

Discrete-Time System Models

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Digital Technium 123

Office Hours: 12:00-13:00 Mondays

You can view the notes for this presentation in [HTML](#) and [PDF](#).

The source code of this presentation is available in Markdown format from GitHub: [dt-models.md](#).

The GitHub repository [EG-247 Resources](#) also contains the source code for all the Matlab/Simulink examples and the Laboratory Exercises.

Scope and Background Reading

This session we will explore digital systems and learn more about the z-transfer function model.

The material in this presentation and notes is based on Chapter 9 (Starting at Section 9.7) of [Steven T. Karris, Signals and Systems: with Matlab Computation and Simulink Modelling, 5th Edition](#). from the **Required Reading List**. I have skipped the section on digital state- space models.

Agenda

- Discrete Time Systems
- Transfer Functions in the Z-Domain
- Modelling digital systems in Matlab/Simulink
- Continuous System Equivalents
- Example: Digital Butterworth Filter

Discrete Time Systems

In the lecture that introduced the z-transform we talked about the representation of a discrete-time (DT) system by the model shown below:

In this session, we want to explore the contents of the central block.

DT System as a Sequence Processor

- As noted in the previous slide, the discrete time system takes as an input the sequence $x_d[n]$.

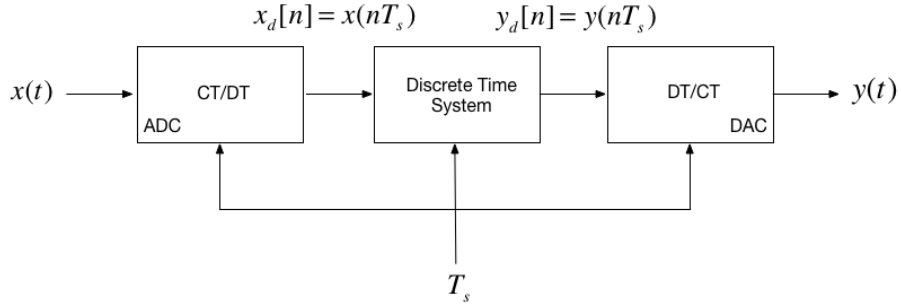


Figure 1: Model of a DT System

- It produces another sequence $y_d[n]$ by *processing* the input sequence in some way.
- The output sequence is converted into an analogue signal $y(t)$ by a digital to analogue converter.

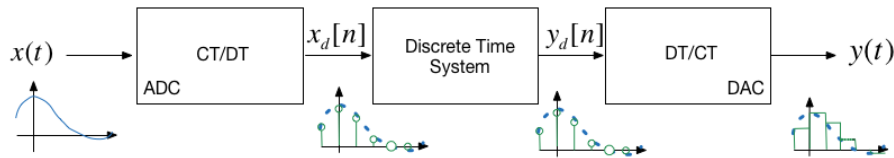


Figure 2: DT System as a Sequence Processor

What is the nature of the DTS?

- The discrete time system (DTS) is a block that converts a sequence $x_d[n]$ into another sequence $y_d[n]$
- The transformation will be a *difference equation* $h[n]$
- By analogy with CT systems, $h[n]$ is the impulse response of the DTS, and $y[n]$ can be obtained by *convolving* $h[n]$ with $x_d[n]$ so:

$$y_d[n] = h[n] * x_d[n]$$

-
- Taking the z-transform of $h[n]$ we get $H(z)$, and from the transform properties, convolution of the signal $x_d[n]$ by system $h[n]$ will be *multiplication* of the z-transforms:

$$Y_d(z) = H(z)X_d(z)$$

- So, what does $h[n]$ and therefore $H(z)$ look like?

Transfer Functions in the z-Domain

Transfer Functions in the z-Domain

Let us assume that the sequence transformation is a *difference equation* of the form^[2]:

$$\begin{aligned} y[n] + a_1y[n-1] + a_2y[n-2] + \dots + a_ky[n-k] \\ = b_0x[n] + b_1u[n-1] + b_2u[n-2] + \dots + b_ku[n-k] \end{aligned}$$

Take Z-Transform of both sides

From the z-transform properties

$$f[n-m] \Leftrightarrow z^{-m}F(z)$$

so....

$$Y(z) + a_1z^{-1}Y(z) + a_2z^{-2}Y(z) + \dots + a_kz^{-k}Y(z) = \dots$$

$$b_0U(z) + b_1z^{-1}U(z) + b_2z^{-2}U(z) + \dots + b_kz^{-k}U(z)$$

Gather terms

$$\begin{aligned} (1 + a_1z^{-1} + a_2z^{-2} + \dots + a_kz^{-k})Y(z) = \\ (b_0 + b_1z^{-1} + b_2z^{-2} + \dots + b_kz^{-k})U(z) \end{aligned}$$

from which ...

$$Y(z) = \left(\frac{b_0 + b_1z^{-1} + b_2z^{-2} + \dots + b_kz^{-k}}{1 + a_1z^{-1} + a_2z^{-2} + \dots + a_kz^{-k}} \right) U(z)$$

Define transfer function

We define the *discrete time transfer function* $H(z) := Y(z)/U(z)$ so...

$$H(z) = \frac{Y(z)}{U(z)} = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2} + \dots + b_k z^{-k}}{1 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_k z^{-k}}$$

... or more conventionally3:

$$H(z) = \frac{b_0 z^k + b_1 z^{k-1} + b_2 z^{k-2} + \dots + b_{k-1} z + b_k}{z^k + a_1 z^{k-1} + a_2 z^{k-2} + \dots + a_{k-1} z + a_k}$$

DT impulse response

The *discrete-time impulse response* $h[n]$ is the response of the DT system to the input $x[n] = \delta[n]$

Last week we showed that $\mathcal{Z}\{\delta[n]\}$ was defined by the transform pair

$$\delta[n] \Leftrightarrow ?$$

$$\delta[n] \Leftrightarrow 1$$

so

$$h[n] = \dots$$

$$h[n] = \mathcal{Z}^{-1}\{H(z).1\} = \mathcal{Z}^{-1}\{H(z)\}$$

Example 1

Karris Example 9.10:

The difference equation describing the input-output relationship of a DT system with zero initial conditions, is:

$$y[n] - 0.5y[n-1] + 0.125y[n-2] = x[n] + x[n-1]$$

Compute:

1. The transfer function $H(z)$
2. The DT impulse response $h[n]$
3. The response $y[n]$ when the input $x[n]$ is the DT unit step $u_0[n]$

1. The transfer function

$$H(z) = \frac{Y(z)}{U(z)} = \dots?$$

1. Solution

$$H(z) = \frac{Y(z)}{X(z)} = \frac{z^2 + z}{z^2 - 0.5z + 0.125}$$

2. The DT impulse response

Start with:

$$\frac{H(z)}{z} = \frac{z-1}{z^2 + 0.5z + 0.125}$$

2. Solution

$$h[n] = \left(\frac{\sqrt{2}}{4}\right)^n \left(\cos\left(\frac{n\pi}{4}\right) + 5\sin\left(\frac{n\pi}{4}\right)\right)$$

Matlab Solution

See [dtm_ex1_2.m](#):

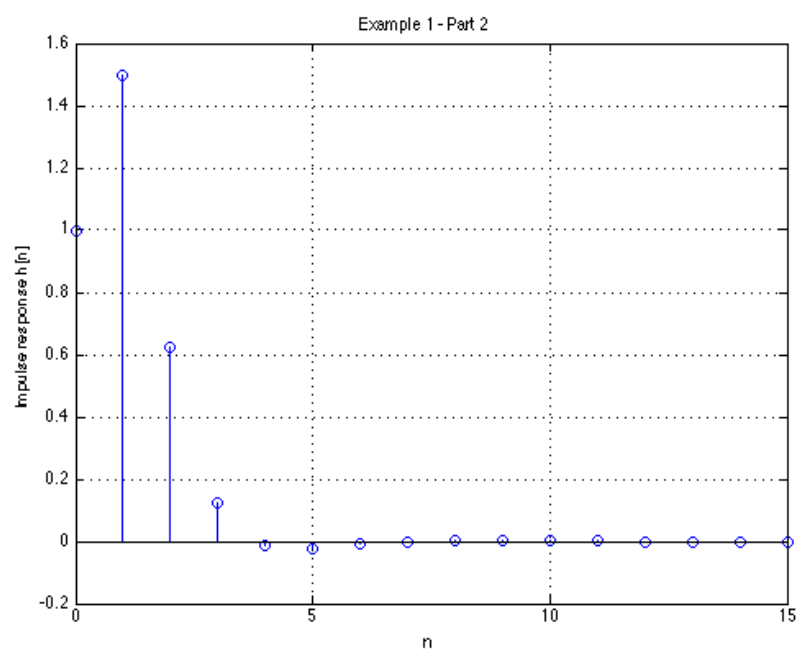


Figure 3:

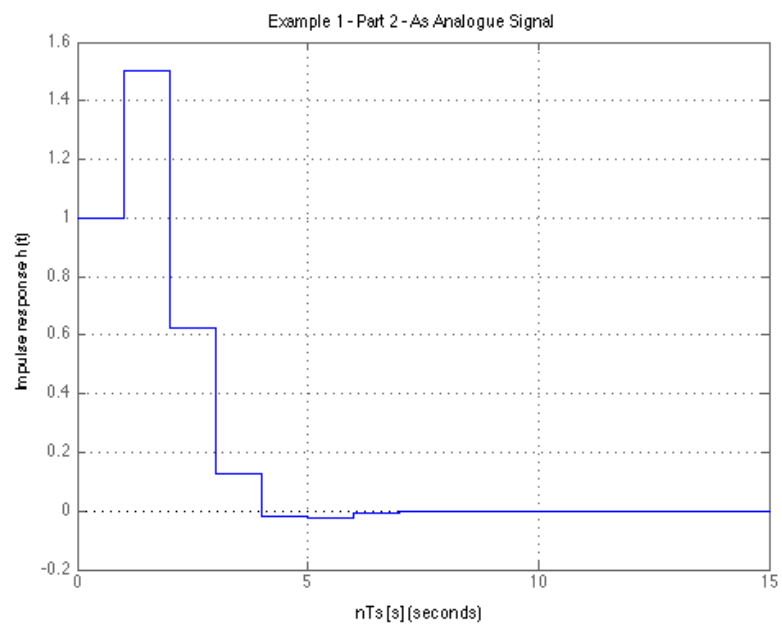


Figure 4:

3. The DT step response

$$Y(z) = H(z)X(z)$$

$$u_0[n] \Leftrightarrow \frac{z}{z-1}$$

$$\begin{aligned} Y(z) = H(z)U_0(z) &= \frac{z^2+z}{z^2+0.5z+0.125} \cdot \frac{z}{z-1} \\ &= \frac{z(z^2+z)}{(z^2+0.5z+0.125)(z-1)} \end{aligned}$$

$$\frac{Y(z)}{z} = \frac{z^2+z}{(z^2+0.5z+0.125)(z-1)}$$

3. Solution

$$y[n] = \left(3.2 - \left(\frac{\sqrt{2}}{4} \right)^n \left(2.2 \cos\left(\frac{n\pi}{4}\right) + 0.6 \sin\left(\frac{n\pi}{4}\right) \right) \right) u_0[n]$$

Matlab Solution

See [dtm_ex1_3.m](#):

Modelling DT systems in Matlab and Simulink

Matlab

Code extracted from [dtm_ex1_3.m](#):

```
Ts = 1;  
z = tf('z', Ts)  
Hz = (z^2 + z)/(z^2 - 0.5 * z + 0.125)  
step(Hz)  
grid  
title('Example 1 - Part 3 - As Analogue Signal')  
xlabel('nTs [s]')  
ylabel('Step response y(t)')  
axis([0,15,0,3.5])
```

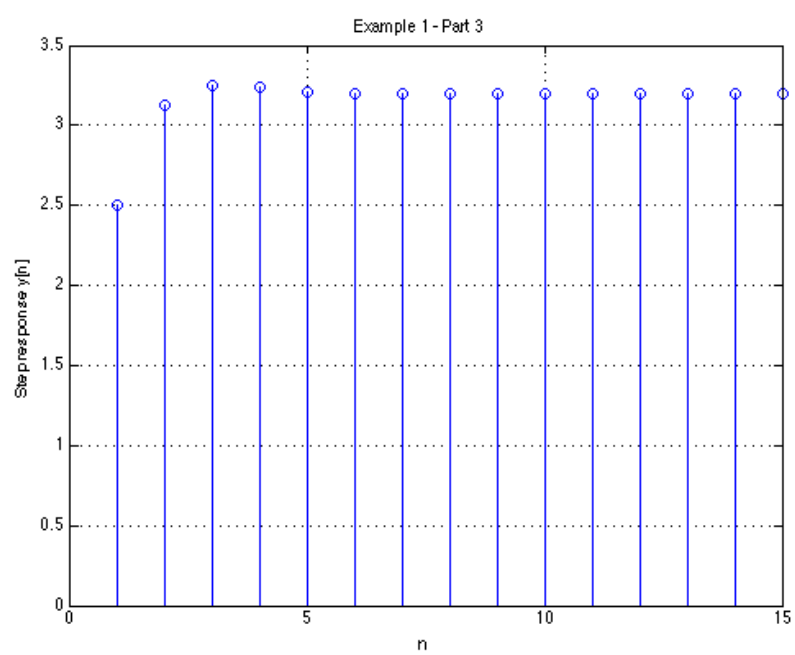


Figure 5:

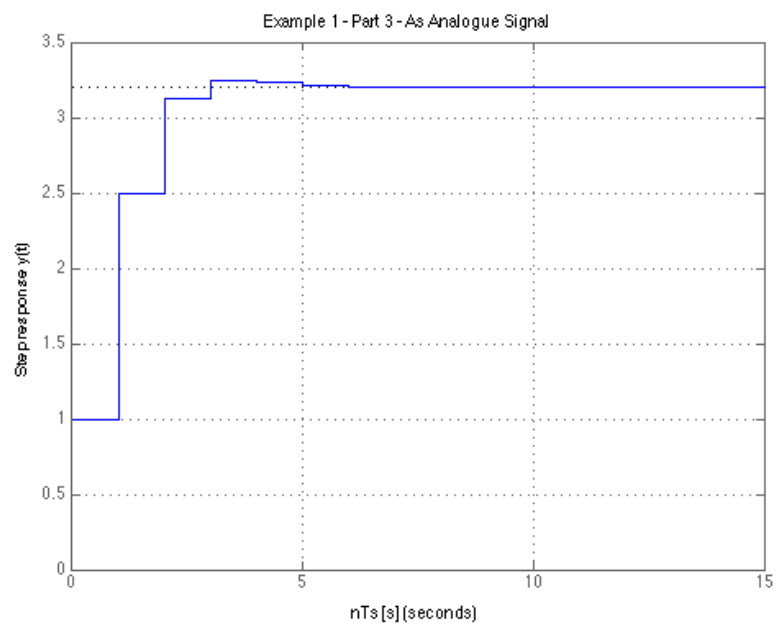


Figure 6: Example 1 - Part 3 - As Analogue Signal

Simulink Model

See [dtm.slx](#):

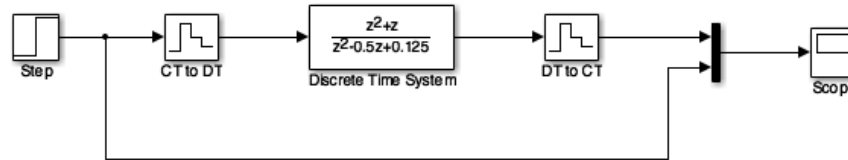


Figure 7: Simulink model

Converting Continuous Time Systems to Discrete Time Systems

Converting Continuous Time Systems to Discrete Time Systems

- In analogue electronics, to implement a filter we would need to resort to op- amp circuits with resistors, capacitors and inductors acting as energy dissipation, storage and release devices.
- In modern digital electronics, it is often more convenient to take the original transfer function $H(s)$ and produce an equivalent $H(z)$.
- We can then determine a *difference equation* that will represent $h[n]$ and implement this as *computer algorithm*.
- Simple storage of past values in memory becomes the repository of past state rather than the integrators and derivative circuits that are needed in the analogue world.
- To achieve this, all we need is to be able to do is to *sample* and *process* the signals quickly enough to avoid violating Nyquist-Shannon's sampling theorem.

Continuous System Equivalents

- There is no digital system that uniquely represents a continuous system

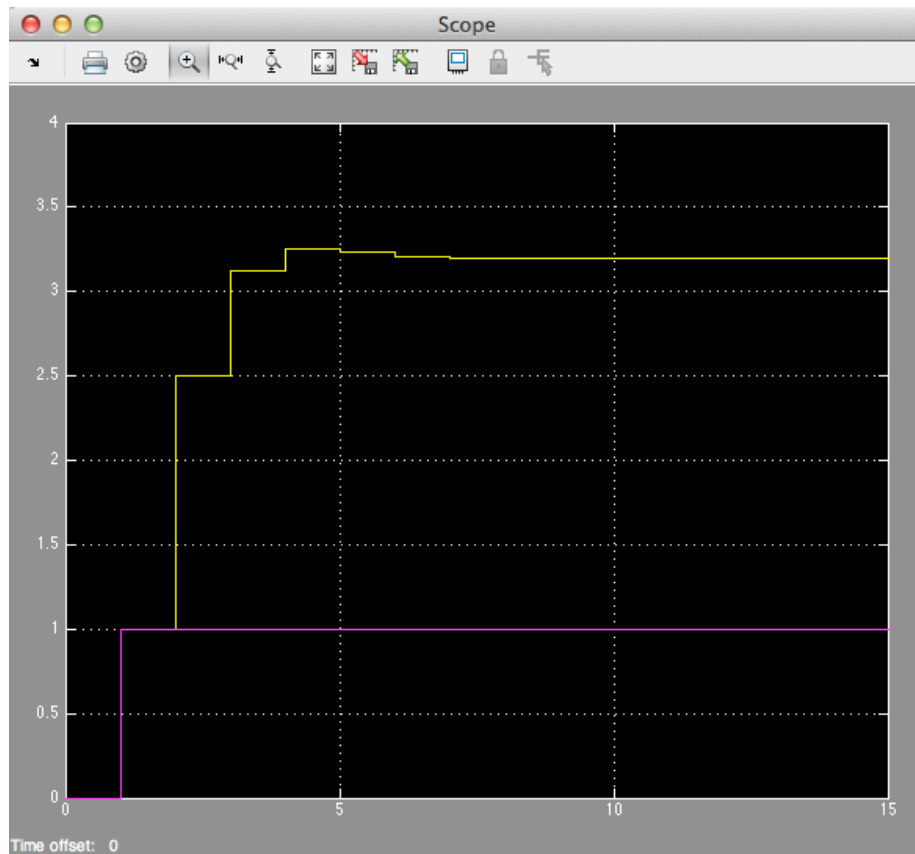


Figure 8: Simulated response

- This is because as we are sampling, we only have knowledge of signals being processed at the sampling instants, and need to *reconstruct* the inter-sample behaviour.
- In practice, only a small number of transformations are used.
- The derivation of these is beyond the scope of this module, but we'll mention the ones that Matlab provides in a function called `c2d`

Matlab c2d function

This is what the help function says:

```
>> help c2d
SYSD = c2d(SYSC,TS,METHOD) computes a discrete-time model SYSD with
sampling time TS that approximates the continuous-time model SYSC.
The string METHOD selects the discretization method among the following:
    'zoh'      Zero-order hold on the inputs
    'foh'      Linear interpolation of inputs
    'impulse'  Impulse-invariant discretization
    'tustin'   Bilinear (Tustin) approximation.
    'matched'  Matched pole-zero method (for SISO systems only).
The default is 'zoh' when METHOD is omitted. The sampling time TS should
be specified in the time units of SYSC (see "TimeUnit" property).
...
```

Example 2

- Design a 2nd-order butterworth anti-aliasing filter with transfer function $H(s)$ for use in sampling music.
- The cut-off frequency $\omega_c = 20$ kHz and the filter should have an attenuation of at least -80 dB in the stop band.
- Choose a suitable sampling frequency for the audio signal and give the transfer function $H(z)$ and an algorithm to implement $h[n]$

Bode plot

Matlab:

```
wc = 2*pi*20e3;
Hs = tf(wc^2,[1 wc*sqrt(2), wc^2]);
bode(Hs,{1e4,1e8})
grid
```

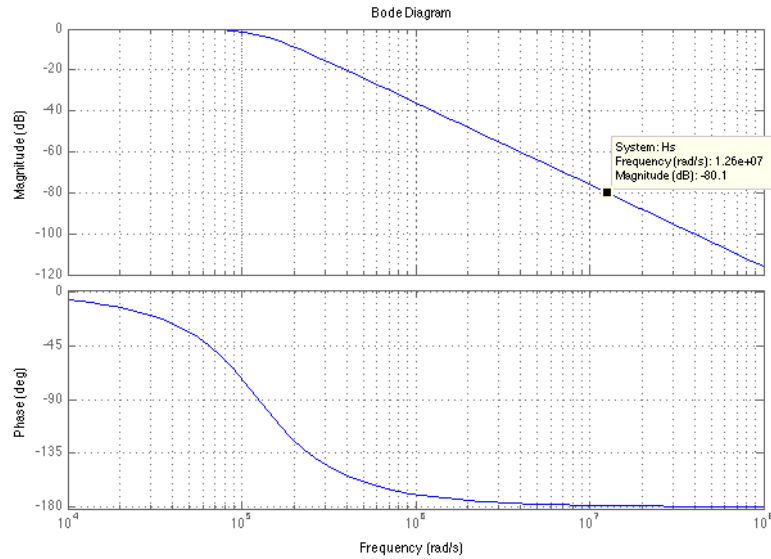


Figure 9: Bode plot

Sampling Frequency

From the bode diagram, the frequency at which $|H(j\omega)|$ is -80 dB is approx 12.6×10^6 rad/s.

To avoid aliasing, we should choose a sampling frequency twice this = ?

So sampling frequency $\omega_s = 2 \times 12.6 \times 10^6 = 25.2 \times 10^6$ rad/s.

Sampling frequency in Hz $f_s = ?$

$$f_s = \omega_s / (2\pi) = 25.2 \times 10^6 / (2 \times \pi) = 40.1 \text{ Mhz}$$

Sampling time $T_s = ?$

$$T_s = 1/f_s \approx 0.25 \mu s$$

Digital Butterworth

```
>> Hz = c2d(Hs, Ts) % zero-order-hold equivalent
```

Hz =

$$\frac{0.0004836 z + 0.0004765}{z^2 - 1.956 z + 0.9567}$$

Sample time: 2.4933e-07 seconds
Discrete-time transfer function.

Step response

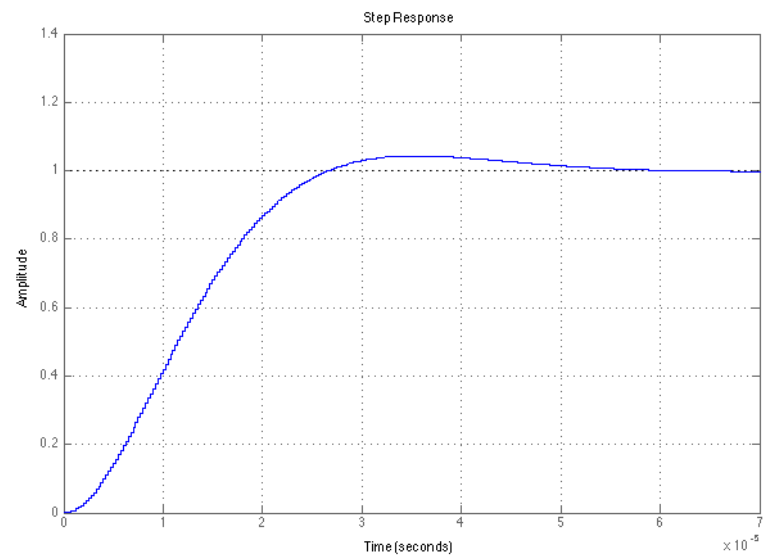


Figure 10:

Algorithm

From previous result:

$$H(z) = \frac{Y(z)}{U(z)} = \frac{486.6 \times 10^{-6} z + 476.5 \times 10^{-6}}{z^2 - 1.956 z + 0.9567}$$

Dividing top and bottom by z^2 ...

$$H(z) = \frac{Y(z)}{U(z)} = \frac{486.6 \times 10^{-6} z^{-1} + 476.5 \times 10^{-6} z^{-2}}{1 - 1.956 z^{-1} + 0.9567 z^{-2}}$$

expanding out ...

$$\begin{aligned} Y(z) - 1.956 z^{-1} Y(z) + 0.9567 z^{-2} Y(z) = \\ 486.6 \times 10^{-6} z^{-1} U(z) + 476.5 \times 10^{-6} z^{-2} U(z) \end{aligned}$$

Algorithm ... continued

Inverse z-transform gives ...

$$\begin{aligned} y[n] - 1.956 y[n-1] + 0.9567 y[n-2] = \\ 486.6 \times 10^{-6} u[n-1] + 476.5 \times 10^{-6} u[n-2] \end{aligned}$$

in algorithmic form (compute $y[n]$ from past values of u and y) ...

$$\begin{aligned} y[n] = 1.956 y[n-1] - 0.9567 y[n-2] + 486.6 \times 10^{-6} u[n-1] + \dots \\ 476.5 \times 10^{-6} u[n-2] \end{aligned}$$

Convert to code

To implement:

$$\begin{aligned} y[n] = 1.956 y[n-1] - 0.9567 y[n-2] + 486.6 \times 10^{-6} u[n-1] + \dots \\ 476.5 \times 10^{-6} u[n-2] \end{aligned}$$

```
/* Initialize */
ynm1 = 0; ynm2 = 0; unm1 = 0; unm2 = 0;
while (true) {
    un = read_adc;
    yn = 1.956 * ynm1
        - 0.9567 * ynm2
        + 486.6e-6 * unm1
        + 476.5e-6 * unm2;
    write_dac(yn);
    /* store past values */
    ynm2 = ynm1; ynm1 = yn;
    unm2 = unm1; unm1 = un;
}
```


Comments

PC soundcards can sample audio at 44.1 kHz so this implies that the anti-aliasing filter is much sharper than this one as $f_s/2 = 22.05$ kHz.

You might wish to find out what order butterworth filter would be needed to have $f_c = 20$ kHz and f_{stop} of 22.05 kHz.

Summary

- Discrete Time Systems
- Transfer Functions in the Z-Domain
- Modelling digital systems in Matlab/Simulink
- Continuous System Equivalents
- Example: Digital Butterworth Filter

The End?

- This concludes this module.
- There is some material that I have not covered, most notably **Discrete Fourier Transform**.
- This is covered in Karris Chapter 10 and Boulet. It will not be examined this year!
- There is a significant amount of additional information about **Filter Design** (including the use of Matlab for this) in Chapter 11 of Karris.

Homework

You should be able to tackle the remaining end of chapter exercises 8-11 (Section 9.10) from Karris. Don't look at the answers until you have attempted the problems.