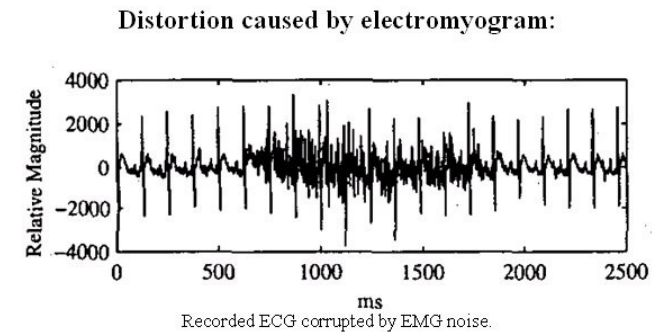


# Signal segmentation and waveform characterization

Biosignal processing I, 521273S

# Short-time analysis of signals

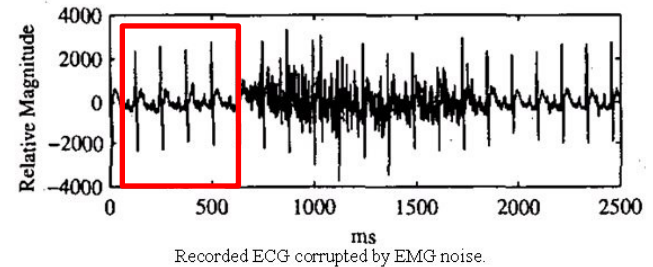
- Signal statistics may vary in time: nonstationary
  - how to compute signal characterizations?
- Signal must be partitioned into homogeneous segments
  - Operation is called segmentation
  - Statistics remain approximately the same in each segment: quasi-stationary segments
  - Characteristics is then computed for each segment separately
- Fixed and adaptive segmentation



# Fixed segmentation

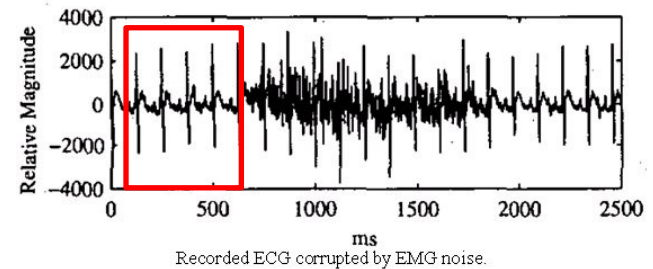
- Partitioning of a signal into fixed-duration segments of length  $M$  samples
  - $M$  must be selected manually (through trial and error, or via domain knowledge)
  - Each segment is then analyzed for some characteristics of interest
  - Segments can have partial overlap to smooth out the abrupt changes between them
- An example: Short-Time Fourier Transform (STFT)
  - STFT is computed for each segment (FFT, windowing, zero-padding)
  - Power spectrum is usually then computed for analyzing rhythmic components
  - Partial overlap of segments is often used for smoother output: e.g. 50%

Distortion caused by electromyogram:



Segment windows:  
no overlap

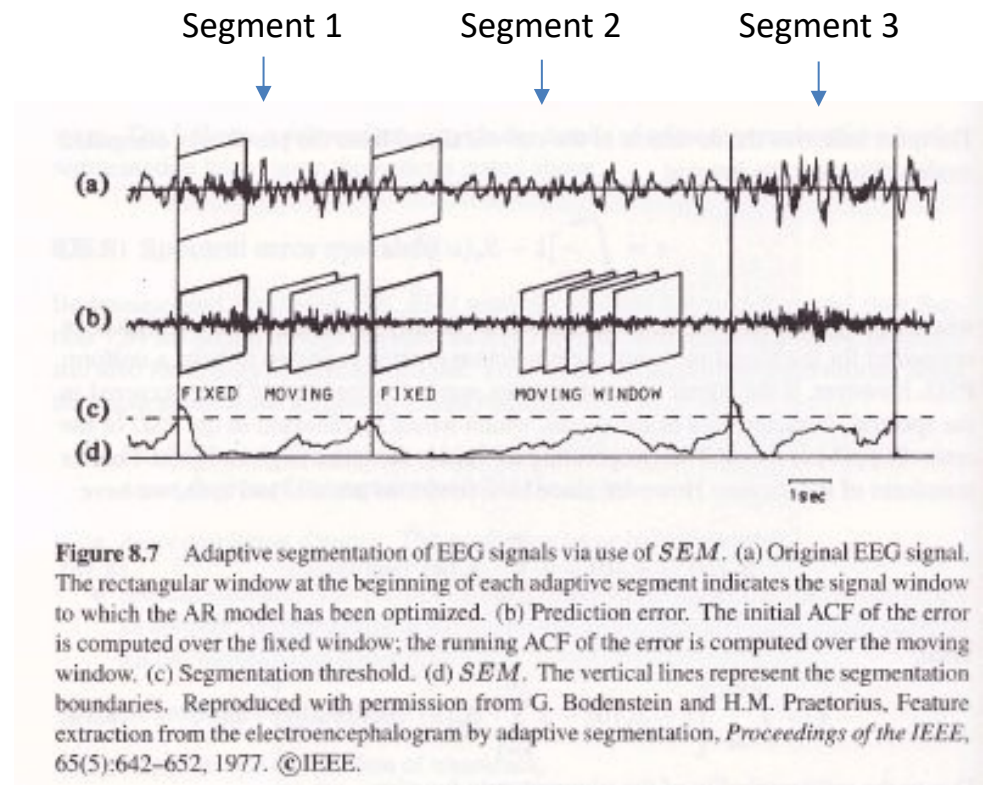
Distortion caused by electromyogram:



Segment windows:  
partial overlap

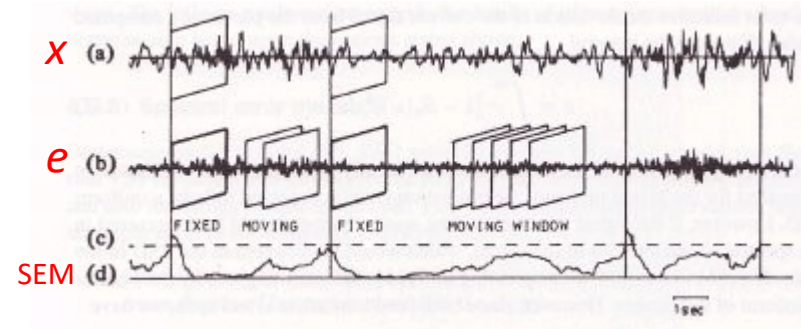
# Adaptive segmentation

- Partitioning of a nonstationary signal into quasi-stationary segments of variable duration
- Typically, a small window (anchor window) is set in the beginning of a segment, and a second window (test window) slides forward for comparisons with the anchor
  - Difference in the contents of the two windows can be measured from their spectrums
  - If the difference exceeds a preset threshold, the ending boundary of the segment has been detected
  - A new anchor window is set, and the process starts over again



# Spectral Error Measure method (SEM)

- SEM applies autoregressive modeling (AR) of the signal spectrum
  - All-pole signal modeling in stochastic process theory: useful when signal contains oscillating components in one or more frequencies
  - Linear prediction of signal samples from  $p$  previous samples
- AR( $p$ ) model is computed from the signal contained in the **anchor window** (at time index  $n=0$  in SEM equation)
  - Anchor window and sliding window are  $N$  samples long
- The AR( $p$ ) model based linear prediction of signal  $x_t$  is applied to the **test window** (at time  $n$ ) to generate prediction error signal  $e_t$  for the test window
- Autocorrelation function  $\phi_e(n, m)$  of  $e_t$  is computed
- SEM value is computed from the autocorrelation function
  - First term compares error signal power in the windows 0 and  $n$
  - Second term considers prediction error whiteness in window  $n$
  - $1 < M \leq N$ , to be specified by application design
- If the SEM value is larger than a preset threshold, the current segment ends at  $n$ 
  - Reset the anchor window at  $n+1$  to initialize a new segment
  - Establish a new AR model and start sliding the test window
- Otherwise, the current segment still continues
  - Increment time index  $n$  to slide the test window to the next sample
  - Compute error signal and autocorrelation function and SEM value for the new window



$$x_t = \sum_{k=1}^p a_k(t) x_{t-k} + e_t$$

$$e_t = x_t - \sum_{k=1}^p a_k(t) x_{t-k}$$

$$\phi_e(n, m) = \frac{1}{N} \sum_{n=0}^{N-|m|-1} e_t(n) e_t(n+m)$$

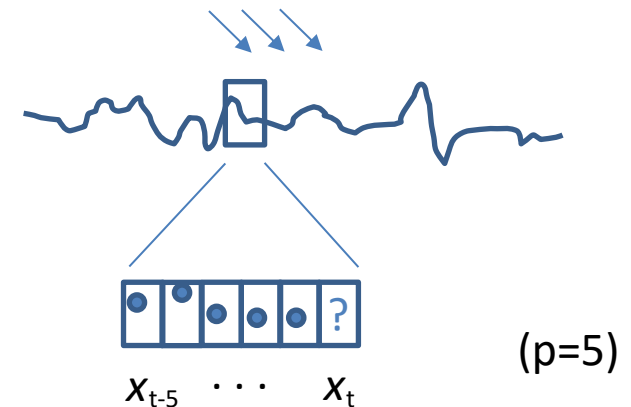
$$SEM(n) = \left[ \frac{\phi_e(0, 0)}{\phi_e(n, 0)} - 1 \right]^2 + 2 \sum_{m=1}^M \left[ \frac{\phi_e(n, m)}{\phi_e(n, 0)} \right]^2$$

# SEM cont'd: explanations

- **Linear prediction** with AR(p) model
  - Establish an AR(p) model for the signal  $x$ 
    - Set the desired model order  $p$
    - Optimize the parameters  $a_k, k=1, \dots, p$
  - By using the model, predict each sample  $x_t$  by using the  $p$  previous samples  $x_{t-i}$
  - The model is not perfect which results in a prediction error  $e_t$ 
    - For example, white additive noise cannot be predicted, or
    - $p$  may be too small to enable modeling of all signal components

-> after going through the entire signal in the window, we get a **prediction error signal  $e$**

predict each sample



$$x_t = \sum_{k=1}^5 a_k(t) x_{t-k} + e_t$$

Real value

Predicted value

Prediction error

The equation shows the real value  $x_t$  on the left. The sum  $\sum_{k=1}^5 a_k(t) x_{t-k}$  is bracketed and labeled 'Predicted value'. The error term  $e_t$  is labeled 'Prediction error'. Arrows point from these labels to their respective parts in the equation.

# SEM cont'd: explanations

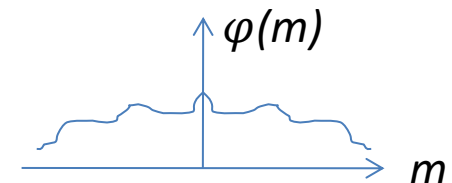
- **Autocorrelation function of prediction error signal**: the correlation of the error signal with itself at different time lags
  - self-similarity of the error signal as a function of time lag  $m$
- It is a mathematical tool for finding repeating patterns, such as the presence of a periodic signal obscured by noise
- Signal power spectrum  $P(\omega)$  can be computed by Fourier transforming the autocorrelation function

Signal segment and its copy:



Time lag  $m$

$$\varphi(m) = \frac{1}{N} \sum_{n=0}^{N-|m|-1} e(n)e(n+m)$$



$$P(\omega) = \sum_{m=-(N-1)}^{N-1} \varphi(m)e^{-j\omega m}$$

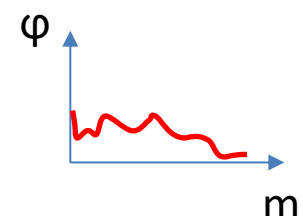
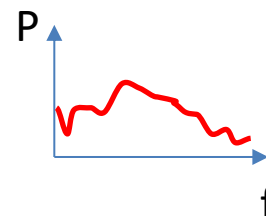
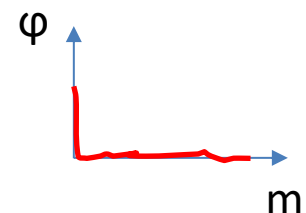
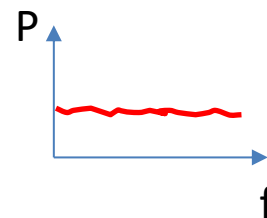
# SEM cont'd: explanations

- If the signal **follows** the set AR model, the prediction error  $e_t$  is white noise
  - The model then explains all structure in the signal (still in the same segment!)
  - Flat power spectrum  $P$  of  $e_t$
  - Autocorrelation function of  $e_t$ : close to Dirac delta function

$$\varphi_e(n, m) > 0, \text{ only for } m = 0$$

- If the signal **does not follow** the set AR model, the prediction error  $e_t$  is not white noise
  - There is some structure in the prediction error signal (possibly the next segment!)
  - Power spectrum  $P$  is not flat for  $e_t$
  - Autocorrelation function of  $e_t$  is more complex containing many non-zero values

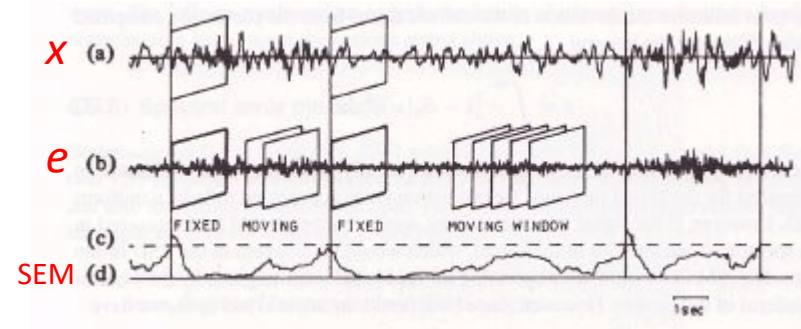
$$x_t = \sum_{k=1}^p a_k(t)x_{t-k} + e_t$$





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$$SEM(n) = \left[ \frac{\phi_e(0, 0)}{\phi_e(n, 0)} - 1 \right]^2 + 2 \sum_{m=1}^M \left[ \frac{\phi_e(n, m)}{\phi_e(n, 0)} \right]^2$$

# Three methods of plotting STFT-spectrums of consecutive segments

- Separate sub-plots of spectrums
- Landscape plot of spectrums
- Spectrograms of spectrums

# STFT example: phonocardiogram

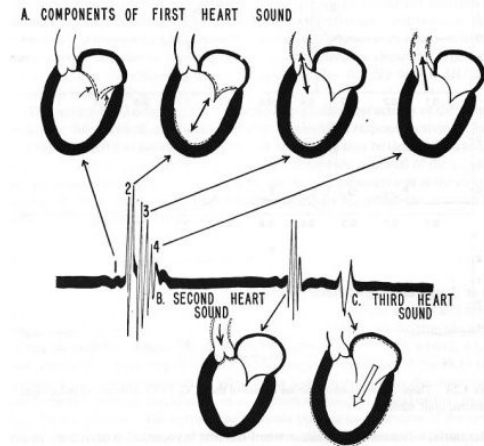
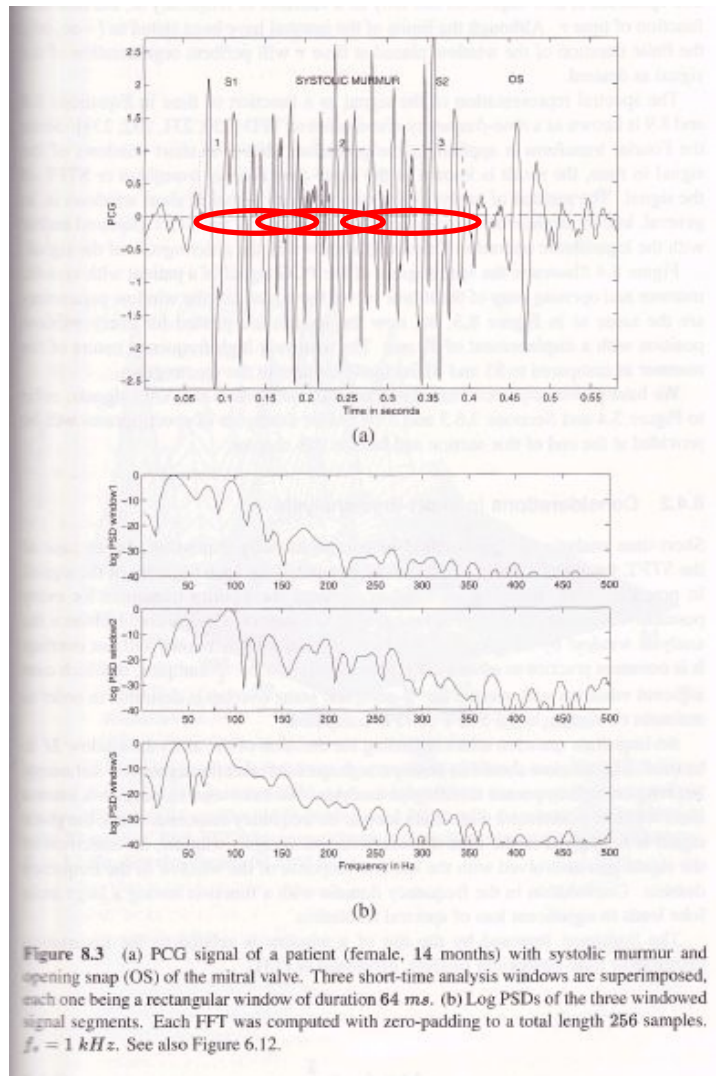


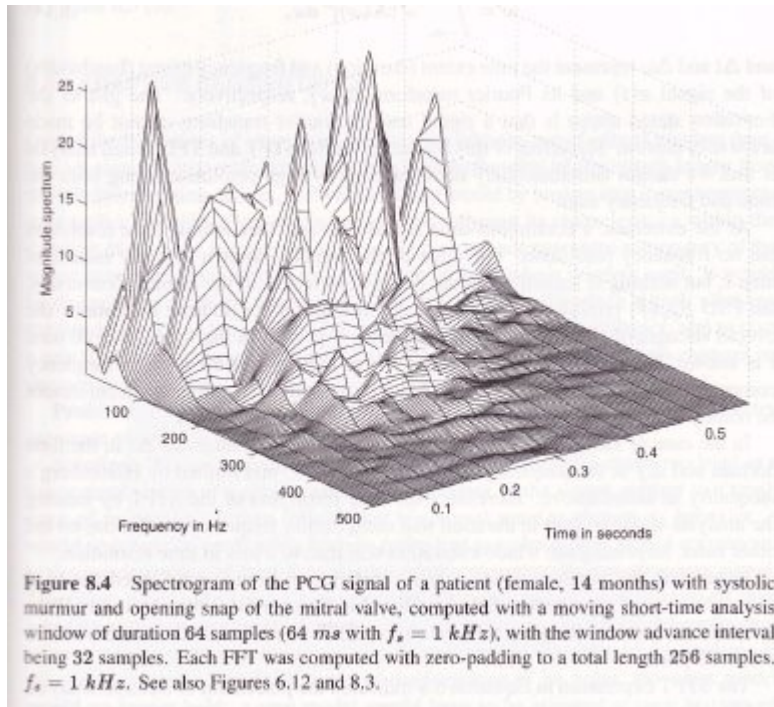
Figure 1.25 Schematic representation of the genesis of heart sounds. Only the left portion of the heart is illustrated as it is the major source of the heart sounds. The corresponding events in the right portion also contribute to the sounds. The atria do not contribute much to the heart sounds. Reproduced with permission from R.F. Rushmer, *Cardiovascular Dynamics*, 4th edition, ©W.B. Saunders, Philadelphia, PA, 1976.

Logarithmic PSD (power spectral density) in three consecutive windows (64 ms), normalized power axis

$$P_{dB}(f) = 10 \log_{10} (P(f)/P_{\max})$$

- 64 sample windows with 32 sample overlap
- 256 sample FFT, zero-padding

# STFT example: phonocardiogram



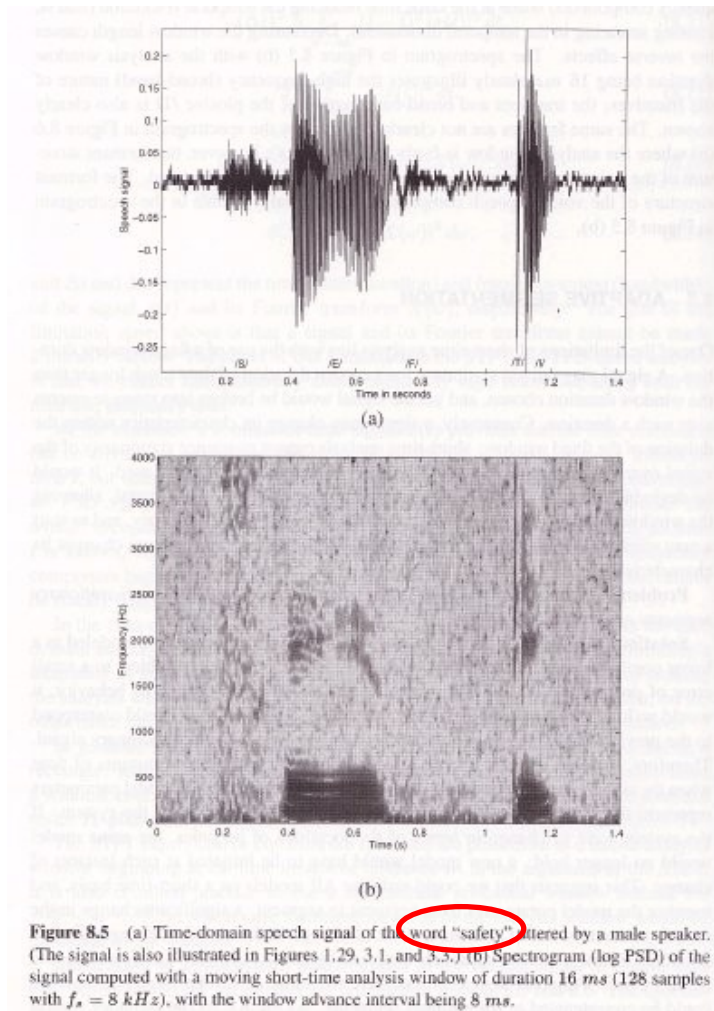
An alternative representation of the spectrogram: interpolated magnitude spectrum as a **landscape plot**

For example:

- 64 sample windows with 32 sample overlap
- 256 sample FFT, zero-padding

More spectrums can be visualized than if plotted separately

# STFT example: speech signal



An alternative representation of PSD in a sliding window: **spectrogram** - gray-level image of power spectrums

- Horizontal axis: start time of window
- Vertical axis: frequency
- Pixel value: signal power at time  $t$  and at frequency  $f$

For example:

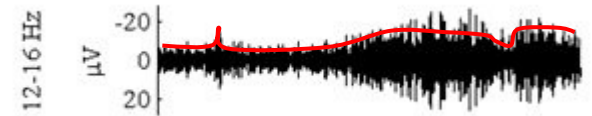
- 128 sample windows with 64 sample overlap
- 128 sample FFT

The most dense representation of spectral information

Activity level of signal

# Envelope extraction

- Estimation of trends in signal activity or energy
  - Amplitude envelope
- A simple method:
  - Full-wave rectify the signal (taking absolute value)
  - Smooth the result (moving average, Butterworth low-pass filter,...)



$$y(t) = \text{smooth}(|x(t)|)$$

# Envelopogram

- Envelopogram estimate: magnitude of the analytic signal  $y(t)$  formed using  $x(t)$  and its Hilbert transform  $x_H(t)$

$$y(t) = x(t) + jx_H(t)$$

$$x_H(t) = \int_{-\infty}^{\infty} \frac{x(\tau)}{\pi(t - \tau)} d\tau$$

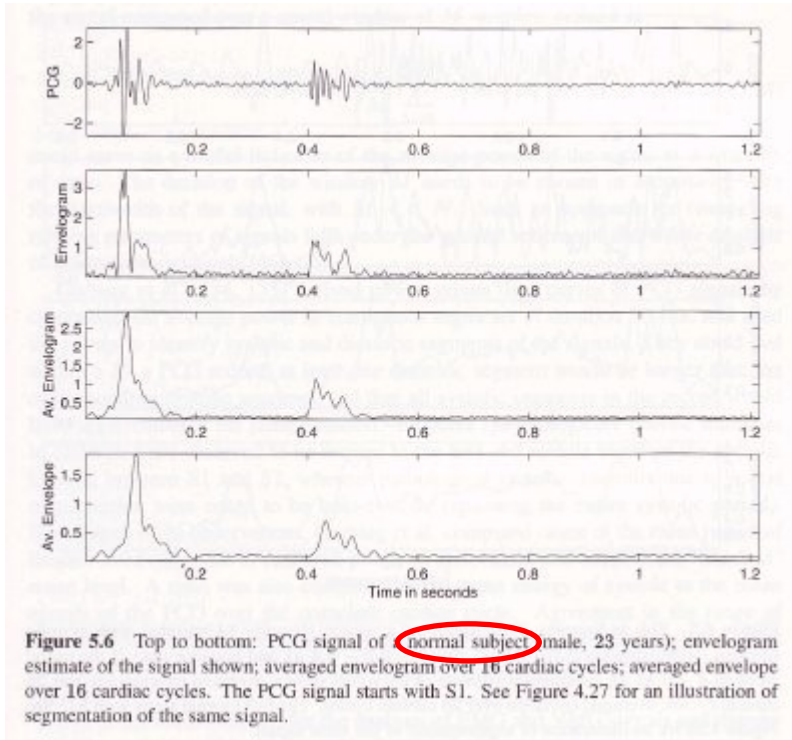
$$|y(t)| = \sqrt{x(t)^2 + x_H(t)^2}$$

- Practical algorithm:
  1. Compute DFT of  $x(t)$
  2. Set the negative-frequency terms to zero
  3. Multiply positive-frequency terms by 2
  4. Compute inverse DFT of the result
  5. The magnitude of the result gives the envelopogram estimate

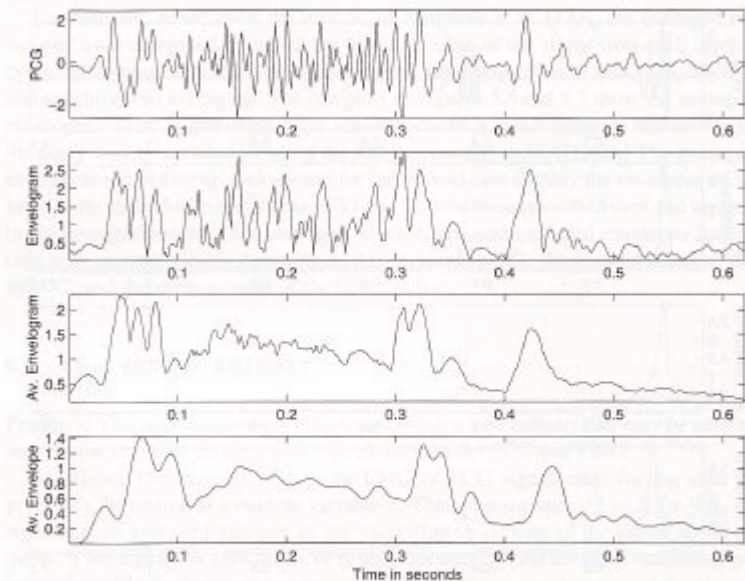
(DFT = Discrete Fourier Transform)



# Envelope examples (phonocardiogram)



- Signal
- Envelopegram of the signal
- Synchronously averaged envelopegram (N=16)
- Synchronously averaged envelope (N=16)



- Signal
- Envelopegram of the signal
- Synchronously averaged envelopegram (N=26)
- Synchronously averaged envelope (N=26)

# Other descriptors of activity level

- Root-mean-square (RMS) value of the signal window
  - Computed over a sliding window of length N
- Zero-crossing rate (ZCR): number of times the signal crosses zero-value per time unit
- Turns count: number of times that the signal amplitude changes direction per time unit

$$RMS = \left[ \frac{1}{N} \sum_{n=0}^{N-1} x^2(n) \right]^{\frac{1}{2}}$$

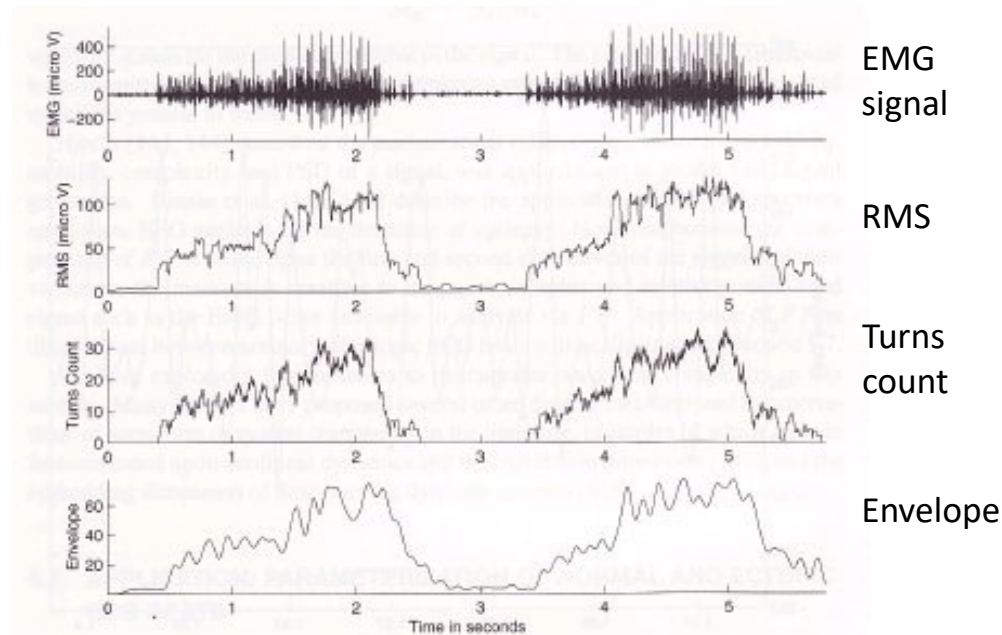


Figure 5.8 Top to bottom: EMG signal over two breath cycles from the neural diaphragm of a dog recorded via implanted fine-wire electrodes; short-time RMS values; turns count using Willison's procedure; and smoothed envelope of the signal. The RMS and turns count values were computed using a causal moving window of 70 ms duration. EMG signal courtesy of R.S. Platt and P.A. Easton, Department of Clinical Neuroscience, University of Calgary.