

Non-stationary acoustic objects as atoms of voiced speech

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In spite of the undisputedly high degree of non-stationarity of speech signals, the present day determination of its acoustic features is based on the assumption that *speech production* can be described as a linear time invariant (LTI) system on the time scale of about 20 ms [1]. In automatic speech recognition, the wide sense stationarity of an LTI-system is used as prerequisite for the consistent estimation of Fourier spectra or of autoregressive models [1]. As an evolutionarily plausible supplement, *speech perception* is also assumed to be focussed on acoustic features which are obtained by using the LTI assumption [2]. Present day models of pitch perception are no exception [2, 3].

The empirical mode decomposition of Huang et al. [4] represents one of very few methods, which are suited to analyze signals without assuming their stationarity. In case of voiced speech, the formant frequencies have been shown to correlate well with frequencies of the empirical modes [5]. However, it is hard to imagine how the so called sifting process of Huang et al. [4] fits into the known function of the peripheral auditory pathway. The present study proposes a different method of empirical mode decomposition, which is based on well known features of the auditory pathway. As a further contrast to decomposition [4], the pitch perception oriented mode decomposition leads to mode reconstructions, which can be confirmed to have uncorrupted phases in comparison to the phases of underlying oscillatory subsystems [6, 7]. Furthermore, it is shown that voiced phones support a fast convergent iterative reconstruction.

The empirical modes can be used advantageously to reconstruct a single fundamental oscillator, the frequency of which can be interpreted as the acoustic correlate of virtual pitch. Virtual pitch perception should be interpreted as an ingenious instrument of time scale separation, which separates the phonological time or frequency scales from the phonetically relevant ones. In contrast to numerous existing pitch tracking methods [2, 3], the virtual pitch oriented time scale separation does not rely on a frequency gap in the long term spectrum (being equivalent to temporary stationarity).

Hearing models are well known to use a subband decomposition with so called critical (audiological) bandwidths. These bandwidths limit the number of separable harmonics to the range 6-8. If we describe the input signal as $S(t)$, the envelope of the impulse response of the bandpass filter as $W_j(t')$, the distance of the envelope maximum to the start of the response as τ_j and the centre filter frequency (CFF) as $\omega_j/2\pi$, we obtain a complex subband of the form

$$X(\omega_j, t) = \int_{-\infty}^{t+\tau_j} d\tau S(\tau) W_j(t-\tau) e^{i\omega_j(t-\tau)}, \quad (1)$$

which represents an analytical signal (due to the limited bandwidth) and is thus well suited to obtain subband phases

$\varphi_j(t)$. An autoregressive approximation to a gammatone bandpass filter represents a widely used example [6, 7]. The CFFs $\{\omega_j | j=1, \dots, J\}$ are usually chosen with the aim to achieve approximate orthogonality (and completeness) of the decomposition. In deviation to common hearing models we start the analysis with CFFs of virtual pitch perception models [2,3], i.e. with CFFs which form a harmonic (equidistant) grid (template) for the harmonics 1-8. Deviating from common pitch models we introduce a time dependent CFF into *subband* (1)

$$X_j(t) = \int_{-\infty}^{t+\tau_j} d\tau S(\tau) W_j(t-\tau) \exp(i \int_{\tau}^t \omega_j(\tau') d\tau') \cdot \quad (2)$$

To demonstrate useful properties of *part-tone* (2), we choose an input signal which is often termed as *sinusoid*

$$S(\tau) = A(\tau) \cos \Phi_j(\tau) \quad \text{with}$$

$$\Phi_j(\tau) = \int_0^{\tau} \Omega_j(\tau') d\tau', \quad (3)$$

where amplitude $A(\tau)$ and phase velocity $\Omega_j(\tau)$ are both limited to positive values. In analogy to phase definition (3), we define filter phase $\psi_j(\tau)$ by a time integral of the respective CFF $\omega_j(\tau)/2\pi$. For times $t \gg \tau_j$, analytic signal (2) simplifies to

$$X_j(t) = \frac{1}{2} e^{i\psi_j(t)} \int_0^{t+\tau_j} d\tau A(\tau) W_j(t-\tau) e^{i(\Phi_j(\tau) - \psi_j(\tau))}.$$

Choosing the special case $\omega_j(\tau') = \Omega_j(\tau')$, i.e. a CFF contour which is precisely adapted to the sinusoid, and assuming a slowly varying amplitude $A(\tau)$, we get

$$X_j(t) = \frac{1}{2} A(t) e^{i\varphi(t)}, \quad (5)$$

i.e. a part-tone which represents the analytic signal of the input signal. Since equation (5) is valid for arbitrary times, result (5) can also be interpreted in a different way. If (in case of a slowly varying amplitude) we succeed to adapt the CFF of a part-tone to its instantaneous frequency, we are assured to have reconstructed a part-tone with an uncorrupted phase.

For a given CFF contour all input signals with a different phase velocity experience a damping due to interference. A filterbank with bandpass filters of form (2) or (4) is thus well suited to separate several empirical modes with different frequencies without corrupting their phases. However, there remains the problem of finding the appropriate CFF contours. In this situation we hypothesize that voiced speech supports a robust and efficient adaptation algorithm on the receiver side and that the adaptation of the CFFs can be achieved iteratively by using the instantaneous frequency contours of the respective part-tones.

Stability of Centre Filter Frequencies

To analyze the convergence of the iterative adaptation of a CFF contour it is useful to approximate the envelope of the impulse response in equation (4) by a Gaussian

$$W_j(t-\tau) = W_j(0) \exp\left(-\frac{\alpha_j}{2}(t-\tau)^2\right) \quad (6)$$

where α_j denotes the inverse of its variance. If we assume amplitude $A(\tau')$ as slowly varying in comparison to envelope (6), part-tone (4) simplifies to

$$X_j(t) = W_j(0)A(t)e^{i\psi_j(t)} \int_{-\infty}^{\infty} d\tau' e^{-i(\psi_j(t+\tau') - \Phi_j(t+\tau'))} e^{-\frac{\alpha_j}{2}\tau'^2}.$$

Furthermore, we assume a smooth instantaneous phase velocity $\Omega_j(\tau')$ of the sinusoid (oscillatory subsystem on the transmitter side). Within a sufficiently short analysis window, the phase of the sinusoid can therefore be described well by a quadratic function. In case of the CFF, we deliberately choose a linear time trend within the current window. The phase difference in equation (7) can thus be written as

$$\begin{aligned} \psi_j(t+\tau') - \Phi_j(t+\tau') &= \psi_j(t) - \Phi_j(t) + (\omega_j(t) - \Omega_j(t)) \tau' \\ &\quad + (\dot{\omega}_j(t) - \dot{\Omega}_j(t)) \tau'^2 / 2 \\ &= \Delta\psi_j(t) + \Delta\omega_j(t) \tau' + \Delta\dot{\omega}_j \tau'^2 / 2, \end{aligned} \quad (8)$$

where we have introduced useful abbreviations in the last step. In case of the error term $\Delta\dot{\omega}_j$ of the CFF chirp, we can drop the time argument in view of the above assumptions. As a further benefit, the integrand in equation (7) simplifies to a complex Gaussian. When the integral is solved analytically, we arrive at phase $\varphi_j(t)$ of part-tone (7)

$$\varphi_j(t) = \Phi_j(t) + \frac{1}{2} \Delta\dot{\omega}_j \frac{\Delta\omega_j(t)^2}{\alpha_j^2 + \Delta\dot{\omega}_j^2} - \frac{1}{2} \arctan\left(\frac{\Delta\dot{\omega}_j}{\alpha_j}\right). \quad (9)$$

To make the convergence property of the iteration visible, we represent the phase error $\varphi_j(t) - \Phi_j(t)$ of the part-tone as function of the error of the respectively used CFF contour, and express this error as function of $\Delta\omega_j(t_0)$ und $\Delta\dot{\omega}_j$, where t_0 denotes the start of the current analysis window. Replacing $\Delta\omega_j(t) = \Delta\omega_j(t_0) + \Delta\dot{\omega}_j(t-t_0)$ in equation (9), we obtain in leading order of the small quantities $\Delta\omega_j(t_0)$ und $\Delta\dot{\omega}_j$

$$\begin{aligned} \varphi_j(t) - \Phi_j(t) &= \varphi_j(t_0) - \Phi_j(t_0) + \Delta\omega_j(t_0) \frac{\Delta\dot{\omega}_j^2}{\alpha_j^2} (t-t_0) \\ &\quad + \frac{\Delta\dot{\omega}_j^3}{\alpha_j^2} (t-t_0)^2 / 2 + \dots \end{aligned} \quad (10)$$

The left hand side can alternatively be expanded into a Taylor series for small $(t-t_0)$. When comparing the expansion coefficients with the corresponding terms of equation (10), we arrive at

$$\begin{aligned} \dot{\varphi}_j(t_0) - \Omega_j(t_0) &= \Delta\omega_j(t_0) \Delta\dot{\omega}_j^2 / \alpha_j^2 \quad \text{and} \\ \ddot{\varphi}_j(t_0) - \dot{\Omega}_j(t_0) &= \Delta\dot{\omega}_j^3 / \alpha_j^2. \end{aligned} \quad (11)$$

In its simplest form, the iteration replaces $\Delta\omega_j(t_0)$ by $\dot{\varphi}_j(t_0) - \Omega_j(t_0)$ and $\Delta\dot{\omega}_j$ by $\ddot{\varphi}_j(t_0) - \dot{\Omega}_j(t_0)$. The described iteration thus converges with the *third power* of the deviation from a stable fixed point, i.e. faster than the well known Newton method and therefore faster than all optimization algorithms. As shown in equation (5), filter stability implies uncorrupted phases of the respective part-tone.

For part-tones with several relevant harmonics, the sinusoid assumption being specified in connection with equation (3) is no longer valid. For such part-tones, convergence criterion (11) does not apply. Such part-tones are well known to have lower priority for virtual pitch perception [2,3] and are out of the scope of the present study. Voiced phones are known to have at least one formant in the range of the separable harmonics. In view of the additional existence of anti-formants, we must be prepared to find several stable fixed points of the iteration. Part-tones which converge to the same stable fixed point, define an empirical mode specific cluster. Within each cluster a second type of distance of a part-tone to the cluster centre can be defined as the number of iterations being necessary to converge. Part-tones with low distances to a cluster centre are synthesized to an empirical mode. The frequency of a formant mode is interpreted as acoustic correlate of spectral pitch [3].

For lower harmonic part-tones the vocal tract causes phase shifts, which introduce non-negligible frequency shifts in a non-stationary setting. This makes re-engineering of virtual pitch perception more difficult. As part of auditory scene analysis, a subset of empirical modes is selected, which is consistent with the current estimate of the fundamental frequency contour $f_0(t)$. As motivated in [6], the selected empirical modes are interpreted as different transients of a two-level cascaded response system which is driven by a common (hidden) fundamental drive in the frequency range of the pitch. For the single harmonic dominated modes, the excitation(s) of the well known secondary (vocal tract) response are described as periodic functions of the *fundamental phase*. (For the other modes their envelopes are described as periodic functions.) The band-limited periodic functions can be represented by finite Fourier series. The empirical modes and the present estimate of $f_0(t)$ are used to estimate the Fourier coefficients and the resonator parameters simultaneously by multiple linear regression. For given excitation- and resonator parameters, the analysis window specific chirp rate $\dot{f}_0(t_0)$ of the fundamental drive is estimated by 1-dim. nonlinear regression. The chirp rate is used to update the fundamental frequency contour $f_0(t)$.

The stable $f_0(t)$ can be interpreted as acoustic correlate of virtual pitch and the resonator and excitation parameters can be used for the distinction of sustainable voiced phones. For a sufficiently broadband excitation of the secondary response, the cascaded response of a non-stationary fundamental drive generates non-stationary acoustic objects, which are suited to be analyzed by the outlined pitch- and vowel perception oriented empirical mode decomposition and time-scale separation.

References

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