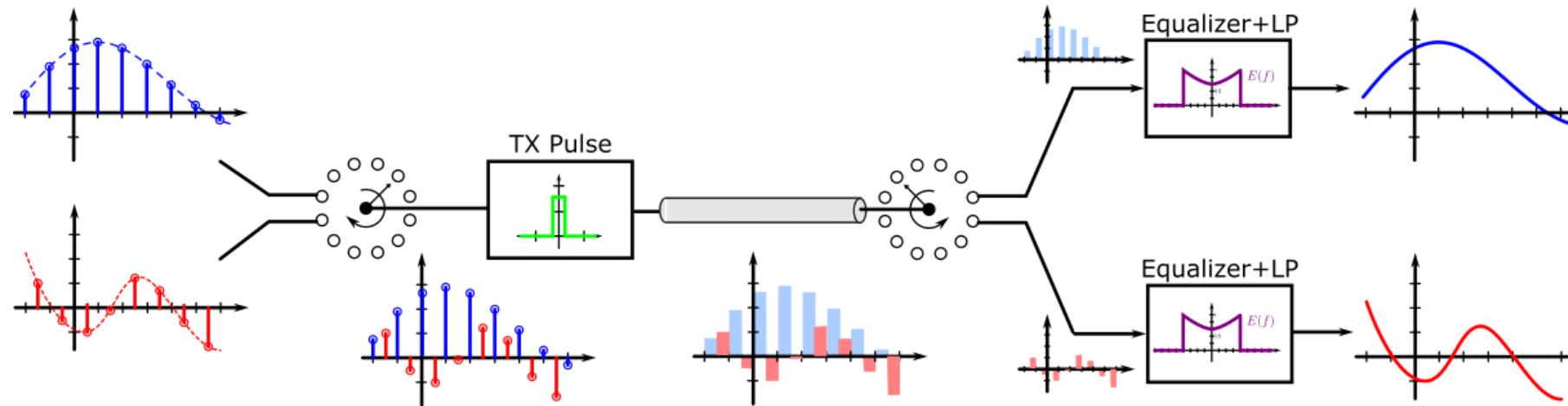


Recap from Week-5

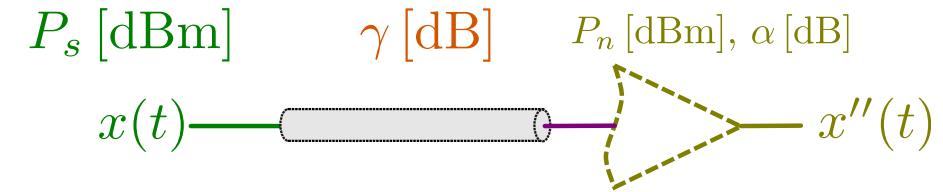
- Communication networks are often built as stars-of-stars
- Between local stars, multiple signals are multiplexed onto a single wire
- Pulse modulation is a straightforward way to “MUX” and “DE-MUX” provided that the bandwidth of the shared wire is sufficient (often the case)



Recap from Week-5

- Transmitting signals over long wires leads to attenuation and degrades the signal-to-noise ratio

$$SNR''[dB] = P_s[dBm] + \gamma[dB] - P_n[dBm]$$



- Periodic analog repeaters can not recover the SNR and even degrade SNR



$$SNR''[dB] = \underbrace{P_s[dBm] + \gamma[dB] - P_n[dBm]}_{\text{No repeaters}} - 10 \log_{10} N$$

EE-432

Systeme de

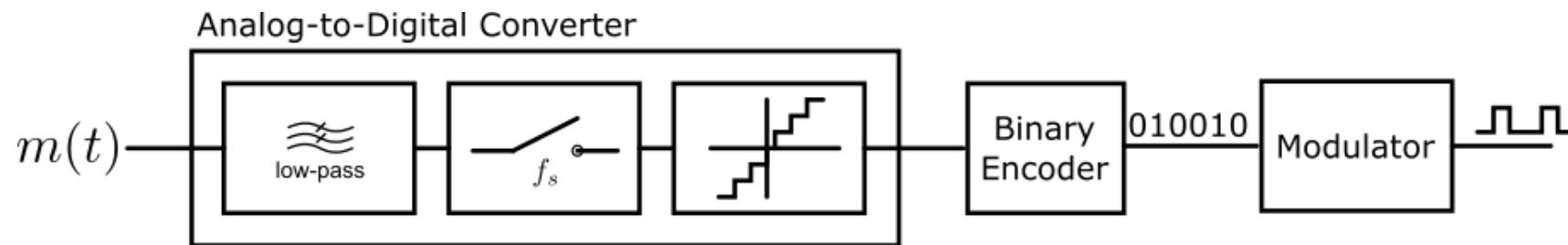
Telecommunication

Prof. Andreas Burg
Joachim Tapparel, Yuqing Ren, Jonathan Magnin

Quantization

Motivation: Binary Pulse Code Modulation (PCM)

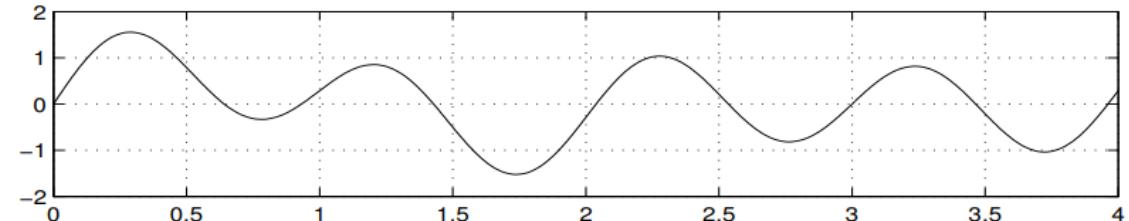
- **Binary PCM: sends a sequence of pulses that encode a binary digits (0/1)**
 - PCM can be generalized to non-binary digits, but we will use the term for binary-PCM
- **Binary PCM transmission of an analog signal**
 1. Analog-to-digital conversion (filtering, sampling, quantization)
 2. Binary encoding as sequence of bits 0/1 (binary-PCM)
 3. Modulation: representing bits as waveform (baseband or RF as appropriate)



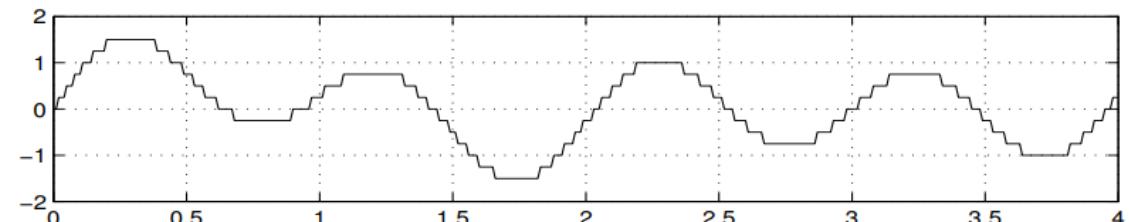
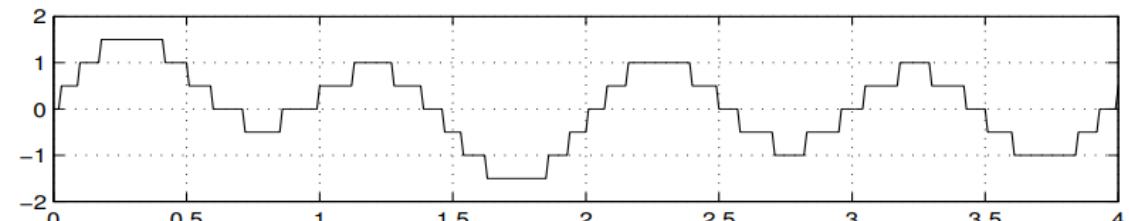
Quantization

- Physical signals are analog. Hence $g(t) \in \mathbb{R}$ and $g[n] \in \mathbb{R}$
- To represent samples $g[n]$ in a digital computer, they must be quantized*
 - Quantized values are chosen from a finite set of symbols \mathcal{Q}
 - The number of symbols L in this set is called the cardinality of the set: $|\mathcal{Q}| = L$
- In general, more quantization levels allow for a better representation of the signal, but also require more bandwidth for transmission

$$|\mathcal{Q}| = 7$$

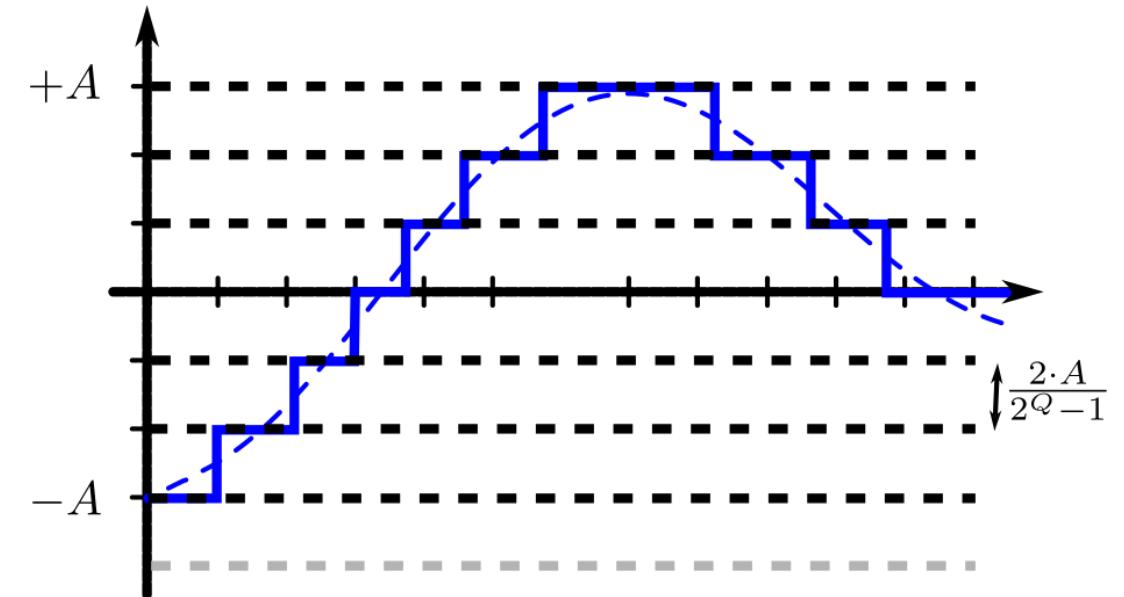


$$|\mathcal{Q}| = 13$$



Uniform Quantization

- Computers typically represent quantized values as sequence of binary digits
 - For efficient binary representation, symbol sets often have power-of-two cardinality $|Q| = 2^Q$
 - For efficient computation, we often use uniform quantization
- Signals with maximum and minimum values $\pm A$ are often DC free and symmetric around zero
 - The range of such signals is $2A$
 - As signals are symmetric, we often use only $2^Q - 1$ from 2^Q available symbols to have also a symmetric quantized range



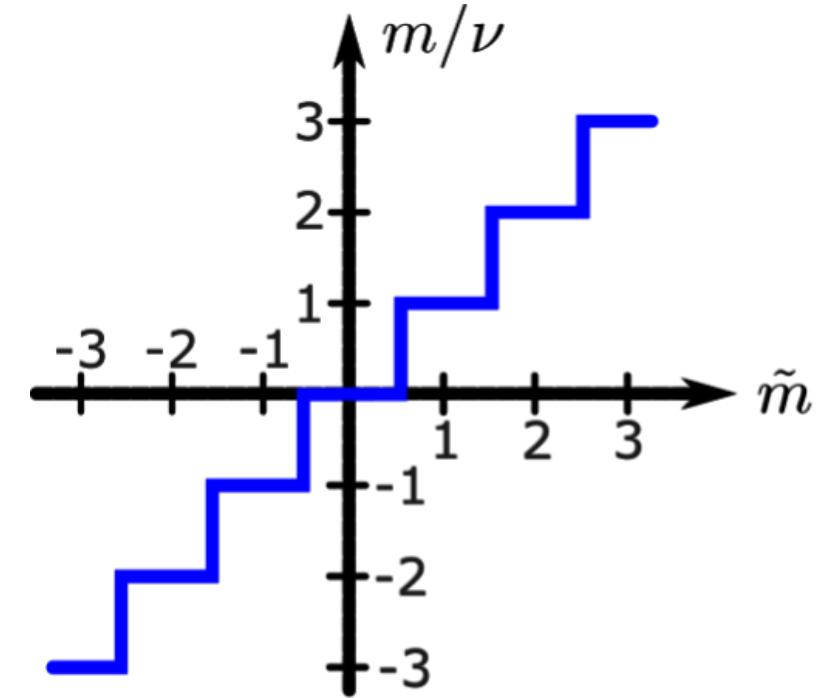
Uniform Symmetric Quantizer for Binary Symbols

- Uniform symmetric quantizer maps values $m[t]$ to signed integers $\tilde{m}[t]$

- Bits per sample: Q
- Effective number of levels: $2^Q - 1$
- Signal range: $\pm A$
- Quantization interval: $\nu = \frac{2 \cdot A}{2^Q - 1}$

$$\tilde{m}[t] = \begin{cases} -N & m[t] < -N\nu \\ n & \left(n - \frac{1}{2}\right)\nu \leq m[t] < \left(n + \frac{1}{2}\right)\nu \\ N & m[t] \geq N\nu \end{cases}$$

$$m[t] \in [-A, +A] \mapsto \tilde{m}[t] \in \left\{ -\frac{2^Q}{2} + 1, \dots, +\frac{2^Q}{2} - 1 \right\}$$



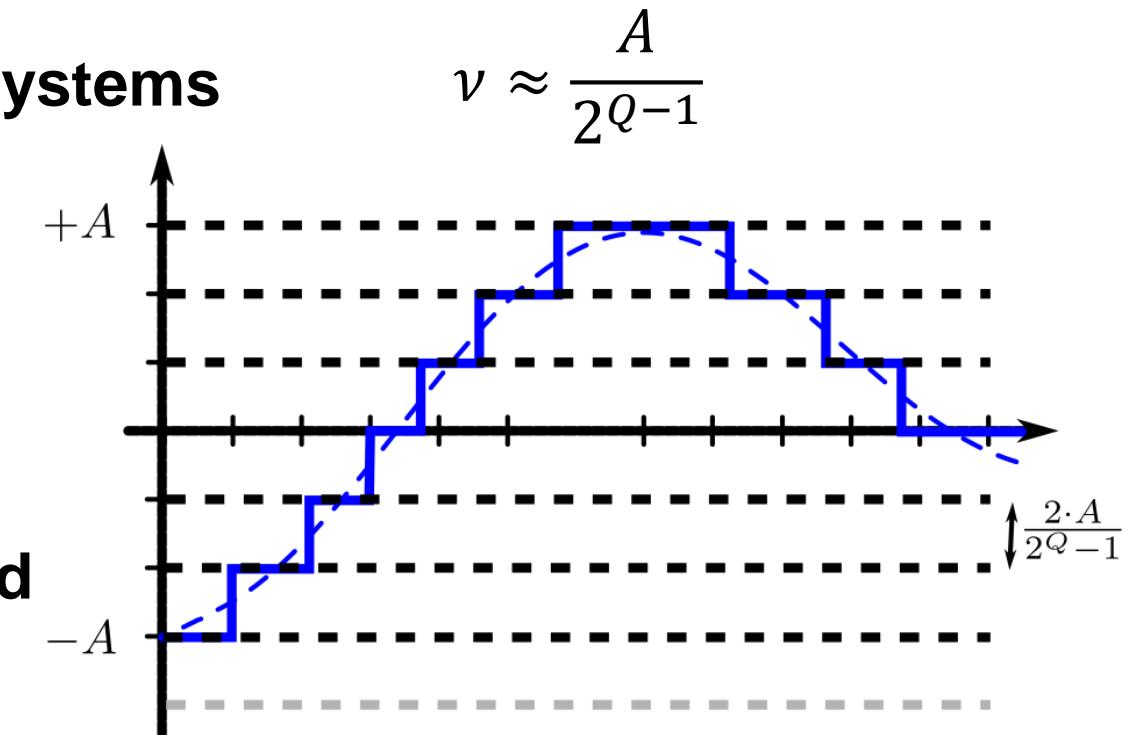
Uniform Quantizer Error (Maximum)

- Quantizers are **nonlinear** time invariant systems

- The maximum quantization error is

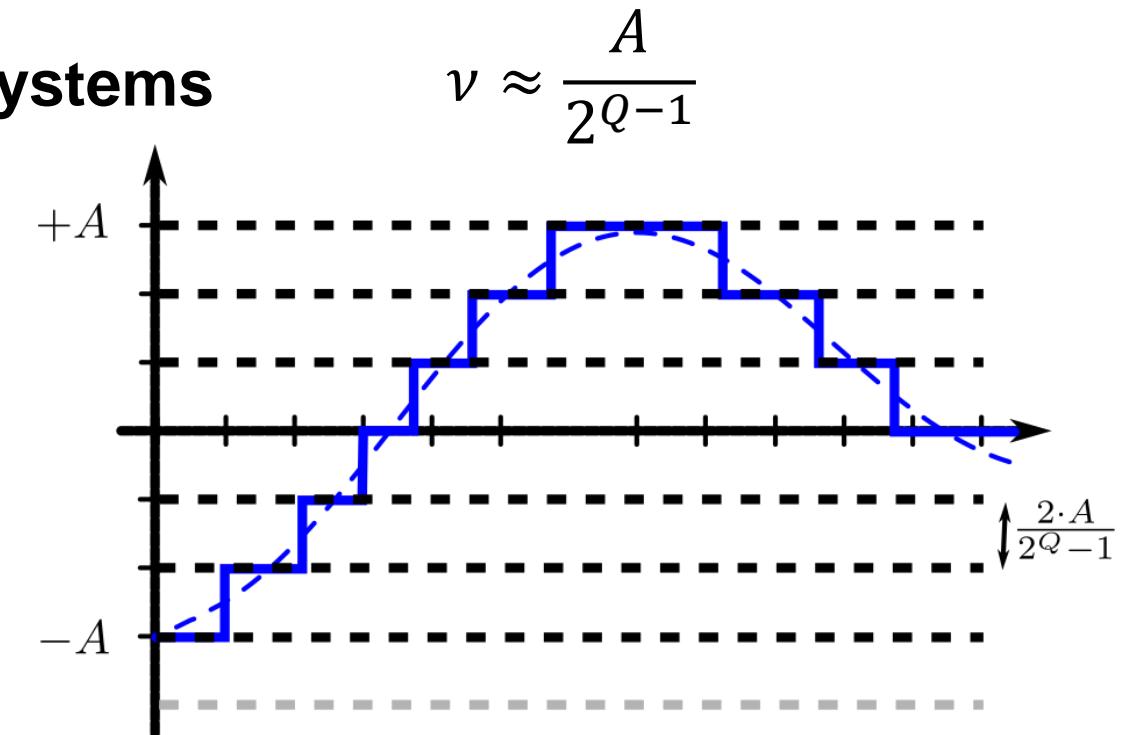
$$\max_m \{ |\tilde{m} \cdot v - m| \} = \frac{A}{2^{Q-1}} \approx \frac{A}{2^Q}$$

- Accurate calculation of the mean squared quantization error is difficult
 - Signal dependent non-linear function



Uniform Quantizer Error (Mean)

- Quantizers are **nonlinear** time invariant systems
- Accurate calculation of the mean squared quantization error is difficult
 - Signal dependent non-linear function
- **Approximation of the mean squared quantization error:**
 - Assume that the signal is uniformly distributed between two quantization levels



$$MSE_Q = \frac{1}{v} \int_{-\frac{v}{2}}^{\frac{v}{2}} |x|^2 dx = \frac{v^2}{12} = \frac{1}{12} \cdot \frac{A^2}{2^{2Q-2}} = \frac{1}{3} \cdot \frac{A^2}{2^{2Q}}$$

Signal-Quantization-Ratio (SQR)

- For quantized signals we are interested in the Signal-to-Quantization Ratio

- Assuming that the signal power is proportional to A^2

- Mean squared quantization error is given by $MSE_Q = \frac{1}{3} \cdot \frac{A^2}{2^{2Q}}$

$$SQR \propto \frac{A^2}{\frac{1}{3} \cdot \frac{A^2}{2^{2Q}}} \propto 2^{2Q}$$

$$SQR_{\text{dB}} = c + Q \cdot \underbrace{2 \cdot \log_{10} 2}_{6\text{dB}} = c + Q \cdot 6\text{dB}$$

Increasing the resolution of the quantizer by 1-bit improves the SQR by 6dB

Signal Quality for Binary PCM

- We assume that the transmission of the PCM signal is error free
- The quality is given by the number of bits we can use for a given channel
- Example: we have the following
 - Audio signal $m(t)$ with bandwidth BW_m → Sampling rate $f_s = 2 \cdot BW_m$
 - Communication channel with bandwidth BW_{ch} → Bit rate $f_{bit} = 2 \cdot BW_{ch}$

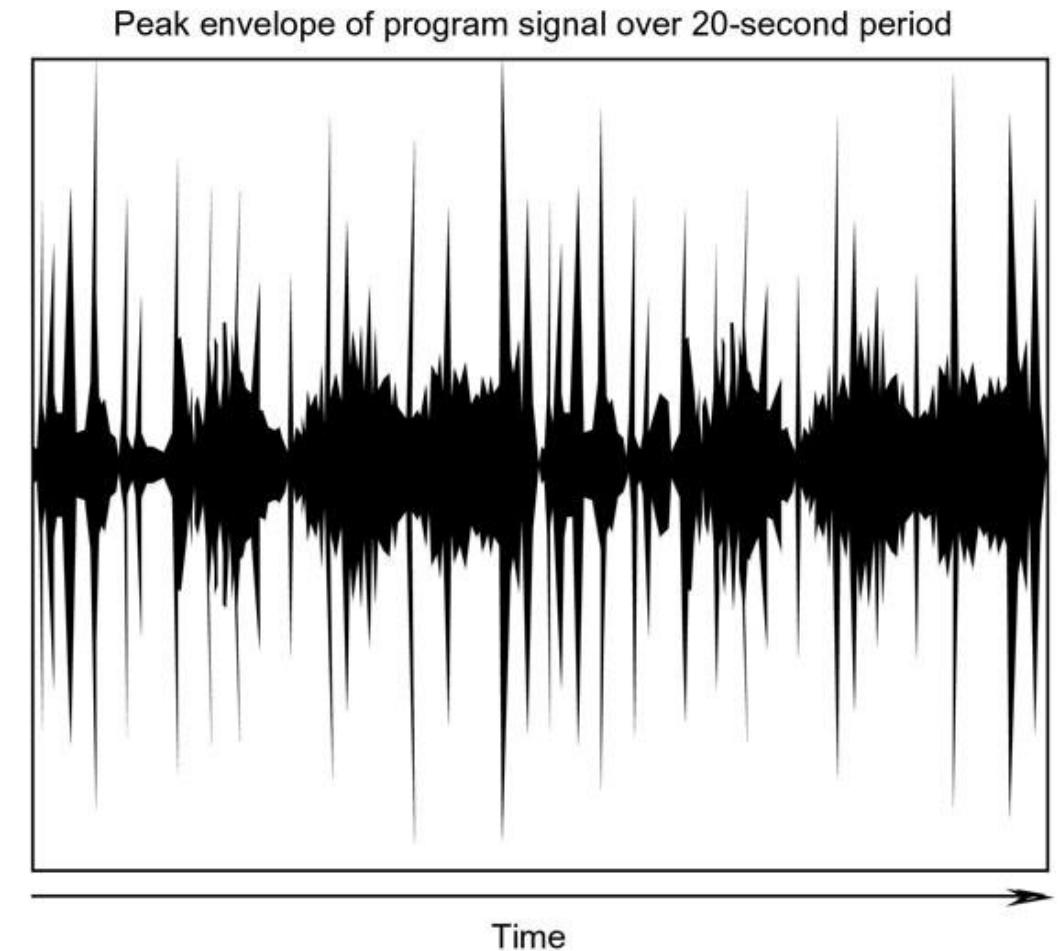
$$Q_{max} = \frac{2 \cdot BW_{ch}}{2 \cdot BW_m} = \frac{BW_{ch}}{BW_m}$$

$$SQR_{dB} = c + \frac{BW_{ch}}{BW_m} \cdot 6dB$$

- Increasing the communication channel BW by BW_m allows to increase the signal quality by 6dB

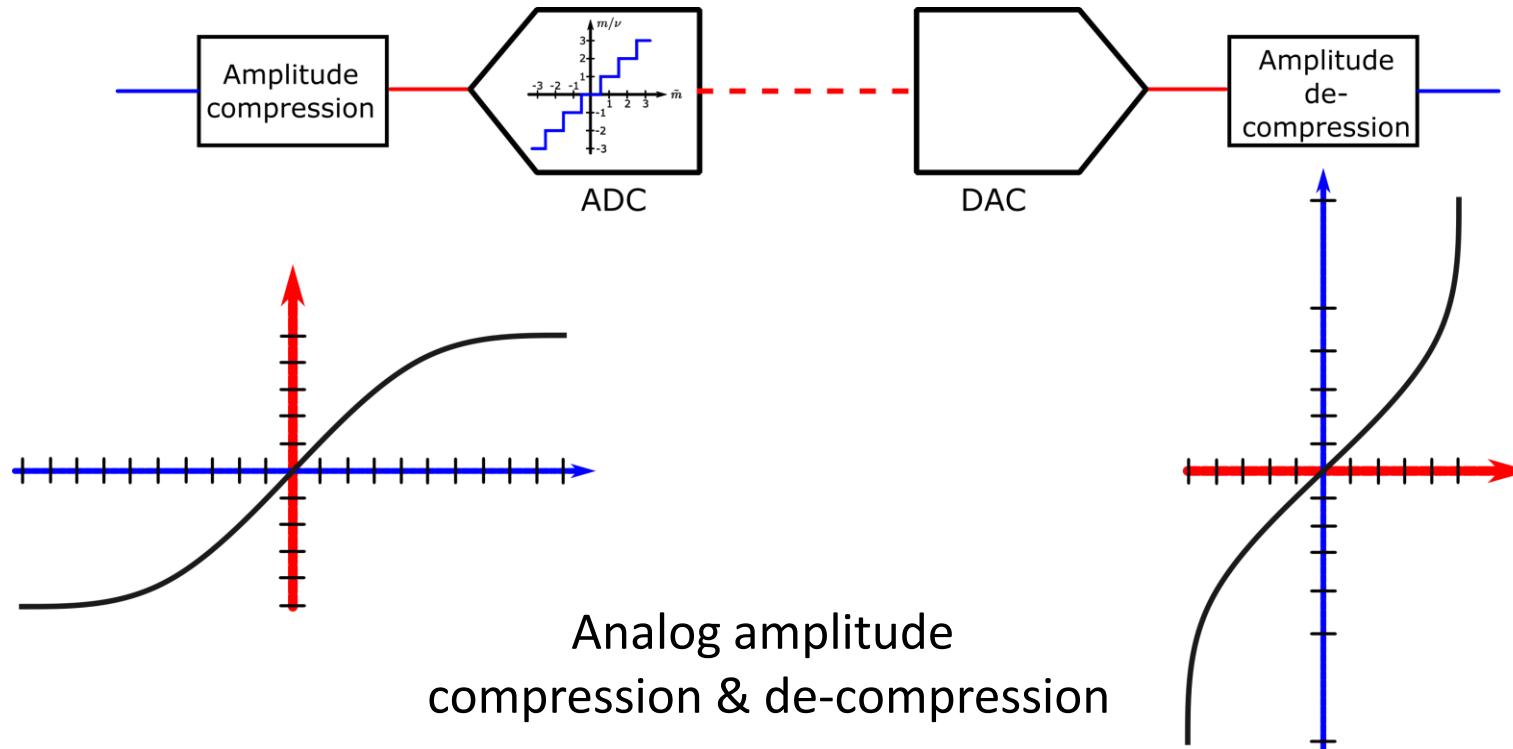
Quantization of Speech Signals

- **Linear quantization step size adapted to peak amplitude**
- **Linear quantization wastes resolution on large amplitudes**, where human hearing is less sensitive
- **Speech signals** are mostly low amplitude
→ **need better resolution near zero**
- **Solution:** Non-uniform quantization



Non-linear Quantization with Amplitude Compression

- **Basic idea:** compress analog signal into a smaller range prior to quantization
 - Analog compression: prior to sampling & quantization
 - Digital compression: first sample with a high resolution, then compress and re-quantize



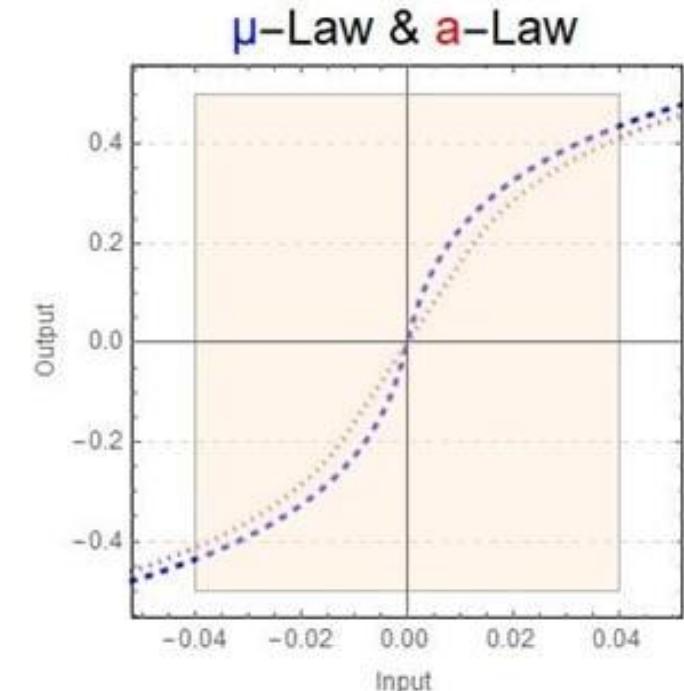
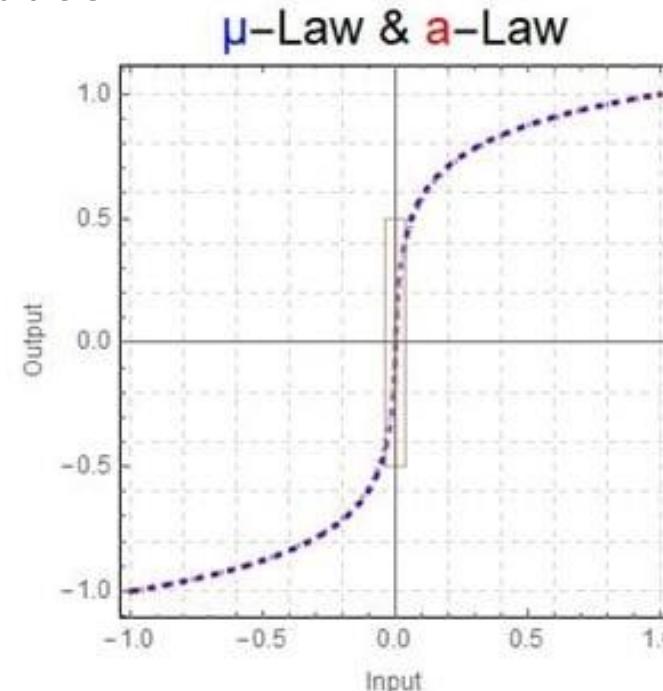
Non-linear Quantization: μ -Law and A-Law

- **Two popular compressions: μ -law and A-law with log-like behaviour**
 - Very similar, especially at high magnitudes
 - Slight differences for low magnitudes
- **U-Law (North America, Japan)**

$$F(x) = \text{sign}(x) \cdot \frac{\ln(1+\mu|x|)}{\ln(1+\mu)}, \mu = 255$$

- **A-Law (Europe, ITU-T global)**

$$F(x) = \begin{cases} \frac{A|x|}{1+\ln A} & |x| < \frac{1}{A} \\ \frac{1+\ln A|x|}{1+\ln A} & \frac{1}{A} \leq |x| \leq 1 \end{cases}, A = 87.6$$



EE-432

Systeme de

Telecommunication

Prof. Andreas Burg
Joachim Tapparel, Yuqing Ren, Jonathan Magnin

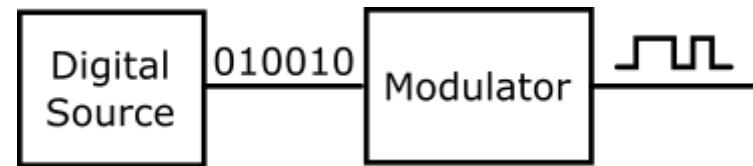
**Digital Data Transmission & Error Rates
in the Baseband**

Week 5: Table of Contents

- **PCM Pulse Shapes and their Bandwidths**
- **Higher Order Digital PAM (M-PAM) for Higher Data Rates**
- **Applications**

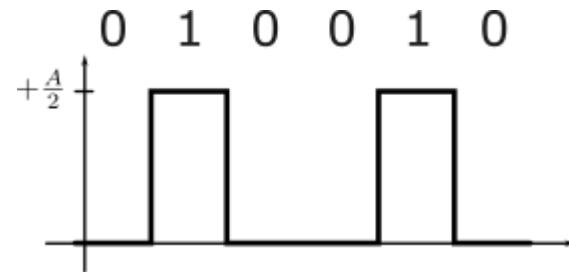
PCM Modulator: Unipolar vs. Bipolar

- PCM encodes bits (0/1) into analog values, chosen from a binary alphabet



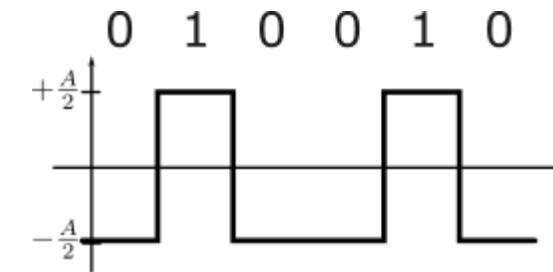
- We distinguish between unipolar (only positive symbols) and bipolar (positive and negative, symmetric symbols) alphabets
 - For reasons we will see later, we keep the “distance” between two symbols the same

Unipolar binary alphabet



$$\{0,1\} \rightarrow \{0, A\}$$

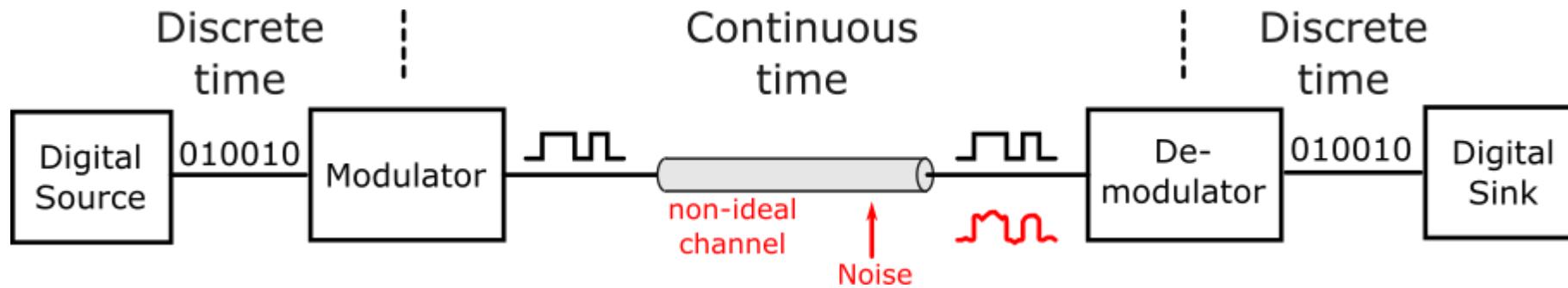
Bipolar binary alphabet



$$\{0,1\} \rightarrow \left\{ -\frac{A}{2}, +\frac{A}{2} \right\}$$

PCM Modulator: Pulse Shapes

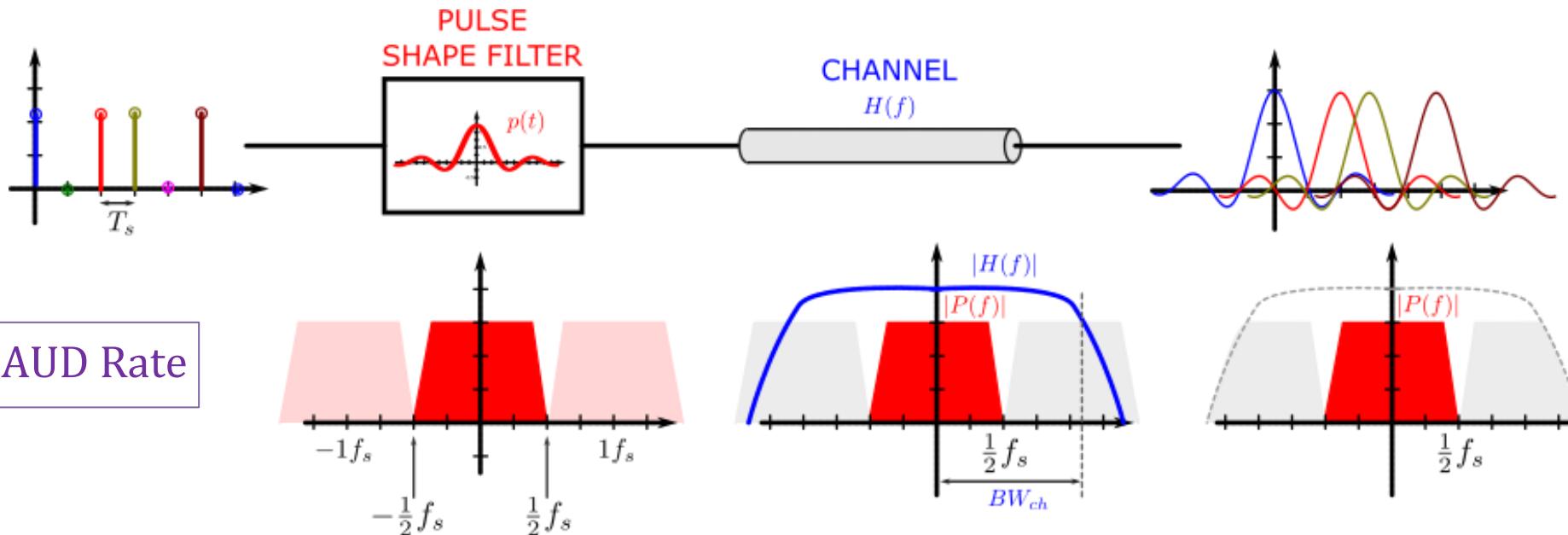
- PCM transmits a series of bits over a continuous time / analog channel**
 - Transmitter: convert discrete time bits into bits as a series of continuous time pulses
 - Receiver: Recover the discrete time bits from the continuous time pulses through sampling



- Transmission channel: distorts the signal by**
 - its typically low-pass frequency characteristics
 - adding noise (actually added at the receiver, but considered part of the channel)

PCM Modulator: Pulse Shapes

- Conversion of discrete bits into a continuous time signal of “pulses” so that
 - are not or almost not affected by the low-pass frequency characteristics of the channel
 - can be recovered easily (independently of each other) at the receiver



- Choose **pulse-shape filter** and **baud rate (sampling frequency)** so that the fundamental spectrum of the signal is not affected by the low-pass channel

$$f_s \leq 2 \cdot BW_{ch}$$

PCM Modulator: Ideal SINC Pulse

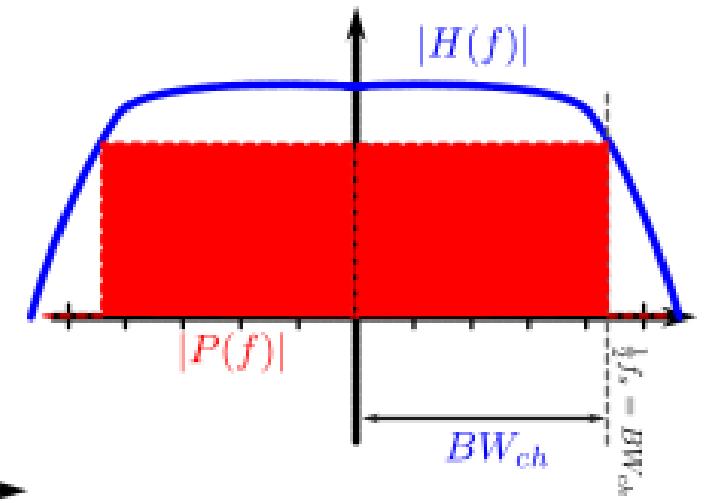
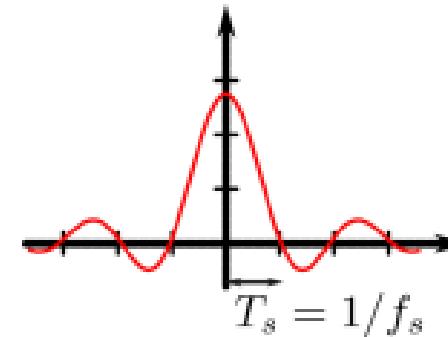
- For $f_s = 2 \cdot BW_{ch}$ we need an ideal (brick-wall) pulse shape filter with the bandwidth of the channel

$$P(f) = \prod \left(\frac{f}{2 \cdot BW_{ch}} \right) = \prod \left(\frac{f}{f_s} \right)$$

- Pulse shape impulse response:

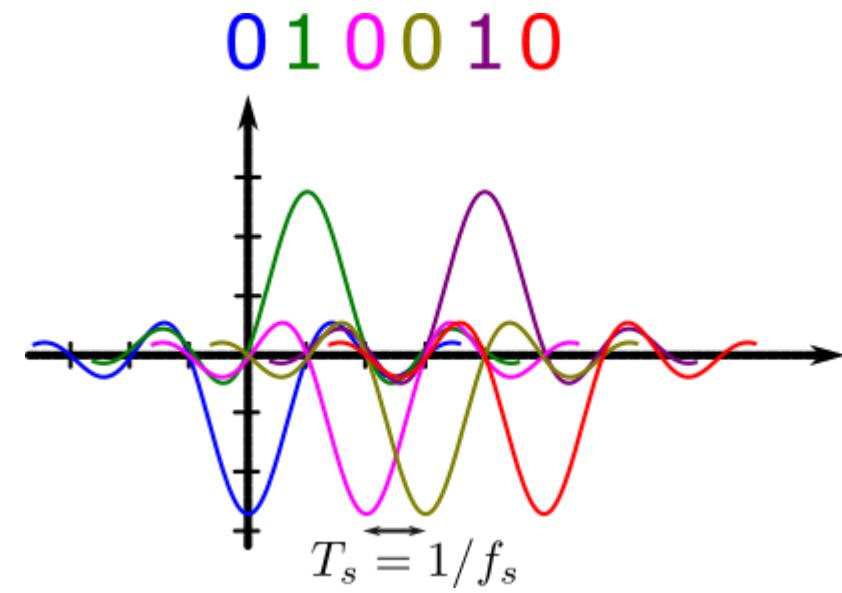
$$p(t) = \frac{\sin(\pi \cdot f_s \cdot t)}{\pi \cdot f_s \cdot t}$$

- Data sequence with bipolar alphabet: $a_k \in \left\{ -\frac{A}{2}, +\frac{A}{2} \right\}$



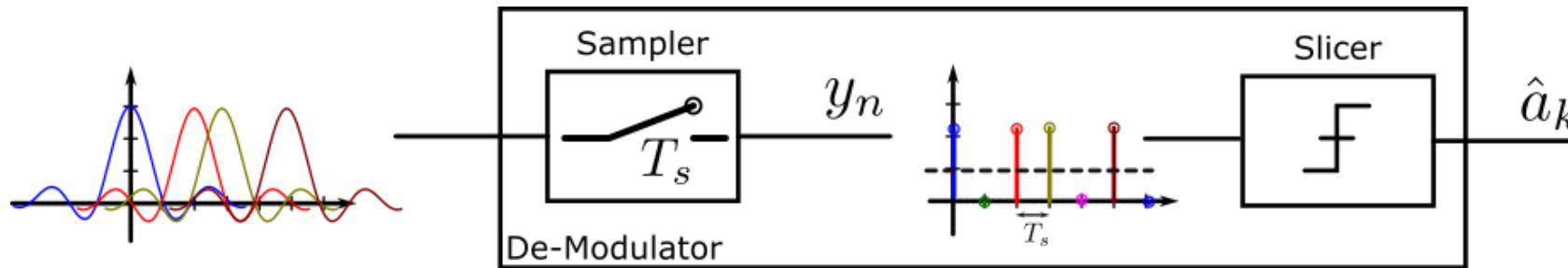
- Received signal is the superposition of weighted transmit pulses delayed by $k \cdot T_s$

$$y(t) = \sum_{k=-\infty}^{+\infty} a_k \cdot p(t - k \cdot T_s)$$

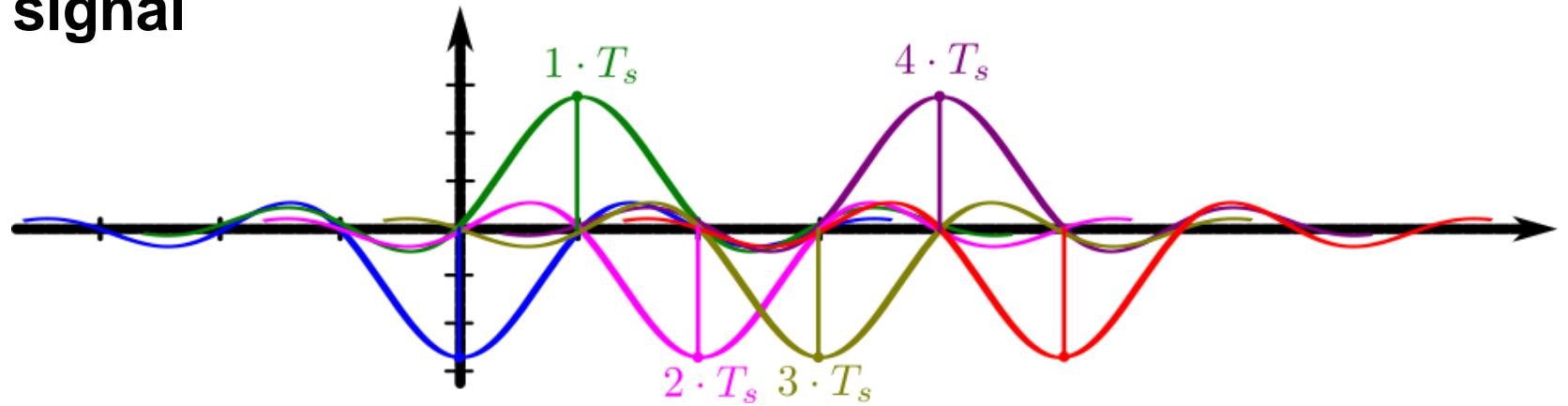


PCM Demodulation with SINC Pulse

- Despite the overlap between Nyquist pulses, we can perfectly recover the original transmitted symbols by sampling at the right moment in time



- At $t = n \cdot T_s$, all overlapping symbols except one are zero and have no impact on the received signal



PCM Demodulation with SINC Pulse

- Despite the overlap between Nyquist pulses, we can perfectly recover the original transmitted symbols by sampling at the right moment in time
- Proof:

- Remember: $\frac{\sin(k \cdot \pi)}{k \cdot \pi} = \begin{cases} 1 & k = 0 \\ 0 & k \neq 0 \end{cases}$

$$y_k = y(t) \Big|_{t=k \cdot T_s} = \sum_{l=-\infty}^{+\infty} a_k \cdot \frac{\sin(\pi \cdot f_s \cdot (n - l) \cdot T_s)}{\pi \cdot f_s \cdot (n - l) \cdot T_s} =$$
$$+ a_k \cdot \underbrace{\frac{\sin(\pi \cdot f_s \cdot (0) \cdot T_s)}{\pi \cdot f_s \cdot (0) \cdot T_s}}_1 + \sum_{l \neq k} a_k \cdot \underbrace{\frac{\sin(\pi \cdot f_s \cdot (l - k) \cdot T_s)}{\pi \cdot f_s \cdot (l - k) \cdot T_s}}_0$$

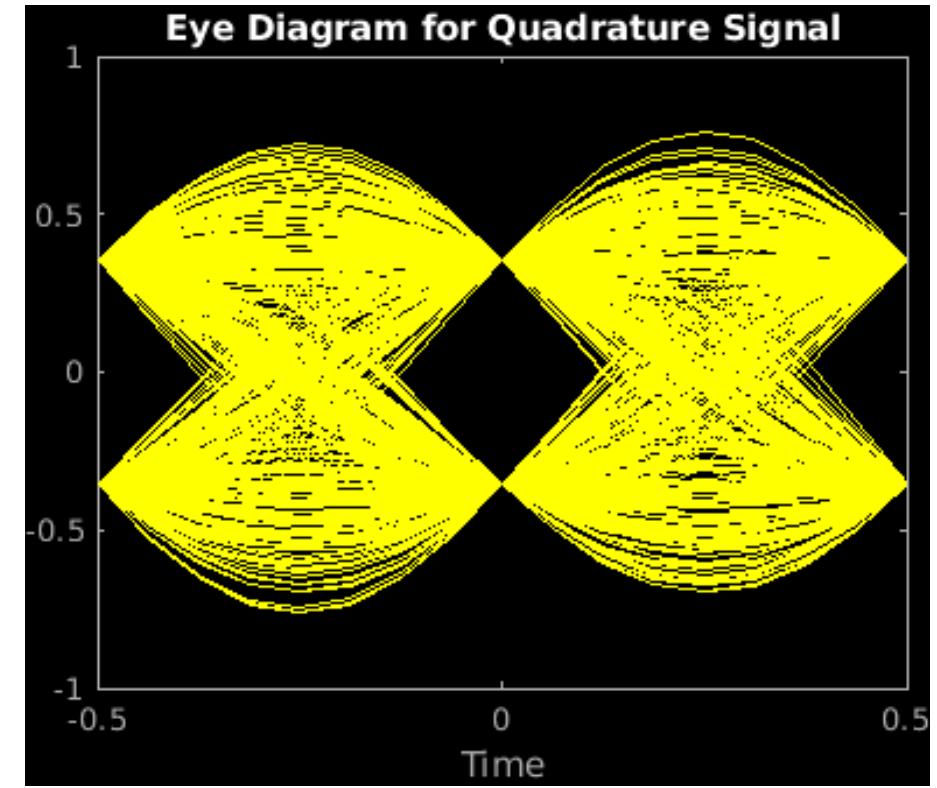
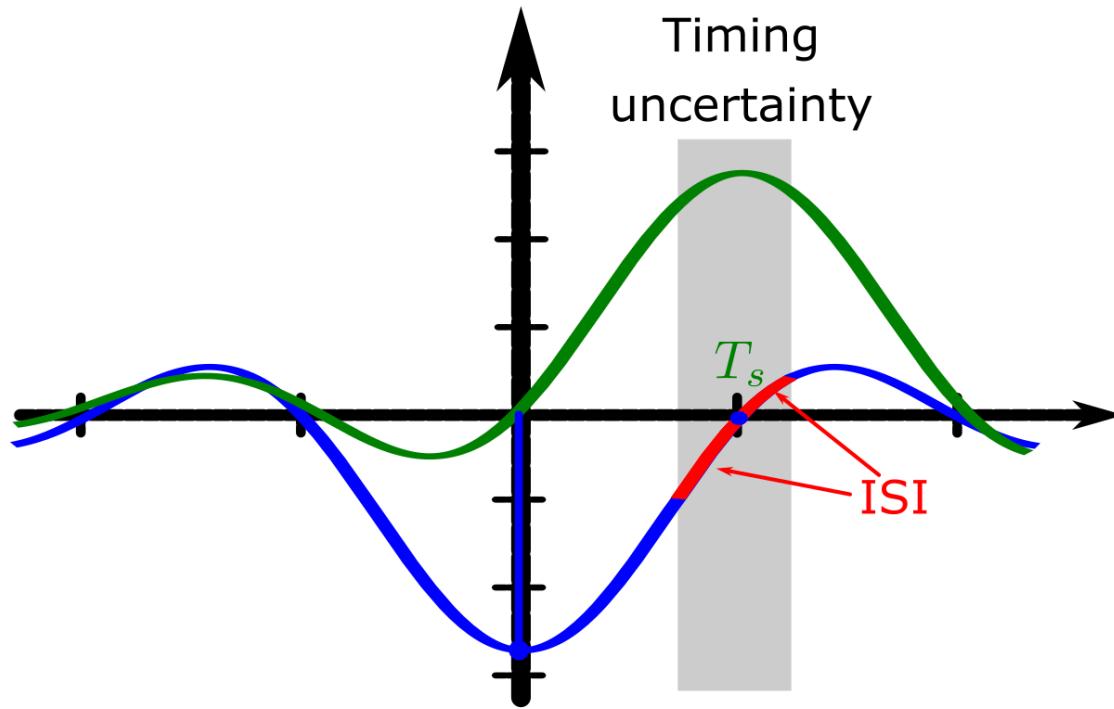
PCM with Non-Ideal SINC Pulses

- **A SINC uses the available bandwidth in the best possible (optimal) way:**
 - For a given baud rate f_s , it uses the least possible bandwidth $BW_{ch} \geq f_s/2$
 - For a given channel bandwidth BW_{ch} it allows for the highest baud rate $f_s \leq 2 \cdot BW_{ch}$
- **Nevertheless, SINC pulses also have a practical disadvantage:**
 1. Any non-ideal sampling instant at the receiver leads to strong interference between symbols
 2. Cutting the tail impacts the spectrum (no longer an ideal brick-wall with strong spectral components far away from the intended bandwidth)

See next 2 slides

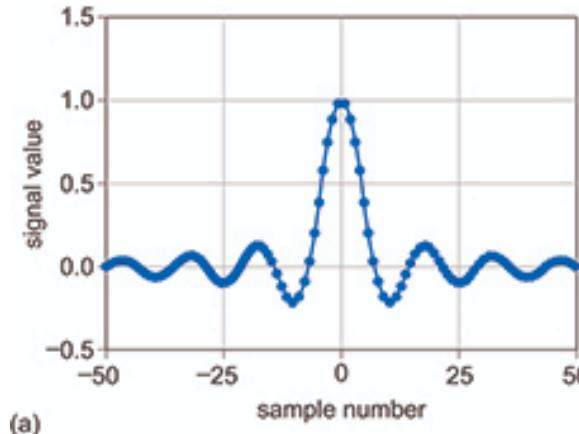
PCM with Non-SINC (non-ideal) Pulses

1. Any non-ideal sampling instant at the receiver leads to strong interference between symbols

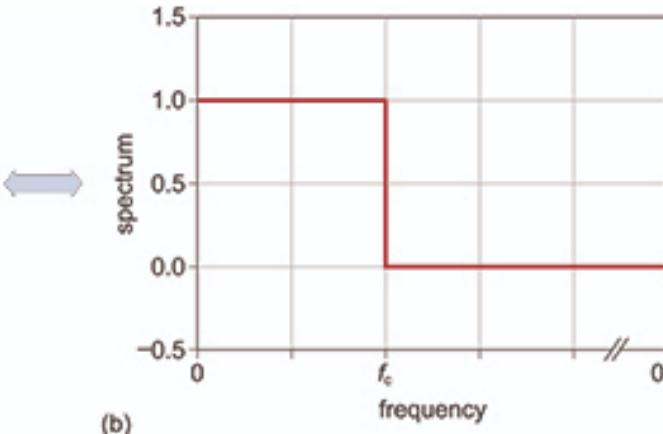


PCM with Non-SINC (non-ideal) Pulses

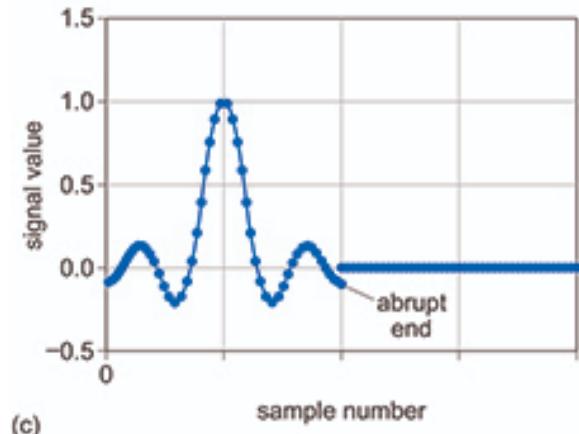
2. Cutting the tail impacts the spectrum (no longer an ideal brick-wall with strong spectral components far away from the intended bandwidth)



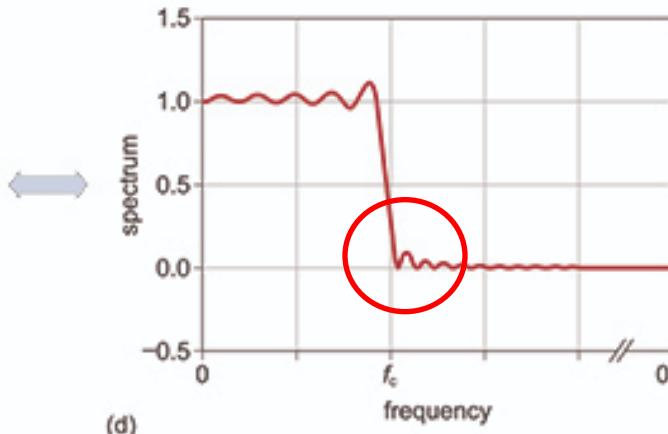
(a)



(b)



(c)



(d)

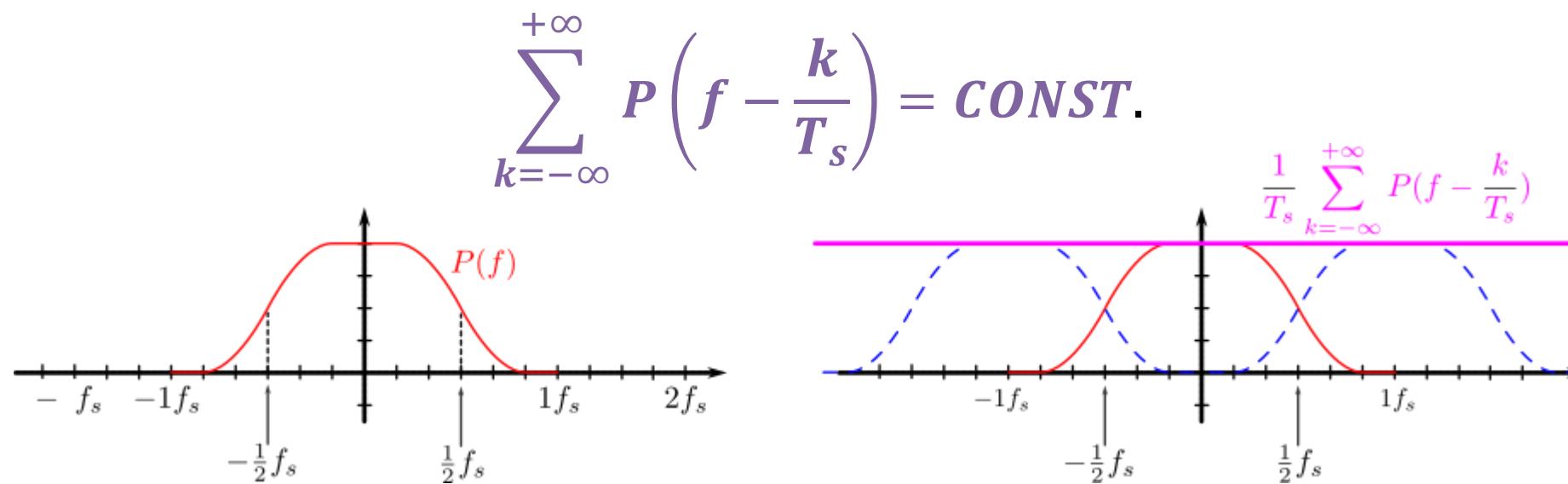
Source: <https://www.open.edu/openlearn/science-maths-technology/electronic-applications/content-section-3.7>

Alternative (Non-Sinc) Pulse Shapes

- To achieve ISI free transmission, we actually only require:

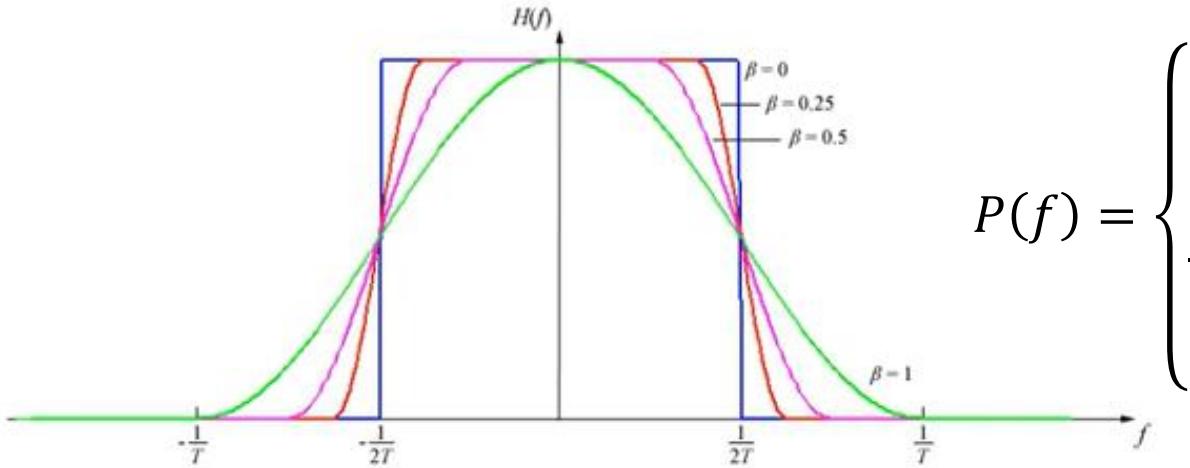
$$p(k \cdot T_s) = \begin{cases} 1 & k = 0 \\ 0 & k \neq 0 \end{cases}$$

- Any pulse with this criterion should be able to perfectly reconstruct the symbols a_k at $t = k \cdot T_s$
- The 2nd Nyquist criterion defines such non-SINC pulses, that are ISI free, but have a BW that is greater than half the sampling frequency $BW_p > \frac{f_s}{2}$

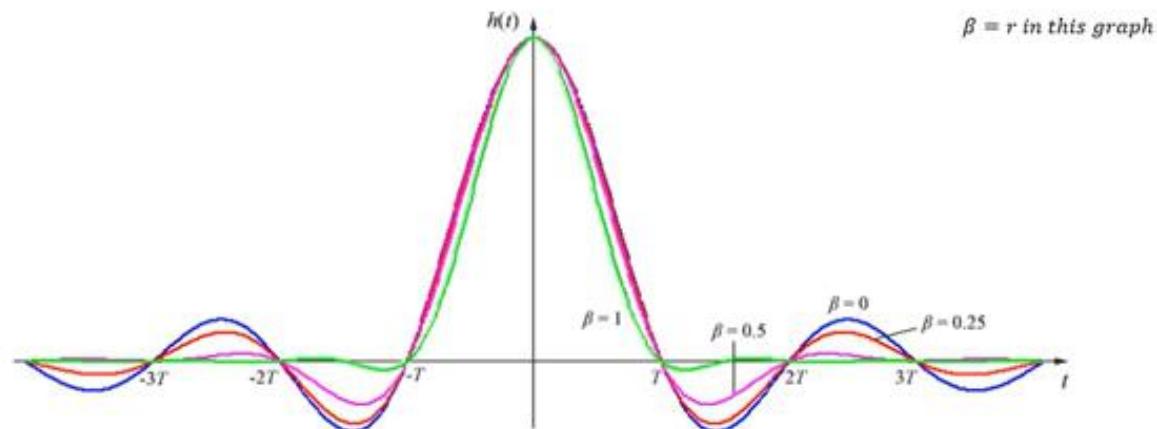


Raised Cosine Pulse (1)

- The most famous Non-SINC pulse is the Raised Cosine Pulse



$$P(f) = \begin{cases} 1 & |f| < \frac{1-\beta}{2T_s} \\ \frac{1}{2} \left[1 + \cos \left(\frac{\pi T_s}{\beta} \left[|f| - \frac{1-\beta}{2T_s} \right] \right) \right] & \frac{1-\beta}{2T_s} < |f| \leq \frac{1+\beta}{2T_s} \\ 0 & \text{else} \end{cases}$$

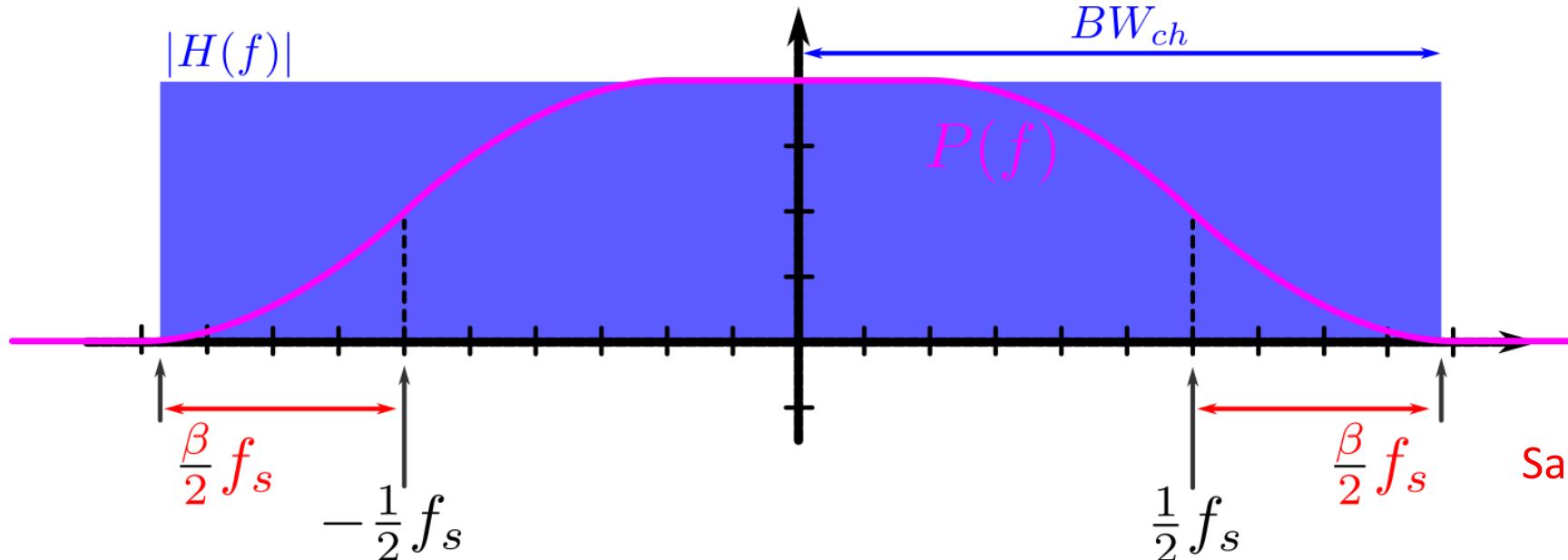


$$p(f) = \begin{cases} \frac{\pi}{4T_s} \operatorname{sinc} \frac{1}{2\beta} & t = \pm \frac{T}{2\beta} \\ \frac{1}{T_s} \operatorname{sinc} \frac{t}{T_s} \frac{\cos \left(\frac{2\beta t}{T_s} \right)}{1 - \left(\frac{2\beta t}{T_s} \right)^2} & \text{else} \end{cases}$$

Raised Cosine Pulse (2)

- For $|f| > \frac{1+\beta}{2T_s} = \frac{1}{2T_s} + \frac{\beta}{2T_s}$, there are no spectral components

- The first part $\frac{1}{2T_s}$ corresponds to the spectrum used by the ideal SINC filter
- The second part $\frac{\beta}{2T_s}$ corresponds to the **excess spectrum**



$$\frac{1 + \beta}{2} f_s < BW_{ch}$$

$$f_s < \frac{2 \cdot BW_{ch}}{1 + \beta}$$

Sampling rate with a band-limited channel and Raised Cosine β

Relationship Between Baud Rate and Bandwidth

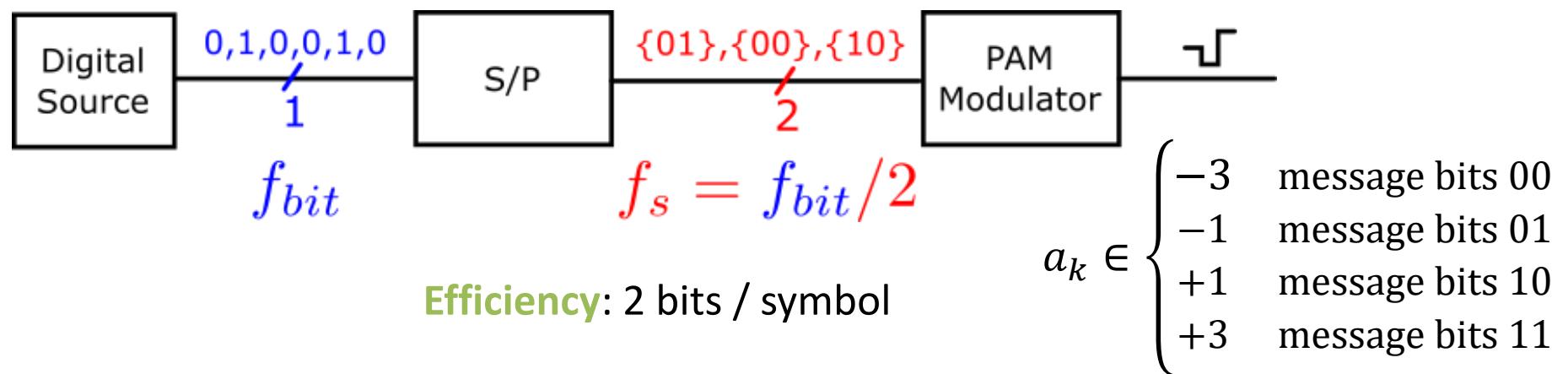
	Maximum baud rate f_s , given a channel bandwidth BW_{ch}	Occupied spectrum BW given a baud rate f_s
Theoretical	$f_s = 2 \cdot BW_{ch}$	$BW = \frac{1}{2} f_s$
Realistic (rolloff β^*)	$f_s = \frac{2}{1 + \beta} \cdot BW_{ch}$	$BW = \frac{1 + \beta}{2} f_s$

* Typical rolloff factors are $\beta = 0.6 - 0.8$

BW is the bandwidth that is occupied by the signal, regardless of if the channel supports it

Higher-Order Pulse Amplitude Modulation (M-PAM)

- For a given bandwidth, the data rate with binary PCM is limited by the bandwidth as we send only 1 bit for every “symbol” ($a_k \in \left\{-\frac{A}{2}, +\frac{A}{2}\right\}$)
- Higher order symbol alphabets: grouping of multiple bits into one symbol
 - Grouping of bits represented as serial-to-parallel (S/P) conversion
- Example 4-PAM: grouping of 2 bits into one symbol with bit rate f_{bit} [bits/s]



M-PAM Constellation Points & Alphabets

- Grouping Q bits into one symbol increases the throughput Q -times, but requires an alphabet with $M = 2^Q$ elements (levels)

Information throughput per symbol

$$I_M = Q = \log_2 M$$

- For a given baud-rate f_s we have $f_{bit} = I_M \cdot f_s$
- Values that represent a combination of bits are called **constellation points**
- The set of constellation points \mathcal{O} to represent groups of $Q = \log_2 M = \log_2 |\mathcal{O}|$ bits is called a **constellation alphabet**

$$\mathcal{O} = \{x_0, x_1, x_2, \dots, x_{M-1}\}$$

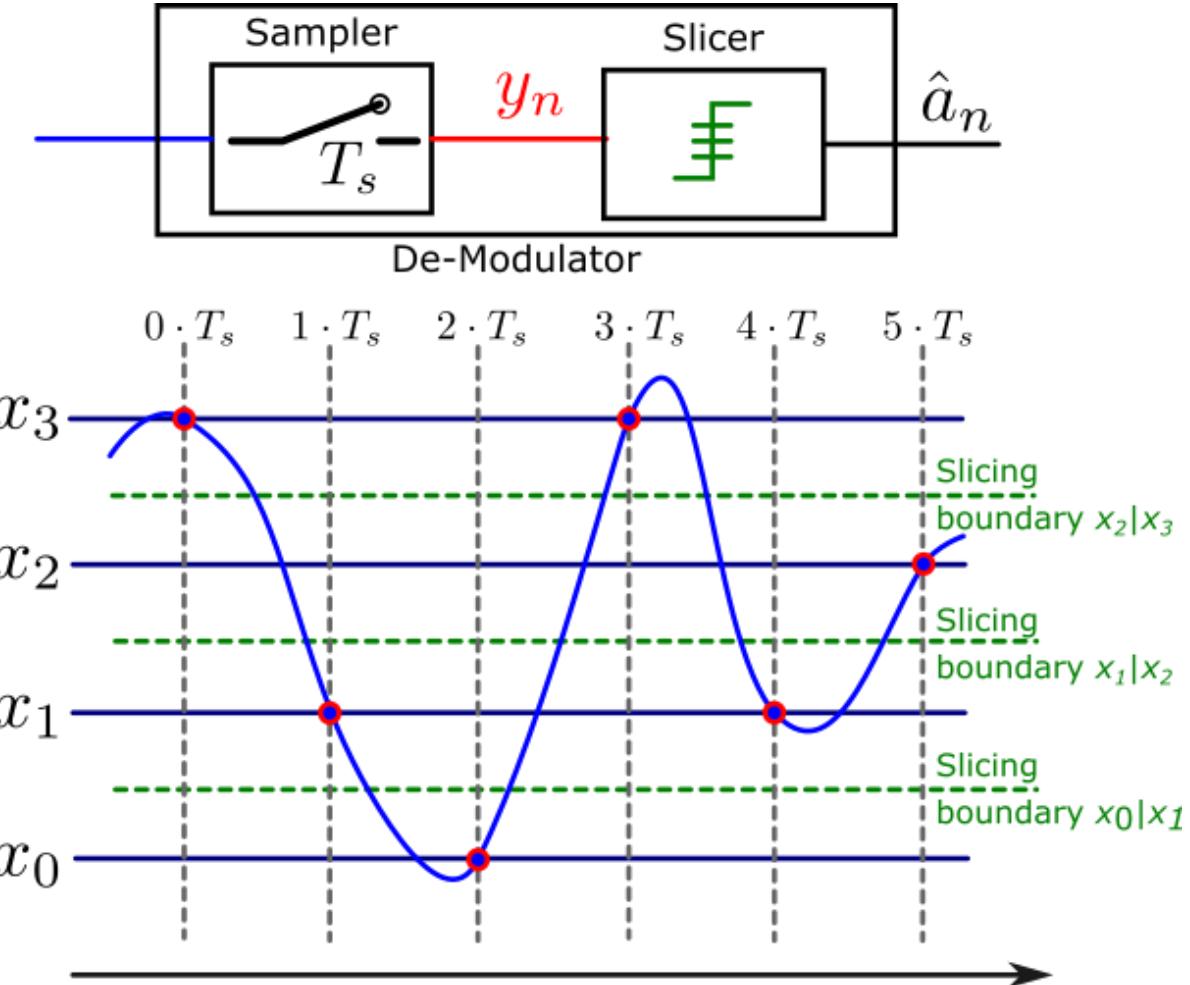
M-PAM Signal Reception w/o Noise

- Consider a 4-PAM signal (received without noise and without attenuation)

- Demodulation

- Sampler:** sample signal at the correct time instant to reveal the transmitted symbol without interference
- Slicer:** decide on the closest valid constellation point based on decision boundaries

$$a_m = k \text{ if } \left(y_n > \frac{x_m + x_{m-1}}{2} \right) \text{ and } \left(y_n < \frac{x_m + x_{m-1}}{2} \right)$$
$$x_{-1} = -\infty ; x_M = +\infty$$



M-PAM Signal Power

- **As symbols are different, the power of each symbol may also be different**
 - Consider the power of the sampled signal: $P_n = |a_n|^2$
- **In general, we are mostly interested in the average signal power \bar{P}_s**
 - For a specific signal example (but with many samples)

$$\bar{P}_s = \lim_{N \rightarrow \infty} \frac{1}{N} \sum_{k=0}^{N-1} |a_k|^2$$

- To be more general, it is reasonable to assume that the symbols $a_k \in \mathcal{O}$ are all equally likely

$$\Pr(a_k = x_i) = \frac{1}{N} \text{ for all } i = 0, \dots, M - 1$$

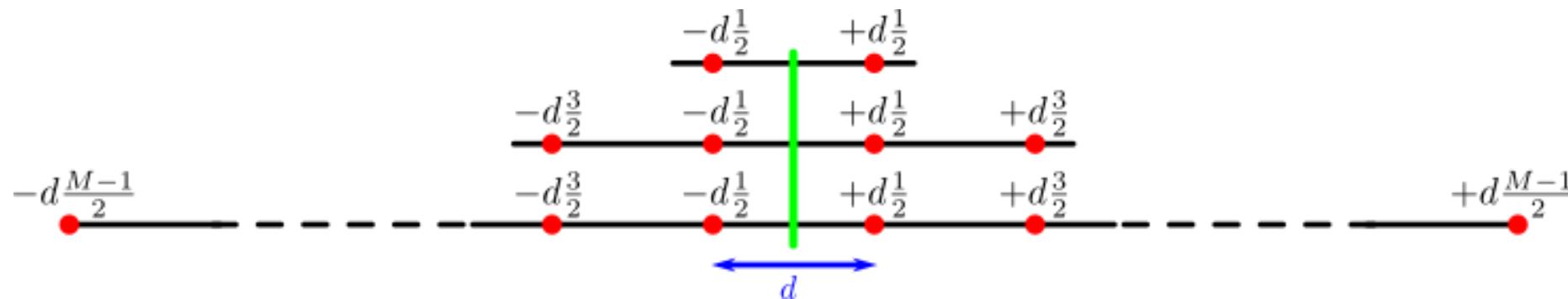
$$\bar{P}_{\mathcal{O}} = \frac{1}{M} \sum_{m=0}^{M-1} |x_m|^2$$

M-PAM Constellation Construction

- **Constructing a constellation involves two steps:**
 - Defining the values of the constellation points $\mathcal{O} = \{x_0, x_1, x_2, \dots, x_{M-1}\}$
 - Assigning a combination of bits to each constellation point
- **There are two strategies to define the values of the constellation points**
 - **Equal Distance Construction:** constellation points have same **distance d** from their neighbours

$$x_{n+1} = x_n + d, n = 0, \dots, M-1, \text{ given } x_0$$

$$\text{Example: } x_0 = -\frac{M-1}{2} \rightarrow \mathcal{O} = \left\{ \pm d \frac{1}{2}, \pm d \frac{3}{2}, \dots, \pm d \frac{M-1}{2} \right\}$$



M-PAM Constellations Construction: Unit Power

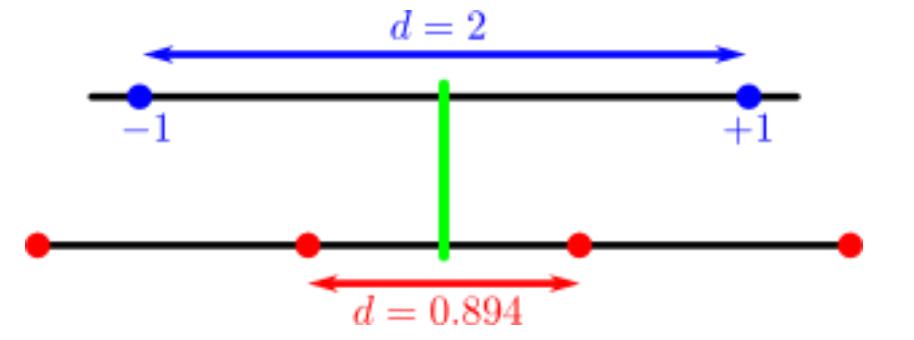
- **Wireless systems:** “average power” constrained by standards & regulations
- **Unit Power** Construction: constellation points yield signal power of one $\bar{P}_s = 1$ while maintaining equal distance (defined by given power)
 1. Start from equal distance construction with arbitrary distance d

$$x_{n+1} = x_n + d, n = 0, \dots, M - 1, \text{ given } x_0$$

$$\bar{P}_o = \frac{1}{M} \sum_{m=0}^{M-1} |x_m|^2$$

2. Scale constellation points to achieve the desired average power level

$$\bar{x}_k = \frac{x_k}{\sqrt{\bar{P}_o}} \quad ; \quad \bar{\mathcal{O}} = \{\bar{x}_0, \bar{x}_1, \bar{x}_2, \dots, \bar{x}_{M-1}\}$$



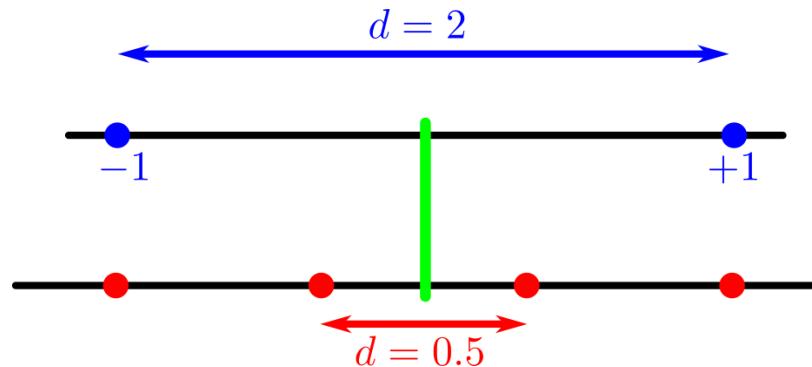
M-PAM Constellations Construction: Max. Amplitude

- **Wireline systems: circuit supply voltages define maximum amplitude**
- **Maximum Amplitude** Construction: constellation points yield signal power of one $\bar{P}_s = 1$ while maintaining equal distance (defined by given power)
 1. Start from equal distance construction with arbitrary distance d

$$x_{n+1} = x_n + d, n = 0, \dots, M - 1, \text{ given } x_0$$

2. Scale constellation points to achieve the desired average power level

$$\bar{x}_k = \frac{A}{M - 1} x_k ; \quad \mathcal{O} = \{\bar{x}_0, \bar{x}_1, \bar{x}_2, \dots, \bar{x}_{M-1}\}$$



EE-432

Systeme de

Telecommunication

Prof. Andreas Burg
Joachim Tapparel, Yuqing Ren, Jonathan Magnin

Applications

Applications of PCM Systems: Wireline



Telephone



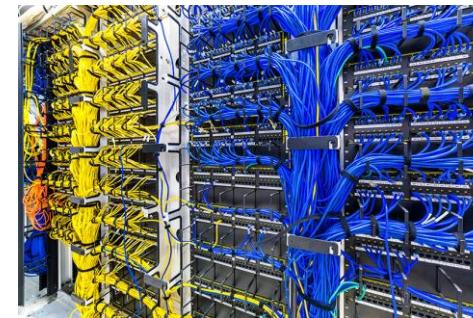
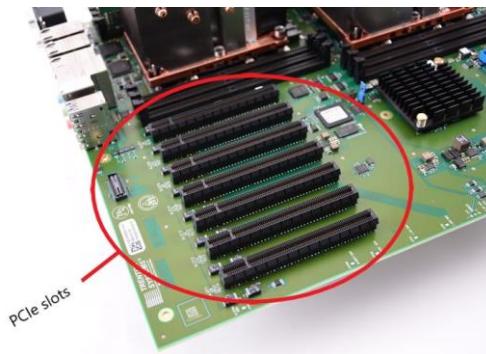
USB



HDMI, Display Port, ...



Memory Interfaces

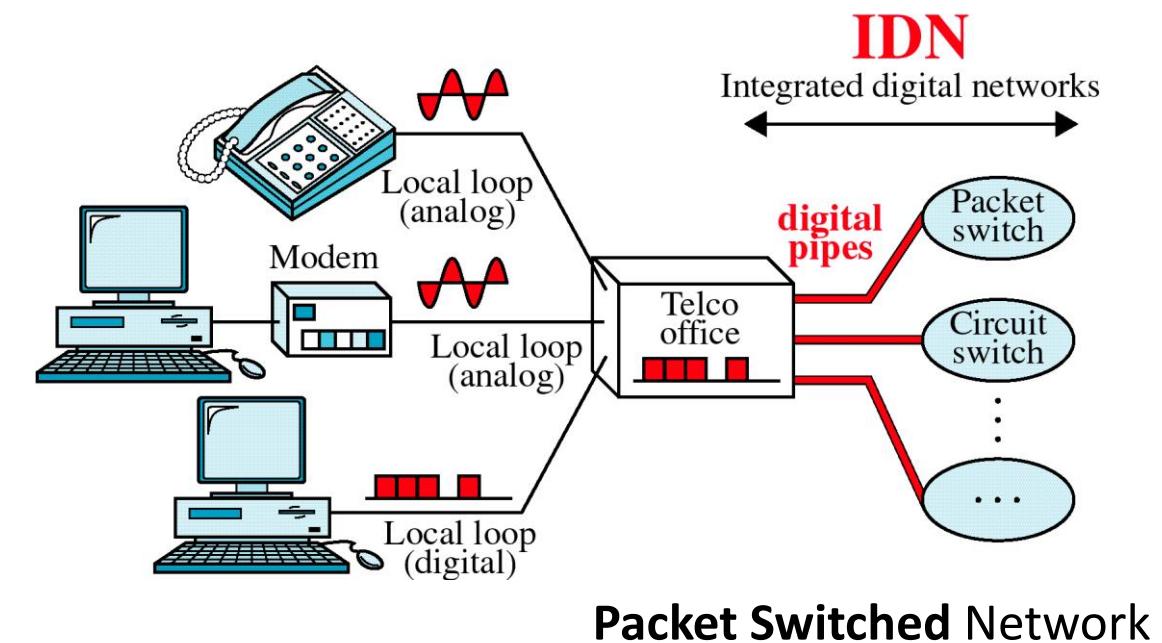
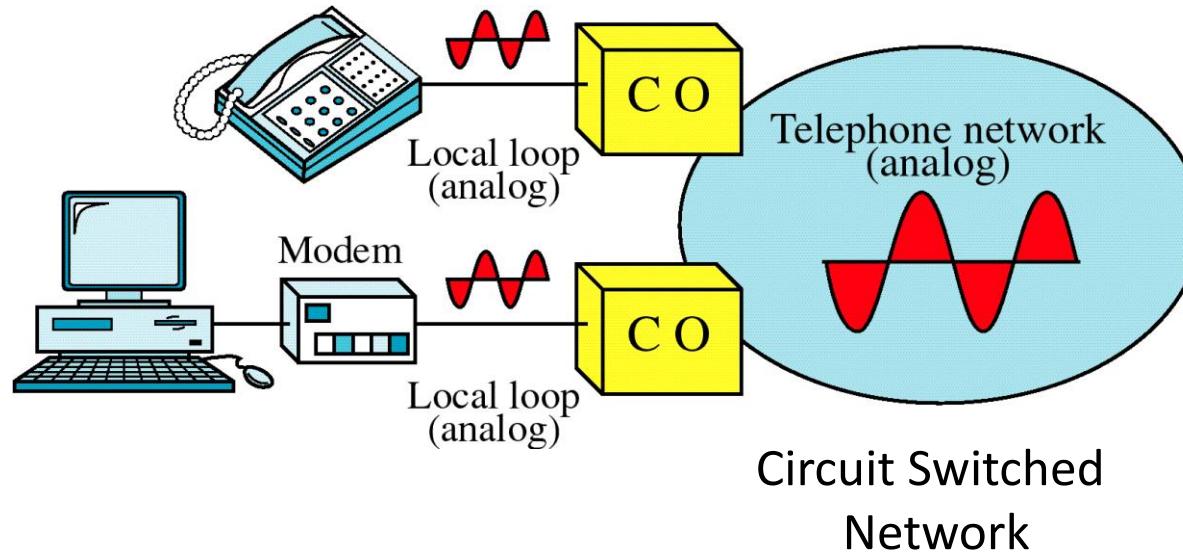


Computer Networks

ISDN Digital Telephony Motivation

- **Motivation in the late 1980s**

- Increasing digitization of the global telephone network (with only local loop analog)
- Increasing need for digital data communication over analog telephone network

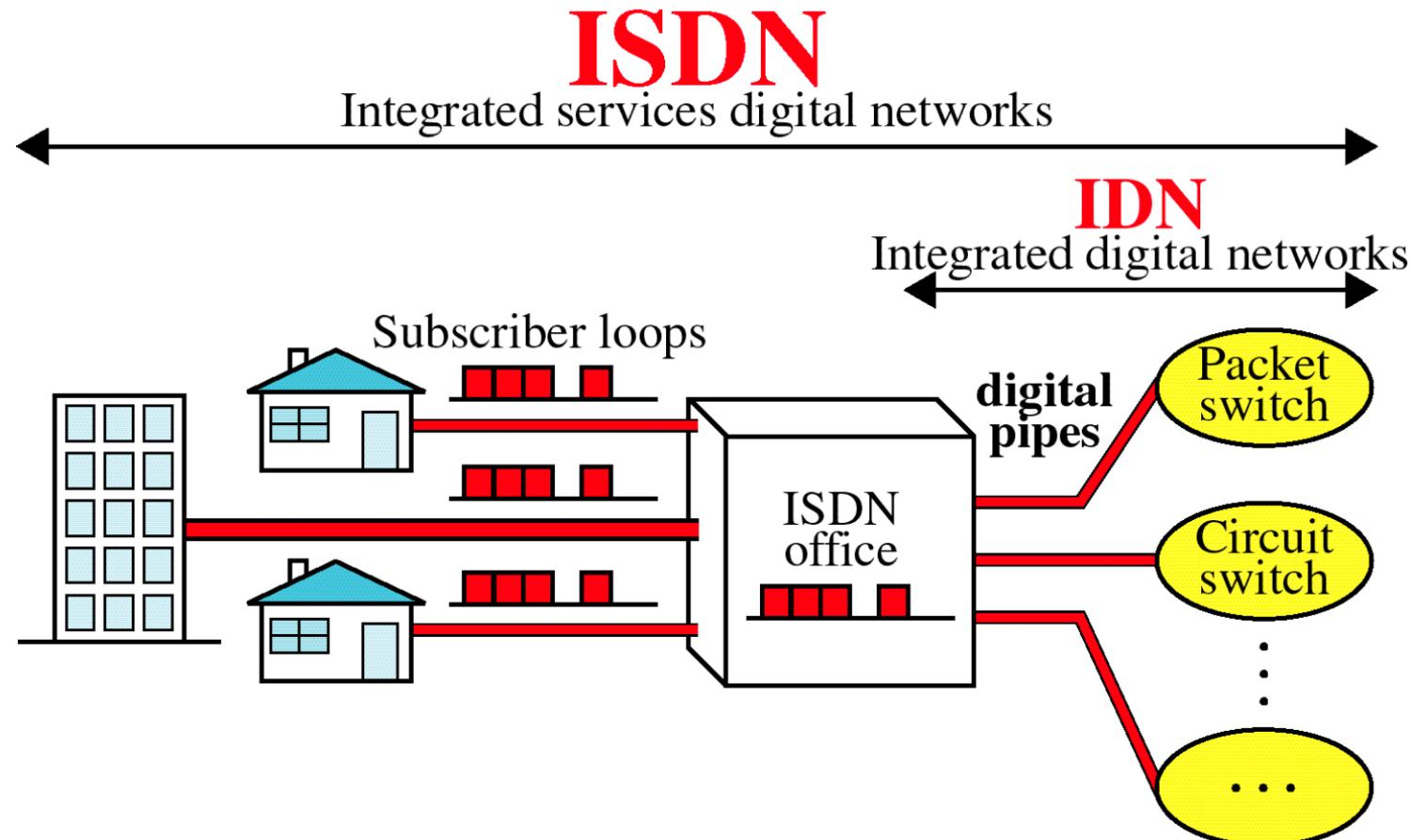


- **Transition from analog local loop (end-user connection) to telephone network was inefficient. Need for new fully digital system**

ISDN Digital Telephony

- **ISDN: Integrated Services Digital Network**

- Proposed in 1984, standardization completed in 1988, commercially available since 1992

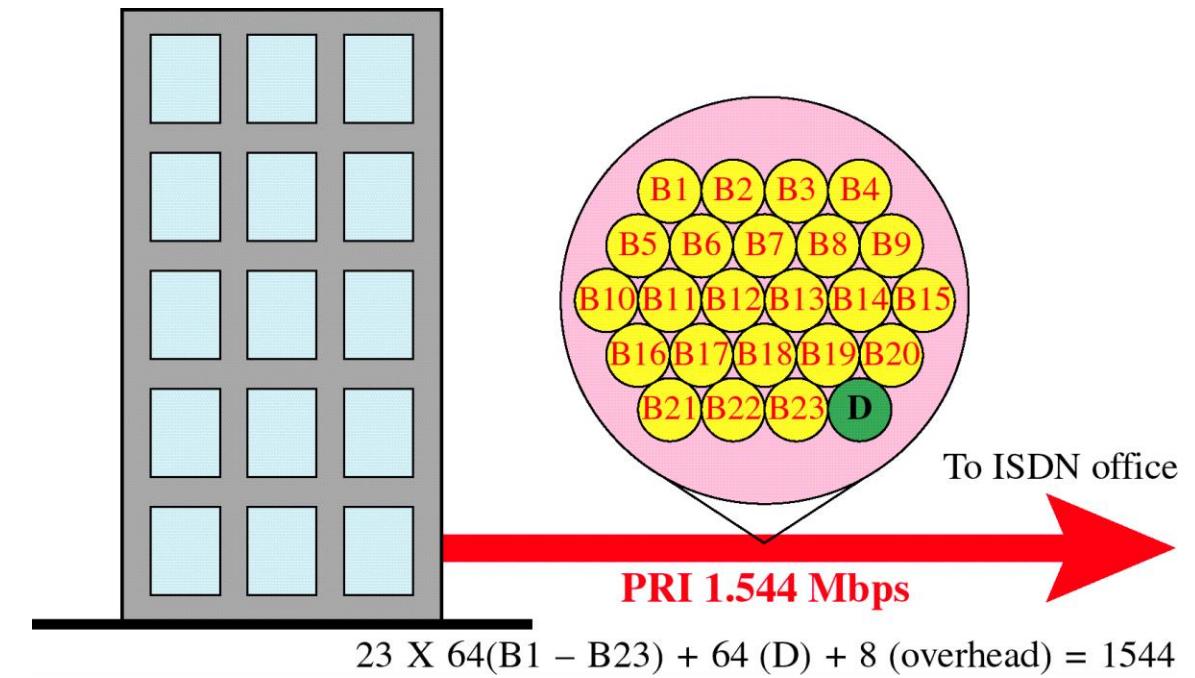
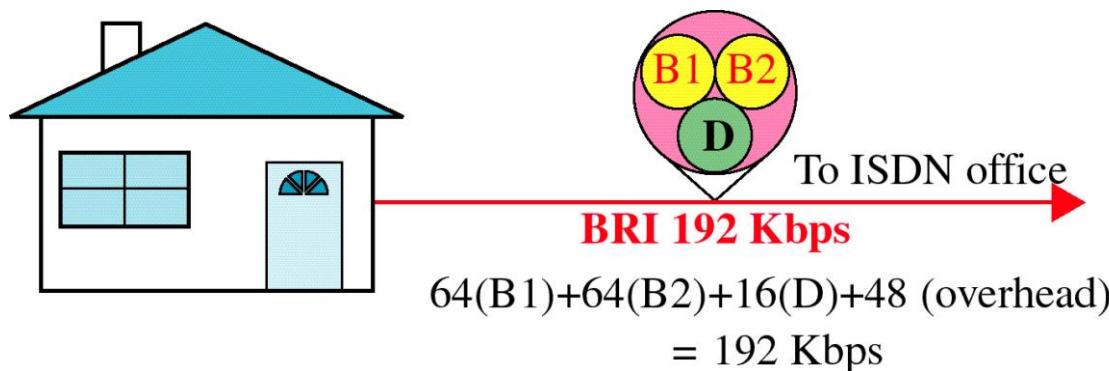


Digital Telephone System: ISDN

- **ISDN:** Carry digitized voice using standard telephony bandwidth
- **ISDN Voice Channels Bandwidth / Quantization:**
 - **Audio bandwidth:** ~300 Hz to 3400 Hz (speech-optimized passband)
 - **Nyquist rate:** To capture this bandwidth, the minimum sampling rate is $2 \times 3400 = 6800$ Hz
 - **Standard choice:** 8000 samples/second (8 kHz) is used in practice — the same rate as traditional PSTN systems.
 - **Voice quantization with 8 bits/sample**
 - **Non-linear analog signal compression** based on A-law in Europe, μ -law in North America/Japan
 - No digital data compression
 - **64 kbps per voice channel** is referred to as a B-channel (Bearer channel) in ISDN
 - **Modulated using “2B1Q” (2 Binary 1 Quaternary) modulation (=4-PAM -3V,-1V,+1V,+3V)**

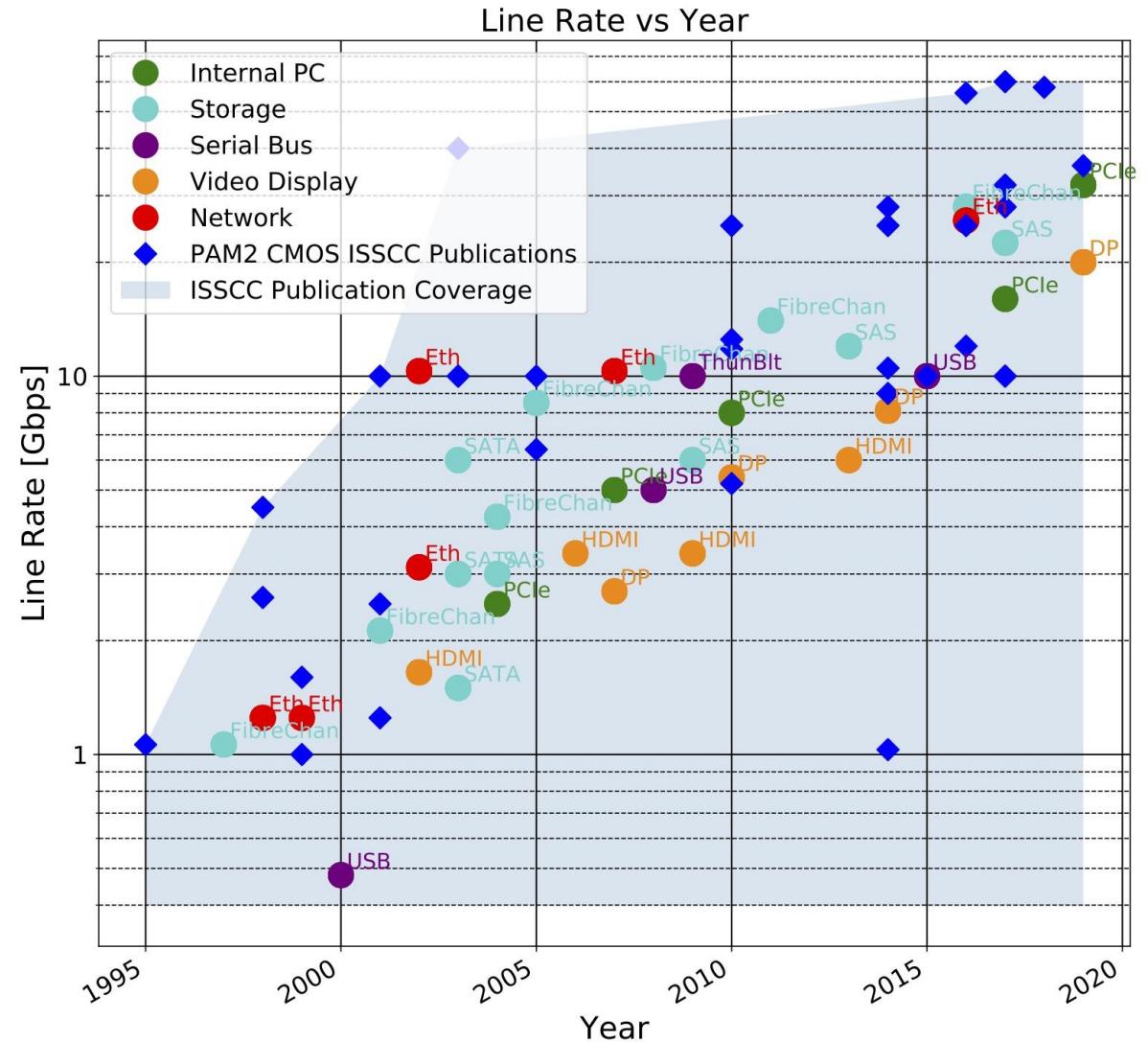
Digital Telephone System: ISDN

- **Basic Rate Interface (BRI):** residential and small-office customers
 - Two B-channels + one 16 Kbps D-channel (2B+D) and 48 Kbps of operating overhead
 - Can use the same **twisted-pair local loop** as analog network
- **Primary rate interface (PRI): business customers**
 - 23 B channels + one 64-kbps D channel and 8 kbps of overhead: 1.544 Mbps
 - **Require updated local wiring** to support the higher signalling bandwidth



SerDes High Speed Chip-to-Chip Links

- High speed serial links are essential for many applications
 - Optical Transmission: OC-192, OC-768, SONET
 - Internal PC: PCIe 1-5
 - Storage: Fibre Channel, SATA, SAS
 - Serial Bus: USB, Thunderbolt
 - Video Display: DisplayPort, HDMI



Ethernet

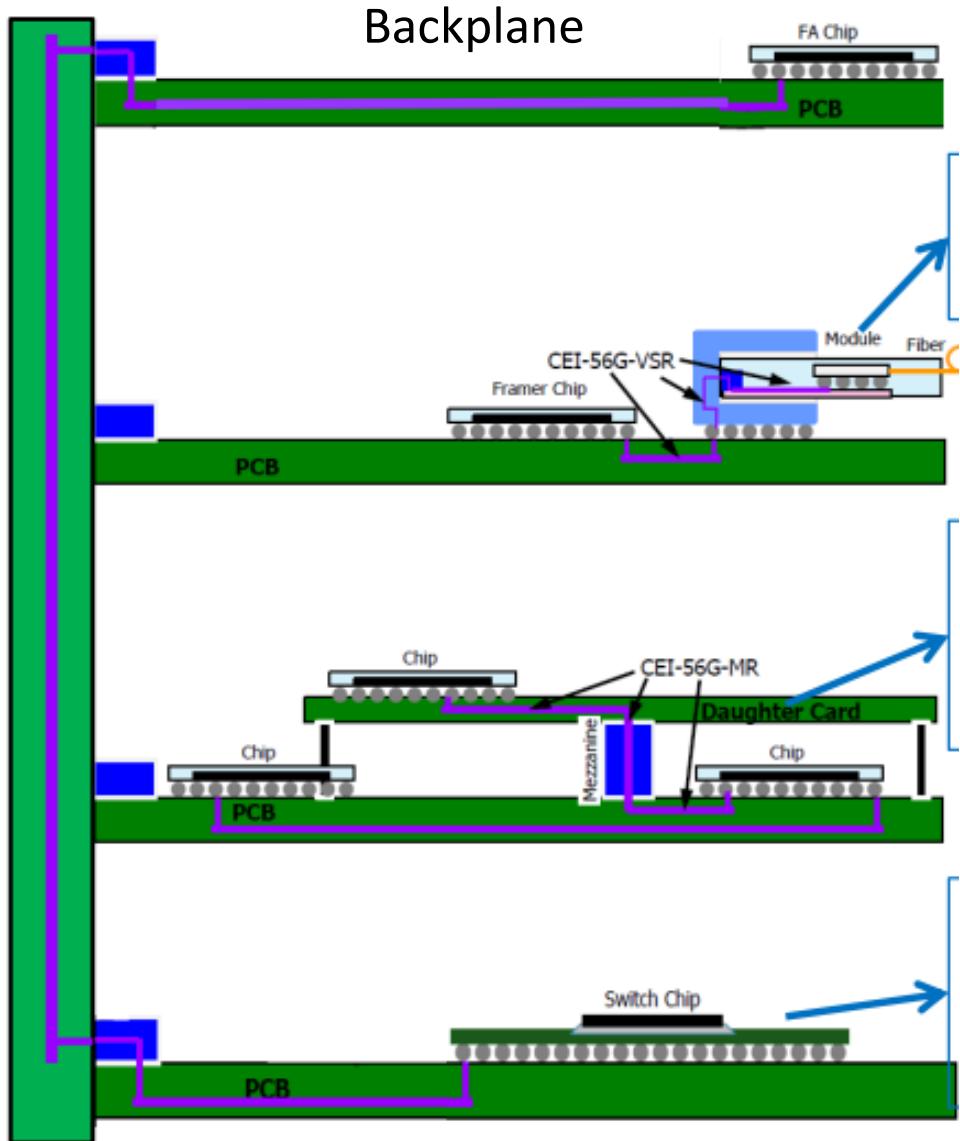
- **Purpose:** robust, and scalable method for local area communication (LAN)
 - Foundations developed in 1973
 - IEEE 802.3 standard since 1983 (10BASE5)



- **Evolution toward higher speed with more parallel lanes, higher bandwidth, and higher order modulation**

Standard	Data Rate	Modulation	Lanes	Baud rate	BW	Medium
10Base-T	10 Mbps	Machester	1	10	10 MHz	Twisted pair
100Base-T	100 Gbps	MLT-3 (PAM-3)	1	125	31.25 MHz	Cat 5
1000Base-T	1 Gbps	PAM-5	4	125	62.5 MHz	Cat 6
10GBase-T	10 Gbps	PAM-16	4	800	500 MHz	Cat 6A
25GBase-T	25 Gbps	PAM-16	4	2500	1250 MHz	Cat 8

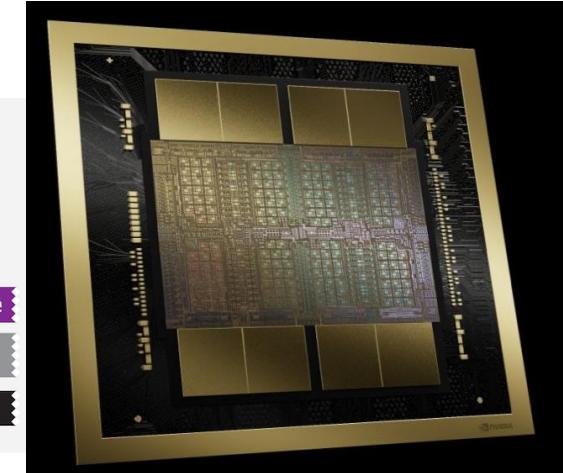
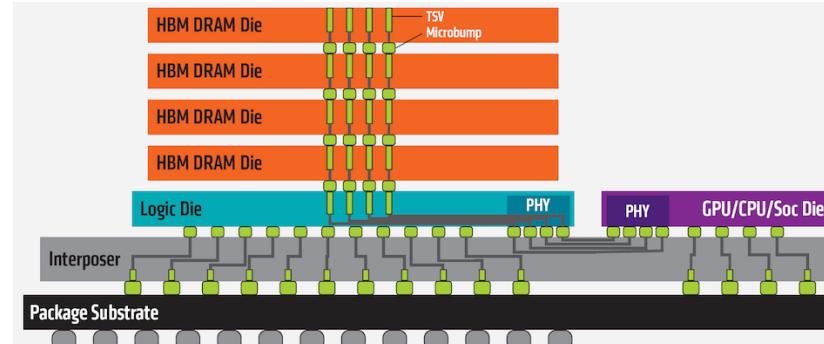
Chip-to-Chip Links in the Gbps Regime



Memory Interfaces: moderate rate due to many parallel pins

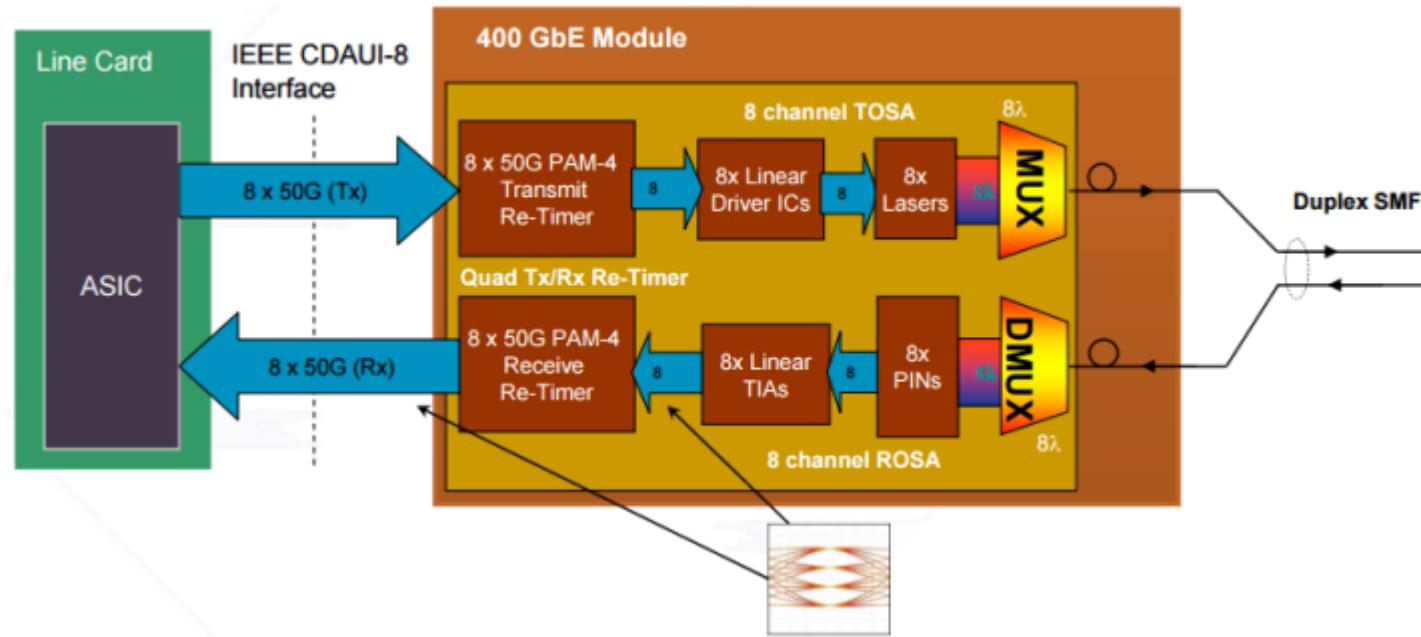
Standard	Year	Baud Rate
DDR	1998	400 Mbaud
DDR 2	2003	400 – 1066 Mbaud
DDR 3	2007	800 – 2133 Mbaud
DDR 4	2014	1600 – 3200 Mbaud
DDR 5	2020	3200 – 6400 Mbaud

Memory Interfaces



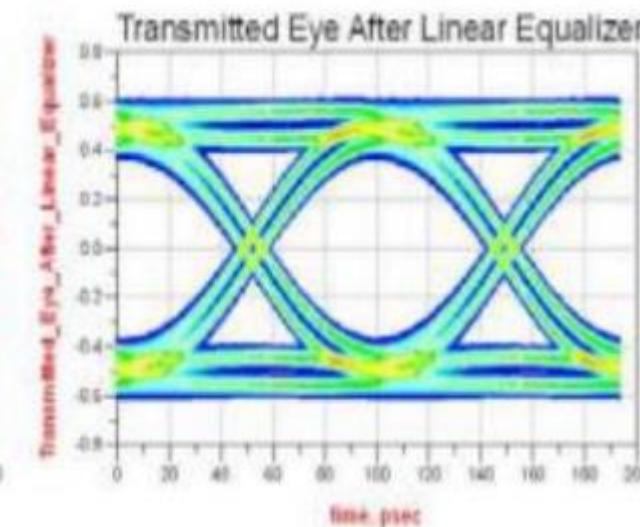
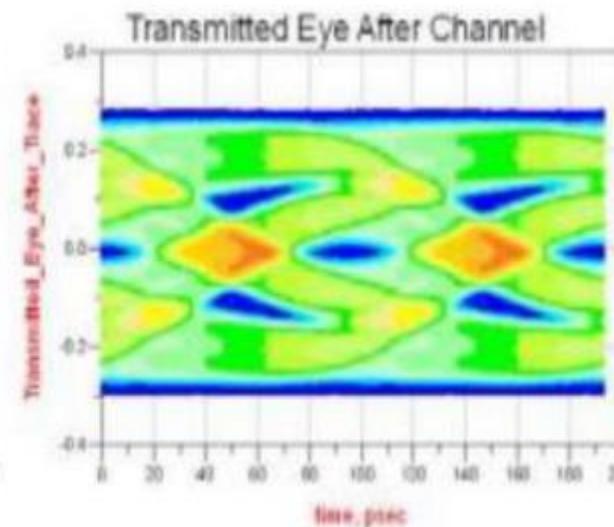
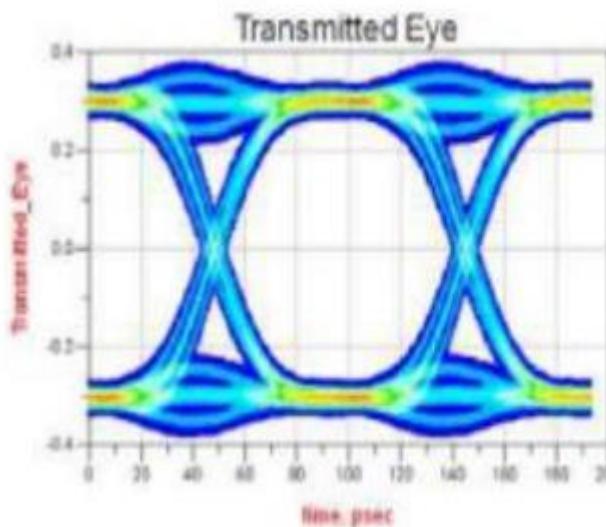
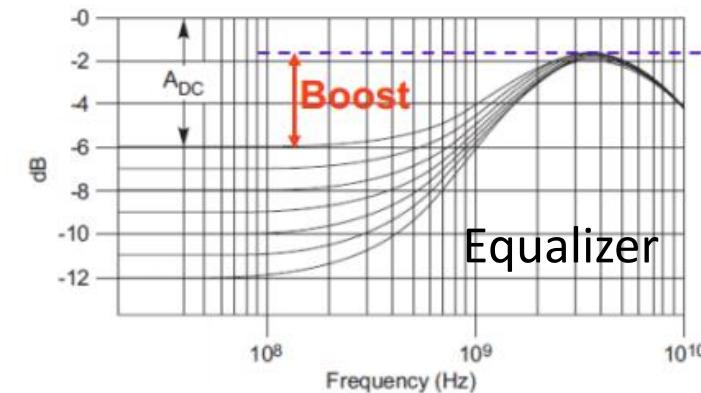
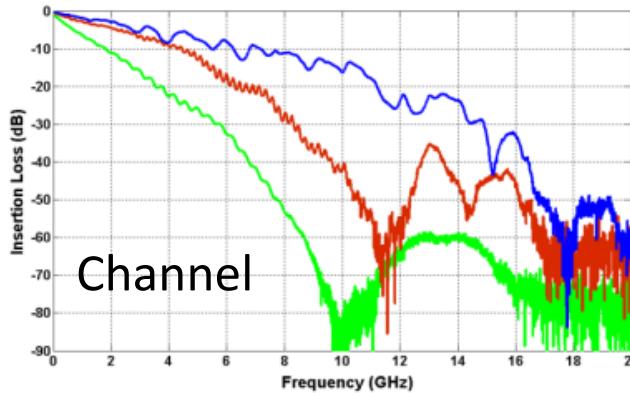
Line Card Interfaces

- Interfaces to optical links are even more challenging
 - Throughput in the 10s of Gbps per single pin



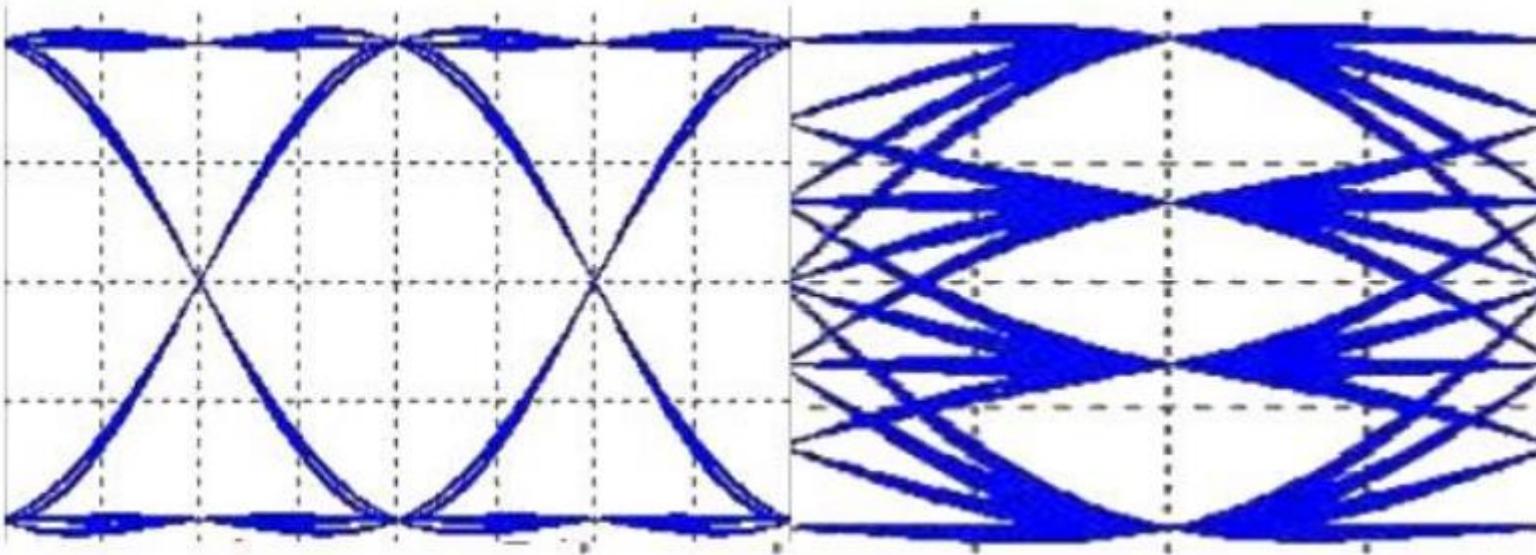
Serial Links Equalization

- Chip-to-chip physical link has limited bandwidth: correction with equalization



Serial Links with Higher PAM Order

- Higher order PAM requires less bandwidth, but eye opening also reduces

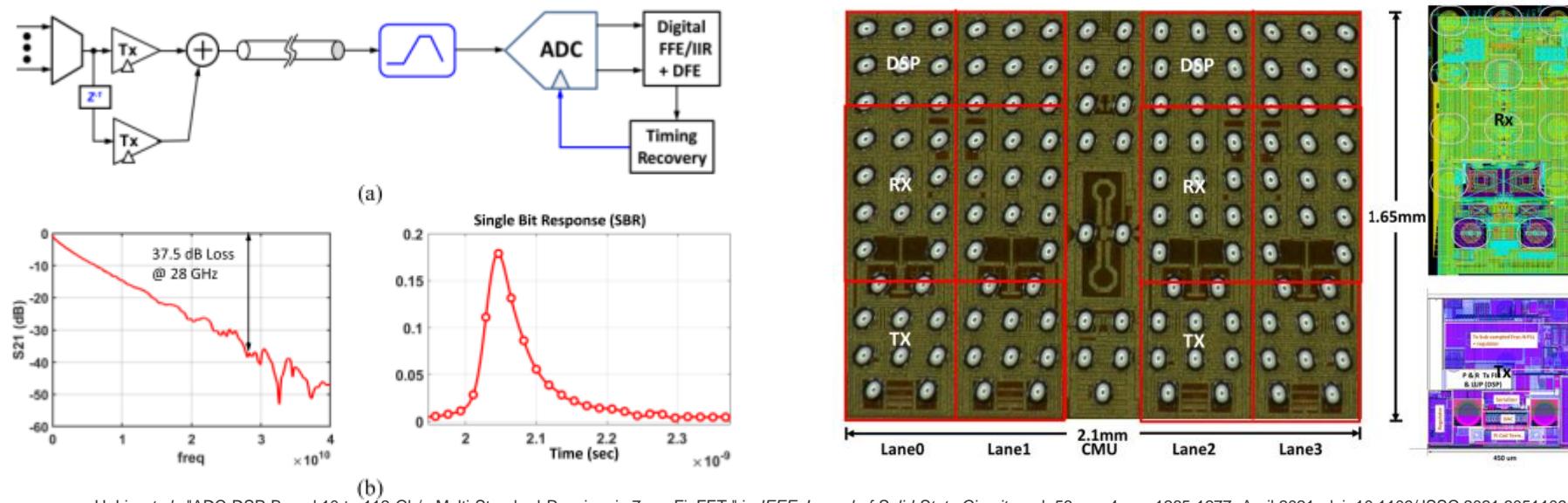


- Eye height for PAM4 is 1/3 of that of PAM2, thus

$$\text{SNR loss} = 20 \cdot \log_{10} \left(\frac{1}{3} \right) \approx 9.5 \text{ dB}$$

Difficulty to Implement Advanced Receivers

- **Serial Links today can operate with Baud Rates in the 10s Gbps regime**
 - Signal processing mostly done in the analog domain
 - Digital signal processing only recently possible with very high-speed data converters



H. Lin et al., "ADC-DSP-Based 10-to-112-Gb/s Multi-Standard Receiver in 7-nm FinFET," in *IEEE Journal of Solid-State Circuits*, vol. 56, no. 4, pp. 1265-1277, April 2021, doi: 10.1109/JSSC.2021.3051109.