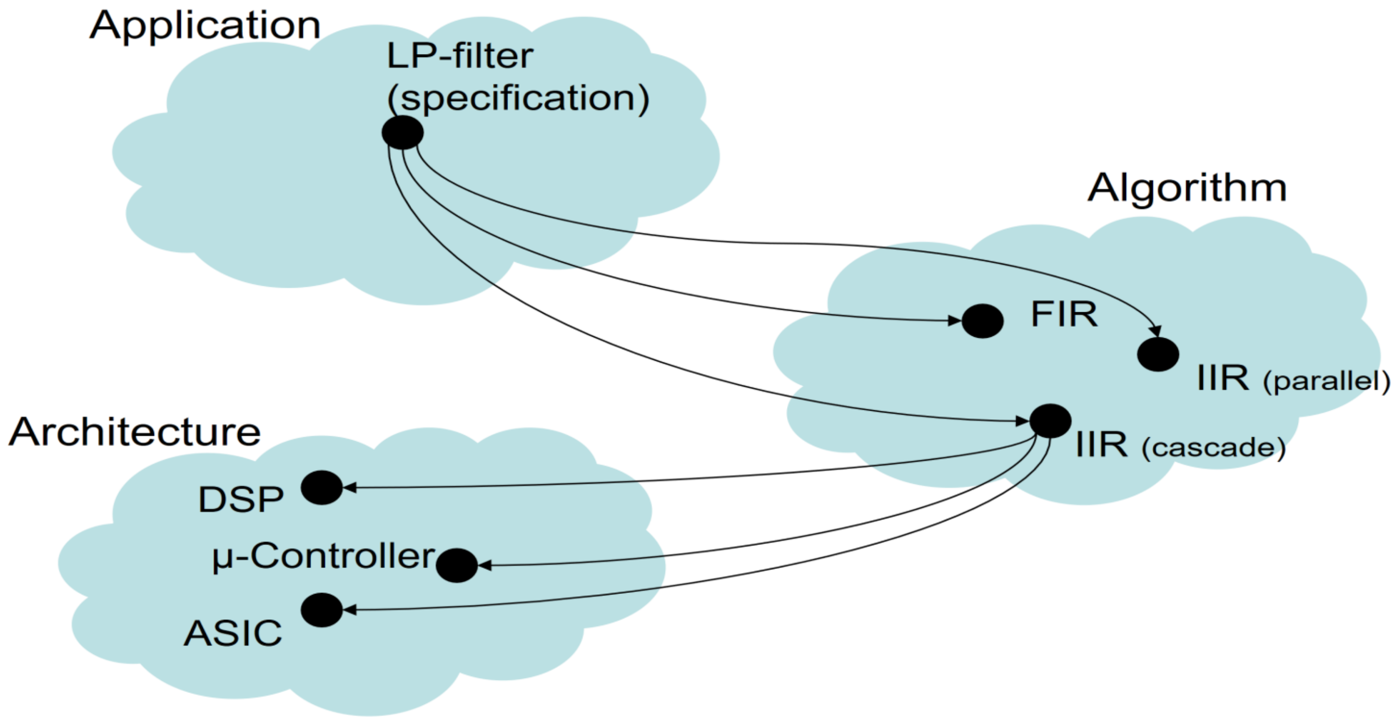
IEE 1711: Applied Signal Processing

Professor Muhammad Mahtab Alam (muhammad.alam@taltech.ee)
Lab Instructor: Julia Berdnikova

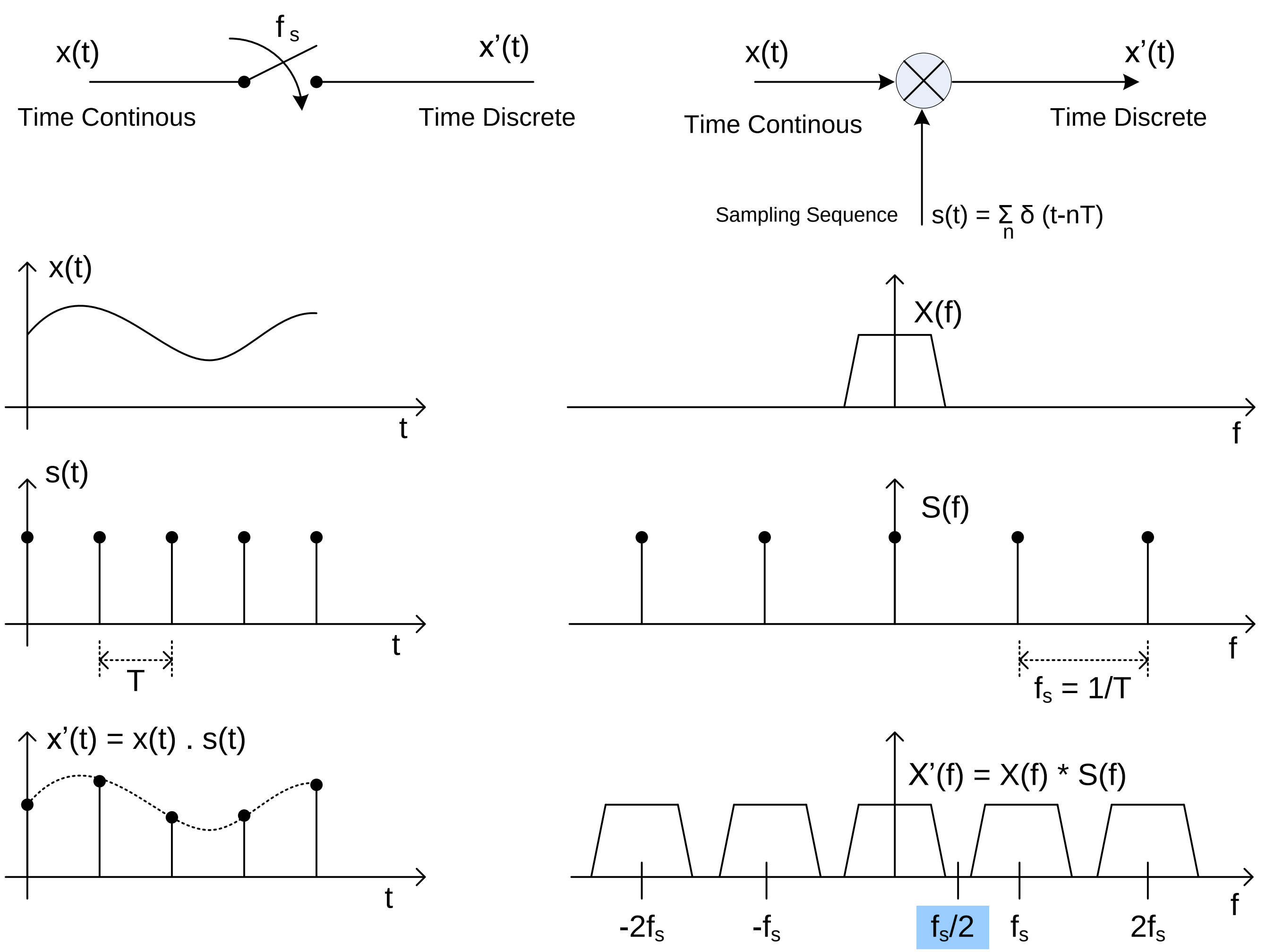


Outline

- Lecture 11: Software Defined Radio
  - Brief Followup

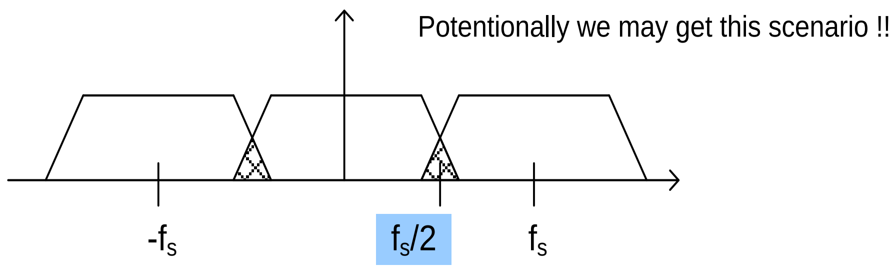
- Lecture 12: SDR Cont…
  - Sampling Principal (multirate sampling)
  - Multistandard Radio and Architectures
  - Bandpass Sampling
  - Channelizers
  - Summary

Sampling Principal 1

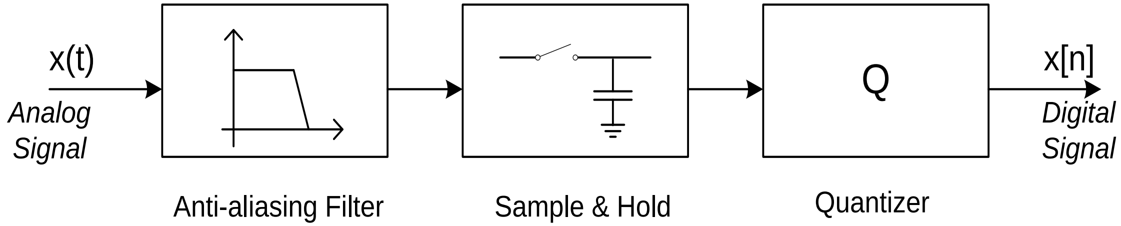


Sampling Principal 2

- Discrete Time ⇨ Periodic Frequency



- Anti-aliasing Filter $f_s > 2 f_{max}$



- Or choose the sampling frequency sufficiently high

Sampling Principal 3

Let’s assume we have a sequence x[n], and on this sequence we want to do re-sampling.

“Sampling of Sampled Signals”

- Pick out Samples → Down Sampling
- Insert new Samples → Up Sampling

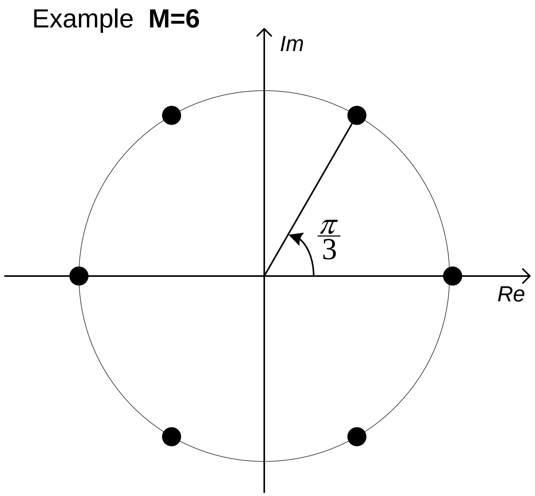
Sampling Sequence described by IDFT

$$\mathrm{S}_M[\mathrm{n}] = \frac{1}{M} \sum_{m=0}^{M-1} \mathrm{e}^{j\frac{2\pi}{M}mn}$$

Sampling Principal 4

Sum of M complex Sequences

$$e^{j\frac{2\pi}{M}mn} = \underbrace{\cos\frac{2\pi}{M}mn + j\sin\frac{2\pi}{M}mn}_{\text{points on the Unit Circle}}$$



n counts the period; n={0, M, 2M, ….}

m counts the individual points within a certain period

$$n = \gamma M \quad \text{where } \gamma \text{ is an integer}$$

Sampling Principal 5

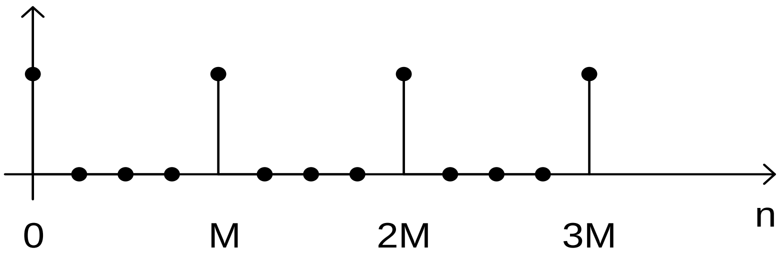
$$e^{j\frac{2\pi}{M}mn} \Rightarrow e^{j\frac{2\pi}{M}m\gamma M}$$

$$e^{j2\pi m\gamma} = 1 \quad \text{since } m \text{ and } \gamma \text{ are integers}$$

$$\mathrm{S}_M[\mathrm{n}] = \frac{1}{M}\sum_{m=0}^{M-1} \mathrm{e}^{j\frac{2\pi}{M}mn} = 1\Big|_{n=\gamma M}$$

$$\text{For } n \neq \gamma M \Rightarrow \sum_{m=0}^{M-1} = 0 \quad \text{(complex exponentials will cancel each other)}$$

$$\mathrm{S}_M[\mathrm{n}] = \begin{cases} 1 & \text{for } n = \gamma M \\ 0 & \text{otherwise} \end{cases}$$

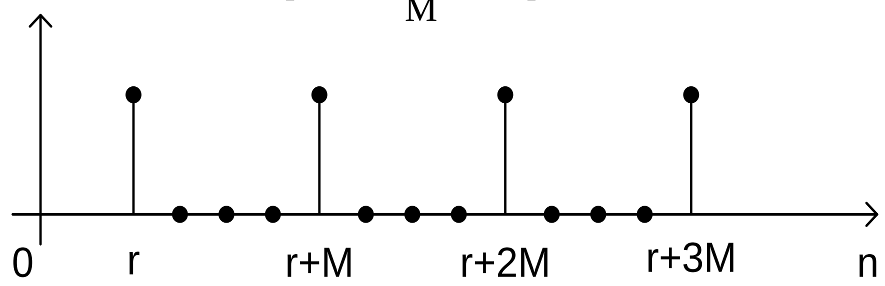


Sampling Principal 6

Sample Sequence off-set from zero

$$S_M[n-r] = \frac{1}{M}\sum_{m=0}^{M-1} e^{j\frac{2\pi}{M}m(n-r)}$$

$$= \underbrace{\frac{1}{M}\sum_{m=0}^{M-1} e^{-j\frac{2\pi}{M}mr}}_{\substack{\text{Complex sine sequence with} \\ \text{amplitude } \frac{1}{M} \text{ and phase } e^{-j\frac{2\pi}{M}mr}}} \cdot \underbrace{e^{j\frac{2\pi}{M}mn}}_{\text{Def. IDFT}}$$



Frequency domain representation of $S_M[n-r]$ is identical to frequency domain representation of $S_M[n]$ except for a phase shift

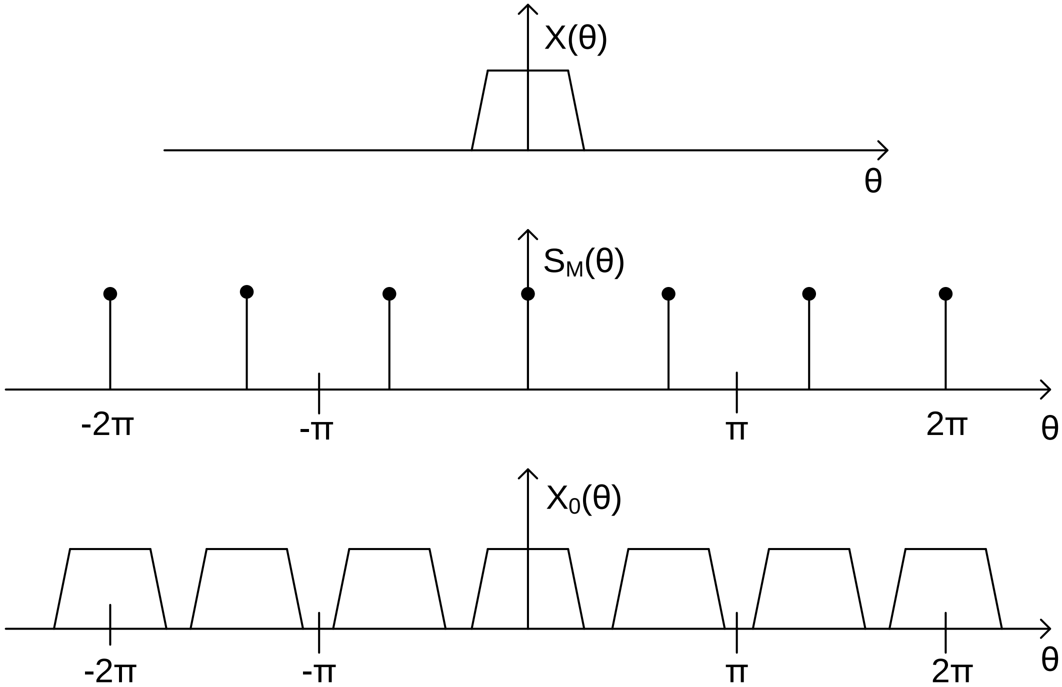
Sampling Principal 7

Re-sampling Operation

$$x[n].S_M[n] = x_0[Mn] \qquad \text{(0 indicates zero offset)}$$

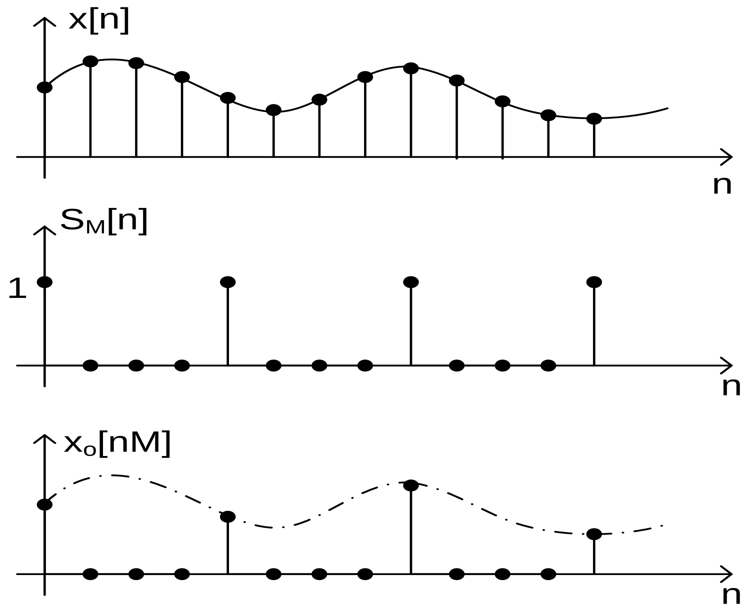
Frequency domain representation of re-sampled signal

$$X_0[\theta] = X[\theta] * S_M[\theta]$$



Re-sampling makes the spectrum periodic within θ ε [-π, π]

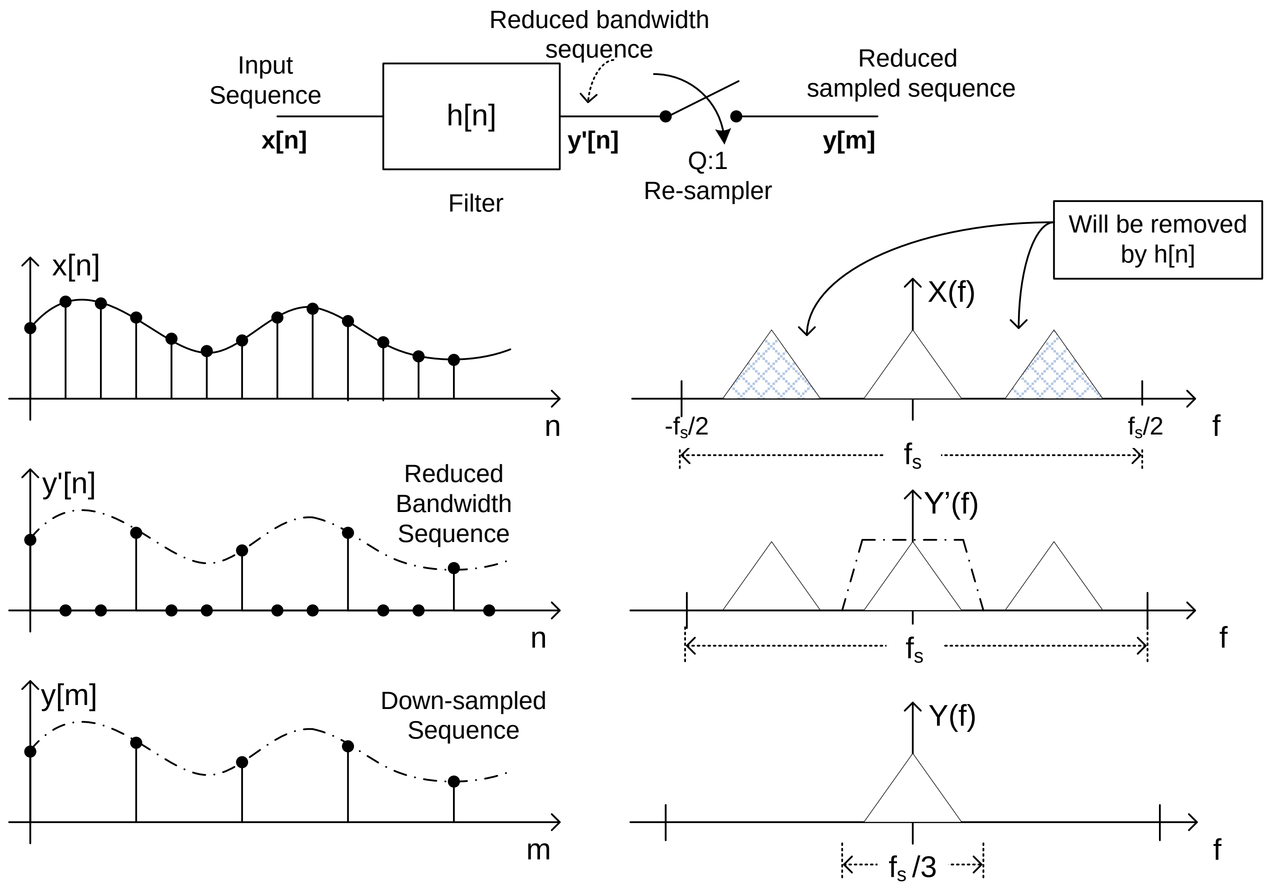
Time Domain



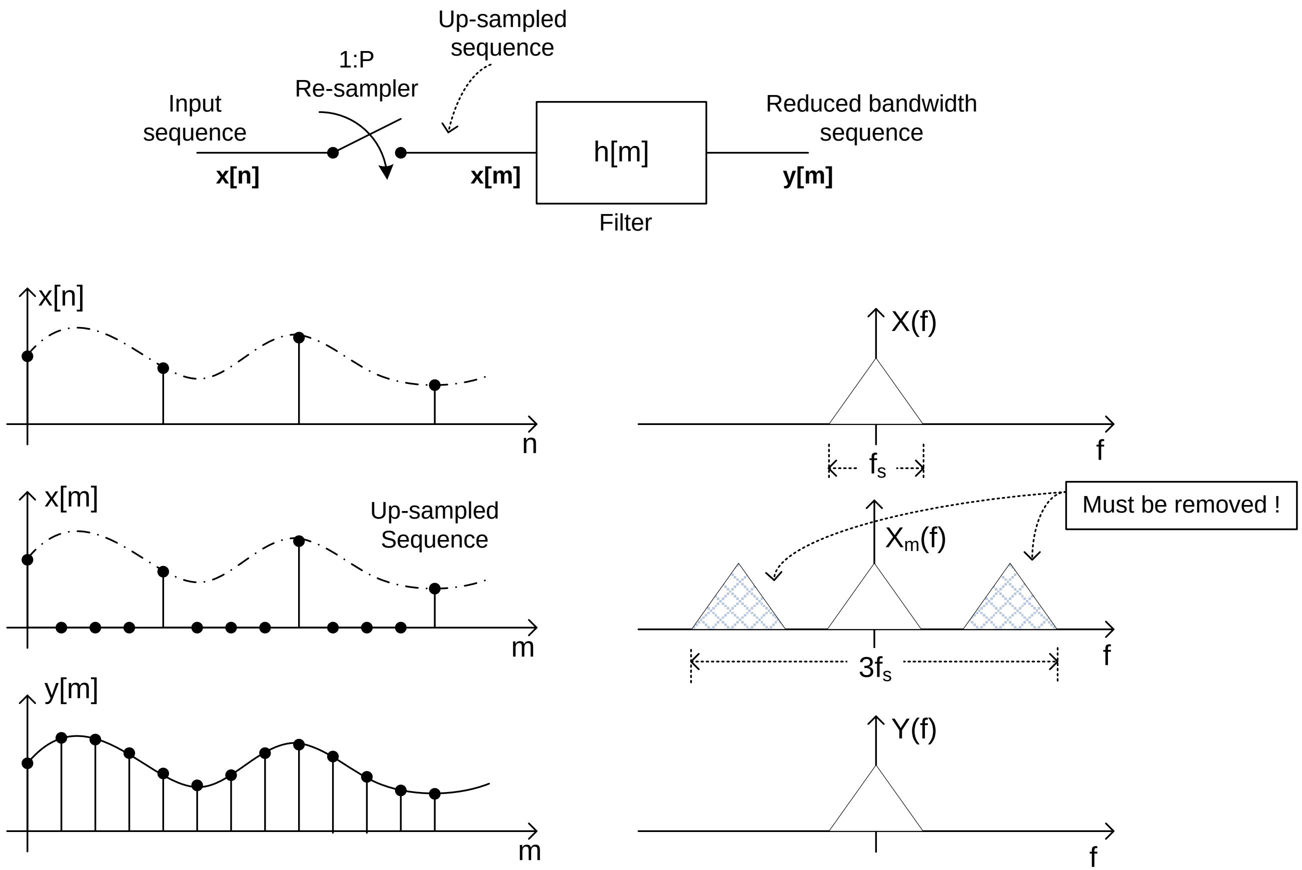
- $x_o[nM]$ contains the same number of the samples as x[n], therefore no sample rate change so far

In order to change the sample rate and at the same time get rid of the freq. domain periodicity, we introduce the concept of Multi-rate filters.

The Concept of Down-sampling



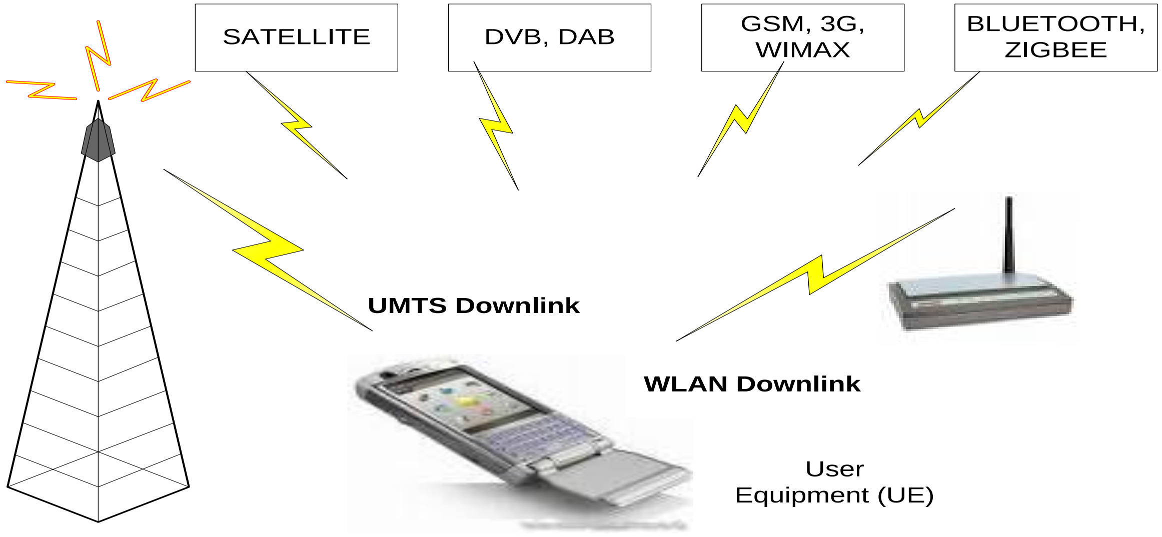
Similarly for Up-Sampling



h[m] performs bandwidth reduction by interpolating between the original samples

Multi-Standard Radio

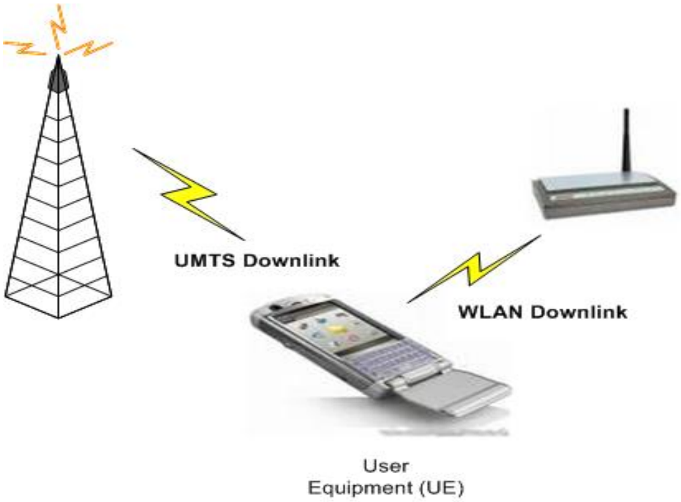
- A transceiver with multi-band and multiple standards is a front-end that can be operated in number of different frequency bands.
- The increasing trend toward a single device integrating several features and capabilities encourage the companies and research centers to develop the multi-standard, mutli-mode 'all-in-one' front end.



Multi-Standard Radio (UMTS & WLAN)

Limitations:

- Channels interference among the standards is very high
- Mobile Applications (UMTS,WLAN)
- Possible Scenario



Specifications:

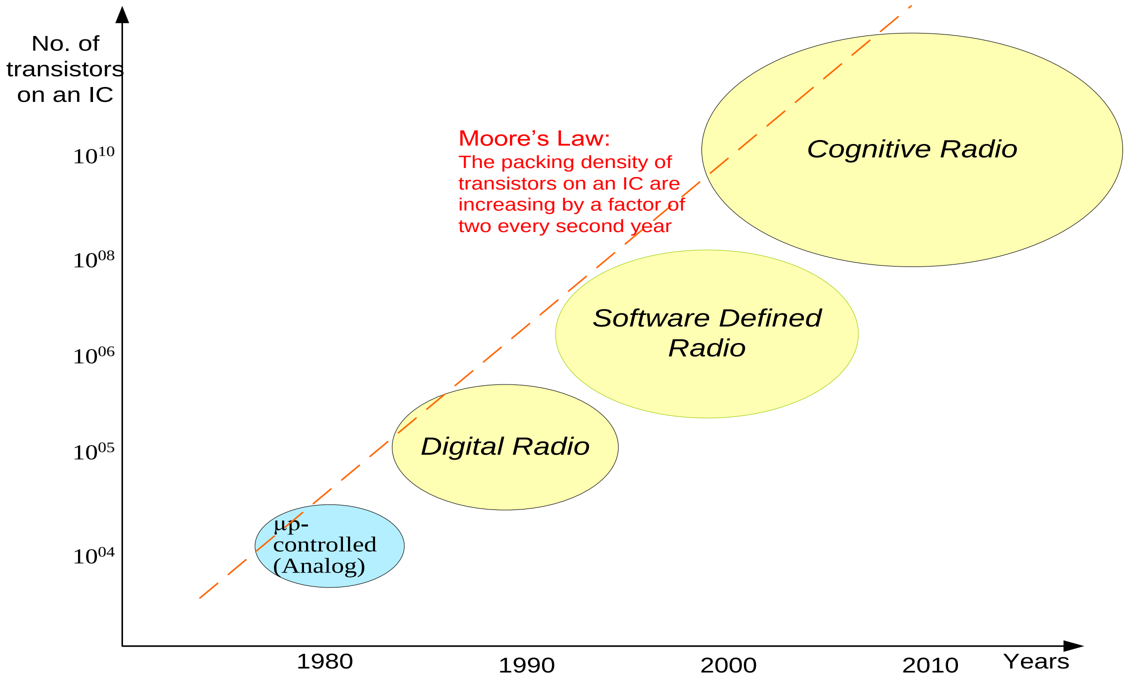
	UMTS	WLAN 802.11g
Frequency Bands	2110 - 2170 MHz: DL	2.4 - 2.4835 GHz
Receiver Sensitivity	-117 dBm	-82 to -65 dBm
Channel Bandwidth	3.84 MHz	16.6 MHz
Number of channels	12	3

Architecture

Moore's Law:

Number of Transistors are increasing by a factor of 2 after every second year.

Key Features: Reconfigurable, Programmable



Possible Choices:

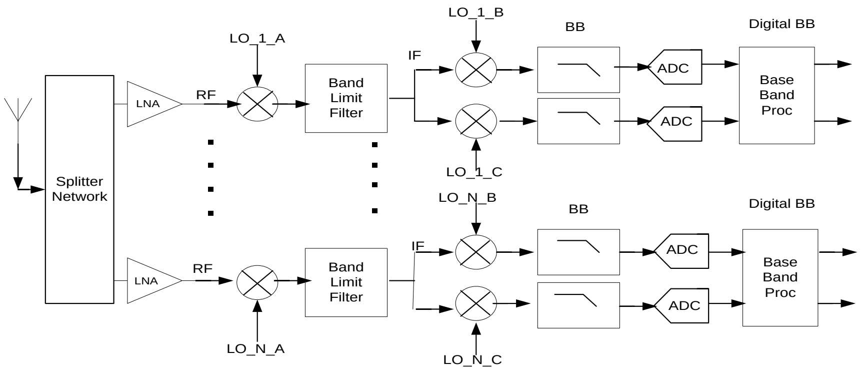
GPP, DSP, FPGA

Digital Radio (Receiver)

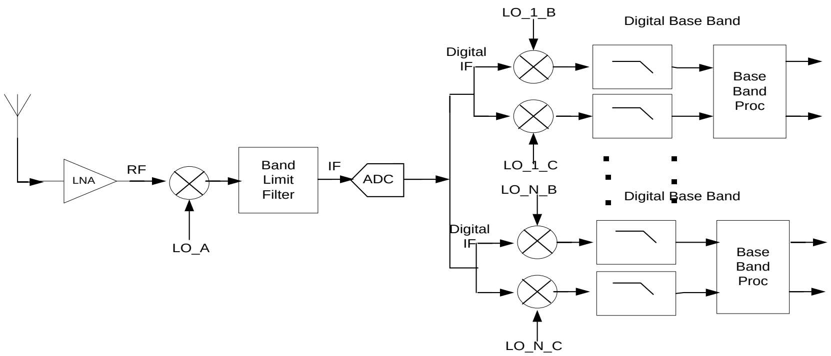
- The gain and phase imbalance between the two paths from the IF stage till the baseband in an N-channel Rx is the cause of crosstalk between the I and Q quadrature components

- The precision of coefficients used in the filtering process sets an upper bound to spectral artifacts levels at −5dB/bit, so the 12bitADC will have an image level below 60dB
    1. Spectral images are below the noise floor of ADC.
    2. Digital filters designed to have linear phase characteristic

First Generation



Second Generation



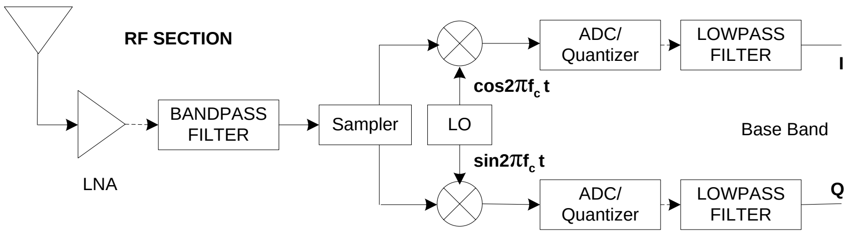
Down Conversion Techniques

- Direct Conversion architecture digitize the input signal at RF and down-convert directly to base band.

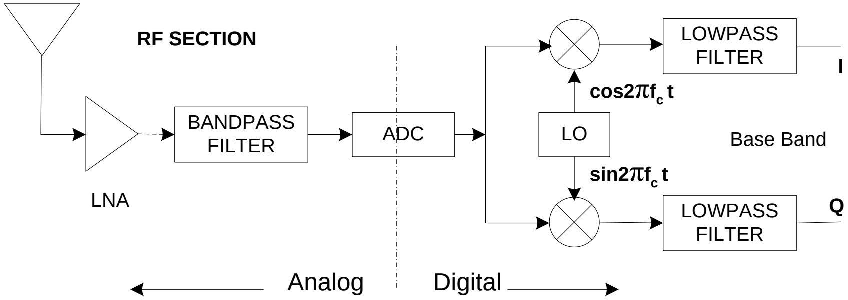
- Direct Conversion architecture requires an additional down-converter at IF before getting the signal at baseband.

- Band pass sampling architecture does not require additional circuits for down conversion prior to quantization.

Direct Conversion Architecture

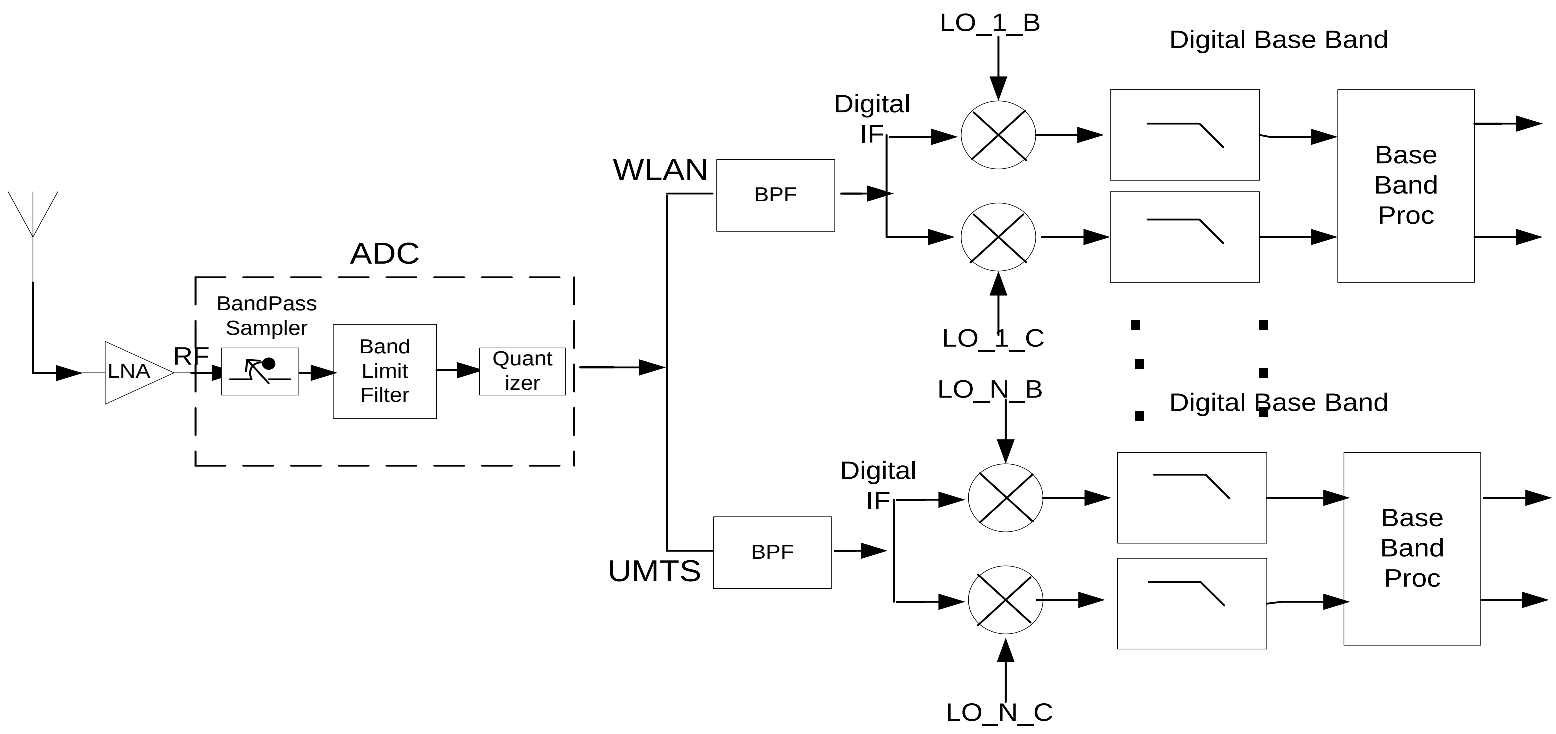


Bandpass Sampling Architecture



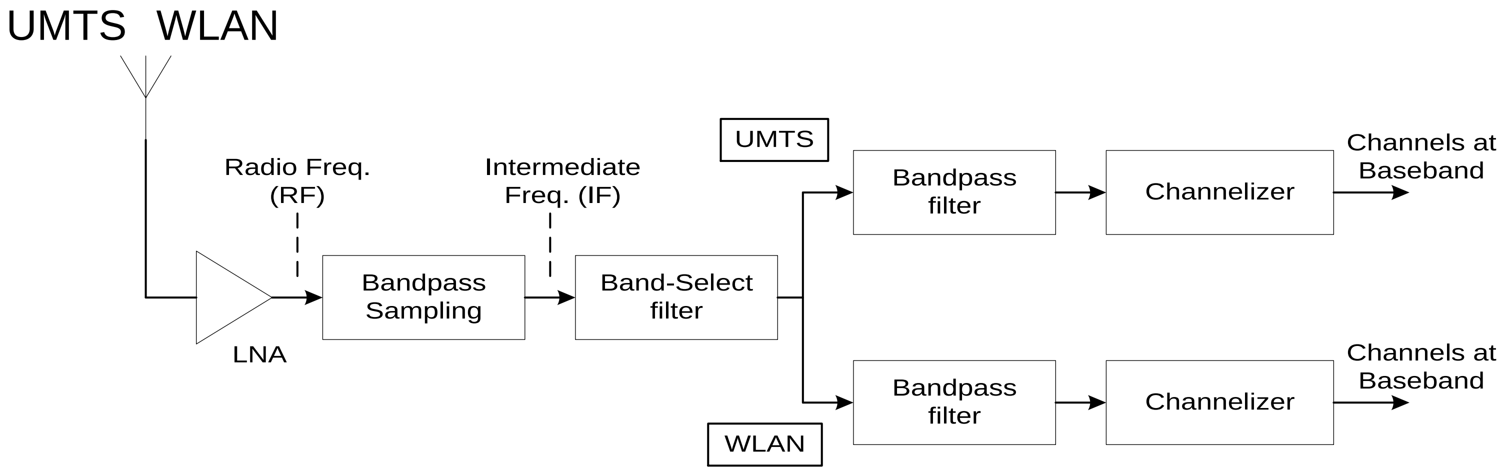
Proposed Software Radio Receiver Architecture

- Digital RF front end
- Digital IF (Decimation & Downconversion)

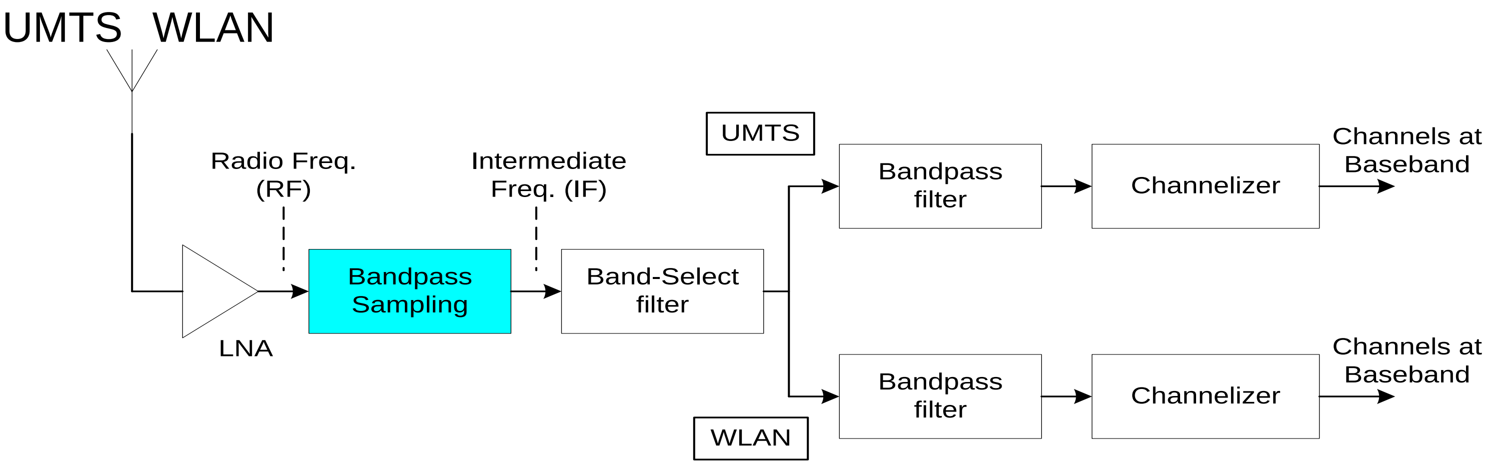


Multi-Standard Software Radio Receiver

System Block Diagram



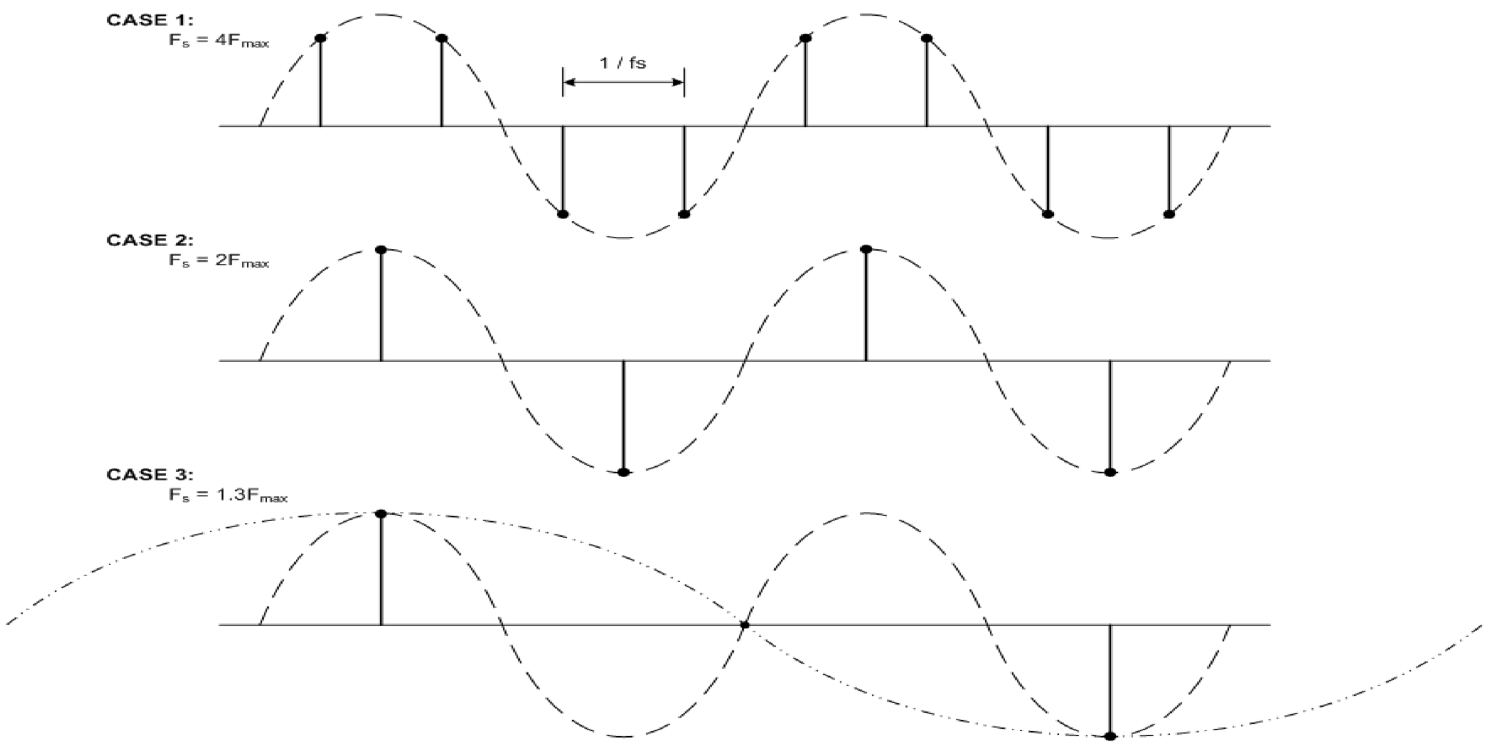
Bandpass Sampling



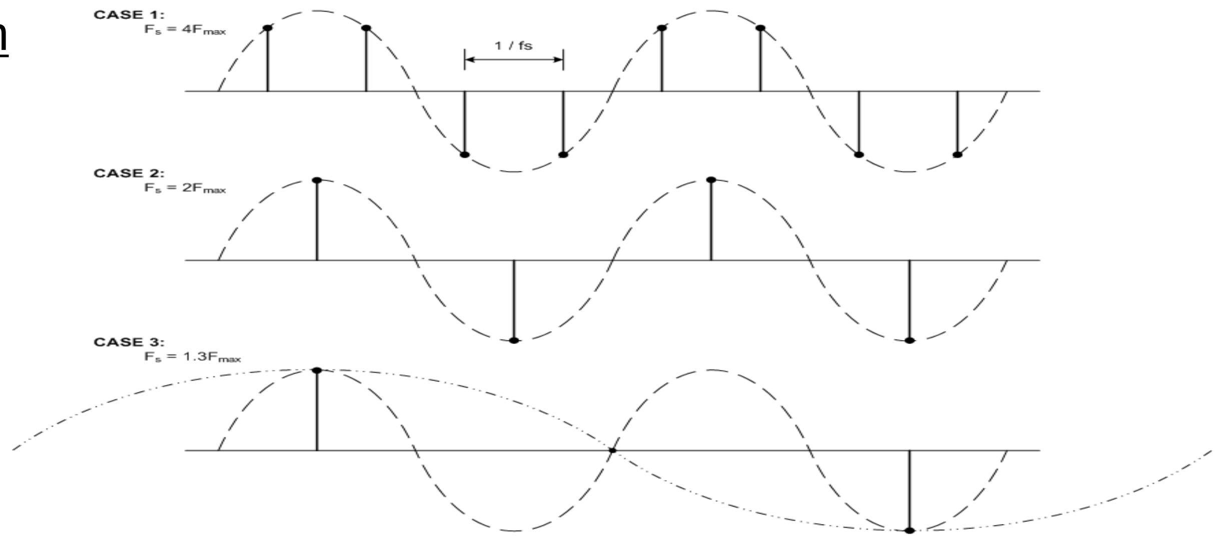
Sampling Theory

Nyquist Theorem:

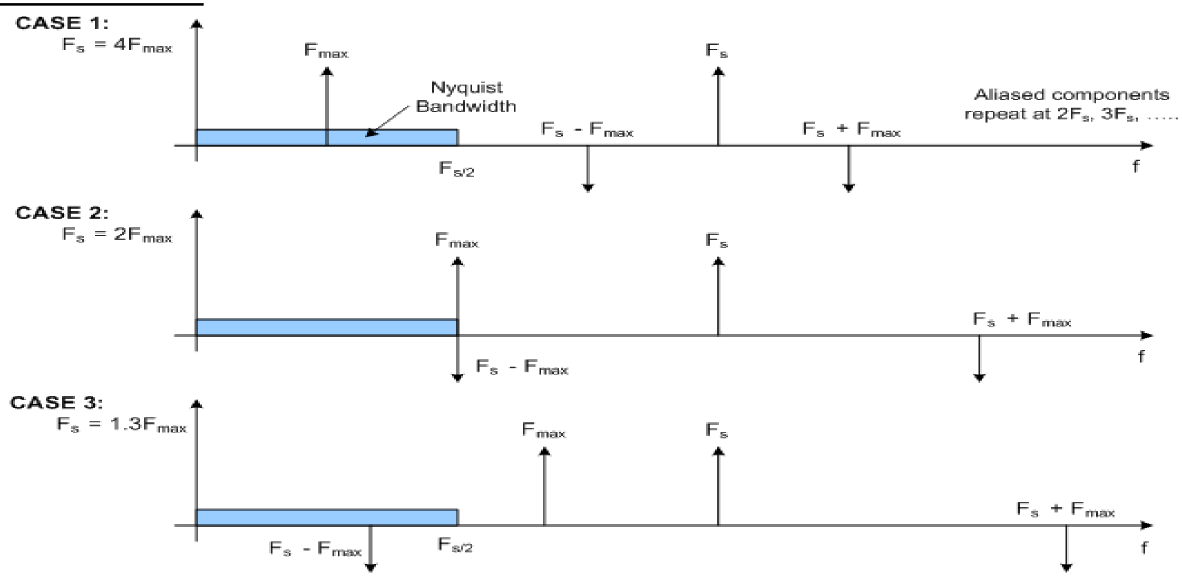
”A signal must be sample at a rate greater than twice of its maximum frequency to ensure un-ambiguous data”



Time Representation



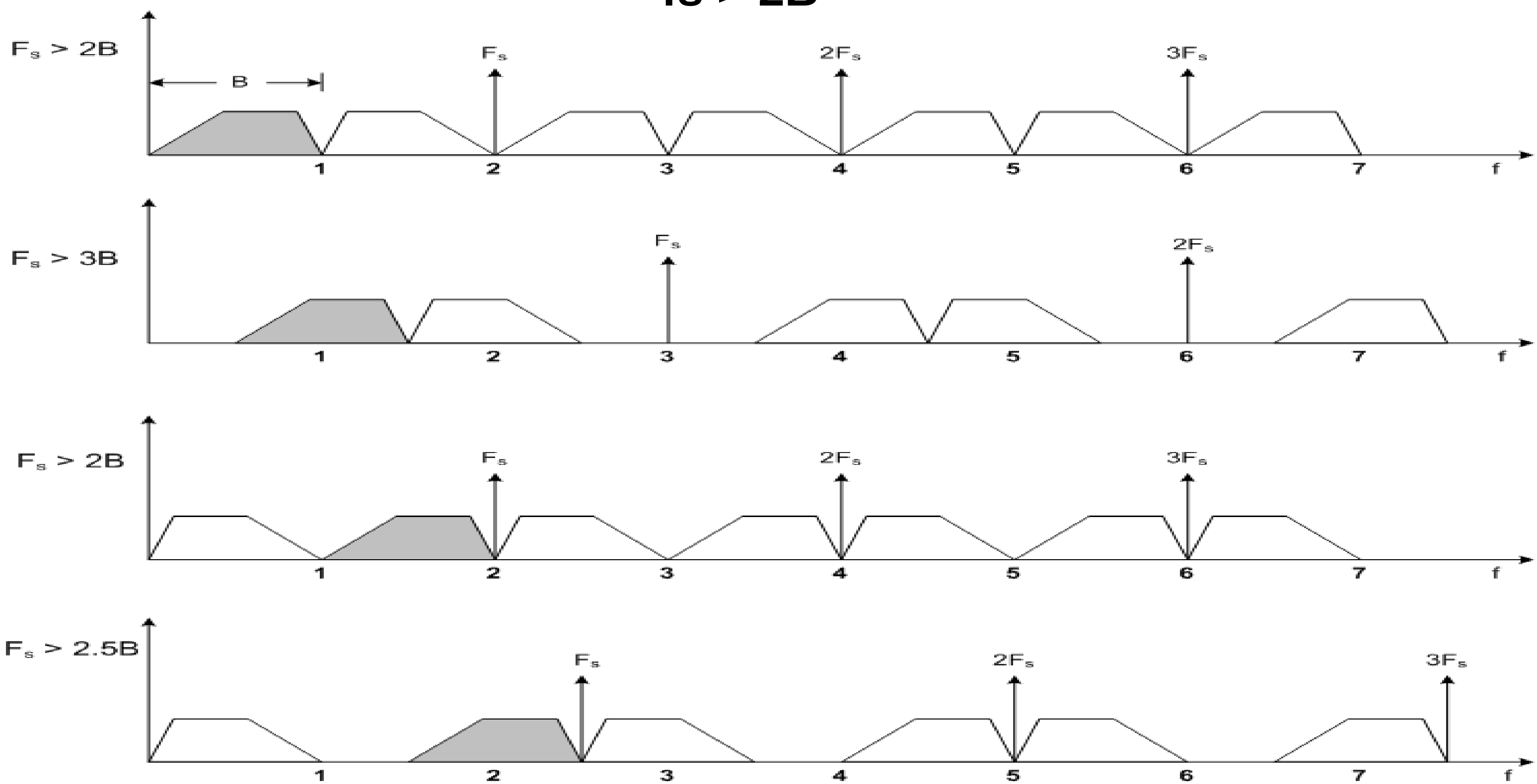
Frequency Representation



Bandpass Sampling Theory

Shannon's Theorem

fs > 2B



Frequency-domain representation of 1MHz signal lying at different spectrum positions, illustrating non-overlap aliases.

Bandpass Sampling Theory

Example:

Frequency-domain representation of 1MHz signal lying between 6 to 7 MHz in the spectrum, under-sampled at 2MSPS produces non-overlap aliases at multiples of sampling frequency. The original signal is aliased to baseband i.e. DC to 1MHz.