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COM3502-4502-6502 SPEECH PROCESSING

Lecture 18

Linear Prediction

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Linear Prediction

'Linear prediction' is the basis for almost all modern speech coding algorithms ...

- mobile phones
- internet telephony (VOIP)
- military communications



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Linear Prediction Analysis

- Linear prediction assumes that each sample (of a speech signal) can be predicted from a weighted sum of the p preceding samples

$$\hat{s}[n] = a_1s[n-1] + a_2s[n-2] + \dots + a_p s[n-p]$$

$$= \sum_{i=1}^p a_i s[n-i]$$
- Exploiting redundancy arising from the *stationarity* assumption, linear prediction allows a speech frame of several hundred samples to be represented by only 10-15 'prediction coefficients' ($a_1 \dots a_p$)

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Linear Prediction Analysis

The diagram illustrates the linear prediction process. It shows a signal $s[n]$ (actual value) and its prediction $\hat{s}[n]$ (predicted value). The prediction is calculated as a weighted sum of the p preceding samples $s[n-1]$ to $s[n-p]$ using coefficients a_1 to a_p . The prediction error $e[n]$ is the difference between the actual value and the predicted value. A red box highlights the linear predictor block with the text "This is a filter".

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Linear Prediction Analysis

- The prediction error for a particular sample $e[n]$ is ...

$$e[n] = s[n] - \hat{s}[n] = s[n] - \sum_{i=1}^p a_i s[n-i]$$
- Taking Z-transforms of both sides we obtain ...

$$E(z) = \left[1 - \sum_{i=1}^p a_i z^{-i} \right] S(z)$$
- Hence the **filter** transfer function is ...

$$H(z) = \frac{E(z)}{S(z)} = 1 - \sum_{i=1}^p a_i z^{-i}$$

An 'all-zero' filter

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Calculating Prediction Coefficients

- The linear prediction coefficients are computed to minimise the sum-squared '**prediction error**' ('**residual signal**') E_p over a frame of speech data

$$E_p = \sum_n e[n]^2 = \sum_n \left(s[n] - \sum_{k=1}^p a_k s[n-k] \right)^2$$

$\hat{s}[n]$ (estimated signal)

$s[n]$ (actual signal)

- This has the effect of '**whitening**' the error signal (i.e. *flattening its spectrum*)

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Calculating Prediction Coefficients

- E_p is minimised by finding the solution to ...

$$\frac{\partial E_p}{\partial a_k} = 0$$
- This results in a set of p simultaneous equations called the 'normal equations' ...

$$\sum_n s[n]s[n-k] = \sum_{i=1}^p a_i \sum_n s[n-i]s[n-k] \quad k = 1 \dots p$$
- Different ranges to the summation gives rise to ...
 - the 'autocorrelation method'
 - the 'covariance method'

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Covariance Method

- In the covariance method, a windowed frame of N samples is considered from the outset
- By defining ...

$$\phi_{i,k} = \sum_{n=0}^{N-1} s[n-i]s[n-k]$$
- The normal equations can be re-written as ...

$$\phi_{0,k} = \sum_{i=1}^p a_i \phi_{i,k}$$
- These are solved using a general purpose matrix inversion routine for symmetric matrices
- Note that the consequence of the summation is that this method considers an overlap that extends into the previous speech frame

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Autocorrelation Method

- In the autocorrelation method, the summation is taken over all samples ...

$$\sum_{n=-\infty}^{\infty} s[n-i]s[n-k] = \sum_{n=-\infty}^{\infty} s[n]s[n+i-k]$$

- This is the autocorrelation sequence r_{i-k}
- Hence the normal equations can be re-written as ...

$$r_k = \sum_{i=1}^p a_i r_{i-k} \quad k = 1 \dots p$$

- Note that, since the signal is non-stationary, it first needs to be windowed
- The normal equations are solved using a matrix inversion method such as '**Durban's algorithm**'

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Comparison of Methods

	Autocorrelation	Covariance
Filter order	10-15	10-15
Windowing	✓	✗
Stability	✓	✗
Window size	25 msec	5-10 msec
Computation	+	+++
Reflection coeffs.	✓	✗

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Linear Prediction

- The filter transfer function **from the signal to the prediction error sequence** is ...

$$H_1(z) = \frac{E(z)}{S(z)} = 1 - \sum_{i=1}^p a_i z^{-i}$$

An 'all-zero' filter

- Hence the filter transfer function **from the prediction error sequence to the signal** is ...

$$H_2(z) = \frac{S(z)}{E(z)} = \frac{1}{1 - \sum_{i=1}^p a_i z^{-i}}$$

An 'all-pole' filter

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Linear Prediction Synthesis

The LP filter can be configured as an **all-pole digital filter** with the error signal as its input ...

$$H(z) = \frac{S(z)}{E(z)} = \frac{1}{1 - \sum_{i=1}^p a_i z^{-i}}$$

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Linear Prediction

- I.e. the LP filter can be used for analysis and synthesis
 - **analysis**
 - input = signal
 - output 1 = prediction coefficients
 - output 2 = 'residual' error
 - **synthesis**
 - input 1 = prediction coefficients
 - input 2 = 'residual' error
 - output = reconstituted signal
- This makes LP particularly suitable for processing speech signals

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Application to Speech


- In speech analysis/resynthesis ...
 - the LP **filter** models the spectral shaping caused by the vocal tract
 - the error sequence models the **source excitation**
- Note the following assumptions ...
 - the **analysis model** assumes impulse/white-noise excitation
(hence the LP filter also captures spectral shaping caused by the glottal source)
 - the **synthesis model** ignores any zeros
(such as the nasal side branch and the path back down the glottis)

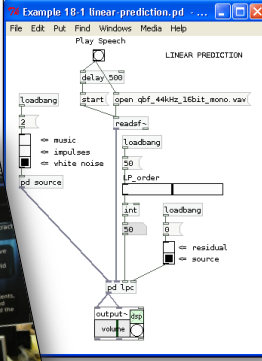
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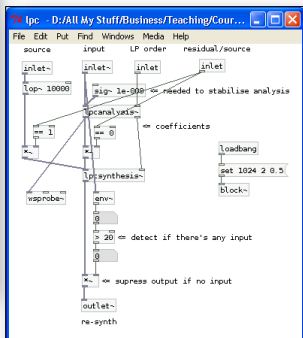
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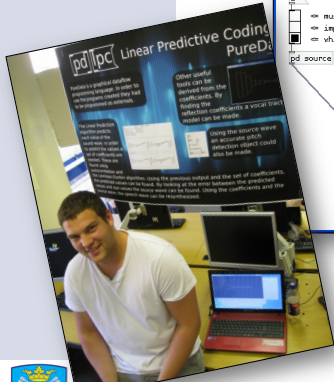
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Linear Prediction of Speech










Hassell, A. (2011). PD LPC: Linear Predictive Coding in PureData. 3rd-Year Project Report, University of Sheffield, Sheffield.



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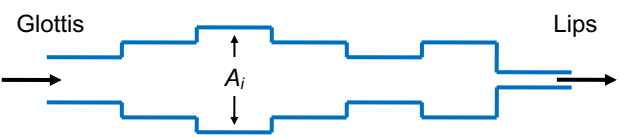
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Vocal Tract Area Estimation

- Durban's algorithm (used in the autocorrelation method) generates parameters that are known as **reflection coefficients**
- It can be shown that these are analogous to the proportion of energy reflected at the boundaries of a lossless acoustic tube model of the vocal tract using a set of cylindrical section of equal length
- The reflection coefficients can thus be used to estimate the cross-sectional area of the vocal tract





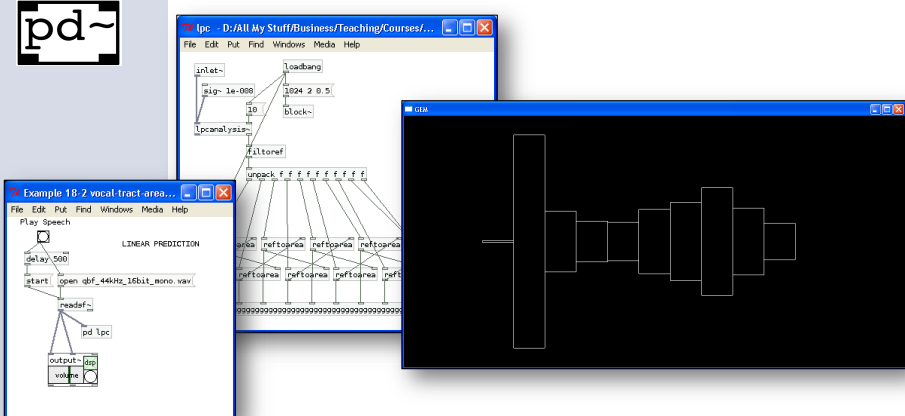
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Vocal Tract Area Estimation



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Applications of Linear Prediction

- pitch extraction
- formant analysis
- speech synthesis
- automatic speech recognition
- linear predictive coding (*LPC*)

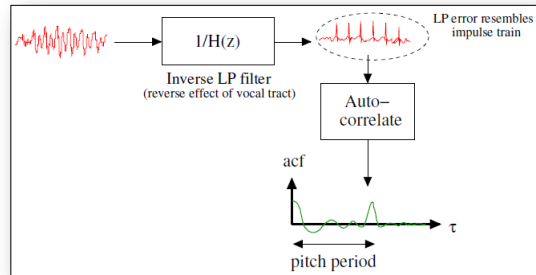
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LP-based Pitch Extraction

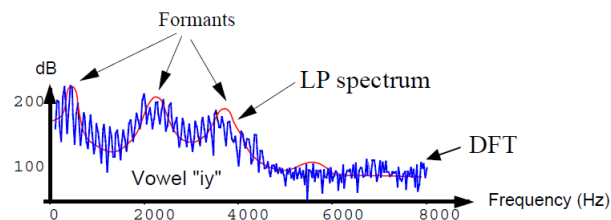
- Obtain LP error signal by passing the speech through the *inverse* filter
- Find the pitch frequency by *autocorrelation* on the error signal



- Use continuity constraints for pitch *tracking*

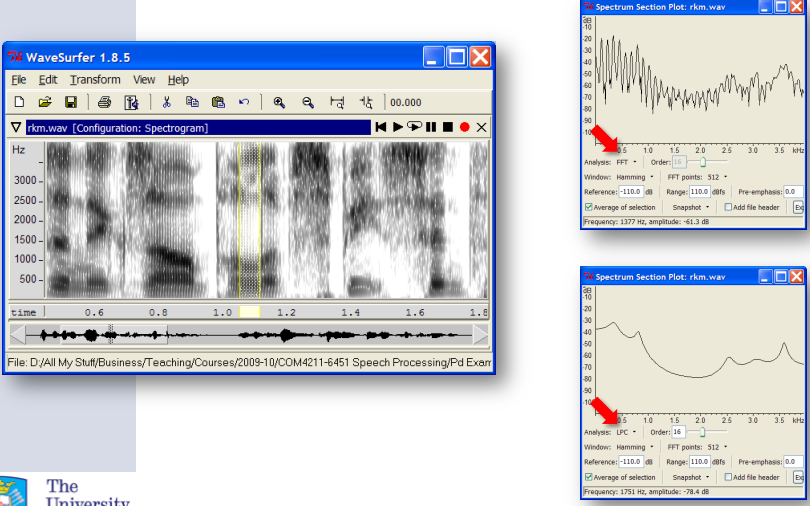
LP-based Spectrum Estimation

- One way to estimate the speech spectrum is to analyse the frequency response of the LP filter
- Since the excitation information is in the error signal, the LP spectrum is a smoothed all-pole approximation to the speech spectrum
- The LP spectrum is thus useful for calculating spectral features (*such as formant frequencies*)



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LP-based Spectrum Estimation



The figure displays the WaveSurfer 1.8.5 interface. The main window shows a spectrogram of a speech signal. Two inset windows, titled 'Spectrum Section Plot: rkm.wav', show the estimated spectrum at different time points. The top inset shows a peak at 1377 Hz with an amplitude of -61.3 dB. The bottom inset shows a peak at 1751 Hz with an amplitude of -78.4 dB. The spectrogram shows a vertical yellow line indicating the time of the analysis.

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LP-based Formant Extraction

- A pole-pair from LP analysis can model a formant
- There are two main approaches to finding formant candidates ...
 - root-solving from the LP polynomial
 - peak-picking from the LP spectrum
- Root-solving finds the frequencies of the pole-pairs with the smallest bandwidths (*i.e. nearest the unit circle*)
- For high-accuracy formant *tracking* it is necessary to consider alternative candidates

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LP-based Speech Synthesis

An implementation of the *source-filter model*

$$s(n) = G.e(n) - a_1s(n-1) - \dots - a_p s(n-p)$$

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LP-based Speech Recognition

- The LP parameters are not guaranteed to be stable from frame-to-frame
- **'Line spectral pairs'** (*LSPs*) are a more stable representation derived from the prediction coefficients.
- Some automatic speech recognition systems use **'perceptual linear prediction'** (*PLP*) for front-end analysis

Hermansky, H. (1990). Perceptual linear predictive (PLP) analysis of speech. *Journal of the Acoustical Society of America*, 87(4), 1738-1752.

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Linear Predictive Coding (LPC)

- LP *analysis* followed by LP *synthesis*
- Need to quantise and code the excitation and LP filter parameters

Transmitter

Receiver

- Used in NATO 2.4 kbps 'LPC-10e vocoder' 🗣️

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Multipulse LPC

Bishnu Atal

- The excitation is set up such that the speech reconstruction error is minimised using an analysis-by-synthesis loop
- The algorithm finds the best pulse position/amplitude, then finds the best position/amplitude of a second, third, etc.
- Pulses can be negative as well as positive


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Mixed-Excitation LPC (MELP)



- Two-state (V - UV) excitation sounds “buzzy”
- Errors in pitch estimation and/or voicing decisions reduce intelligibility
- ‘Mixed-excitation’ models avoid hard V - UV decisions
- MELP has been shown to offer excellent intelligibility at 2400, 1200 and 600 bps
- The US DoD ‘**MELPe**’ was adopted in 2002 as the NATO standard vocoder for secure speech communications (*STANAG-4591*)



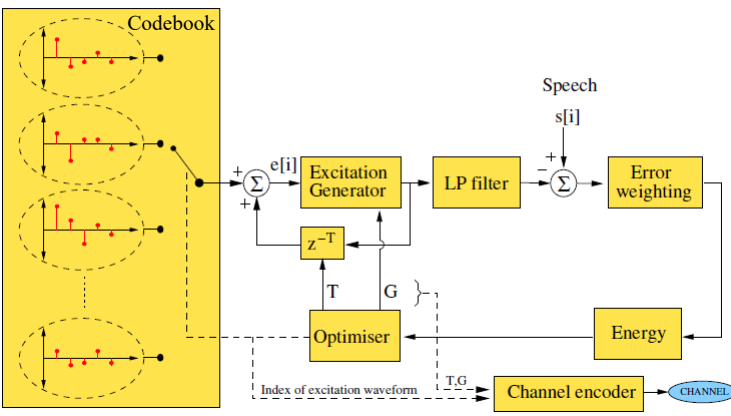
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Code-Excited LPC (CELP)



- Produces good quality speech over 4.8 kbps
- The basis of most modern speech coding systems




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GSM Speech Codec



- The digital mobile cellular radio network which is extensively used throughout Europe and many other parts of the world is termed the 'Global System for Mobile communications' (GSM)
- The GSM full rate speech codec operates at 13 kbps
- Basic operating principle ...
 - 'CELP'
 - 8 kHz sample rate
 - 20 msec frames
 - 8 short-term predictor coefficients per frame
 - each frame is then further split into four 5 msec sub-frames
 - for each sub-frame the encoder finds a delay and a gain for the codec's long-term predictor
 - the residual signal after both short and long term filtering is quantised for each sub-frame





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This lecture has covered ...

- Linear prediction
- LP analysis of speech signals
- Calculating LP coefficients
- Vocal tract area estimation
- LP-based applications
- Linear predictive coding (*LPC*)




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Any Questions ?



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Next time ...

Cepstral Analysis

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