Daltech, Dalhousie University
Department of Electrical and Computer Engineering

ECED 4502 – Digital Signal Processing

Lab 4 –Infinite Impulse Response Filtering in Matlab

1. You need the DSP System Toolbox installed on your Matlab.
   Please see Appendix B to learn how to install it if you do not have it.
2. Matlab 2018 release or newer versions is recommended.

Objectives

The objectives of this lab are to observe the characteristics of infinite impulse response (IIR) filters and to design your own IIR filters in Matlab.

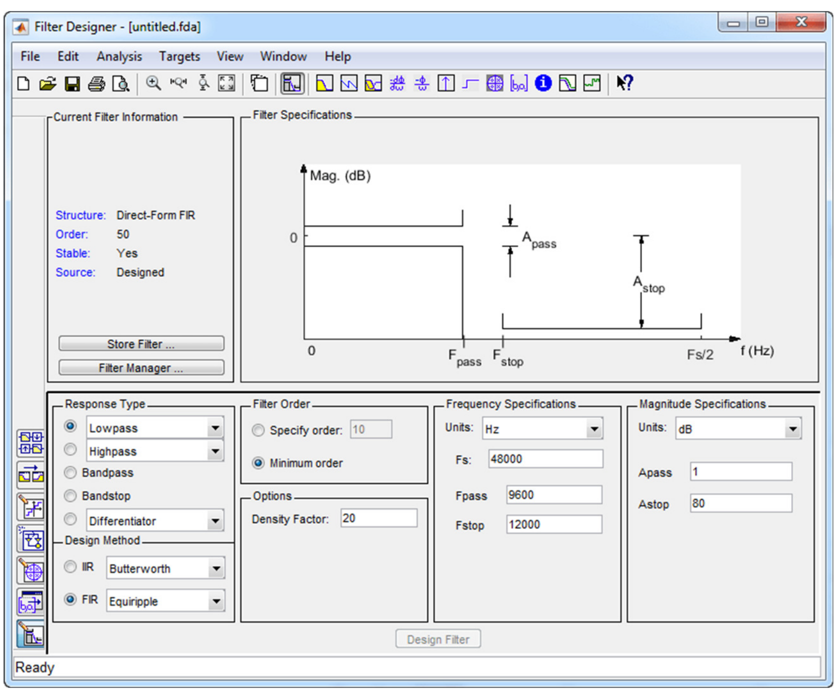
In the first section, Background, you will become familiar with using the Filter Designer toolbox. In the second section, you are expected to design different IIR filters with specified parameters, comment on your design and answer some questions about it.

1- Background: Using Filter Designer

To open filter designer, type

filterDesigner

at the MATLAB® command prompt. The filter designer opens with the Design filter panel displayed.



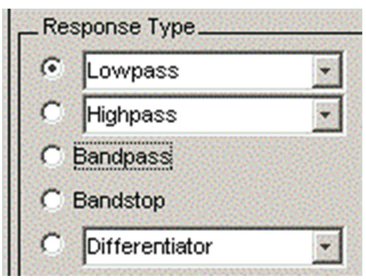
Note that when you open filter designer, **Design Filter** is not enabled. You must make a change to the default filter design in order to enable **Design Filter**. This is true for each time you want to change the filter design. Changes to radio button items or drop down menu items such as those under **Response Type** or **Filter Order** enable **Design Filter** immediately. Changes to specifications in text boxes such as **Fs**, **Fpass**, and **Fstop** require you to click outside the text box to enable **Design Filter**.

1.1 Choosing a Response Type

You can choose from several response types:

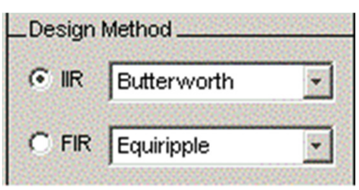
- Lowpass
- Raised cosine
- Highpass
- Bandpass
- Bandstop
- Differentiator

To design a Lowpass filter, select the radio button next to Lowpass in the Response Type region of the app.



Choosing a Filter Design Method

Select IIR methods listed in the app.



Viewing Filter Specifications

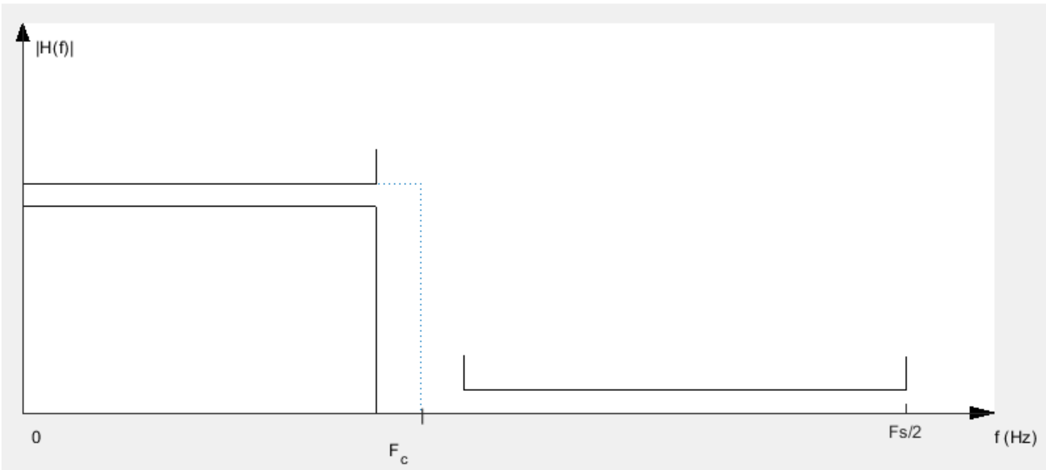
The filter design specifications that you can set vary according to response type and design method. The display region illustrates filter specifications when you select **Analysis > Filter Specifications** or when you click the **Filter Specifications** toolbar button.

You can also view the filter specifications on the Magnitude plot of a designed filter by selecting **View** > **Specification Mask**.

Filter Order

You have two mutually exclusive options for determining the filter order when you design an equiripple filter:

- **Specify order**: You enter the filter order in a text box.
- **Minimum order**: The filter design method determines the minimum order filter.



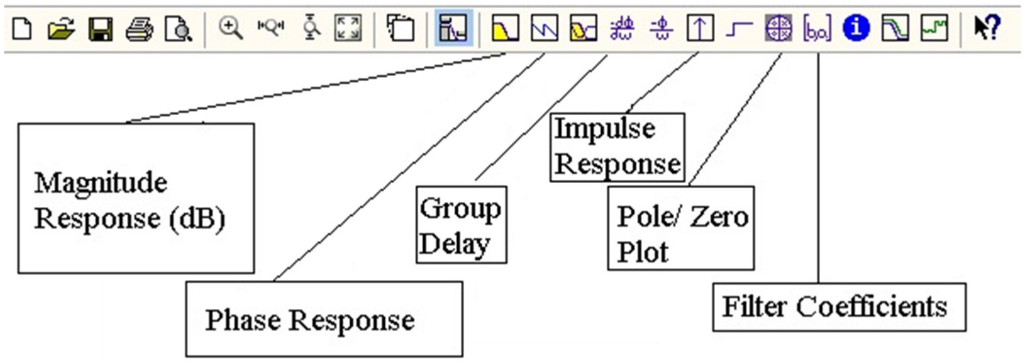
- The parameters of your filter are given in the "**Frequency Specification**" part of the screen where you can select:
  - the filter type – you can try first the low-pass filter.
  - the order of your filter (the higher the order, the better approximation of the ideal filter you will get)
  - the cut-off frequency of your filter for the passband F_p and the cut-off frequency of your filter for the stopband F_s – both of them should be always smaller than *fs/2* with *fs* selected in the part of the screen "**Acquire Settings**".

You should try to design your filter using different filter design methods (i.e. approximations of the ideal frequency response) such as Butterworth, Chebyshev I & II and Elliptic.

You should make observations about the behavior of the frequency responses and pole-zero plots for the different design methods. The primary focus should be on the magnitude of the frequency response; however, you should also comment on the linearity of the phase response in the passband.

Now that you have specified the filter design, click the **Design Filter** button to compute the filter coefficients.

Notice that the Design Filter button is disabled once you've computed the coefficients for your filter design. This button is enabled again once you make any changes to the filter specifications.



The coefficients in your design are in the top bar.

Example: For a 10th order, Direct Form II second-order sections, Low pass, Elliptic IIR filter with Fs =48 kHz and Fpass= 9.6 kHz we have

Structure:	Direct-Form II, Second-Order Sections
Order:	10
Sections	5
Stable:	Yes
Source:	Designed

```
Section #1
--------------------------
Numerator:
 1
-0.33490998214784406528110594081226736
 1
Denominator:
 1
-0.67197985508723045988688227225793525557
 0.93884502325395124966433968438650481
Gain:
0.80632608282072504302107063267612829800
```

```
--------------------------
Section #2
--------------------------
Numerator:
 1
-0.082690163875476563326571977086132392287
 1
Denominator:
 1
-0.826475991007310950564601625956129282713
 0.856176158887234861616377656901022419333
Gain:
0.876445215850252035139078543579671531916
--------------------------
```

```
Section #3
--------------------------
Numerator:
1
0.552120446488081006997106214839732274413
1
Denominator:
 1
-1.109622999149721955802760930964723229408
 0.714120252982946346342885135527467355132
Gain:
0.813151668106534608604363256745273247361
```

```
--------------------------
Section #4
--------------------------
Numerator:
1
1.705652561120809096451012010220438241959
1
Denominator:
 1
-1.413006561918870751526355888927355408669
 0.563838496243366016358322667656466364861
Gain:
1.2282771247617729759582516635418869555
--------------------------
```

```
--------------------------
Section #5
--------------------------
Numerator:
 1
-0.421822725739562587321529463224578648806
 1
Denominator:
 1
-0.613949779495460923328664648579433560371
 0.983404881078795090054711636184947565198
Gain:
0.004316398556150480639304056751370808342
--------------------------
Output Gain:
1
```

Note that coefficients of the current filter, which depend on the filter structure (e.g., direct form or second-order cascaded sections, etc.) are shown in the text box.

Use Convert structure or Convert to Second-Order Sections on the Edit menu to transform a filter from one structure to another, or use the Import Filter Panel to import a filter with the desired structure directly (see Appendix A).

By adjusting the index of the displayed coefficient you can get the coefficients in the transfer function of the biquads and from there the frequency response of your filter $H(z)=\prod_{k=1}^{N_1} H_k(z)$, where $H_k(z)=\dfrac{b_{k,0}+b_{k,1}\cdot z^{-1}+b_{k,2}\cdot z^{-2}}{1-(a_{k,1}\cdot z^{-1}+a_{k,2}\cdot z^{-2})}$ and $2N_1$ is equal to the order of the filter (e.g., if the order of the filter in your design is four you should have 6 forward coefficients ($b_{1,0}, b_{1,1}, b_{1,2}, b_{2,0}, b_{2,1}, b_{2,2}$) and four reverse coefficients ($a_{1,1}, a_{1,2}, a_{2,1}, a_{2,2},$)

2. Lab Report Exercise

In your lab report you should answer the questions below and include your frequency magnitude response, group delay & pole-zero plots for Butterworth, Chebyshev Type I, Chebyshev Type II and Elliptic filters, and comment on the differences among them.

<u>Exercise 1.</u>

Design a high-pass IIR filter using the Butterworth method. Specify the order to be 8 with a default attenuation of 3 dB at a cut-off frequency Fc = 14400 and sampling frequency Fs = 48000 Hz.

1- Capture the magnitude response, the group delay and the pole-zero plot of your filter.
2- Click on the filter coefficients tab to see your Numerator and Denominator coefficients. You will see the coefficients for a filter structure of four cascaded Direct Form II second-order sections. Click on store filter to store your filter in the filter manager. Once you save the filter open filter manager and click on FVtool to see the filter you have designed.

<u>Exercise 2.</u>

Now, without changing the filter order, response type and frequency specification try other IIR design methods: Chebyshev Type I, Chebyshev Type II and Elliptic.

<u>Exercise 3.</u>

Change the design criterion to Minimum Filter Order. Keep Fs as it was 48000 Hz.
And use Fstop = 9600Hz, Astop = 80dB, Fpass = 12000Hz, Apass = 1dB

- How does the order of the filter affect the frequency response of IIR filters in a particular design?
- To meet the frequency response specifications, which design requires in general the lowest order for the transfer function?
- How does the magnitude, phase response, group delay and pole-zero plots of the filters compare among various designs?
- How far does the transition zone extend in the various designs?

Comment on your observations.

Appendix A

IIR Direct Form Structures (Digital Filter Design) [1]

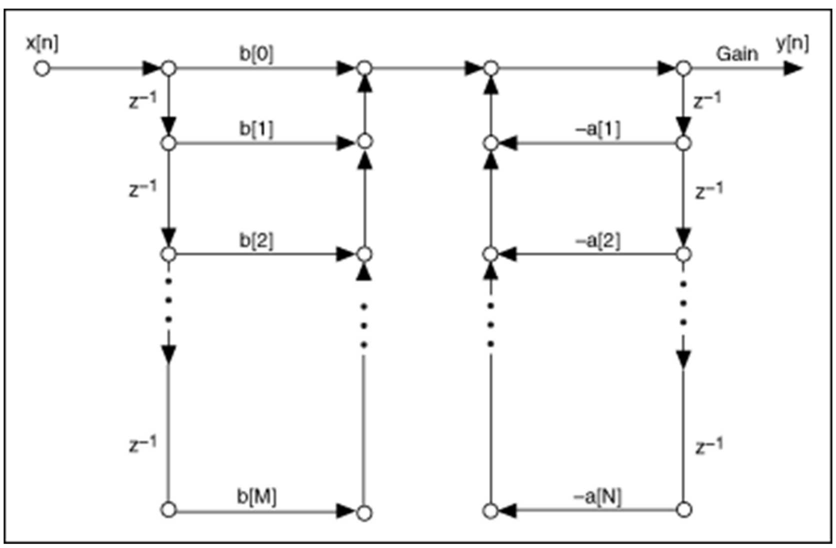
The transfer function of an infinite impulse response (IIR) filter is defined as follows:

$$H(z) = Gain \bullet \frac{\sum_{n=0}^{M} b[n] z^{-n}}{1 + \sum_{n=1}^{N} a[n] z^{-n}}$$

where *z* is a complex variable, *M* is the order of the numerator, *N* is the order of the denominator, *a* is the set of reverse coefficients, and *b* is the set of forward coefficients.

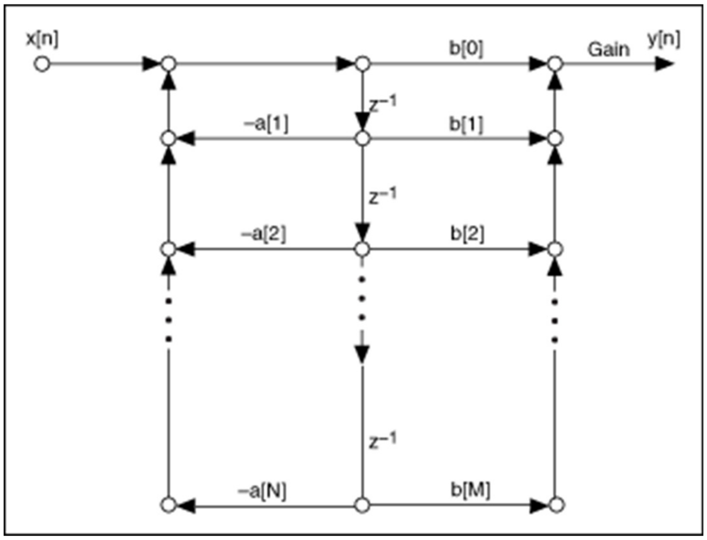
IIR Direct Form I

The IIR Direct Form I structure is the most straightforward IIR structure from a filter transfer function perspective. The following figure represents the IIR Direct Form I structure. Refer to the Understanding Filter Structure Graphs topic for information that helps you read and understand a filter structure graph.



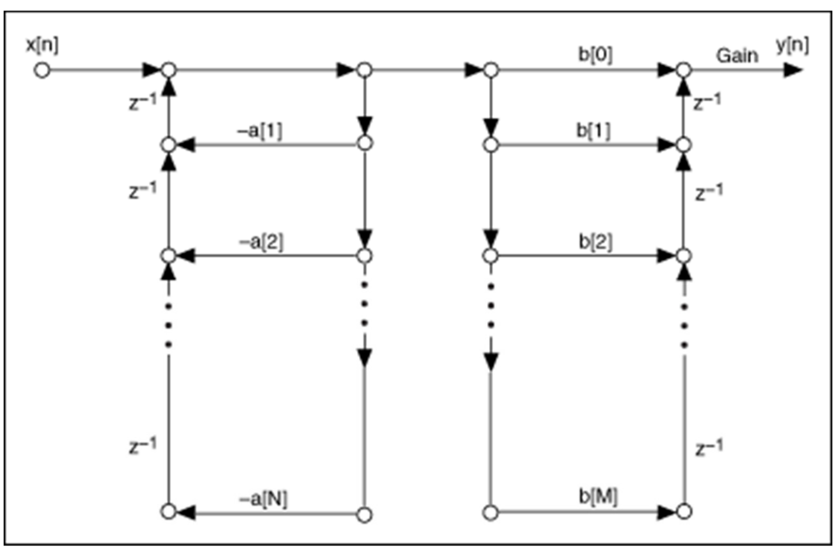
IIR Direct Form II

The following figure represents the IIR Direct Form II structure. You can see that this structure contains fewer delays thus less storage is required.



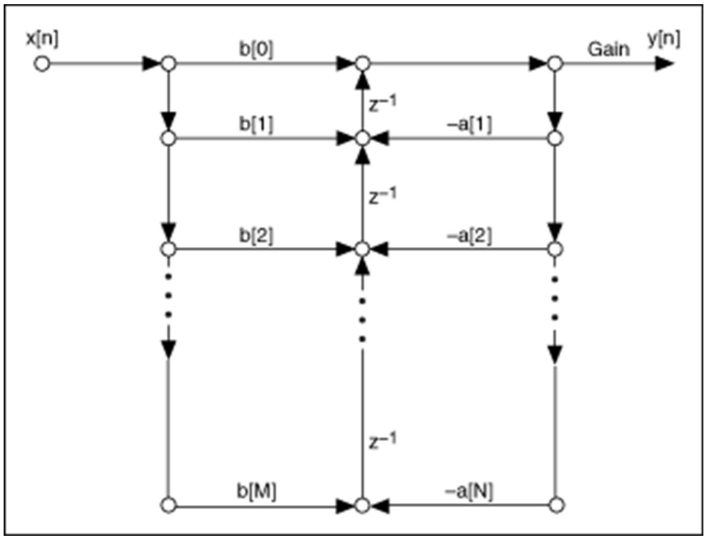
IIR Direct Form I Transposed

The following figure represents the IIR Direct Form I Transposed structure.



IIR Direct Form II Transposed

The following figure represents the IIR Direct Form II Transposed structure.



The IIR Direct Form I and Form II Transposed structures implement forward coefficients first. The Form I Transposed, and Form II structures implement reverse coefficients first. The IIR Direct Form structures usually require few mathematical operations. However, the sensitivity to finite word length effects limits the use of this form in fixed-point implementations. Use the IIR Cascaded Second-Order Sections Form structures to alleviate finite word length effects.

[1] https://zone.ni.com/reference/en-XX/help/371325F-01/lvdfdtconcepts/iir_direct_specs/

Appendix B

Downloading the DSP System Toolbox

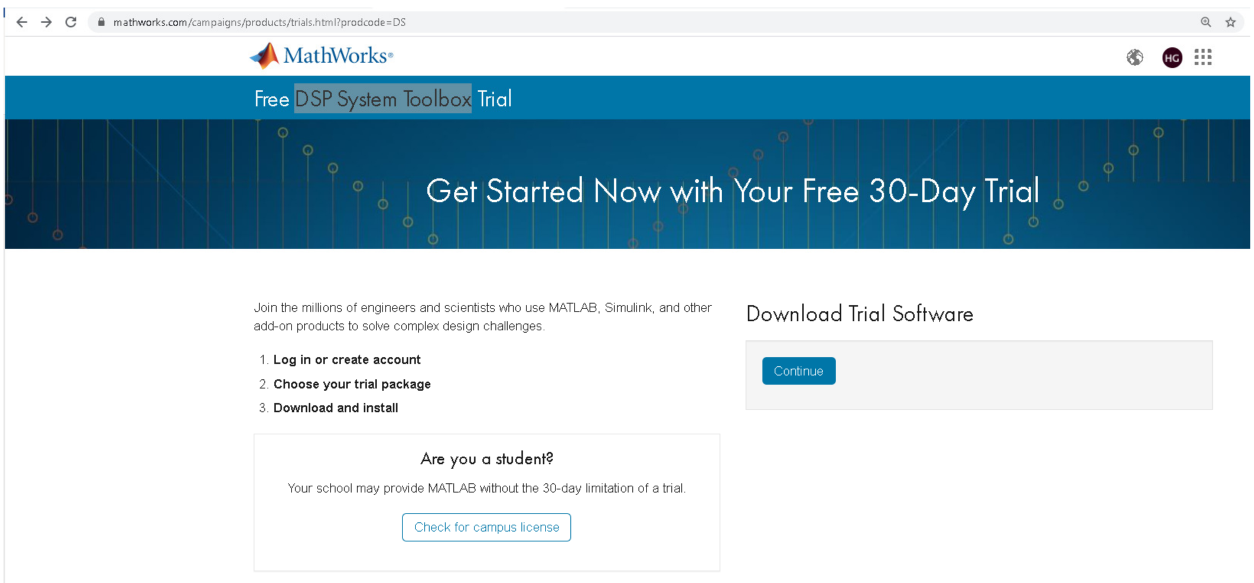
With DSP System Toolbox you can design and analyze FIR, IIR filters.

https://www.mathworks.com/products/dsp-system.html

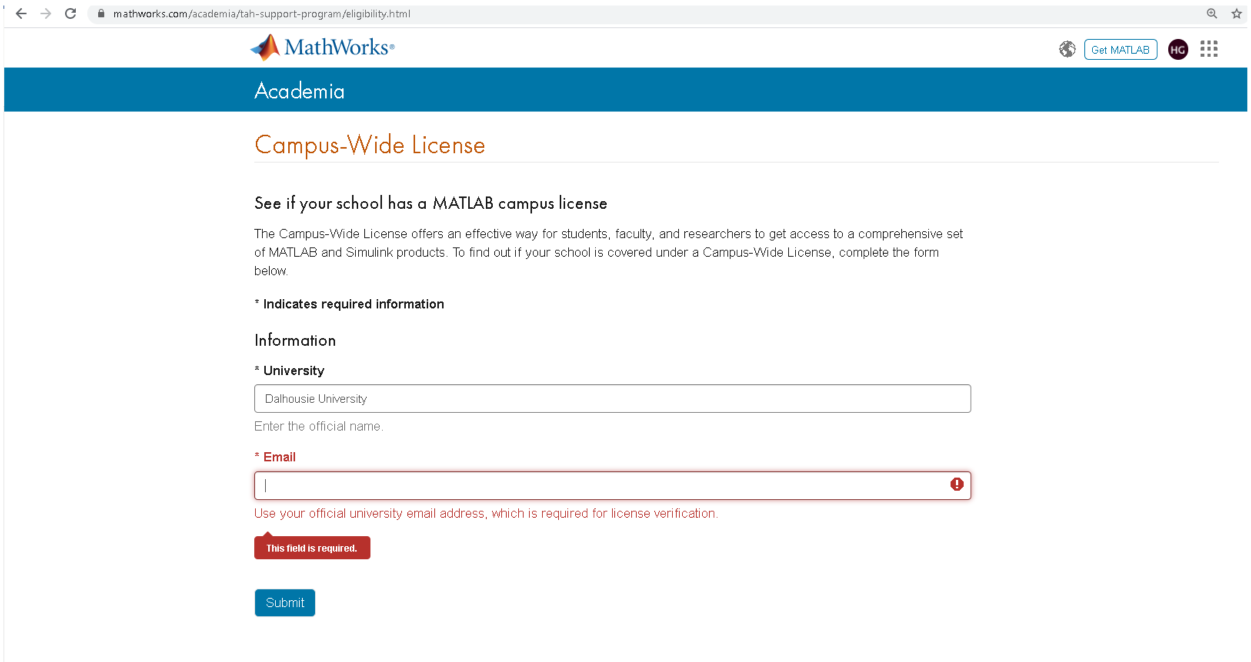
You can stream signals from variables, data files, and network devices for system development and verification. The Time Scope, Spectrum Analyzer, and Logic Analyzer let you dynamically visualize and measure streaming signals. For desktop prototyping and deployment to embedded processors, including ARM® Cortex® architectures, the system toolbox supports C/C++ code generation. It also supports bit-accurate fixed-point modeling and HDL code generation from filters, FFT, IFFT, and other algorithms.

Use the following link to download the toolbox

https://www.mathworks.com/campaigns/products/trials.html?prodcode=DS

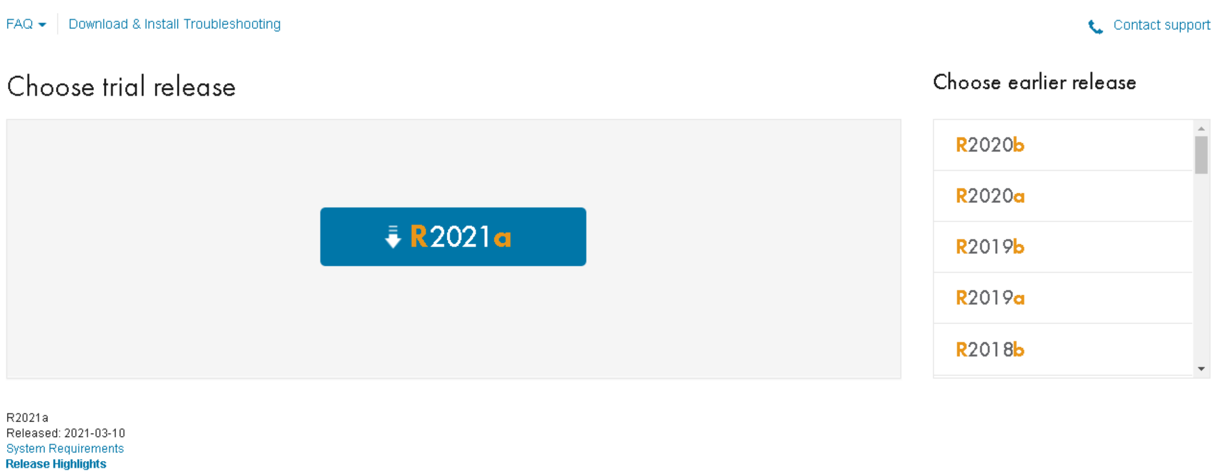


If you do not have the Academia license. Click on **Check for the campus license**.

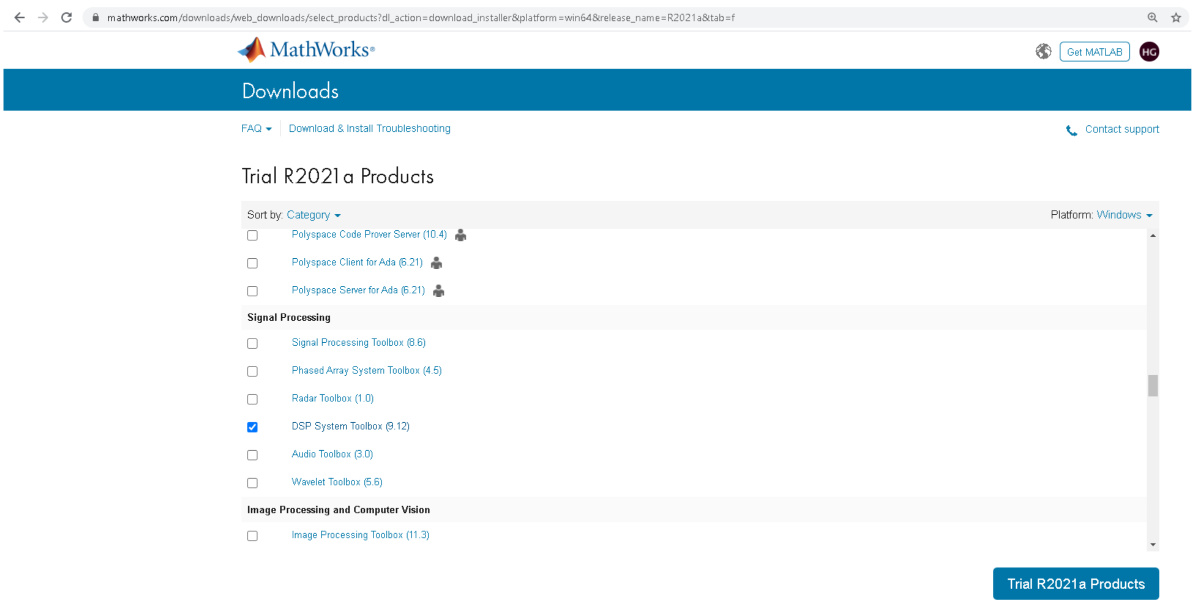


After entering your email you will receive an email to add Dalhousie license into your Mathworks account

After getting the academic license choose your Matlab release version



Then choose DSP System Toolbox corresponding to your Matlab version. From the download page



In the next page choose your operating system and download the installer