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COM3502/4502/6502 SPEECH PROCESSING

Lecture 16 The 'Z' Transform

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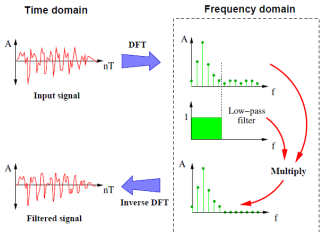
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Filtering

- Recall that a signal may be filtered by ...
 - transforming it into the frequency domain (*using DFT*)
 - multiplying the signal spectrum by the filter spectrum
 - transforming it back into the time domain (*using IDFT*)



- An alternative is to characterise the action of a filter in the time domain (*i.e. on the waveform itself*)
- This is done using '**difference equations**'

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Difference Equations

- Consider a linear filter where ...
 - $x[nT]$ is the input
 - $y[nT]$ is the output (*response*)

- '**Difference equations**' relate the current filter output to the current and past inputs and outputs

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Difference Equations: *an example*

Difference equation

→

$$y[nT] = ay[(n-1)T] + x[nT]$$

Equivalent processing diagram

→

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Difference Equations: *an example*

Difference equation

Input Signal $x[nT]$ → Filter → $y[nT]$ Output Signal

$$y[nT] = ay[(n-1)T] + x[nT]$$

Equivalent processing diagram

Input

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Difference Equations: *an example*

Difference equation

Input Signal $x[nT]$ → Filter → $y[nT]$ Output Signal

Result is a simple low-pass filter

$$y[nT] = ay[(n-1)T] + x[nT]$$

Equivalent processing diagram

Input

Output

$a < 1$

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Difference Equations: *an example*

Example 16-1 difference-equation.pd - D:/All My Stuff/Business/Teaching/C...

File Edit Put Find Windows Media Help

loadbang

pd dsp 1 metro 25

0.75

phasor~ 500

cos~

1e+006

clip~ -1 1

tabwrite~ Input

clear

0.866

fexpr~ \$f2*\$y+\$x ← difference equation

Gain

tabwrite~ Output

Input

Output

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Linear Time-Invariant Filters

- A **linear filter** obeys the principle of super-position ...
 - if $x_1[kT]$ yields $y_1[kT]$
 - and input is $ax_1[kT] + bx_2[kT]$
 - then $y[kT] = ay_1[kT] + by_2[kT]$
- The response of a **time-invariant filter** is the same for all times ...
 - if $x[kT]$ yields $y[kT]$
 - then $x[(k-k_0)T]$ yields $y[(k-k_0)T]$

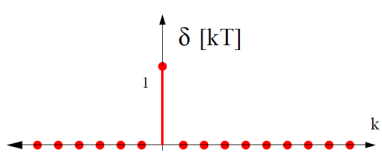
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Linear Time-Invariant Filters

- Linearity and time-invariance allows the input (*and the response*) of a filter to be constructed from a weighted sum of time-shifted impulses $\delta[kT]$ where $\delta[kT]$ is 1 when $k = 0$ (and 0 otherwise)



- Time-invariance and super-position thus allows the input signal to be expressed as ...

$$x[kT] = \sum_{n=-\infty}^{\infty} x[nT] \delta[(k-n)T]$$

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Linear Time-Invariant Filters

- Given the **'impulse response'** of a filter $h[kT]$, the output is ...

$$y[kT] = \sum_{n=-\infty}^{\infty} x[nT] h[(k-n)T]$$

- This is the **'discrete convolution'** of the input signal with the filter impulse response, and it can be written as ...

$$y[kT] = x[kT] * h[kT]$$

convolution operator

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$$y[0] = \sum_{i=-4}^4 h[i] \cdot x[0-i] = 1 \cdot 5 = 5$$

$$y[1] = \sum_{i=-4}^4 h[i] \cdot x[1-i] = 1 \cdot 5 + 1 \cdot 4 = 9$$

$$y[6] = \sum_{i=-4}^4 h[i] \cdot x[6-i] = 1 \cdot 2 + 1 \cdot 1 = 3$$

$$h[n] * x[n] = \sum_i h[i] \cdot x[n-i]$$

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Convolution

<http://en.wikipedia.org/wiki/Convolution>

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The Z-Transform

- Analysis of filters in the time domain (*i.e. using difference equations*) is cumbersome
- The '**Z-transform**' is a power series representation of a discrete-time sequence
- E.g. for the sequence $x[0] x[1] x[2] x[3]$, the Z transform simply multiplies each coefficient in the sequence by a power of z corresponding to its index ...

$$X(z) = x[0]z^0 + x[1]z^{-1} + x[2]z^{-2} + x[3]z^{-3}$$
- By convention, *negative* powers of z are used for *positive* time indices (*i.e. z^{-n} represents a delay of n samples*)

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The Z-Transform

- For a general sequence, the Z transform is written as ...

$$X(z) = \sum_{k=-\infty}^{\infty} x[kT]z^{-k}$$
- The transform is denoted by the $Z\{\cdot\}$ operator ...

$$X(z) = Z\{x[kT]\}$$

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The Z-Transform

Key properties of the transform are ...

- linearity

$$Z\{ax_1[k] + bx_2[k]\} = aZ\{x_1[k]\} + bZ\{x_2[k]\}$$
- time-shifting

$$Z\{x[k - k_0]\} = z^{-k_0} Z\{x[k]\}$$
- convolution

$$Z\{x[k] * y[k]\} = Z\{x[k]\} Z\{y[k]\}$$

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The Z-Transform

- As we saw, the output of a linear filter is described by the discrete convolution ...

$$y[k] = x[k] * h[k]$$
- This can be expressed using the Z transform as a product ...

$$Y[z] = X[z] H[z]$$
- Hence the '**z-plane transfer function**' $H[z]$ may be written as ...

$$H[z] = \frac{Y[z]}{X[z]}$$

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Common Z-Transform Pairs

Unit impulse

$h[k] = \delta[k]$

$\Rightarrow H(z) = 1$

Shifted impulse

$h[k] = \delta[k-m]$

$\Rightarrow H(z) = z^{-m}$

Unit step

$h[k] = u[k]$

$\Rightarrow H(z) = \frac{1}{1-z^{-1}}$

Decaying exponential

$h[k] = a^k u[k] ; |a| < 1$

$\Rightarrow H(z) = \frac{1}{1-a \cdot z^{-1}}$

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Link with the Fourier Transform

Z-Transform

$\rightarrow Z\{x[kT]\} = \sum_{k=-\infty}^{\infty} x[kT]z^{-k}$

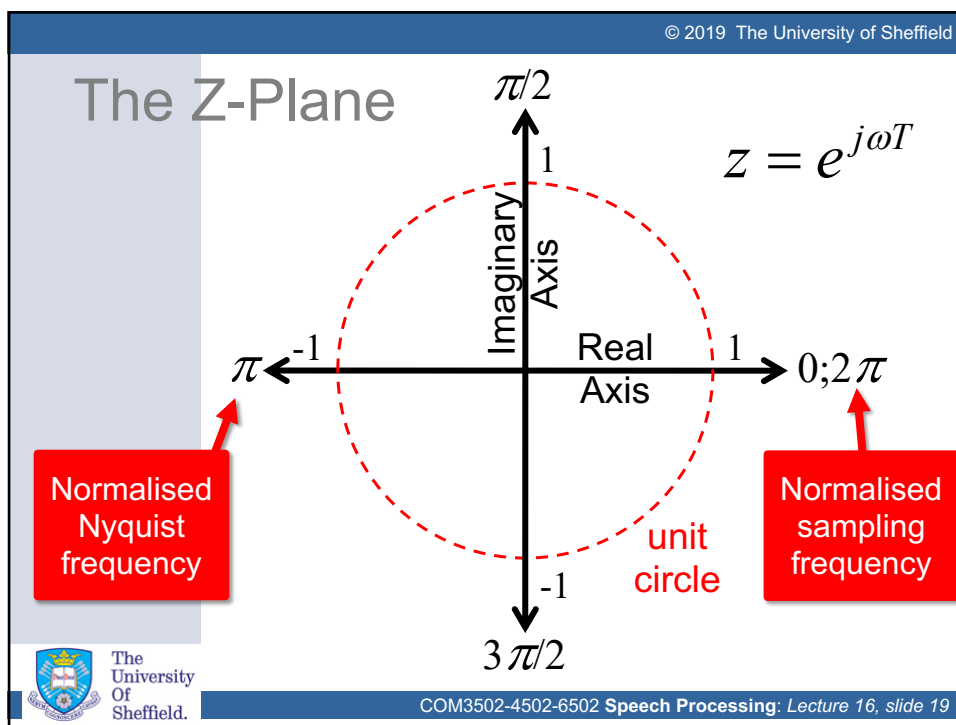
- If $z > 1$, then the transform is equivalent to multiplying the sequence by a *rising* exponential
- If $0 < z < 1$, then the transform is equivalent to multiplying the sequence by a *falling* exponential
- If z is a complex number *lying on the unit circle*, then the transform is equivalent to multiplying the sequence by a real cosine and imaginary sinusoid
i.e. this is directly equivalent to the Fourier transform!
- Hence, given a filter transfer function $H[z]$, the filter frequency response can be evaluated at any frequency ω by setting ...

$z = e^{j\omega T}$

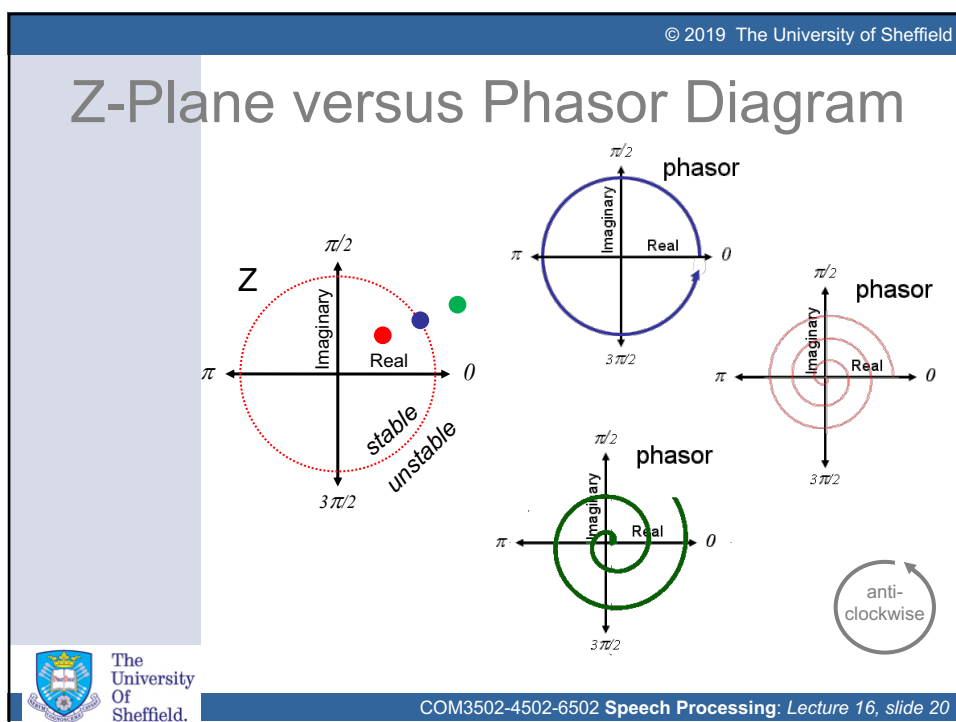
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Poles and Zeros

- Linear filter transfer functions (*arising from general difference equations*) can be written as the ratio of two polynomials in z ...

$$H[z] = \frac{P[z]}{Q[z]}$$
- The values of z for which $P[z]=0$ are known as the '**zeros**' of $H[z]$
- The values of z for which $Q[z]=0$ are known as the '**poles**' of $H[z]$
- Poles correspond to frequencies at which a filter transfer function tends to infinity (*i.e. 'resonances'*)
- Zeros correspond to frequencies at which a filter transfer function tends to zero (*i.e. 'anti-resonances'*)
- A filter is completely characterised by its poles and zeros

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Poles and Zeros

- The z -domain polynomials are often written in terms of their poles and zeros z_i ...

$$P[z] = \prod_{i=1}^p (z - z_i) \quad Q[z] = \prod_{i=1}^q (z - z_i)$$
- The poles and zeros correspond to the '**roots**' of the difference equations
- They can be visualised by plotting their positions in the '**Z-Plane**' ...

Zeros

Poles

Zeros are signified by 'o's'

Poles are signified by 'x's'

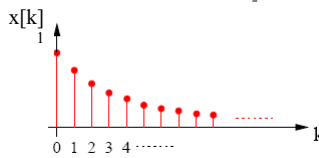
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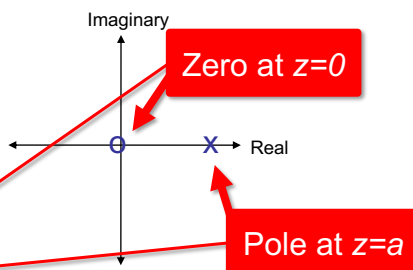
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Poles and Zeros: *an example*

$x[k] = 0$ for $k < 0$
 $x[k] = a^k$ otherwise



$$\begin{aligned}
 X(z) &= \sum_{k=-\infty}^{\infty} x[k] z^{-k} \\
 &= \sum_{k=0}^{\infty} a^k z^{-k} \\
 &= \sum_{k=0}^{\infty} (a \cdot z^{-1})^k \\
 &= \frac{1}{1 - a \cdot z^{-1}} \\
 &= \frac{z}{z - a}
 \end{aligned}$$


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Poles and Zeros

- The magnitude response of a filter can be quickly understood based on the location of its poles and zeros
- By starting with a pole/zero plot, it is possible to design a filter and obtain its transfer function very easily ...

$$\begin{aligned}
 H[z] &= \frac{\prod_{i=1}^p (z - z_i)}{\prod_{i=1}^q (z - z_i)} \\
 &= \frac{\prod \text{"distance of zeros from unit circle"}}{\prod \text{"distance of poles from unit circle"}}
 \end{aligned}$$

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Filter Frequency Response

- The frequency response of a filter can be plotted using the distances from the 'unit circle' to the poles and zeros
- While moving around the unit circle ...
 - if close to a zero, then the magnitude is small
 - if close to a pole, then the magnitude is large
- If a *zero* is on the unit circle, then the frequency response is zero at that point
- If a *pole* is on the unit circle, then the frequency response goes to infinity at that point

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Filter Response: *an example*

- Consider the simple averaging filter with the following difference equation ...


$$y[n] = ay[n-1] + x[n]$$
- Taking Z transforms, this becomes ...

$$Y = aYz^{-1} + X$$
- Hence the transfer function (Y/X) is given by ...

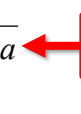
$$Y - aYz^{-1} = X$$

$$Y(1 - az^{-1}) = X$$


$$\frac{Y}{X} = \frac{1}{1 - az^{-1}} = \frac{z}{z - a}$$



Zero at $z=0$



Pole at $z=a$

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Filter Response: *real pole*

(normalised Nyquist)

Unit Circle

Spectrum

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Filter Response: *real pole*

Example 16-2 difference-equation-spectrum.pd

wsprobe.pd

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Filter Response: *complex pole*

(normalised Nyquist)

Imaginary

Real

ω

π

0

Spectrum

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Poles & Zeros in Pd

- As well as [fexpr~], Pd provides several 'raw' pole/zero filter objects:
 - real pole [rpole~]
 - real zero [rzero~]
 - complex pole [cpole~]
 - complex zero [czero~]
- Each of these objects takes the appropriate real/complex filter coefficients as inputs

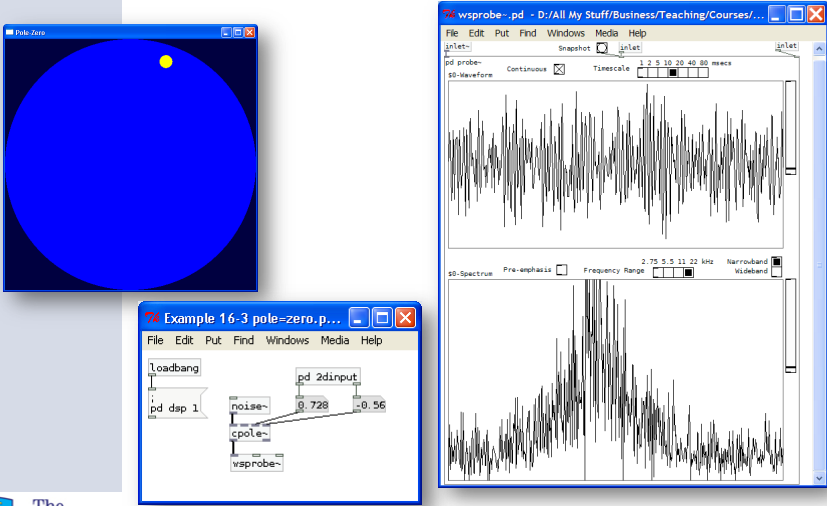
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Filter Response: *complex pole*



The top-left screenshot shows a complex plane with a yellow dot representing a pole. The top-right screenshot shows a waveform and its spectrum. The bottom-left screenshot shows the patch code:

```

loadbang
pd dsp 1
noise~
|
cpole~ 0.728 -0.56
|
wsprobe~

```

The top-right screenshot shows a Pure Data window titled 'wsprobe~.pd' with a waveform and a spectrum plot. The spectrum plot shows a peak at approximately 2.75 kHz.

The bottom-left screenshot shows a Pure Data patch titled 'Example 16-3 pole-zero.p...'. The patch includes a 'loadbang' object, a 'pd dsp 1' object, a 'noise~' object, a 'cpole~' object with arguments '0.728' and '-0.56', and a 'wsprobe~' object.

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This lecture has covered ...

- Filtering
- Difference equations
- Convolution
- The Z transform
- Link with the Fourier transform
- Poles and zeros
- The Z Plane


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Any Questions ?



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Next time ...

Linear Filters

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