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Close Talking Autodirective Dual Microphone

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ABSTRACT

The paper presents Close Talking mode of Autodirective Dual Microphone (ADM) technology developed by Alango Ltd. ADM is an adaptive beamforming technology having two operational modes. In Far Talk mode ADM provides optimal directivity for every frequency region such that sounds coming from the back plane are cancelled. In Close Talk mode all sounds originating outside a close proximity to the microphone are (theoretically) completely cancelled. ADM fast adaptation time leads to excellent noise cancellation in changing noisy environments. ADM technology has a low demand for placing, matching and distance between individual sensors. This simplifies its integration into mobile and other devises. ADM operational mode is defined by DSP algorithm easily switching according to situation.

1. BACKGROUND

Close talking (or noise canceling) microphone is essentially a directional microphone in which differential properties together with the proximity effects are used to attenuate distant and preserve close sounds. A directional microphone may be constructed either acoustically or electronically. Figure 1 shows a schematic of electronic directional microphone. Such microphone consists of two omni-directional microphones with matching characteristics. The rear microphone signal $R(t)$ is (optionally) delayed by τ and electronically subtracted from the front signal $F(t)$ producing the output signal $D(t)$. For our purposes this scheme may be regarded as an equivalent scheme for an acoustic directional microphone as well.

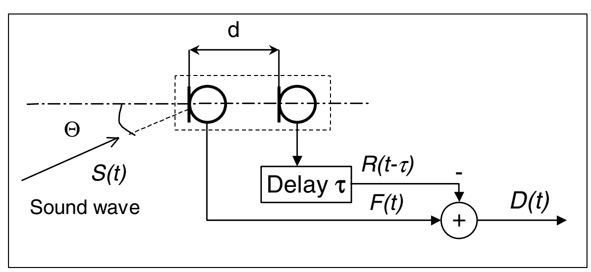

Figure 1 Electronic bi-directional microphone

In complex form the output of such microphone for a plane acoustic wave of frequency f, unit amplitude and incidence angle Θ is given as

$$S(t)=e^{j2\pi ft} \qquad T=d/V_s$$
$$D_{f,\Theta}(t)=e^{j2\pi ft}\left(1-q\cdot e^{-j2\pi f(\tau+T\cos\Theta)}\right) \tag{1}$$

where T is the sound propagation time between the front and rear microphones, V_s is the sound velocity, q is relative signal amplitude difference between the front and rear microphones. q depends on the distance to sound source, incidence angle and sensitivities match between the microphones. Equation (1) may be rewritten as

$$D_{f,\Theta}(t)=e^{j2\pi ft}(1-q)+ \\ +e^{j2\pi ft}\cdot q\cdot\left(1-e^{-j2\pi f(\tau+T\cos\Theta)}\right) \tag{2}$$

Sound pressure level is inverse proportional to distance, hence, for small d and distant sounds, the amplitudes may be regarded equal independently of the incidence angle Θ. Consequently $q \approx 1$ and the output is defined by the second term of (2) so that

$$D_{f,\Theta}(t)=e^{j2\pi ft}\left(1-e^{-j2\pi f(\tau+T\cos\Theta)}\right)$$

$$\left|D_{f,\Theta}(t)\right|=2\left|\sin\left(\pi f\left(\tau+T\cos\Theta\right)\right)\right| \tag{3}$$

Assuming a relatively small spacing between the microphones ($fd \ll V_s$) and $\tau\le T$ gives

$$\left|D_{f,\Theta}(t)\right|=2\pi f\left|\tau+T\cos\Theta\right|=\frac{2\pi f\,d\,P(\Theta)}{V_s} \tag{4}$$
$$P(\Theta)=\left|\tau/T+\cos\Theta\right|$$

where $P(\Theta)$ is the microphone polar pattern. Figure 2 shows examples of $P(\Theta)$ for bi-directional ($\tau=0$), cardioid ($\tau=T$) and super-cardioid ($\tau=0.5T$) types of directional microphones. From equations (3), (4) and Figure 2 it is seen that independent of the microphone polar pattern relative attenuation of front ($\Theta=0$), distant ($q \approx 1$) sounds is defined by equation (3) only.

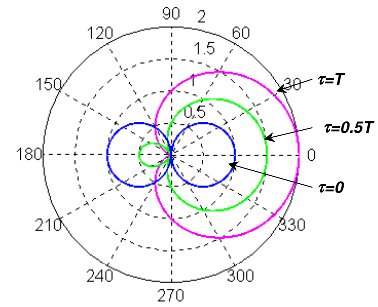

Figure 2 **Polar patterns for different values of τ**

Alternatively, for close, on axis sounds there is a large difference in sound pressure level on the front and rear microphones so that $q \ll 1$ and the output is defined by the first term of (2) as

$$D_{f,\Theta}(t)=e^{j2\pi ft} \qquad \left|D_{f,\Theta}(t)\right|=1 \tag{5}$$

Figure 3 shows additional attenuation of distant, on-axis ($\Theta=0|\pi$) sounds provided by a bi-directional microphone ($\tau=0$) as a function of frequency for different spacing between constituting microphones according to (3).

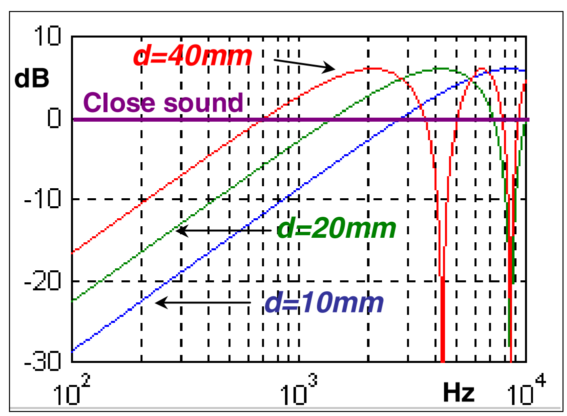

Figure 3 Far sound attenuation for different inter-microphone distances

Examining equation (3) and Figure 3 it is seen that, after some frequency, a directional microphone actually amplifies sounds coming from some directions. For bi-directional microphone and on-axis sounds this “crossover” frequency has the wavelength λ approximately equal to six times the distance between the microphones d. 10dB distant sounds attenuation is achieved with

$\lambda \quad 20d$. As such a directional microphone provides a practical solution for only small (up to 10mm) distance between constituting microphones (or acoustic ports for its acoustical equivalent). This fact renders using built-in, regular directional microphones (either acoustic or electronic) practically useless in such mobile applications as cellular phones, portable voice recorders and similar.

Close-talking Autodirective Dual Microphone technology solves the above problems by allowing polar pattern to vary gradually from forward-looking to backward-looking cardioid steering its null to any direction. This allows canceling distant sounds coming from all directions. Close sounds are still preserved due to the proximity effect.

2. PRINCIPALS OF ADM TECHNOLOGY FOR FAR TALK

Autodirective Dual Microphone (ADM) is a novel digital signal processing technology developed by Alango Ltd. ADM technology creates an adaptive gradient microphone that automatically changes its polar pattern to provide the best signal-to-noise ratio. ADM also resolves problems associated with conventional directional microphones: ADM is easy to build into any device, it does not have a proximity effect and it is much less sensitive to wind noise. This makes it a perfect solution for mobile and outdoor applications.

Figure 4 shows that ADM is an inherently digital technology. The superior directivity and other

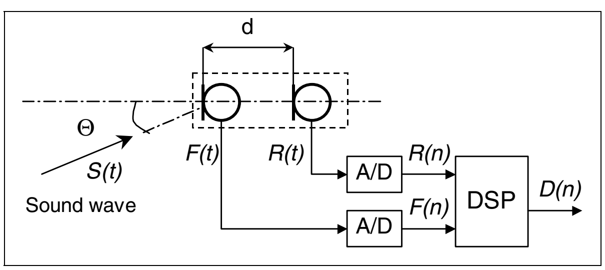

Figure 4 Basic schematic of ADM

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

Figure 5 shows that ADM is a subband technology.

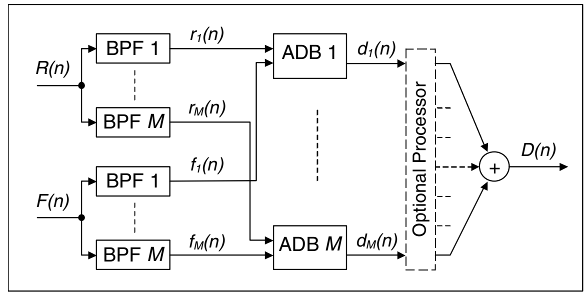

Figure 5 ADM is a subband technology

Signals *F(n)* and *R(n)* are first divided on frequency subbands by blocks of bandpass filters. 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 subband signals { $f_k(n)$, $r_k(n)$ } constitute inputs of identical Adaptive Directivity Blocks (ADB). Each subband is then processed independently to provide the best signal-to-noise ratio in it. The outputs of each ADB are then combined back into the full band signal. Optional processor may provide additional useful subband functions such as noise suppression, multiband compression and others.

Figure 6 shows simplified schematic of every ADB block.

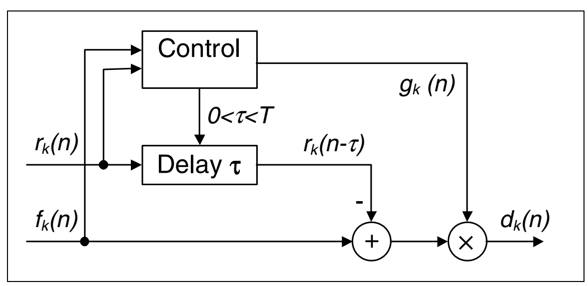

Figure 6 Simplified Adaptive Directivity Block

ADB implements adaptive directivity by varying the additional delay τ between the subband front and rear microphone signals. The control block is responsible for choosing τ between 0 and T that provides the best signal-to-noise ratio for the corresponding band. Interpolation procedure may be used to achieve sub-sample resolution if necessary.

According to equation (3) the output amplitude of ADB 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 $g_k(n)$ which depends on the frequency band and the current value of delay τ .

Figure 7 shows ADM prototype polar pattern measured in an anechoic chamber.

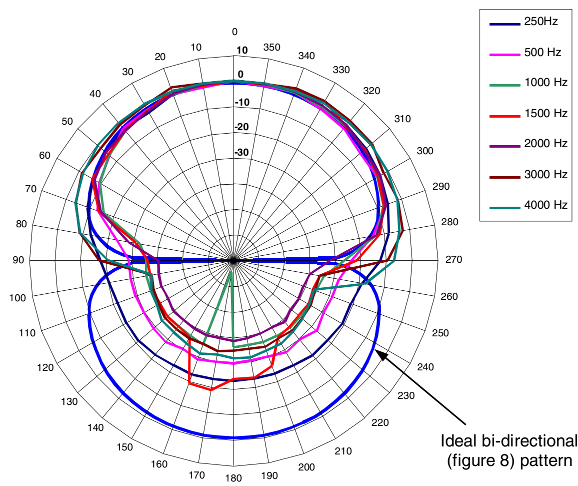

Figure 7 Measured ADM polar pattern

It is seen that the measured characteristics provide good correspondence with ADM theoretical pattern that is given by the front part of the bi-directional pattern.

3. PRINCIPALS OF ADM TECHNOLOGY FOR CLOSE TALK

Close talking (or noise canceling) microphones exploits proximity effect to boost close sounds relative to distant sounds attenuated according equation (3) and Figure 3. As such, all distant sounds are preferably attenuated irrespective of their direction of arrival. This is achieved by a small modification of ADM technology described above for far talk.

Ability to cancel front distant sounds is achieved when the null of the microphone polar pattern is allowed to be steered into the front plane ($-\pi/2 < \Theta < \pi/2$). According to equations (3), (4) this corresponds to introducing a negative delay τ that is obviously impossible. Practical solution may be achieved by inserting a constant delay T into the front microphone signal path and allowing τ to vary between 0 and $2T$. As before, T is the sound propagation time between the front and rear microphones. Figure 8 shows the modified schematic for Adaptive Directivity Block working in the close talk mode.

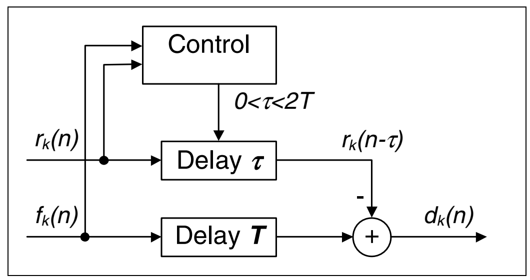

Figure 8 Adaptive Directivity Block for close talk

The difference between ADB schematic for close talk (Figure 8) compared to the one for far talk (Figure 6), is in the additional delay introduced into the front signal and removed equalization. Since the proximity effect is used (the output signal is defined by the front microphone), no equalization is necessary.

Ability to steer the null of the polar pattern into any direction independently in different frequency subbands provides significant improvements over a standard, fixed pattern noise canceling microphone:

- Cancellation distant on-axis sounds in the whole frequency range by steering null of the polar pattern into the corresponding direction;
- Improved noise cancellation of off-axis sounds;
- Possible larger distances between constituting microphones. This virtue significantly simplifies ADM integration into portable and other devices.

4. ADM APPLICATIONS

From application point of view ADM technology provides two significant advantages over conventional directional microphones:

- 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.
- ADM allows "software" switch between close and far talk modes. As such it may be used for devices used interchangeably in close and far talk modes;

By far the most mass-market application of ADM is mobile phones. Since many modern mobile phones have the speakerphone option, ADM is ideal for such application providing easy switch between the handset and speakerphone modes significantly reducing interfering noises and echoes.

Other applications include almost all usages of directional microphones such as but not limited to:

- Portable voice recorders;
- Camcorders;
- Conference microphones;
- Surveillance microphones;

ADM may be easily combined with other digital signal processing technologies using the same subband scheme. In Alango implementation it is combined with stationary noise suppression and multiband compression followed by Automatic Gain Control. As such ADM may deliver much better quality and flexibility than any standard directional microphone.

Since omnidirectional microphones are generally much cheaper than unidirectional microphones, the overall cost of ADM may be comparable with unidirectional microphones.