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Autodirective Dual Microphone

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ABSTRACT
The paper describes Autodirective Dual Microphone (ADM) technology and its applications. ADM digital signal processing technology developed by Alango Ltd. is an adaptive beamforming technology that uses only two closely spaced omnidirectional sound pressure sensors. ADM technology provides the optimal and variable directivity in every frequency region. The adaptation time is very fast leading to very good improvement in signal-to-noise ratio in fast changing noisy environments. Contrary to regular directional microphones, an ADM technology based microphone is much less sensitive to wind noises and does not a proximity effect. Its DSP implementation is relatively simple requiring very modest computational resources.

1. BACKGROUND

First order, pressure-gradient, directional microphones may be built using either electronic or acoustic means. Figure **1** shows a schematic of electronic directional microphone

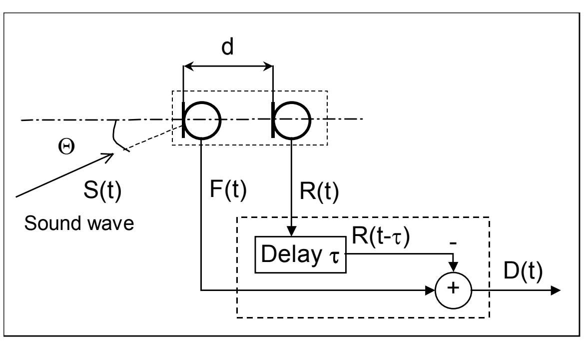

Figure 1. Electronic directional microphone

The microphone consists of two omnidirectional sound pressure sensors separated by a small distance d. The rear sensor signal *R(t)* is delayed by τ and subtracted from the front sensor signal *F(t)*. It may be shown that the output of such microphone is given by the following equation

$$Y(f,\Theta) = S(f)(1 - e^{-j2\pi f(\tau + T\cos\Theta)}) \tag{1}$$

where **Θ** is the angle of incidence relative to the microphone axis, f is frequency, *T=d/c* is sound propagation time between the sensors, *d* is the distance between the sensors and *c* is sound velocity. Taking the magnitude of Eq.1 yields

$$\left|Y(f,\Theta)\right| = 2\left|S(f)\sin\frac{2\pi f(\tau + T\cos\Theta)}{2}\right| \tag{2}$$

Assuming a relatively small distance between the sensors and a small delay ($fd/c \ll 1$ and $\tau \le T$)

$$|Y(f,\Theta)| = 2\pi f\,|S(f)(\tau + T\cos\Theta)| = 2\pi fT|S(f)|P(\Theta) \qquad (3)$$
$$P(\Theta) = |\tau/T + \cos\Theta|$$

where P(Θ) is the microphone polar pattern. Varying the delay τ (either electronically or acoustically) between **0** and T, it is possible to get different polar patterns of the constructed directional microphone. For example, τ=0 corresponds to the bi-directional (figure eight) pattern, τ=T to the cardioid pattern, τ=$0.5T$ to the super-cardioid pattern. Figure 2 shows these examples.

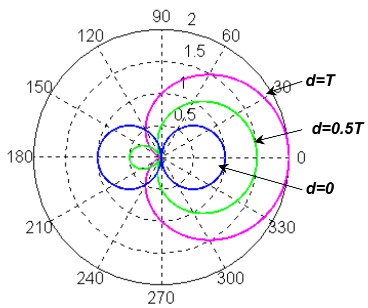

Figure 2. Polar patterns for different values of τ

Acoustic directional microphone is based on the same principle but implemented by acoustic means. In such a microphone the difference between sound pressure levels in two points is detected by exposing the microphone membrane via the front and rear microphone ports. The delay is usually achieved by using a special porous material between the rear port and the membrane.

From Eq.3 and Figure 2 it is seen that varying τ between 0 and T, it is possible to steer the null in the back plane between 90 and 270 degrees. The null cannot be moved to the front plane, and thus, the signal coming from front directions with θ between –90 and 90 degrees cannot be canceled.

In principal, it is possible to make the delay τ adjustable resulting in a microphone with variable polar pattern. The optimal choice of τ_{opt} depends on the acoustic conditions such as the room reverberation as well as the number, spectral content and direction of interfering signals. Some professional directional microphones allow selection of its polar pattern from a predefined set of patterns to match the situation at hand. However microphone polar pattern is set manually and it is constant during operation while the operating conditions may change dramatically. Examples include but not limited to hearing aids, cellular phones, reporters microphones, surveillance and security. In case when interferences have different spectral contents, the optimal delay τ_{opt} may be different if optimization is done in different frequency bands. It is thus obvious that a polar patter with a uniform delay over the whole frequency range does not allow achieving the maximal possible SNR improvement.

Another problem with conventional directional microphones is proximity effect when the microphone frequency response changes with distance between the sound source and the microphone. Gradient type microphones are also relatively sensitive to wind noise often requiring large windshields when used in open air.

2. ADM TECHNOLOGY

Autodirective Dual Microphone (ADM) is a patent pending technology developed by Alango Ltd. that solves all the above problems. The technology creates an adaptive gradient microphone that automatically provides optimal polar pattern that may be different for different frequencies. An ADM technology based microphone does not have a proximity effect and it is as low sensitive to wind noise as its constituting sound pressure sensors.

Figure 3 shows that ADM is an inherently digital technology.

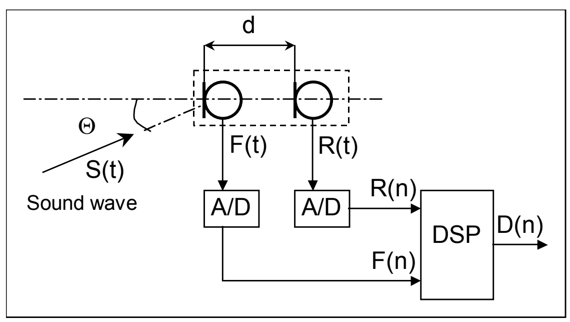

Figure 3. ADM microphone structure

The superior directivity and other virtues are achieved by digital signal processing of digitized versions of signals $F(n)$ and $R(n)$ acquired by the front and rear sensors respectively. The output signal $D(t)$ may be used directly or converted to analog form again by a digital-to-analog converter (not shown).

Figure 4 shows schematic of ADM algorithm. Signals $F(n)$ and $R(n)$ are each divided on M frequency bands by identical filter banks of bandpass IIR filters BPF 1…M .

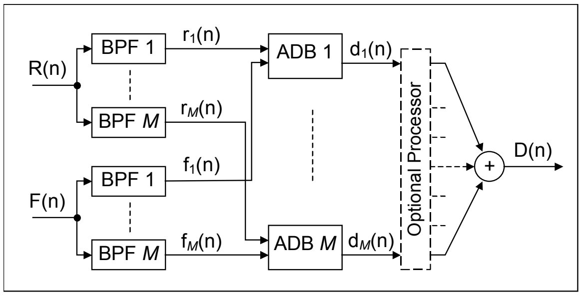

Figure 4. ADM algorithm schematic

The phase and amplitude characteristics of the filters are designed to provide a good reconstruction of the original signal when individual bands are combined. IIR filter bank is chosen in favor of FIR or FFT based approaches to provide the minimal signal delay. Pairs of corresponding bandpass filters $\{f_k(n), r_k(n)\}$ constitute inputs of identical Adaptive Directivity Blocks (ADB). Figure 5 shows the schematic of every ADB block.

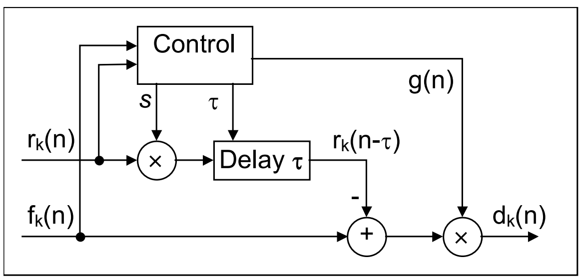

Figure 5. Schematic of Adaptive Directivity Block

First each ADB block compensates for possible different frequency responses of individual sound pressure sensors. This is done by multiplying the rear signal $r_k(n)$ by factor s computed according to the ratio of long term energies of $f_k(n)$ and $r_k(n)$.

ADB block k implements adaptive directivity in the frequency band k by varying the additional delay τ between the subband front and rear sensor signals. The control block is responsible for choosing τ_{opt} between **0** and T that provides the best signal-to-noise ratio. Interpolation procedure may be used to achieve subsample resolution if necessary. The implementation of optimization procedure for τ is very efficient. The optimal adaptation time is almost instantaneous taking about 5 milliseconds.

According to Eq.3 the output amplitude of ADB block depends on the frequency f and the delay τ . The output signal must hence be equalized to provide a constant frequency response. This is achieved by multiplying the output signal by gain $g(n)$ which depends on the frequency band and the current value of delay τ . Multiplication by gain $g(n)$ also serves to neutralize the proximity effect and reduce sensitivity to wind. This is done by limiting the output amplitude $d_k(n)$ to the amplitude of the front sensor signal $f_k(n)$.

Outputs of ADB blocks constitute a set of subband signals $\{d_k(n)\}$, each signal having an optimal directivity in its frequency band. Since every signal $d_k(n)$ is a full band signal, they may be easily combined by simple summation. Optional Processor block shown on Figure 4 may provide additional functionality that is frequency dependent. Examples include simple frequency equalization, dynamic multiband compression and stationary noise suppression.

3. ADM PERFORMANCE

Figure 6 shows theoretical (ideal) polar pattern off ADM technology based microphone as it had been measured in an anechoic chamber.

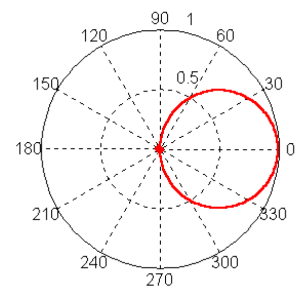

Figure 6. Theoretical polar pattern of ADM

Sound source located in the rear hemisphere (left part) is completely cancelled by a proper choice of τ . The right part Figure 6 corresponds to a bi-directional microphone as it provides the maximal directivity and maximal attenuation for sounds coming from the front hemisphere. Real performance may differ from the ideal due to multiple sound sources, strong reverberation and imperfect sound pressure sensors used to build the microphone. However, existing ADM prototypes demonstrate that very fast adaptation time (<5 milliseconds) combined with frequency selective optimization and using good quality sensors with smooth frequency response provide very good performance significantly exceeding performance of traditional directional microphones. Measurements show that in diffused ambient noise with no clear direction ADM technology provides about 3-4dB improvement in signal-to-noise ratio over an acoustic directional microphone. In better acoustic environments with more directional sounds, the advantage is generally more significant. The final sound quality provided by ADM technology is very good matching the sound quality provided by its constituting omnidirectional microphones.

4. ADM APPLICATIONS

An important advantage of ADM technology is that ADM microphone is easy to build as a standalone unit as well as a part of a portable device. This allows its application in any area where directional microphones are preferable.

Examples of possible applications of ADM technology include but not limited to:

- Cellular phones. ADM technology may provide significant advantage significantly improving the signal-to-noise ratio in noisy environments in both handset and speakerphone modes.
- Conference microphones. Good directivity and lack of proximity effect together with additional noise suppression and multiband signal compression make it ideal microphone suitable for experienced and novice speakers.
- Reporters and camcorder users may significantly benefit from ADM technology due to high directivity and low sensitivity to wind noise.
- Surveillance and security. ADM high directivity, easy integration into other devices and low sensitivity to wind noise are key factors for such applications.

5. CONCLUSION

ADM technology provides an alternative to acoustic directional microphones. Its high, optimal, self-adjusted directivity significantly improves signal-to-noise and direct-to-reflected sound ratios. Lack of proximity effect makes it ideal for using in conference and sound reinforcement systems. ADM is easy to integrate into different devices requiring no special acoustic design. ADM low sensitivity to wind makes it ideal for outer, “field” use.