity to noise is also indicative of robustness issue in the differential beamformer requiring good matching of both amplitude and phase.

As an example, assume that the spacing of two MEMS active areas is 0.5 mm. A dipole microphone (formed by subtracting the output of two closely spaced pressure microphones) will have a maximum output when the spacing is $\frac{\lambda}{2}$. This spacing results in a maximum output for signals propagating along the microphone pair axis at 343 kHz where the SNR will be 3 dB higher than the SNR for a single microphone. At 1 kHz, the output will have an SNR that is more than 55 dB below the SNR at 343 kHz. Clearly this loss of SNR to obtain directionality by a differential design is a severe, if not insurmountable, technical challenge. With the knowledge that small MEMS pressure microphones already have SNR design limitations, one can conclude that directional microphones based on combinations of closely-spaced (MEMS-scale spacing) MEMS acoustic pressure sensing microphones will not be practically realizable.

Fig. 5 shows the frequency responses for signals incident along the microphone pair axis ($\theta = 0$) for a dipole microphone, a cardioid-derived dipole microphone, and a cardioid-derived omnidirectional microphone. Note that the cardioid-derived dipole microphone and the cardioid-derived omnidirectional microphone have the same frequency response. In each case, the microphone-element spacing is 2 cm. At this angle, the zero occurs in the cardioid-derived dipole term at the frequency where $kd = 2\pi$.

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Beamformer designs based on delay-sum or additive filter-sum beamforming are not appropriate for MEMS-sized arrays due to the small size of the microphone array relative to the acoustic wavelength.

Adaptive differential microphone

Fig. 3 showed the configuration of an adaptive differential microphone as introduced in Ref. [1]. As represented in Figure 1 and 3, a plane-wave signal $s(t)$ arrives at two omnidirectional microphones at an angle $\theta$. The microphone signals are sampled at the frequency $1/T$ by analog-to-digital (A/D) converters and filtered by

$$E_{c-\text{omni}}(kd, \theta) = \frac{1}{2} \left[ C_F(kd, \theta) + C_B(kd, \theta) \right] = -2 j S_o \sin(kd/2) \cos([kd/2] \cos \theta).$$

(7)

For small $kd$, Equation (7) has a frequency response that is a first-order high-pass, and the directional pattern is omnidirectional.

The subtraction of the forward-facing and backward-facing cardioids yields the dipole response of Equation (8) as follows:

$$E_{c-\text{dipole}}(kd, \theta) = C_F(kd, \theta) - C_B(kd, \theta) = -2 j S_o \cos(kd/2) \sin([kd/2] \cos \theta).$$

(8)

A dipole constructed by simply subtracting the two pressure microphone signals has the response given by Equation (9) as follows:

$$E_{\text{dipole}}(kd, \theta) = -2 j S_o \sin([kd/2] \cos \theta).$$

(9)

One observation to be made from Equation (8) is that the dipole's first zero occurs at twice the value ($kd = 2\pi$) of the cardioid-derived omnidirectional and cardioid-derived dipole term ($kd = \pi$) for signals arriving along the axis of the microphone pair.

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updata to the gradient of the surface with respect to the adaptive weight parameter. The steepest-descent algorithm due to its simplicity and ease of implementation.

The steepest-descent algorithm finds a minimum of the error surface $E[e^2(t)]$ by stepping in the direction opposite to the gradient of the surface with respect to the adaptive weight parameter $\beta$. Thus, we can write [1] an LMS update equation as follows:

$$\beta_{t+1} = \beta_t + 2\mu e(t)c_B(t).$$

A desired signal $S(j\omega)$ arriving from straight on ($\theta = 0$) is distorted by the factor $|\sin(kd)|$. For a microphone used for a frequency range from about $kd = 2\pi \cdot 100\text{Hz} \cdot T$ to $kd = \pi 2$, first-order recursive low-pass filter can equalize the mentioned distortion reasonably well. There is a one-to-one relationship between the adaptation factor $\beta$ and the null angle $\theta_n$ as given by Equation (12) as follows:

$$\beta = \frac{\sin \frac{kd}{2}(1 + \cos \theta_n)}{\sin \frac{kd}{2}(1 - \cos \theta_n)}.$$  

Since it is expected that the sound field changes with time, it is of interest to allow the first-order microphone to adaptively compute a response that minimizes the output under a constraint that signals arriving from a selected range of direction are not impacted. An LMS or Stochastic Gradient algorithm is a commonly used adaptive algorithm due to its simplicity and ease of implementation.

The subtraction node generates the unfiltered output signal $e(n)$ according to Equation (13) as follows:

$$e(t) = c_F(t) - \beta c_B(t).$$

$$E(j\omega,d) = e^{-jkd} \cdot 2j \cdot S(j\omega) \cdot [\sin(\frac{kd}{2}(1 + \cos \theta)) - \beta \sin(\frac{kd}{2}(1 - \cos \theta))]$$

$C_F(j\omega,d) = S(j\omega) \cdot [e^{jkd\cos \theta} - e^{-jkd(1+\cos \theta)}]$, 

$C_B(j\omega,d) = S(j\omega) \cdot [e^{-jkd\cos \theta} - e^{-jkd(1+\cos \theta)}]$
The intervals $\beta \in [0,1]$ and $\beta \in [1,\infty)$ are mapped onto $\theta \in [0.5\pi,\pi]$ and $\theta \in [0,0.5\pi]$, respectively. For negative $\beta$, the directivity pattern does not contain a null. Instead, for small $|\beta|$ with $-1 < \beta < 0$, a minimum occurs at $\theta = \pi$; the depth of which reduces with growing $|\beta|$. For $\beta = -1$, the pattern becomes omnidirectional and, for $\beta < -1$, the rear signals become amplified relative to the front. An adaptive algorithm chooses $\beta$ such that the energy of $e(n)$ in a certain exponential or sliding window becomes a minimum. As such, $\beta$ should be constrained to the interval $[-1,1]$. Otherwise, a null may move into the front half plane and suppress the desired signal. For a pure propagating acoustic field (no wind or self-noise), it can be expected that the adaptation selects a $\beta$ equal to or bigger than zero. For wind and self-noise, it is expected that $-1 \leq \beta < 0$. An observation that $\beta$ would tend to values of less than 0 indicates the presence of uncorrelated signals at the two microphones. Thus, one can also use $\beta$ to detect (1) wind noise and conditions where microphone self-noise dominates the input power to the microphones or (2) coherent signals that have a propagation speed much less than the speed of sound in the medium (such as coherent convected turbulence).

**Velocity sensing MEMS microphone**

As shown in the preceding section, using the pressure-difference approach to realize a first-order differential microphone array can be severely limited in SNR and sensitivity to microphone mismatch if the spacing of the measurement of the acoustic pressure field is small (< 10 mm). Since MEMS microphones are typically around 1 to 2 mm in size, it is clear that with present MEMS pressure microphone technology, that the SNR would not allow one to build a usable device whose overall size is much less than 10 mm. Of course one could resort to using porting with tubes to effectively sample the sound field at wider spacing. But, this would require a more difficult construction as well as potentially introduce tube resonance and matching issues. A potential solution to this problem would be to build a MEMS microphone device that responds directly to the acoustic particle velocity with a frequency response that is inherently flat. Ribbon microphones utilizing this transducer approach have been known for more than 80 years when they were popularized by RCA. Ribbon velocity microphones are still produced for professional. Fig. 4 shows a prototype of one scheme to realize a MEMS velocity microphone.

![Prototype velocity MEMS microphone array mechanical construction.](image)

**FIGURE 4.** Prototype velocity MEMS microphone array mechanical construction.

The multiple silicon “sails” are physically connected to obtain some local velocity averaging and for force addition in a force-feedback design. The angular rotation of the sails causes the displacement of a set of interlocking fingers that are used in a capacitive read-out scheme. The compliance of the restoring spring is set to so that the
microphone acts in the mass controlled region. Bernoulli’s equation can be used to compute the net force on the sails due to fluid flow past the sails. Another technique that was under investigation was to use force feedback to keep the sails vertical. Unfortunately work stopped on this project before the capacitive readout component and force feedback control was finished so no data was taken with this prototype.

The major practical issue for velocity microphones is their innate sensitivity to airflow. Commercial velocity microphones are surrounded by large windscreen structures to reduce this sensitivity. Small consumer electronics would not allow much in the way of a mechanical windscreen. However, since MEMS dimensions are naturally small in size there is the possibility of using the physics of the boundary layer due to air viscosity at the surface of a rigid device. Fig. 5 shows the tangential velocity of the boundary layer over the surface of the rigid planar surface. If the MEMS velocity microphone sail height is set to be less the boundary layer thickness, then it would be possible to use the buffering of the boundary layer to reduce the sensitivity to airflow. An important factor that works in the favor of using the boundary layer is that the boundary layer thickness is a function of inverse of the square root of the frequency. The thickness as a function of frequency can be shown to be approximately,

$$\delta(\omega) = \frac{2.1}{\sqrt{\omega}}$$

(15)

Figure 6 shows a plot of the boundary layer thickness as a function of frequency. Although the thickness is less than 1 mm, this dimension is much larger than most MEMS active sensor sizes. Thus, it should be possible to use the boundary layer as a natural windscreen. It is fortuitous that the boundary layer thickens at lower frequencies. Wind airflow is low frequency dominant thereby allowing us to attain more wind protection due to this effect. Other techniques can also be utilized to further reduce the influence of wind sensitivity using passive and active means.

![FIGURE 5. Tangential particle velocity close to the boundary surface where Ue is the free stream flow velocity.](image-url)
By combining the velocity microphone signal with an acoustic pressure-sensing microphone any first-order differential microphone can be realized. It is interesting to note that first-order directional microphones were made with this principal in the 1930’s. Care is needed to ensure that the two transducers are matched in magnitude and phase, but that could be easily handled by the internal ASICs used to form the differential beamformer output signal. One can even build a steerable first-order microphone by utilizing two velocity microphones rotated 90 degrees from each other to obtain the particle velocity in two orthogonal directions.

**SUMMARY**

It was shown that small directional microphones made from the differential pressure between two closely-space pressure microphones are inherently sensitive to self thermal noise and wind-induced noise as well as microphone matching. Small (>10 mm) dual element differential arrays are now practical due to the inherent matching of microphone amplitude and phase due to semiconductor manufacturing techniques but output SNR still remains an issue. For smaller microphone arrays that are on the size of 1 mm it will be necessary to utilize a microphone transducer that natively responds to the acoustic particle velocity. One possible design using silicon "sails" was shown. It was further shown that limiting the height of the velocity microphone to be well inside the viscous boundary layer could reduce the sensitivity to airflow for a velocity microphone. Finally, it was shown that the boundary layer had the desirable quality of being thicker at lower frequencies where wind airflow would be dominant.

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