Lab 1

5ETA0 - Intro Telecommunications

February 11, 2021

NOTE: Functions `calculateSpectrum` and `calculateSpectrumdB` are not functions of MATLAB - they were constructed specifically for this course, and should be in your MATLAB path when you're solving exercises (by putting them in the same folder as your executable).

Exercise 1: Sampling theorem

1. In MATLAB, plot the signal $\omega(t) = \cos(2\pi 50 \cdot 10^3 \cdot t + \frac{\pi}{3})$ at sampling frequencies $f_s = B$, $f_s = 2B$ and $f_s = 4B$, for 20 periods of the signal. For each f_s, plot the time domain representations as points (use `stem(x,y)` or `plot(x, y, 'o')`).
2. Calculate the spectra of the sampled signals using the function `calculateSpectrumdB` (see Appendix). Plot these frequency spectra for each f_s.
   (a) What can you conclude from your results for questions 1) and 2)? What phenomena do you see and what theorem can associate with these figures?
3. What f_s should you choose to get a 25kHz signal?
   HINT: The Fourier transform integrates over time - the more periods of the sine wave you use to calculate the spectrum, the more accurate the spectrum will be.

Exercise 2: Signal reconstruction

Download, install and open in MATLAB the `appchapter_2_7`, the instructions on how to do this can be found on canvas.

1. Upload the frequency sweep by clicking on the upload input button and look at the input signal. If you sample and reconstruct the signal with the pre-set variables, an incorrect signal is obtained. What is the lowest sampling frequency f_s you have to choose to reconstruct the signal correctly?
2. Choose this f_s with the slider (it might not give you the exact f_s you calculated) and press sample and reconstruct. Has the signal been reconstructed properly? Why or why not? (HINT: Zoom in to inspect the reconstructing signal)
3. What T should you choose for a correct reconstruction of the signal?
4. Can the 500 Hz sinusoidal signal be reconstructed with the same sampling frequency that yielded a correctly sampled sweep as found above?

Exercise 3: Idealized sampling

Load the waveform titled ”D-LORENTZ” into your MATLAB workspace and plot its spectrum using the `calculateSpectrumdB` function.

1. Find the bandwidth of the D-LORENTZ signal using its spectrum plot.Use $f_s = 10^6$.
2. As shown in Figure 1, the D-LORENTZ signal has been sampled with a pulsetrain with $\tau = \frac{1}{6}T_s$, where $T_s = 30\mu s$. The resulting signal is ”WAVE_SAMP”. Open the MATLAB script `Lab1_ex3`, where the multiplication with the pulsetrains to get ”WAVE_SAMP” has been done for you. Plot its spectrum with `calculateSpectrumdB` in MATLAB.
   What phenomena can you see happening in the spectrum? Why? Can the signal be recreated without loss of information?

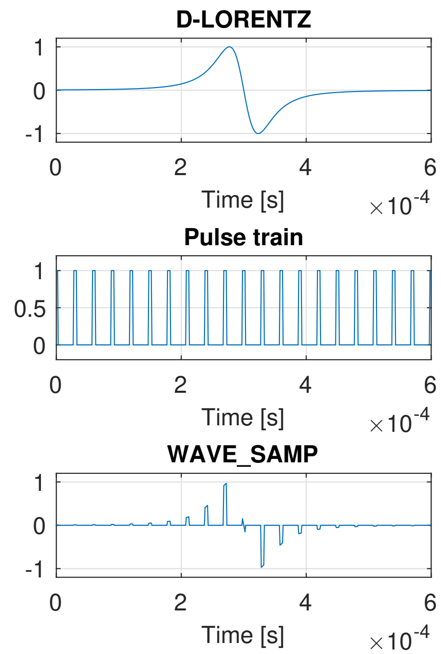

Figure 1: Waveform ”D-LORENTZ” sampled with a pulse train

3. Two pulse trains have been made for you in the MATLAB script `Lab1_ex3` (Figure 2):

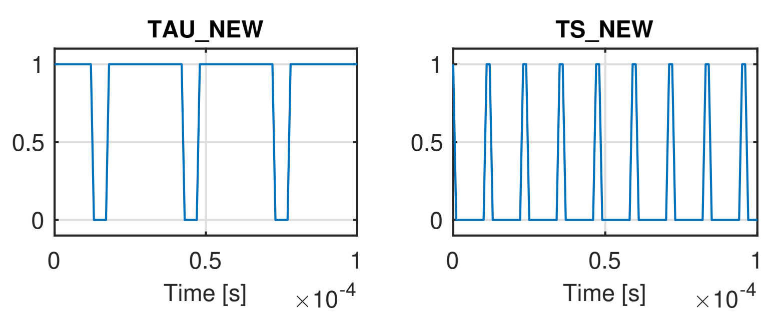

Figure 2: New pulse trains

(a) "TAU_NEW", with $\tau_1 = \frac{5}{6}T_{s1}$ with $T_{s1} = 30\mu s$

(b) "TS_NEW", with $\tau_2 = \frac{1}{6}T_{s2}$, with $T_{s2} = 12\mu s$

Which one of these can be used to make the received signal more accurate? Why?
Choose this pulsetrain and multiply it with the D-Lorentz waveform. Plot its time and frequency domain spectra, the latter using `calculateSpectrumdB`, what do you see?

4. What sampling technique is used in this exercise, and what are its downsides?

Exercise 4: Quantizing

Download, install and open in MATLAB the `appChapter_3_1_quantizing_noise`, the instructions on how to do this can be found on canvas.
This app displays how a PCM transmitter (analog-to-digital conversion) works.

1. Upload the analog signal.
2. Plot the flat-top PAM signal.
   (a) How does the spectrum change?
   (b) In lectures, you saw how the spectrum changes when sampled with flat-top PAM where $\tau < T_s$. In this exercise, $\tau = T_s$. How does this influence the spectrum?
3. Plot the quantized PAM signal.
   (a) How many bits do you need at the least to represent 32 levels without redundancy?
   (b) How does the SNR change with increasing levels?
   (c) According to theory, in this situation, what should the SNR be at M levels (in dB)?
   (d) Does the displayed SNR comply with theory? If no, can you think of reasons why?
4. Now, the signal needs to go through an encoder. Plot the PCM signal.
   (a) What is the pre-set bit rate R? Does it change with increasing levels?
   (b) What is the required bandwith for this PCM signal, B_{PCM}?
   (c) What R should you choose so that $T_b = 0.0001$ s?
   (d) If $T_b = 0.0001$ s, what is the symbol time T_s of a 32 level signal?

Appendix

MATLAB function	**Description**
`t`	Vector representation of time
`[f, magdB] = calculateSpectrumdB(x, fs)`	Calculates frequency spectrum of signal `x` using sampling frequency `fs`. Outputs frequency vector `f` and spectrum magnitude vector `magdB` in dB.
`[f, mag] = calculateSpectrum(x, fs)`	Calculates frequency spectrum of signal `x` using sampling frequency `fs`. Outputs frequency vector `f` and spectrum magnitude vector `mag`.
`Y = sin(X)`	Returns the sine of the elements of `X`