Integrated Analog Electronics - Industrial Applications

Digital Audio Power Amplifiers

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Outline - Monday

- – Introduction to TI Denmark (Toccata Techn.)
- – Advantages of Digital Amplifiers
- – System Overview
- – Why is analog design important for a ”true Digital amplifier”
  - I.e. why are you guys sitting listening to this...
- – Intro. to Pulse Width Modulation (PWM)
- – Demo of a digital amplifier solutions (silent demo)
- – Q & A

TI-Denmark (former Toccata Techn.)

- Toccata was founded in 1997 by Lars Risbo
  - Spin-off from DTU Ph.D project ('92-'94)
- 1998: "Millennium" hi-end amp in co-op with TacT audio
- 1999: licence deal with TI
- 2000: Toccata acquired by TI

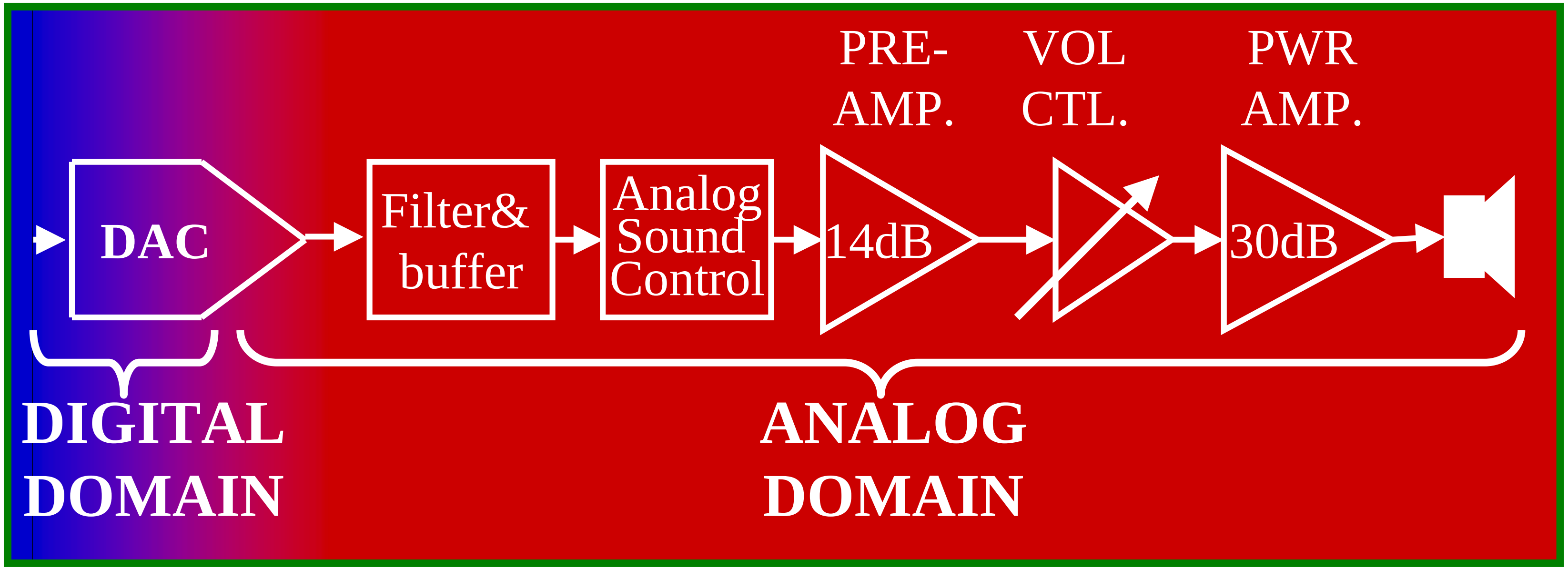
Introduction to equibit/TDAA

Why and how does it work???

Conventional signal path

DAC needed
Analog processing
Complex & long analog signal path:
    Sound degradation!!!

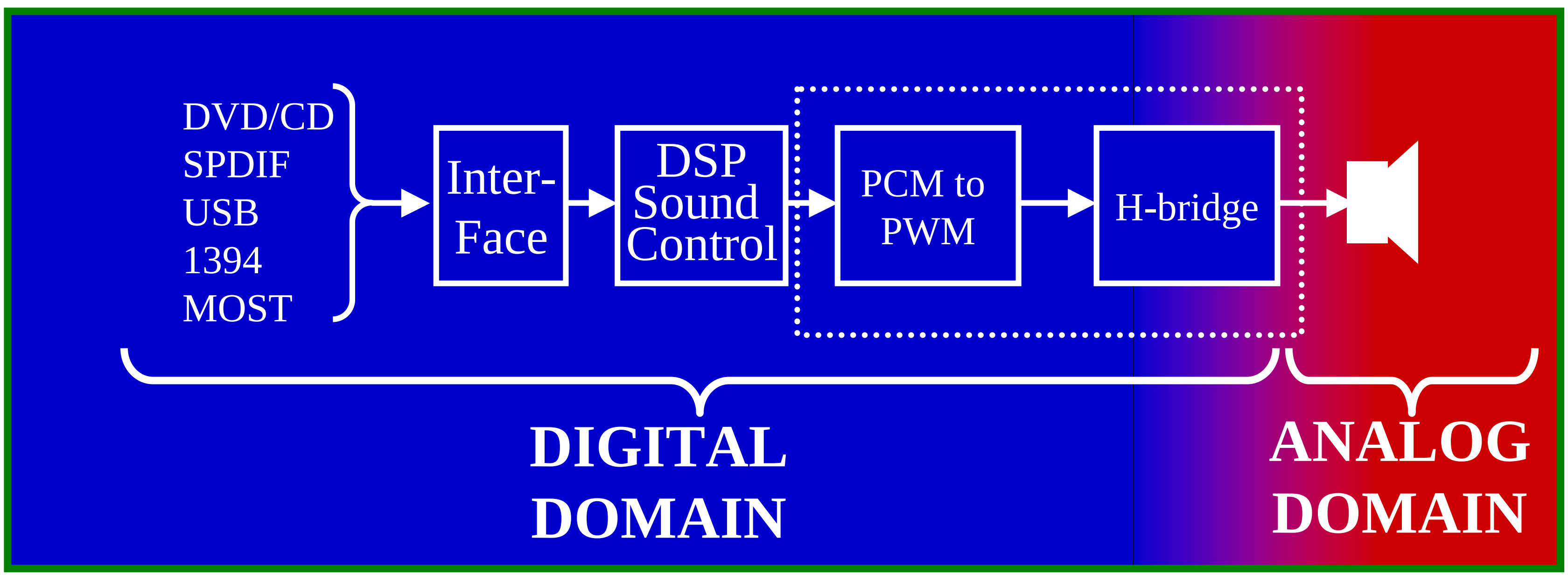
High Analog Gain:
    Sensitive to noise & Hum pick up:
    SNR at speakers is much lower than theoretical DAC spec



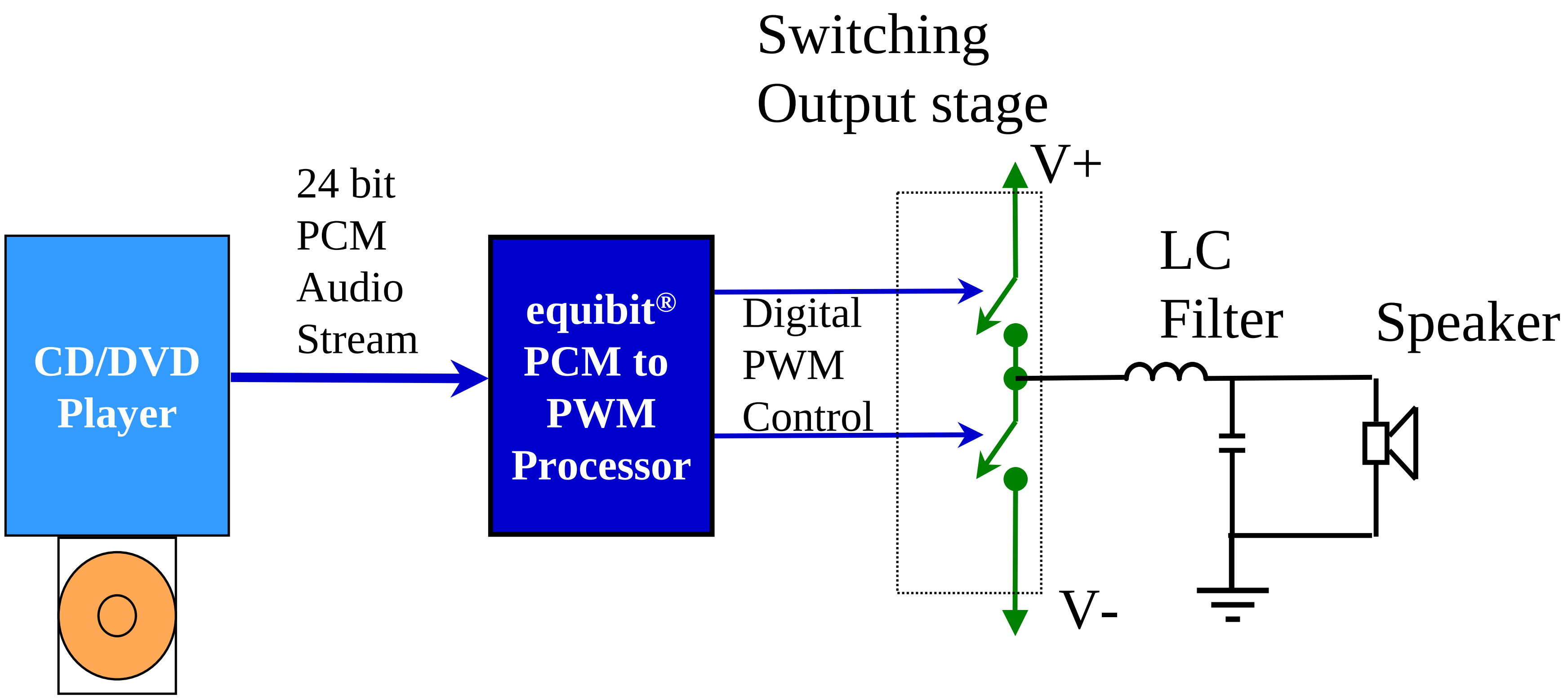
Equibit® Power Processor signal path

True digital path
Only digital processing
No analog processing
No D/A converter used

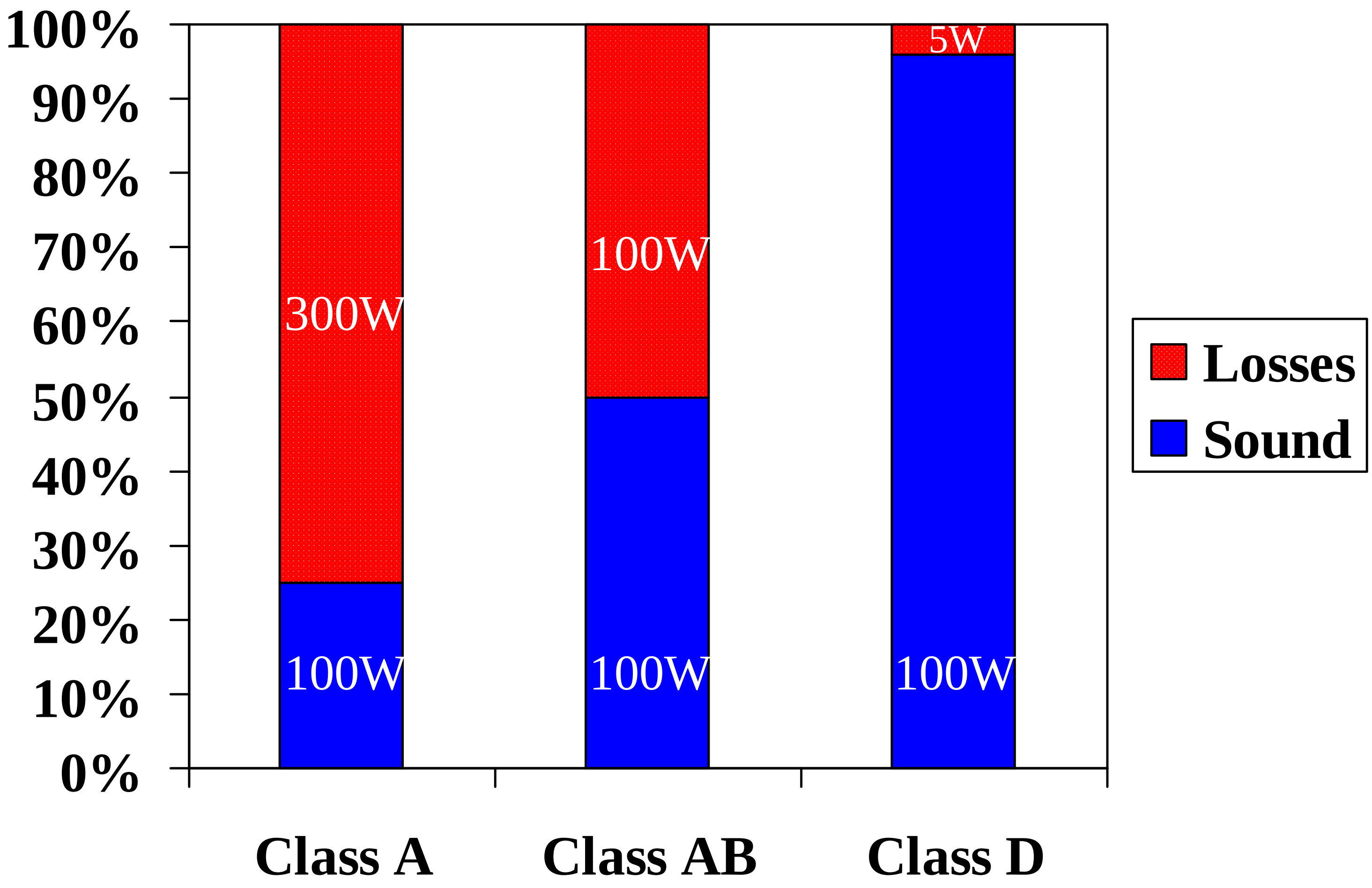
No-loss DSP sound ctrl.
No gain or amplification
Digital is immune to noise/hum pick-up



Equibit® Power Processor (Basic Principle)

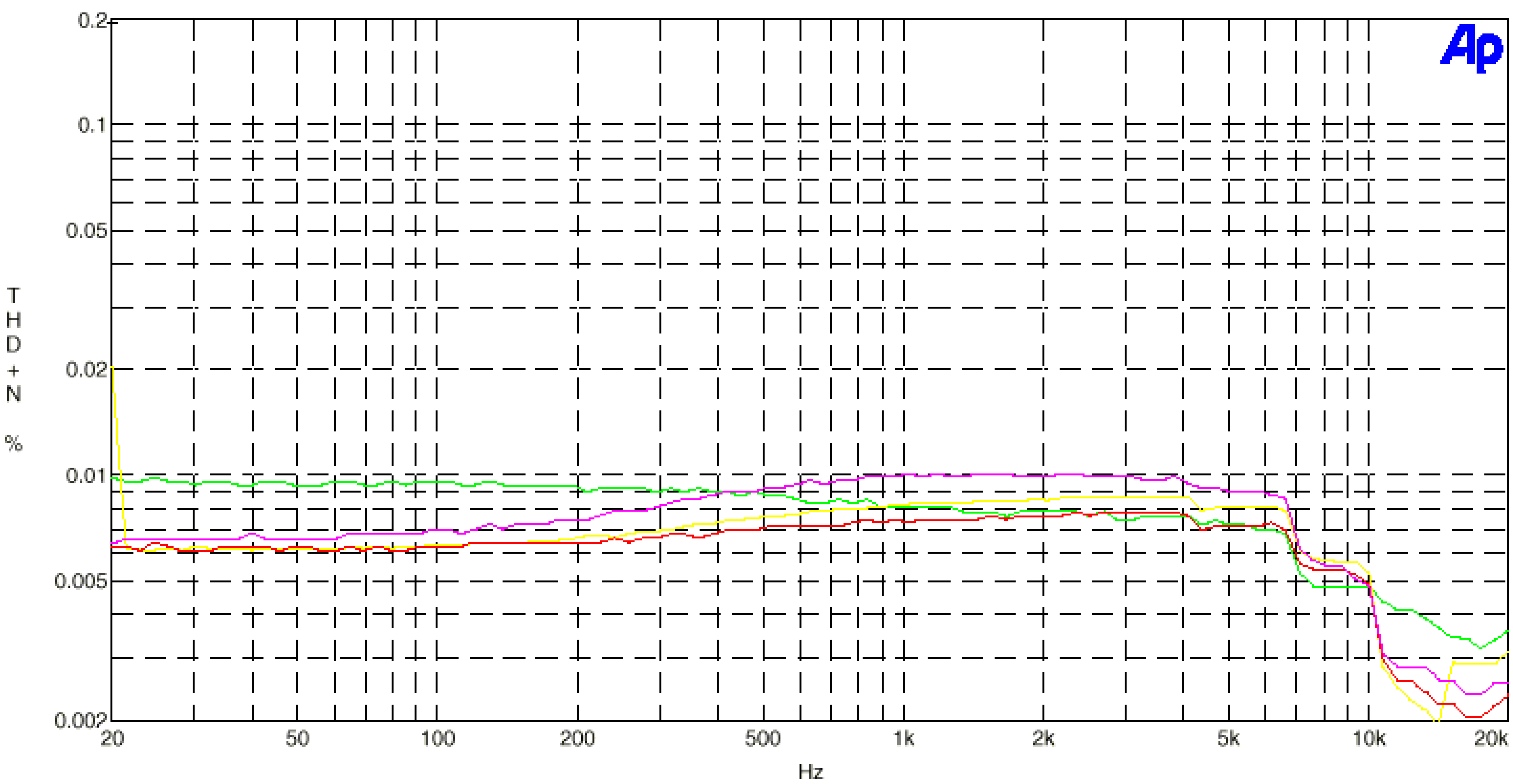


Digital power processing efficiency

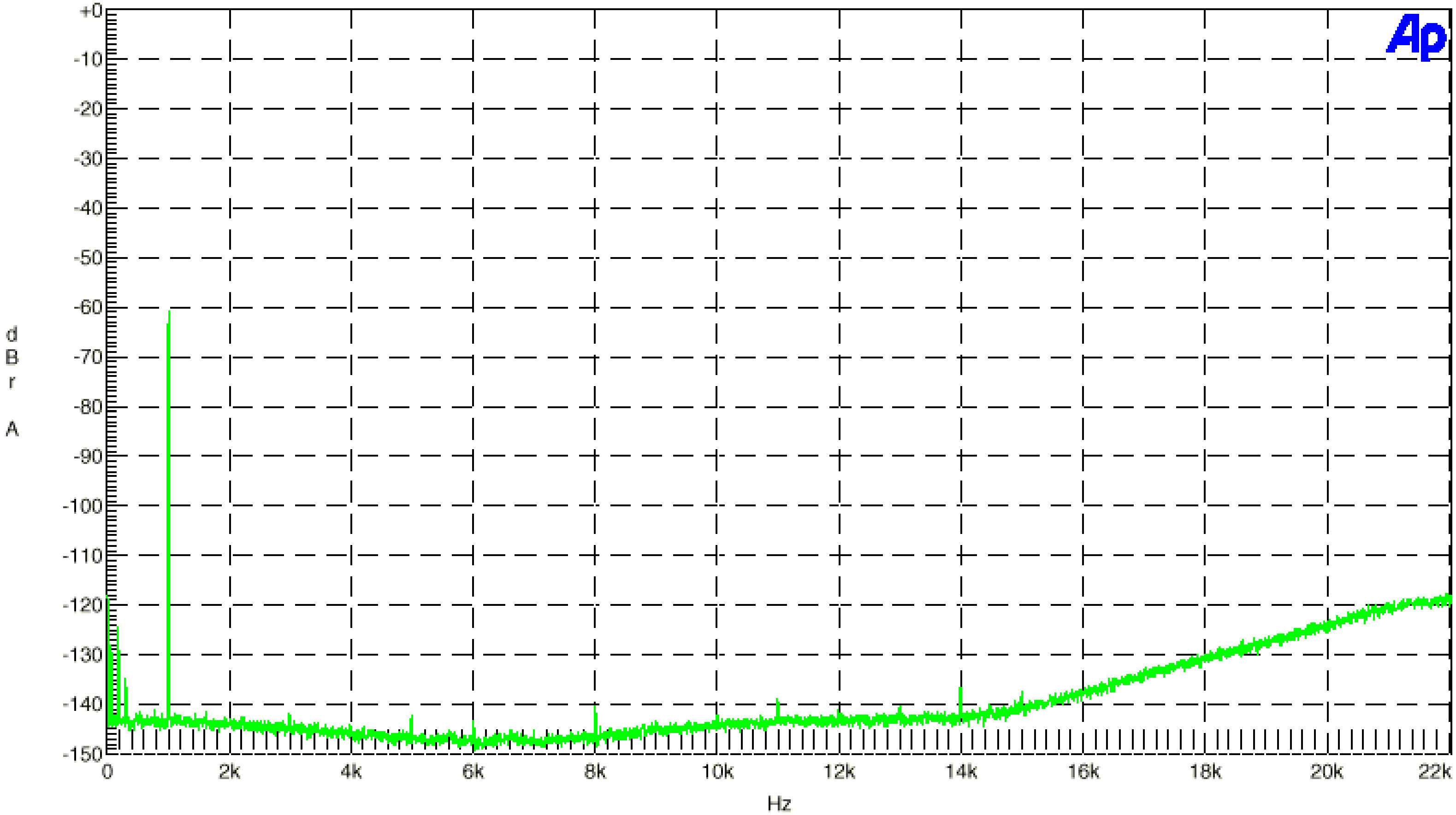


Digital power processor, THD+N

100mW-100W

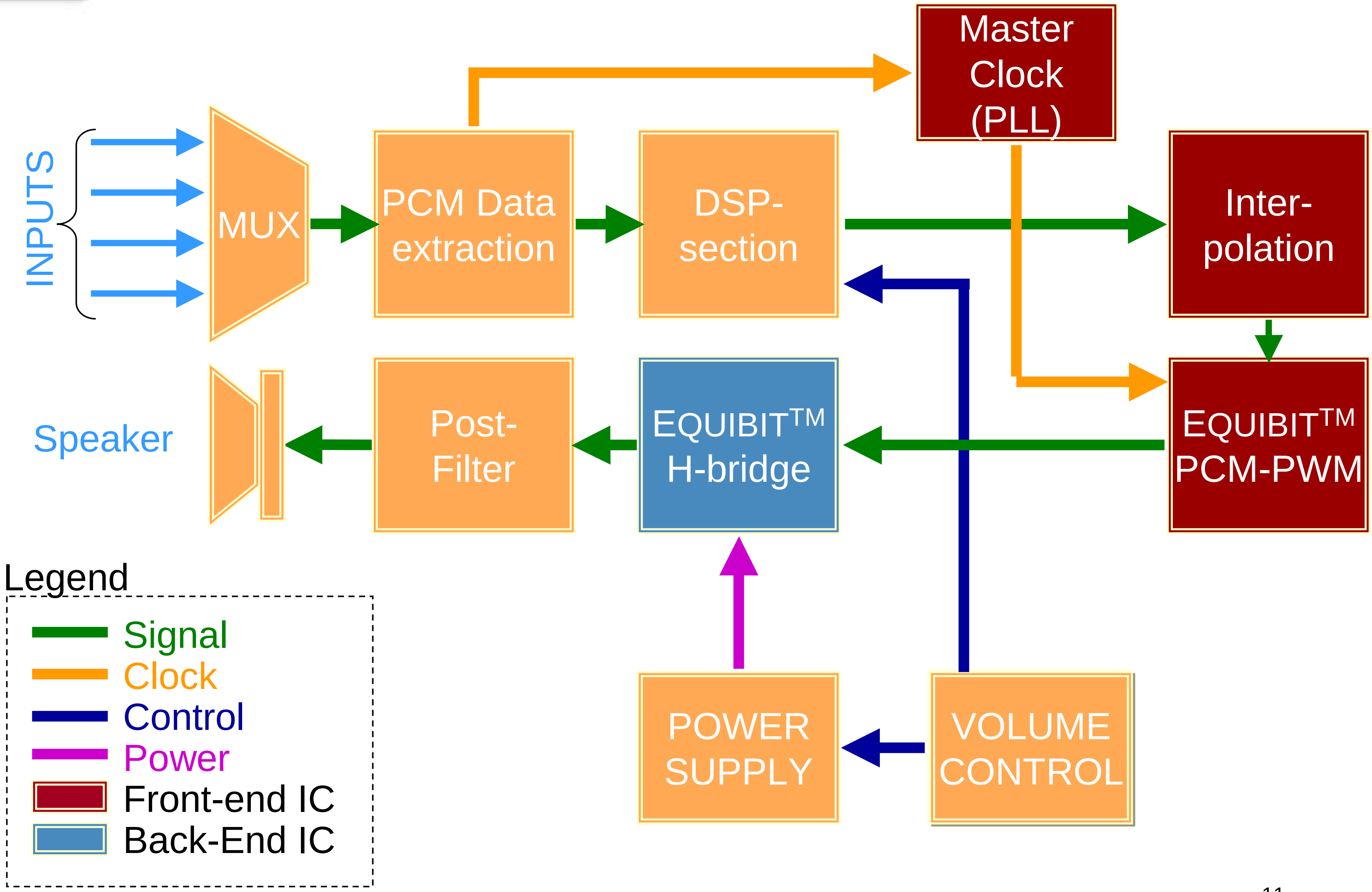


Digital power Processor, Noisefloor



Pure Path Digital
Texas Instruments

System Block Diagram



Why is analog design important for a ”true Digital amplifier”?

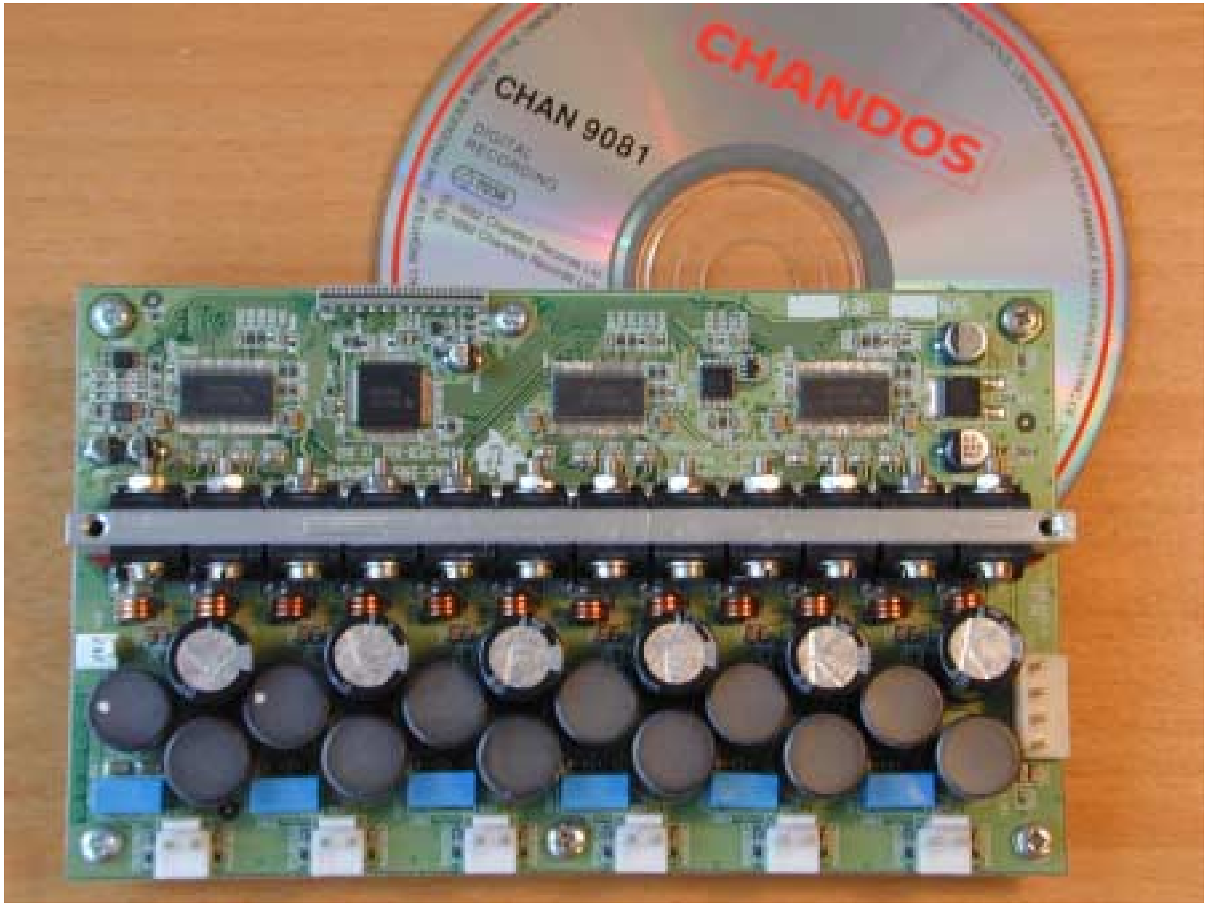
- **Phase Locked Loop (PLL)**
  - **The PLL is an analog circuit**
  - **Low clock jitter is critical for obtaining low noise**
  - **The PLL is a very critical system component**

- **Power Stage**
  - **Mostly analog circuit**
  - **Must achieve <0.1% THD => high precision waveforms**
  - **Timing control is critical**
  - **Protection system is challenging**

Silent DEMO

System solutions with TI Pure Path Digital amplifier chips

6x100W/6ohmsEVM including 6ch modulato+volume chip



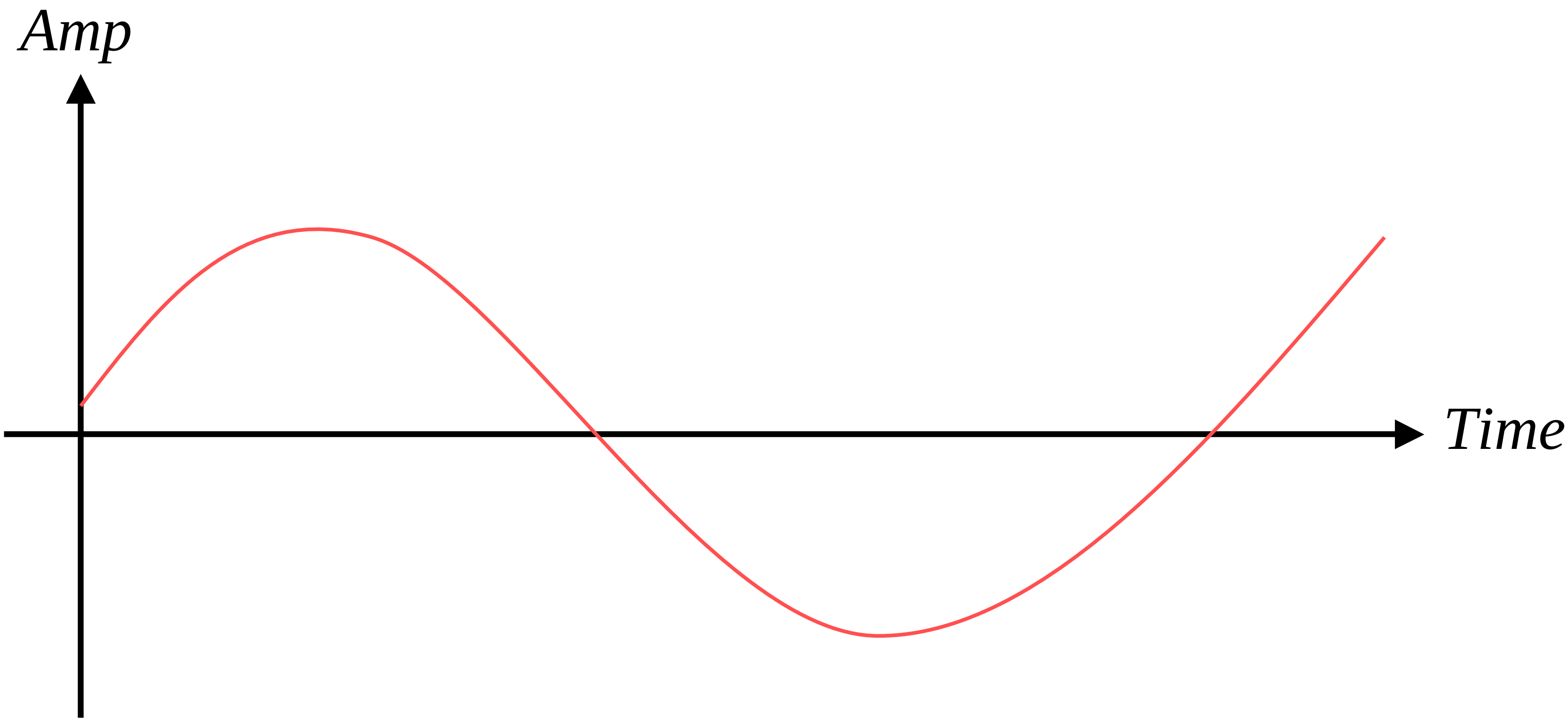
Panasonic DT100 DVD mini-compo, 6x30W

Panasonic XR10 AV receiver, 6x100W

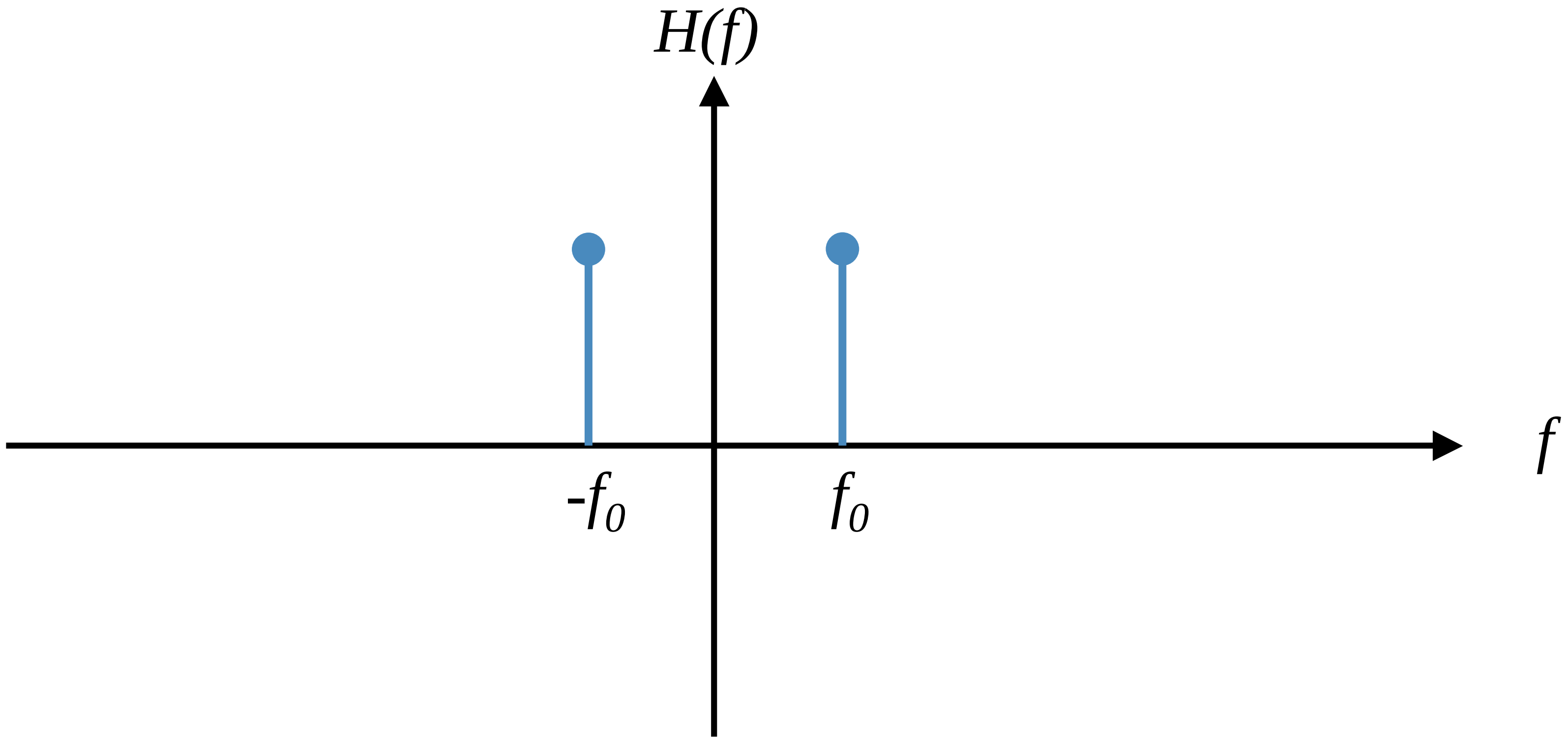
PWM techniques

Overview

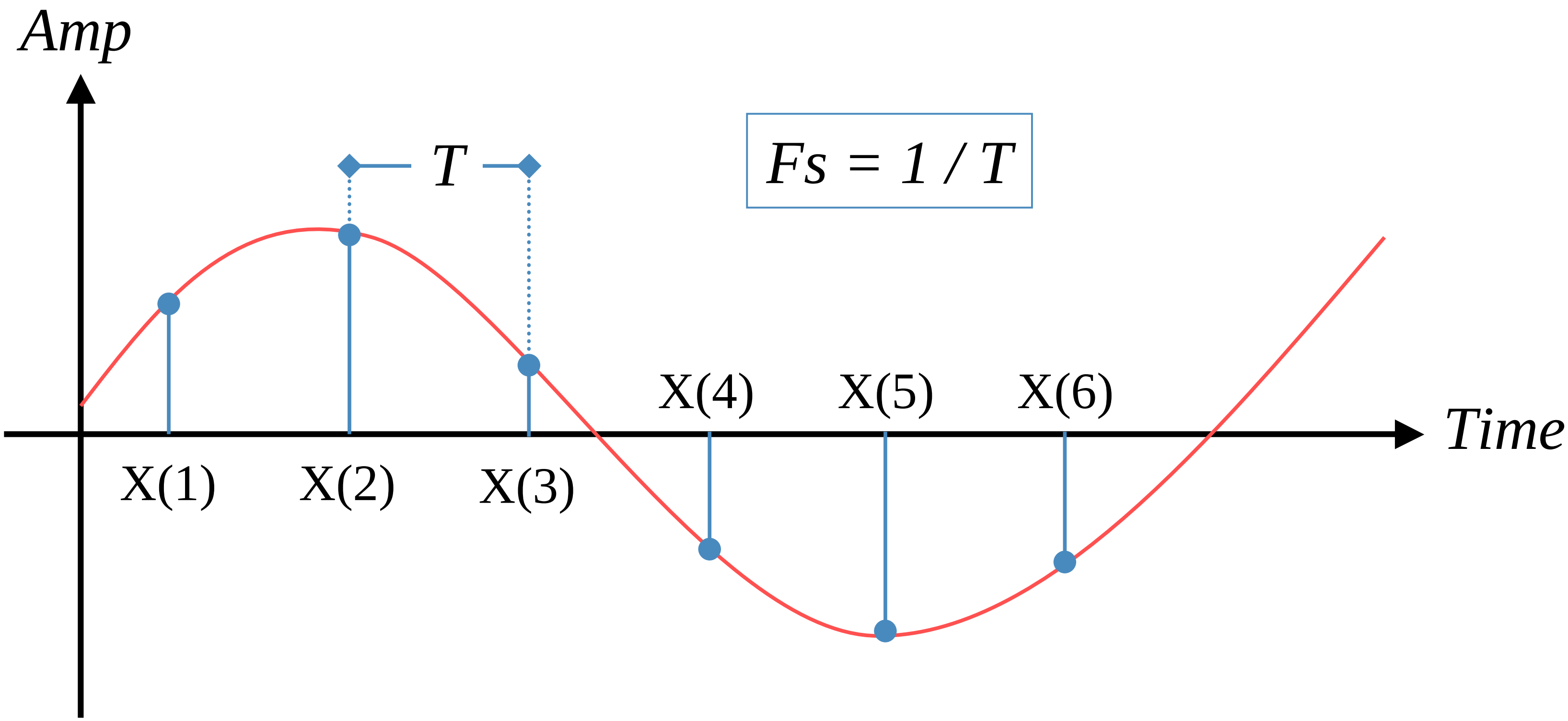
Analog Continuos Signal



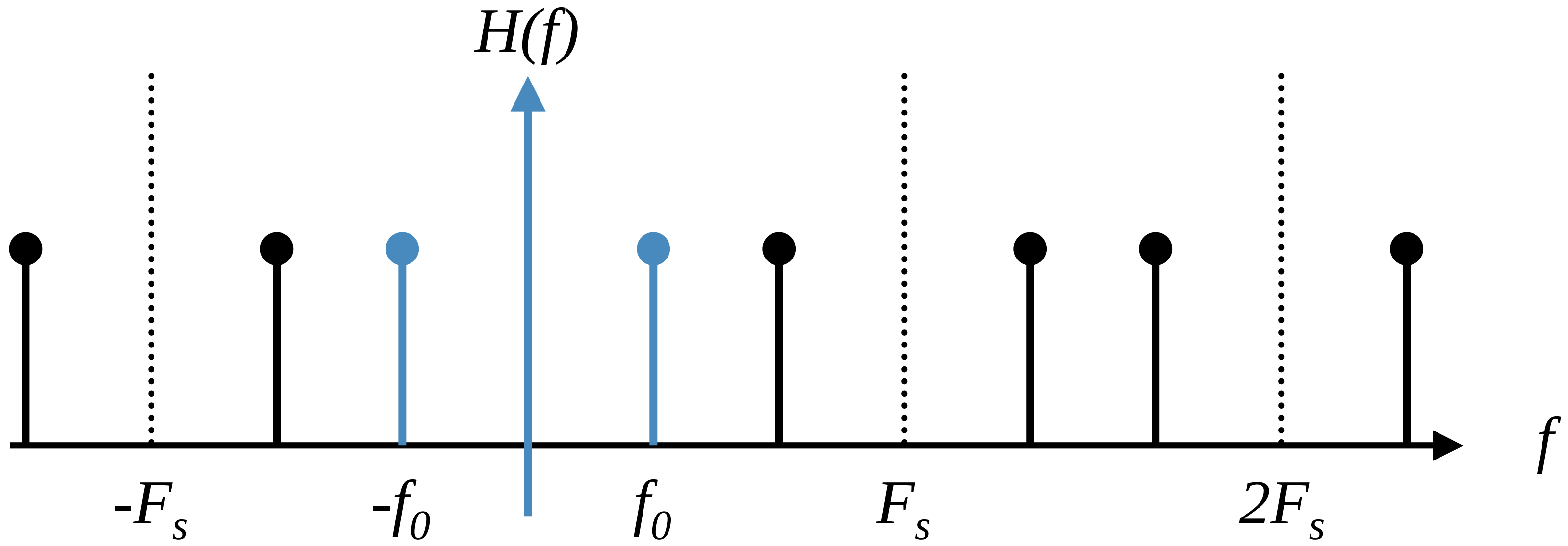
Continuos Signal: Spectrum



PCM Signal

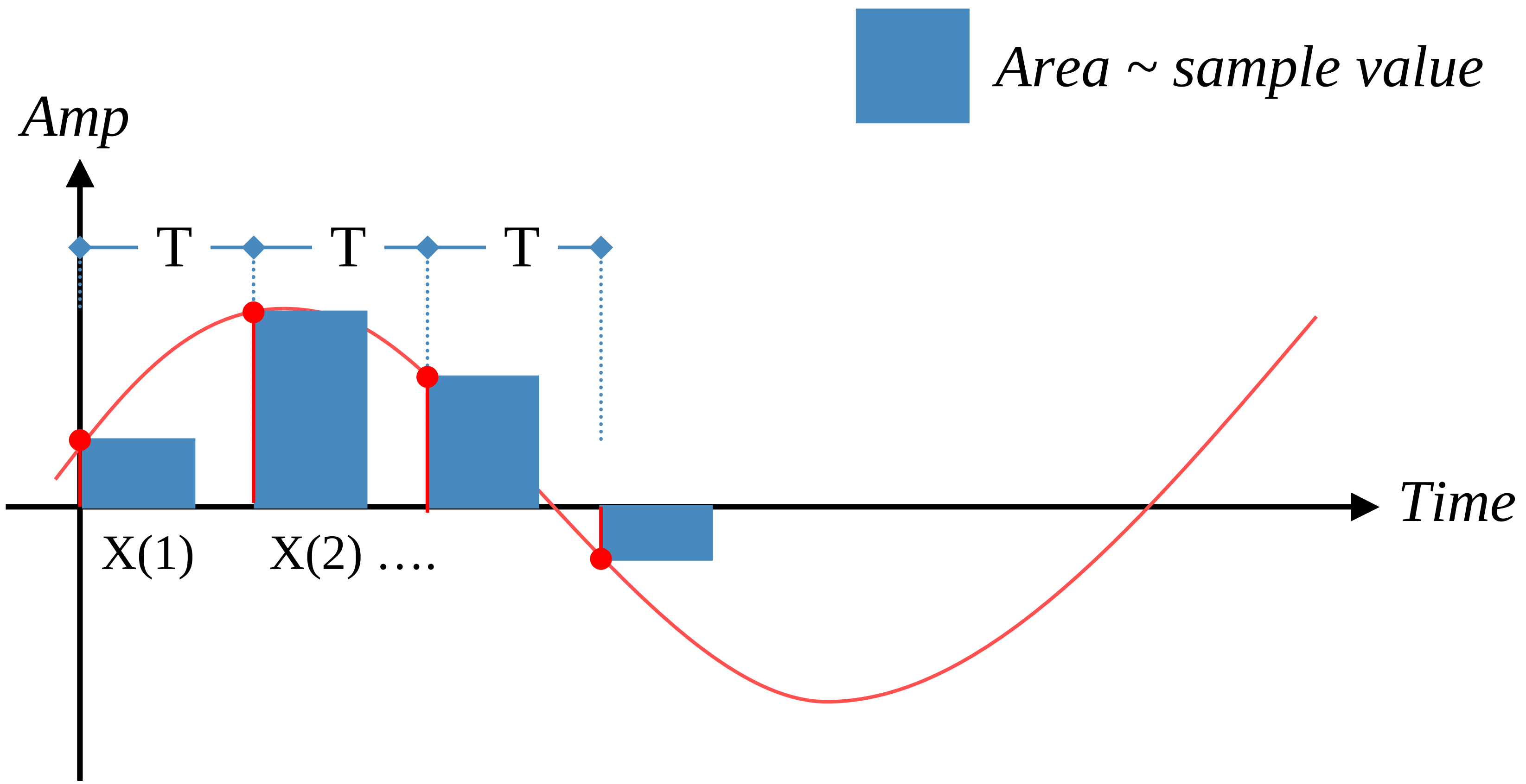


PCM Signal: Spectrum



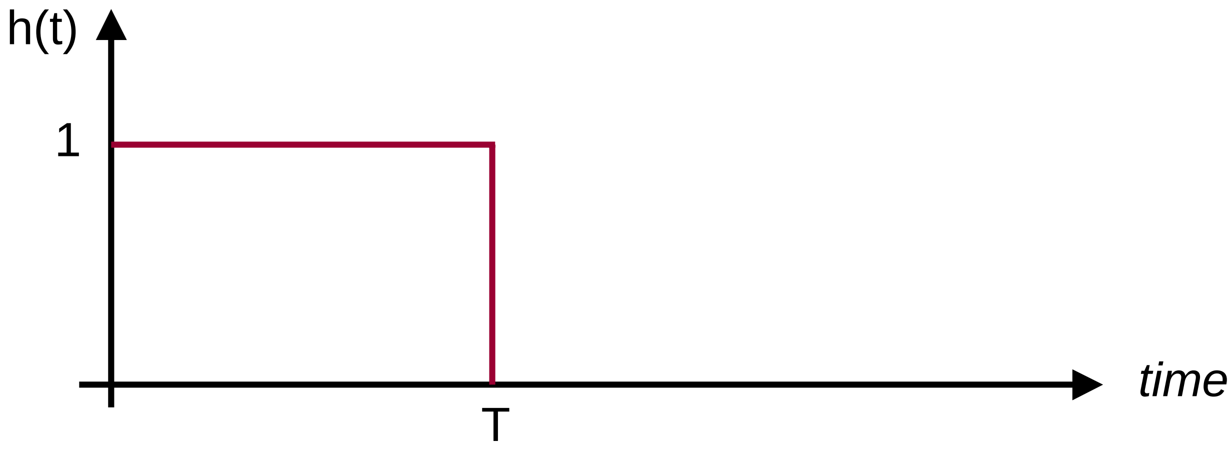
PCM=Pulse Code Modulation

PCM Signal Reconstruction (1)

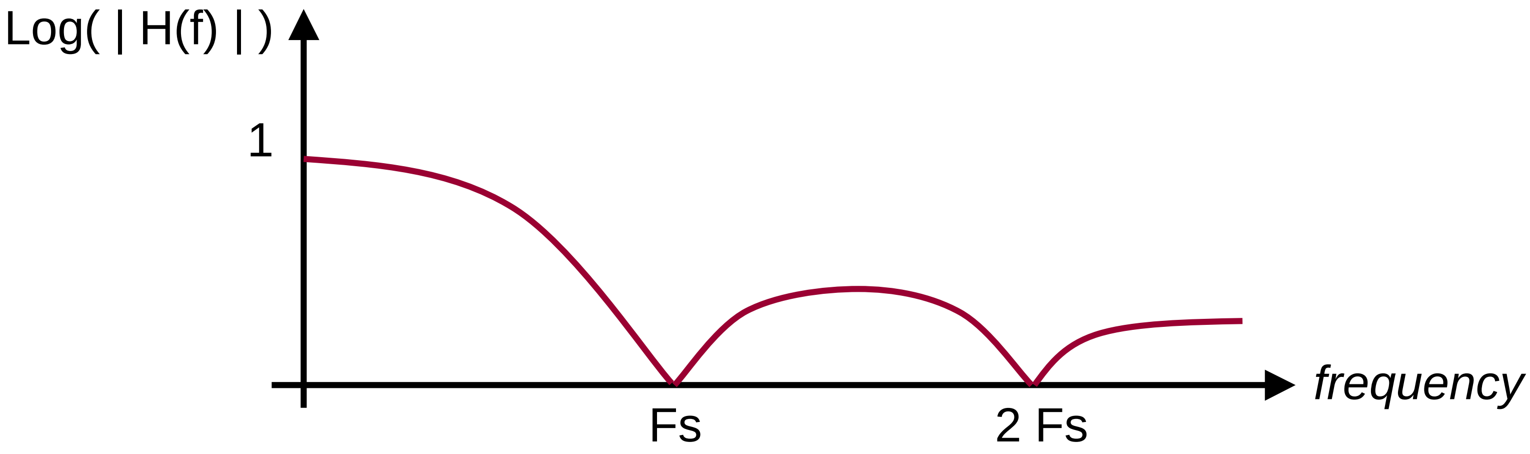


Pulse: Variable Amp + Fixed duration

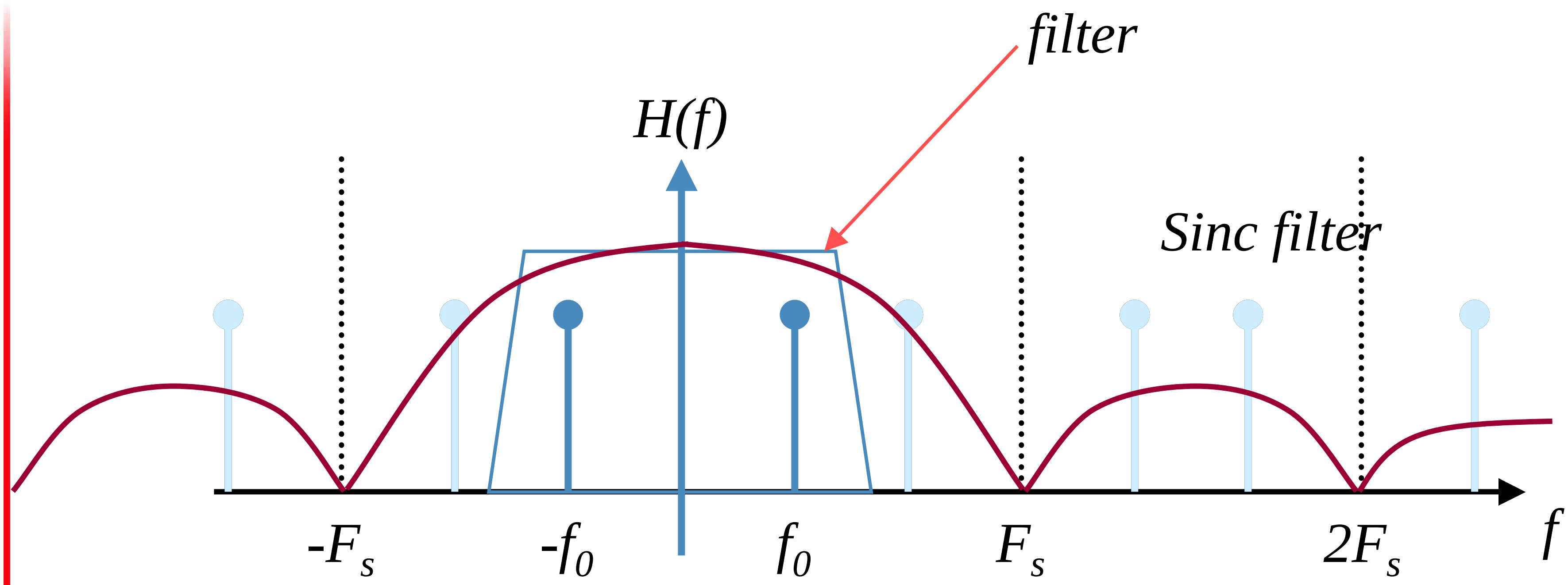
SINC Filter Impulse Response



SINC Filter Frequency Response

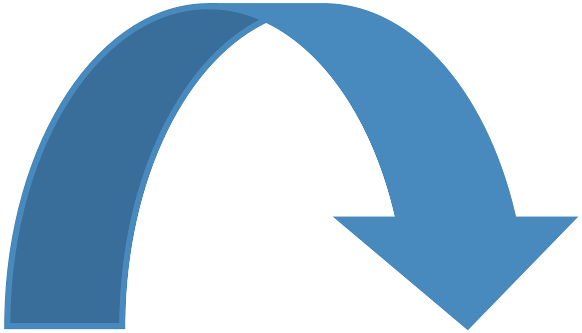


PCM Reconstruction (2)



Purpose of the PWM

ENCODING



PCM-

SIGNAL

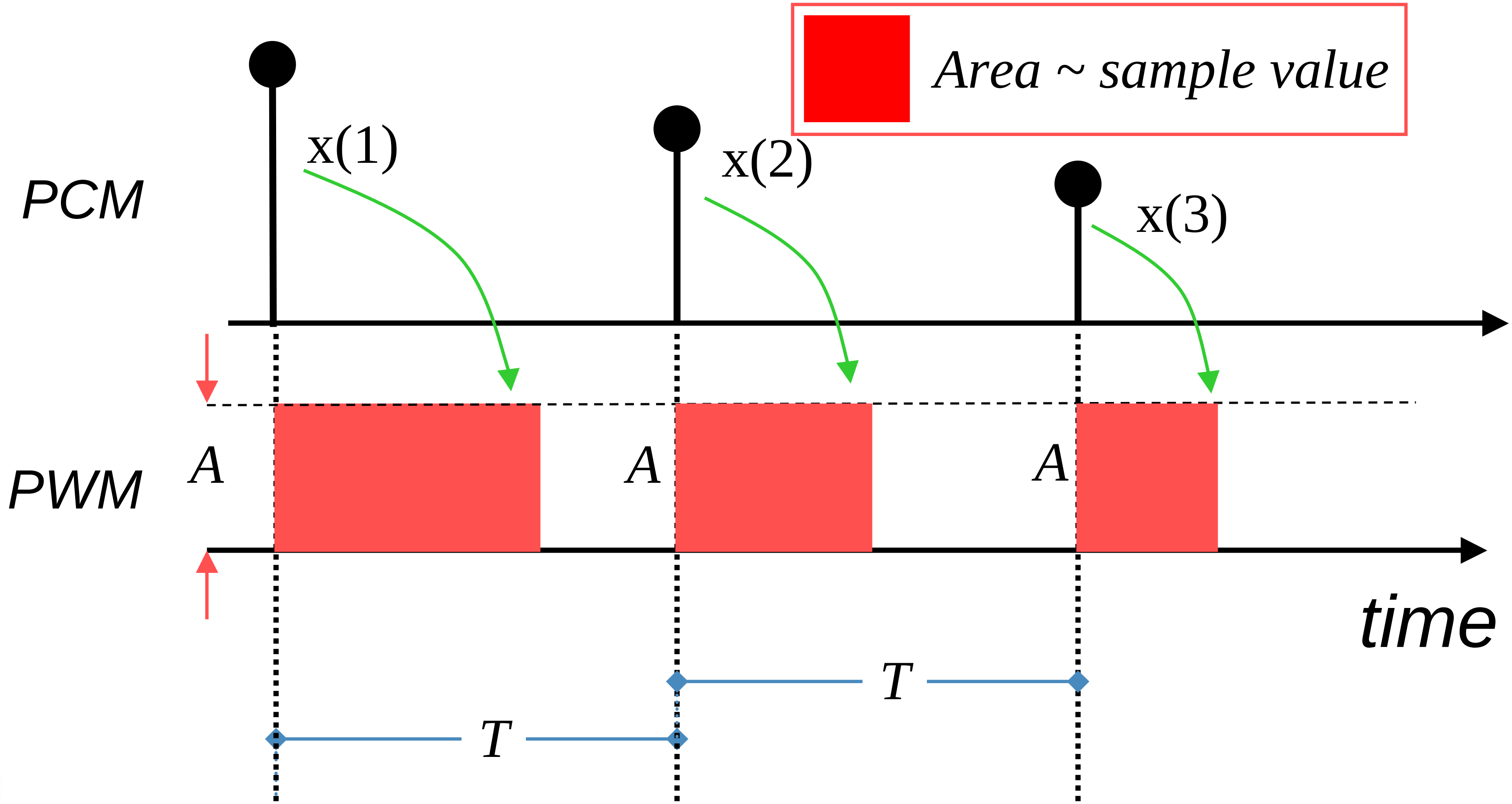
Two-level

(one-bit)

SIGNAL

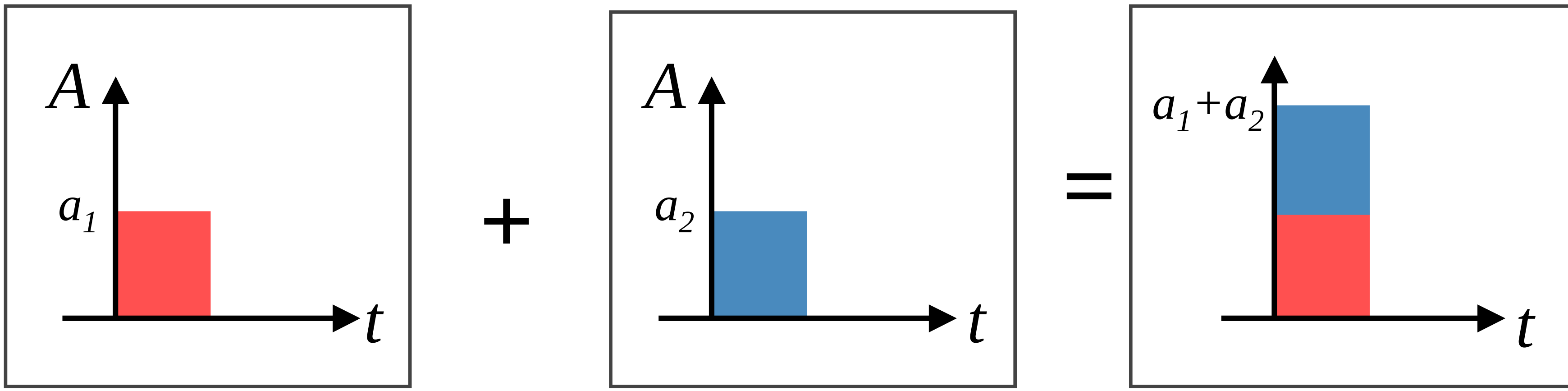
- **To accomplish a PCM to 1-bit signal ENCODING that:**
  - **Has a low Pulse Repetition Frequency (PRF)**
  - **Controlled by a digital clock**
  - **Decoded by a low-pass filter**
  - **High Fidelity (low THD, low noise)**

PCM → PWM



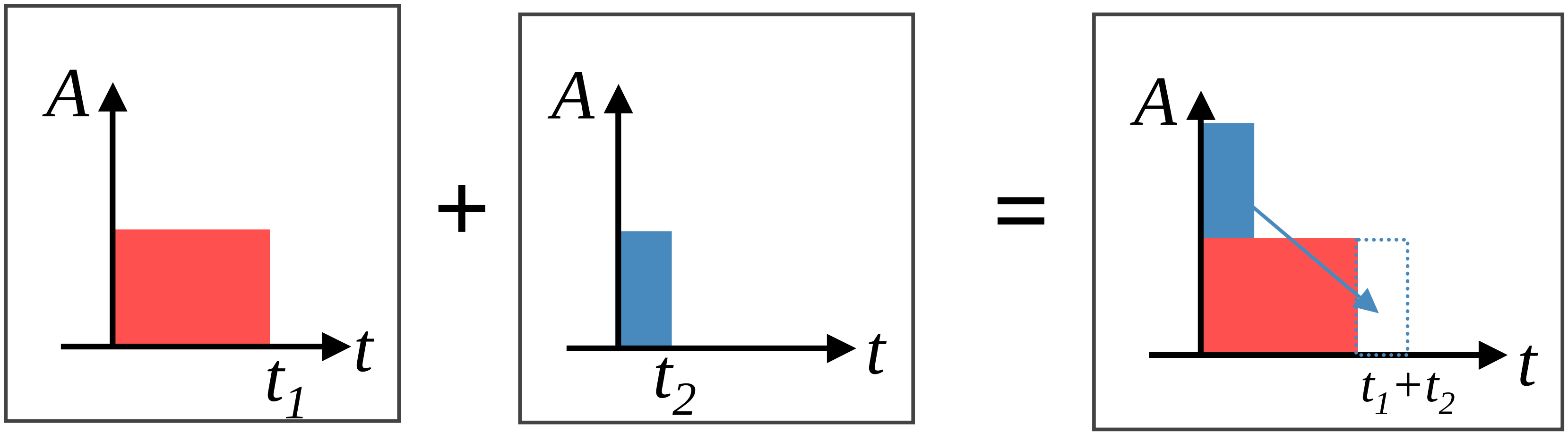
Pulse: Fixed Ampl. + Variable duration

Linear Axiom - PCM



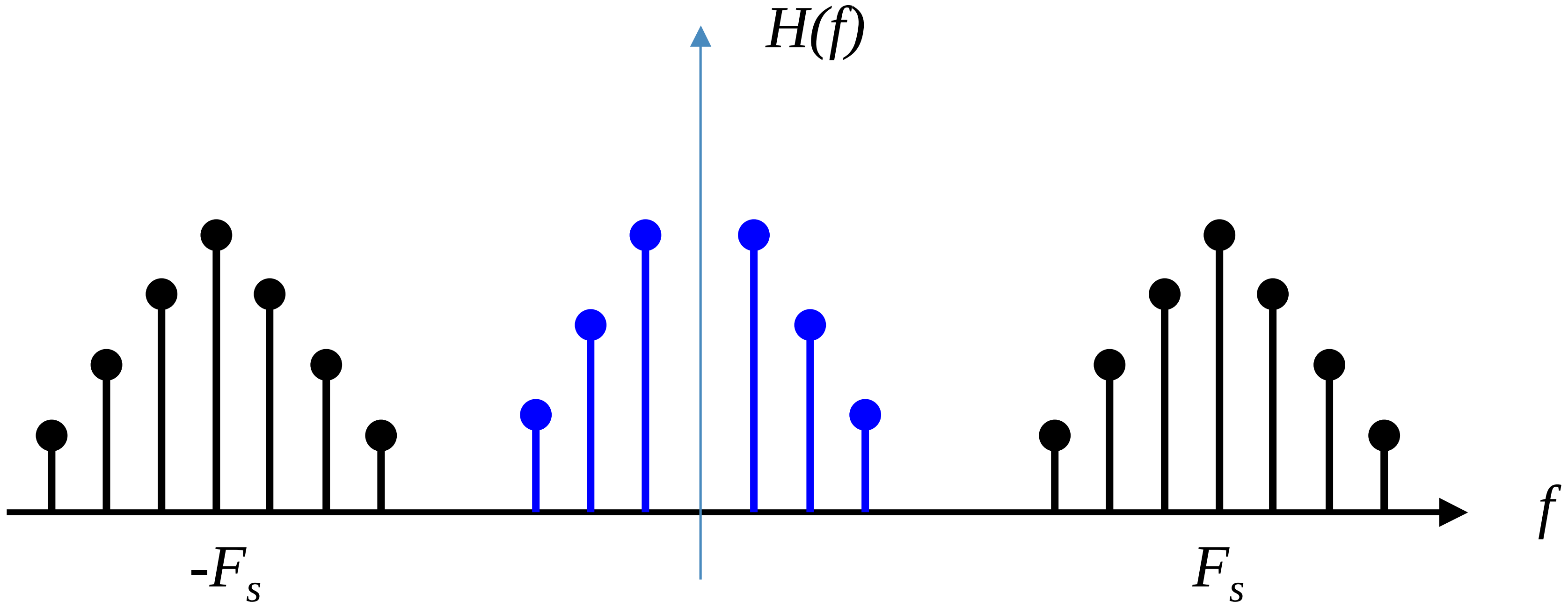
Linear Operation: $f(a_1)+f(a_2)=f(a_1+a_2)$!!

Linear Axiom - PWM

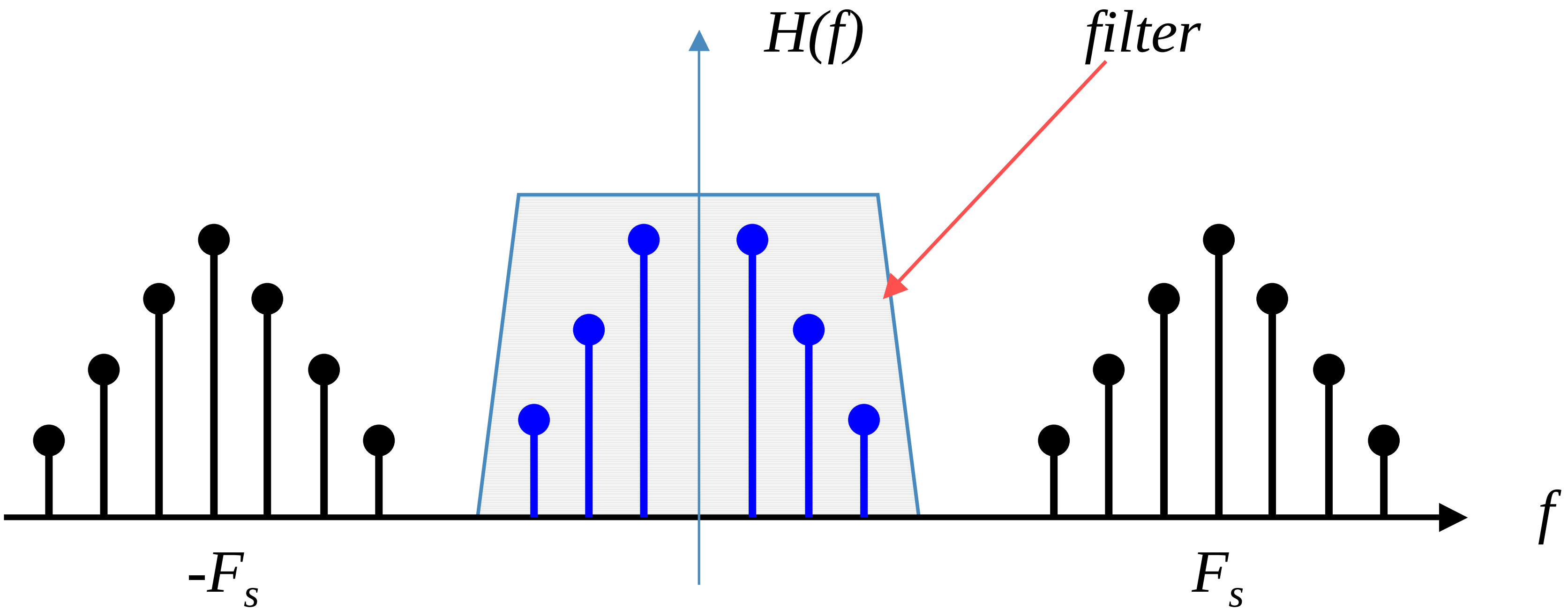


Non-linear Operation !!

PWM Signal Spectrum

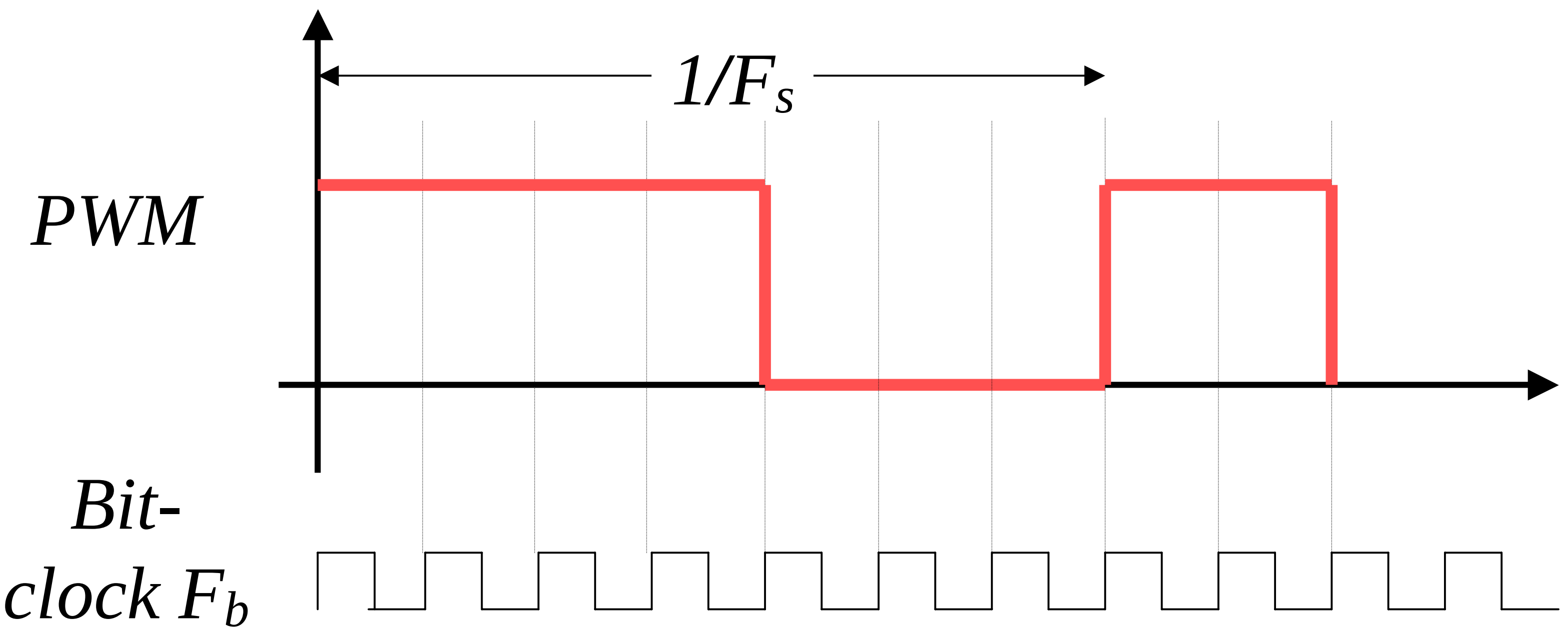


PWM Signal: Reconstruction Filter



Discrete-Time UPWM

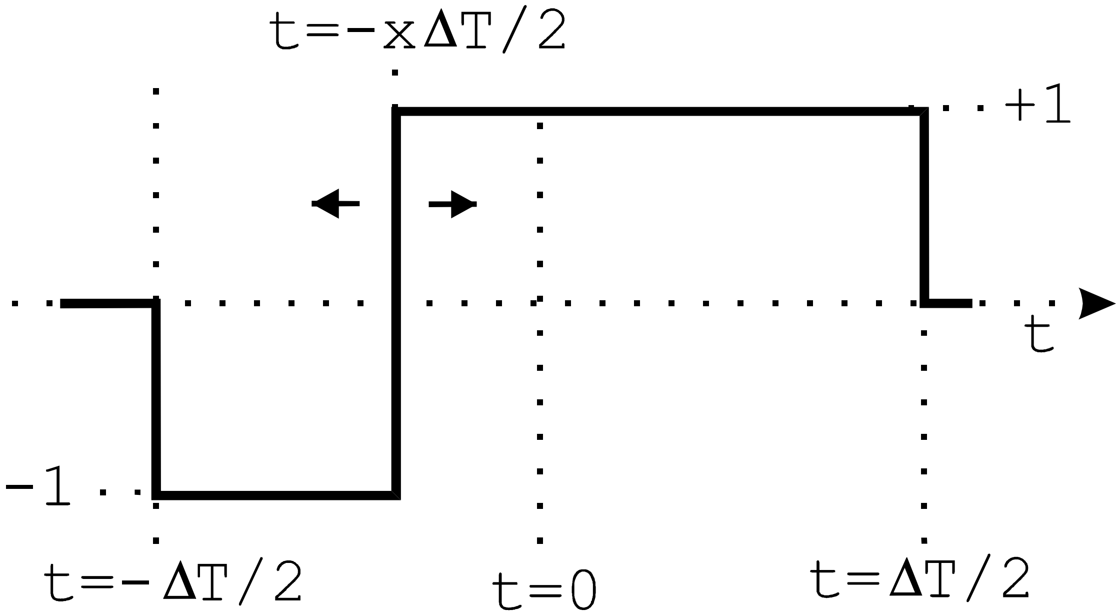
Pulse edges defined by a bit-clock F_b, $F_b = N \times F_s$



→ Coarse Quantization to N-levels

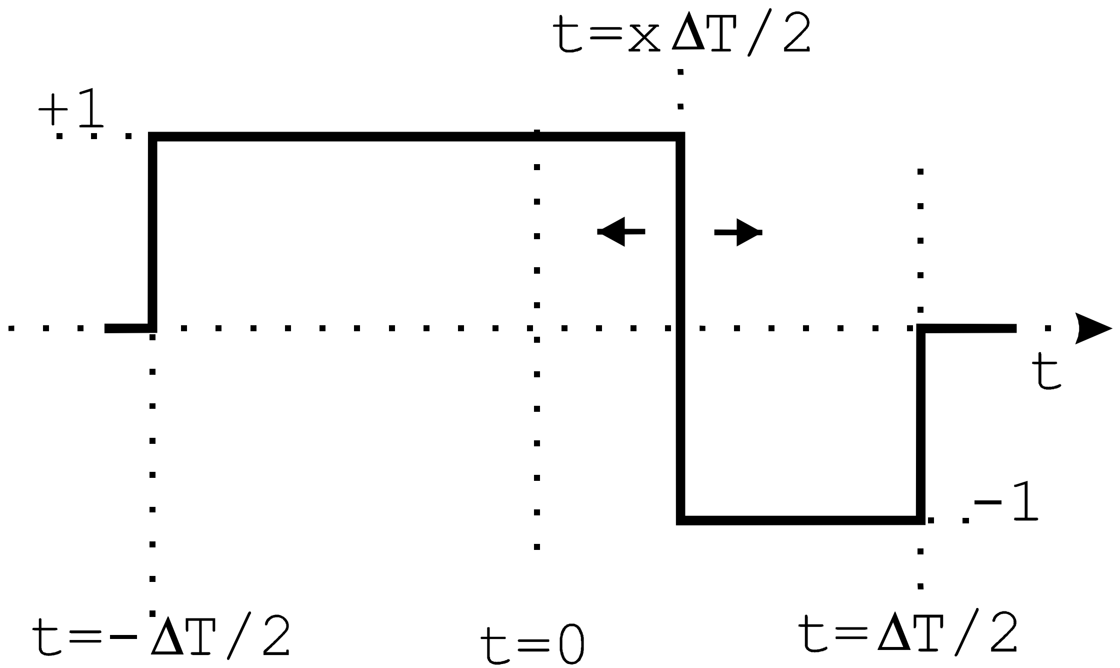
Pulse function: Single sided leading edge modulation

P(x,t):



Pulse function: Single sided trailing edge modulation

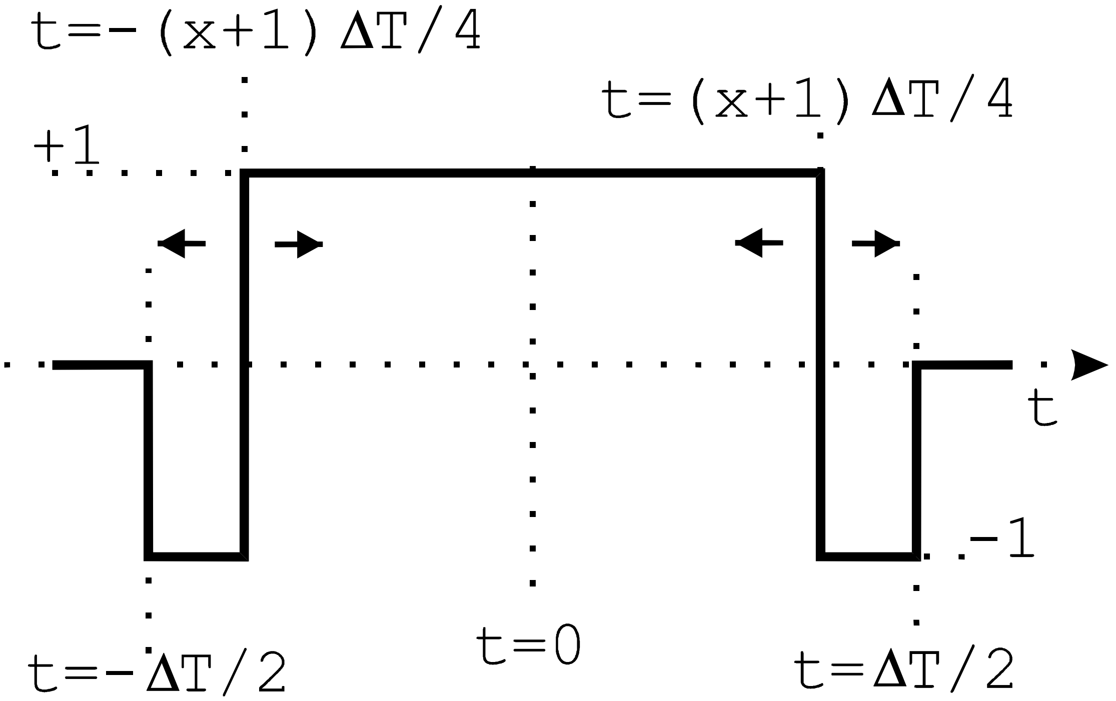
P(x,t):



Double Sided Symmetric Modulation

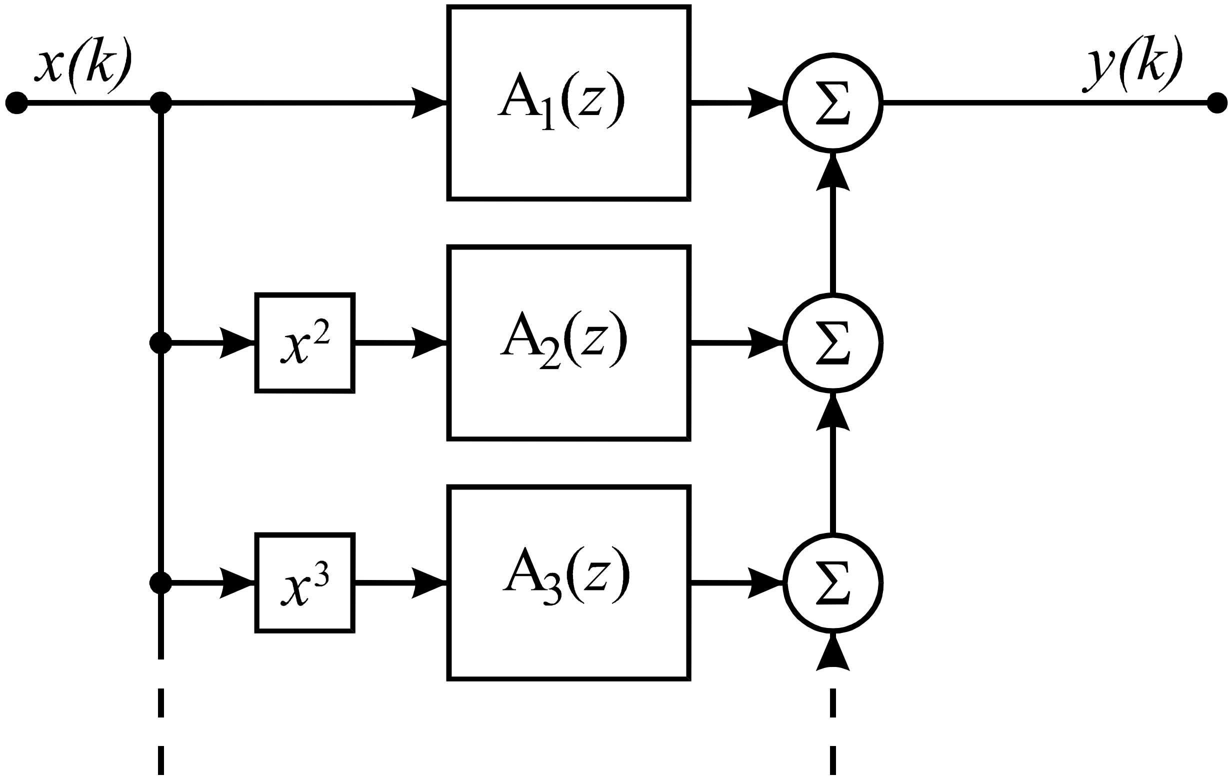
- **Both edges are modulated:**

P(x,t):



PWM Error Modeling

- **Hammerstein Model:**

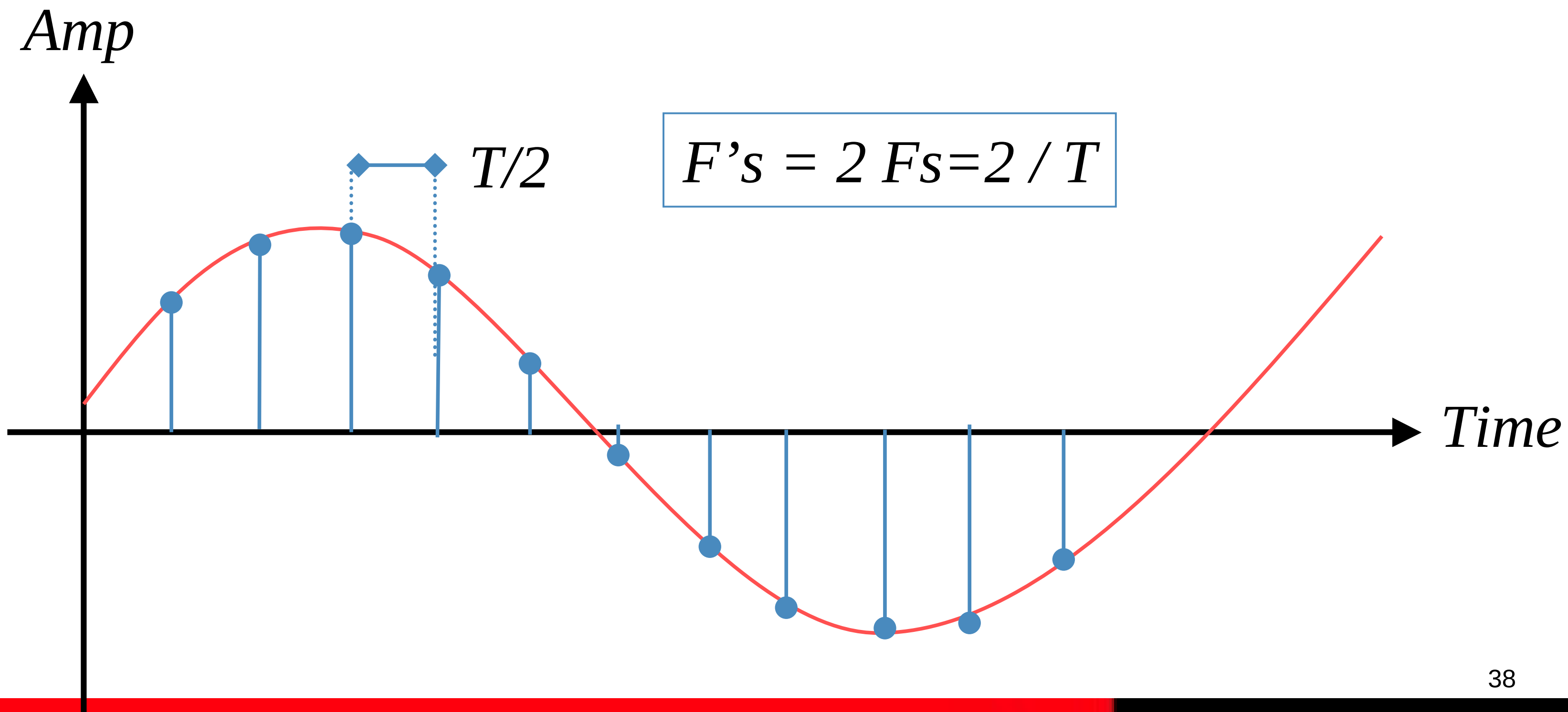


PWM errors

- **PWM errors:**
  - Harmonic + IM Distortion (rising with signal frequency)
  - Intermodulation noise due to intercations with the noise shaper (we will explain later)
- **Errors are corrected by digital signal processing in PCM domain prior to PWM (Equibit® algorithms):**
  - Harmonic + IM Distortion
  - Inter-Modulation (IM) Noise
  - Quasi-Symmetry (QS) Noise

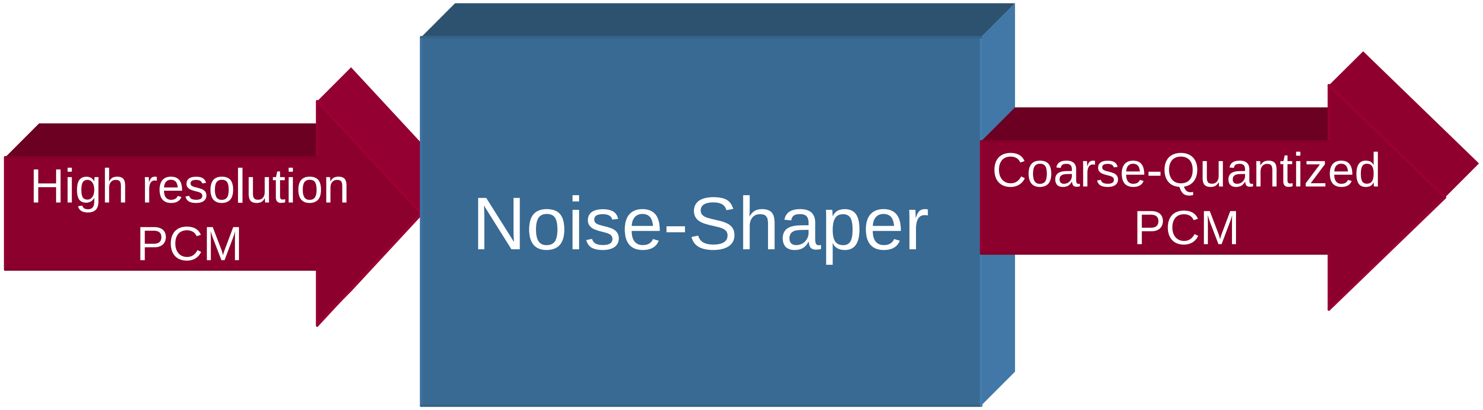
Oversampling

- Sampling much faster than the Nyquist Criterion
- Relaxed Filter requirements
- Allows room for Noise-Shaping



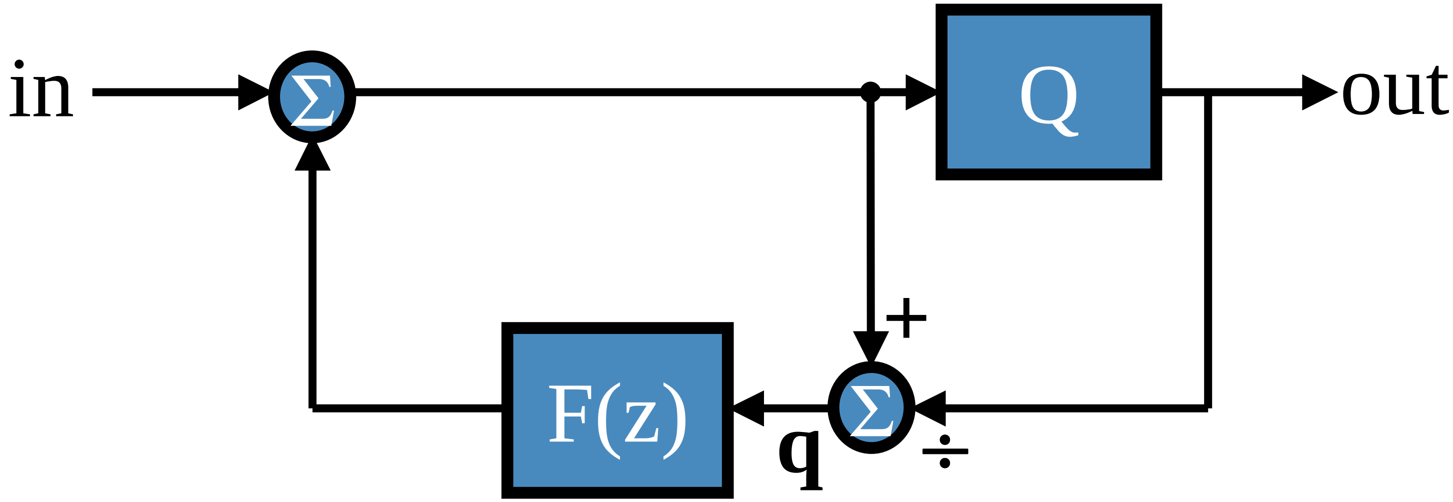
Noise-Shaping

- Interface to discrete UPWM: quantized output
- Spectral redistribution of the Q-noise
- Q-noise suppressed in the audio band

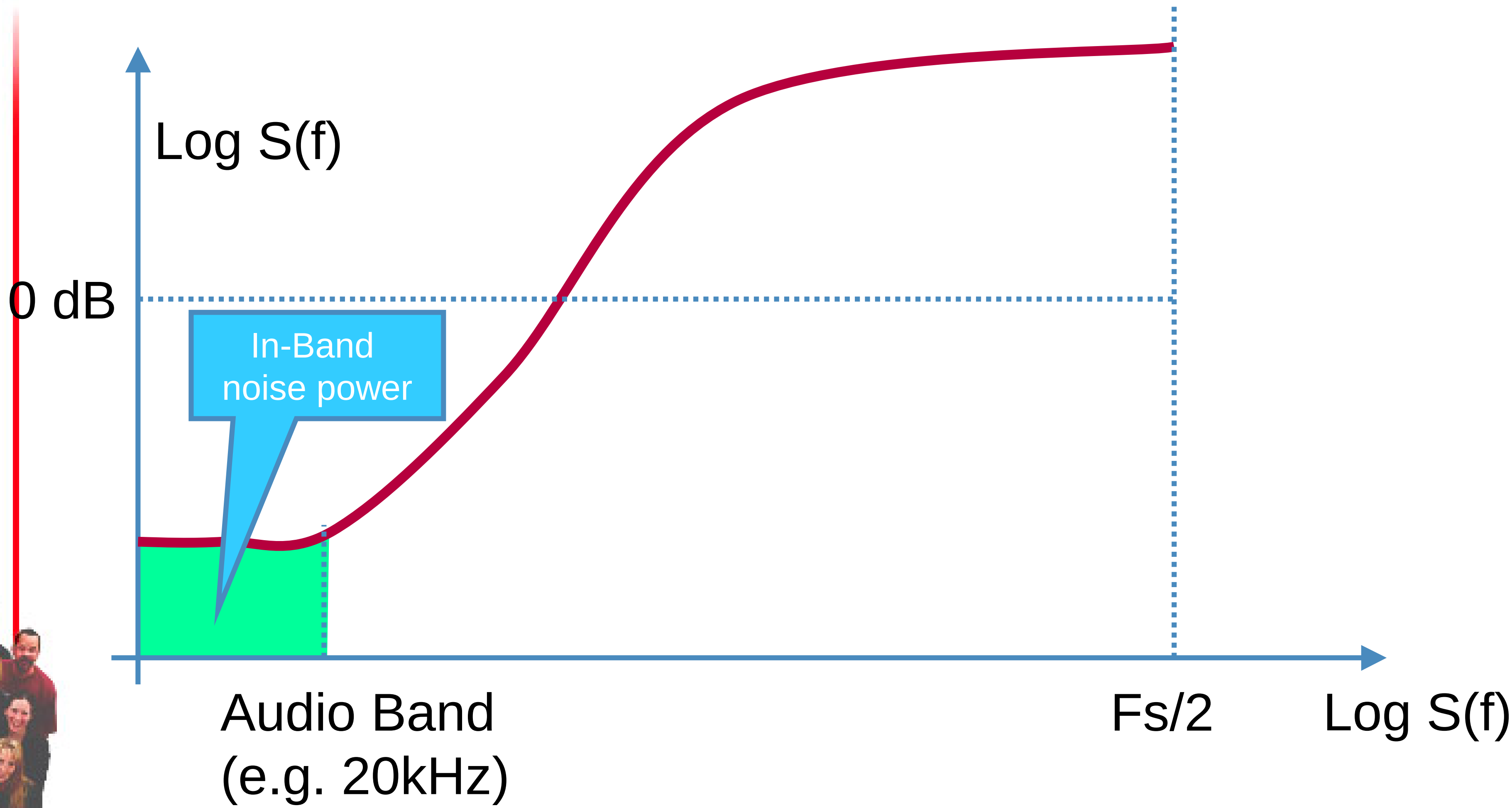


Typical Noise-Shaper

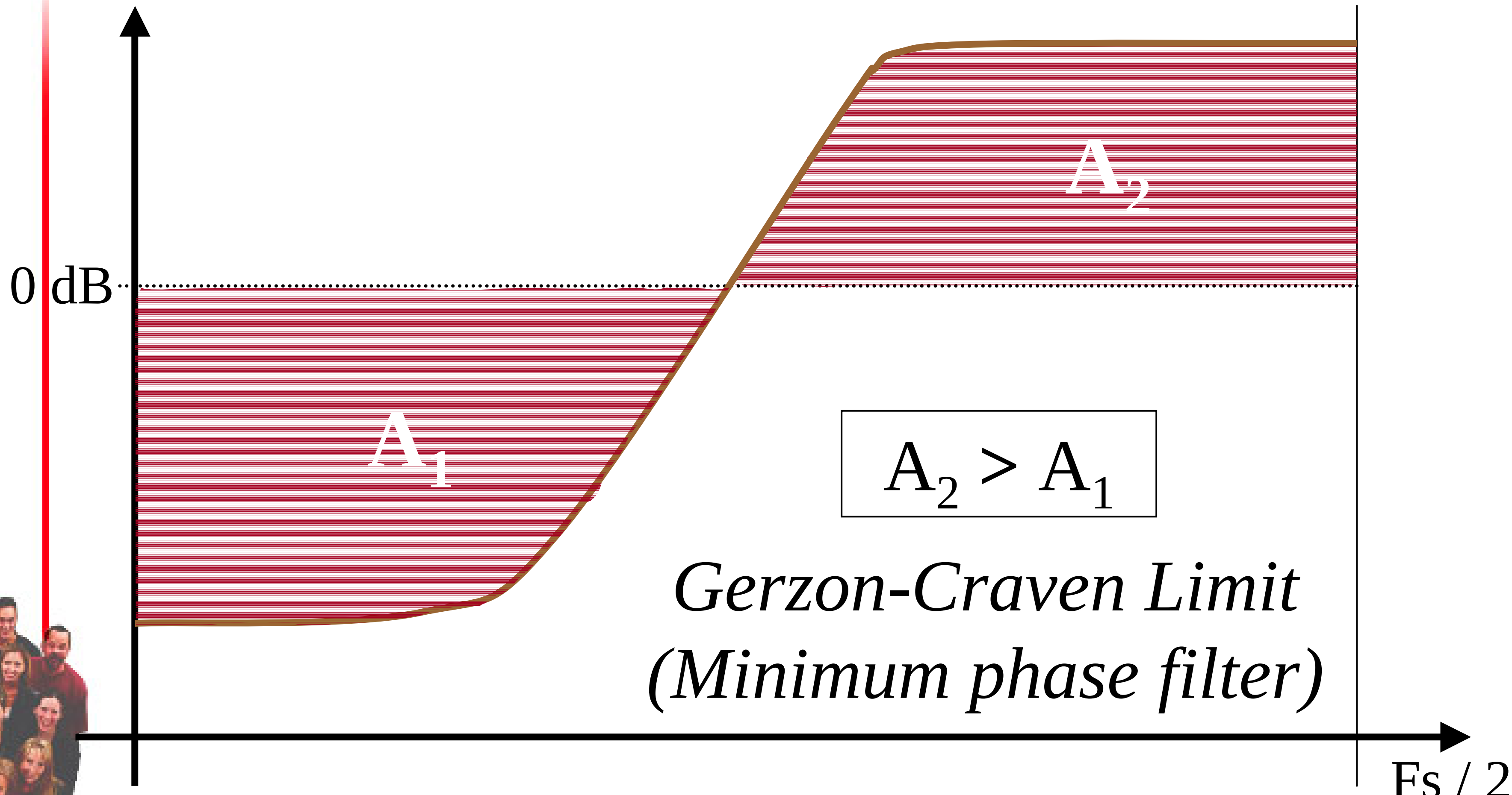
- The FeedBack filter F(z) weights past history of the Q-noise
- E.g. $F(z) = 2z^{-1} - z^{-2}$ gives a 2nd-order shaper



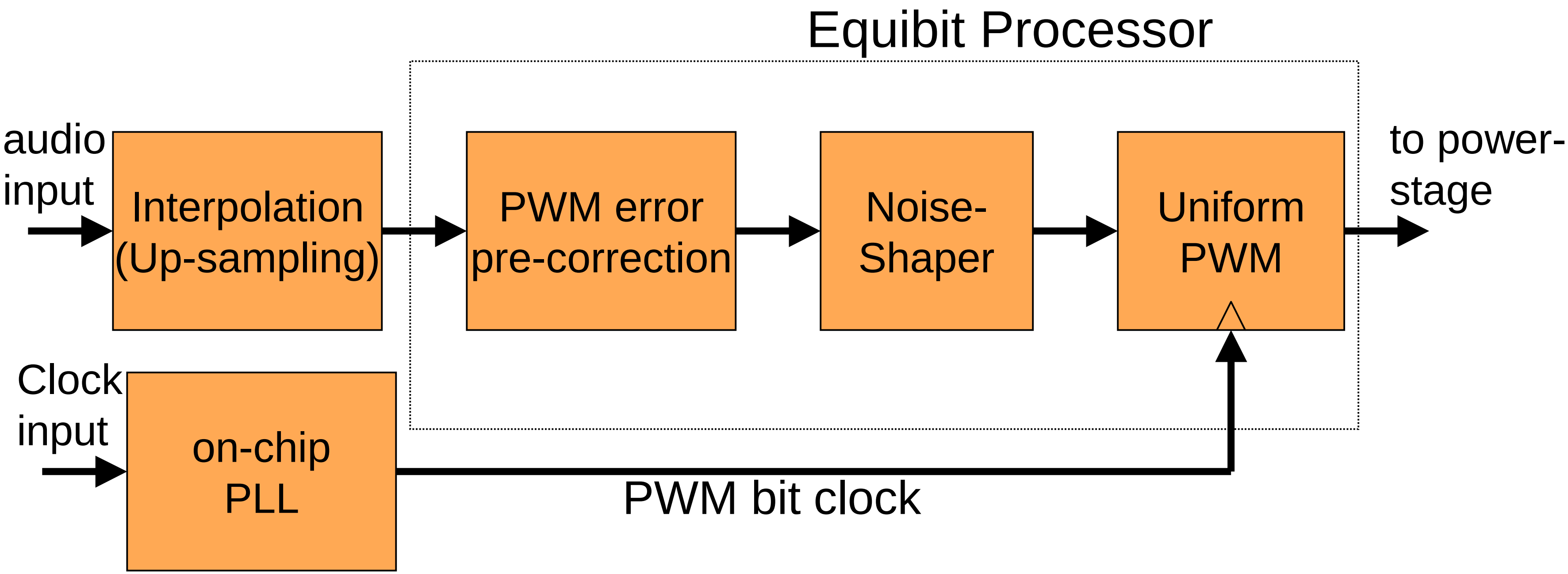
Noise-Shaper spectrum



Total Noise Amplification



PWM modulator Architecture



- **Fsw:352.8 /384 kHz (8x normal 44.1/48 rate)**
- **4/5th order noise shaper**
- **90.3/98MHz PWM bit clock (2048 fs-in)**
  - **Next generation runs at ~200MHz!**

PURE PATH DIGITAL
TEXAS INSTRUMENTS

Thursday lecture:
Power Stage Design