# ELEC6021 — COMMS Assignment I

- Complete the set of simulations as indicated in Assignment I
- Hand in your assignment to the ECS Student Services (no longer called Course Office) on ———
- Mark contribution to the unit: 25% (Total 25 marks)
- A few "hints" are given in this lecture
- Assignment and this lecture notes can be downloaded from

  http://www.ecs.soton.ac.uk/∼sqc/EZ619/

- There will be no lecture for 2nd lecture time slot – use it for you to do this assignment





# Simulation of Carrier Communication Systems

- In Assignment, we are going to simulate carrier communication systems
- A simple way of representing a continuous-time signal is to use a discrete-time signal obtained by fast sampling
- Thus for carrier of $f_c$ we may need a sampling rate, say, approximately $10 \times f_c$, but in real world carrier frequencies are in GHz
- What is the carrier frequency we will be used in our simulation? Why could we use a more realistic carrier, say 1 GHz, in simulation?
- Since we use sampled discrete-time signal to represent continuous-time signal, we must first know it is OK to do so
- Intuitively, if we sample very fast the sampled signal will look like the original continuous-time signal, but of course it is more than that





# Normalisation

- Many Matlab functions such as those for filter design prefer normalised frequency parameters, and such normalisations are with respect to the sampling rate
- Therefore, if $f$ and $f_\mathrm{s}$ are the absolute frequency and absolute sampling rate, the $f_\mathrm{norm}$ and $f_\mathrm{s,norm}$ are the relative, normalised parameters, which have to fulfill

$$\frac{f}{f_\mathrm{s}} = \frac{f_\mathrm{norm}}{f_\mathrm{s,norm}} \tag{1}$$

  Therefore

$$f_\mathrm{norm} = f_\mathrm{s,norm} \cdot \frac{f}{f_\mathrm{s}} = 2 \cdot \frac{f}{f_\mathrm{s}} \tag{2}$$

  whereby $f_\mathrm{s,norm} = 2$ has been inserted as normalised sampling rate (see e.g. `help fir1` or `help psd` to check this)

- What this means: Frequency $f = \frac{f_s}{2}$ is normalised as $f_\mathrm{norm} = 1$



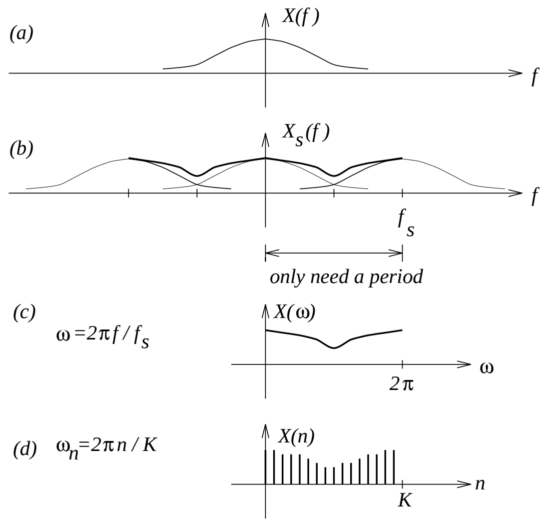


# What Happens When Sampling

- (a) the true spectrum $X(f)$ of a continuous-time signal $x(t)$, and (b) the spectrum $X(f)$ of the sampled signal $x[k] = x_s(t)$ at a sampling rate $f_s$

  $X_s(f)$ is periodic with a period $f_s$, and we only need a period of $X_s(f)$ to represent it. Also distortion occurred around $\frac{f_s}{2}$ (unless $X(f)$ is bandlimited with a bandwidth less than or equal to $\frac{f_s}{2}$)

- In fact we only need to show $X_s(f)$ from $f = 0$ to $f = \frac{f_s}{2}$



# Written Assignment

- No point to display the spectrum of a sampled signal beyond $f = \frac{f_s}{2}$ $\rightarrow$ those are "artifacts" due to sampling operation

- Only the spectrum within $f = 0$ to $f = \frac{f_s}{2}$ is relevant to the original spectrum of the continuous-time signal

  That is, the spectrum in frequency range 0 to 1 (recall $f = \frac{f_s}{2}$ corresponds to $f_{\rm norm} = 1$)

- Written assignment should be **concise** and contains a set of **clearly labelled figures**

  Do not forget **units**, unless for normalised quantities

- Overall system gain should be kept to unity, otherwise you may have "unexpected" results



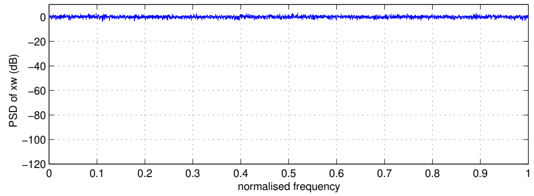


# Example of Written Assignment

## 1 AM/FM Analogue Modulation

### 1.1 AM Modulation

**a) Input Signal.** White noise xw with a flat power spectral density (PSD):




The required lowpass filter can be constructed according to $f_{\rm cutoff,norm} = 2 \cdot 10$ kHz/2 MHz $= 0.01$; the resulting PSD of the signal after filtering $x(t)$: