

## ELG3175 INTRODUCTION TO COMMUNICATION SYSTEMS

### LABORATORY I: Signals, Systems and Spectra

#### Introduction:

In this lab, students will familiarize themselves with the lab instruments and equipment, will generate and measure signals and their spectra as well as the frequency and impulse responses of filters.

#### Reference:

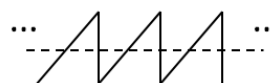
1. Chapters on Fourier series and frequency/impulse response of linear systems in the course textbook.
2. Manuals of Waveform Generator (Agilent 33500B), Oscilloscope (Rohde & Schwarz HMO722), and Spectrum Analyzer (Rohde & Schwarz HMS3010), available on the course website.

#### Preparation:

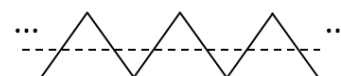
1. Read the extracts from the manuals of Waveform Generator (Agilent 33500B), Oscilloscope (Rohde & Schwarz HMO722), and Spectrum Analyzer (Rohde & Schwarz HMS3010), posted on the course web page to become familiar with the instruments and how to operate it. Full manuals are also available for future reference.

2. Find the Fourier series of a rectangular pulse train with duty cycle  $d = \tau/T$ , where  $\tau$  is the pulse duration and  $T$  is the period. Assuming that the amplitude is 1, the period is  $1 \mu\text{s}$  and  $d=50\%$  (% duty cycle is  $(\tau/T)100\%$ ), what are the fundamental frequency, harmonics and their amplitudes? Repeat the same for  $d=10\%$ . What are the frequencies of the missing harmonics (i.e. those for which the amplitude is zero) in both cases ?

3. Do the same for a sawtooth pulse train (waveform), as shown below (no duty cycle here, just amplitude and period from Q2).



Repeat the same for a triangle pulse train.



4. What is the output of an ideal low pass filter if the input is a square wave and  $f_c = 1.5f_0$ , where  $f_c$  and  $f_0$  are cut-off frequency of the filter and the square wave fundamental frequency, respectively? How can you approximate the output if the input is such that  $f_c = 10f_0$  ? You will test your conclusions in the lab experiment.

5. Bring at least 2 USB sticks to the Lab to record all your measurements (as screenshots in .png or .jpg files), 2<sup>nd</sup> stick is required in case 1<sup>st</sup> one does not work/fails. The format of the USB should be FAT32. All screenshots must be recorded on the USB (pictures taken by cell phones are **not acceptable**) and must be included in the reports in the original form, **no editing at all** (e.g. cropping etc.).

#### Apparatus:

1. Waveform (signal) Generator (Agilent 33500B)
2. Spectrum Analyzer (Rohde & Schwarz HMS3010)
3. Dual Channel Oscilloscope (Rohde & Schwarz HMO722)
4. Krohn-Hite 3384 Filter Unit



for every measurement you make and include them in the report. Compare the results and explain what you observe.

3. Set the RBW to 30 kHz. Change the video bandwidth (VBW) on the SA to 100 kHz, 30 kHz, 10 kHz, 3 kHz, and 1 kHz by pressing “BANDW” button and using “VBW” softkey. Comment on the impact of the VBW on the measured spectrum.

4. Set the VBW to 30 kHz. Change the signal to a square waveform (50% duty cycle) with 1 V<sub>pp</sub> and observe the output on both the OS and the SA. Set the units on the SA to V and the start and stop frequencies on the SA to 100 kHz and 20.1 MHz, respectively, to clearly see the spectrum lines. Compare the amplitudes and frequencies of first 3 harmonics to your computations. Do the results agree with each other? Set the SA unit to dB $\mu$ V and, for more precise measurements of the amplitude, you can change the Range to 5 dB/DIV. Now change (double/half) the amplitude of the signal and observe the impact of this change on the spectrum. Do the same for the fundamental frequency.

5. Set the amplitude to 1 V<sub>pp</sub> and the frequency to 1 MHz on the SG. Decrease the duty cycle to 40%, 20%, 10%, and 2% and observe the spectrum on the SA. Comment on the impact of the duty cycle on the signal’s spectrum. Determine the frequencies of the missing harmonics for each duty cycle and compare them with your computations.

6. Change the signal to a sawtooth pulse train (ramp waveform 100% symmetry) with 1 V<sub>pp</sub> amplitude and the fundamental frequency of 100 kHz. Set the RBW to 10 kHz on the SA and the start and stop frequencies to 100 kHz and 1.1 MHz, respectively. Observe the output on both the Oscilloscope and the SA. Compare the amplitudes and frequencies of first 3 harmonics to your computations. Do the results agree with each other? Now, change the signal to a triangular waveform (50% symmetry) and repeat the previous steps.

7. Include in your report a detailed description of your lab activities in each step (including the screenshots), interpretation/explanation of the measurements and also the comparison between the measurements and your computations; explain any difference between the latter two.

## Part II: Periodic Signals, Fourier Series and Low-Pass Filter (LPF)

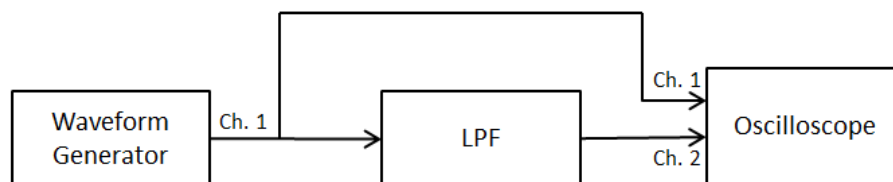


Fig. 2: measurement setup for Part II

1. In this part, we will observe the impact of a low-pass filter on periodic signals via the Fourier series representation of such signals. Assemble the experimental setup as shown in Fig. 2. Generate a 10 kHz square wave (50% duty cycle) signal with 1 V<sub>pp</sub> amplitude in Channel 1 of the SG (Waveform Generator). Set the low-pass filter (LPF) cut-off frequency  $f_c$  to 15 kHz, and the filter type to Butterworth (bu). Observe the input and output of the LPF on the oscilloscope. Explain the waveform of the output. Measure its amplitude and compare it with your prelab calculations. Do they agree with each other? For

all calculations, you may assume that the LPF is the ideal (brick-wall) low-pass filter with the given cut-off frequency. Increase  $f_c$  to 25 kHz, 35 kHz, 45 kHz and 55 kHz; each time observe the output and explain the changes you see. Finally, set  $f_c$  to 200 kHz, observe the output and explain what you see. How many terms of the input Fourier series will appear at the output each time?

2. Change the signal to a triangular pulse train (ramp with 50% symmetry), set  $f_c$  to 15 kHz and observe the output of the LPF. Compare its amplitude to your prelab calculations. Set now  $f_c = 200$  kHz, observe the output and explain the difference.

3. Generate a 10 kHz sinusoidal signal with 1 V<sub>pp</sub> amplitude in Channel 1 of the SG (waveform generator). Set the LPF to Butterworth (bu) type and  $f_c$  to 50 kHz. Observe both the input and the output of the filter on the Oscilloscope and measure the delay between the input and output. How does it relate to the filter's bandwidth? To test your guess on the relationship between the delay and the bandwidth, change  $f_c$  to 100 kHz and measure the delay again. Does it agree with what you have expected?

### Part III: Frequency Response of an LPF

1. Keep the setup of Fig. 2. Set the LPF  $f_c$  to 50 kHz and the low-pass mode, Butterworth type.

2. Apply a 1 kHz sinusoidal signal with 1 V<sub>pp</sub> amplitude to the input of the filter. Measure precisely the output signal's amplitude on the oscilloscope (hint: this can be done via the screen grid or via the measure function of the OS).

3. Repeat step 2 for at least 10 frequencies in the range 10-100 kHz to determine the overall frequency response. Measure precisely the frequency at which the filter output is half of its maximum output in power (i.e., the -3 dB point in the frequency response). This is the so-called -3 dB bandwidth of the filter. How does it compare to  $f_c$  ?

4. Plot the magnitude frequency response. Does it look similar to the ideal low-pass filter? What are the differences/similarities?

5. Change the type of filter to Bessel and repeat steps 2-4. Plot both frequency responses on the same graph and compare them. What are the differences/similarities? Which one is more close to the ideal LPF?

### Part IV: Impulse Response of an LPF

1. In this step we are going to measure the impulse response of the same LPF (at this stage, you should already have some idea as to how it should look like). This can be done by applying a rectangular input with sufficiently short pulse duration (which approximates the delta function in practice) to the LPF. To do so, start by generating a square wave signal (50% duty cycle) with 100 mV<sub>pp</sub> amplitude and 10 kHz frequency. Set the cutoff frequency of the LPF  $f_c$  to 50 kHz and observe the output on the oscilloscope. Now decrease duty cycle to 25% (which halves the pulse duration), double the amplitude (to keep the area under the pulse the same – explain why this is required) and observe the output again, compare it to the previous one. Has it changed significantly? Keep decreasing  $d$  to 10% (increasing the amplitude to keep

the area the same) and observing the output. If required, decrease  $d$  further until you stop seeing significant changes in the output (minor variations are OK). At this point, calculate the filter impulse response via the measured filter output (recall our discussion in the class on the relationship between the filter output and impulse response when the input is a narrow rectangular pulse approximating delta function). The corresponding input pulse serves as a good approximation the delta-function for this particular LPF. Note the pulse duration and relate it to the LPF bandwidth, i.e. given the LPF bandwidth, what rectangular pulse is short enough to approximate the delta function?

2. Compare the measured impulse response in step 1 to the impulse response of the ideal LPF. What are the similarities/differences? Measure the null-to-null time of the measured impulse response and compare it that of the ideal LPF. This measurement is better to be done between 1<sup>st</sup> and 2<sup>nd</sup> nulls after the peak value. Relate this quantity to the filter's bandwidth.

3. Change the type of the filter to Bessel (bES) and repeat steps 1-2. Compare its impulse response to that in steps 1-2. What are the differences/similarities? How can you explain this based on the frequency responses you have measured in Part II?