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AN APPROACH TO AMBIENT NOISE CANCELLATION FOR MOBILE COMMUNICATION

N. R. Raajan
Y. Venkatramani
T. R. Sivaramakrishnan

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N.R.Raajan
ECE/School of EEE,
SASTRA University,
Thanjore,
Tamil Nadu,
India.
nrraajan@ece.sastra.edu

Y.Venkatramani
Principal,
Saranathan Collegeof Engineering,
Trichy,
Tamil Nadu,
India.
principal@saranathan.ac.in

T.R.Sivaramakrishnan
Dean Research,
ECE / SEEE,
SASTRA University,
Thanjore,
Tamil Nadu,
India.
deantrs@sastra.edu

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Abstract

A great problem in speech processing is to represent the shape and characteristics of the vocal tracts. This task is normally done by using an acoustics tube model, based on the calculation of the area function would be performed. We will show that these models have good performance in experiments. A Mathematical model of Vocal fold has been obtained as part of new approach for Ambient noise cancelation.

1. Introduction

The capability of speech communication is essential for the interaction between human beings. In the process of speech production, several processes are involved simultaneously. One of these processes results in the production of the source sound for speech. This process is called phonation and is performed by the vocal folds. This source sound, or voice, passes the vocal tract, consisting of the air channels between the vocal folds on the one end and the lips and the nostrils on the other end. The geometrical configuration of this vocal tract can be adapted by movements of the articulators. The configuration of the articulators determines the resonance characteristics of the vocal tract. In this way, the vocal tract acts as an adjustable acoustic converter that converts the source sound to the desired speech sound.

The success of a Ambient noise processing method depends on its ability to characterize and model the noise process, and to use the noise characteristics advantageously to differentiate the signal from the noise.

Depending on its source, a noise can be classified into a number of categories, indicating the broad physical nature of the noise, as follows:
1) Acoustic noise,
2) Thermal noise and shot noise,
3) Electromagnetic noise
4) Electrostatic noise
5) Channel distortions, echo and fading
6) Processing noise .

2. Ambient Noise in mobile communication

The Ambient noise depends on the speed and the sound isolation between the air and the compartment while the acoustic feedback, the number of the Persons in the area, the sound absorption of the interior and so on. Thus it is clear that an efficient suppression of these disturbances is possible only by applying an adaptive system. The most popular system against the sound and all ambient noises has been known for more than 25 years [1] It requires multiple reference microphones and a complicated processing technique. The acoustic noise and echo cancellation problem is also widely investigated [2] but most of the results are valid for low noise environments like conference rooms or offices. An entirely new approach to handle all sources of disturbances was proposed in [3].

The system developed in [3] incorporates also the surround audio and is called an integrated system. This system is very efficient and its only weak point is the very complicated adaptive filter (non-recursive of order 256) for suppression of the engine noise. In this paper a new adaptive system for this purpose is proposed and its performance is investigated. This system may be incorporated in the integrated system [3] or could be used in some simpler implementations including also acoustic Noise echo suppression. Also it is well known that two of most frequently applied algorithms for noise cancelation [4] are normalized least mean squares (NLMS) [5] and recursive least squares (RLS) [6, 7] algorithms. Thus, it is clear that the choice of the adaptive algorithm to be applied is always a tradeoff between computational complexity and fast convergence. In the present work we propose a new adaptive algorithm with averaging applied for

Ambient noise cancelation. The extensive experiments conducted reveal its robustness fast convergence and at the same time low level computational complexity.

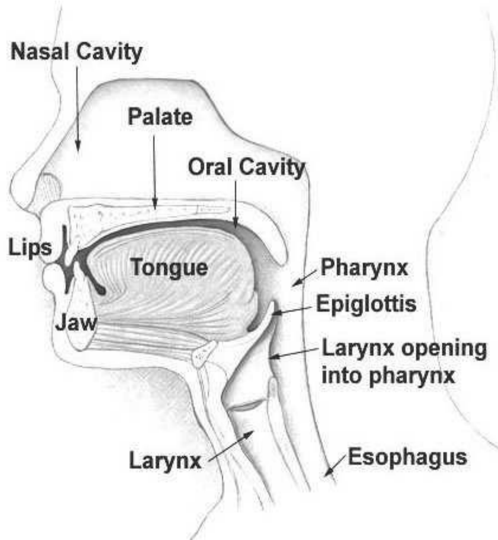

Figure 1. Anatomy of vocal fold

A novel approach to discord the Ambient noise with respect voice is archived by formation of vocal cord (Mathematical model) will eliminate the ambient noise.

3. Voicing

One of the features which has bothered researchers in the area of speech synthesis in the past has been voicing. We discuss this here because it is a good example of how failure to understand the differences between abstract and physical modeling can lead to disproportionate problems (Keating 1984). The difficulty has arisen because of the nonlinearity of the correlation between the cognitive phonological voicing and how the feature is rendered phonetically. Phonological voicing is a distinctive feature that is, it is a parameter of phonological segments the presence or absence of which is able to change one underlying segment into another. For example, the English alveolar stop /d/ is [+voice] (has voicing) and differs on this feature from the alveolar stop / t / which is [-voice] (does not have voicing). Like all phonological distinctive features, the representation is binary, meaning in this case that [voice] is either present or absent in any one segment.

The most frequent phonetic parameter to correlate with phonological voicing is vocal cord vibration the vocal cords usually vibrate when the underlying plan is to produce a [+voice] sound, but usually do not when the underlying plan is to produce a [-voice] sound.

Many synthesis models assume constant voicing vocal-cord vibration, but it is quite clear that the binary distinction of vocal-cord vibration vs. no vocal-cord vibration is not accurate. Vocal-cord vibration can begin abruptly (as when there is a glottal stop onset to make this possible singers regularly do this), gradually (the usual case), or at some point during the phone, although it may be phonologically voiced. Similarly for phonologically voiceless segments, it is certainly not the case that on every occasion there is no vocal-cord vibration present at some point during the phone. Figure gives some idea of the range of possibilities. We know of no model which sets out the conditions under which these variants occur.

These examples serve once again to underline a repeating theme in this book: phonological characterizations of segments should not be read as though they were phonetic, and sets of acoustical features should not be given one-to-one correlation with phonological features. More often than not the correlation is not linear nor, apparently, consistent- though it may yet turn out to be consistent in some respects. Phonology and phonetics cannot be linked simply by using phonological terms within the phonetic domain such as the common transfer of the term voicing between the two levels. Abstract voicing is very different from physical voicing, which is why we consistently use different terms for the two. The basis of the terminology is different for the two levels; and it is bad science to equate the two so directly.

4. The Human Vocal Apparatus (Modeling)

Figure 2 shows a representation of the mid sagittal section of the human vocal tract due to Coker [8]. In this model, the cross-sectional area of the oral cavity A(x), from the glottis, x = 0, to the lips, x = L, is determined by five parameters: a , tongue body height; b , anterior/posterior position of the tongue body; c , tongue tip height; d , mouth opening; and e , pharyngeal opening. In addition, a sixth parameter, f, is used to additively alter the nominal 17-cm vocal tract length. The articulatory vector is (a, b,.. . , f).

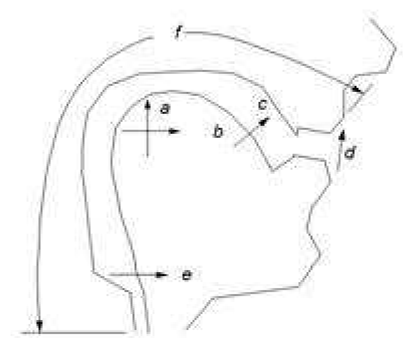

Figure 2. Articulatory model

The vocal tract model has three components: an oral cavity, a glottal source, and acoustic impedance at the lips. We shall consider them singly first and then in combination. As is commonly done, we assume that the behavior of the oral cavity is that of a lossless acoustic tube of slowly varying (in time and space) cross-sectional area, A(x), in which plane waves propagate in one dimension (see Fig. 2). Sondhi [11] and

Portnoff [9] have shown that under these assumptions, the pressure, p(x, t), and volume velocity, u(x, t), satisfy which express Newton's law and conservation of mass, respectively.

In is the equilibrium density of the air in the tube and c is the corresponding velocity of sound.

.

$$-\frac{\partial P}{\partial x} = \frac{\rho}{A(x,t)}\frac{\partial u}{\partial t} \qquad (1)$$

$$-\frac{\partial u}{\partial x} = \frac{A(x,t)}{\rho c^2}\frac{\partial P}{\partial t} \qquad (2)$$

Differentiating (1) and (2) with respect to time and space, respectively, and then eliminating the mixed partials, we get the well-known Webster equation [12] for pressure, The eigen values are taken as formant frequencies. We elect to use the Webster equation (in volume velocity) to compute a sinusoidal steady-state transfer function for the acoustic tube including the effects of thermal, viscous, and wall losses.

$$\xi(x,t) = e^{-\frac{1}{2}\frac{(b-\sqrt{b^2-4*k(x)*M})t}{M}} + e^{-\frac{1}{2}\frac{(b+\sqrt{b^2-4*k(x)*M})t}{M}} + \frac{1}{\sqrt{b^2-4*k(x)*M)}}
\left(\begin{array}{l}
\left(\left(\int P(x,t)*e^{-\frac{1}{2}\frac{(-b-\sqrt{b^2-4*k(x)*M})t}{M}}dt\right)e^{\frac{(\sqrt{b^2-4*k(x)*M})t}{M}} \right. \\
\left. *\left(\int P(x,t)*e^{\frac{1}{2}\frac{(b+\sqrt{b^2-4*k(x)*M})t}{M}}dt\right)\right) \\
*e^{-\frac{1}{2}\frac{(b+\sqrt{b^2-4*k(x)*M})t}{M}}
\end{array}\right)$$

Where P – pressure, K (x) – Damping coefficient, M – M ass of speech, ξ(x, t) – resultant Value.

5. Voice and noise

A reasonably good voice can be achieved by pressing air from the esophagus into the pharynx. The passing air starts the mucosa at the entrance of the esophagus to vibrate leading to esophageal voice production.

Table 1: Properties of the glottal waves(normal phonation)

	Female	Male
Fundamental frequency F_0 (Hz)	207	119
	0.14	0.23
Glottal peak flow	0.26	0.39

Table 2: configuration of lips and mouths

configuration	Thickness lip	Length lip
Basic	0.25	7
Long lip	0.25	9
Short lip	0.25	5
Higher opening	0.25	7
Tapered	0.25 (bottom)-0.125 (free tip)	7
Thin lip	0.125	7

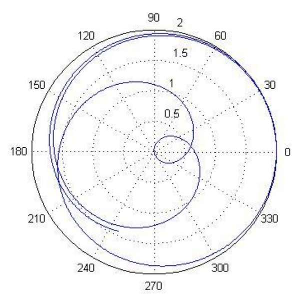

Figure 4. The acoustic tube model of the vocal tract with basic damping efficiency

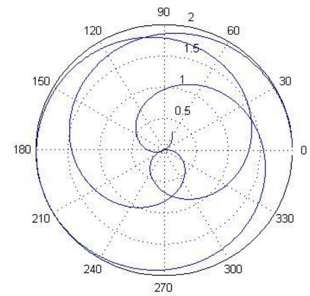

Figure 5. The acoustic tube model of the vocal tract with Mid damping efficiency

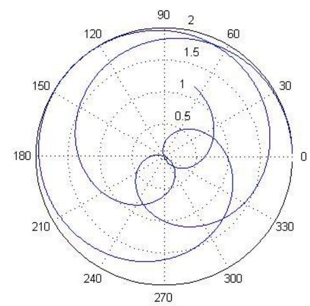

Figure 6. The acoustic tube model of the vocal tract with high damping efficiency

Table 3: types of nose and its quality

Background Noise	Parametric Background Quality
Background noise high	Hissing – fizzing Rushing - roaring
Background noise Mid	Rumbling - rolling
Background noise low	Humming – buzzing
Background buzz	Bubbling - percolating
Background flutter	Crackling – staticky
Background static	Chirping - clicking
Background chirping	Chirping - staticky

6. Results and conclusion

The Mathematical modeling of the vocal track has been obtained and implemented. After implementing ξ(x, t) in MATLAB, The Mathematical modeling of the vocal track has been obtained and implemented (Table 4). By this model the sound source (Voice) and the other sound sources (Noise) has been separated and Noise is fully eliminated.

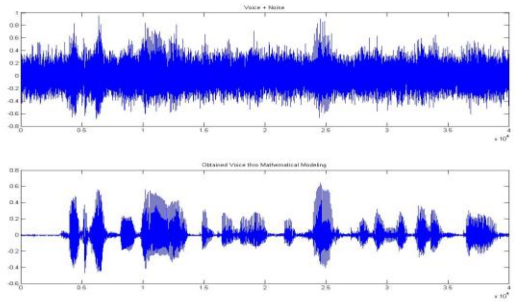

Figure 7: (a) voice + Noise (b) obtained through mode

Table 4: Results obtained

Source	Damping	Various Coefficient	Noise
0.472	0.252	-3.675	-6.848
-0.693	-0.615	-0.22	-8.809
-0.9	-0.127	2.145	-9.685
0.136	0.777	3.285	-9.178
0.984	0.615	2.439	-8.713
0.472	-0.515	0.277	-9.455
-0.693	-1.071	-1.262	-10.693
-0.9	-0.233	-1.131	-10.815
0.136	0.924	-0.322	-9.538
0.984	0.881	-0.382	-8.297
0.472	-0.252	-1.288	-8.269
-0.693	-0.892	-1.569	-8.756
-0.9	-0.172	-0.465	-8.159
0.136	0.872	0.91	-6.203
0.984	0.745	1.013	-4.336
0.472	-0.428	-0.046	-3.745
-0.693	-1.069	-0.674	-3.74

-0.9	-0.318	0.013	-2.73
0.136	0.776	1.002	-0.447
0.984	0.705	0.815	1.657

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