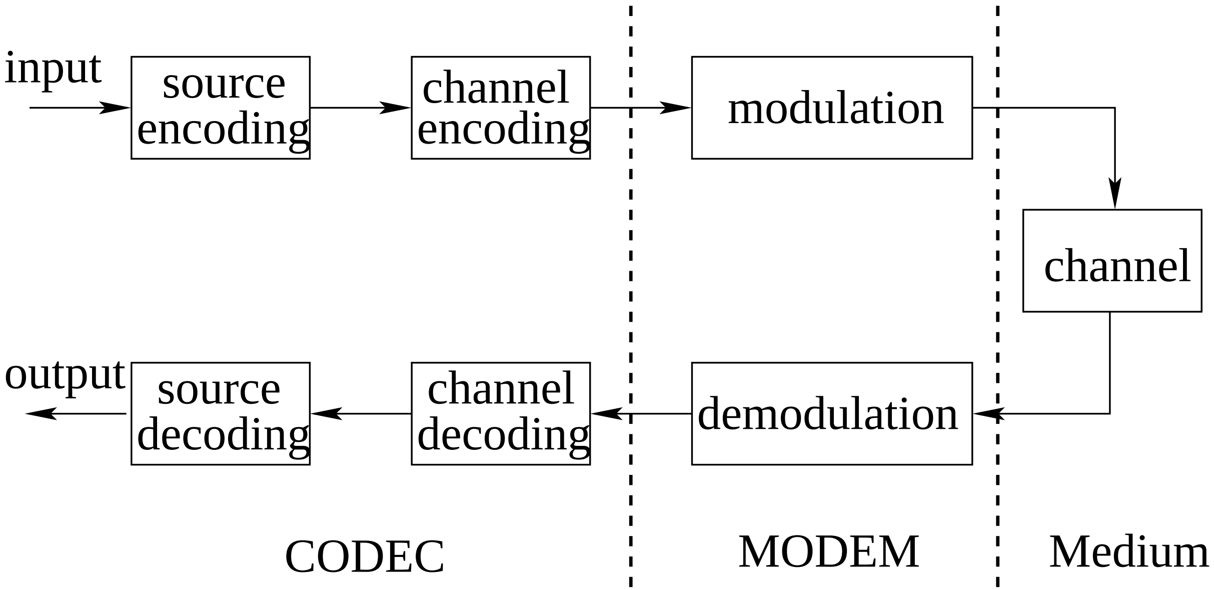
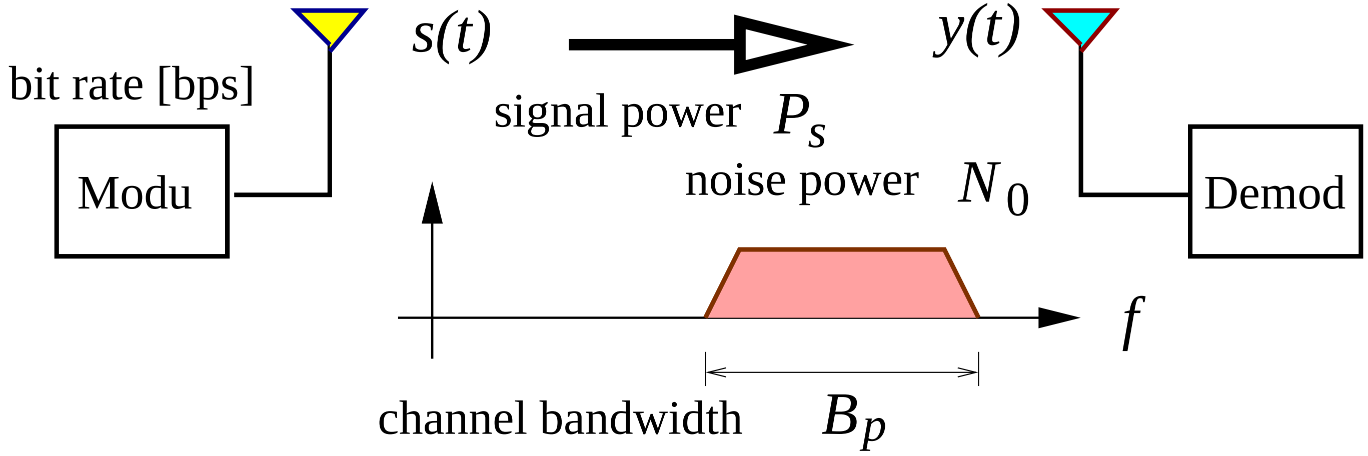
Digital Communication System

- Purpose: communicate information at required **rate** between geographically separated locations reliably (**quality**)
  - Important point: rate, quality ↔ **spectral bandwidth**, **power** requirements
  - Information theory provides guiding principles for everything in communication
- Major components: a pair of transmitter and receiver called transceiver
  - "Horizontal" partition: transmitter, receiver and channel (transmission medium)
  - "Vertical" partition: CODEC, MODEM and channel



Recap of Channel Capacity

- Information theory provides us basic theory for communication system design, including MODEM



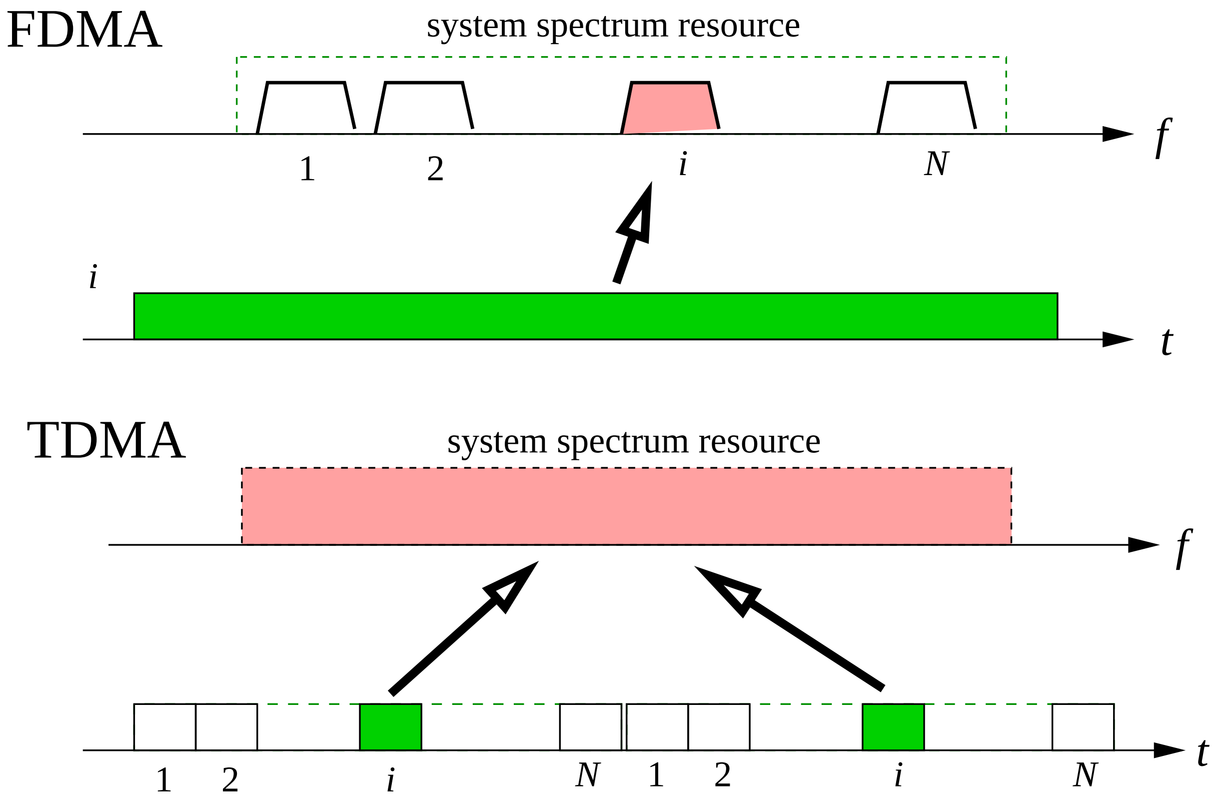
  - Assuming AWGN channel with Gaussian signal $s(t)$, channel capacity

$$C = B_p \log_2 \left(1 + \frac{P_s}{N_0} \right) \quad \text{[bps]}$$

  - Maximum rate could be achieved, i.e. upper limit
- MODEM responsible: **transfer the bit stream at required rate over the communication medium reliably**
  - Required rate [bps] with required quality ↔ **spectral bandwidth** and **power** requirements
- **Carrier** communication: $s(t)$ is radio frequency signal, because low frequency signal cannot travel far, also spectral resource (channels) are in RF

Channel Partition

- Frequency division multiple access: system spectral band is divided into frequency slots
  - A user is assigned with a frequency slot (channel), who can transmit continuously in time, but its signal spectrum must be inside its allocated frequency slot



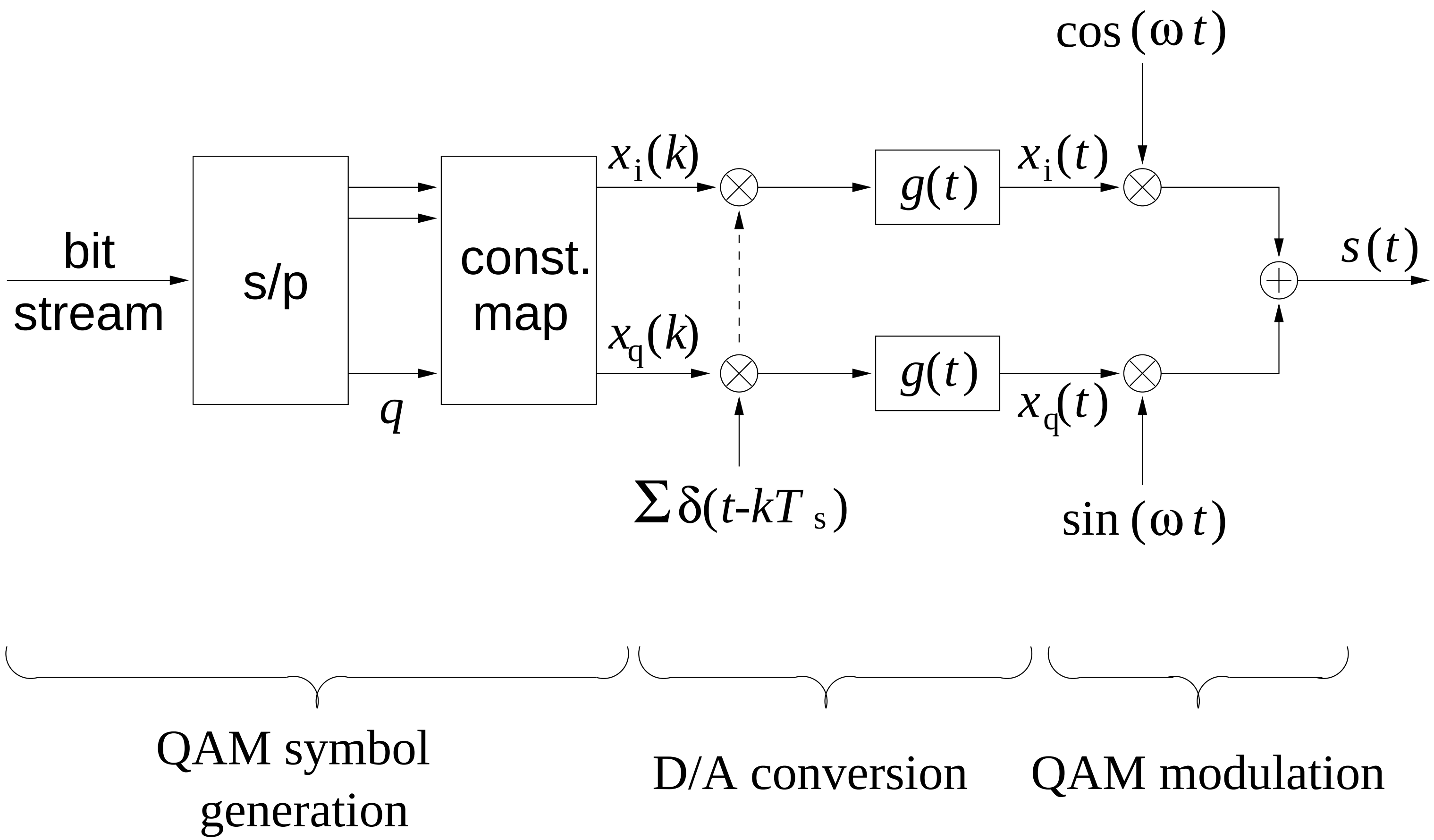
- Time division multiple access: transmission in time frames, and each frame divided into time slots
  - A user is assigned with a time slot (channel), who can only transmit in time bursts, i.e. in its allocated time slots, and its signal spectrum can occupy whole system spectral band

Digital Modulation

- In old day, communications were *analogue*, analogue modulation techniques include
  - amplitude modulation (AM), frequency modulation (FM), and phase modulation (PM)
- Communications today are all *digital*, and equivalent digital modulation forms
  - amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK)
- Carrier signal in digital communication is sin waveform $A \sin(2\pi f_c t + \theta)$, specified by amplitude A, frequency f_c, and phase θ
  - Use amplitude, frequency, or phase of the carrier to "carry" information leads to ASK, FSK, or PSK
  - A large number of digital modulations are in use, and often combinations of these three basic ways are employed
- We will consider quadrature amplitude modulation (QAM), which is a combination of ASK and PSK

Quadrature Amplitude Modulation

- Let us start our study from transmitter

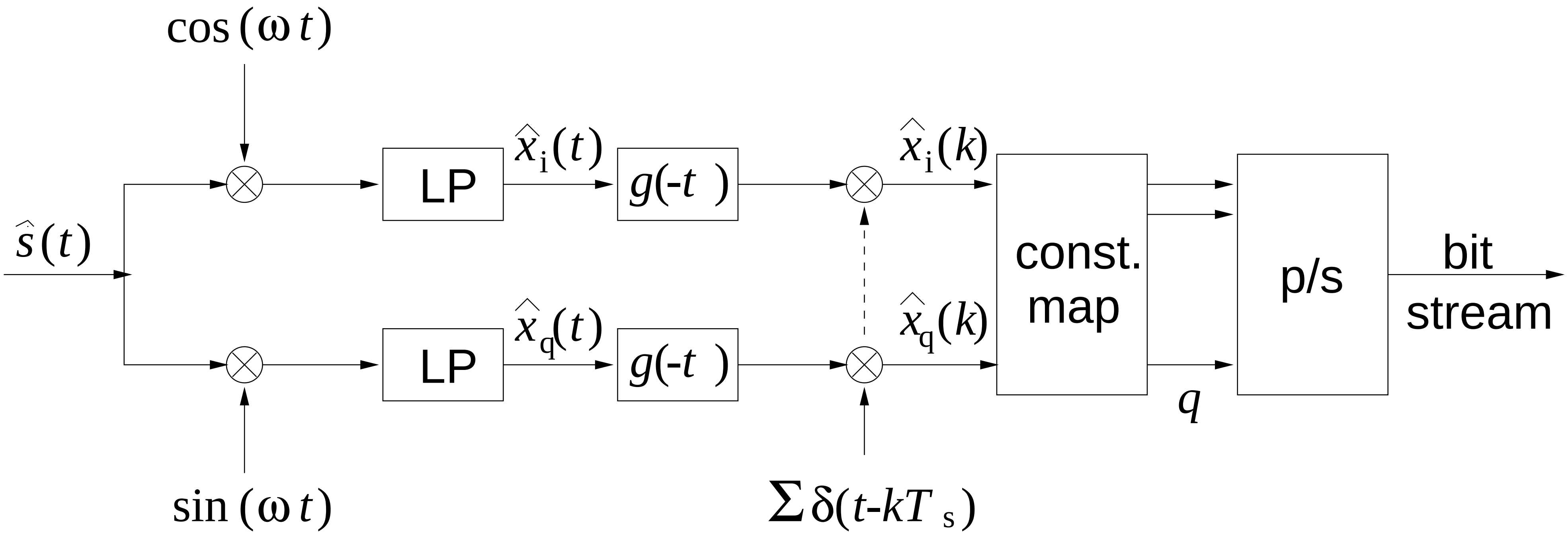


Note: e.g. odd bits go to form $x_{\rm i}(k)$ and even bits to form $x_{\rm q}(k)$; $x_{\rm i}(k)$ and $x_{\rm q}(k)$ are in-phase and quadrature components of the $x_{\rm i}(k) + jx_{\rm q}(k)$ QAM symbol; $x_{\rm i}(k)$ and $x_{\rm q}(k)$ are M-ary symbols

D/A conversion is not "correct name", should be "transmit filter", part of pulse shaping filter pair

Quadrature Amplitude Demodulation

- At receiver



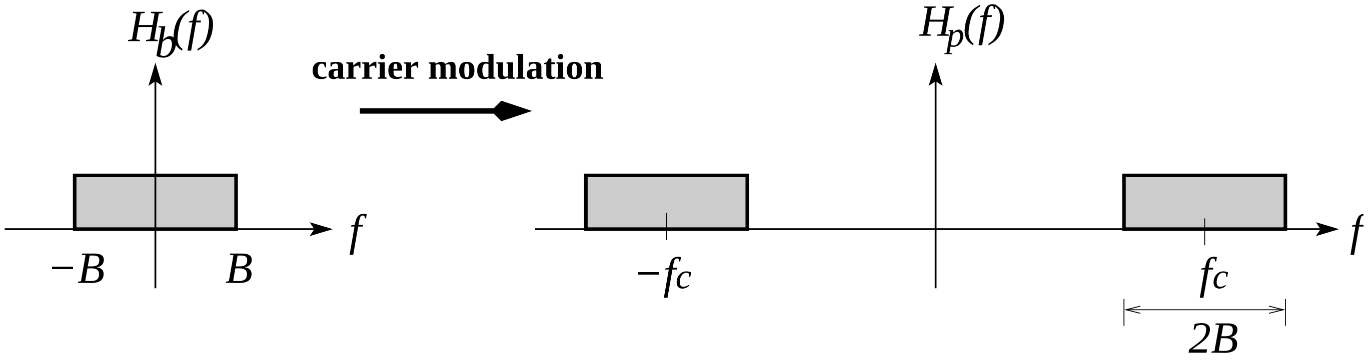
QAM demodulation symbol detection bit recovery

- In-phase and quadrature branches are "identical"
  - many issues, such as design of Tx/Rx filters $g(t)/g(-t)$, carrier recovery, synchronisation, can be studied using one branch

Channel Characteristics

- Between modulator and demodulator is medium (channel)

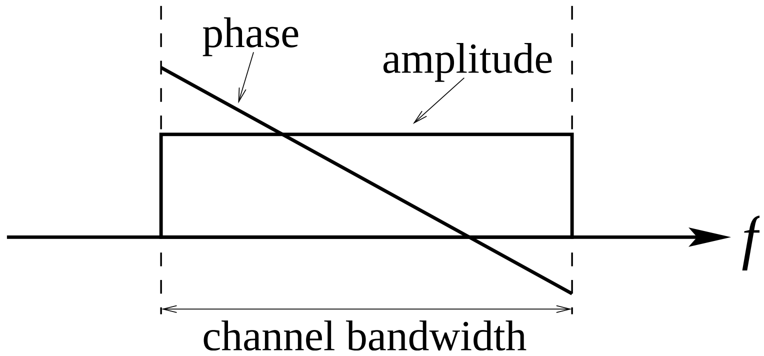
- **Passband** channel and **baseband** (remove modulator/demodulator) equivalence:



  – Baseband channel bandwidth $B \leftrightarrow$ passband channel bandwidth $B_p = 2B$

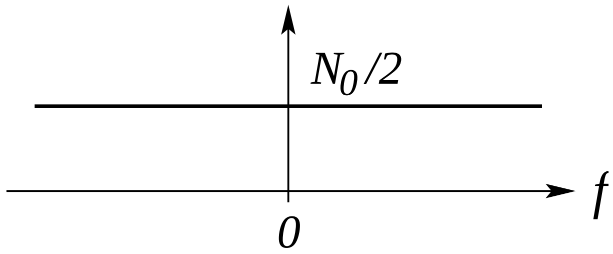
- Communication is at passband channel but for analysis and design purpose one can consider equivalent baseband channel

- Channel has **finite** bandwidth, **ideally** phase spectrum is linear and amplitude spectrum is flat:

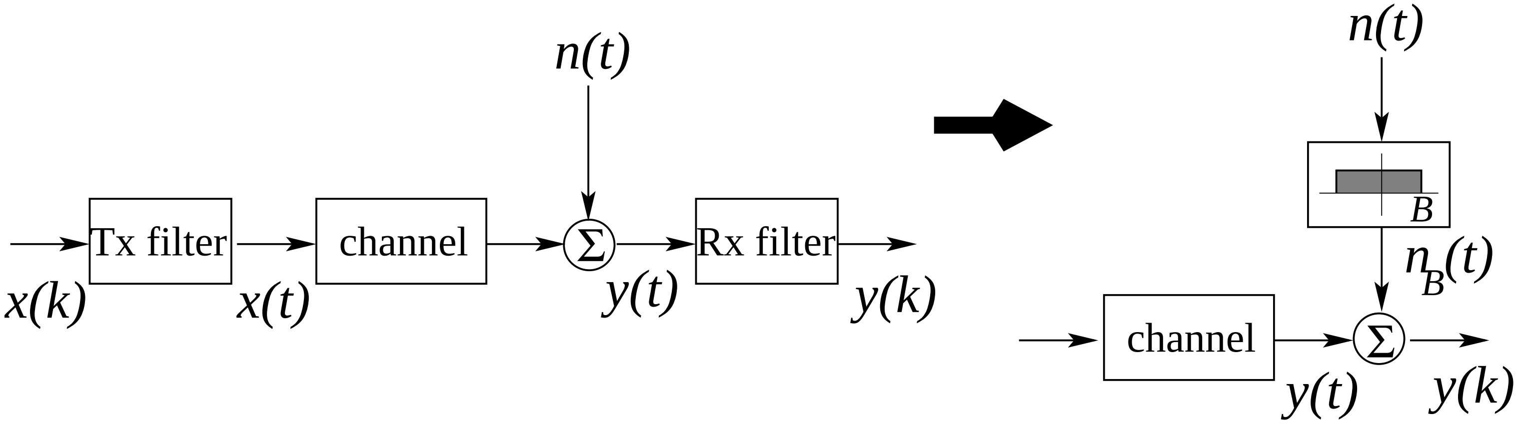


Channel Noise

- **Bandwidth** is a prime consideration, and another consideration is **noise** level
  - At receiver, power amplifier amplifies weak received signal also introduces noise
  - How serious power amplifier introducing noise is quantified by noise figure of amplifier
- Channel noise: AWGN with a constant power spectrum density (PSD) $N_0/2$
- Power is the area under PSD, so a white noise has infinitely large power



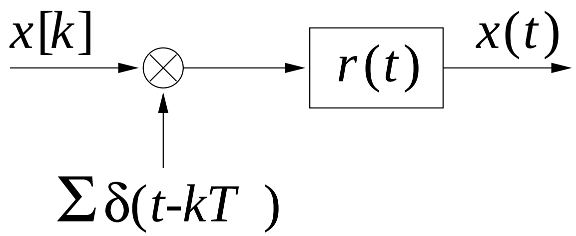
- But communication channels are **bandlimited**, so noise is also bandlimited and has a finite power
  - Noise $n(t)$ introduced by power amplifier passes through Rx filter who has a bandwidth of B
  - Thus noise at the receive signal, $n_B(t)$, has a power of $N_0 \cdot B$



Pulse Shaping — Starting Point

- Unless transmission symbol rate f_s is very low, one cannot use impulse, narrow pulse or rectangular pulse to transmit data symbols
  - Such pulses have very large (infinite) bandwidth, but we only have finite baseband bandwidth B
- Discrete samples have to be **pulse shaped**
  - $\{x[k]\}$: transmitted symbols
  - $\sum \delta(t - kT_s)$: pulse clock (every T_s s a symbol is transmitted)
  - $r(t)$: combined impulse response of Tx/Rx filters, and channel

    $r(t) = g(-t) \star c(t) \star g(t)$ or $R(f) = G_R(f) \cdot C(f) \cdot R_T(f)$



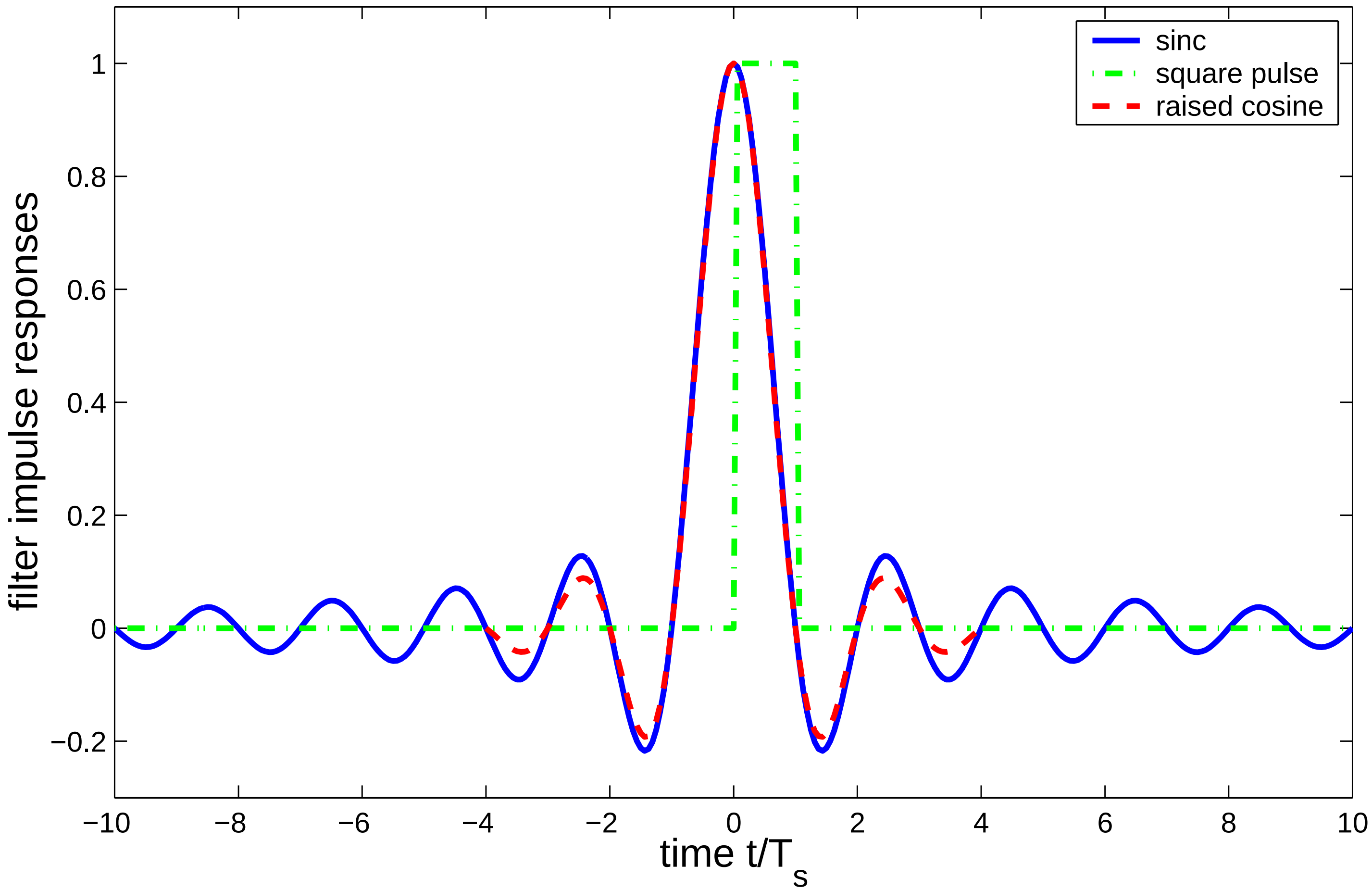
  - Baseband (received) signal, assuming no noise

$$x(t) = r(t) \star \left(\sum x[k]\delta(t - kT_s) \right) = \int \sum r(t - \tau) \cdot x[k]\delta(\tau - kT_s)\, d\tau$$

$$= \sum_{k=-\infty}^{+\infty} x[k] \cdot r(t - kT_s)$$

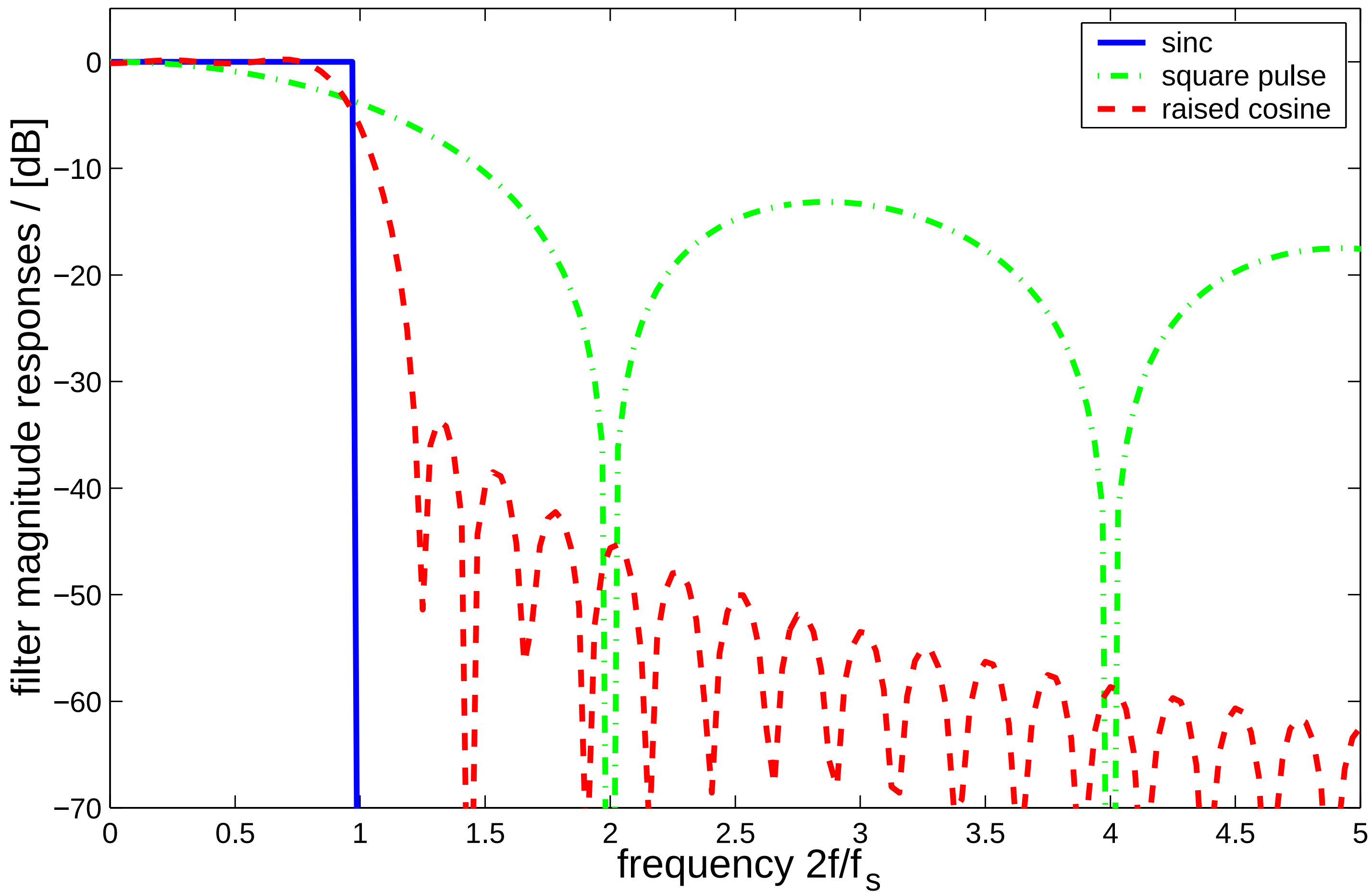
- What are the requirements for $r(t)$? or how should we choose this combined impulse response $r(t)$ so that we can retrieve the original data sample $x[k]$ from $x(t)$?
- To transmit at symbol rate f_s needs certain bandwidth B_T and B_T depends on which pulse shaping used — does the channel bandwidth B enough to accommodate signal bandwidth B_T?

Pulse Shaping — Time Domain



- **1.** square: last one T_s; **2.** sinc: assume $t \to \pm\infty$; and **3.** raised cosine: truncate to 8 T_ss
  - Time support of square is one T_S, looks suitable for continuously sending $\{x[k]\}$ at $t = kT_S$ or is it?
  - Time supports of sinc and truncated raised cosine last many T_S, how could we send continuously $\{x[k]\}$ at $t = kT_S$? Mixed up!
- All these filters have regular zero-crossing at symbol-rate spacing except $t = 0$ (**Nyquist** system)

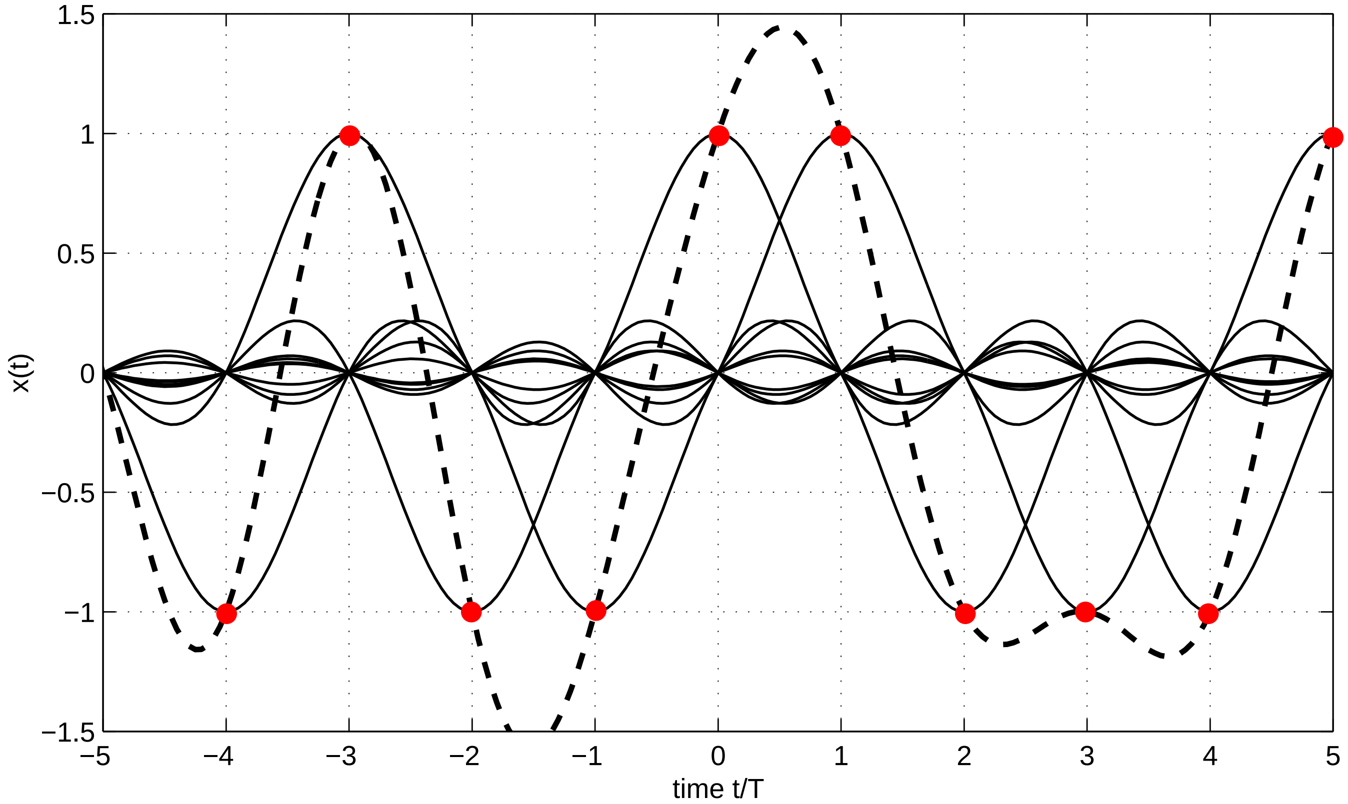
Pulse Shaping — Frequency Domain



- Remember channel bandwidth B is finite, and signal bandwidth B_T must fit in it
  - square pulse produces considerable large excess bandwidth well beyond symbol rate f_s
  - sinc pulse has exactly finite bandwidth of $f_s/2$, but impractical to realize
  - truncated raised cosine has main bandwidth within f_s, and easy to realize

Right Pulse Shaping

- Recall we are discussing how to choose $r(t)$ so that we can recover $\{x[k]\}$ from $x(t)$
- Example – binary (± 1) $\{x[k]\}$, each is transmitted as a sinc pulse: the peak of different shifted sinc functions (different $x[k]$) coincide with zero crossings of all other sincs (other data symbols)



  – At receiver, sampling at correct symbol rate enables recovery of transmitted $x[k]$!
- Right pulse shaping seems: combined impulse response $r(t)$ has regular zero-crossing at symbol-rate spacing except it peaks at $t = 0$, a **Nyquist** filter

Transmit and Receive Filters

- Pulse shaping fulfils two purposes
  - limit the transmission bandwidth B_T so it fits in channel bandwidth B, and
  - enable to recover the correct sample values of transmitted symbols

- Such a pulse shaping $r(t) = g(-t) \star c(t) \star g(t)$ is called a Nyquist system

  1. (Infinite) sinc has a (baseband) bandwidth $B_T = f_s/2$, (infinite) raised cosine has $f_s/2 \leq B_T \leq f_s$ depending on roll-off factor
  2. A Nyquist time pulse have regular zero-crossing at symbol-rate spacings to avoid interference with neighbouring pulses at correct sampling instances

- Assuming ideal channel $c(t)$, Nyquist system $r(t)$ is separated into transmit filter $g(t)$ and receive filter $g(-t)$

  1. The filter $g(-t)$ in the receiver is also called a matched Filter (to $g(t)$); $g(t)$ and $g(-t)$ are basically identical (square-root of $r(t)$)
  2. This division of $r(t)$ enables suppression of out-of-band noise and results in the maximum received signal-to-noise ratio (SNR)

Summary

- Revisit major blocks of a digital communication system
  - "Horizontal" partition: transmitter and receiver (transceiver), and channel
  - "Vertical" partition: CODEC, MODEM, and channel

- MODEM: responsible for transferring the bit stream at required rate rate over the communication medium reliably
  - Required rate [bps] with required quality $\leftrightarrow$ **spectral bandwidth** and **power** requirements, as highlighted by channel capacity
  - Transmission channel (medium) has finite bandwidth and introduces noise, two factors that have to be considered in design

- Purpose of pulse shaping, how to design transmit and receive filters
  - limit the transmission bandwidth so it can fit in channel bandwidth
  - enable to recover the correct sample values of transmitted symbols