LING 490 - SPECIAL TOPICS IN LINGUISTICS

Fundamentals of Digital Signal Processing

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Week 15

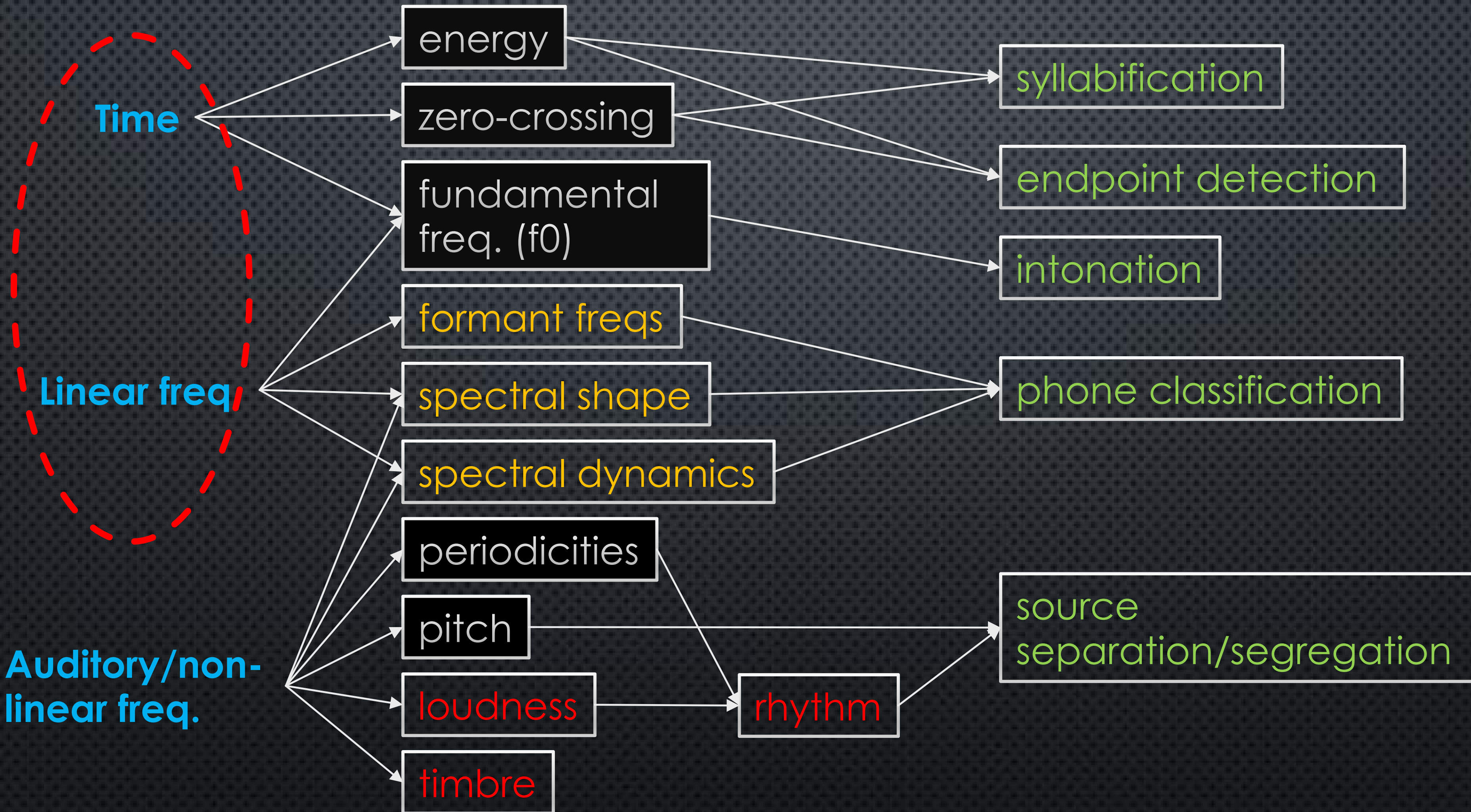
Last week…

- Pitch and fundamental frequency (f0)
    - Pitch is an attribute of sensation
    - F0 is a physical property of a periodic sound

Last week…

- F0 estimation using **autocorrelation**:
  - The correlation of a signal with itself being delayed
  - $\emptyset(\tau) = \sum_{n=0}^{N-1} x(n)x(n+\tau)$
  - The effect of window type
  - The effect of window size
  - Centre clipping: enhance periodicity
- F0 estimation using the average magnitude difference function (**AMDF**)
  - Look for dips instead of peaks
  - $M(\tau) = \sum_{n=0}^{N-1} |x(n) - x(n+\tau)|$

Useful parameters of speech signals



Spectral Centre of Gravity

- Also known as Spectral Centroid
- A measure of the distribution of energy at different frequencies in a spectrum.
- Midpoint of the spectral energy distribution of that sound.
- It can be considered a kind of "balancing point" of the frequency spectrum.

Spectral Centre of Gravity

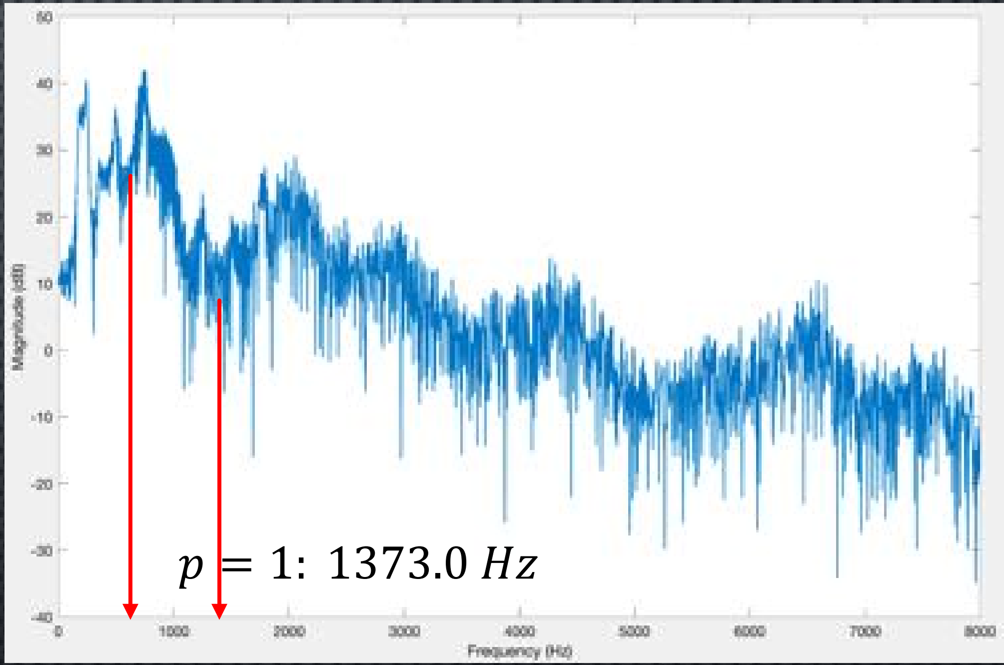
- The spectral centroid measured at one instant in time probably will not prove to be very useful: sounds tend to constantly change.

- If it is measured at several points during the course of the signal, it could reveal how dynamic the signal

Spectral Centre of Gravity

- Spectral Centre of Gravity in Hz is defined as:

$$CoG = \frac{\sum_{k=0}^{K-1} f(k)X(k)^p}{\sum_{k=0}^{K-1} X(k)^p}$$



$p = 2$: $701.1\ Hz$

$f(k)$: centre frequency of spectral bin k

$X(k)$: magnitude of spectral bin k

p: 1 – magnitude; 2 - power

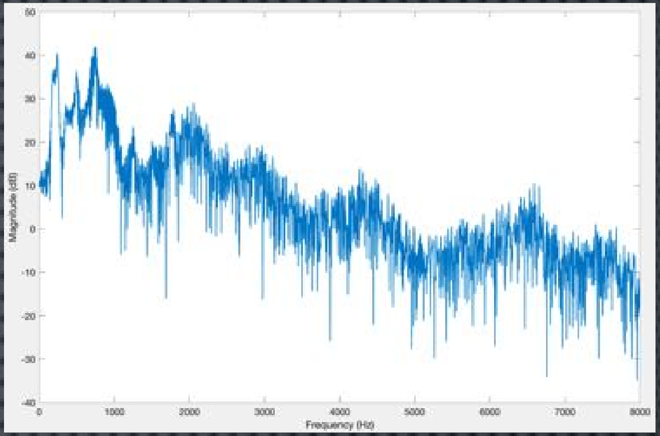
Why is CoG interesting?

Amongst other things:

- Related to impression of "brightness" of a sound
- Essential information for computing spectral moments
- Related to spectral tilt
- Lombard effect: higher spectral CoG – a flattening of spectral tilt
- For English (and Dutch), a more level spectral slope (decreased spectral tilt), i.e., a higher COG, strongly correlates with perceived sentence accent

Sluijter, A.M.C. "Phonetic correlates of stress and accent", HIL dissertations 15, Ph.D. Thesis, University of Leiden, 1995. See also Eurospeech 95, pp. 941-944

Freq-domain speech processing



Why not simply use the (log) magnitude spectrum?

- source and filter are mixed
- difficult to choose time-frequency resolution to always get formants
  - spurious peaks
  - merged peaks
  - can be solved using pitch-synchronous analysis, but that is difficult too
- redundant for clean speech

- DFT is a general tool: not specialised for speech

Linear Predictive Coding (LPC)

- LPC is a powerful speech analysis technique
- One of the most useful methods for encoding good quality speech at a low bit rate.
- Provides extremely accurate estimates of speech parameters, and is relatively efficient computationally

Why LPC

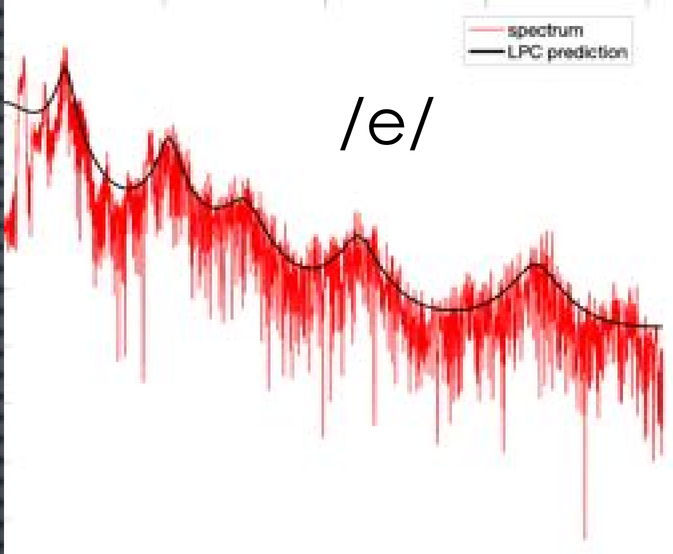
- The voiced speech signal is quasi-periodic
- Anything that is periodic is, in principle, predictable.
- Can we take a number of speech samples and try to predict what comes next?
  - … Yes, and with reasonable results too!

LPC: Basic principles

source → filter → signal

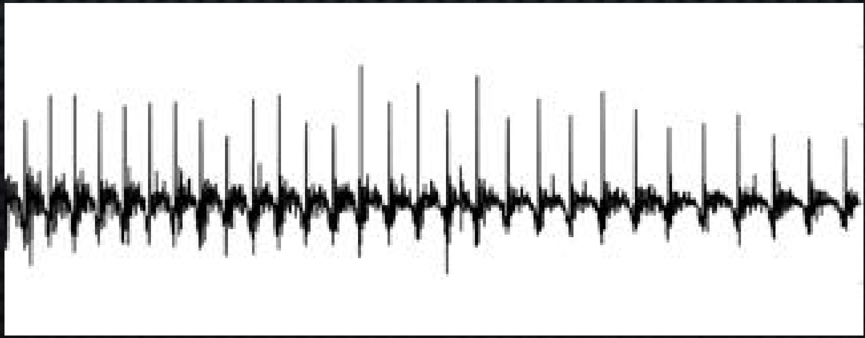
- LPC starts with the assumption that the speech signal is produced by a buzzer (producing pulse train) at the end of a tube. source
- The glottis (the space between the vocal cords) produces the buzz, which is characterized by its intensity (loudness) and frequency (pitch)
- The vocal tract (the throat and mouth) forms the tube, which is characterized by its resonances, which are called formants filter

How does LPC do it?



filter

- LPC analyses the speech signal by:
  - estimating the formants and removing their effects from the speech signal
  - estimating the intensity and frequency of the remaining buzz sound
- The process of removing the formants is called *inverse filtering*, and the remaining signal is called the *residue*



source

The solution

- The basic solution is a *difference equation*, which expresses each sample of the signal as a linear combination of previous samples

- Hence ‘Linear’ and ‘Prediction’ in LPC

So how to predict?

$= 0$

$$y(t) = \sum_{j=0}^{M} b_j x(t-j) - \sum_{i}^{N} a_i y(t-i)$$

- We assume that whatever comes next in $y(n)$ is some linear combination of what's gone before:

$$\hat{y}(n) = a_1 y(n-1) + a_2 y(n-2) + \cdots + a_N y(n-N)$$

$a_1, a_2, \ldots, a_N$: the prediction coefficients

Prediction coefficients

$$\hat{y}(n) = a_1 y(n-1) + a_2 y(n-2) + \cdots + a_N y(n-N)$$

- LPC system needs to estimate these coefficients.
- In other words, what values of $a_1, a_2, \ldots, a_N$ give us the closest approximation to the true signal?

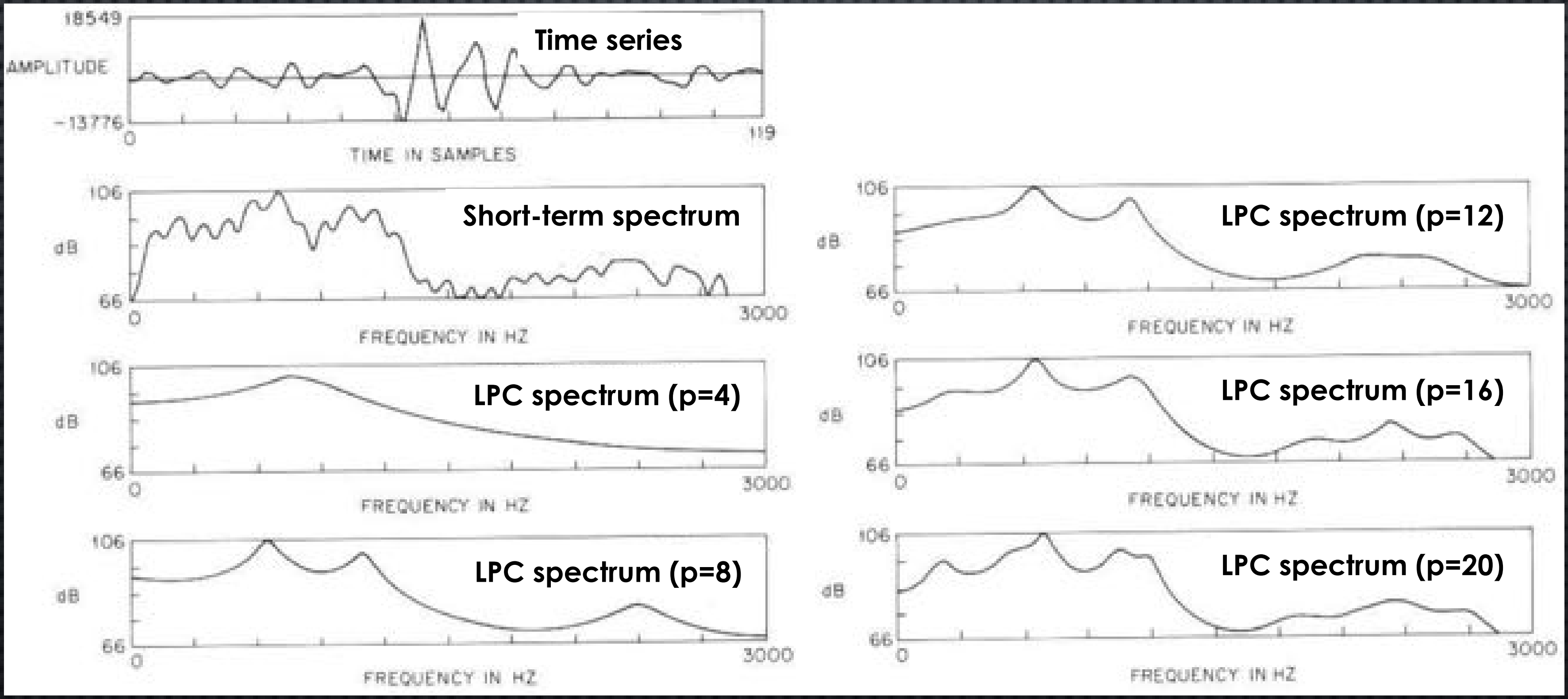
How's it done? (technical aside)

- This is a straightforward problem, in principle.
- In practice, it involves
  - 1. the computation of a set of coefficient values
  - 2. the solution of a set of linear equations.

- These can be solved using a variety of numerical analysis techniques (we're not going to cover this here)

LPC spectra and formant estimation

- Frequency response can be determined by the LPC coefficients



Spectra for /a/ sampled at 6 kHz

Have we done a good job?

- Given the predicted value and the actual value of $y(n)$, we can measure the error (difference):

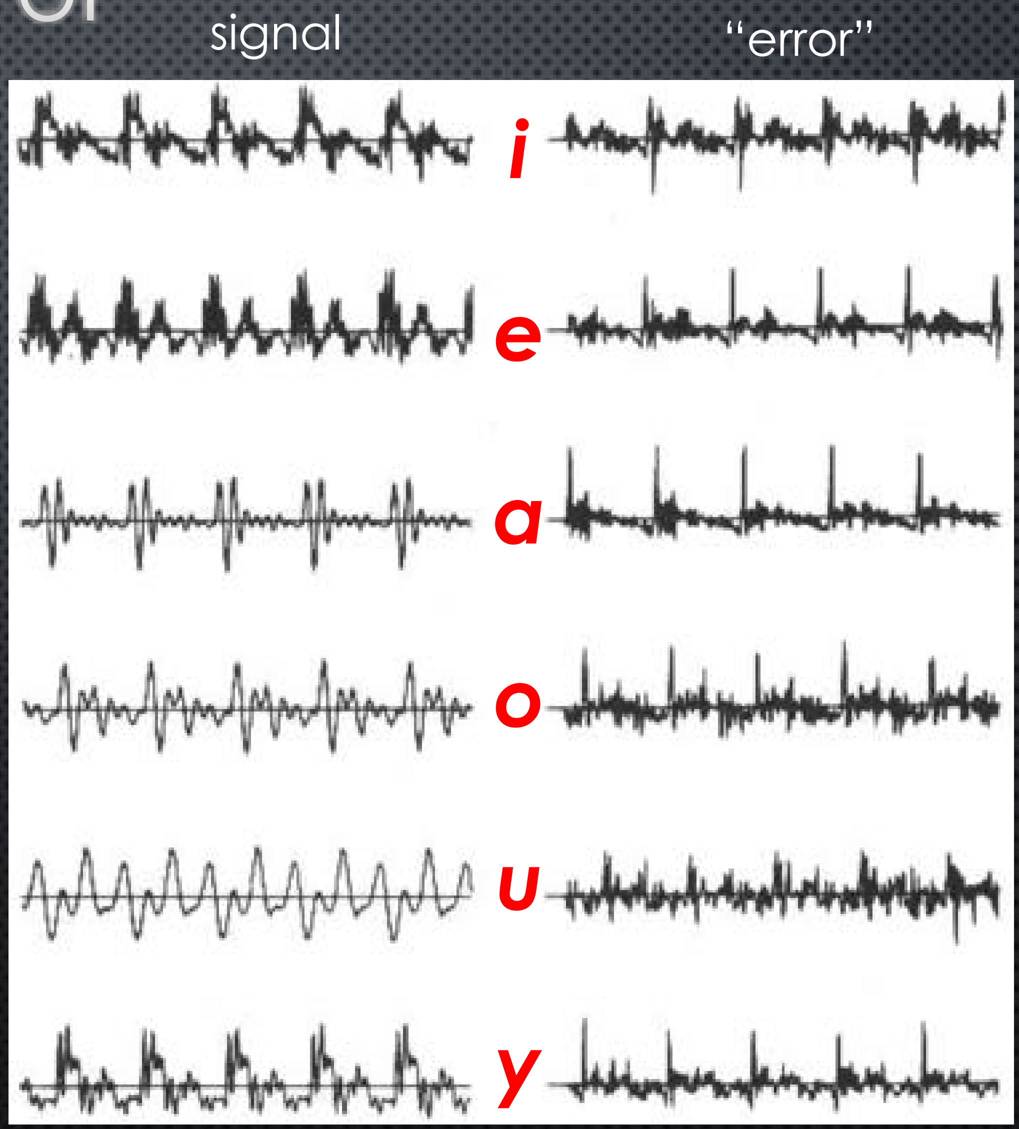
$$e(n) = y(n) - \hat{y}(n)$$

- Ideally, we want $e(n)$ to be as small as possible i.e. our prediction is near perfect

Prediction error

- Even if we have the best set of coefficients, the error may still be non-zero. Why?

Prediction error $e(n)$ covaries with signal



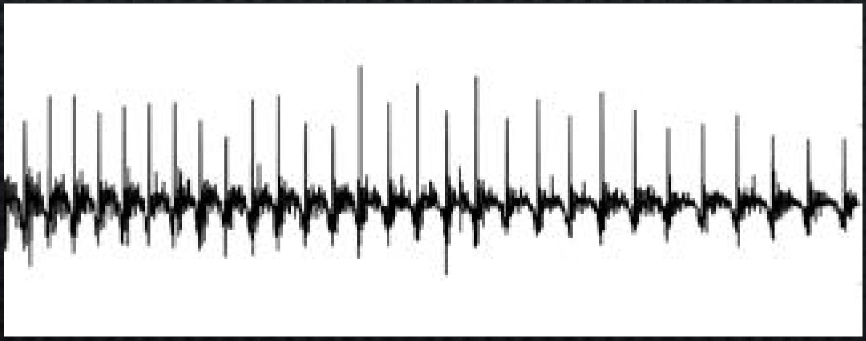
The source

- LPC models the vocal tract filter

filter

- The source is accounted for by the error $e(n)$
  - $e(n)$ is not really an error at all!
  - $e(n)$ can be used as the basis for F0 estimation

source



LPC remarks

- Not good for sounds like nasals
- Need to choose number of coefficients
- C in LPC stands for ‘coding’ – coefficients can be used as a more efficient means of representing a signal.
- Still need to code the error signal, or we could use a different one all together