# Contents



# 1 Week 8 Module 6:

## 1.1 Digital Communication Systems

### 1.1.1 Introduction to digital communications

1. The success factors for digital communications

   (a) Power of the DSP paradigmw

   - integers are easy to **regnerate**
   - good phase control
   - adaptive algorithms

   (b) Algorithmic nature of DSP is a perfect match with information therory:

   - Image Coding: JPEG's entropy coding
   - Encoding of of accustic or video informatiion: CD's and DVD's error correction
   - Communication Systems: trellis-coded modulation and Vierbi coding

   (c) Hardware advancement

   - minituarization
   - general-purpose platforms
   - power efficiency

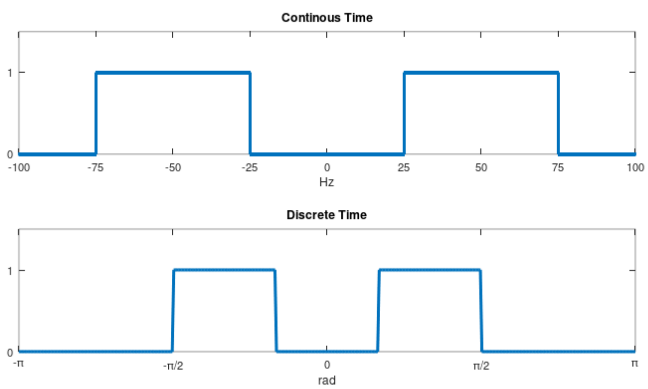
2. The analog channel constraints

   - unescapable "limits" of physical channels:
     - **Bandwith:** the signal that canbe send over a channel has a limited frequency band

    - **Power:** the signal has limited power over this band, e.g. due to power limit of the equipment
- Both constraints will affect the final capacity of the channel.
- The maximum amound of information that can be reliably delivered over a channel - bits per second -
- Bandwidth vs. capacity:
    - small sampling period $T_s \Rightarrow$ high capacity
- but the bandwidth signal grows as $\frac{1}{T_s} \Rightarrow \$ \ \Omega_{\mathrm{N}} = \pi_{\overline{T_s\$}}$

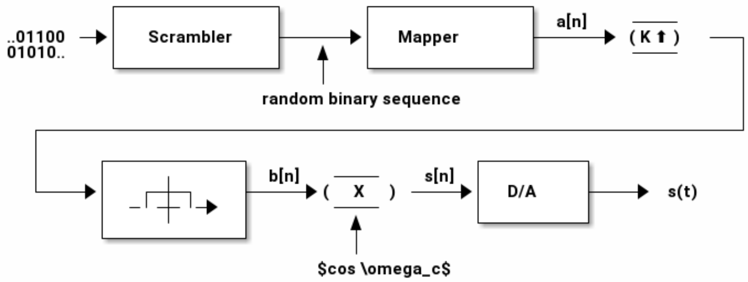
3. The design Problem

- We are going to adapt the all-digital paradigm
- Converting the specs to digital design




- with:
    - Sampling Frequency $F_S \geq 2f_{max}$
    - Continuous Time $F_s/2$: Nyquist frequency
    - Maximum Frequency: $\frac{F_s}{2} \Rightarrow \pi$
    - Bandwidth: $\omega_{min,max} 2\pi \frac{f_{min,max}}{F_s}$
- Transmitter design
    - convert a bitstream into a sequence of symbols a[n] via a mapper
    - model a[n] as white random sequence $\Rightarrow$ add a scrambler

- no we need to convert a[n] into a continuous-time signal within the constraints




If we assume that the data is randomized and therefore the symbol sequence is a white sequence, we know that the power spectral density is simply equal to the variance. And so the power of the signal will be constant over the entire frequency band. But we actually need to fit it into the small band here as specified by the bandwidth constraint. So how do we do this? Well, in order to do that, we need to introduce a new technique called upsampling, and we will see this in the next module.

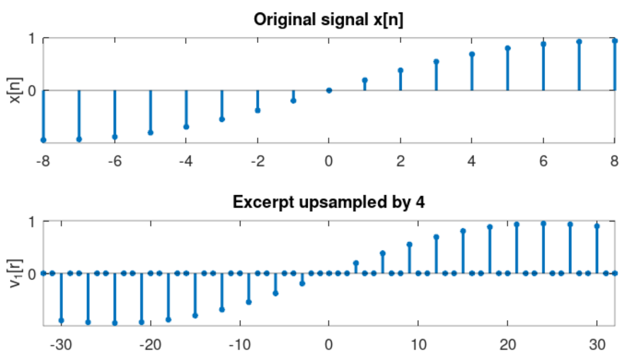
We are talking about digital communication systems and in this lesson we will talk about how to fulfill the bandwidth constraint. The way we are going to do this is by introducing an operation called upsampling and we will see how upsampling allows us to fit the spectrum generated by the transmitter onto the band allowed for by the channel.

### 1.1.2 Controlling the bandwidth

- Shaping the bandwidth Remember that our assumption is that the signal generated by the transmitter is a wide sequence and therefore its power spectral density will be full band. What we need to do is to shrink the support of its power spectral density so.
  - the answer is multirate techniques

1. Upsampling
   - Our Problem

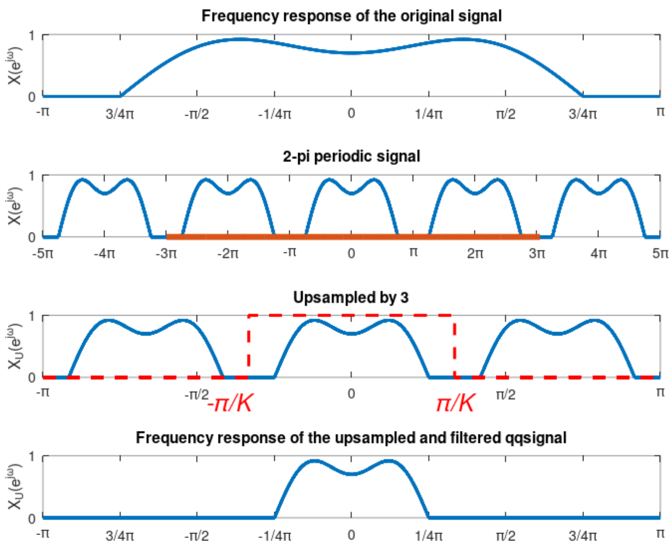
- bandwith constraint requires us to control the spectral support of a signal
  - we need to be able to shrink the support of a full-band signal
- Upsampling can be obtained by interpolating a discrete time sequence to get a continuous time signal. And resample the signal with a sampling period which is k-times smaller than the original interpolation sample.
- Or we do it entirly digitally.
  (a) we need to ”increase” the number of samples by k
  (b) obviously $x_U[m] = x[n] whenmmultipleofK$
  (c) for lack of better strategy, put zeros elsewhere
- Upsampling in the time domain




- Upsampling in the digital domain

$$\begin{aligned} X_U(e^{j\omega}) &= \sum_{m=-\infty}^{\infty} x_U[m]e^{-j\omega m} \text{ with } x_U = 0 \text{ if } mnK \\ &= \sum_{m=-\infty}^{\infty} x[n]e^{-j\omega nK} \\ &= X(e^{j\omega K}) \end{aligned}$$

This is simply a scaling of the frequency axis by a factor of K. Graphical interpretation: since we are multiplying the frequency axis by a factor of K, there will be a shrinkage of the frequency axis.




- $\frac{\pi}{K}$ : $FilterCut - OffFrequency$
- The bandwith of the signal was shrinked by factor K=3: from $\frac{3}{3}\pi$ to $\frac{1}{4}\pi$
- back in the time domain
  (a) insert K-2 zeros after every sample
  (b) ideal lowpass filtering with $\omega_c => frac\pi K$
  x$^[n] = x_U(n) * sinc(n/K)$
  $= x_U[i]sinc\left(\frac{n-i}{K}\right)$
  $= x[m]sinc\left(\frac{n}{K} - m\right)$

### 1.1.3 Fitting the transmitter spectrum

The bandwith constrainth says that only frequencies between $F_{min}$ and $F_{max}$ are allowed. To translate it to the digital domain, follow the preceeding steps:

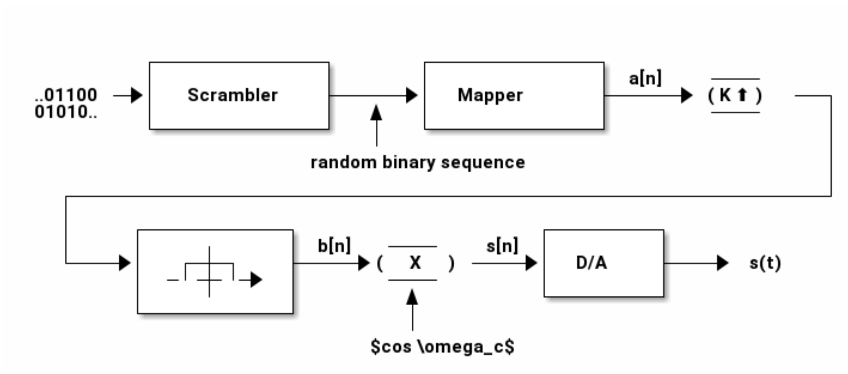
- let $W = F_{max} - F_{min}$
- pick $F_s$ so that:

- $F_s > 2F_{max}$
- $F_s = KW$, $k\mathbb{N}$

- $\omega_{max} - \omega_{min})2\pi\frac{W}{F_s} = \frac{2\pi}{K}$
- we can simply upsample by K

**Bandwith constrainth**

And so we can simply upsample the sample sequence by K, so that its bandwidth will move from 2pi to 2pi/K, and therefore, its width will fit on the band allowed for by the channel.




- Data Rates
  - upsampling does not change the data rate
  - we produce (and transmitt) W symbols per seconds
  - W is sometimes called the Baud rate of the system and is equal to the available bandwith.
- Raised Cosine

# 2 Week 8 Module 7:

## 2.1 Image Processing