

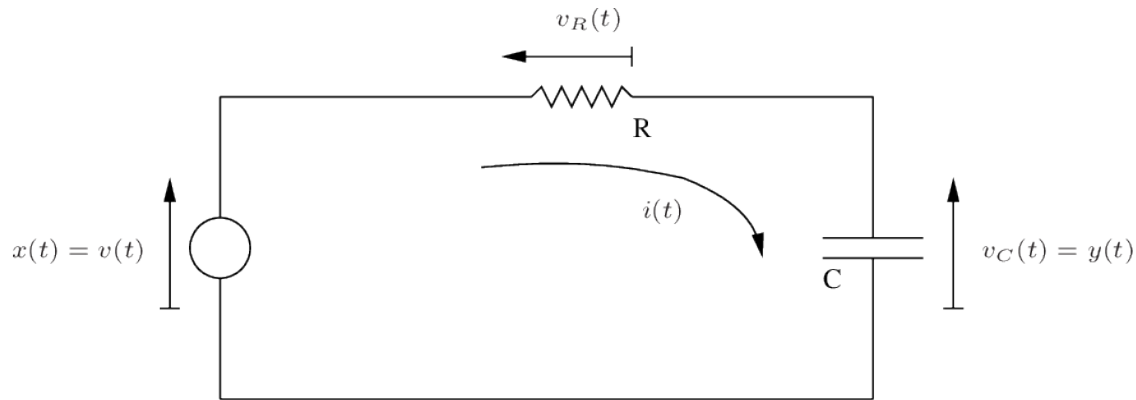
EEE2047S 2024 Lab 3

July 4, 2024

1 Lab 3: Frequency response

1.1 The problem

Consider numerically solving the first-order lowpass filter circuit below:



The input signal $x(t)$ is the source voltage and the output $y(t)$ is the voltage across the capacitor.

Using standard techniques the governing differential equation is found to be

$$y(t) + RC \frac{dy(t)}{dt} = x(t).$$

For known $x(t)$ we can use this equation to solve for the corresponding output $y(t)$. This is a first-order differential equation in $y(t)$, so one auxiliary condition will be required to specify $y(t)$ fully (think of homogeneous and particular solutions).

Forming the discretised signals $x[n] = x(nT)$ and $y[n] = y(nT)$ for some small $T > 0$, the standard Euler method can be used to derive the forward and reverse iterations

$$y[n+1] \approx y[n] + \frac{T}{RC}(x[n] - y[n]) \quad \text{and} \quad y[n-1] \approx y[n] - \frac{T}{RC}(x[n] - y[n]).$$

We could also derive the backward or implicit Euler method for the discretisation, where the forward and backward iterations are

$$y[n] = \frac{1}{1 + T/(RC)}y[n-1] + \frac{T/(RC)}{1 + T/(RC)}x[n] \quad \text{and} \quad y[n] = \frac{1}{1 - T/(RC)}y[n+1] - \frac{T/(RC)}{1 - T/(RC)}x[n].$$

If the value of T is too large then the Euler method becomes unstable. The implicit Euler method has better stability, but in most cases requires more computation to solve.

In this lab we want to consider the response of the system to the input $x(t) = \cos(\omega_0 t)$ for different values of ω_0 . This is called the *steady state response* for the frequency ω_0 .

However, an auxiliary condition is still required to fully specify the output. To investigate this effect we will first consider the input $x(t) = \cos(\omega_0 t)u(t)$, which is zero for negative time. In other words the signal suddenly “starts” at time $t = 0$. We will consider initial conditions of the form $y(t_0) = y_0$ for some time t_0 and given value y_0 .

1.2 Basic frequency response

The following function uses the discrete formulation above to approximately find the output signal $y[n]$ for a given input $x[n]$, subject to the initial condition $y[n_0] = y_0$. The initial condition is first used to populate one element of the output, and the recursions are then used in both directions to obtain the full solution.

```
[1]: import numpy as np
import matplotlib.pyplot as plt
%matplotlib inline

def rclowpass_resp(xv,n0,y0,TdRC,bflag=0):
    """RC lowpass output from given input using standard Euler
    xv: input signal (array)
    n0, y0: auxiliary condition yv[n0] = y0
    TdRC: value T/RC
    bflag: method Euler forward (0) or backward (1)
    returns yv: output signal (array)
    """

    yv = np.zeros(xv.shape);
    yv[n0] = y0;

    if bflag==0:
        # Forward Euler in each direction
        for n in range(n0,len(xv)-1):
            yv[n+1] = yv[n] + TdRC*(xv[n] - yv[n]); # forward recursion
        for n in range(n0,0,-1):
            yv[n-1] = yv[n] - TdRC*(xv[n] - yv[n]); # reverse recursion
    else:
        # Backward (implicit) Euler in each direction
        for n in range(n0,len(xv)-1):
            yv[n+1] = 1/(1+TdRC)*yv[n] + TdRC/(1+TdRC)*xv[n+1]; # forward
            ↪ recursion
        for n in range(n0,0,-1):
            yv[n-1] = 1/(1-TdRC)*yv[n] - TdRC/(1-TdRC)*xv[n-1]; # reverse
            ↪ recursion

    return yv;
```

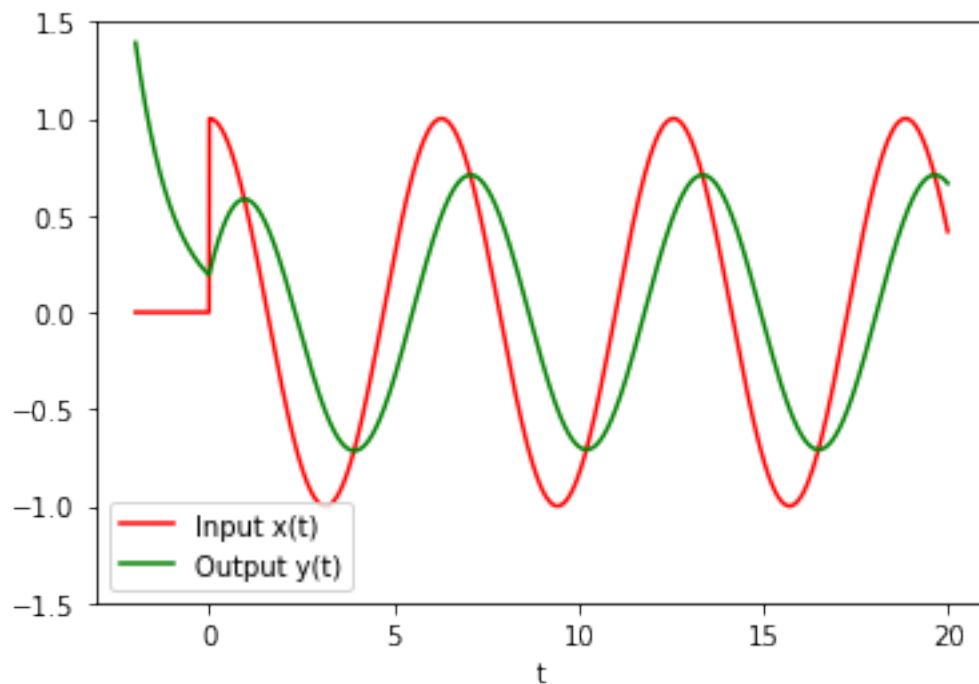
We generate an input xv containing the values of $x(t) = \cos(\omega_0 t)u(t)$ for a fixed frequency $\omega_0 = 1$

and a range of values tv :

```
[2]: # Input signal and discretisation
omega0 = 1; # current frequency
T = 0.01; # small number
if omega0==0: T0 = np.inf;
else: T0 = 2*np.pi/omega0; # waveform period (seconds) for current omega0
nv = np.arange(-200,2000); # discrete signal indices n
tv = nv*T; # time values corresponding to indices t=nT
xv = np.zeros(tv.shape);
for i in range(0,len(xv)): xv[i] = np.cos(omega0*tv[i])*(tv[i]>=0);
```

Suppose the initial condition on the output is $y(0) = 0.2$. We can find the corresponding output, and plot both the input and the output on the same set of axes:

```
[3]: # Response for given initial condition
RC = 1; TdRC = T/RC;
n0 = np.where(nv==0)[0][0]; # find location in nv with zero value
y0 = 0.2;
yv = rclowpass_resp(xv,n0,y0,TdRC,0);
fig = plt.figure(); ph = plt.plot(tv,xv,'r-',tv,yv,'g-');
plt.xlabel('t'); plt.legend(['Input x(t)', 'Output y(t)']); plt.ylim((-1.5,1.5));
```



Some interesting observations can be made:

- Prior to time $t = 0$ we see the output approaching the value 0.2 from above. We can understand this in terms of the original circuit. Remember that for negative values of time the input voltage $x(t)$ is zero, so the voltage source on the left is a short circuit. The capacitor is therefore simply discharging through the resistor. Since the auxiliary condition says that at time $t = 0$ the capacitor voltage must be 0.2 volts, for negative time this voltage must have been positive and much bigger.
- The initial onset of the signal occurs at $t = 0$. After some time has passed we see a simple relationship between the output and the input: both are sinusoids at the same frequency, but the output is delayed relative to the input and its amplitude has been reduced. We call this the steady-state response. Essentially, for large t the input is $x(t) = \cos(\omega_0 t)$ and the output can be written as $y(t) = A \cos(\omega_0 t + \theta)$ for some A and θ .
- Just after $t = 0$, but before the system has reached steady state, the output has a period of disturbance. This transient depends on the initial conditions, but (since the system is stable) will die down after enough time has passed. The steady-state response is independent of the initial conditions.

For the case shown above we can characterise the steady-state response for the driving frequency ω_0 by estimating A and θ for large t . The following block of code calculates these estimates by finding the difference in position between the last positive peak in the output signal and the most recent earlier peak in the input, and expressing this as a phase. Multiples of 2π are then added or subtracted until this phase is in the normal range $-\pi$ to π .

```
[4]: # Use local maxima for input and output signals to estimate gain and phase
from scipy.signal import argrelmax

xv1mi = argrelmax(xv)[0]; # locations of local maxima
yv1mi = argrelmax(yv)[0];
if len(xv1mi)<2 or len(yv1mi)<2: raise RuntimeError # not enough peaks found
xlmi = xv1mi[-1]; ylmi = yv1mi[-1]; # locations of last maximum

# Required quantities
gain = yv[ylmi]/xv[xlmi]; # ratio of peak amplitudes
tdelay = tv[xlmi] - tv[ylmi]; # time delay (seconds) output peak relative to
    ↪ input
theta = 2*np.pi*tdelay/T0; # delay in units of radians
while theta<=np.pi: theta = theta + 2*np.pi; # add or subtract multiples of
    ↪ 2*pi until in (-pi.pi]
while theta>np.pi: theta = theta - 2*np.pi;

print('Last output peak: y(', tv[ylmi], ') =', yv[ylmi]);
print('Preceeding input peak: x(', tv[xlmi], ') =', xv[xlmi]);
print('Gain: A =', gain);
print('Phase lag: theta =', theta, 'radians');
```

Last output peak: y(19.64) = 0.7088803724640561

Preceeding input peak: x(18.85) = 0.9999999013971618

Gain: A = 0.7088804423616797

Phase lag: `theta = -0.7899999999999991` radians

Some discussion of how the phase is represented is useful. Shifting a signal by a phase of 2π radians corresponds to a time shift of one cycle or one wavelength T_0 , where $T_0 = 2\pi/\omega_0$. Thus, for example, a phase shift of π corresponds to a time shift of half a wavelength, and $\pi/2$ to a shift of a quarter wavelength. The actual length of this shift therefore depends on the particular frequency we're considering.

1.3 Investigation

The code in the previous section is combined into a single script below.

```
[5]: %run src/labX_preamble.py # For internal notebook functions
```

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```
[6]: %%writefileexec src/lab_freqresprclp-1.py -s # dump cell to file before execute
```

```
import numpy as np
from scipy.signal import argrelmax
import matplotlib.pyplot as plt
%matplotlib inline

def rclowpass_resp(xv,n0,y0,TdRC,bflag=0):
    """RC lowpass output from given input using standard Euler
    xv: input signal (array)
    n0, y0: auxiliary condition yv[n0] = y0
    TdRC: value T/RC
    bflag: method Euler forward (0) or backward (1)
    returns yv: output signal (array)
    """

    yv = np.zeros(xv.shape);
    yv[n0] = y0;

    if bflag==0:
        # Forward Euler in each direction
        for n in range(n0,len(xv)-1):
            yv[n+1] = yv[n] + TdRC*(xv[n] - yv[n]); # forward recursion
        for n in range(n0,0,-1):
            yv[n-1] = yv[n] - TdRC*(xv[n] - yv[n]); # reverse recursion
    else:
        # Backward (implicit) Euler in each direction
        for n in range(n0,len(xv)-1):
            yv[n+1] = 1/(1+TdRC)*yv[n] + TdRC/(1+TdRC)*xv[n+1]; # forward
            ↪ recursion
        for n in range(n0,0,-1):
            yv[n-1] = 1/(1-TdRC)*yv[n] - TdRC/(1-TdRC)*xv[n-1]; # reverse
            ↪ recursion
```

```

    return yv;

# Input signal and discretisation
omega0 = 1; # current frequency
T = 0.01; # small number
if omega0==0: T0 = np.inf;
else: T0 = 2*np.pi/omega0; # waveform period (seconds) for current omega0
nv = np.arange(-200,2000); # discrete signal indices n
tv = nv*T; # time values corresponding to indices t=nT
xv = np.zeros(tv.shape);
for i in range(0,len(xv)):
    xv[i] = np.cos(omega0*tv[i])*(tv[i]>=0);

# Response for given initial condition
RC = 1; TdRC = T/RC;
n0 = np.where(nv==0)[0][0]; # find location in nv with zero value
y0 = 0.2;
yv = rclowpass_resp(xv,n0,y0,TdRC,0);
fig = plt.figure(); ph = plt.plot(tv,xv,'r-',tv,yv,'g-');
plt.xlabel('t'); plt.legend(['Input x(t)','Output y(t)']); plt.ylim((-1.5,1.5));

# Use local maxima for input and output signals to estimate gain and phase
xvلمي = argrelmax(xv)[0]; # locations of local maxima
yvلمي = argrelmax(yv)[0];
if len(xvلمي)<2 or len(yvلمي)<2: raise RuntimeError # not enough peaks found
xlلمي = xvلمي[-1]; ylلمي = yvلمي[-1]; # locations of last maximum

# Required quantities
gain = yv[ylلمي]/xv[xللمي]; # ratio of peak amplitudes
tdelay = tv[xللمي] - tv[ylلمي]; # time delay (seconds) output peak relative to
    input
theta = 2*np.pi*tdelay/T0; # delay in units of radians
while theta<=np.pi: theta = theta + 2*np.pi; # add or subtract multiples of
    2*pi until in (-pi,pi]
while theta>np.pi: theta = theta - 2*np.pi;

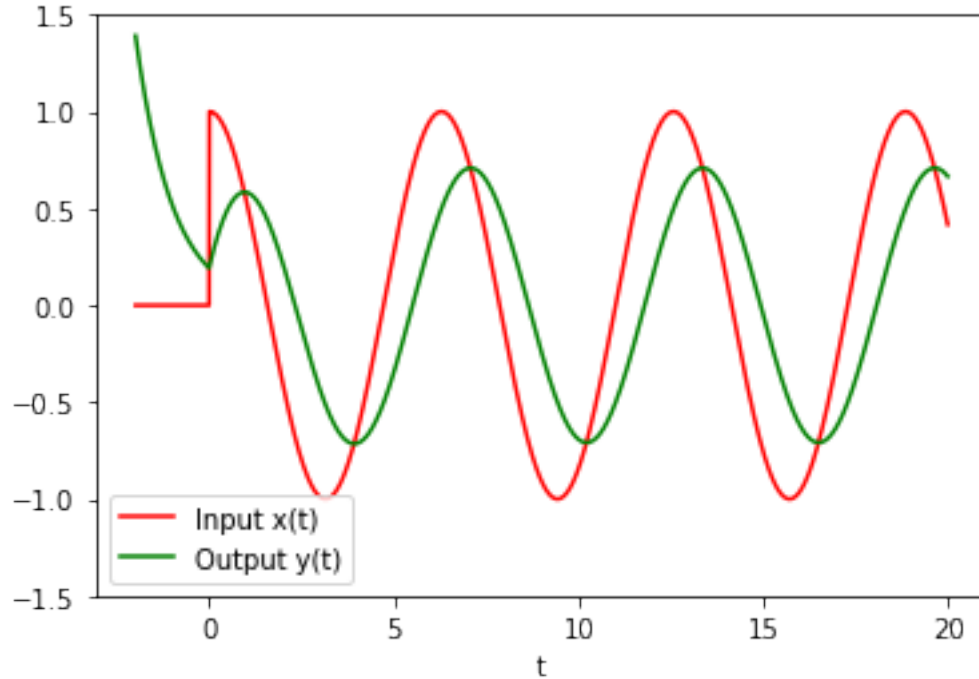
print('Last output peak: y(', tv[ylلمي], ') =', yv[ylلمي]);
print('Preceeding input peak: x(', tv[xللمي], ') =', xv[xللمي]);
print('Gain: A =', gain);
print('Phase lag: theta =', theta, 'radians');

```

```

Last output peak: y( 19.64 ) = 0.7088803724640561
Preceeding input peak: x( 18.85 ) = 0.9999999013971618
Gain: A = 0.7088804423616797
Phase lag: theta = -0.7899999999999991 radians

```



The following items involve investigations that you should perform to investigate the nature of the problem being addressed:

- Run the code above for the case of $\omega = 0$, where the input simply becomes the step function at the origin. The method used for phase estimation will fail because there are no peaks, but the input-output plot is still correct. By changing the value of RC confirm that after about $5RC$ seconds have passed the transient has decayed almost entirely away, and the system reaches its steady state (in this case DC) response. Convince yourself that this is true regardless of the initial value y_0 . The quantity $\tau = RC$ is called the *time constant* of the circuit, and it determines the time duration of transients in the system. Specifically, after one time constant the transient response will have decayed to $e^{-1} = 0.368 \approx 37\%$ of its initial value and after k time constants the it will have decayed to e^{-k} times its initial value.
- Consider changing the frequency ω_0 of the input. Try for example $\omega_0 = 1, 2, 5, 10$ radians per second. Make a note of the gain A and phase lag θ for each case. You should observe that the gain *decreases* as ω_0 increases. In other words the circuit attenuates high frequencies more than low frequencies, and is hence a lowpass filter. Confirm that the steady-state gain and phase lags are independent of the initial conditions.
- The method of using locations of peaks in input and output signals to determine the phase response requires T to be sufficiently small for it to be reliable. If we want an estimate accurate to one part in 60, for example, then we need more than 60 samples over a wavelength T_0 , or $T < \frac{1}{60}T_0$. With $\omega_0 = 2\pi/T_0$ and the value $T = 0.01$ this provides a usable limit of $\omega_0 \leq 10$.
- Observe that the code as written doesn't work for $\omega_0 = 0.1$, because the interval over which we've calculated signal values is too short for the peak search method to work. Modify `nv` to calculate the signal over a longer time period using say `nv = np.arange(-2000,20000)`:

the method then works for $\omega_0 = 0.1$ but now fails for $\omega = 0.005$. It also requires more computation.

1.4 Frequency representation and Bode plot

In this course we will see (or might already have seen) that the RC circuit has a transfer function

$$H(\omega) = \frac{1/RC}{1/RC + j\omega}.$$

This system has a real impulse response so $H(-\omega) = H^*(\omega)$. The exact analytical form for the steady-state response to the input signal $x(t) = \cos(\omega t)$ is quite easily shown to be

$$y(t) = |H(\omega)| \cos(\omega t + \angle H(\omega)).$$

This workbook investigates ways of visualising the transfer function $H(\omega)$. This is not trivial because, even though ω is real, $H(\omega)$ takes on complex values. Thus we need to plot both magnitude and phase as functions of frequency. Also, it turns out that expressing both domain and range on logarithmic axes makes it much easier to characterise the behaviour of the system. This leads to the conventional *Bode plot*.

The simple way to visualise $H(\omega)$ is to choose a set of frequencies of interest and store them in an array `wv`. We can then evaluate $H(\omega)$ at these points, storing the results in another array `Hv`. Note that since `Hv` will be complex we cannot just plot it. Instead for any ω we can write the frequency response in magnitude-phase form

$$H(\omega) = |H(\omega)|e^{j\angle H(\omega)}$$

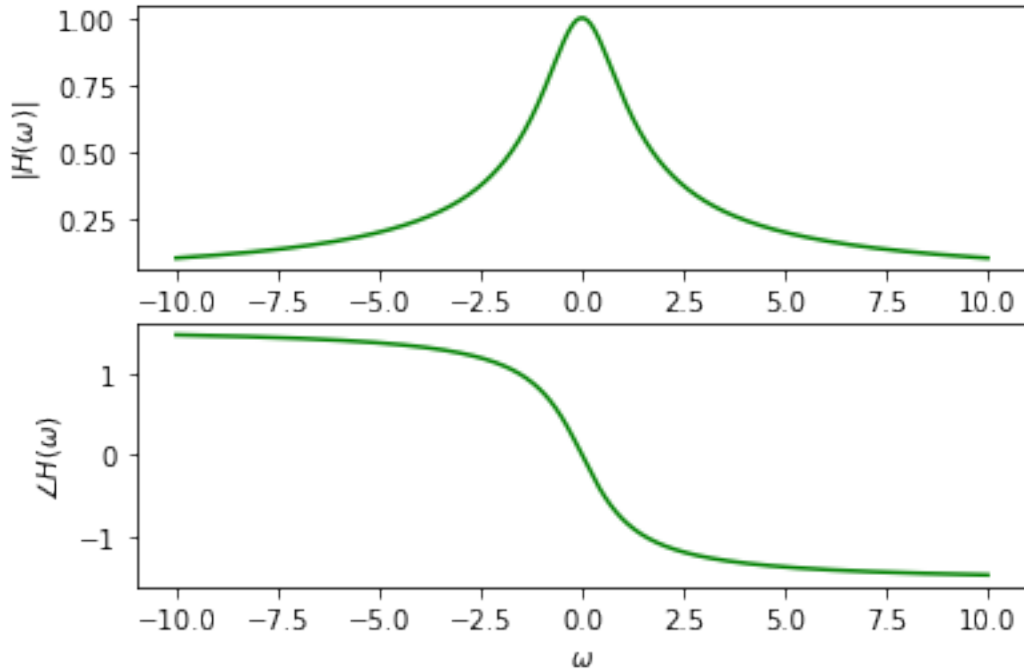
and make separate plots of $|H(\omega)|$ and $\angle H(\omega)$.

```
[7]: # Two-sided Bode plot linear-linear

import numpy as np
import matplotlib.pyplot as plt
%matplotlib inline

# Frequency response
RC = 1;
wv = np.linspace(-10, 10, 1000);
Hv = (1/RC)/((1/RC)+1j*wv);

# Display
fh, ax = plt.subplots(2);
ax[0].plot(wv, np.abs(Hv), c='g'); ax[0].set_ylabel(r'$|H(\omega)|$');
ax[1].plot(wv, np.angle(Hv), c='g'); ax[1].set_ylabel(r'$\angle H(\omega)$');
plt.xlabel('$\omega$');
```

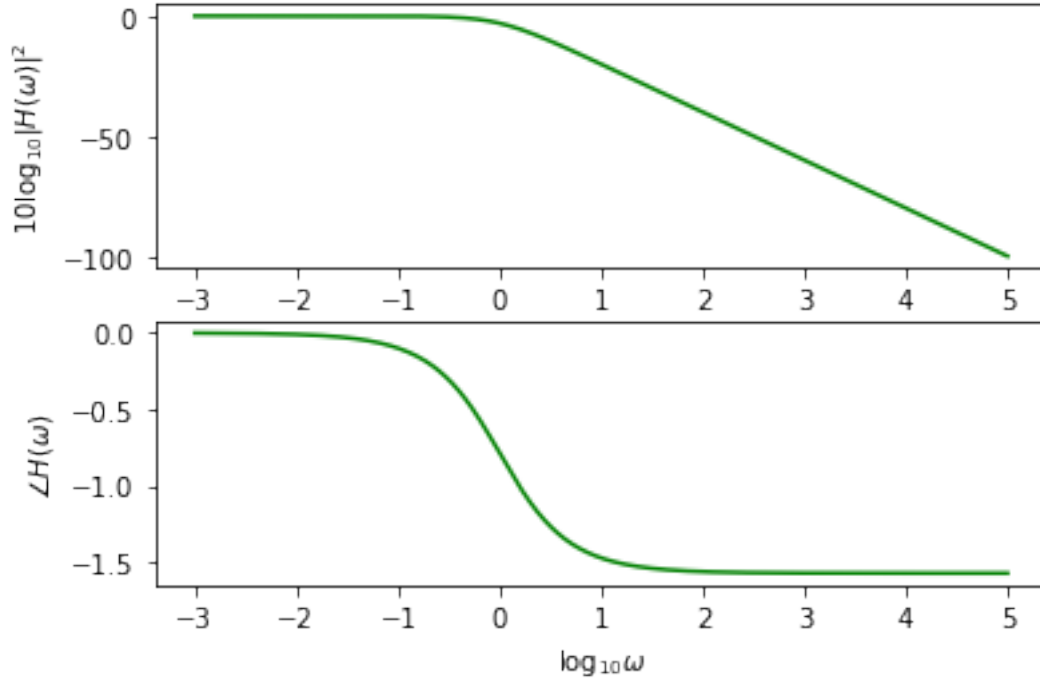
It is difficult from the plots above to see the effect of the RC value for the circuit. If you increase it the magnitude plot for example become more “peaky”, but the effect is hard to characterise.

We note firstly that the condition $H(-\omega) = H^*(\omega)$ means that the magnitude $|H(\omega)|$ is always even and the phase $\angle H(\omega)$ is always odd. We may as well therefore only plot them for positive frequencies $\omega > 0$ - all the useful information is still available.

Secondly, the magnitude plot above is linear in frequency ω and linear in the gain $|H(\omega)|$. A log-log plot turns out to be more useful. We can redo the plots above, but this time with the gain in decibels $G_{dB}(\omega) = 10 \log_{10} |H(\omega)|^2$ plotted against logarithmic frequency $\log_{10} \omega$:

```
[8]: # One-sided Bode plot log-log
lrv = np.linspace(-3, 5, 1000); # linear points in log space
wv = 10**lrv; # actual frequencies
Hv = (1/RC)/((1/RC)+1j*wv); # frequency response
dbHv = 10*np.log10(np.abs(Hv)**2); # magnitude response in dB

fh, ax = plt.subplots(2);
ax[0].plot(lrv, dbHv, c='g'); ax[0].set_ylabel('$10 \log_{10} |H(\omega)|^2$');
ax[1].plot(lrv, np.angle(Hv), c='g'); ax[1].set_ylabel(r'$\angle H(\omega)$');
plt.xlabel('$\log_{10} \omega$');
```



The magnitude plot now has two clear regions: a flat passband and a stopband with a linear roll-off, separated by a “knee”. We can investigate these two regions in more detail. First we note that the transfer function can be written as

$$H(\omega) = \frac{1}{1 + j\omega RC}.$$

The gain in dB can thus be written as

$$G_{dB}(\omega) = 10 \log_{10} |H(\omega)|^2 = 10 \log_{10} \left(\frac{1}{1 + j\omega RC} \frac{1}{1 - j\omega RC} \right) = -10 \log_{10}(1 + (\omega RC)^2).$$

Consider the term in the logarithm:

- For the case $\omega RC \ll 1$, or $\omega \ll 1/(RC)$, we have approximately

$$G_{dB}(\omega) \approx -10 \log_{10}(1) = 0.$$

This is the one asymptote.

- For the case $\omega RC \gg 1$, or $\omega \gg 1/(RC)$,

$$G_{dB}(\omega) \approx -10 \log_{10}(\omega RC)^2 = -20 \log_{10}(\omega RC) = -20[\log_{10} \omega] - 20 \log_{10}(RC).$$

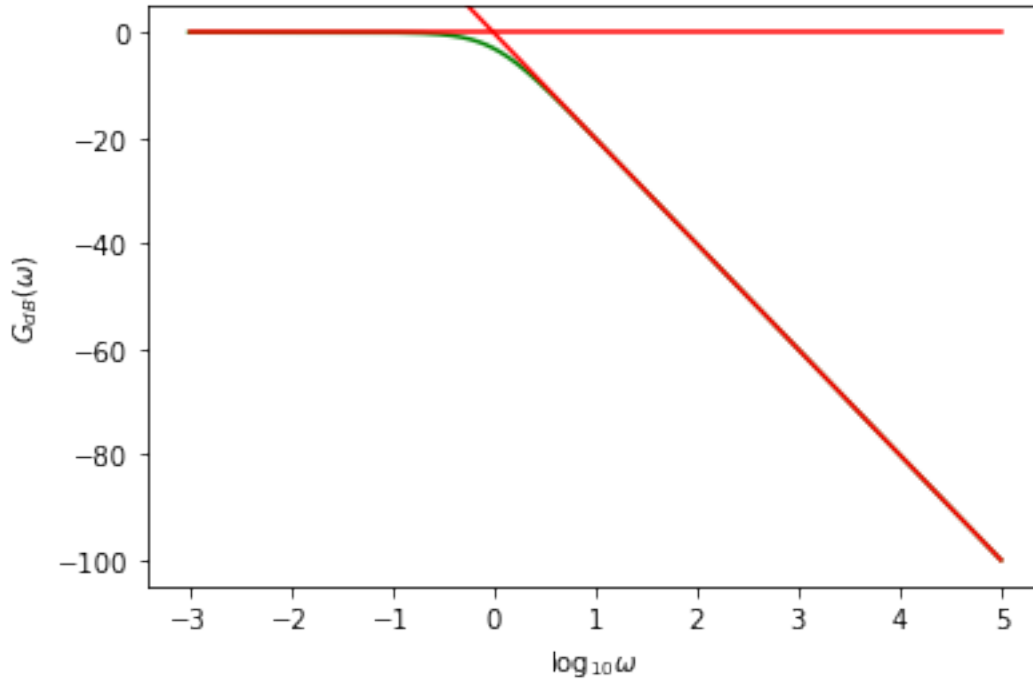
This is another asymptote.

These two asymptotes cross at $\omega = 1/(RC)$, the location of the knee. This is called the *cutoff frequency* of the filter, and it marks the transition from the passband to the stopband.

We can redo the magnitude plot and show these two asymptotes:

```
[9]: fh = plt.figure();
plt.plot(lwv, dbHv, c='g'); plt.ylabel('$G_{dB}(\omega)$');
plt.xlabel('$\log_{10} \omega$');
yax = plt.gca().get_ylim();

as0v = np.zeros(lwv.shape);
as1v = -20*lwv - 20*np.log10(RC);
plt.plot(lwv, as0v, 'r', lwv, as1v, 'r');
plt.gca().set_ylim(yax);
```



In this last plot the slope of the roll-off is seen to be 20dB for every unit increment of the x-axis $\log_{10} \omega$. However, note that $\log_{10} \omega = 0$ corresponds to $\omega = 1$, $\log_{10} \omega = 1$ corresponds to $\omega = 10$, $\log_{10} \omega = 2$ corresponds to $\omega = 100$, and so on. Thus an increase of one unit on the log frequency axis corresponds to an increase in frequency by a factor of 10. We call a factor of 10 increase in frequency a *decade*.

Thus the first-order lowpass filter has a roll-off of 20dB per decade once above the cutoff at $\omega_c = 1/(RC)$.

We could also plot the gain in dB against $\log_2 \omega$. Since $\log_{10} \omega = \log_2(\omega)/\log_2(10)$, once above the knee we can write

$$G_{dB}(\omega) \approx -\frac{20}{\log_2(10)}[\log_2 \omega] - 20 \log_{10}(RC).$$

An increase in $\log_2 \omega$ by one unit corresponds to a doubling of the frequency, called an *octave*. We see that a one-unit increase in \log_2 frequency results in a reduction in gain of $20/\log_2(10) \approx 6$ dB.

In other words, the first-order lowpass filter has a roll-off of 6dB per octave once above the cutoff.

2 Tasks

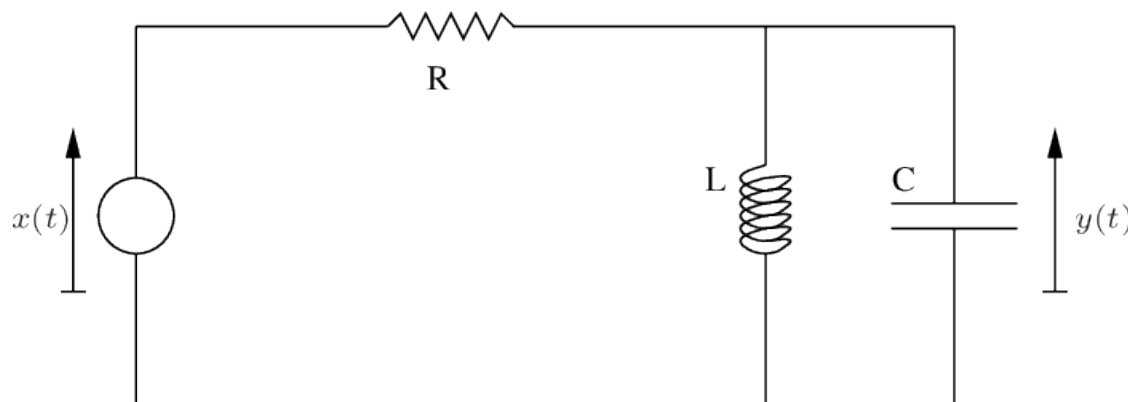
These tasks involve writing code, or modifying existing code, to meet the objectives described.

1. Make a plot of the step response $g(t)$ of an RC lowpass filter for the case $RC = 1$. The domain of the plot should be from $t = -2$ to $t = 15$. Assume an initial rest condition $g(-2) = 0$. (6 marks)
2. The first-order **highpass** filter - which has R and C swapped in the previous RC circuit - has a frequency response given by

$$H(\omega) = \frac{j\omega RC}{1 + j\omega RC}.$$

Generate a Bode plot for this system, both magnitude and phase, for the case of $RC = 1$. The gain should be in dB, and both plots should use $\log_2 \omega$ as the independent variable. The frequency range should extend from $\log_2 \omega = -6$ to $\log_2 \omega = 6$. Find the two asymptotes and include them in the magnitude plot. (6 marks)

3. A second-order RLC circuit



has transfer function

$$H(\omega) = \frac{\frac{1}{RC}(j\omega)}{(j\omega)^2 + \frac{1}{RC}(j\omega) + \frac{1}{LC}}.$$

It turns out that the fundamental parameters for this circuit are the resonant frequency $\omega_0 = 1/\sqrt{LC}$ and the damping factor $\alpha = 1/(2RC)$, giving

$$H(\omega) = \frac{2\alpha(j\omega)}{(j\omega)^2 + 2\alpha(j\omega) + \omega_0^2}.$$

Generate a Bode plot for the system for $R = 10$ and $L = C = 1$. The gain should again be expressed in dB with $\log_2 \omega$ as the independent variable, and the response should be shown for values of $\log_2 \omega$ over the range -6 to 6 . Comment on the filter characteristic, i.e. lowpass, highpass, bandpass, etc. (8 marks)