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Long Range Noise Canceling Microphone

Alexander A. Goldin
Alango Ltd., Haifa, 39100, Israel

ABSTRACT

The paper presents Long Range Noise Canceling (LRNC) microphone array technology developed in Alango Ltd. LRNC is a digital signal processing technology utilizing near field signals from two unidirectional or four omnidirectional microphones. It allows differentiation between user's voice originating in a closed region in front of LRNC microphone and other sounds that are effectively blocked. The pick up range of LRNC microphone may be as large as 70cm in front of the microphone and, if necessary, may be easily reduced by changing corresponding software parameters. This unique property makes LRNC microphone attractive for a variety of voice applications where distant sounds, noises or acoustic echoes must be blocked.

1. DISTANCE LIMITATIONS OF NOISE CANCELING MICROPHONES

1.1. Conventional Noise Canceling Microphones

Conventional noise canceling microphone is simply a bi-directional microphone in which differential properties together with near field sound properties are used to attenuate distant and preserve close sounds. Directional microphone may be constructed either acoustically or electronically. Figure 1 shows schematic of an electronic directional microphone. Such microphone consists of two omni-directional microphones with matching frequency responses. The rear microphone signal $R(t)$ is (optionally) delayed by τ and subtracted from the front signal $F(t)$ producing the output signal $D(t)$. For our analysis purposes this scheme serves as the equivalent scheme for an acoustic directional microphone as well.

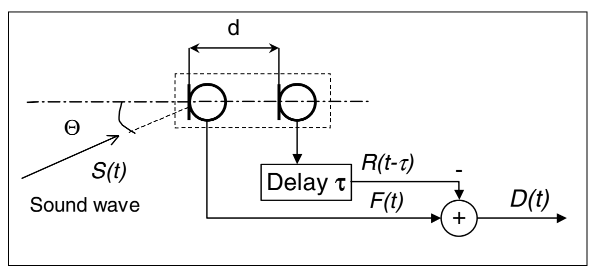

Figure 1 Electronic bi-directional microphone

The output of such microphone for an acoustic wave of frequency f 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 the signal amplitude is normalized to generate unit amplitude on the front microphone, T is the sound propagation time between the front and rear microphones, V_s is the sound velocity, q is the relative signal amplitude difference between the rear and front microphones. Factor q depends on the distance to the sound source, spacing between the microphones, incidence angle and sensitivities match between the microphones. With amplitude-normalized signal on the front microphone, the output signal amplitude may be regarded as the microphone gain relative to the omnidirectional microphone and it is given as

$$G_{\tau,\Theta}\left(f,q\right)=\left|1-q\cdot e^{-j2\pi f\left(\tau+T\cos\Theta\right)}\right|$$

For on-axis ($\Theta=0\,|\,\pi$) sounds and a bi-directional microphone ($\tau=0$) the gain formula is simplified as

$$G\left(f,q\right)=\left|1-q\cdot e^{-j2\pi fT}\right|$$

For matching microphones and on-axis sounds q is a function of distance x between the sound source and the front microphone.

$$q\left(x,d\right)=\frac{x}{x+d} \tag{2}$$

As such, performance on noise canceling microphone given as relative attenuation of distant sounds (q=1) to close sounds (q<1) is given as

$$g\left(f,q\left(x,d\right)\right)=\frac{\left|1-e^{-j2\pi fT}\right|}{\left|1-q\left(x,d\right)\cdot e^{-j2\pi fT}\right|} \tag{3}$$

Figure 2 illustrates the best possible performance (x=0, q=0) of a bi-directional microphone ($\tau=0$) as a function of frequency for different spacing between constituting microphones according to equation (3). It is seen that, after some frequency, a conventional directional microphone may actually amplify distant sounds relative to close sounds. For a bi-directional microphone and on-axis sounds this “crossover” frequency has the wavelength λ approximately equal to six times the spacing between the microphones d. A reasonable, -10dB attenuation of distant sounds is achieved with $\lambda \quad 20d$. For these reasons the maximal distance between the microphones (or acoustical ports) for a noise canceling microphone is limited to 20mm for most practical applications.

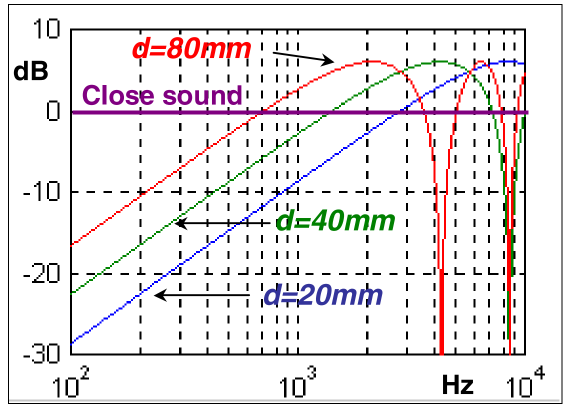

Figure 2 Far sound attenuation for different inter-microphone spacings

Performance demonstrated on Figure 2 is purely theoretical because there is always some distance between the sound source and the front microphone so that q is never equal to zero. Equation (3) also predicts

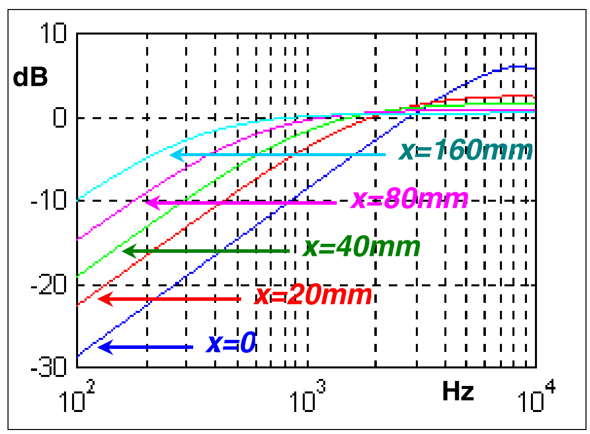

Figure 3 Relative attenuation of distant sounds for spacing d=20mm and different distances x

that there is performance degradation with an increase of distance between the sound source and the front microphone. Figure 3 shows relative attenuation of distant sounds for spacing d=20mm and different distances between the sound sources and the front microphone.

From equation (3), Figure 2 and Figure 3 it follows that noise canceling microphones provide a practical solution for:

- Only small, up to 20mm spacing between constituting microphones (or acoustic ports for its acoustical equivalent).

- Only small, about 20mm distance between the microphone and the sound source (mouth).

1.2. Autodirective Dual Microphone (ADM)

Autodirective Dual Microphone (ADM) technology [1], [2] provides an alternative to conventional directional and noise canceling microphones. As shown on Figure 4 ADM is an inherently digital technology.

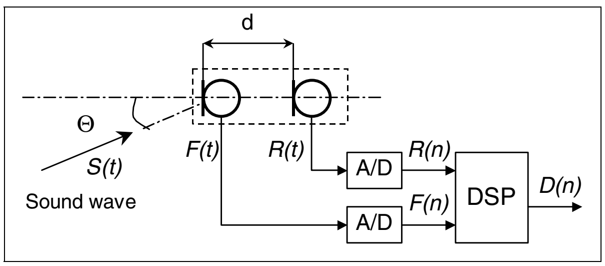

Figure 4 ADM and LRNC microphones

ADM advanced properties are achieved by digital signal processing of discrete time versions *F(n)*¸ *R(n)* of signals 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. Signals *F(n)* and *R(n)* are first divided on frequency subbands by blocks of bandpass filters.

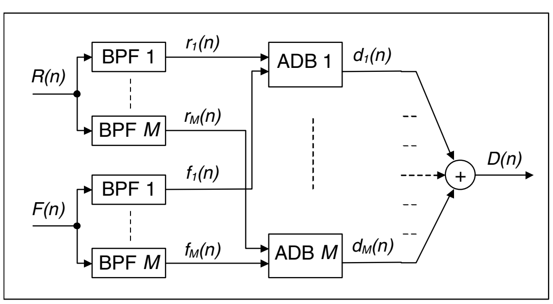

Figure 5 Subband structure of ADM processing

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 and the outputs of each ADB are then combined back into the full band signal. Figure 6 shows a simplified schematic of ADB block

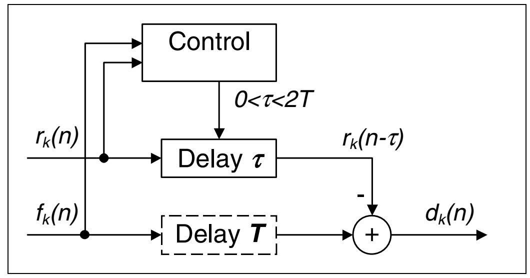

Figure 6 Adaptive Directivity Block of ADM

ADB implements adaptive directivity by varying the delay τ between the subband front and rear microphone signals. The control block is responsible for choosing τ providing the best signal-to-noise ratio for the corresponding band. As shown in [1], [2], this is equivalent to minimization of the output signal energy.

Autodirective dual microphone can work in two modes: far-talk and close-talk. In the far-talk mode delay τ is varied between 0 and T steering the null of the microphone polar pattern in the back hemisphere. In the close talk mode additional delay T is added into the front subband signal and delay τ is varied between 0 and $2T$. This allows steering the null of the microphone polar pattern in every direction canceling distant sounds irrespective of their direction of arrival. Close sounds are preserved for any value of τ due to amplitude difference on the front and rear microphones.

Close talking ADM provides multiple advantages over a standard, fixed pattern noise canceling microphone:

- Cancellation of distant, noise sounds in the whole frequency range irrespective of the direction of arrival by steering the null of the polar pattern into the corresponding direction;
- Larger possible spacing between constituting microphones. This significantly simplifies ADM integration into portable and other devices and increases the microphone operational distance.

In practice, the best performance of ADM is achieved with spacing between the constituting microphones up to 50mm range. This limits the operational distance between the microphone and the sound source to about 60-80mm. Larger spacing leads to a degradation in noise canceling properties due to reduced coherency of signals from the constituting microphones in practical, reverberant environments.

2. LONG RANGE NOISE CANCELING (LRNC) MICROPHONE

2.1. Principals

Long Range Noise Canceling (LRNC) microphone is an end-fire microphone array intended for voice applications. The technology utilizes near field properties of sound and non-linear, digital signal processing algorithm to effectively limit the microphone range of sensitivity and block sounds outside that range. Figure 7 shows two microphone LRNC array, the Bubble Of Sensitivity (BOS) in front of it, analog-to-digital converters and the DSP.

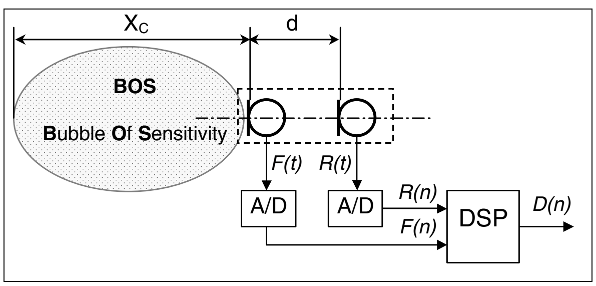

Figure 7 LRNC microphone

Similar to ADM, LRNC is a subband technology. The structure of LRNC microphone shown on Figure 8 is actually identical to ADM microphone shown on Figure 5.

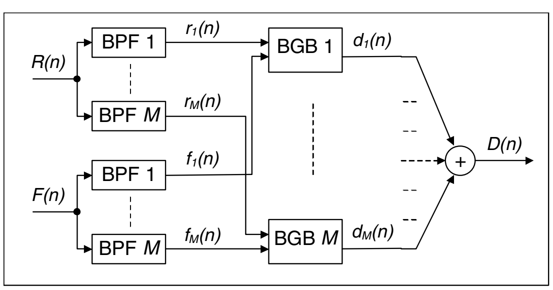

Figure 8 Structure of LRNC microphone

However, its processing blocks called Band Gain Blocks (BGB) are different. Figure 9 illustrates the BGB structure.

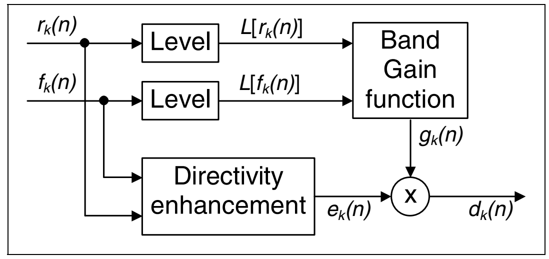

Figure 9 Band Gain Block structure

Based on the front/rear subband signal levels $L[r_k(n)]$, $L[f_k(n)]$, BGB computes a band gain that is applied to the (optionally enhanced) subband signal $e_k(n)$. The Directivity Enhancement block may implement ADM or other type of processing improving the signal-to-noise ratio. In the simplest case or when no enhancement is possible, the front subband signal $f_k(n)$ can be taken as $e_k(n)$ without any processing. Understanding of BGB logic is facilitated when noticed that in a free field:

- Close sounds in front of LRNC microphone (user's voice) produce greater signal level on the front microphone;
- Close sounds from the rear hemisphere (noises) produce larger signal levels on the rear microphone;
- Far sounds (noises) produce equal signal levels on the front and rear microphones independent on the direction of arrival;

In a free field (no reverberations) and a positive signal-to-noise ration (signal, when present, is stronger than noise), a simple logic may block unwanted noises out. Each BGB simply passes or attenuates the bandpass signal $e_k(n)$ depending on the ratio of levels $R_k(n)$

$$R_k(n) = L[r_k(n)] / L[f_k(n)]$$

For example, the following logic will effectively block sounds that are father away than the critical distance x_C (see Figure 7)

$$g'_k(n) = \begin{cases} 1, & R(n) \le R_C \\ g_{\min}, & R(n) > R_C \end{cases} \tag{4}$$

where R_C is the critical ratio corresponding to the critical distance and computed as

$$R_C = \frac{x_C}{x_C + d} \tag{5}$$

To avoid voice distortions caused by often gain switching when the sound source is at the critical distance x_C ($R \quad R_C$), the gain may be smoothed as

$$g_k(n) = \alpha \cdot g'_k(n) + (1-\alpha) \cdot g_k(n-1), \quad 0 < \alpha < 1$$

For far signals, ratios $R_k(n)$ fluctuate around unity so that values R_C close to unity cannot provide stable differentiation between close and far sounds. The amplitude of the fluctuations depends on the reverberations and the spacing between the constituting microphones. Inevitable microphones mismatch also prevents using values R_C close to unity. The lower R_C, the more reliable is the differentiation but smaller the critical distance x_C. It was experimentally found that for two unidirectional microphones in a typical office environment a reliable detection is achieved for $R_C < 0.9$.

From equation (3) it follows that for a fixed critical ratio R_C, the critical distance x_C is directly proportional to the spacing between the microphones d:

$$x_C = \frac{d \cdot R_C}{1 - R_C}$$

As such, it would have been beneficial to increase the distance if larger operating distance is required. Unfortunately, in practical, reverberant environments the maximal spacing between the microphones is limited. Large spacing leads to reduced coherency between the microphone signals and, as a result, larger fluctuations in the ratios $R_k(n)$. This requires using smaller critical ratios R_C eliminating the benefits of the increased spacing d. The optimal spacing for two unidirectional microphones was experimentally found to be about 80mm. This, together with the value for the critical ratio $R_C = 0.9$, gives the maximal critical distance as $x_C = 70\text{cm}$.

Fast and reliable decision on attenuating a frequency band when no close sound is detected allows blocking distant sounds. When a band is open due to a detected close sound in that band, the recorded signal includes the main, close signal mixed with far signals (noises). However, when the signal-to-noise ratio is sufficiently positive, decisions are fast and reliable and the bands are narrower than the corresponding frequency resolution of the human ear, the noises are effectively masked by the main signal. The requirement for the signal-to-noise ration is not very demanding. In general exceeding the noise by just 6dB (two times in a band amplitude) provides sufficient masking. Assuming 30cm operating distance, the normal speech level is 74dB SPL on the microphone. As such, normal operation is achieved with up to 68dB SPL of noise level that far exceeds environmental noise in normal office or home environments.

2.2. Using ADM technology in LRNC microphone

As it was mentioned, operation of LRNC microphone relies on a positive signal-to-noise ratio. As such using unidirectional microphones provides significant benefits by significant attenuation of noisy sounds coming from directions different from the front one. However, providing a good match of unidirectional microphones is rather difficult and expensive for mass-market applications. The problem is that not only front sensitivities but polar patterns have to match each other as well. Otherwise, a far sound coming from some direction may generate larger amplitude on the first microphone and, consequently, be confused with the front, close signal. Using two Autodirective Dual Microphones as building blocks for LRNC microphone array provides improved directional characteristics as well as a solution for microphones matching problem.

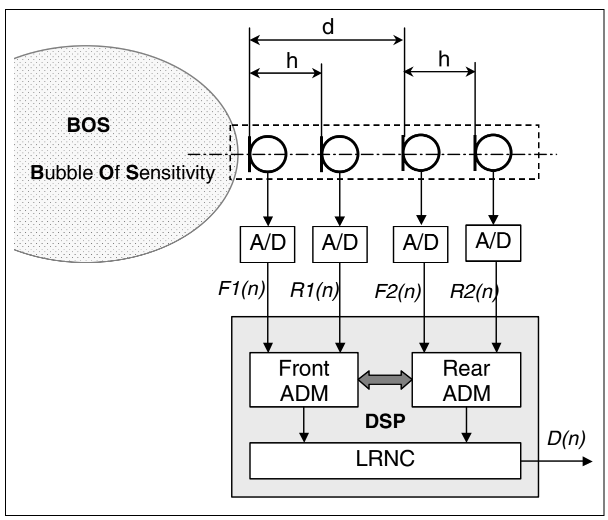

Figure 10 LRNC microphone using ADM technology

The idea is illustrated on Figure 10. The microphone is composed of four omnidirectional microphones. The first and second pairs of microphones are used to build the front and rear Autodirective Dual Microphones according to the technology described in [1] and outlined in Section 1.2. Synchronizing the processing between the rear and front subband signals of each pair ensures the same instantaneous polar patterns for each ADM. Outputs of the two ADM microphones are used as inputs for LRNC algorithm described above.

The optimal distance h between the microphones of each pair is 40mm. Due to better directional characteristics and matching of each pair, the spacing between the pairs d may be increased to 100mm potentially allowing larger operational range or working with lower signal-to-noise ratios.

3. CONCLUSIONS

In a far field conventional directional microphones attenuate unwanted sounds according to their polar pattern. Polar pattern defines the relative microphone gain as a function of the angle of incidence. Within a specific angle of incidence, sounds attenuate according to the inverse distance low. For many applications such attenuation is not enough. For example, when a desktop microphone is used for hand-free (e.g. VoIP communications), sound produced by loudspeaker s together with its reflections is picked up by the microphone and transmitted to the far side of the connection as acoustic echo. What would be needed is the ability to cancel sounds originating outside of a closed region in front of the microphone.

Noise canceling microphones provide only a limited solution. Due to a necessary small spacing between acoustical ports or constituting microphones, advantages of conventional noise canceling microphones over omnidirectional microphones disappear very quickly with increase of the distances (see equation (3) and Figure 3). As such, noise canceling microphones may be used for close-talking applications only such as communication headsets and hand-held microphones.

Close-talking Autodirective Dual Microphone permits larger spacing between the individual microphones. As a result, its operational distance predicted by equation (3) is increased allowing it to be used effectively in mobile phones, radios and voice recorders. However, it is still too small to provide a solution for desktop, conference and other "hands-free" microphones.

Long Range Noise Canceling microphone delivers a practical solution for a variety of voice applications where the desired microphone pick-up area does not exceed 70cm in front of the microphone. Its desktop version allows full duplex, hands-free voice communication over Internet. It provides significant benefits for video and voice conferencing systems where each participant in a room has its own microphone. With LRNC microphone each microphone picks up only his owner voice blocking all the neighbors. Conference microphones and microphones for public announcement systems may be an attractive application for LRNC technology due to elimination of an acoustic feedback and automatic "muting" when the user increases the distance to the microphone.

4. REFERENCES

[1] Alexander Goldin, "Autodirective Dual Microphone", presented at the 114th Convention of Audio Engineering Society.

[2] Alexander Goldin, "Close-talking Autodirective Dual Microphone", presented at the 116th Convention of Audio Engineering Society.