电子科技大学
格拉斯哥学院
Glasgow College, UESTC

Communication Circuits Design – 2018-19, semester II

Lab 2 – Week5

Student's Chinese name	郑长刚
Student's English name	Changgang Zheng
Student's UESTC ID#	2016200302027
Student's UoG ID#	2289258Z

Nonlinear mixing principles

The objective of this lab is to become familiar with the process of generating new frequencies by mixing together two signals at different frequencies as inputs of a nonlinear device. In this case the nonlinear device will be a simple Germanium diode.

Theory recap.

When two sinusoidal signals of different frequencies f_1 and f_2 are applied simultaneously to a nonlinear device, the mixing effect generates several different output frequencies including:

- The first and second harmonics of the original signals, so f_1 and f_2, as well as $2f_1$ and $2f_2$
- The sum and difference components so f_1+f_2 and $|f_1-f_2|$
- A DC offset at 0 Hz

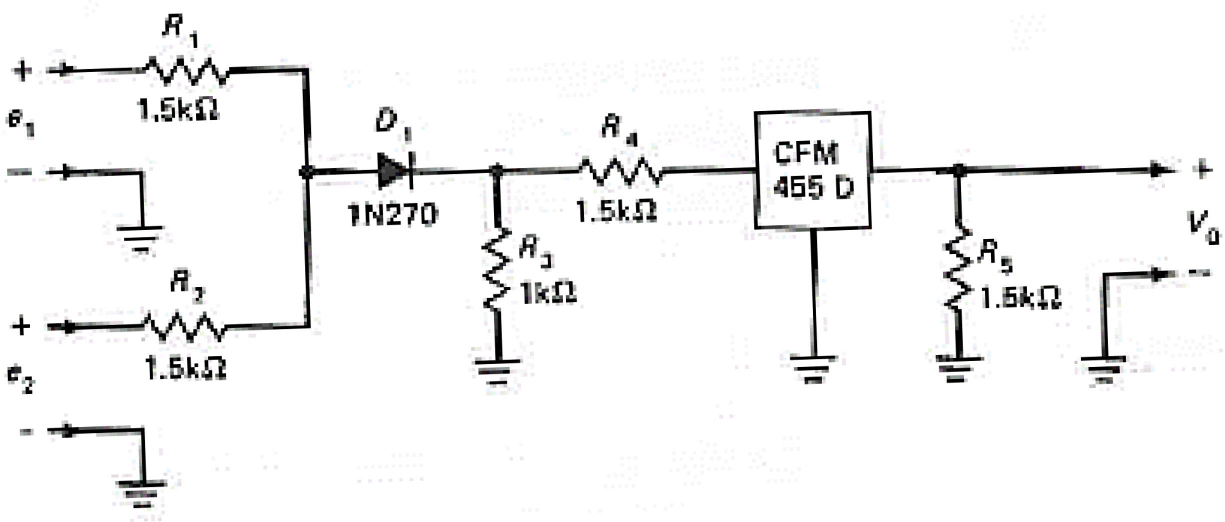
One method to prove this principle is to feed the output signal of the mixer through a sharp band-pass filter tuned to each of the expected output frequencies, and then check for the presence of a sinusoidal signal of that particular expected frequency at the output of the band-pass filter.

However, sharp filters, such as ceramic or crystal filters, are not available for any possible frequency, but typically only for standard frequencies used in radio design such as 455 kHz and 10.7 MHz

So, an alternate approach used in this experiment, is to carefully select the two input frequencies so that only one of the expected output frequencies of nonlinear mixing will be near the standard pass band of the available band-pass filter (in this case, the 455 kHz one).

Practical procedure.

1. **Build the circuit shown in the figure below – Note the two inputs e1 e2 from the signal generator into the voltage divider of 2 resistors R1 R2, followed by diode and band-pass filter. The final output Vo goes to the oscilloscope.**



2. **Set the amplitude of the two input signals to 10 V peak to peak. Set the frequency of signal e1 to 455 kHz and e2 to 200 kHz. Monitor the output voltage using the oscilloscope.**

   Carefully fine-tune the frequency of one of the two input signals so that the output reaches its maximum amplitude. Use the oscilloscope to measure the maximum amplitude and exact frequency at that point, and note the values in the table below.

3. **Repeat the step 2 for the following combinations of input frequencies to populate the table.**
   a. **f1 = 200 kHz and f2= 455 kHz**
   b. **f1 = 227.5 kHz and f2= 300 kHz**
   c. **f1 = 300 kHz and f2= 227.5 kHz**
   d. **f1 = 355 kHz and f2= 100 kHz**
   e. **f1 = 755 kHz and f2= 300 kHz**
   f. **f1 = 295 kHz and f2= 80 kHz**

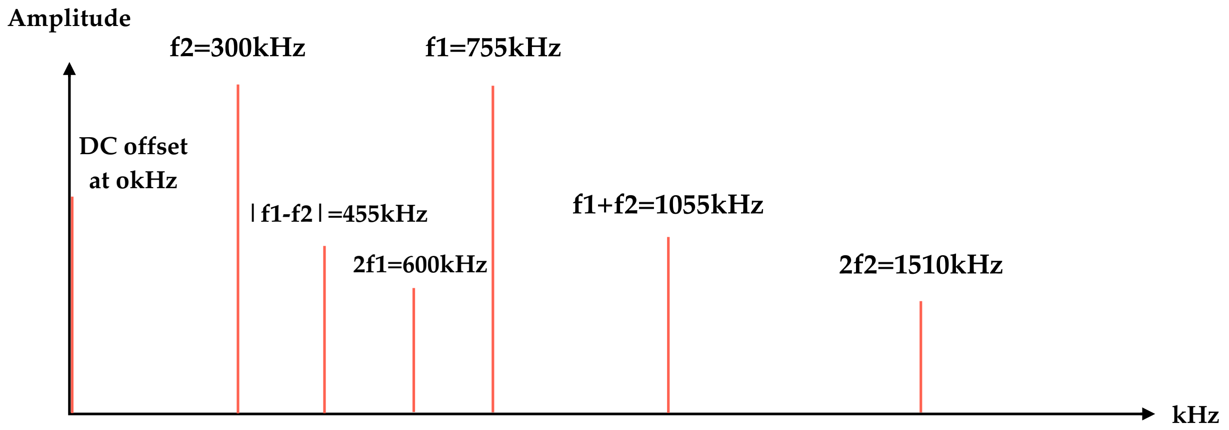
   (if in serious trouble for this, carry on and see more info at question 7)

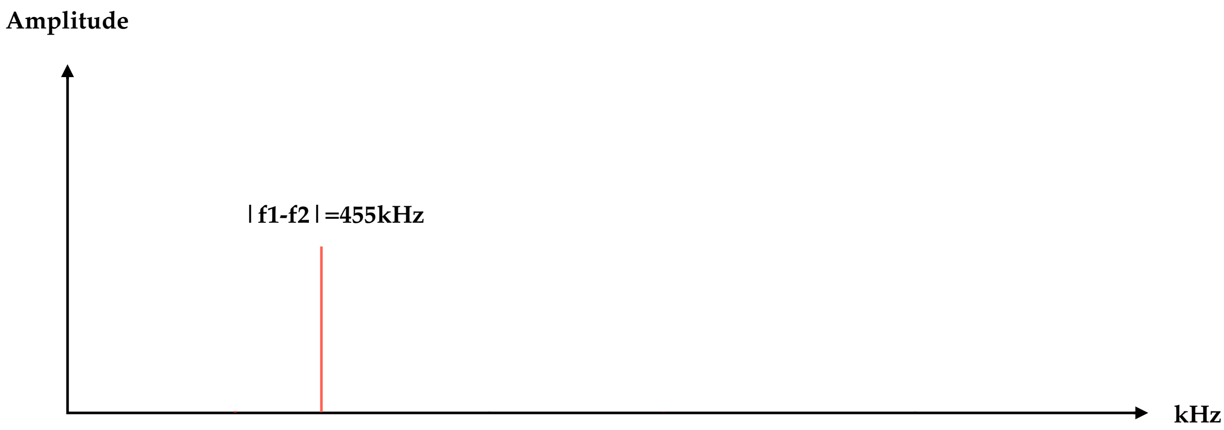
Input f1	Input f2	OA_1	OF_2	2f1	2f2	f1+f2	|f1-f2|
455kHz	200 kHz	7.84V	454 kHz	910 kHz	400 kHz	655 kHz	255 kHz
200 kHz	**455 kHz**	7.85V	454 kHz	400 kHz	910 kHz	655 kHz	255 kHz
227.5 kHz	300 kHz	1.58V	458 kHz	**455 kHz**	600 kHz	528 kHz	72.5 kHz
300 kHz	227.5 kHz	1.58V	458 kHz	600 kHz	**455 kHz**	528 kHz	72.5 kHz
355 kHz	100 kHz	4.22V	455 kHz	710 kHz	200 kHz	**455 kHz**	255 kHz
755 kHz	300 kHz	4.32V	454 kHz	1510 kHz	600 kHz	1055 kHz	**455 kHz**
295 kHz	30 kHz	0.308V	455 kHz	590 kHz	60 kHz	325 kHz	265 kHz

OA_1: Output Amplitude

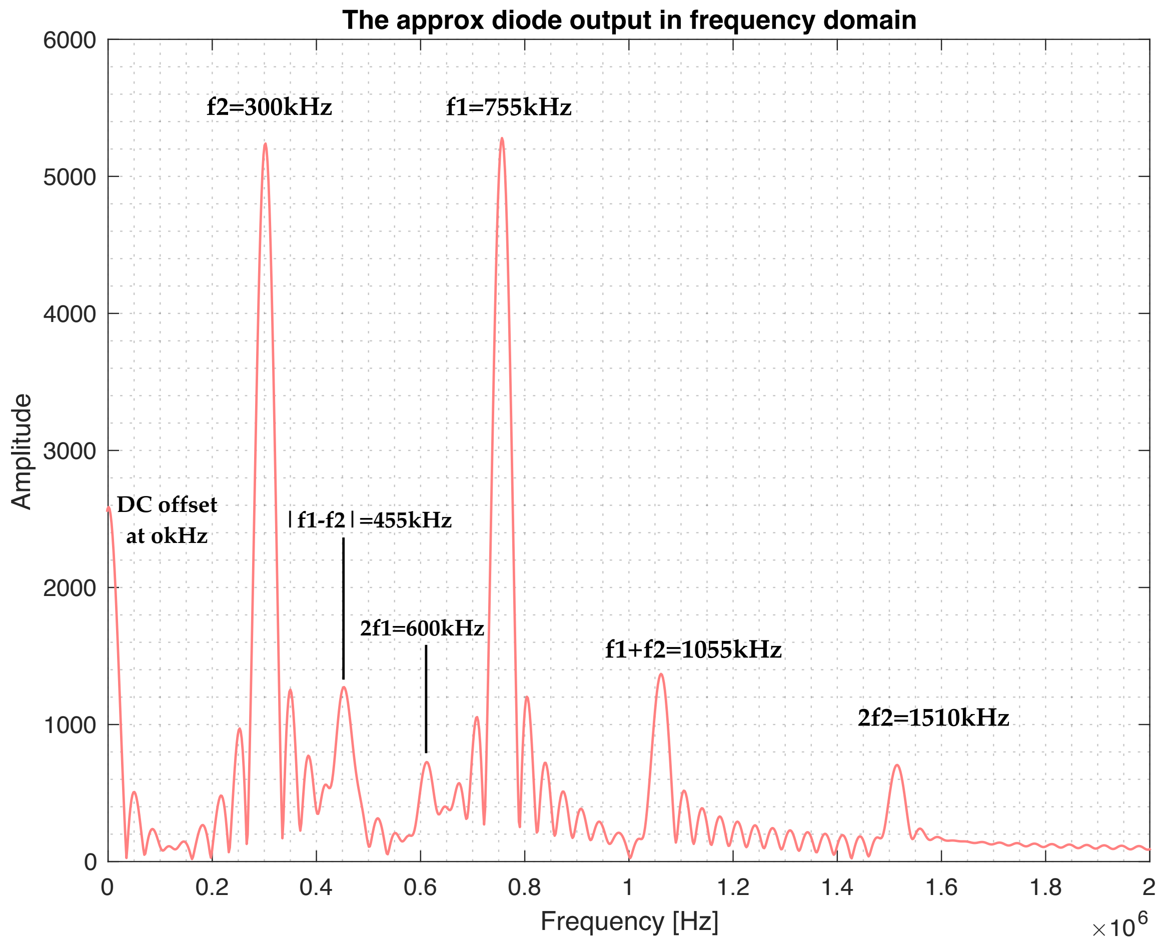
OF_2: Output frequency

4. **Sketch the expected spectrum of the output signal just after the diode (between the diode and resistor R3) and after the ceramic filter (on resistor R5 basically) when the input signals are those given <u>in case 3e</u> above. Do not worry about the relative amplitude of each component, just make sure you sketch all the components**





I also use the program from MATLAB to plot the result, which is similar to what I plot above. The code used to generate this is based on the program download from Moodle.



5. Which outputs were of largest amplitude in the table at step 3? (tick in the table the corresponding box to the nearest expected output among f_1 f_2 $2f_1$ $2f_2$ f_1+f_2 $|f_1-f_2|$)

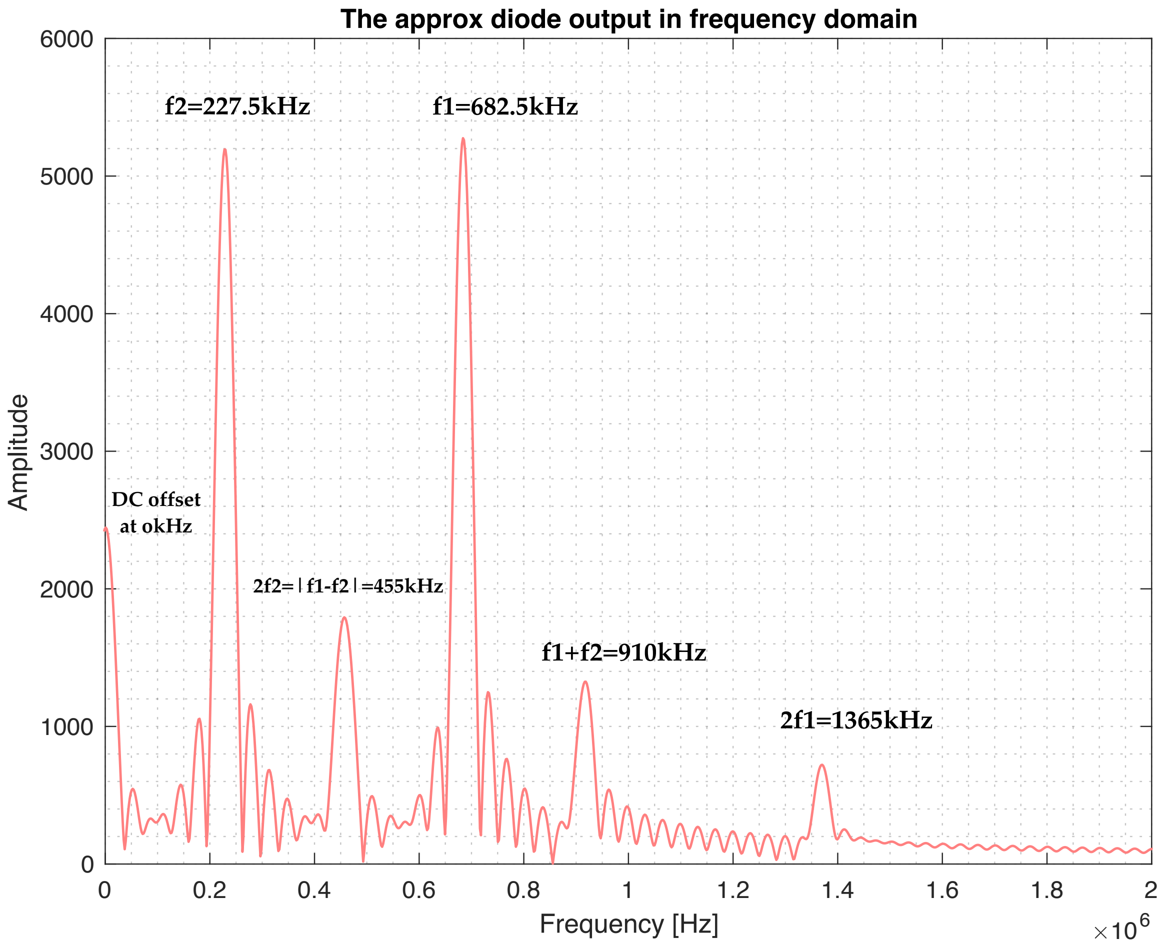
Input f1	Input f2	OA_1	OF_2	2f1	2f2	f1+f2	|f1-f2|
455kHz	200 kHz	7.84V	454 kHz	910 kHz	400 kHz	655 kHz	255 kHz
200 kHz	**455 kHz**	7.85V	454 kHz	400 kHz	910 kHz	655 kHz	255 kHz
227.5 kHz	300 kHz	1.58V	458 kHz	**455 kHz**	600 kHz	528 kHz	72.5 kHz
300 kHz	227.5 kHz	1.58V	458 kHz	600 kHz	**455 kHz**	528 kHz	72.5 kHz
355 kHz	100 kHz	4.22V	455 kHz	710 kHz	200 kHz	**455 kHz**	255 kHz
755 kHz	300 kHz	4.32V	454 kHz	1510 kHz	600 kHz	1055 kHz	**455 kHz**
295 kHz	30 kHz	0.308V	455 kHz	590 kHz	60 kHz	325 kHz	265 kHz

OA_1: Output Amplitude

OF_2: Output frequency

6. If we completed step 2 using a new combination f1 = 682.5 kHz and f2= 227.5 kHz, what problem would occur in looking for the output difference frequency component, which is $|f_1-f_2|$?

For this question, we can first calculate and find: 2f1=1365kHz, f1+f2=910kHz, 2f2=|f1-f2|=455kHz. We find that 2f2=|f1-f2|, which means it is difficult to extract the frequency component |f1-f2| as it overlaps by other frequency range. So, we need to change the frequency to deal with the overlapping issue. As shown in the following figure.



7. In question 3f, the only output frequency that is within the pass band of the band-pass filter would be equal to f_1+2f_2, which is typically far smaller than the desired sum or difference components. This is called a 3rd order component (as generated by the sum of one input component plus twice the other).
How many dBs down is this output frequency compared to the sum or difference components (2nd order harmonics) that you have recorded at question 3e and 3d?

According to the equation $20\log_{10}$(Output/Input), we can calculate the reduction in dBs. So, for question 3d, we have reduction:

$$20\log_{10}\left(\frac{\text{Output}}{\text{Input}}\right) = 20\log_{10}\left(\frac{0.308\text{V}}{4.22\text{V}}\right) = -22.792\text{ dB}$$

So, for question 3e, we have reduction:

$$20\log_{10}\left(\frac{\text{Output}}{\text{Input}}\right) = 20\log_{10}\left(\frac{0.308\text{V}}{4.32\text{V}}\right) = -22.995\text{ dB}$$

We can see that the reduction of 3e and 3f is close. And the result tells us that the same order component have similar amplitude and the higher the order is, the higher the reduction is.

8. What would be the expression of the other 3rd components? And what would be their frequency values if we refer to the case 3f?

From the 3f, I can find:

f1 = 295 kHz
f2= 80 kHz

The expression of 3rd components is 3f1, 3f2, 2f1+f2, 2f2+f1, |2f1-f2|, |2f2-f1|:

3f1= 885 kHz
3f2 = 240 kHz
2f1+f2 = 670 kHz
2f2+f1= 455 kHz
|2f1-f2|= 510 kHz
|2f2-f1|= 135 kHz

Thus, we can get the final result shown in the following table:

Components	Frequency(kHz)
3f1	885
3f2	240
2f1+f2	670
2f2+f1	455
|2f1-f2|	510
|2f2-f1|	135

Reference : The following code is used to generate the above graph, which is basically got from Moodle:

```
clear all
clc

%%
fs=500e6; %Sampling frequency of the signals (500 MHz, high enough to consider the signal almost continuous)
ts=1/fs;  %Sampling time, the inverse of sampling frequency
t=[0:ts:0.03e-3]; %time axis
f1=682.5e3; %input frequency 1
f2=227.5e3; %input frequency 2
phase=0; %the phase shift (can be zero, pi, pi/2...MATLAB wants it in radians)
sig2=cos(2*pi*f1.*t);
sig1=cos(2*pi*f2.*t+phase);
figure
title('The two inputs in time domain')
plot(t,sig1,'LineWidth',1)
hold on
plot(t,sig2,'color',[1,0.5,0.5],'LineWidth',1)
xlabel('Time [s]'); ylabel('Amplitude')
legend('sig1','sig2')

nfft=2.^nextpow2(length(sig2))*16; %Length of the FFT
sig1f=fft(sig1,nfft);
sig2f=fft(sig2,nfft);
faxis=linspace(-fs/2,fs/2,length(sig1f));
grid minor

figure
plot(faxis,fftshift(abs(sig1f)),'LineWidth',1)
hold on
plot(faxis,fftshift(abs(sig2f)),'LineWidth',1,'color',[1,0.5,0.5])
xlim([0 1e6])
title('The two inputs in frequency domain')
xlabel('Frequency [Hz]'); ylabel('Amplitude')
legend('sig1','sig2')
grid minor

%%
vt=0.7; %A general value for the Vt of the diode, say 0.7 V, not crucial here
siglin=(sig1+sig2)./vt/2; %linear term of Taylor series approximation
sigsquare=((sig1+sig2).^2)./8/vt; %square term of Taylor series approximation
sigdiode=siglin+sigsquare; %sum them as Taylor series
figure
plot(t,sigdiode,'LineWidth',1,'color',[1,0.5,0.5])
title('The approx diode output in time domain')
xlabel('Time [s]'); ylabel('Amplitude')
grid minor

sigdiodef=fft(sigdiode,nfft);
figure
plot(faxis,fftshift(abs(sigdiodef)),'LineWidth',1,'color',[1,0.5,0.5])
xlim([0 2e6])
title('The approx diode output in frequency domain')
xlabel('Frequency [Hz]'); ylabel('Amplitude')
grid minor
```