Signal Processing via Sampled-data Control – Beyond Shannon

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A moving image demo:

Superresolution via bicubic and Proposed filters

Where is the difference?

- Intersampling information
- Beyond the Nyquist frequency (Shannon limit)
-) Sampled-data control theory

Message of this talk

- You can do better signal processing using sampled-data control theory
- ⇒ Optimal recovery of freq. components beyond the Nyquist freq. (=1/2 of sampling freq.)

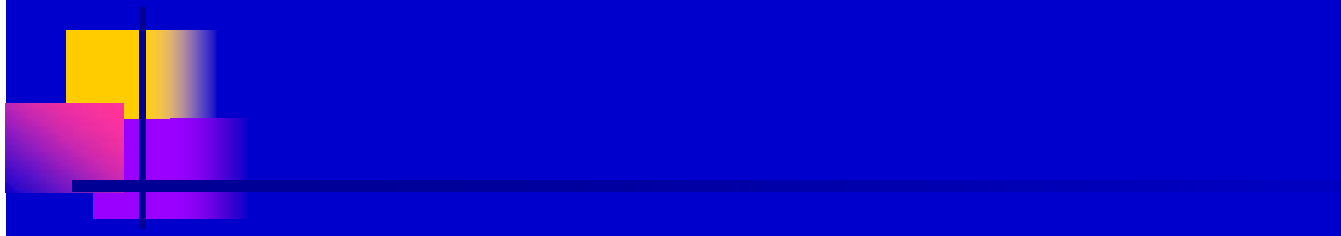
Let's first listen to a demo

Sound demo:

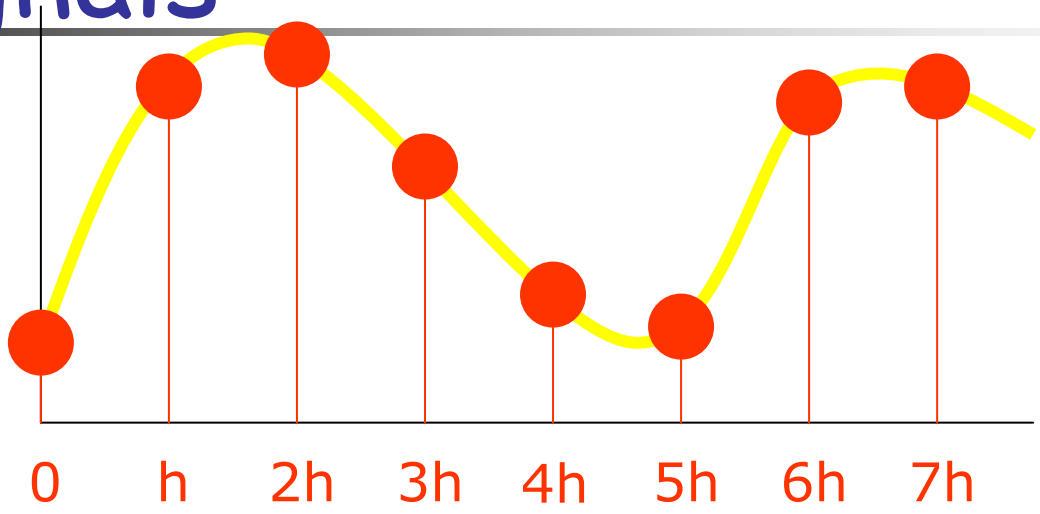
Red: Original (up to 22kHz)
Blue: downsampled to 11k, and then processed 4 times upsampled via YY filter

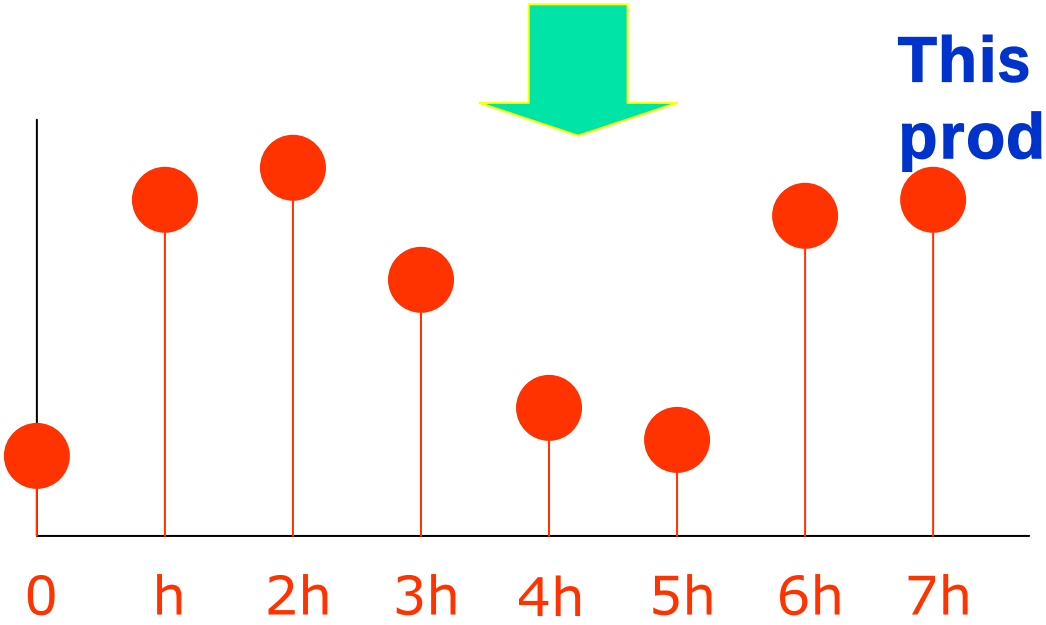
Did you hear the difference?

Part I: Current digital signal processing – Basics☺



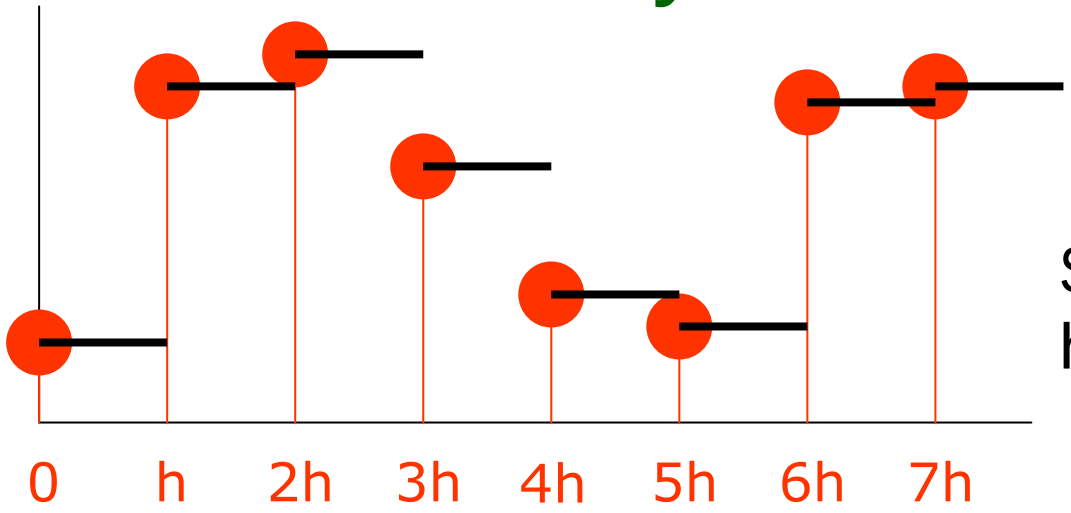
Sampling continuous-time signals





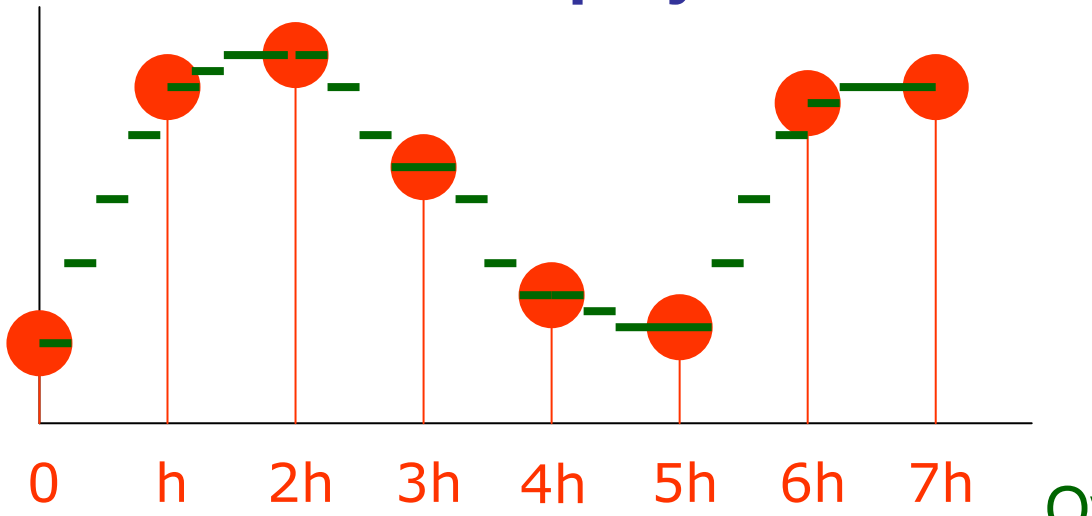
This does not produce a sound

Hold device is necessary



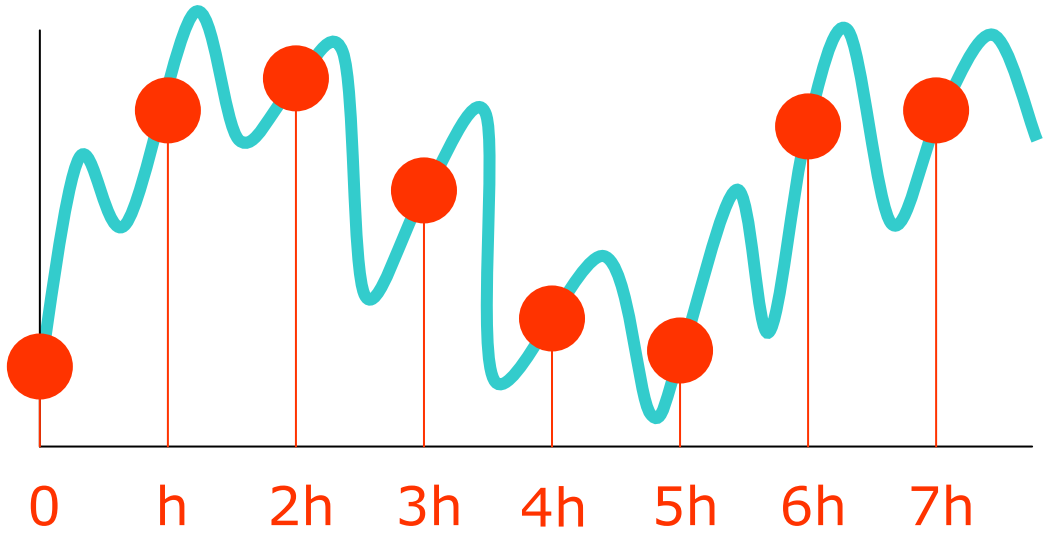
Simple 0-order hold

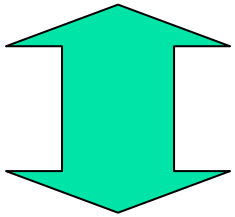
Old CD players

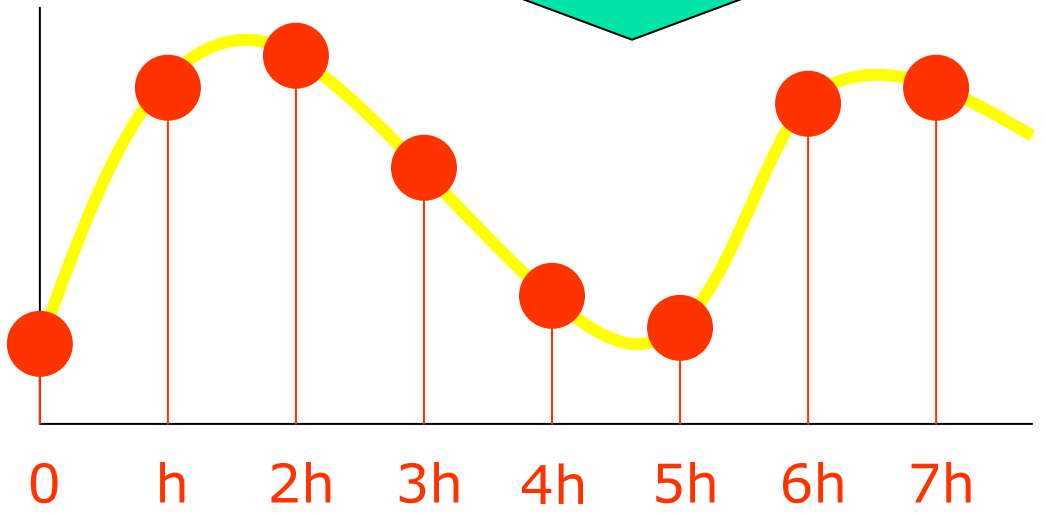


Oversampling DA converter

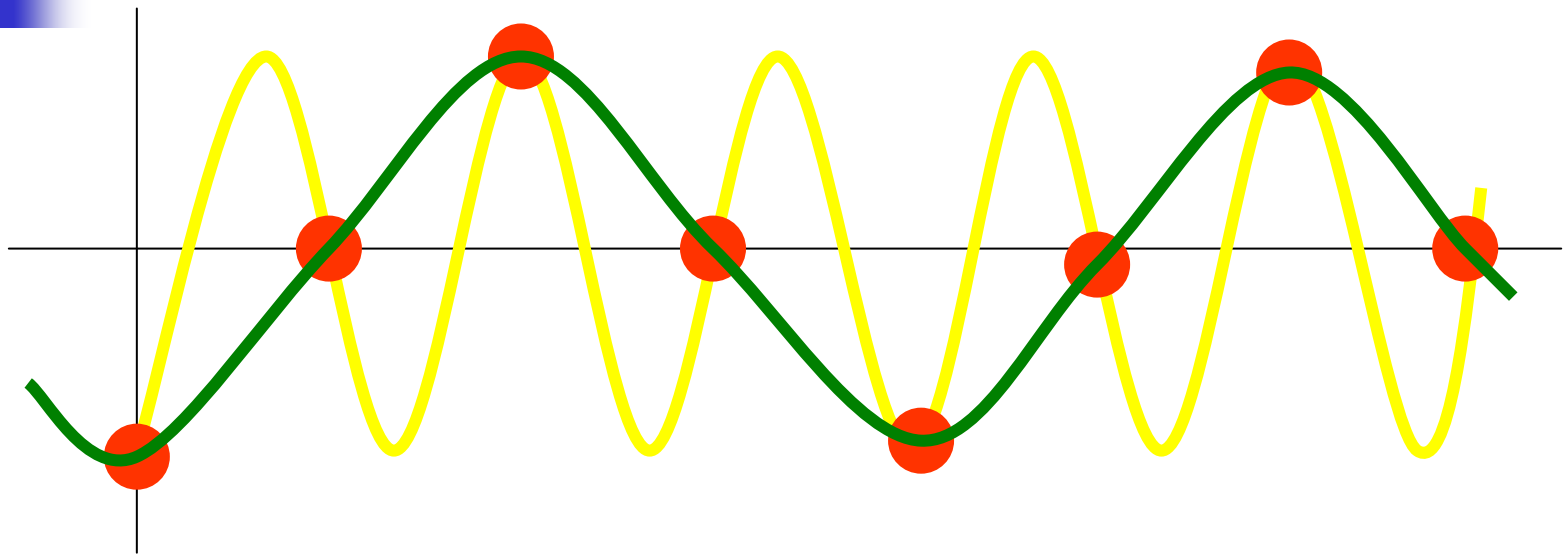
More recent players


0
h
2h
3h
4h
5h
6h
7h




0
h
2h
3h
4h
5h
6h
7h

Typical problem: Sampling ! Aliasing



- High-freq. intersample information can be lost
- If no high-freq. components beyond the Nyquist frequency (=1/2 of sampling freq.) ! unique restoration

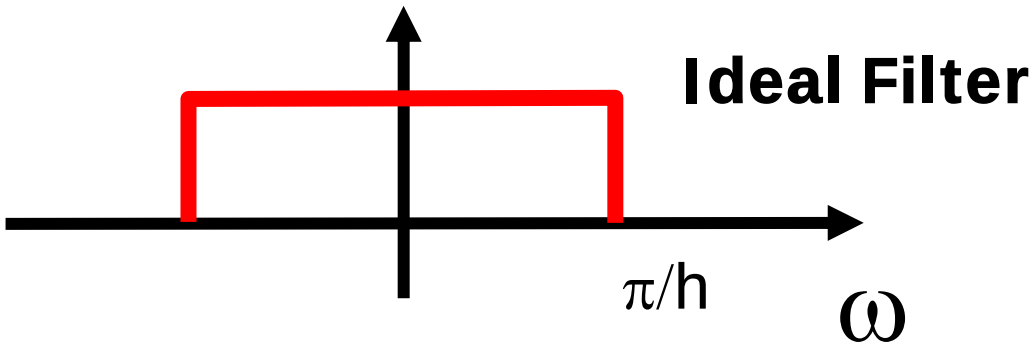
  ! Whittaker-Shannon-Someya sampling theorem

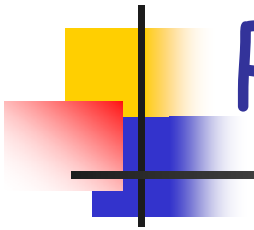
Sampling Theorem

- Band limiting hypothesis) unique recovery

$$\hat{f}(j\omega) = 0 \text{ for } |\omega| > \pi/h \Rightarrow$$

$$f(t) = \sum_{n=-\infty}^{\infty} f(nh) \frac{\sin \pi(t/h - n)}{\pi(t/h - n)}$$

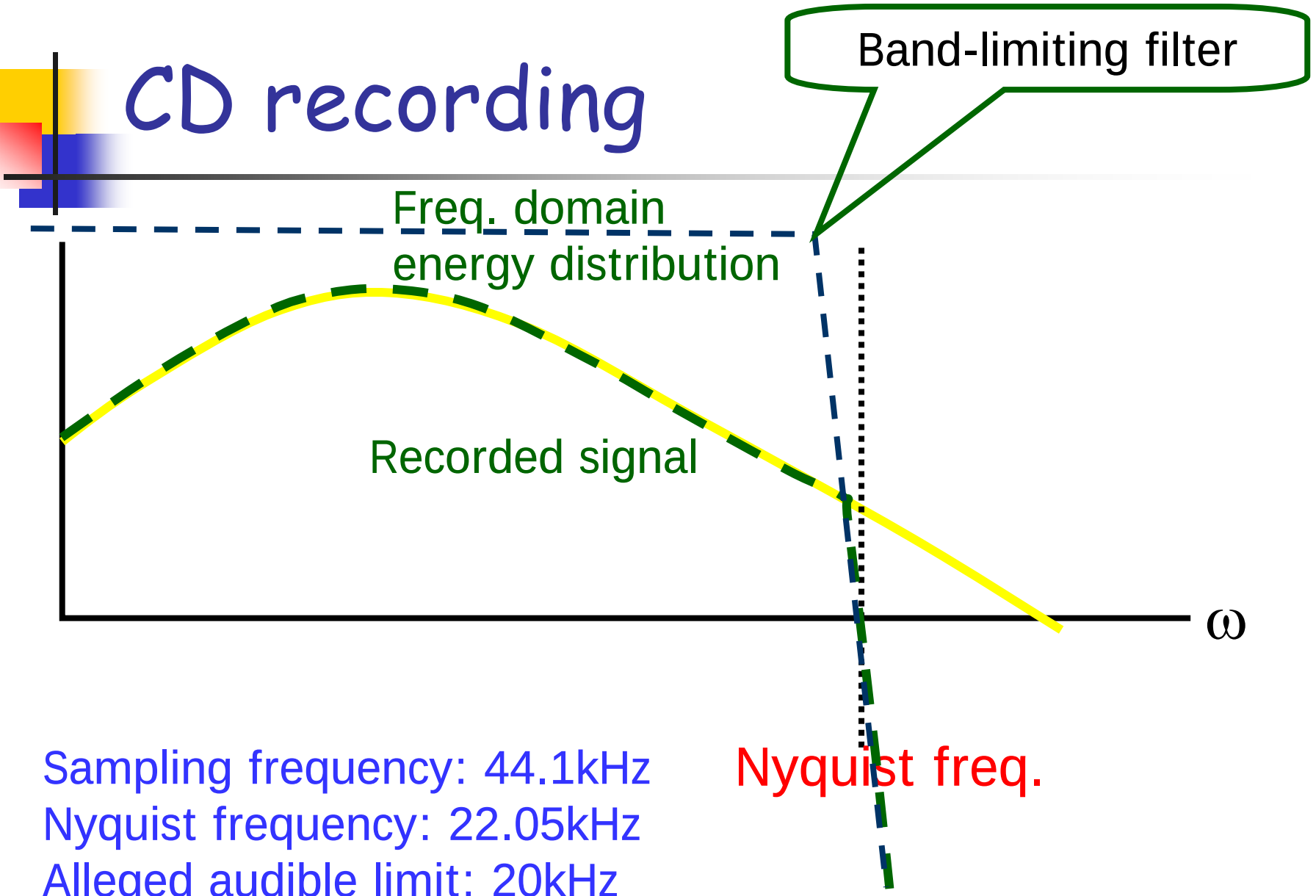




Problems

- Band limiting hypothesis unrealistic
- The formula not causal (you need infinitely many future sampled values)
- Very slow convergence
- Must approximate the ideal filter in practice
- ⇒ Sharp cut-off characteristics
  - High degree of filters
  - Distortion due to the Gibbs phenomenon

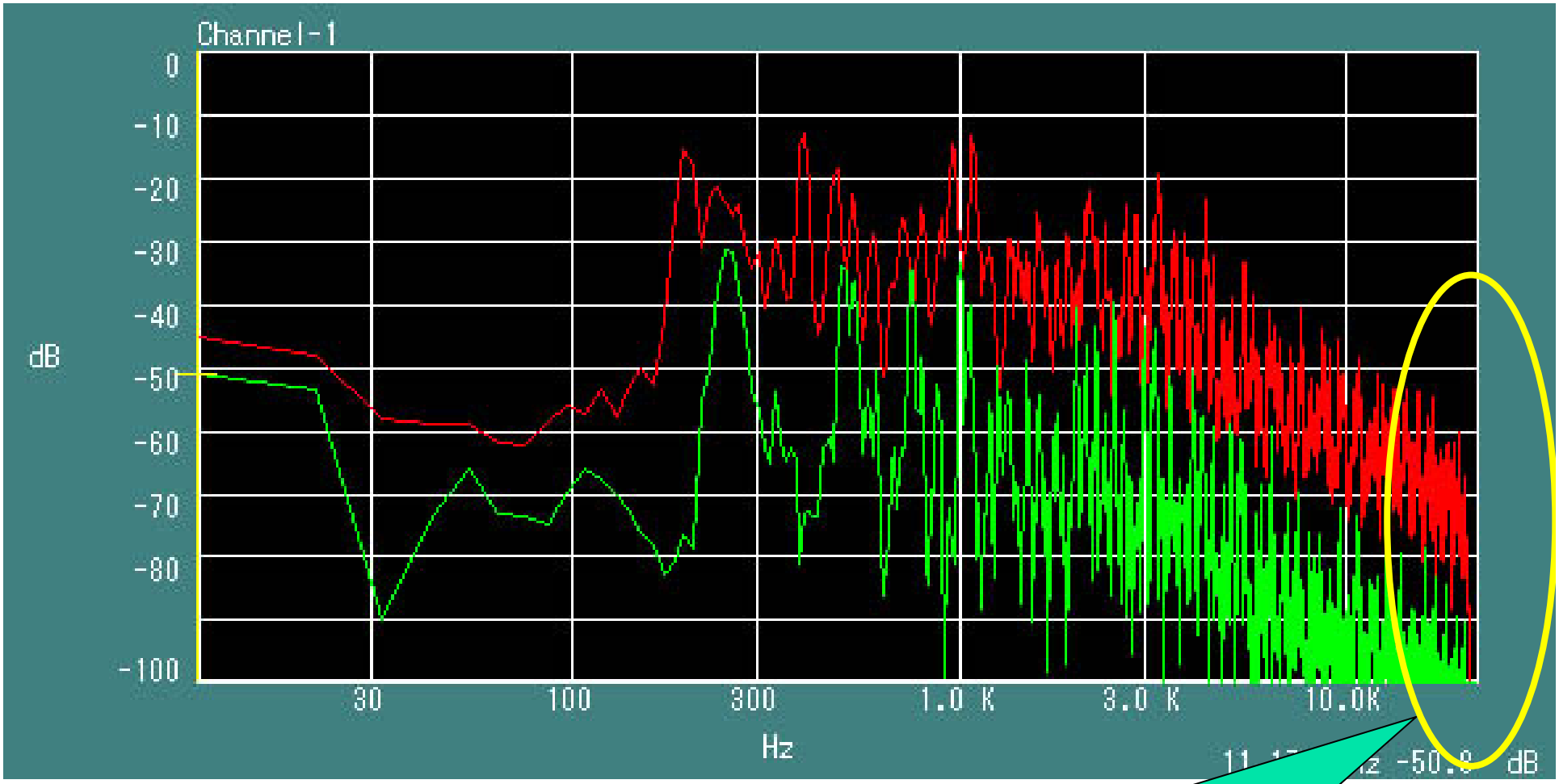
CD recording



Sampling frequency: 44.1kHz
Nyquist frequency: 22.05kHz
Alleged audible limit: 20kHz

Example: Digital Recording (CD): sharp anti-aliasing filter

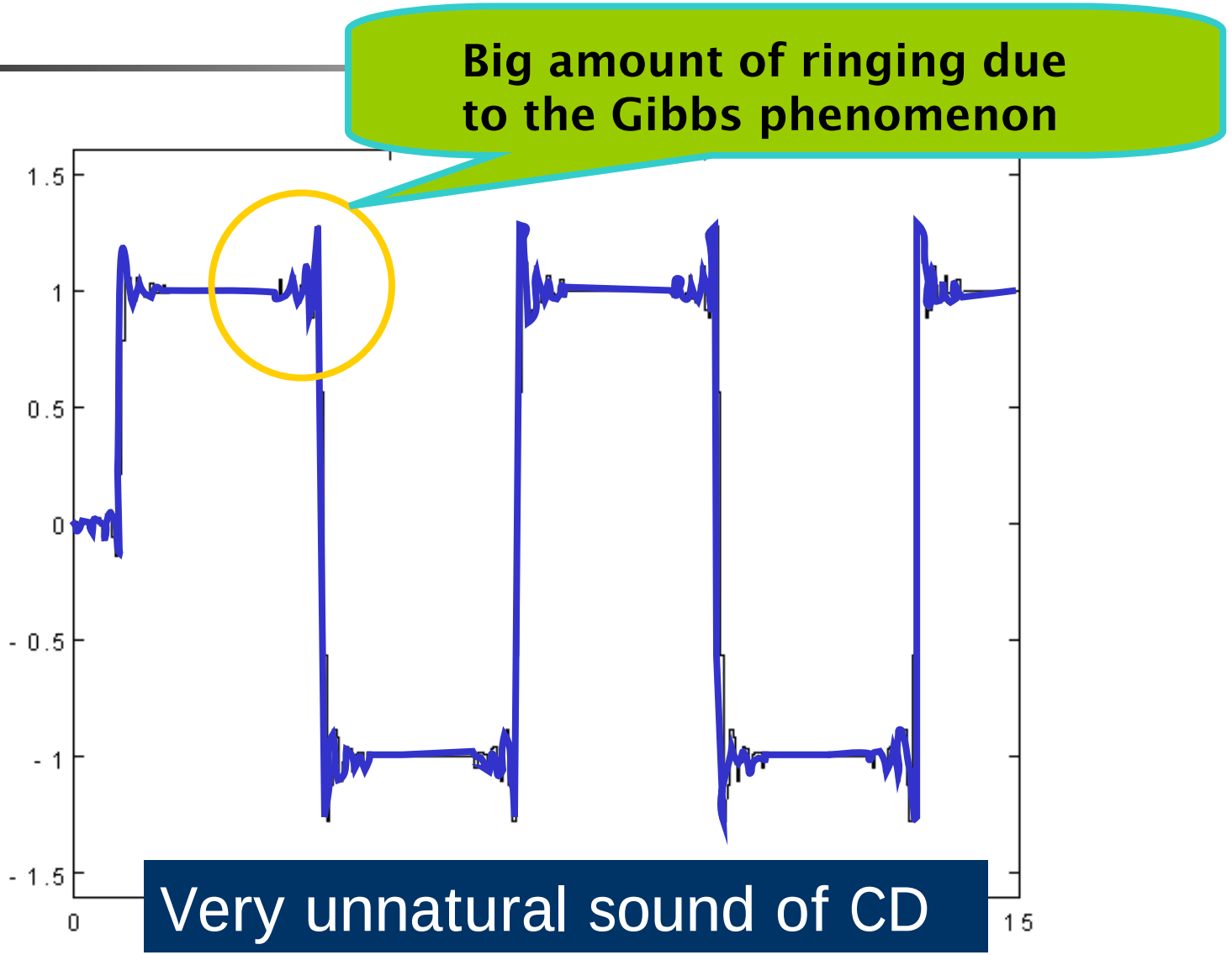
No signal beyond 20kHz



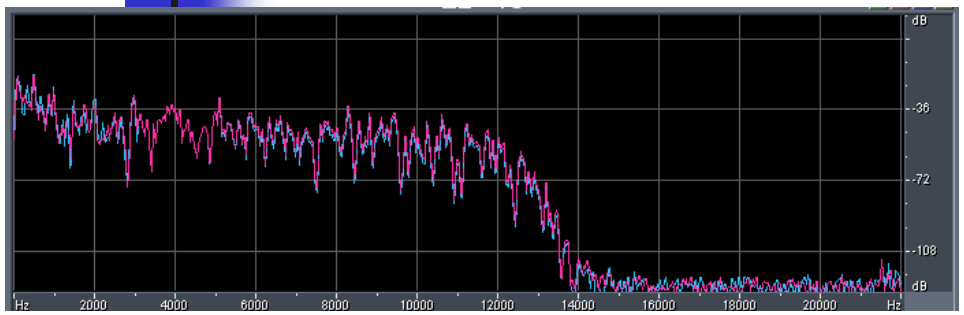
But you won't be able to hear them anyway??

Very sharp anti-aliasing filter

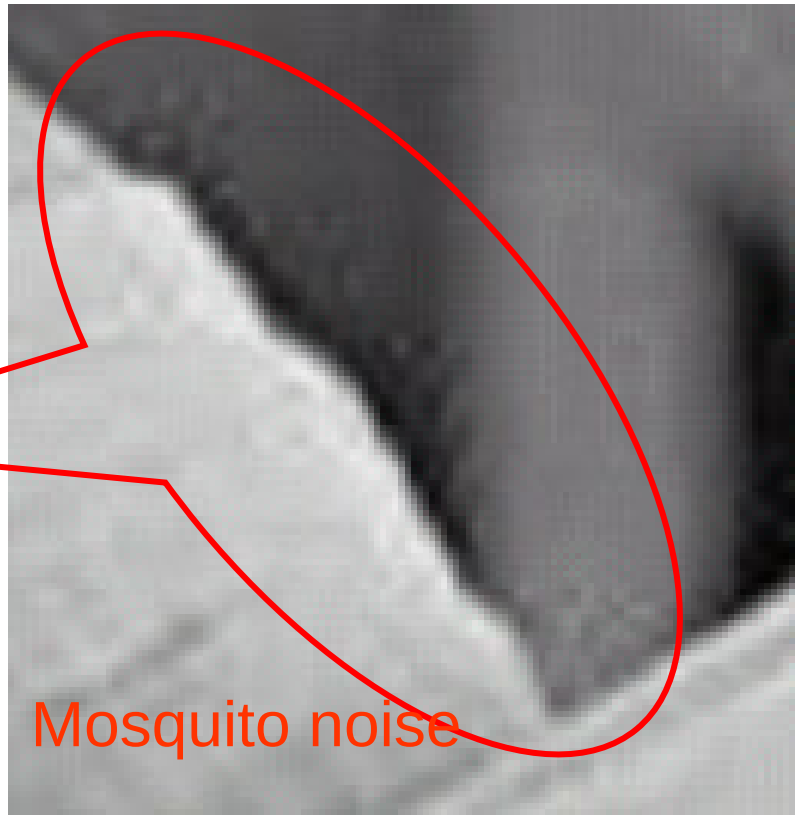
Effect of a band-limiting filter

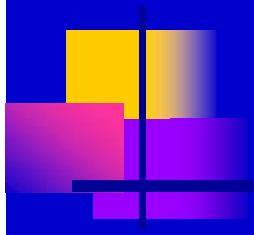


Mosquito Noise



Truncated freq. response

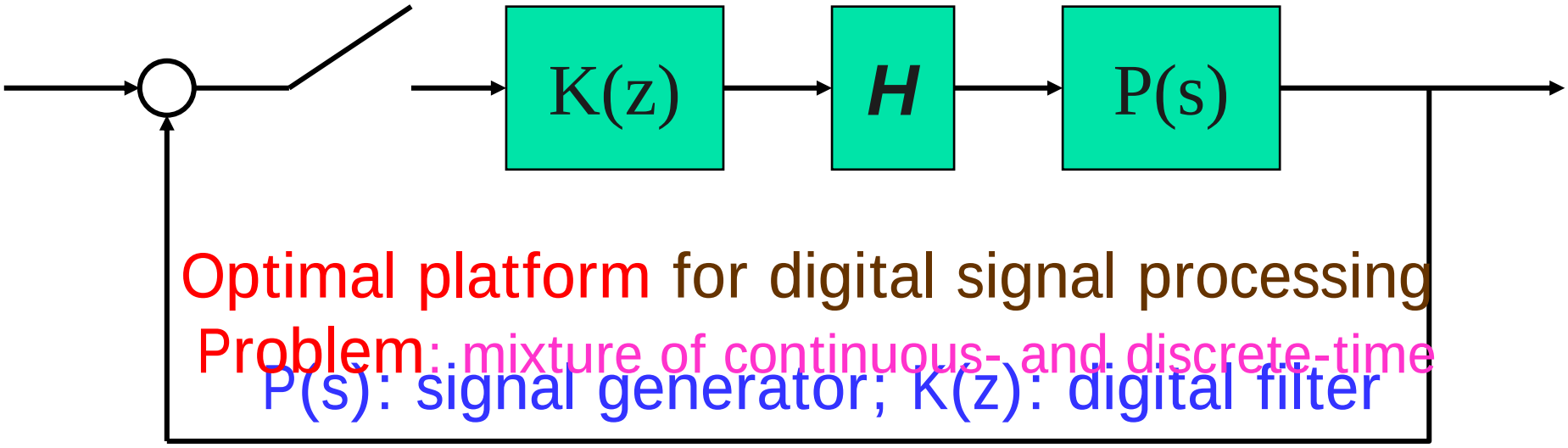




Part II: Why sampled-data theory effective?
- Review of the theory

Sampled-data Control Systems – What are they?

- Continuous-time plant
- Discrete-time controller
- sample/hold devices



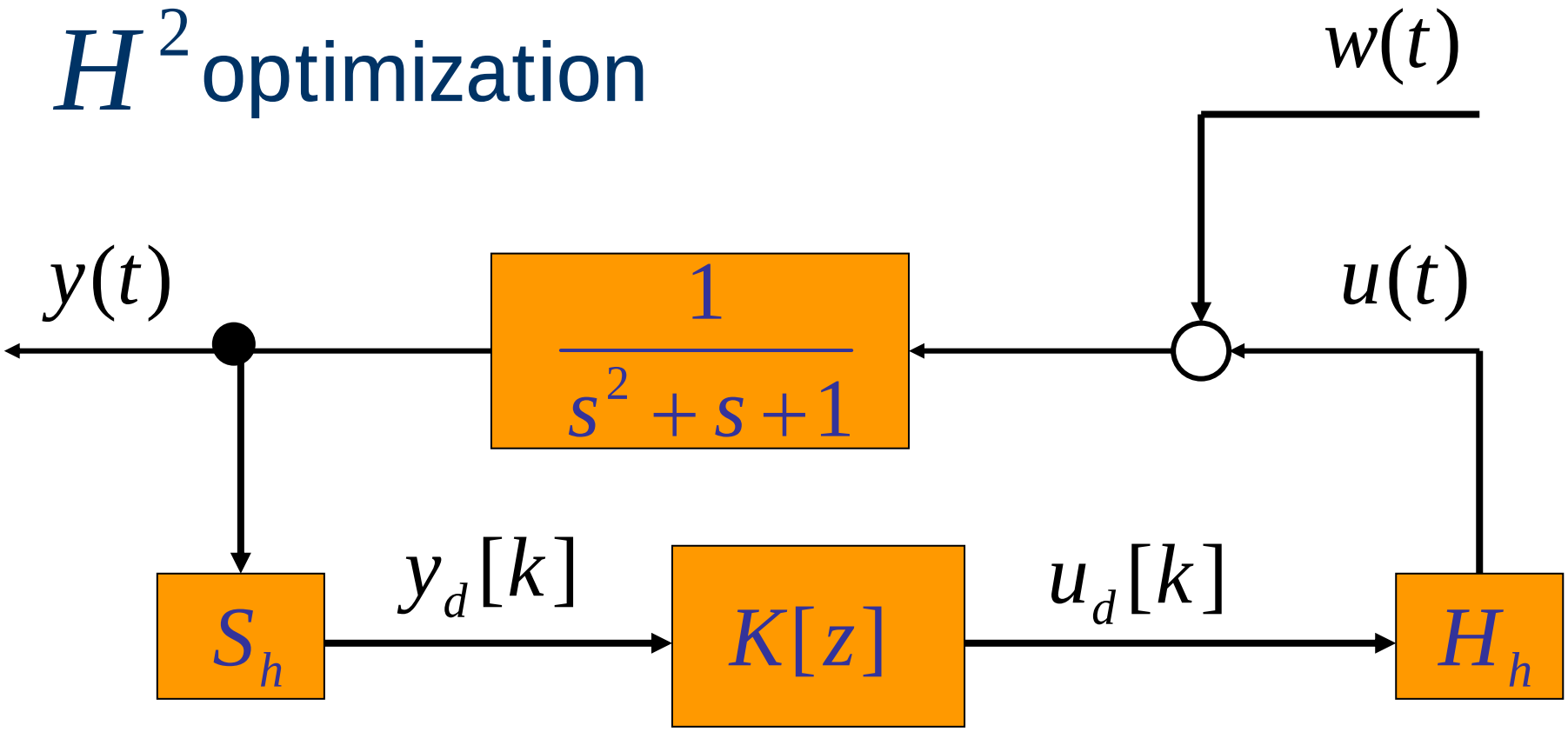
Optimal platform for digital signal processing

Problem: mixture of continuous- and discrete-time

P(s): signal generator; K(z): digital filter

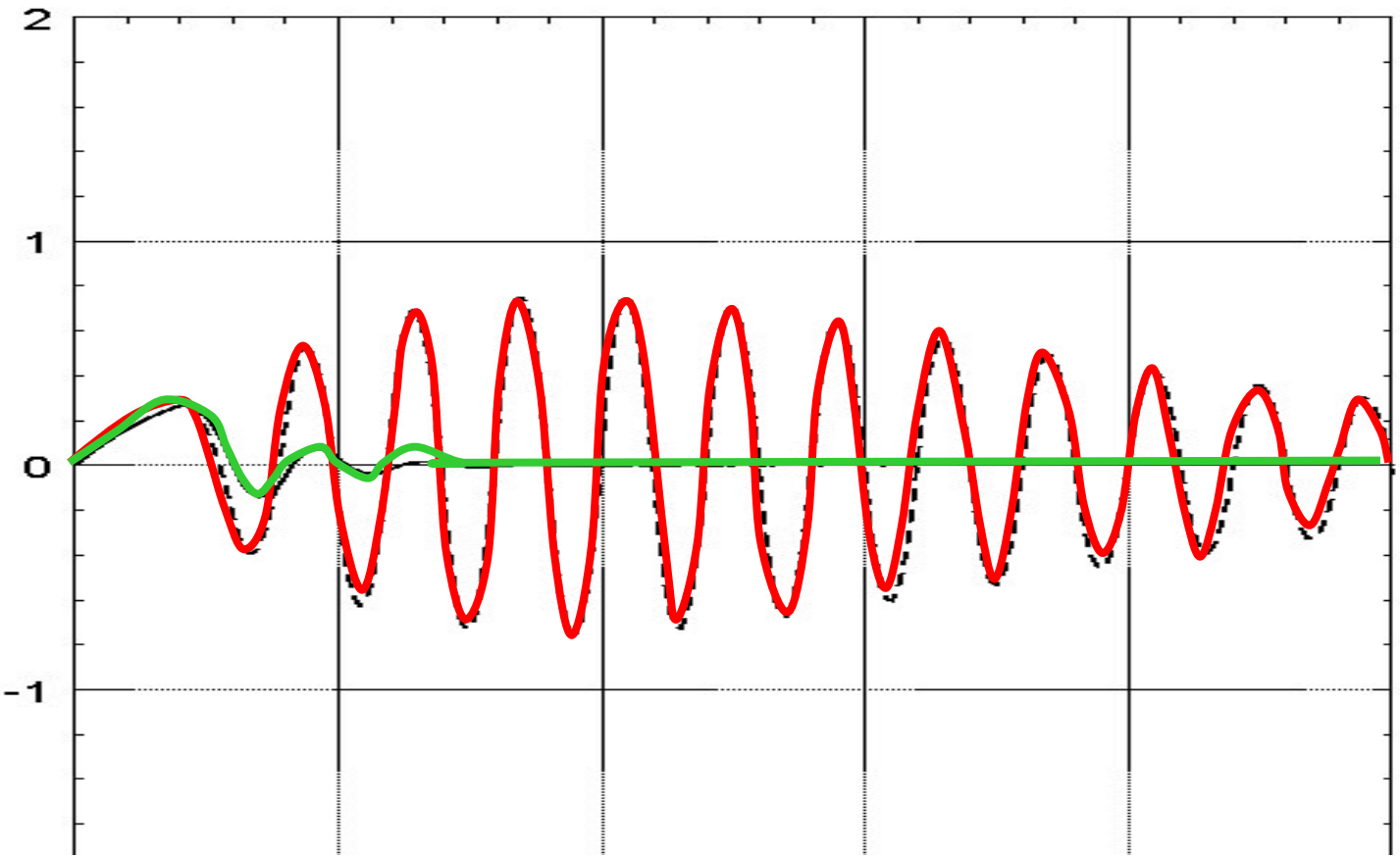
Does this make a difference? ---Yes

H^2 optimization



a) Discrete-time H^2 with no intersample consideration
b) sampled-data design

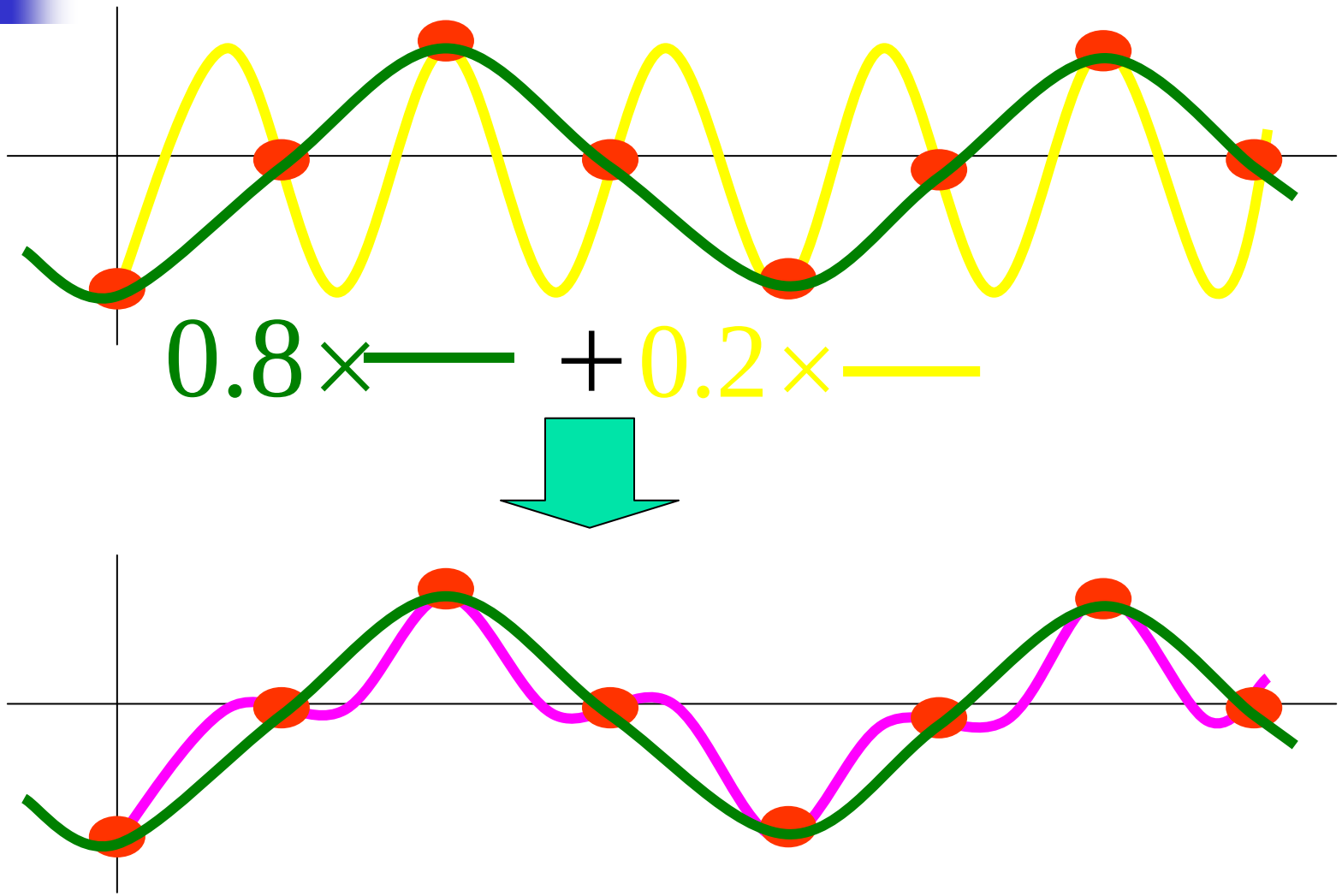
Time Response



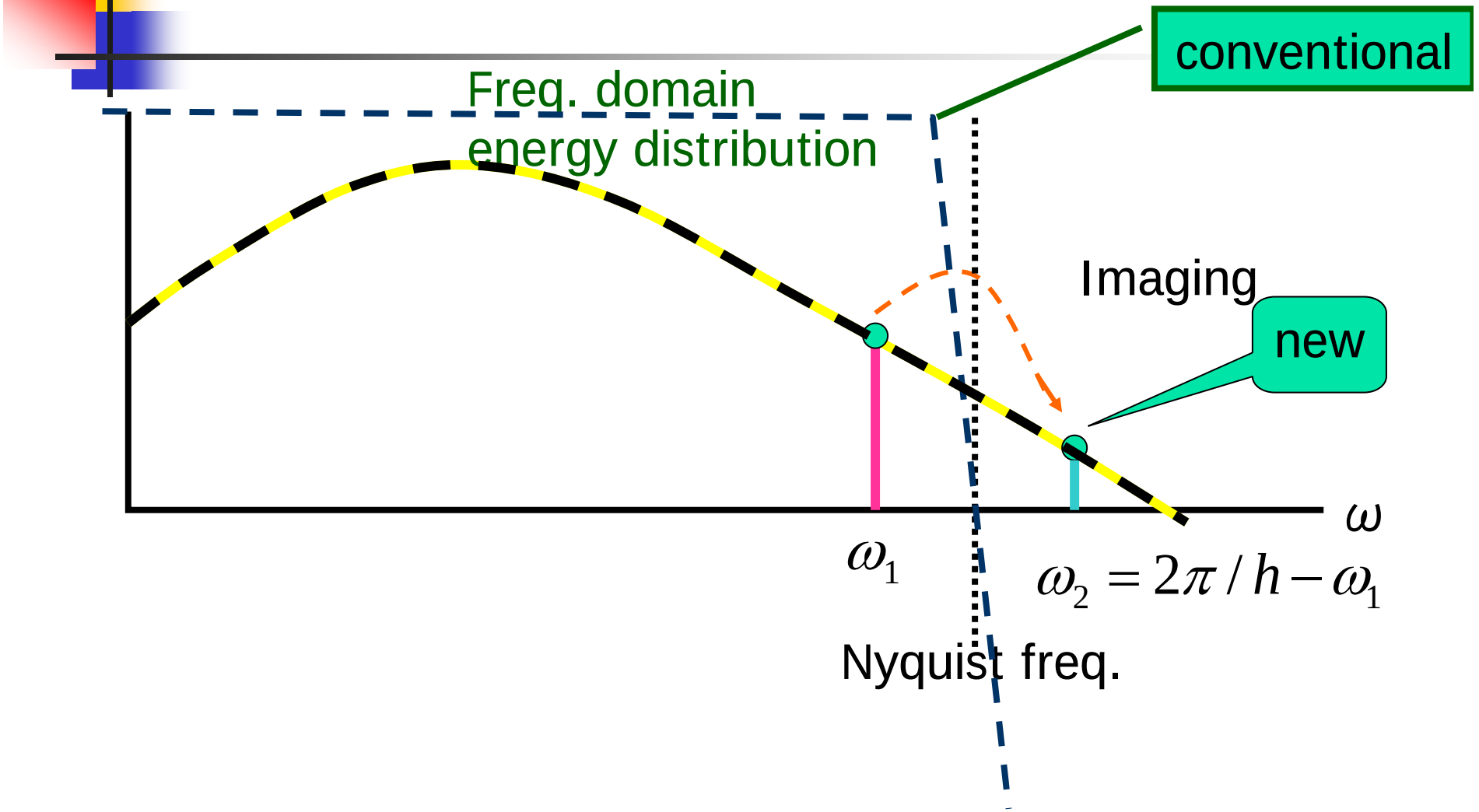
a) Discrete-time H^2 with no intersample consideration
b) sampled-data design

Part III: How can sampled-data theory help signal processing?

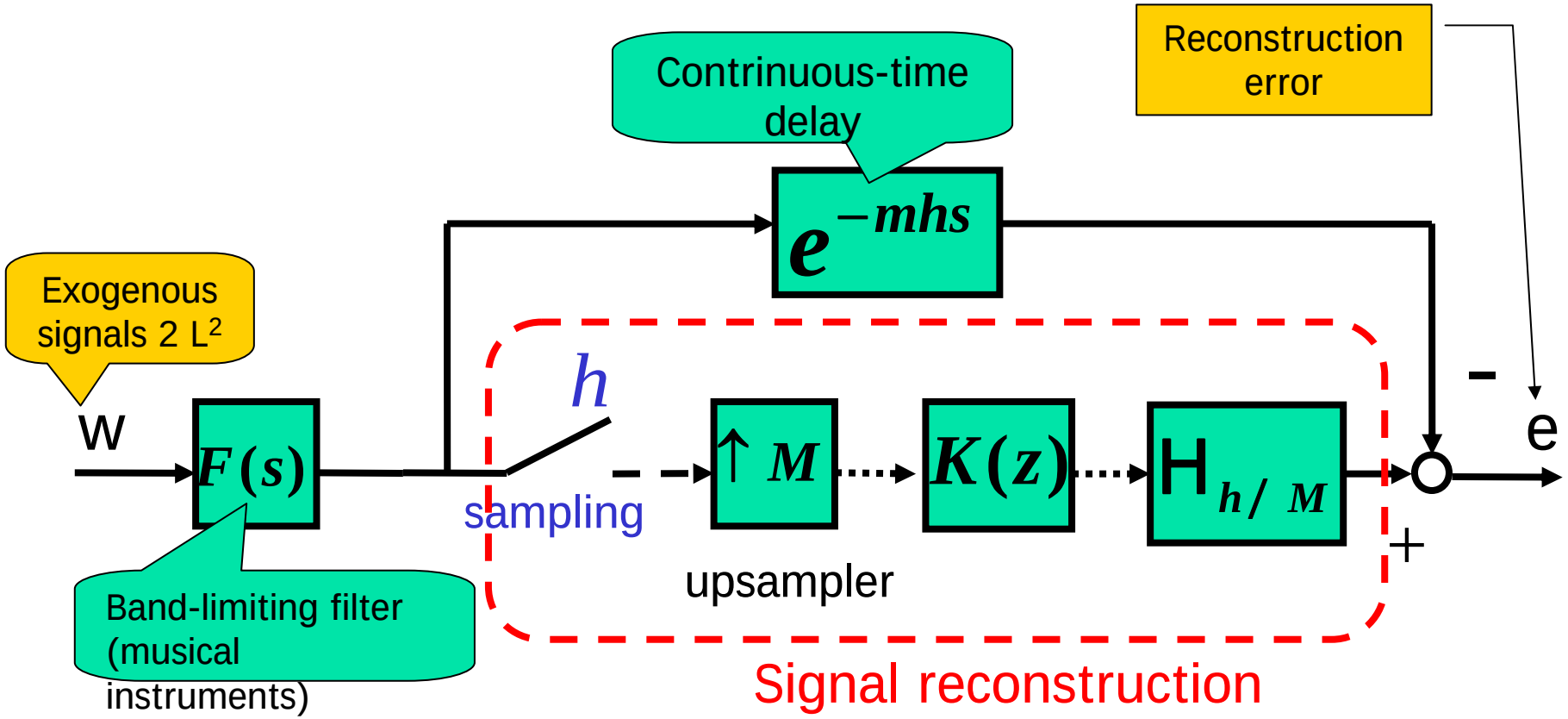
With a little bit of a priori information…



Utilizing analog characteristic



Sampled-data Design Model



Problem: Find K[z] satisfying

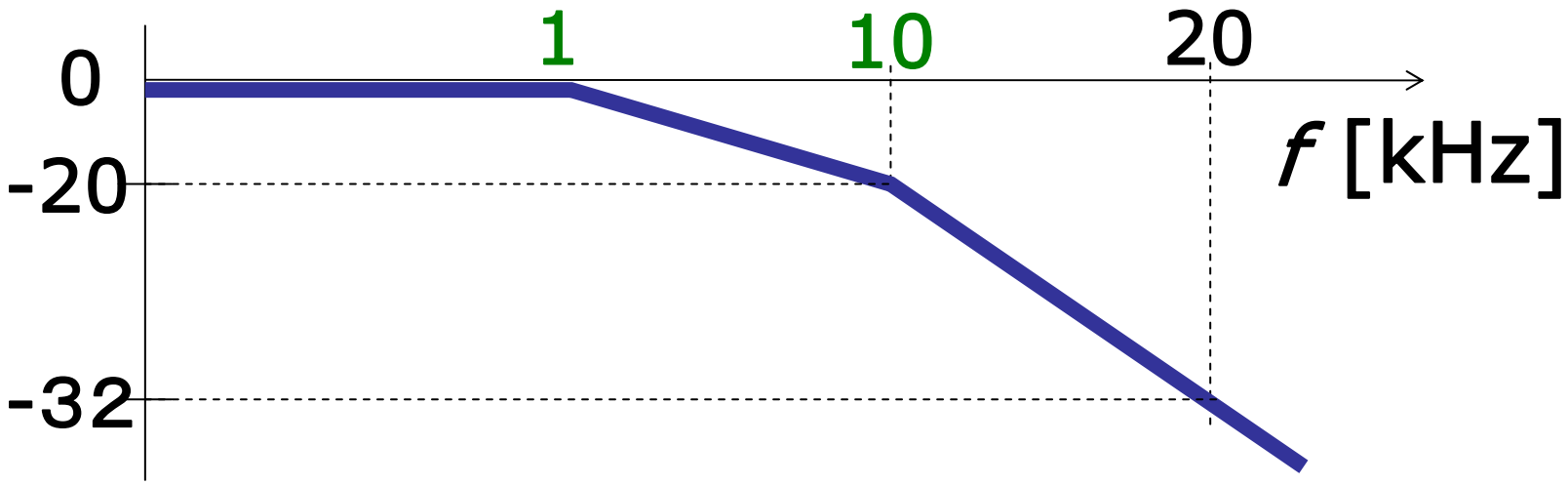
$$\|T_{ew}\| < \gamma$$

Sampled-data H^1 control problem

Example of F(s)

Sampling period h=1, delay step m=2

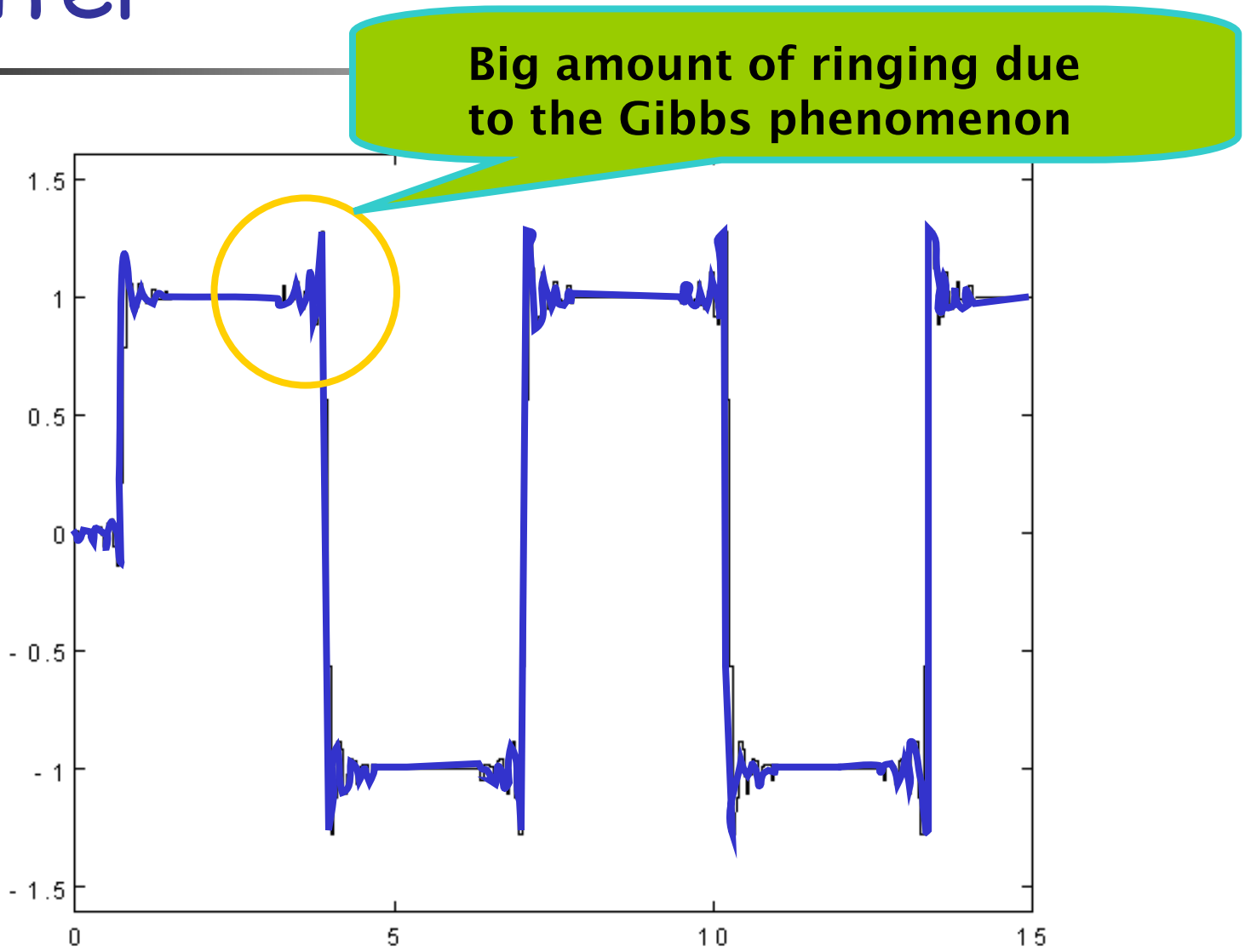
$$F(s) = \frac{1}{(0.70187s+1)(7.0187s+1)}$$

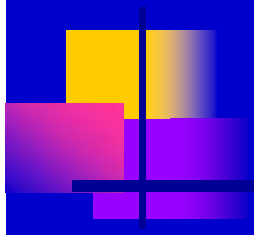


Interpolator via the proposed method

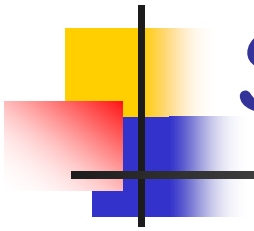
Proposed
Square wave resp.

Response of the Johnston filter





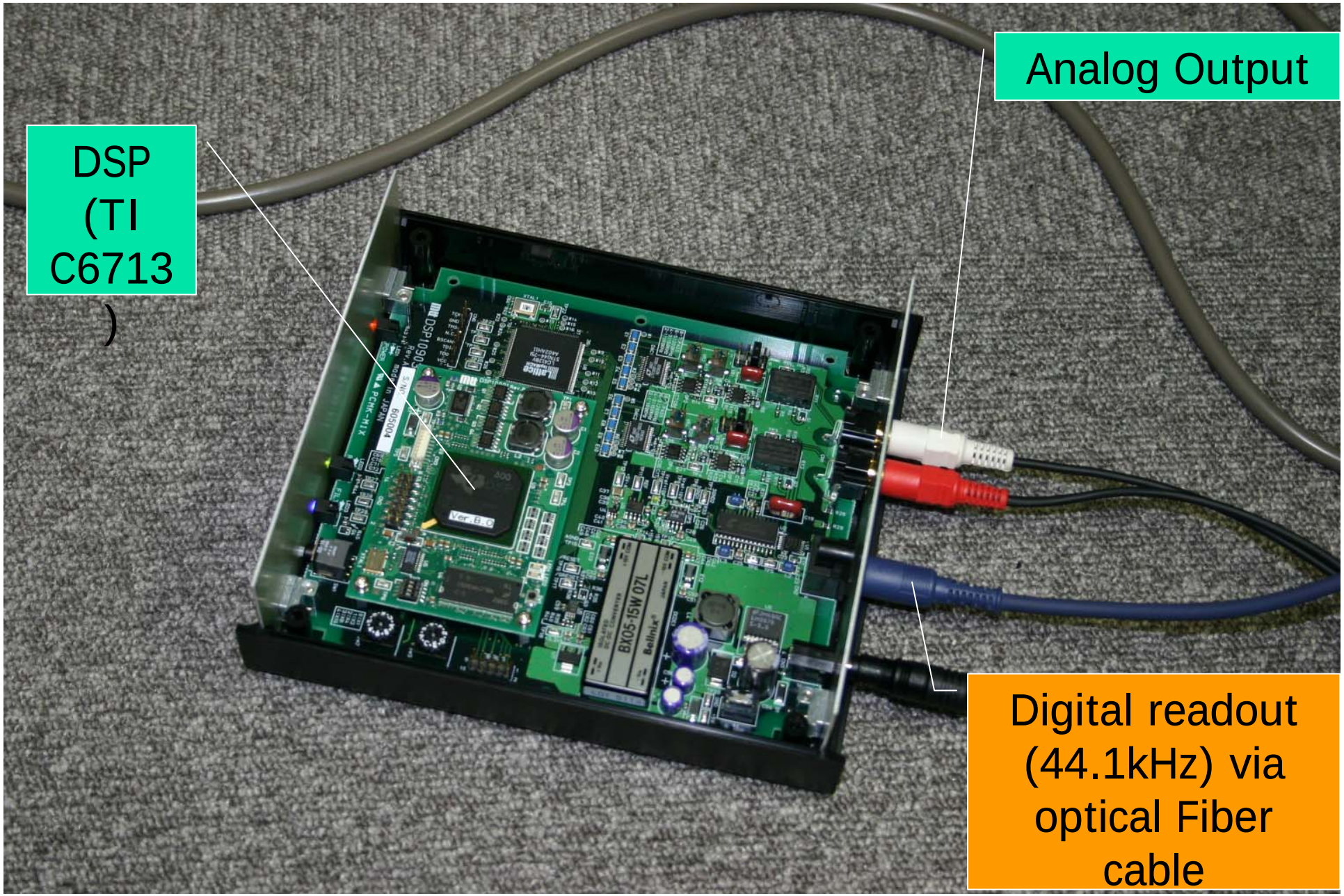
Part IV: Application to Sound Restoration



Sound restoration

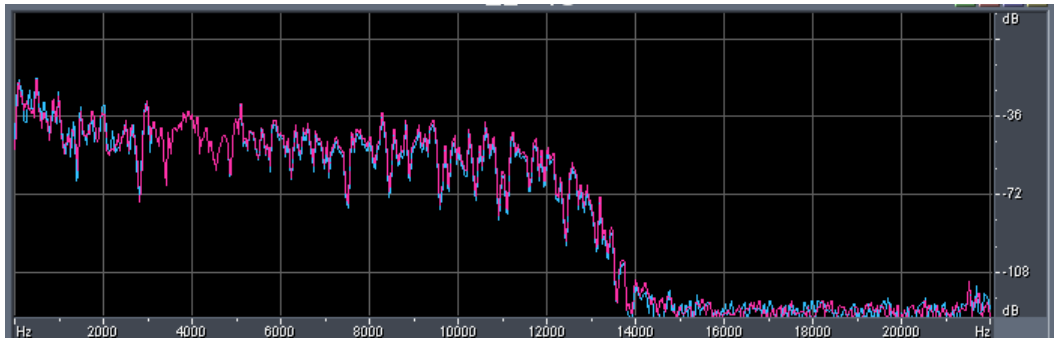
Sound demo:

YY Lab

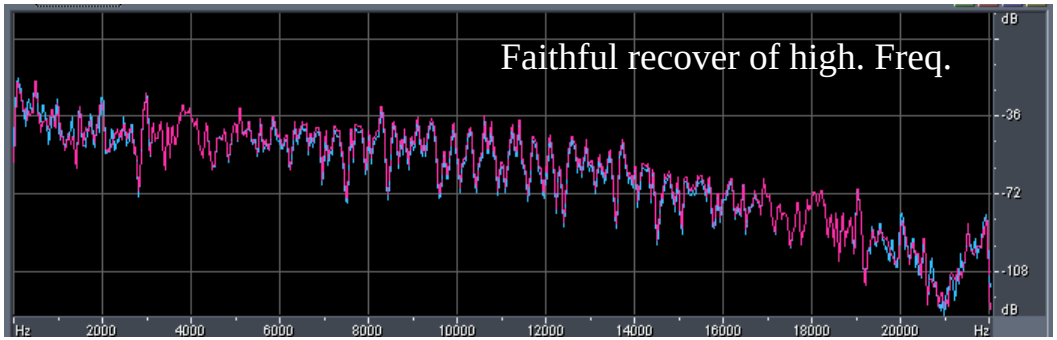

DSP
(TI
C6713
)
Analog Output
Digital readout
(44.1kHz) via
optical Fiber
cable

Example in MD(mini disk) players

MDLP4(66kbps)



After "YY"



More natural high freq. response

By the courtesy of SANYO Corporation

This "YY filter" is implemented in custom LSI sound chips by SANYO Coop., and being used in MP 3 players, mobile phones, voice recorders. The cumulative sale has reached over 40 million units.

Effect evaluation on compressed audio via PEAQ program

- Tested on 100 compresed music sources via PEAQ (Perceptual Evaluation of Audio Quality)
- PEAQ values:

  0…indistinguishable from CD

  -1…distinguishable but does not bother the listener

  -2…not disturbing

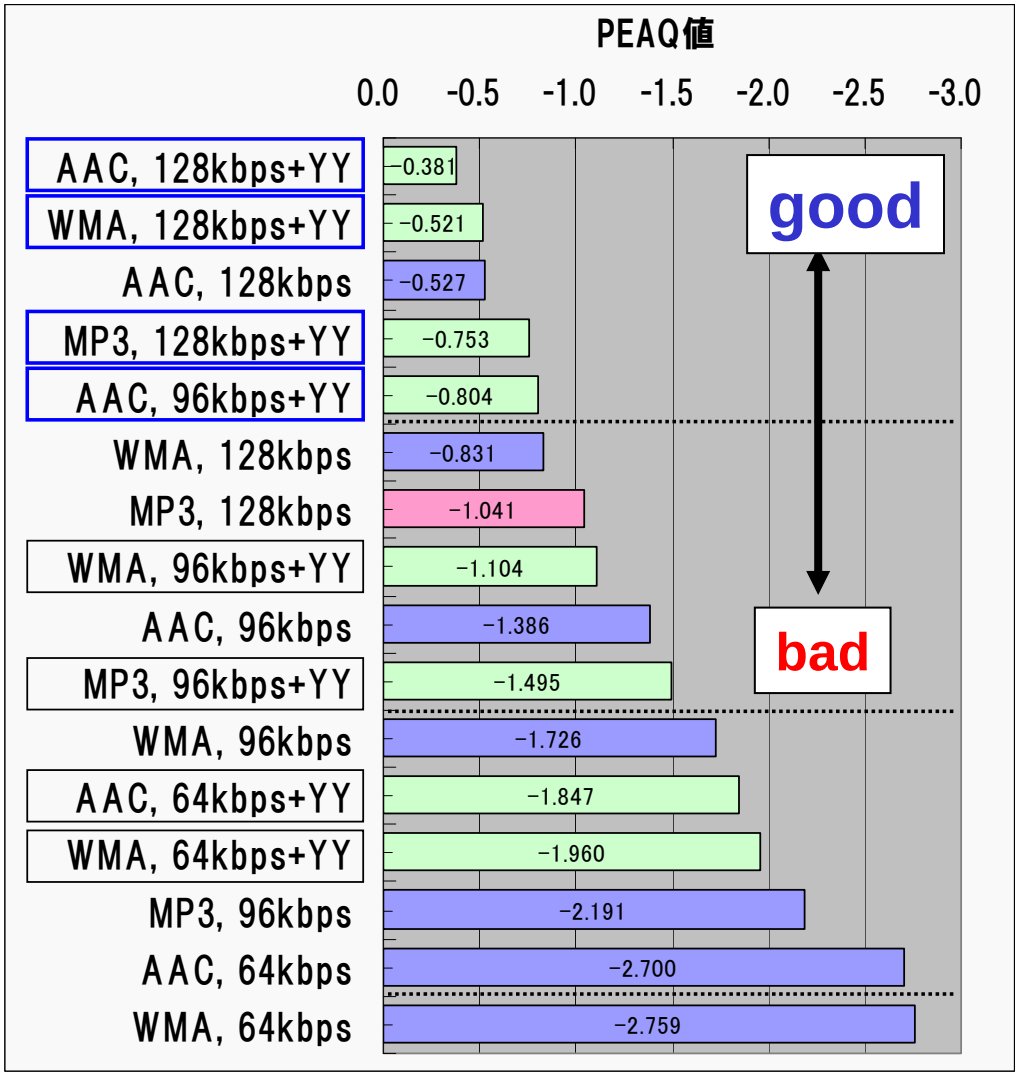
  -3…disturbing

  -4…very disturbing

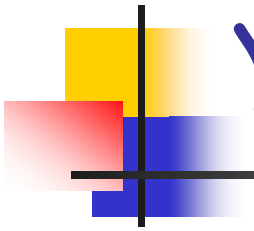
- Note how YY improves the sound quality

http://en.wikipedia.org/wiki/PEAQ

By the courtesy of SANYO corporation



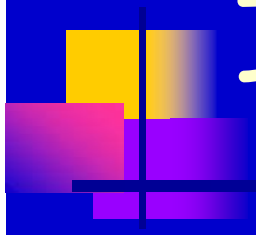
Compression formats: MP3, AAC, WMA
Bitrates: 64kbps, 96kbps, 128kbps
Showing average values

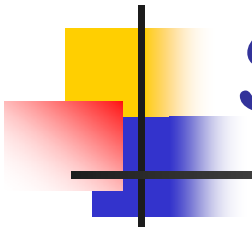


Yet one more application

- FANTABIT iPod/iPhone App

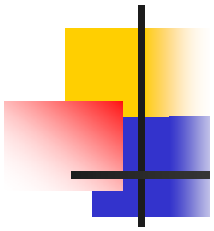
Part V: Application to Images





Same Problems as Sounds

- Block and Mosquito noise
- Lack of sufficient bandwidth
- Mosquito noise – Gibbs phenomenon
- Can sampled-data filter help?

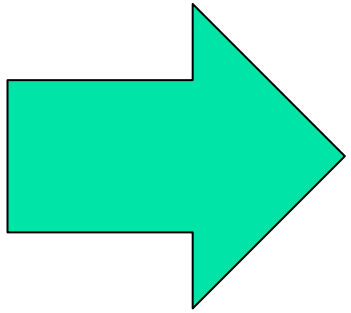


Another application:
How can we zoom “digitally”?

Interpolation via bicubic filter

Interpolation via sampled-data filter

Superresolution: Comparison with the bicubic filter



A moving image demo:

Superresolution via bicubic and Proposed filters


Thank you!