TELECOMMUNICATION SYSTEMS AND SERVICES

Session 4:

Transmission Impairments

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This session is designed to enhance knowledge from lesson 5: transmission impairments. In particular, distortion (linear and nonlinear) and noise (Additive White Gaussian Noise) in telecommunications systems will be studied in the laboratory.

1. Linear distortion

Linear distortion appears when the transfer function does not behave as an ideal channel (constant attenuation propagation and constant speed for all frequencies). In this case, the signal is distorted in its waveform. As any signal can be decomposed into a weighted sum of pure tones (sine or cosines), in order to see the effects of linear distortion, a sum of several sinusoids will be implemented.

Being:

$$
\begin{aligned}
x_1(t) &= v_1 \cos(2\pi f_1 t) \\
x_2(t) &= v_2 \cos(2\pi f_2 t) \\
x_3(t) &= v_3 \cos(2\pi f_3 t) \\
x &= x_1 + x_2 + x_3
\end{aligned}
$$

And transmitted by a non ideal channel, the resulting signal will not be a delayed and scaled version of the original signal (ideal channel). By means of the decomposition of a signal into sinusoids, and by applying its properties regarding invariant linear system, the output signal can be computed as:

$$
\begin{aligned}
y_1(t) &= v_1 \left|H(f_1)\right| \cos\left(2\pi f_1 t + \phi(f_1)\right) \\
y_2(t) &= v_2 \left|H(f_2)\right| \cos\left(2\pi f_2 t + \phi(f_2)\right) \\
y_3(t) &= v_3 \left|H(f_3)\right| \cos\left(2\pi f_3 t + \phi(f_3)\right) \\
y &= y_1 + y_2 + y_3
\end{aligned}
$$

Not being fulfilled $y = x(t) * h(t) \neq Kx(t-\phi)$, if h(t) is not an ideal channel.

For the forthcoming exercises, represent signals in the range [0, *T*) seconds for:

- T=2
- V_1=1.2, V_2= -1/3, V_3=1/5
- f_1=2 Hz, f_2=3f_1, f_3=5f_1.
- Sampling frequency F_s = 1000 samples/s, ***æt*** =1/F_s.
- Remember that you can create temporary axis as (1: ***æt*** : *T*)

1.1 Exercise 1: amplitude distortion.

Write a MATLAB program to show two figures:

1. The three component signals (using red, green and blue), and the signal resulting from the sum of all of the (in black) [signal *x (t)*]
2. One figure with the same signals after modifying the second and third component the voltage in 0.6 and 0.4 respectively (x_1 and x_3). The first signal (x_1) is not modified (x_1).

Draw the distorted signal.

1.2 Exercise 2: phase distortion.

Write a MATLAB program to show two figures:

1. One figure with the three component signals (using red, green and blue), and the signal resulting from the sum of all of the (in black) [signal *x (t)*]
2. One figure with the same signals after modifying the second and the third component the phase delay (- ū *(f)*, 0 radians) in 3´ / 4 and 5´ / 4 respectively.

Draw the distorted signal.

1.3 Exercise 3: amplitude and phase distortion.

Combine in a MATLAB code both distortions from 1.1 and 1.2, i.e. amplitude and phase distortions.

Draw the distorted signal.

1.4 Exercise 4: amplitude and phase distortion in a real channel

In this exercise we are going to represent the transmission channel by:

$$H(f) = K_0 e^{-j2\pi f t_0} + K_1 e^{-j2\pi f t_1}$$

Represents the transfer function of the channel in the frequency range from 0 to 10 Hz (using, for example, a step of 0.1 Hz).

Remember that the frequency axis can be express as (0:1/*T*:1/**œt**), and Fourier transformed is computed in Matlab as as fft (x). It is recommended to limit the abscissa to 100 Hz in order to better display the results (xlim([0 100]).

Calculate the phase and amplitude distortions for each components for K_0=1 K_1=0.3, t_0=1 s, t_1=1.7 s. Also, represent the total output signal with and without distortion in time domain. To do that, you have to multiply each component by the absolute value of H (f) in an appropriate frequency (for example for x_1 you must v_1 by the absolute value of H (f_1) and applying a phase delay in each component (for example, for x_1 there is also a phase delay equal to the angle of H(f_1)).

2. *Nonlinear distortion*

Nonlinear distortion modifies the waveform of the input signal due to the generation of new tones due to nonlinear behavior of the transmission system (usually by amplifiers working in the area of quasi-linearity). These tones can harmonics or intermodulation products (for two or more components tones). Being:

$$x_1(t) = v_1 \cos(2\pi f_1 t)$$
$$x_2(t) = v_2 \cos(2\pi f_2 t)$$

and

$$y(t) = k_1 x(t) + k_2 x^2(t) + k_3 x^3(t),$$

being k_1 = 10, k_2 = 4.3 y k_3 = -5.

For the flowing exercises, represent signals in the range [0, *T)* seconds, with T = 0.4 s, in intervals of t = 0.001 sec.

2.1 Exercise 5: Distortion of a tone

Being $x(t) = x_1(t)$, and V_1 = 1.5, f_1 = 20.

Write a MATLAB program to represent a figure with four graphs:

1. *x (t)*
2 The absolute value of *X (f). [Even if it is a continuous signal, it is recommended to use* **stem** for a proper *"visualization" when "counting" the harmonics]*
3. *y (t)*
4. Absolute value of *Y (f) [Use once again the command* ***stem].***

What differences can be seen between the output signal and the original one? What type of components are (harmonics or intermodulation products) do you recognize?

2.2 Exercise 6: Distortion of Two-tones

Being $x(t) = x_1(t) + x_2(t)$, and V_1 = 1.3, f_1 = 5 Hz y V_2 = 1/2, $f_2 = 5\, f_1$.

Write a MATLAB program to present a figure four graphic:

1. Input signal with the two component signals (red and green) and the resulting signal from the sum (in black)
2. Absolute value of *X (f).*
3. *y (t)*
4. Absolute value of *Y (f).*

What differences can be seen between the output signal and the original one? What type of components are (harmonics or intermodulation products) do you recognize?

3 Noise

Besides distortion, in a real situation involved signals are contaminated by several noise sources that cause that the behavior is different from the ideal model. Noise is a random signal that is added to our signal, and it can be often be modeled as a stochastic process. The most important noise source is the thermal noise due to electronic components. It is modeled as white Gaussian noise. The quality of the received signal depend greatly on the signal to noise ratio (S/N or SNR) between the desired signal and the random noise is noise. One way to decrease noise power generated is through a filter with a bandwidth equal to (or slightly higher if not ideal) the signal.

White noise is characterized by a constant power spectral density η. The power of white noise from the power spectral density is calculated as

$$P_n = \frac{\eta}{2}\int_{-\infty}^{\infty} df$$

That is, the noise power is theoretically infinite since the spectrum is infinite. Therefore, in practice we consider the output power of the system (which acts as a filter). For Matlab simulations, the spectrum is limited, with equal spaced samples at $\Delta t=1/F_s$, being represented only in the range [- F_s / 2, F_s / 2). Therefore, we can calculate the noise power simulation considering itself as a low-pass filter with bandwidth F_s / 2:

$$P_n = \frac{\eta}{2}\int_{-\infty}^{\infty} |H(f)|\,df = \frac{\eta}{2}\int_{-Fs/2}^{Fs/2} df = \frac{\eta}{2}F_s$$

3 1 Exercise 7: Adding white Gaussian noise

In this last exercise we will use the same values of T and F_s, as well as $x_1(t)$ and $x_2(t)$. That is: T=0.4, F_s= 1000,

$$x_1(t) = v_1 \cos(2\pi f_1 t)$$
$$x_2(t) = v_2 \cos(2\pi f_2 t)$$

Write a MATLAB program to analyze the effect of noise and to represent a figure with four graphs:

1. Input signal A n to the input signal $x(t) = x_1(t) + x_2(t)$ with V_1=1.2, V_2=1.5, f_1=5 and $f_2=5f_1$;
2. *X (f)*.
3. A third signal $y(t) = x(t)+n(t)$. Use **n=tco_wgn(size(x,1),size(x,2),eta,Fs)** where **eta** is the power spectral density of noise η. In a first implementation, use **eta** = 0. 1.
4. *Y (f)*

Calculate the power spectral density for the following values of noise power (Eta = 0.01 and 1).

Run the program for the previous values and discuss the results that can be seen in the graphs. How noise affects the waveform and the power of the desired signal?