Small Microphone Array Design and Processing for Speech Recognition of Vocal Orders in Restaurants

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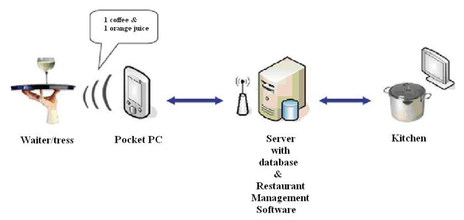
Summary

Veovox is a project led by a swiss company *Veovox*® in collaboration with swiss research institutes whose purpose is to market an order-taking device, enabling a waiter in a restaurant to take orders by voice. With this device, the waiter only needs to pronounce the order to a personal digital assistant (PDA), which directly translates the voice order into text through an integrated voice recognition engine. The text is then submitted to the restaurant management software solution for further processing. We present here the developed audio processing system of a such device developed by the LEMA. In the context of a more or less noisy restaurant composed of distributed parasitic speakers and diffuse noise (cocktail noise), the developed audio processing consists in enhancing the speech of the waiter from the recorded audio signal in order to provide a high quality signal to the recognition engine.

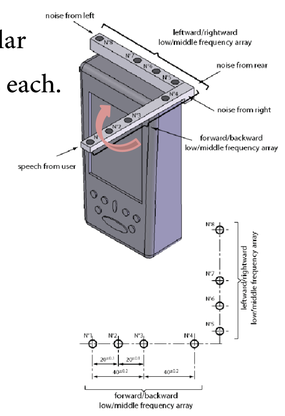


Array Design and Signal Processing

1. Array Design

Eight microphones which ***steer in four different directions*** : forward, backward, left and right based on differential beamforming theory.

The global array consists in two perpendicular and orientable line-arrays of 4 microphones each.

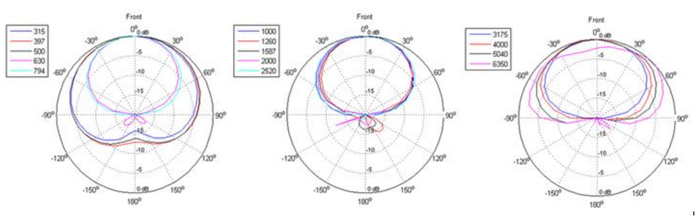


Three sub-arrays per line-array :

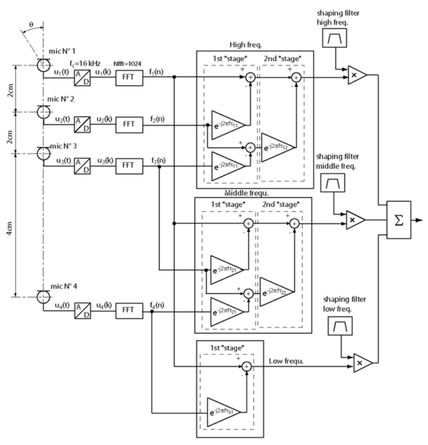
- 3 mic, d = 2 cm 1.2 - 6 kHz
- 3 mic, d = 4 cm 600 - 1200 Hz
- 2 mic, d = 8 cm 300 – 600 Hz

For each sub-array :

- If 2 mic : 1st order array (cardioid)
- If 3 mic : 2nd order differential array (cardioid + hypercardioid)



Directivity diagram of each sub-array.



Block diagram of each line array -> four spectra corresponding to each direction : X_{frwd} (towards the speaker), X_{bckwd}, X_{left}, X_{right}.

2. Noise Reduction

In order to clean the signal, a ***statistical noise reduction*** is done on each channel using the spectral substraction technique. In the frequency (ω) domain and assuming the observation Y is composed of the speech X and an additive noise N, the magnitude of speech can be estimated by the formula :

$$|X(\omega)| = |Y(\omega)| - |\tilde{N}(\omega)|$$

Where $\tilde{N}(\omega)$ is an estimation of $N(\omega)$.

3. Spatial Masking

This step consists in ***enhancing the separation between punctual sources*** distributed in a room (like parasitic speakers).

$$X_{frwd} = \begin{cases} \boldsymbol{X}_{frwd} & if\ X_{frwd} \geq \alpha \times \max(X_{frwd}, X_{bckwd}, X_{left}, X_{right}) \\ 0 & else \end{cases}$$

The comparison of the four channels in the frequency domain allows the design of adaptive filters, one per channel, which, by spectral subtraction, decrease the signals from other directions.

4. Cube Coherence Masking

In order to ***reduce distorsion*** due to previous processes, a cube coherence masking which consists in only keeping coherent frequency bins through all the process is applied. X_{raw} : spectrum of the raw signal taken on one arbitrary microphone.

$$\left.\begin{aligned} C_1 &= X_{raw}.\overline{X_{raw}} \\ C_2 &= X_{frwd}.\overline{X_{frwd}} \\ C_3 &= X_{frwd}.\overline{X_{raw}} \end{aligned}\right\} \quad X_{frwd} = \boldsymbol{X}_{frwd} \cdot \left(\frac{|C_3|^2}{C_1.C_2}\right)^3$$

5. VAD

Automatic speech recognition systems generally need an input signal where ***all what is non speech is eliminated***. This is the goal of Voice Activity Detection (VAD) algorithm.

D : spectral distance between the variance of the tested signal σ_y^2 and the variance of noise σ_n^2 .

$$D = 10\log\frac{\sigma_y^2}{\sigma_n^2}$$

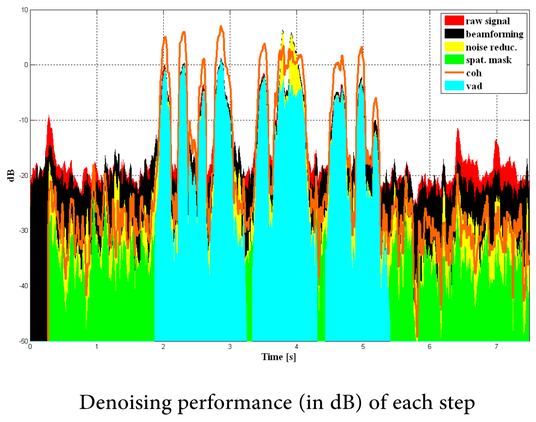
H : a boolean variable which is equal to 1 if speech is present and 0 otherwise.

$$H = \begin{cases} 1 & if \quad D > \lambda \\ 0 & else \end{cases}$$

Where λ is a threshold automatically determined or empirically chosen.

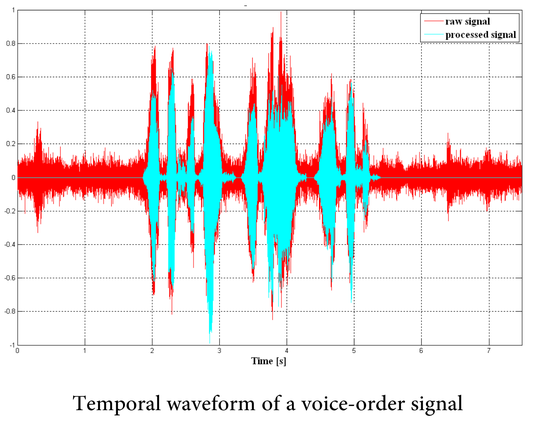
The audio processing system of LEMA has been successfully tested in different environments (from gastronomical restaurant to very noisy cafeterias with music).

Conclusion



Denoising performance (in dB) of each step

This algorithm has been developed in the context of **real time audio processing**. A first version has been developed on Matlab which has been tranposed on Simulink and then implemented on DSP. Sensibility studies are in progress but first results of our experimental tests in different real situations are very promising : we can see on both opposite figures the comparison between the raw and the processed signal (in a Leq and in a temporal waveform point of view). An attenuation of 15 dB is obtained with a very low distorsion on the speech signal giving a very good score in speech recognition. Further work consists in reducing both the number of microphones and the needed time to well estimate noise spectral properties (approximatively 500 ms at 16 kHz today). A patent is pending.



Temporal waveform of a voice-order signal in a cafeteria (EHL) before and after processing.

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