

COM3502/4502/6502 SPEECH PROCESSING

Lecture 12

Frequency Analysis



1

Time vs. Frequency Domain

- A signal may represent a sequence of *any* kind of measurement
- It is common for signals to represent a sequence of measurements in *time*
- A useful technique for signal analysis is to decompose it into a set of basic pieces that are easier to understand (e.g. *linear trends*)
- Decomposing into *periodic* functions can be thought of as transforming a signal into the '**frequency domain**' (i.e. *repetitions in time can be expressed as a frequency*)

$$f = 1/\tau \text{ cycles-per-second/Hertz}$$



2

Fourier Analysis



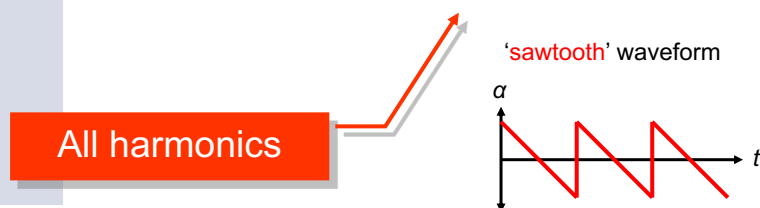
- Introduced by **Joseph Fourier** (1768–1830)
- **Theorem:** any periodic signal of frequency f_0 can be constructed exactly by adding together '**sinusoids**' (*sine waves*) with frequencies ...
 - $f_0, 2f_0, 3f_0, 4f_0, 5f_0$, etc.
- Each sinusoid in this '**Fourier series**' is characterised by its ...
 - frequency
 - amplitude
 - phase
- f_0 is termed the '**fundamental frequency**'
- $2f_0, 3f_0, 4f_0$, etc. are termed the '**harmonics**'

3

Fourier Analysis

$$a(t) = \frac{2}{\pi} \sum_{k=1}^{\infty} \frac{\sin(k\omega t)}{k}$$

$$= \frac{2}{\pi} \left(\sin(\omega t) + \frac{\sin(2\omega t)}{2} + \frac{\sin(3\omega t)}{3} + \dots \right)$$



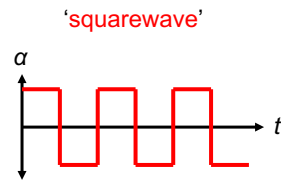
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Fourier Analysis

$$a(t) = \frac{4}{\pi} \sum_{k=1}^{\infty} \frac{\sin((2k-1)\omega t)}{(2k-1)}$$

$$= \frac{4}{\pi} \left(\sin(\omega t) + \frac{\sin(3\omega t)}{3} + \frac{\sin(5\omega t)}{5} + \dots \right)$$

Odd harmonics



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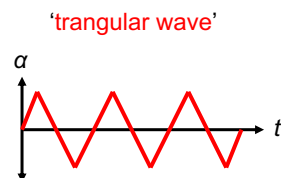
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Fourier Analysis

$$a(t) = \frac{8}{\pi^2} \sum_{k=0}^{\infty} (-1)^k \frac{\sin((2k+1)\omega t)}{(2k+1)^2}$$

$$= \frac{8}{\pi^2} \left(\sin(\omega t) - \frac{\sin(3\omega t)}{9} + \frac{\sin(5\omega t)}{25} - \dots \right)$$

Odd harmonics
(alternating sign)



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Fourier Analysis

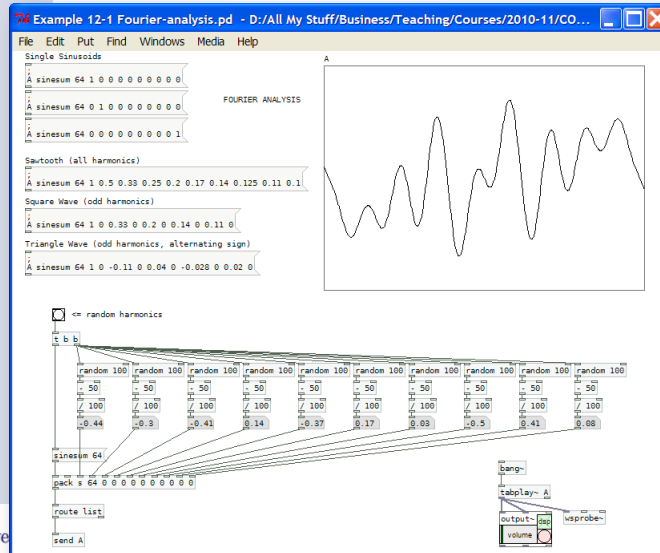
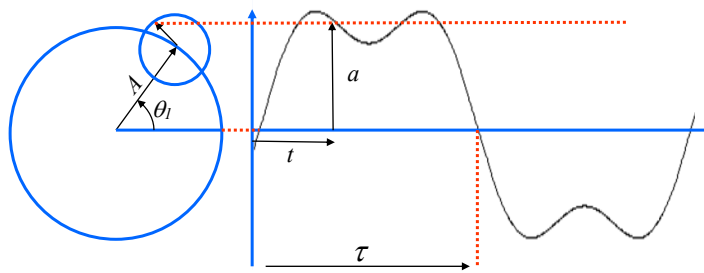


Figure 12, slide 7

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Fourier Analysis

Projection of multiple vectors rotating at constant velocities



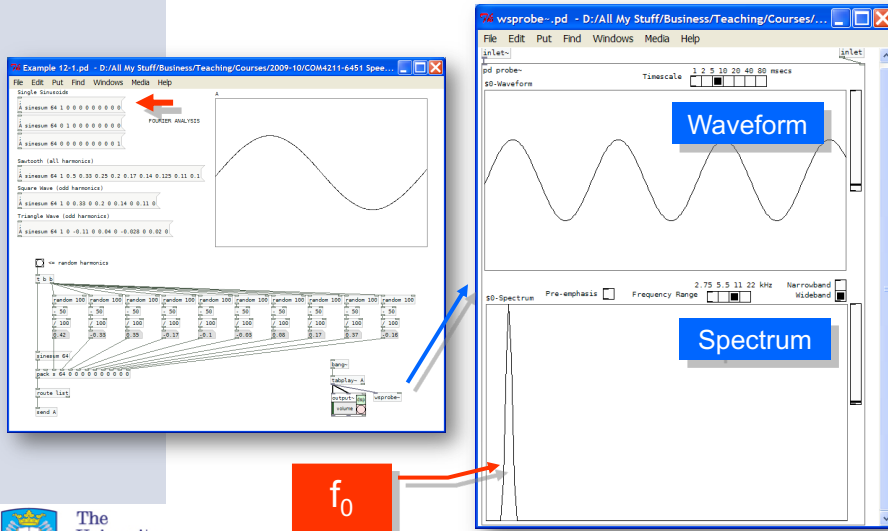
- A magnitude of 1st rotating vector
- θ phase angle of 1st vector (*radians*)
- $f_0 = 1/\tau$ fundamental frequency (*cycles per second, Hertz*)
- $\omega = 2\pi f_0$ fundamental angular frequency (*radians/second*)
- $\theta = \omega t$
- $a = A \cdot \sin(\omega t) + A/3 \cdot \sin(3\omega t)$



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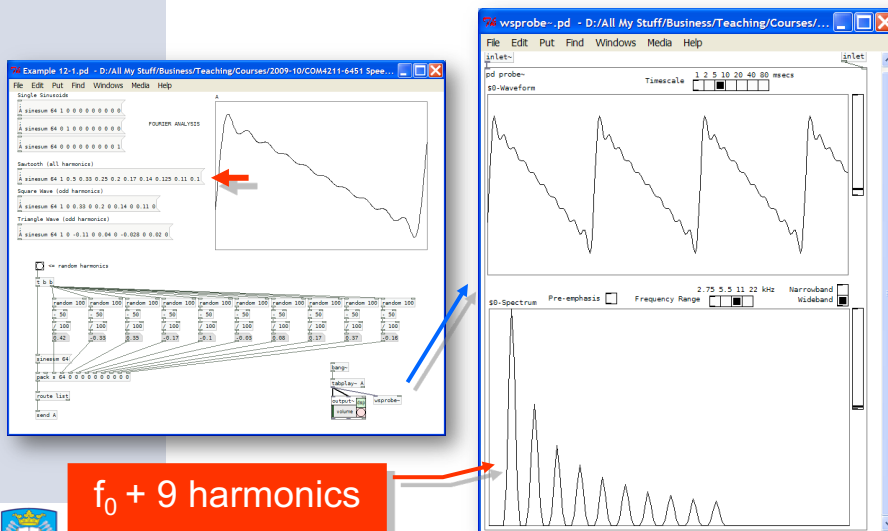
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The Spectrum



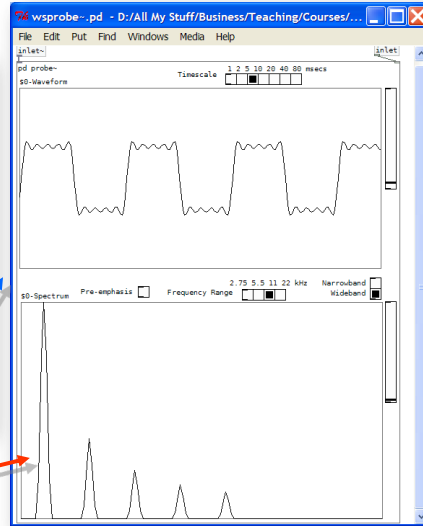
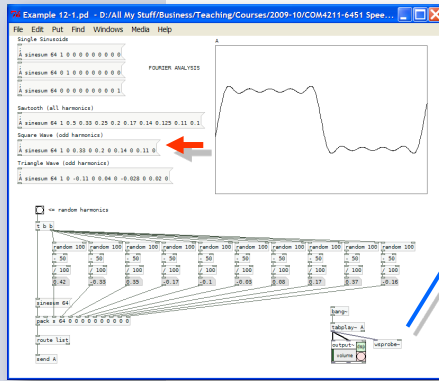
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The Spectrum



12

The Spectrum

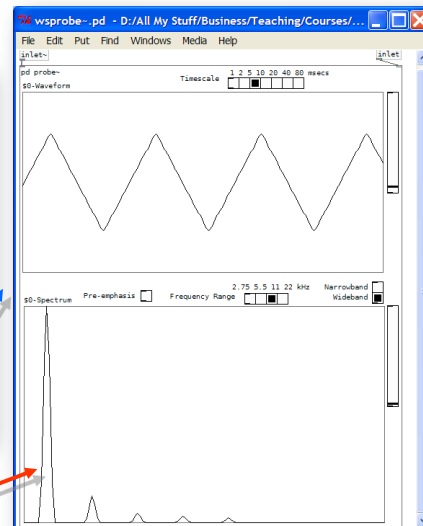
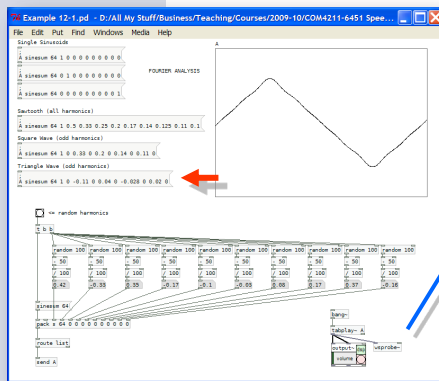


$f_0 + \text{odd harmonics}$



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The Spectrum

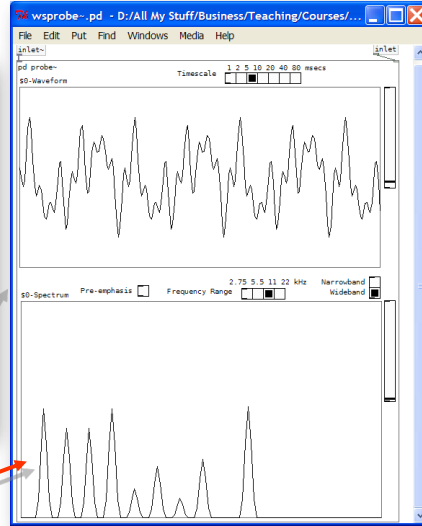
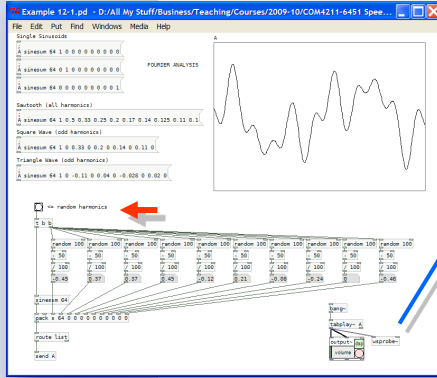


$f_0 + \text{odd harmonics, alternating sign}$



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The Spectrum

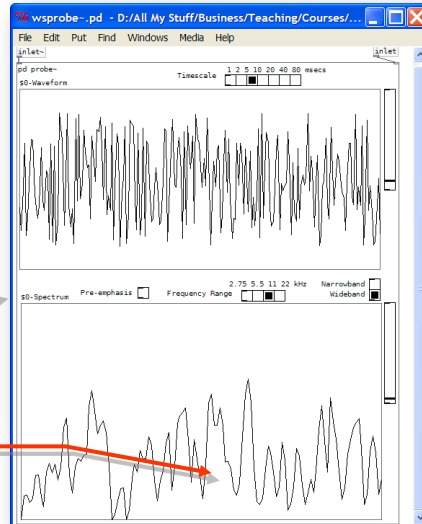
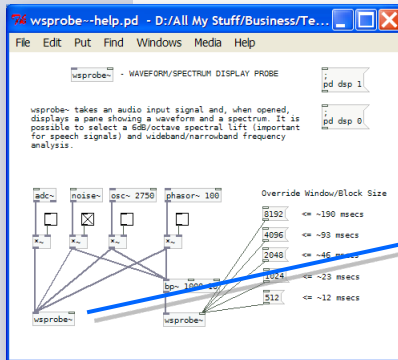


$f_0 + 9$ random harmonics



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The Spectrum



Energy at all frequencies



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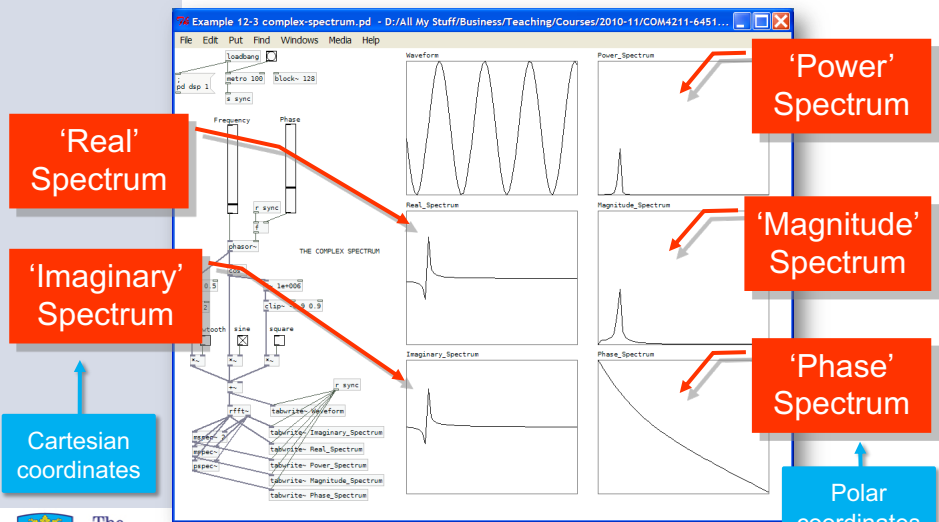
The Power Spectrum

- Previous examples have illustrated the 'power spectrum', i.e. the energy present at each frequency
- The power of a signal at each frequency is the sum of the squares of its real and imaginary components (e.g. a sine wave has varying amplitude but constant power)
- Information (*the phase relationships*) is lost during the calculation of the power spectrum, so this means that the original signal cannot be recovered accurately
- However, a signal's spectrum is fully specified by the magnitude and phase at each frequency ... this is the 'complex spectrum'



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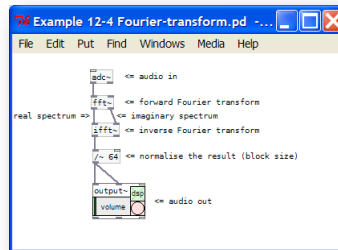
The Complex Spectrum



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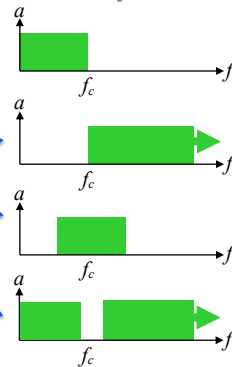
The Fourier Transform

- The complex (magnitude and phase) spectrum is obtained from a time signal by means of the 'Fourier transform'
- No information is lost during the transformation
- This means that the time signal can be recovered from the complex spectrum by means of the 'inverse Fourier transform'



Filters

- A 'filter' modifies the properties of a signal
- A filter's effect can be analysed in the time domain or in the frequency domain
- A filter is typically characterised by its 'frequency response'
- Types of filter ...
 - low-pass
 - high-pass
 - band-pass
 - band-stop



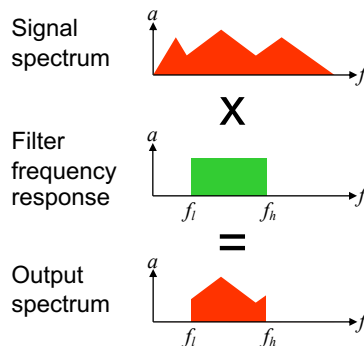
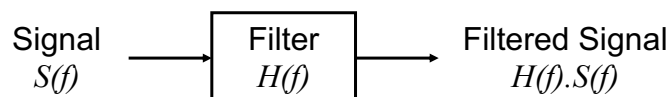
Filters

- The '**frequency response**' of a *low-pass* or a *high-pass* filter is characterised by ...
 - the '**cut-off frequency**' f_c (Hz)
 - the '**roll-off rate**' (dB/octave)
- A '**first-order filter**' attenuates at a roll-off rate of 6 dB/octave
- The frequency response of a *band-pass* filter is characterised by
 - the '**centre frequency**' f_c (Hz)
 - the '**bandwidth**' (Hz) or '**Q**' ($=cf/bw$)
- When a signal passes through a linear filter, its spectrum is *multiplied* by the frequency response of the filter



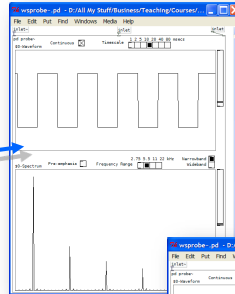
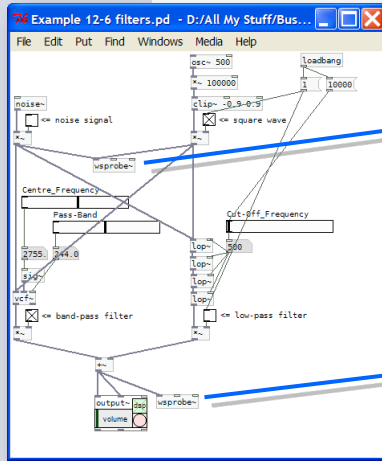
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Filters

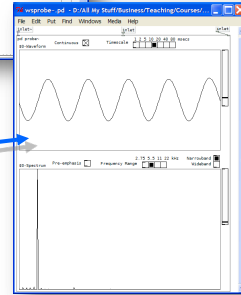


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Filters



Before filtering



After filtering



Impulse Response

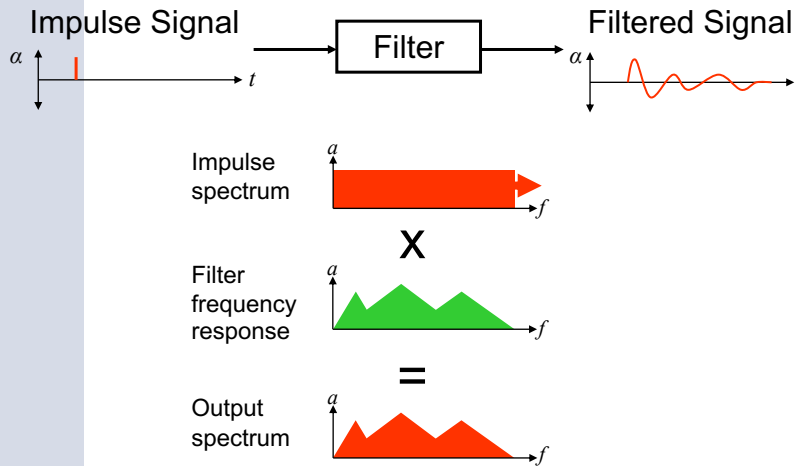
- An 'impulse' signal is an instantaneous pulse of energy
- An idealised impulse signal has a *flat* frequency spectrum (*i.e. equal energy at all frequencies*)



- Hence the 'impulse response' of a filter has a spectrum that is *equal* to its frequency response
- Note that white noise *also* has a flat spectrum



Impulse Response



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Impulse Response

Example 12-7 impulse-response... patch components: loadbang, metro 1000, impulse generator, bandpass filter, output, volume.

wsprobe window showing a time-domain waveform and a frequency spectrum plot with a peak at approximately 2.75 kHz.



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Source-Filter Modelling

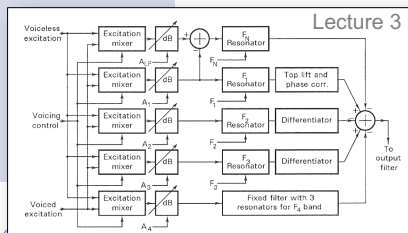
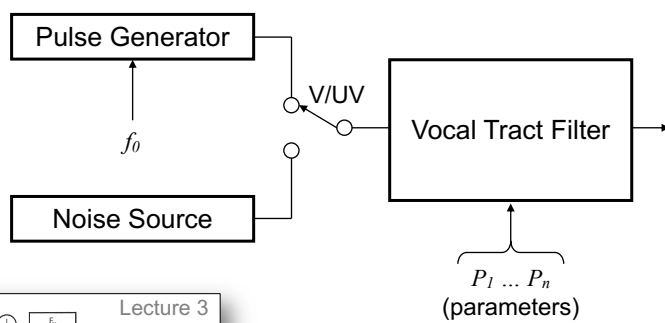
- If we design a filter with a frequency response equal to the spectrum of a vowel sound and input a sequence of impulses, we will generate the original vowel sound ...



- Also, if we design a filter with the frequency response of an unvoiced fricative and input a white noise signal, we will generate the original fricative sound ...



Source-Filter Model of Speech



This lecture has covered ...

- Fourier analysis
- Fundamental frequency
- Harmonics
- The power spectrum
- The complex spectrum
- The Fourier transform
- Filters
- Impulse response
- The source-filter model of speech



Any Questions ?



MOLE Quiz #1

QUIZ

- 16:00 Wednesday 13th November in North Campus PC room
- 20 questions (*multiple-choice, true/false, etc.*)
- Covers material from Lecture 1 to Lecture 10
- Worth 20% of the overall course mark
- 1hr in duration (*longer if you have an LSP in place*)
- You may start the test anytime up to the final deadline
- Only one attempt is permitted
- You may save your work part way through, but the clock will keep running
- You may continue after the time expires, but the system will record that you took more than the allocated time



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MOLE Quiz #1

QUIZ

- Any phonetic symbols are displayed using SAMPA
- The quiz is marked automatically, so avoid making any spelling mistakes (*e.g. it's "vocal cords" not "vocal chords"*)
- This is an individual 'open-book' assessment (*i.e. you may refer to your course notes, but you **must not collude with anyone** during the test*)
- Marks will appear immediately after submission
- Feedback (*i.e. the correct answers*) will appear one week after the deadline
- You may pack-up and leave when you are finished, but please do so quietly



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Next time ...

Digital Signals



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