IDIAP RESEARCH REPORT

STOCHASTIC TECHNIQUES IN DERIVING PERCEPTUAL KNOWLEDGE

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Abstract

The paper argues on examples of selected past works that stochastic and knowledge-based approaches do not contradict each other. Frequency resolution of human hearing is decreasing with increasing frequency. Spectral basis designed for optimal discrimination among different phonemes of speech have similar property. Further, human hearing is most sensitive to modulations with frequency around 4 Hz. Filters on feature trajectories, designed for optimal discrimination among phonemes of speech are bandpass with central frequency around 4 Hz.

1. Introduction

Speech signal originates in speaker vocal organs, it is to be processed by human auditory system, and it has its purpose, which is a communication among human beings. This knowledge can be used to advantage in designing speech processing techniques. Such techniques are often called the knowledge-based processing techniques.

Variability of speech signal due to non-linguistic information sources such as environmental noise, or anatomical and idiosyncratic differences among various speakers is not well understood and its influence on the signal appears almost random. The so called stochastic cpeech processing techniques that attempt to deal with this random component in the signal are currently dominating the field.

When it comes to information rate reduction for speech recognition, both the deterministic and the stochastic techniques use knowledge. The difference is that in the knowledge–based techniques the knowledge may come from relevant speech production and speech perception experiments while in stochastic techniques, the knowledge comes from large amounts of training data.

Most would agree that the strategies using the knowledge derived from data appear to work better. On the other hand, the stochastic techniques do require large amounts of the data to get the knowledge. One cannot help wondering if the stochastic techniques do not waste the data on re-learning the same speech-specific knowledge every time again and again. Is there any way to use the knowledge derived by the stochastic techniques from one data set on a new problem? What is it that the stochastic techniques derive from the data?

Given that speech evolved to be heard, it should not be surprising to find out that an examination of stochastic techniques optimized on large amounts of speech data turn out to be consistent with relevant properties of human speech production and perception.

2. Linear discriminant analysis

The linear discriminant analysis (LDA) is a stochastic technique that attempts to optimise the linear discriminability between classes in the presence of undesirable within-class variability (see e.g. [Hunt 1979, Brown 1987] for some examples of previous use of LDA in ASR). It requires that class affiliation of each vector in the data used for the analysis is known (the database is labelled).

LDA is most often applied to sequences of several short-term feature vectors [Braun 1987]. In such applications, the resulting linear discriminants form two-dimensional filters that are

to be applied to time-feature plane. In this paper, however, we review works that allow for interpretation of LDA results in terms of either a) variable-resolution spectral bases that may indicate certain spectral resolution of speech analysis, or b) the FIR RASTA filters that may indicate certain range of modulation frequencies, that are desirable for classification of speech.

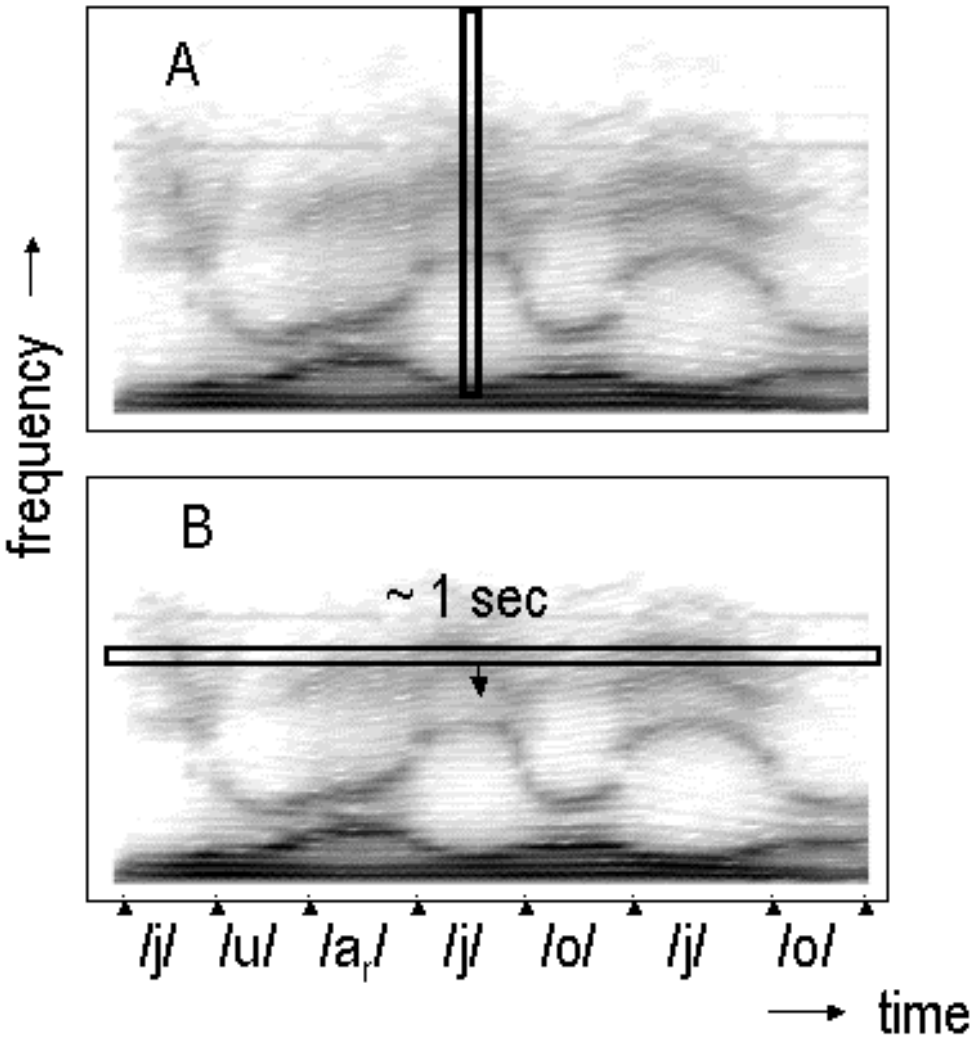

Fig. 1 Two possible ways of forming labelled vectors for LDAT analysis of short-term spectra.

In the first case, reported in [Malayath and Hermansky 2002, 2003] and shown in the A part of the Figure 1 , LDA was applied to vector space of logarithmic Fourier spectra of speech,. In this case, the resulting LDA discriminant matrix consists of basis for projection of the short-term spectra . Such basis represent an alternative to the conventional cosine basis of the cepstral projection. The vector labelling is in this case trivial since each vector is clearly affiliated only with a single class. The second way of applying LDA was reported in [van Vuuern and Hermansky 1997] and shown in the B part of the Figure 1. In this case, the LDA was applied to the vector space formed from segments of temporal trajectories of spectral energies. The LDA discriminant matrix in this case consists of FIR filters to be applied on time trajectories of logarithmic spectral energies. In reported experiments about 1 s long temporal vectors were used. Each vector spanned much more than a single phoneme, and was labelled by the phoneme at the centre of the vector.

3. Critical-bands of hearing

Fletcher observed that signal with frequency components outside a certain "critical band" does not affect detection of an another signal with its frequency components inside this "critical band" (see [Fletcher 1953] for the review of their earlier experiments that revealed existence of the critical bands). This is a clear evidence of the ability of human hearing to separate different spectral components of the acoustic signal into individual bands for the further processing. An important property of such "critical-band-like" spectral analysis is that its frequency resolution is lower at higher frequencies [Fletcher 1953]. Similar spectral scale has been observed in experiment with perception of pitch of tones [Shower and Biddulph 1931]. Even though neither the masking experiments nor the experiments with perception of pitch suggest that spectral profiles of sounds are the entities extracted and used by hearing for the sound classification, benefits of the critical-band-like spectral resolution in ASR seems to be well established through years of comparative ASR experiments. The critical-band-like spectral analysis is often emulated in speech processing by weighted summation of the short-term Fourier spectrum [Davis and Mermelstein 1980, Hermansky 1990].

Malayath and Hermansky [Malayath and Hermansky 2002, 2003] applied LDA to short-term spectral vectors from Fourier analysis (20 ms hamming window, 10 ms analysis step) of all monophthong vowels from OGI Stories database (OGI Stories contain about 3 hours of fluent American English telephone-quality speech from more than 200 adult speakers of both genders, hand-labelled by phonemes). The LDA was used to find such projections of the logarithmic short-term spectrum (spectral basis) that would allow for optimal discrimination among the vowels.

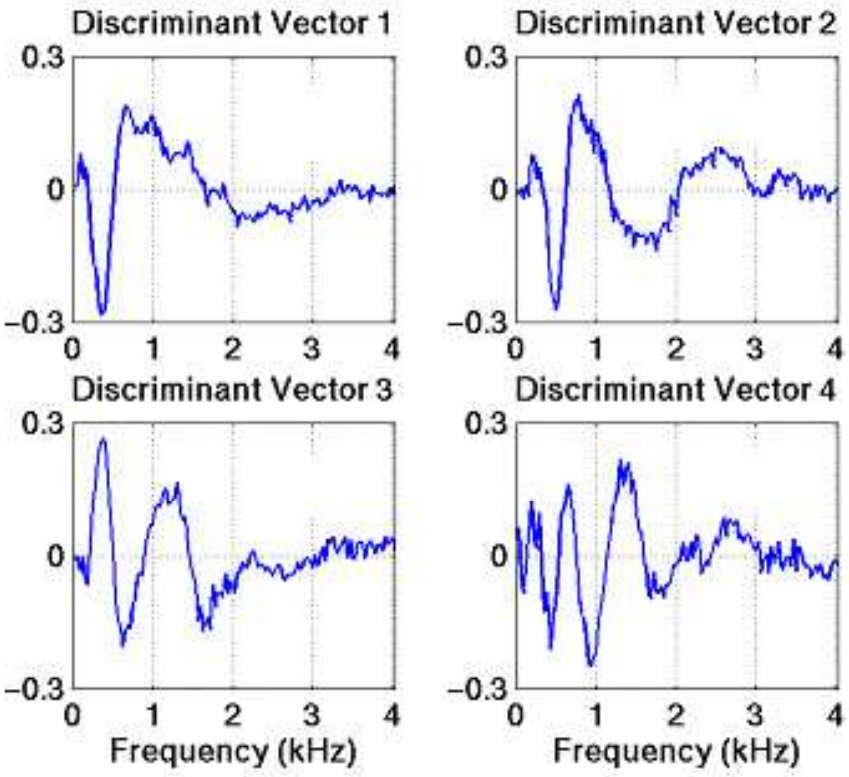

Figure 2 Spectral basis derived by LDA technique

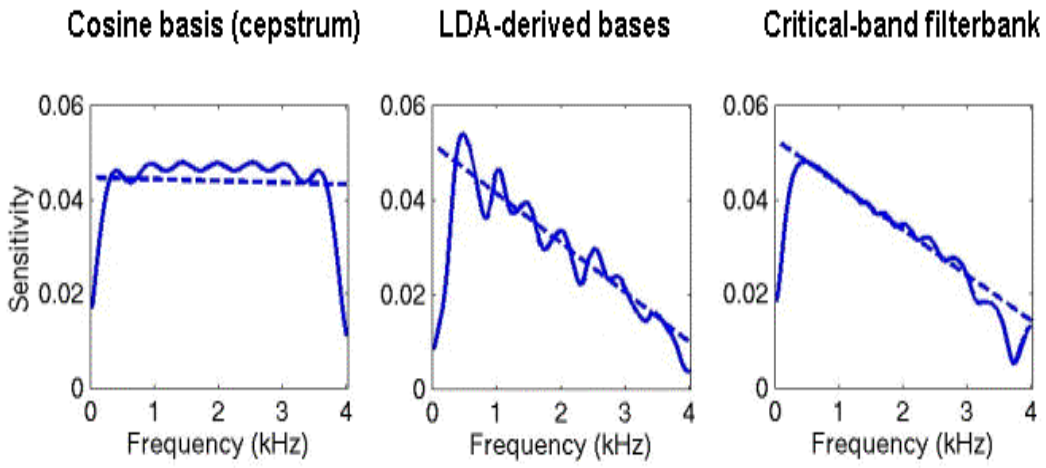

Figure 3 Sensitivity to frequency change of a synthetic formant for three different spectral analysis techniques. As expected, the cepstral projection yields approximately uniform sensitivity while the sensitivity of the LDA-derived projection is similar to the sensitivity of the emulated critical-band filter-bank

The first four spectral basis from their LDA analysis are illustrated in Fig. 2. Notice that period of these spectral basis is shorter at lower frequencies. Subsequently, speech analysis that employs such spectral basis has higher spectral resolution at lower frequencies. [Malayath and Hermansky 2002, Malayath and Hermansky 2003] show by sensitivity analysis of emulated critical-band filter-bank and the LDA-derived spectral projection that the spectral resolution implied by spectral basis in Fig. 2 is very similar to spectral resolution of auditory-like Bark frequency scale (Fig. 3).

This finding supports earlier results of [Umesh et al. 1997] who derived auditory-like frequency warping by minimizing differences between speech from different talkers.

The optimality of logarithmic-like spectral scale in discrimination of vowels is not surprising. Changes in the position of the tract constriction cause roughly equal relative changes in formant frequencies. That is: when, as a result of change in the front cavity shape the first formant changes from its initial position around 500 Hz by say 10 Hz, the second formant moves from its initial 1500 Hz by about 30 Hz and the third by about 50 Hz) i.e. the resulting vowel spectrum changes about uniformly on the logarithmic frequency scale.

4. Perception of modulations

Since early experiments in perception of modulated signals [Riesz 1928] it is known that the ear is most sensitive to modulations around 4 Hz. This finding has been subsequently verified a number of times (see [Kay 1982] for a review). Further, extensive experiments of Drullman and his colleagues [Drullman et al 1994] and Arai and his colleagues [Arai et al 1999] have shown that for maintaining the high intelligibility of speech, only the spectral envelope changes between about 1 and 15 Hz are necessary.

In the model, the sensitivity of human hearing to spectral envelope changes can be emulated by filtering temporal trajectories of computed parameters. This has been done in RASTA processing of speech [Hermansky and Morgan 1994] to attenuate the features with rate-of-

change that is not expected for speech. The initial ad hoc form of the RASTA filters was optimised on a relatively small series of ASR experiments with noisy telephone digits. The form of the optimised RASTA filter independently confirmed Drullman et al and Arai et al. – the filter passes components between 1 and 15 Hz and attenuated slower and faster changing parameters.

Van Vuuren and Hermansky formed 101-dimensional vector space from logarithmic outputs of the emulated critical-band filter-bank [Hermansky 1990] with vectors labelled by their respective phoneme classes. Each vector then spanned about 1 s at 100 Hz sampling frequency. LDA analysis yielded a 101 X 101 scatter matrix, decomposed into its principal components. Then the principal vectors represented FIR filters, which most efficiently (with respect to the within-class and the across-class variability) mapped the 101-dimensional input space to a single point of the output space. Since the target classes were context independent phonemes (just like in the previous experiment in the LDA design of the spectral basis), such designed FIR filters attempted to compensate for the coarticulation with neighbouring phonemes. Further, they also attempted to compensate for other sources of non-linguistic variability such as noise.

Frequency responses of the first three FIR filters derived from about 60 hours of forcefully-aligned Switchboard database are shown in Fig. 3 from [Hermansky 1997]. Filters for different frequency channels are similar. The frequency characteristic (shown at in the right part of the figure) are generally consistent with RASTA [Hermansky and Morgan 1994], and delta, and double-delta feature of speech [Furui 1981]. However, impulse responses of the data-derived filters shown in the upper part of the figure suggest preference for the zero-phase filters. Effective parts of the impulse responses appear to span at least 250 ms. Interesting fact is that the LDA filters derived at different frequencies (not shown here) are roughly the same, i.e. the filters at say 500 Hz do not noticeably differ from the filters derived at 3 kHz. This result would support the notion of the second (post-cochlear) time constant, hypothesized since early works of Gabor [Gabor 1946].

The general characteristics of the data-derived RASTA filters appear to be relatively independent of the particular database used for their design. The most important processing involves a mild temporal lateral inhibition in which the average of several spectral values around the current time instant is subtracted from the weighted average of spectral values from surrounding past and future contexts. Next is the difference between weighted averages from left and right contexts of the current frame (the first derivative of the first discriminant vector), followed by an aggressive mexican hat temporal lateral suppression (the second derivative of the first discriminant vector) implying quite narrow band-pass filter with 12dB/oct slope. Such impulse responses can be interpreted as a difference of two Gaussians (the first discriminant) and its derivatives (higher discriminants). Mexican hat-like dynamics-enhancing functions are hypothesized to be important for scene interpretation by human visual system [Marr 1982].

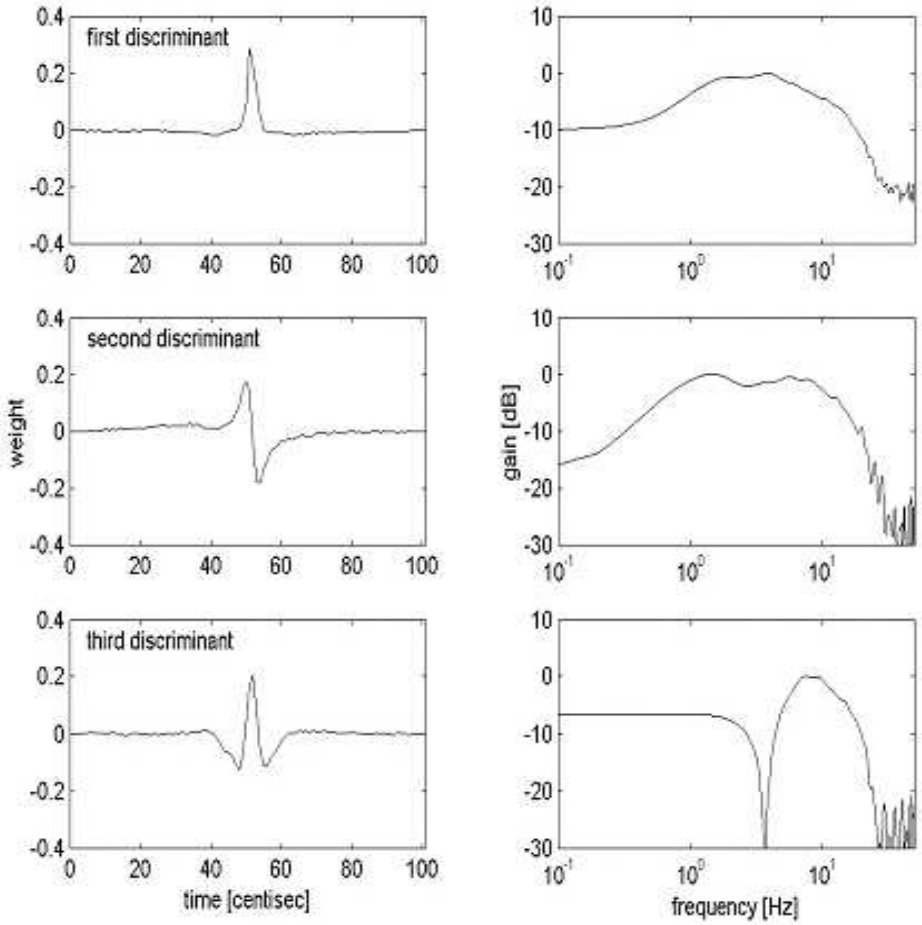

Figure 3 Impulse and frequency responses of the first three discriminant vectors from the LDA-derived discriminant matrix. The filters for the 5 Bark frequency channel are shown here. Filters for the other carrier frequencies studied (between 1 and 14 Bark) are very similar.

5. Data-guided processing and human auditory perception.

Spectral basis derived by LDA and shown in Fig. 2, were applied for deriving of small number of linearly-separable features from the short-term FFT logarithmic power spectrum. The only build-in prior knowledge from hearing is the use of the power spectrum. (This may be justified by frequency selectivity of human cochlea and by one-way rectification of auditory hair-cell firings). The LDA technique is otherwise rather ignorant in matters of human psychophysics and/or physiology and merely attempts to do the engineering job of efficient separation of speech sound classes. Yet, it delivers spectral resolution that is consistent with human hearing!

The same may be said of LDA-derived RASTA filters. The impulse responses could have been highly concentrated in time but they are not, implying that it is beneficial for identification of phonemes in running speech to collect data from relatively large time spans, significantly exceeding the typical 10-20 ms length of the analysis window [Yang et al. 2000]. Rather, consistently with the "critical time interval" observed in forward temporal masking and many other perceptual phenomena, the time span for the information extraction is several hundreds of ms. Frequency responses of the dominant discriminants are band-pass, passing the range of modulation frequencies between roughly 1 Hz and 15 Hz where human

hearing is the most sensitive. Thus, again, the temporal processing that is needed for a good classification of phoneme-like speech sounds is quite consistent with temporal properties of human hearing.

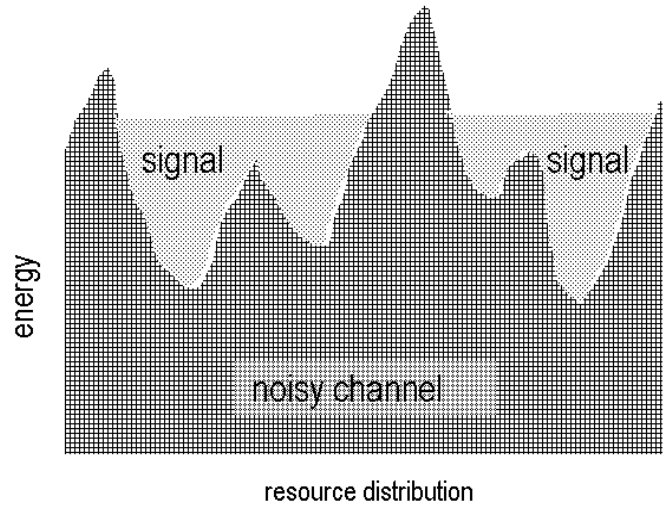

Fig. 4 Optimal distribution of signal energy in noisy channel (adopted from [Galagher 1968])

Why would optimization of signal processing module on speech data result in human-like processing? Are the properties of human hearing imprinted on speech signal? Come to think of it, what else should we expect? Information theory teaches that optimal signal for communication through noisy channel should confirm to properties of the channel [Gallager 1968]. Then, imprinted on the signal, one should see some properties of the communication channel for which the signal was designed. Hearing likely existed before speech evolved - all parts of the human speech production also serve more life-sustaining function than speech production. It thus appears likely that they were adopted for speaking later in the course of human evolution. Why would not forces of nature follow the same optimal strategy and form speech obeying the same optimal principles, i.e. form it in such a way that it is well heard? As a result, when engineer attempts to design an optimal processing strategy, she could end up with the strategy that emulates human hearing!

6. Summary

We discussed speech processing techniques that attempt to optimise processing in such a way that the goal of the processing, i.e. the extraction of information from the speech signal, is better achieved. Such techniques form a bridge between signal processing and stochastic pattern classification and subsequently are trained on large amounts of speech data. The consistency of resulting signal processing modules with some basic properties of human hearing support optimality of hearing in decoding the linguistic message in speech.

7. Acknowledgements

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