IEE 1711: Applied Signal Processing

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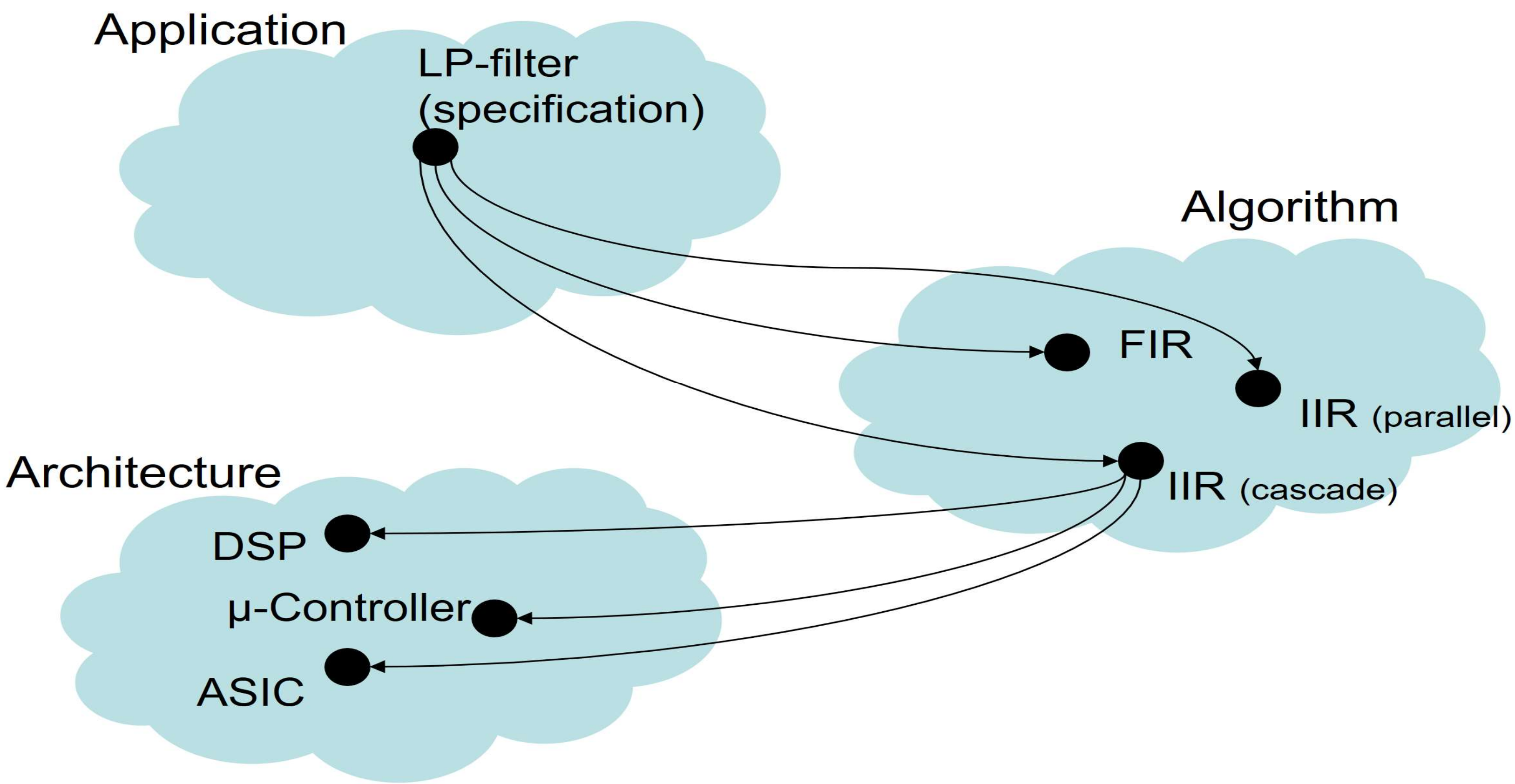


Outline

- Lecture 3: Digital Image Processing
  - Followup
- Lecture 4: Digital Communication - Transmitter
  - Overview of Digital Communication
    - Transmitter
    - Channel
    - Receiver
  - Tranmitter
    - Digital Communication System
    - Passband vs Baseband tranmission
    - Constellation of waveform
    - Bit Error Rate
- Summary

Analog vs. Digital Communication (1/4)

Analog Communication:

- Transmission of signals that are inherently analog (speech, video, etc..)
- Baseband or passband (AM, FM, ..)
  - While **baseband** is the original signal, **passband** is the filtered signal that is eventually converted back to **baseband**. Some short-distance systems do not have to modulate **baseband** to higher frequencies before transmission.
- Bandwidth = signal bandwidth

  Example: speech signal 0..4kHz -> BW=4kHz
- Received signal subject to channel impairments, transmitter/receiver impairments, etc..

Analog vs. Digital Communication (2/4)

Digital Communication:

- Transmission of signals that are inherently digital (`data') *or* analog (speech, video, etc..)
- Analog signals are converted into digital signals by *sampling & quantization* (A-to-D conversion)

Example : =PCM (pulse code modulation)

- speech 0…4kHz

- sampled at 8kHz (cfr. Nyquist criterion),

- each sample converted into 8 bits number-> 64kbits/sec

Analog vs. Digital Communication (3/4)

Digital Communication

- What?

A principle feature of a digital communication system is that during a finite interval of time, it sends a waveform from a *finite set of possible waveforms*.

The objective of the receiver is not to reproduce the transmitted waveform, **but (only) to determine *which of the possible waveforms has been sent*.**

Analog vs. Digital Communication (4/4)

Digital Communication Key Features:

- source coding/compression:

  Example: speech signal

  64kbits/sec-> 11kbits/sec...4kbits/sec

  (through `signal modeling')

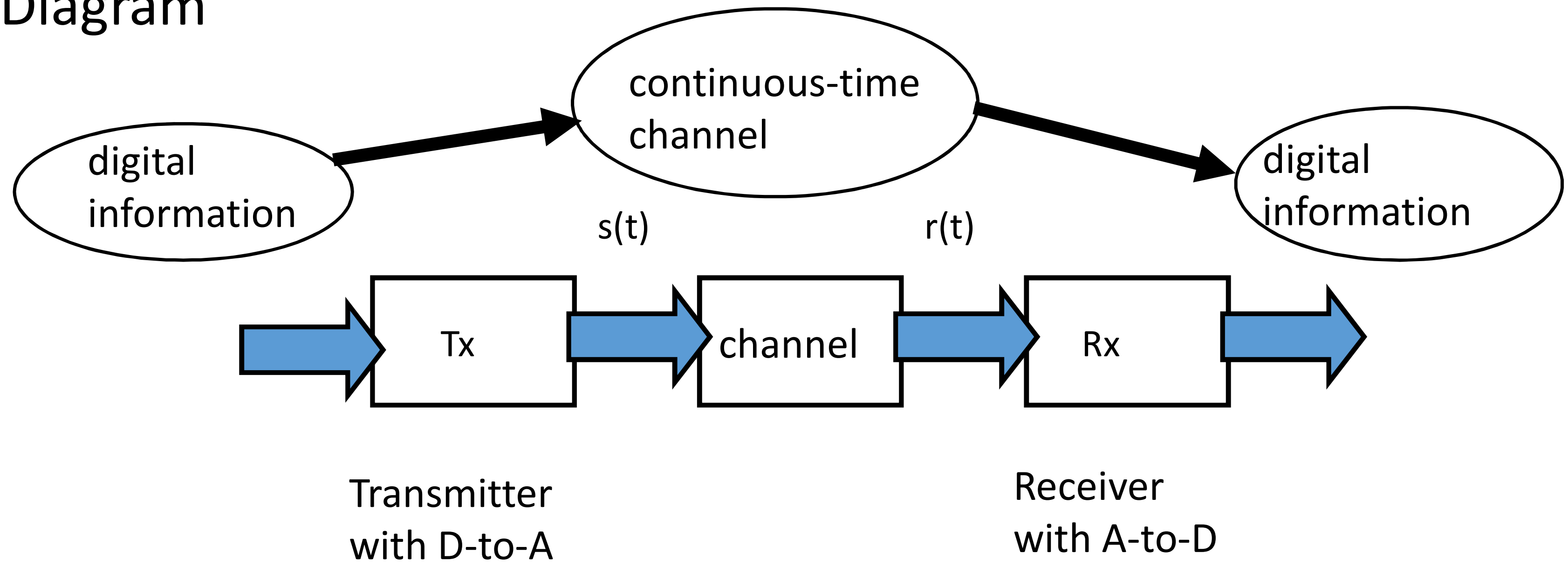
- channel coding/error correction
- increased spectral efficiency through coding, signal processing, etc.

  Example: v.34 voice-band modem

  33.6 kbits/sec in 4kHz voice-band (=8bits/sec/Hz)

Digital Communication System (1/4)

- Block Diagram



- Digital Information is digital signal (data) or `sampled+quantized' analog signal (speech,..)

Digital Communication System (2/4)

Transmitter

- converts bit sequence into waveform s(t)

  (=`modulation’)
- bits are grouped into `symbols’

  (n bits per symbol, hence M=2^n different symbols)

  (=`symbol alphabet’, `constellation’)
- each symbol corresponds to a different waveform segment
- symbol rate = # transmitted symbols/sec = Rs

  (`Baud rate’, after Baudot, French telegraph engineer)

Digital Communication System (3/4)

Channel

- physical medium :

  twisted pair, coax, optical fiber, radio

- channel impairments :

  noise, attenuation/distortion, cross-talk,

  interference, etc…

$$r(t) \neq s(t)$$

Digital Communication System (4/4)

Receiver

- Converts received signal r(t) into bit sequence
  (=`demodulation/detection')

- Receiver performance :
  Bit Error Probability (BEP) or Bit Error Rate (BER)
  BER = (#bit errors) / (#transmitted bits)
  example : voice : BER <1E-3
  data : BER <1E-10

Transmitter (I/3)

- Transmitted bits are grouped into symbols
  (n bits per symbol, hence M=2^n symbols)
- Transmitted symbols are

$$a_1, a_2, a_3, a_4, \ldots \qquad a_k \in \{\alpha_1, \alpha_2, \alpha_3, \ldots, \alpha_M\}$$

- Transmitted signal is

$$s(t) = \sqrt{E_s} . \sum_k p(t - kT_s; a_k)$$

where p(t) is transmit pulse, and E_s is symbol energy
(a_k and p(t) are energy-normalized), Ts is symbol period

Transmitter (2/3)

- Transmitted signal is

$$s(t) = \sqrt{E_s}.\sum_k p(t - kT_s; a_k)$$

- Linear modulation (e.g. PAM, QAM, PSK)

  all signal segments are proportional to the same pulse p(t)

$$s(t) = \sqrt{E_s}.\sum_k a_k.p(t - kT_s)$$

- Non-linear modulation (e.g. FSK)

Transmitter (3/3)

- Constellations for linear modulation
  (=`symbol alphabet')

PAM — pulse amplitude modulation

QAM — quadrature amplitude modulation

PSK — Phase-shift keying

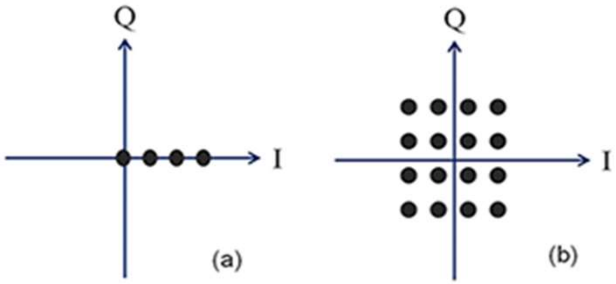
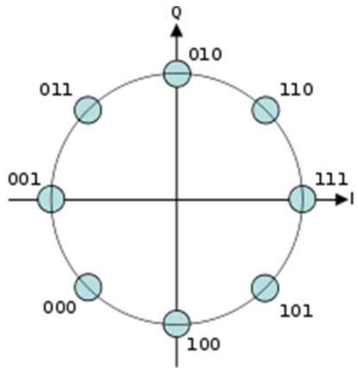


Fig. 1 (a) PAM-4 and (b) 16-QAM signal constellation diagrams.
I=in phase, Q=quadrature phase.



Channel (1/3)

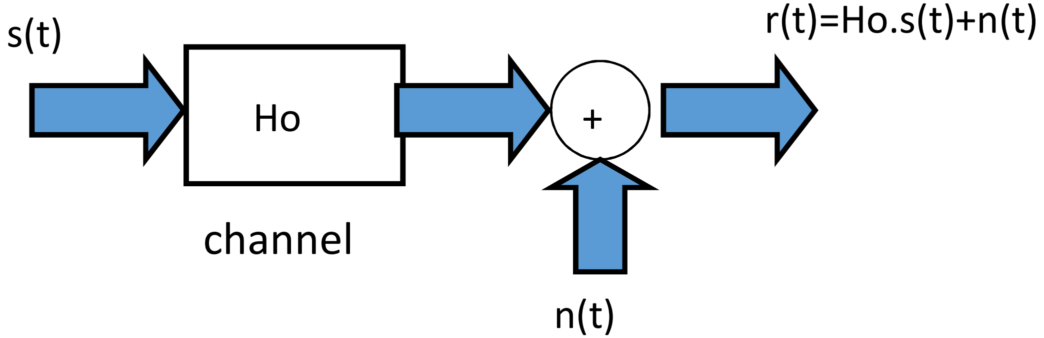
Channel impairments:

- attenuation/distortion (linear/non-linear)
- noise (linear/non-linear)
- cross-talk (1 or many)
- echo (e.g. hybrid impedance mismatch)
- RFI (e.g. amateur radio)

Channel (2/3)

- Mostly simple linear channel models
- Example: AWGN-channel

  (additive white Gaussian noise channel)

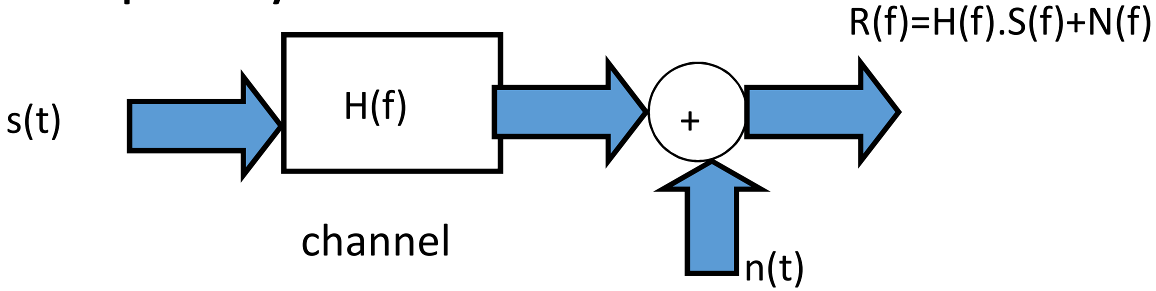


n(t) is zero-mean Gaussian process with power spectrum No/2 for |f|<B (B=bandwidth)

Channel (3/3)

- PS: Gaussian noise model justified through central limit theorem (ex: 1 cross-talker is non-Gaussian, 30 cross-talkers approx. Gaussian)

- Example: frequency-selective channel



frequency-dependent channel attenuation/phase distortion

(example: twisted pair, coax)

Receiver (1/3)

- Receiver retrieves transmitted symbols $a_1, a_2, a_3, \ldots$ from received signal r(t)

- This leads to an *optimization problem*

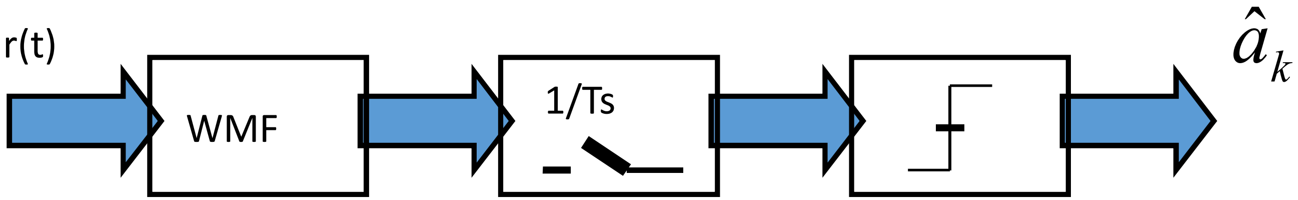
  Example: minimum distance receiver

$$\min_{a_1, a_2, a_3, \ldots} \int | r(t) - \sqrt{E_s} . \sum_k a_k . p'(t - kT_s) |^2 \, dt$$

  where p'(t) is transmit pulse p(t), modified by channel

Receiver (2/3)

- For AWGN channels (<->frequency-selective channels), a receiver may consist of :
  - (a *front-end* `(whitened) matched filter’, WMF)
  - a *symbol-rate sampler* (i.e. 1 sample/symbol interval)
  - a (memory-less) *decision device* that decides on the nearest symbol in the symbol alphabet



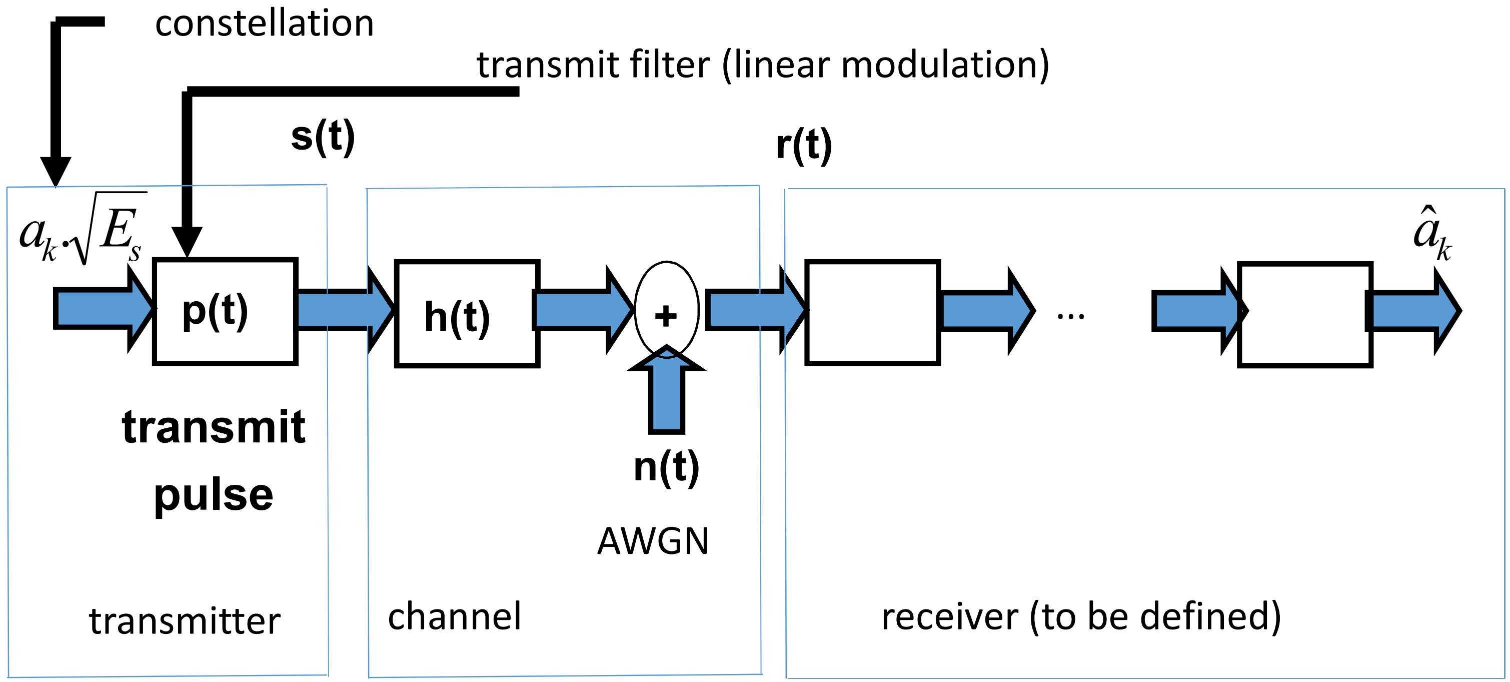
- Timing instant for symbol-rate sampling is crucial, hence *synchronization scheme* needed !

Receiver (3/3)

- For frequency-selective channels, the receiver may consist of
  - WMF + symbol-rate sampling front-end, or
  - anti-alias filtering + Nyquist-rate sampling front-end

  followed by more complicated processing:
  - Maximum-likelihood sequence estimation (e.g. Viterbi algorithms)
  - Equalization + decision device
  - …

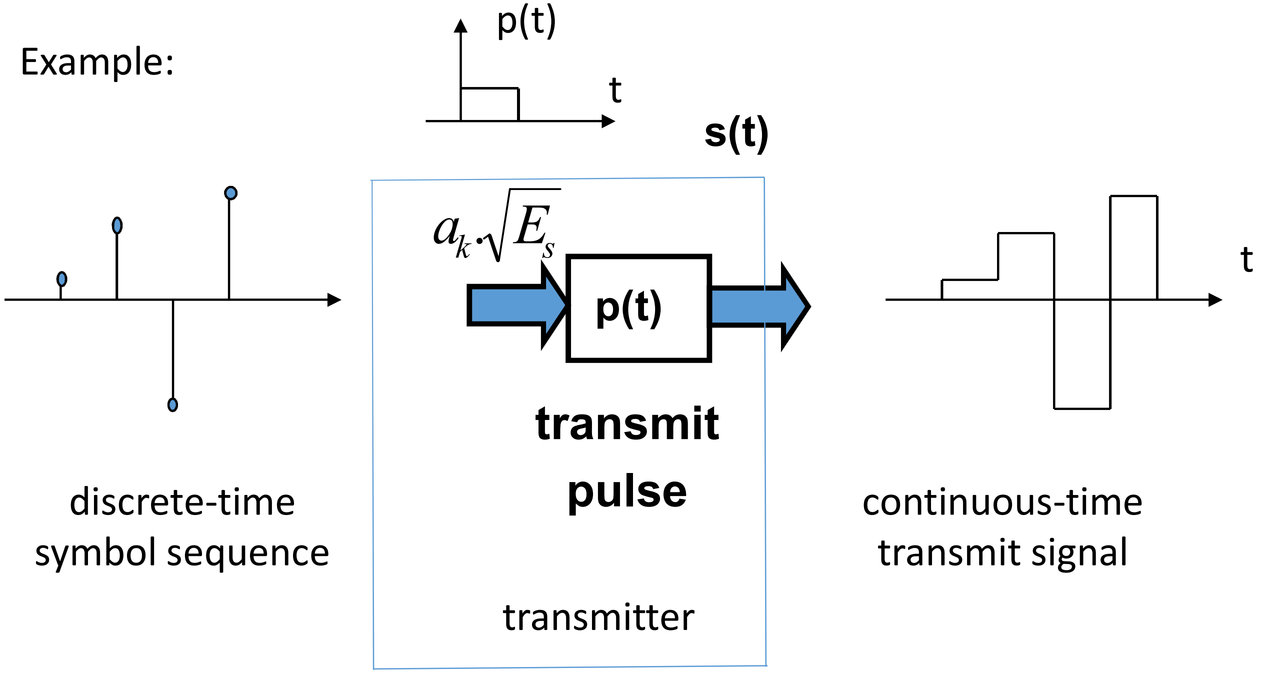
Transmitter: Constellation + Transmit Filter (1/2)



$$s(t) = \sqrt{E_s} . \sum_k a_k . p(t - kT_s)$$

PS: channel coding (!) not considered here

Transmitter: Constellation + Transmit Filter (2/2)



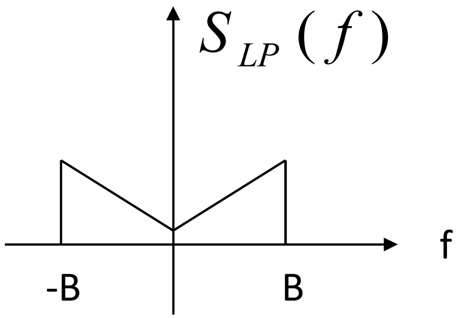
-> s(t) with infinite bandwidth, not the greatest choice for p(t)..

-> implementation: upsampling/digital filtering/D-to-A/S&H/...

Preliminaries: Passband vs. baseband transmission (I)

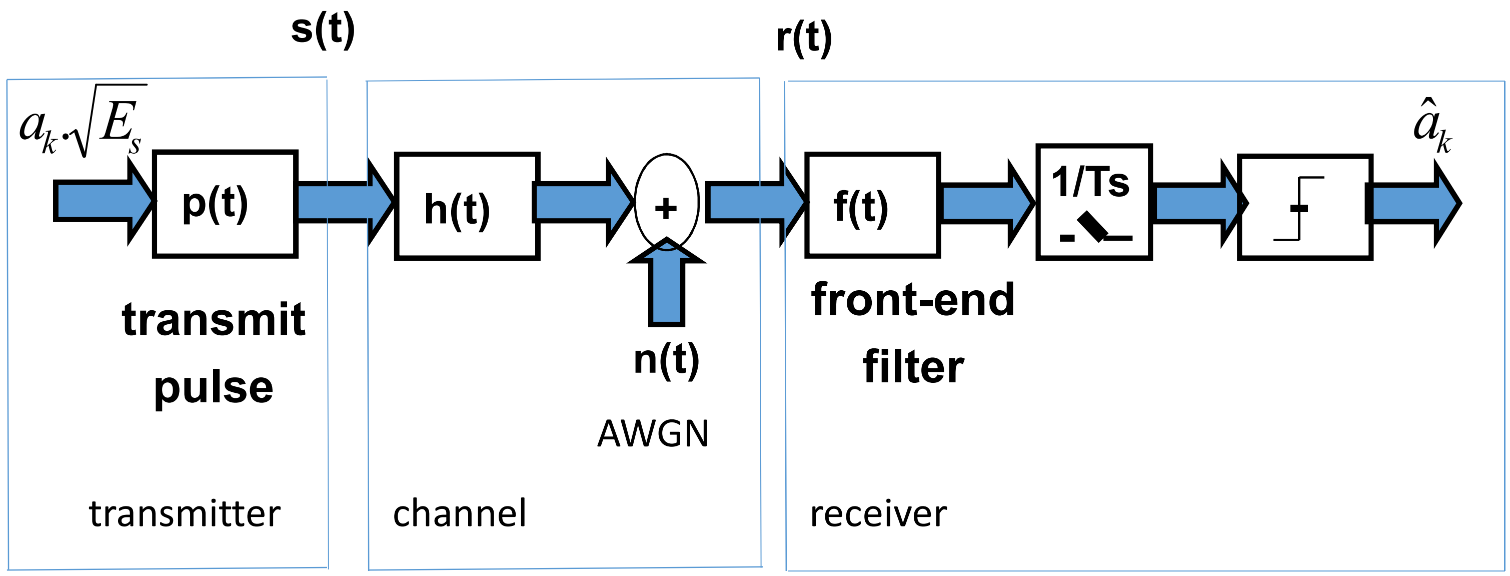
Baseband transmission

- transmitted signal is $s_{LP}(t) = \sqrt{E_s}.\sum_k a_k.p(t - kT_s)$ (linear modulation)
- transmitted signals have to be real, hence a_k = real, p(t)=real
- baseband means $S_{LP}(f) = 0$ for $|f| > B$



Preliminaries : Passband vs. baseband transmission (II)

Baseband transmission model/definitions



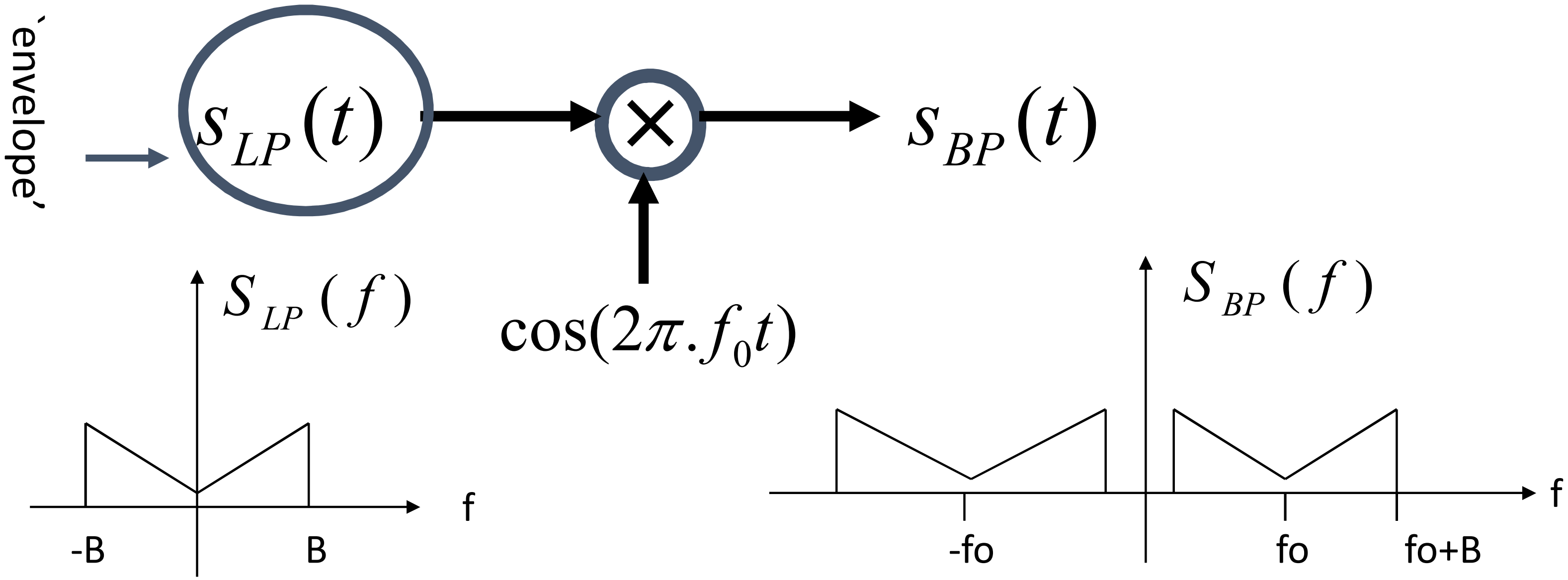
g(t)=p(t)*h(t)*f(t) (convolution)

everything is real here!

Preliminaries : Passband vs. baseband transmission (III)

Bandpass transmission

transmitted signal is modulated baseband signal

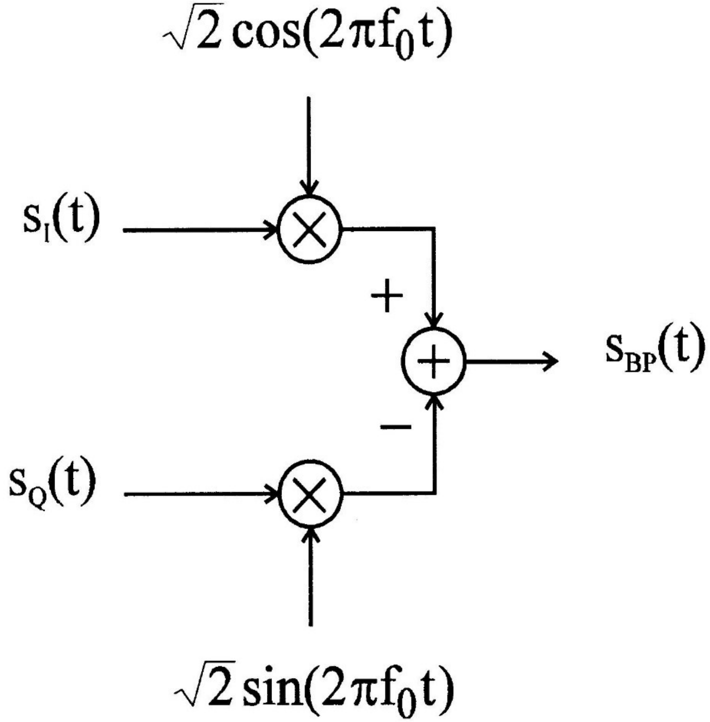


Preliminaries : Passband vs. baseband transmission (IV)

Bandpass transmission:

- note that modulated signal has 2x larger bandwidth, hence inefficient scheme !
- solution = accommodate 2 baseband signals in 1 bandpass signal :

  I =`in-phase signal'
  Q=`quadrature signal'



$\sqrt{2}$ such that energy in BP is energy in LP

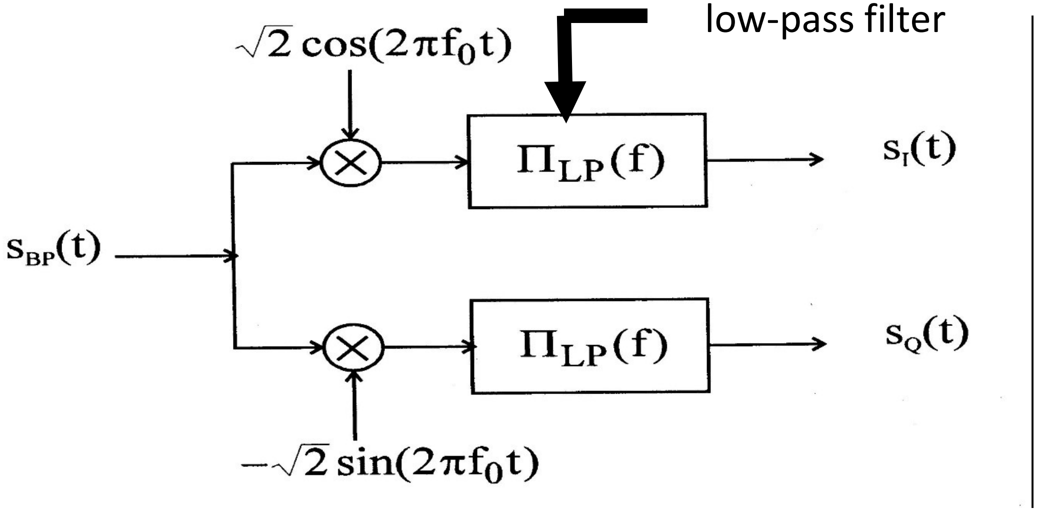
Preliminaries : Passband vs. baseband transmission (V)

- Convenient notation for `two-signals-in-one' is complex notation :

$$s_{LP}(t) = s_I(t) + j.s_Q(t)$$

`complex envelope'

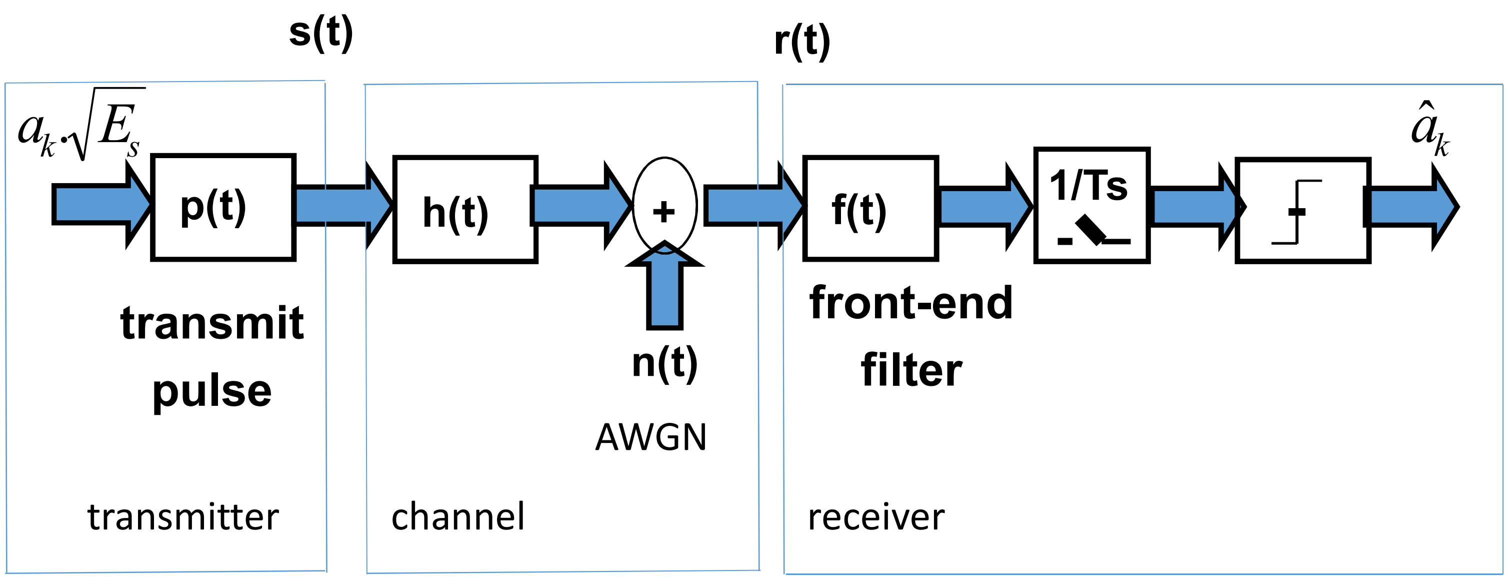
- re-construct `complex envelope' from BP-signal



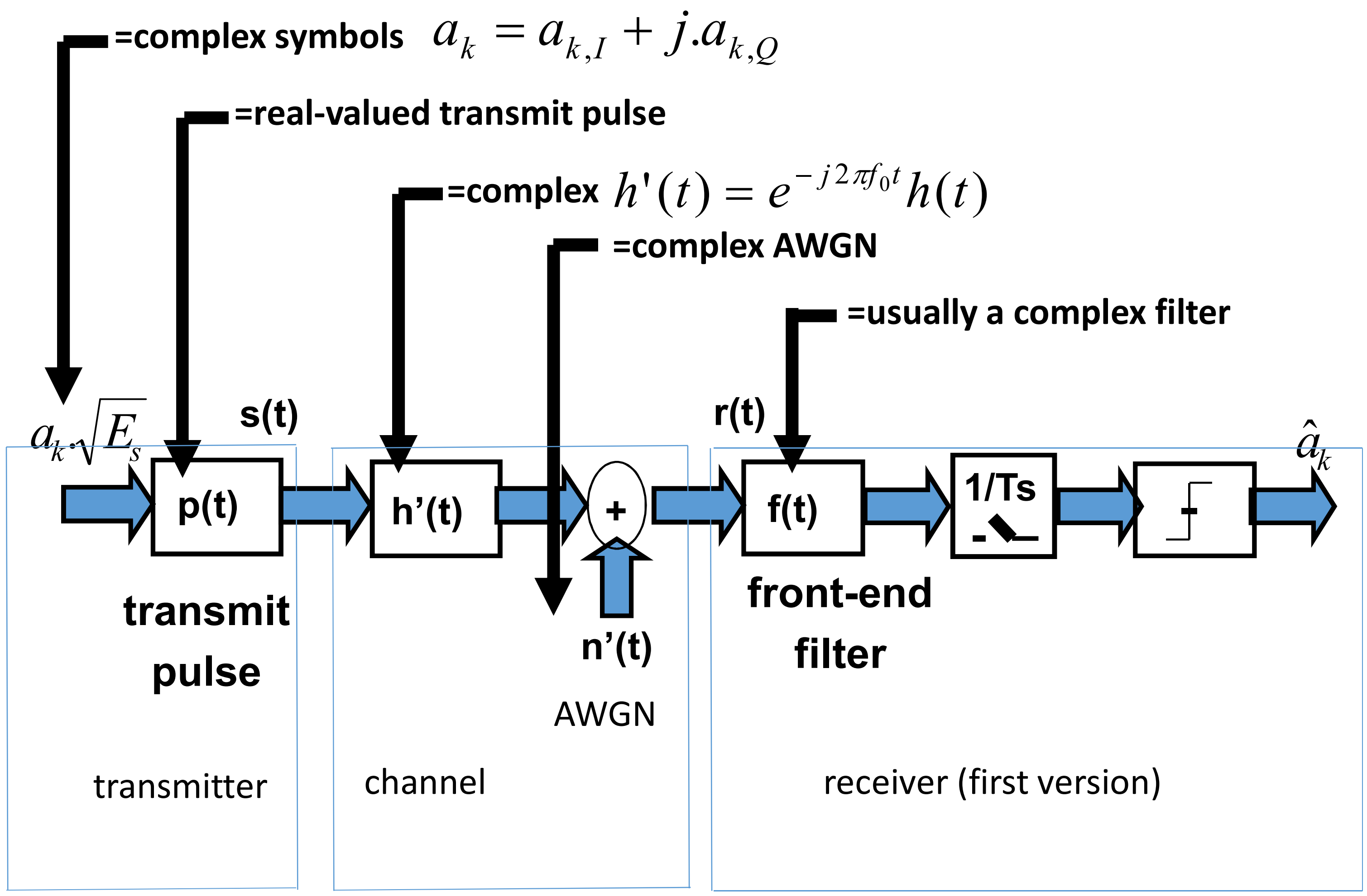
Preliminaries : Passband vs. baseband transmission (VI)

Passband transmission model/definitions

a convenient and consistent (baseband) model can be obtained, based on complex envelope signals, that does not have the modulation/demodulation steps:



Preliminaries : Passband vs. baseband transmission (V)



Preliminaries : Passband vs. baseband transmission (VI)

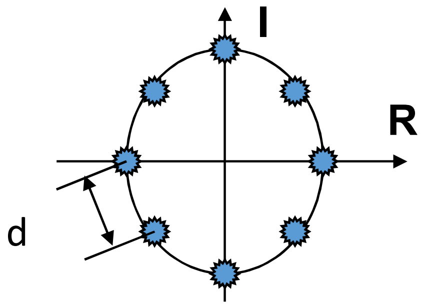
- No major difference between baseband and passband transmission/models (except that many things (e.g. symbols) can become complex-valued).
- PS: modulation/demodulation steps are transparent (hence may be omitted in baseband model) only if receiver achieves perfect *carrier synchronization* (frequency fo & phase).
  Synchronization not addressed here
  (see e.g. Lee & Messerschmitt, Chapter 16).

Constellations for linear modulation (I)

M-PSK phase-shift keying

$$a_k \in \left\{ \exp(j.2\pi \frac{m}{M}) \mid m = 0,1,\ldots, M-1 \right\}$$

- a_k energy-normalized if ....
- Then distance between nearest neighbors is



$$d_{PSK}(M) = 2.\sin(\frac{\pi}{M})$$

Constellations for linear modulation (II)

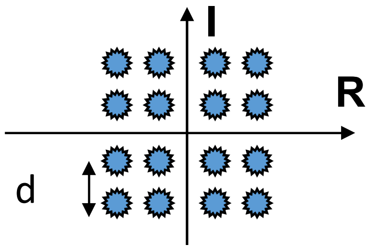
M-QAM quadrature amplitude modulation

$$a_k = a_{I,k} + j.a_{Q,k}$$

$$a_{I,k}, a_{Q,k} \in \left\{ \pm A_{QAM}, \pm 3A_{QAM}, \ldots\ldots, \pm(\sqrt{M}-1)A_{QAM} \right\}$$

$$d_{QAM}(M) = \sqrt{\frac{6}{M-1}}$$

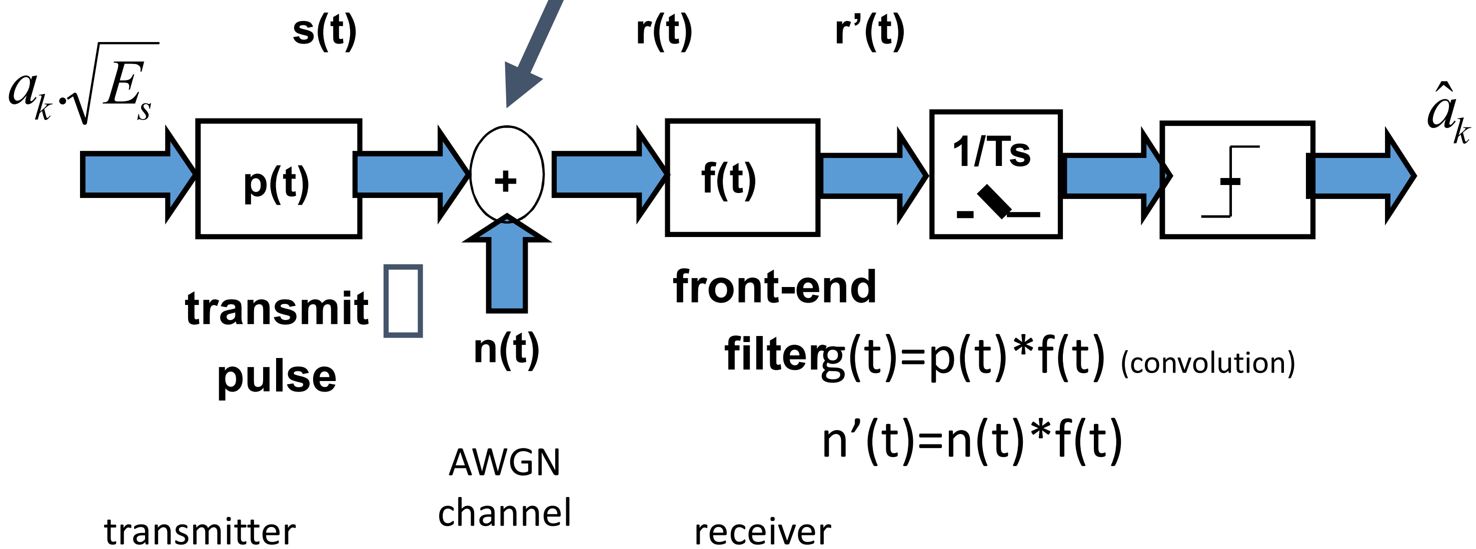
- distance between nearest neighbors is



$$d_{PAM}(M) \le d_{PSK}(M) \le d_{QAM}(M)$$

BER Performance for AWGN Channel

BER=(# bit errors)/(# transmitted bits)



g(t)=p(t)*f(t) (convolution)

n'(t)=n(t)*f(t)

BER for different constellations?

BER Performance for AWGN Channel

definitions:

- transmitted signal $$s(t) = \sqrt{E_s}.\sum_k a_k.p(t - kT_s)$$

- received signal (at front-end filter) $$r(t) = \sqrt{E_s}.\sum_k a_k.p(t - kT_s) + n(t)$$

- received signal (at sampler) $$r'(t) = \sqrt{E_s}.\sum_k a_k.g(t - kT_s) + n'(t)$$

g(t) =p(t)*f(t) = transmitted pulse p(t) filtered by front-end filter

n'(t) =n(t)*f(t) = AWGN filtered by front-end filter

BER Performance for AWGN Channel

Received signal sampled @ time t=k.Ts is...

$$r'(k.T_s) = \underbrace{\sqrt{E_s}.a_k.g(0)}_{1} + \underbrace{\sum_{m\neq 0} a_{k-m}.g(m.T_s)}_{2} + \underbrace{n'(k.T_s)}_{3}$$

1 = useful term

2= `ISI', intersymbol interference (from symbols other than a_k)

3= noise term

Strategy :

a) analyze BER in absence of ISI (=`transmission of 1 symbol')

b) analyze pulses for which ISI-term = 0 (such that analysis under a. applies)

c) for non-zero ISI, see Lecture 4-5