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MIXED-DOMAIN CIRCUITS AND SYSTEMS

Yannis Tsividis

Department of Electrical Engineering
Columbia University, New York

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Introduction

- Today, much innovation happens at boundaries, combining more than one fields (e.g., MEMs).
- In a similar manner, new possibilities can result by mixing domains within a field, in this case circuits and systems.
- This talk will summarize recent instances in the research of the speaker and his group, as examples in support of this claim.
- At the request of the organizers, some brief historical information is included.

1. Analog-digital MOS ICs – the early days

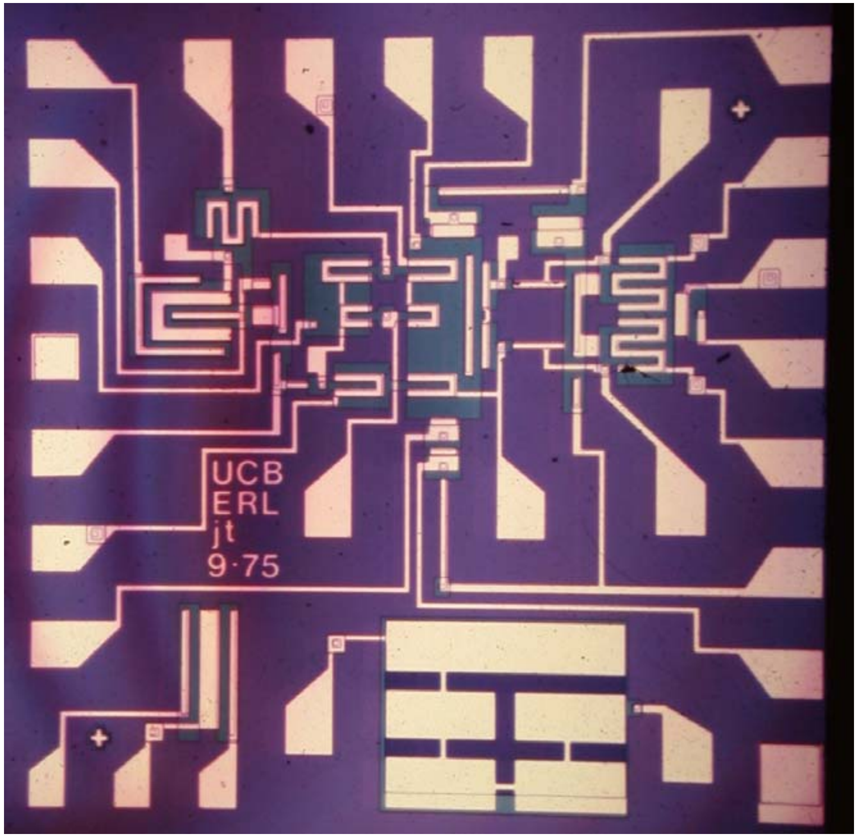
- **Early 1970s**:
    - The MOS transistor was well-established as a good digital switch; digital MOS ICs were becoming widespread.
    - The MOS transistor was not considered a viable analog IC component; analog circuits were made using bipolar transistors on *separate* chips.

- **Mid-1970s, UC Berkeley:**
  - MOS A/D converters were demonstrated (McCreary and Gray, 1975; Suarez et al., 1975).
  - The development of the “workhorse” of analog ICs – the op amp – in MOS technology was next.

    The task fell on this speaker.
  - Initial comment from an industrial colleague:
    “*So, you will make an op amp out of switches?*”

First fully-integrated MOS op amp

(Tsividis and Gray, 1976)



- Hand-made
- 12 µm, 4-mask process (!)
- Supply: +15, -15, and -20 V!
- 26 transistors
- DC gain: 350
- Unity-gain bandwidth: 5 MHz
- Power dissipation: 150 mW!

- Clearly, not a good op amp – but good enough to demonstrate that fully integrated analog MOS ICs are possible.

- Op amp was demostrated as part of a per-channel PCM telephony codec, a truly mixed analog-digital system (Tsividis et al., 1976).

- Industrial version (Siliconix) of codec was developed (Smarandoiu and Hodges, 1978).

- **Late 1970s:** Very high volume production of fully integrated MOS PCM codecs begins (one needed per telephone subscriber world-wide!)

- In retrospect, this was not difficult:
  - We were practically the only ones doing it, and had time to think.
  - Being in the right place, at the right time, helped!

- The speaker's group has followed this domain-combining model ever since.

- The rest of this talk presents some example of our recent attempts to combine domains in circuits and systems.

- "Domain" in this talk can mean: *analog; digital; linear; non-linear; time-invariant; time-varying; continuous-time; discrete-time; etc.*

2. Continuous-time digital filters

Tsividis, El. Letters 2003
and ICASSP 2004

Signal processor possibilities (1-D)

Time	*Amplitude*	
Discrete	Discrete	DSP
Continuous	Continuous	Classical analog
Discrete	Continuous	Sampled-data analog
Continuous	*Discrete*	*This work*

In the systems we are about to discuss,

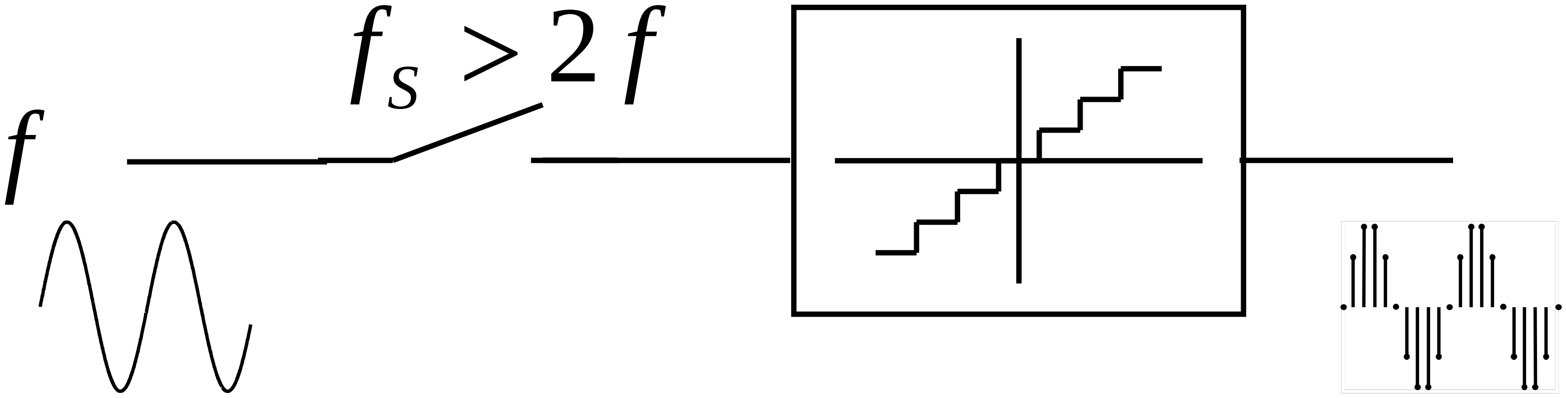
continuous-time signals…

…are processed with digital hardware
(which handles only 0s and 1s),

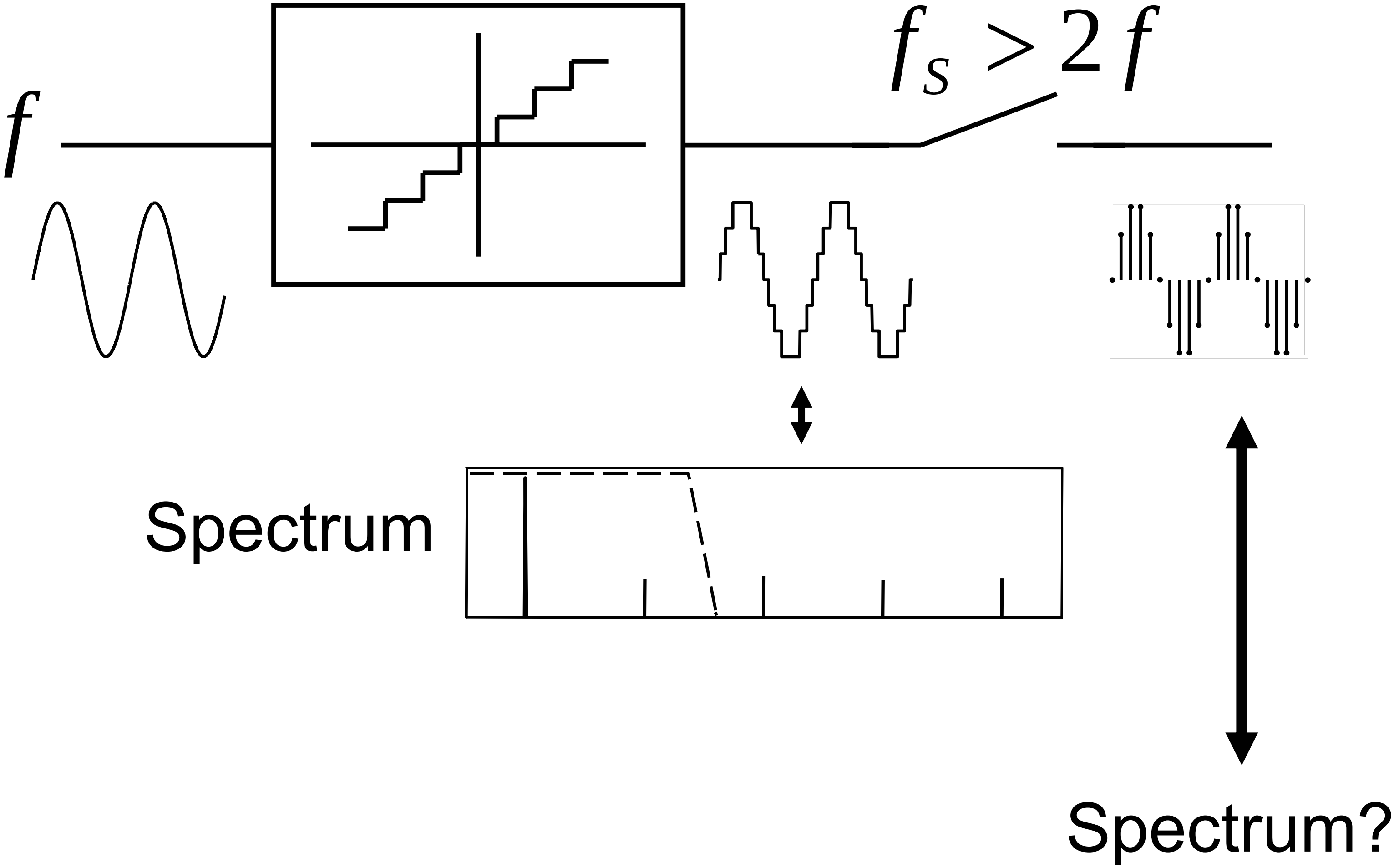
…operating in continuous-time.

These systems will be called
"continuous-time digital signal processors".

Conventional A/D-D/A equivalent:



To make easier to interpret the output, note that it does not change if the order is reversed:



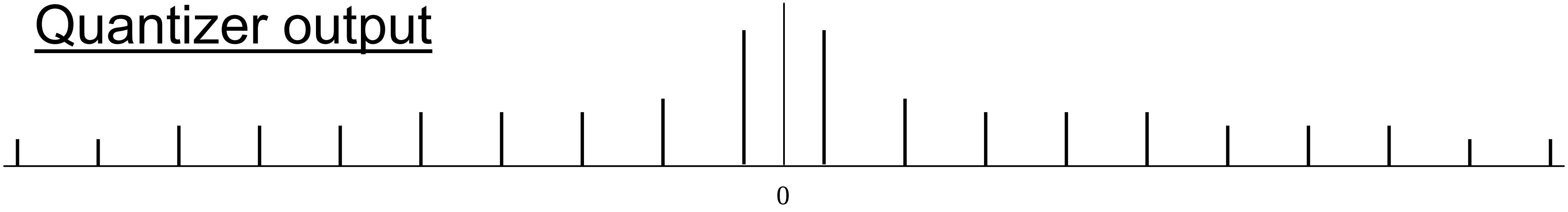
f

$f_s > 2f$

Spectrum

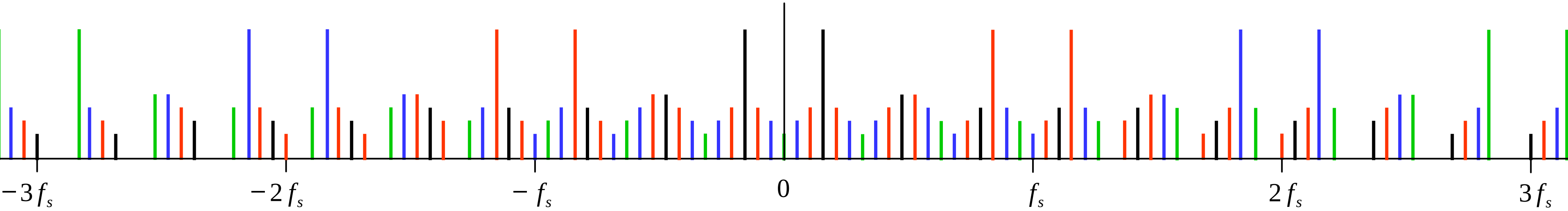
Spectrum?

Quantizer output



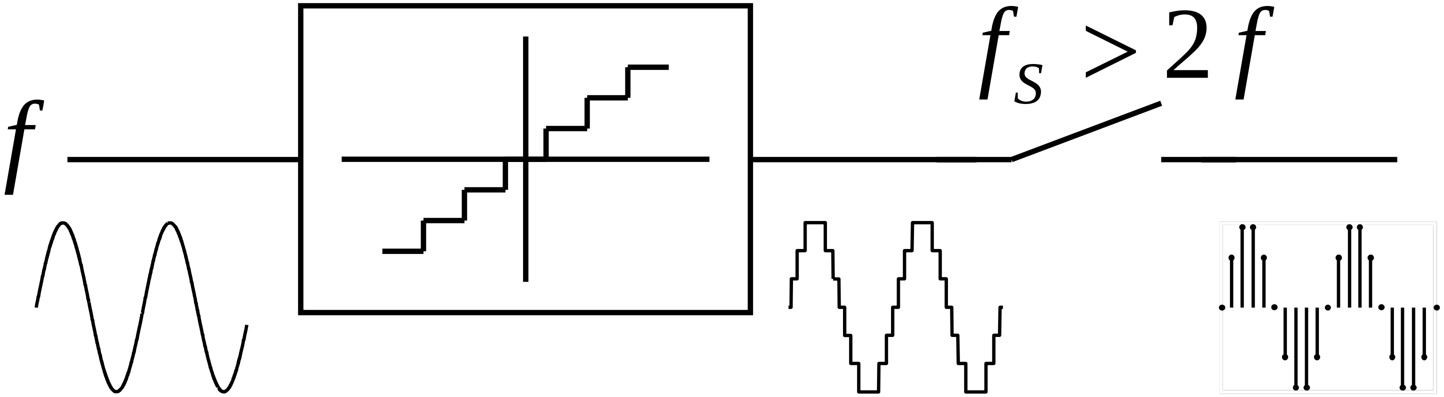
Quantizer in-band spectrum

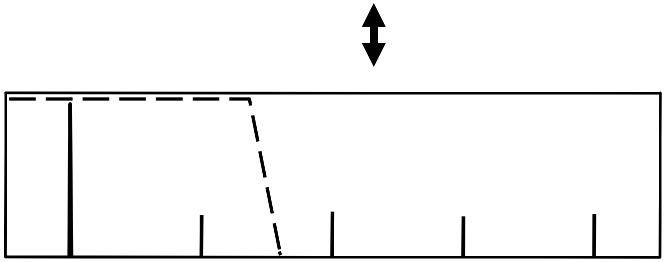
Quantizing + Sampling



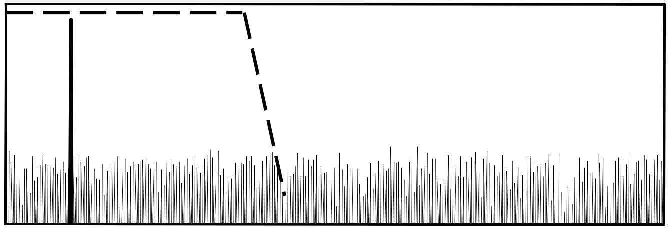
In-band spectrum contains aliased out-of-band harmonics

Thus:





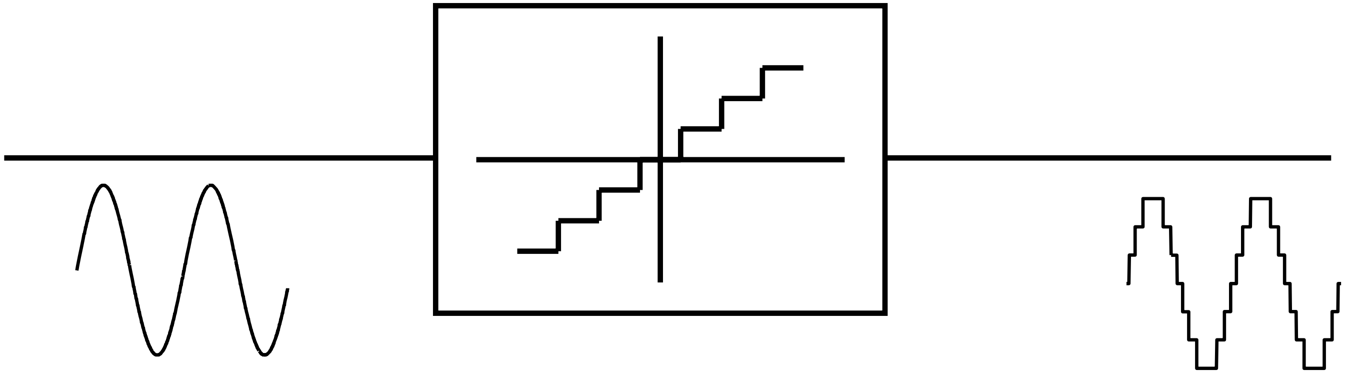
Small in-band error;
depends on ratio of
band-edge frequency to input frequency



Large in-band error

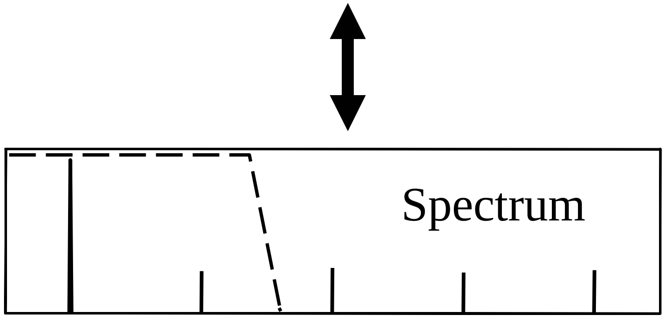
What if we eliminate sampling?

Using quantization only:

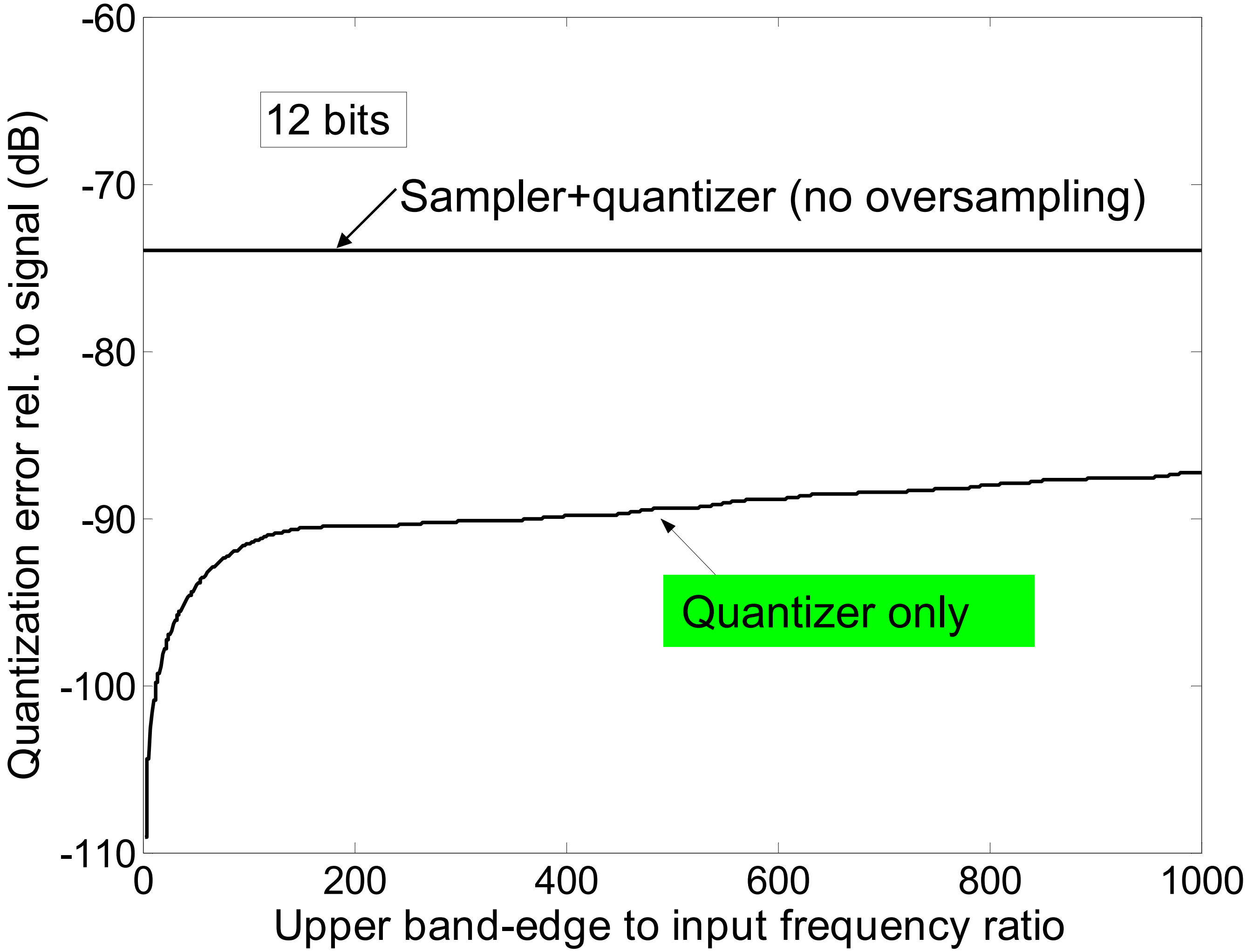


- No signal aliasing.

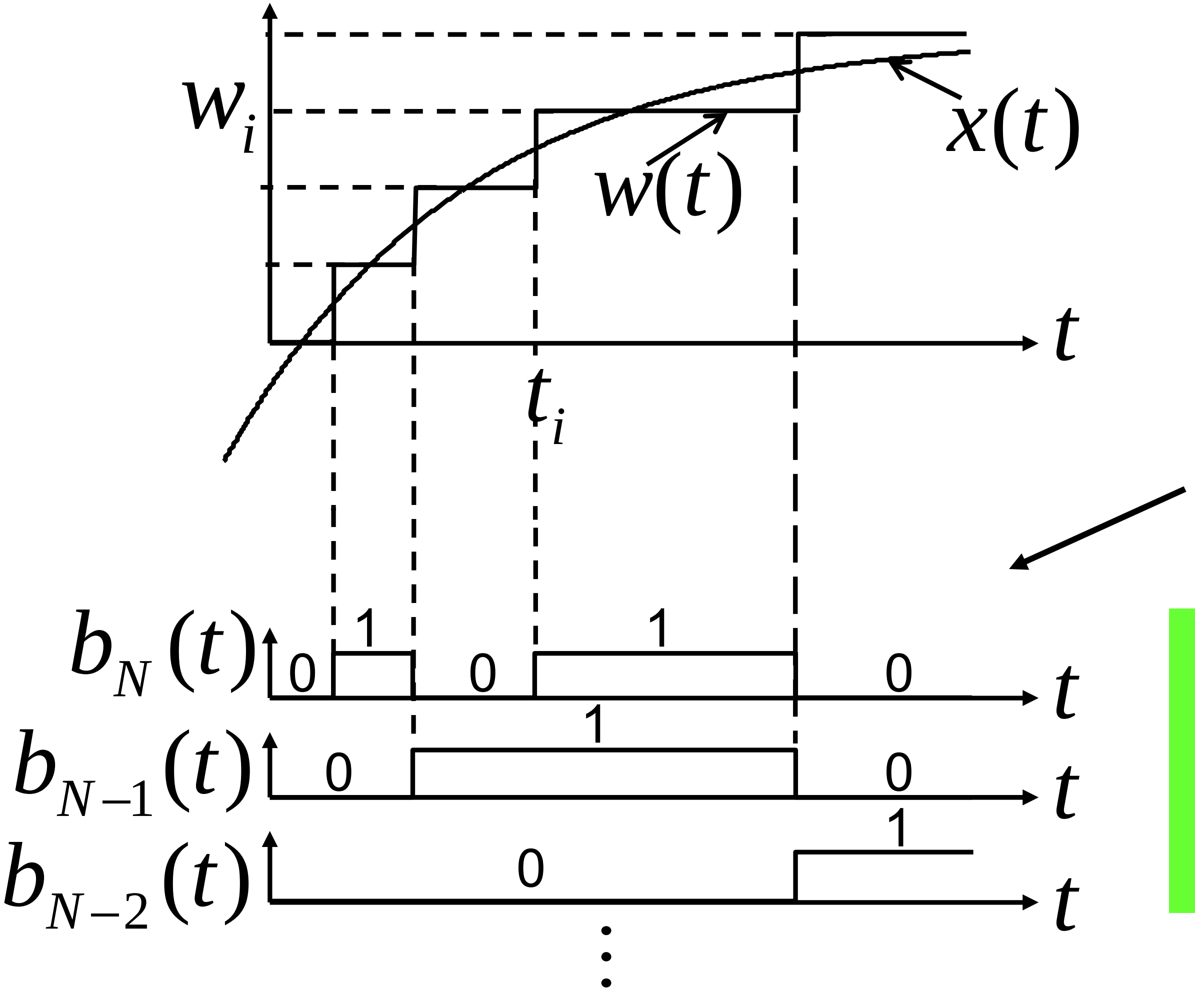
- No quantization error other than harmonic distortion.



In-band quantization error (12 bits):



Convert input to continuous-time digital form:



$$t_i \in \mathbf{R}$$

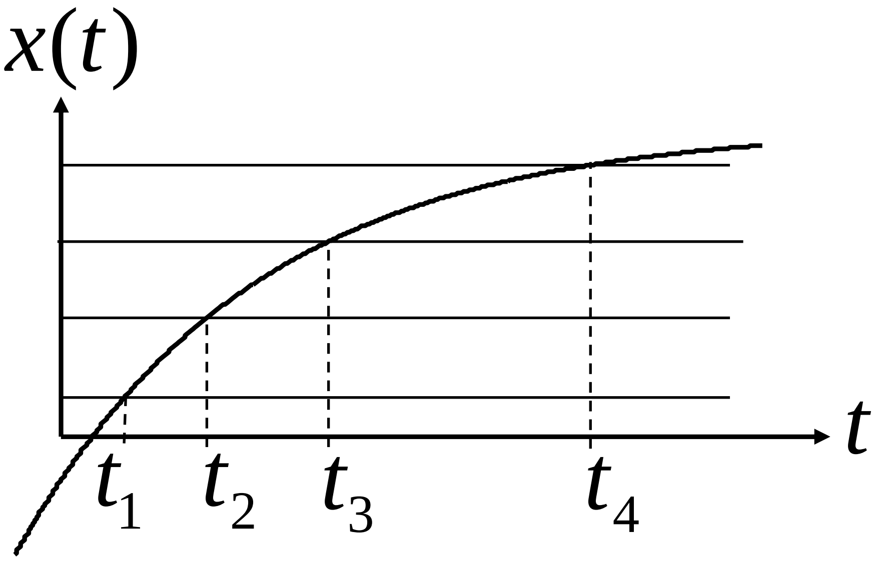
Continuous-time binary waveforms

$$w(t) = \sum_{n=1}^{N} 2^{-n} b_n(t) \qquad (1)$$

Switching activity decreases with decreasing input activity.

Relation to coding:

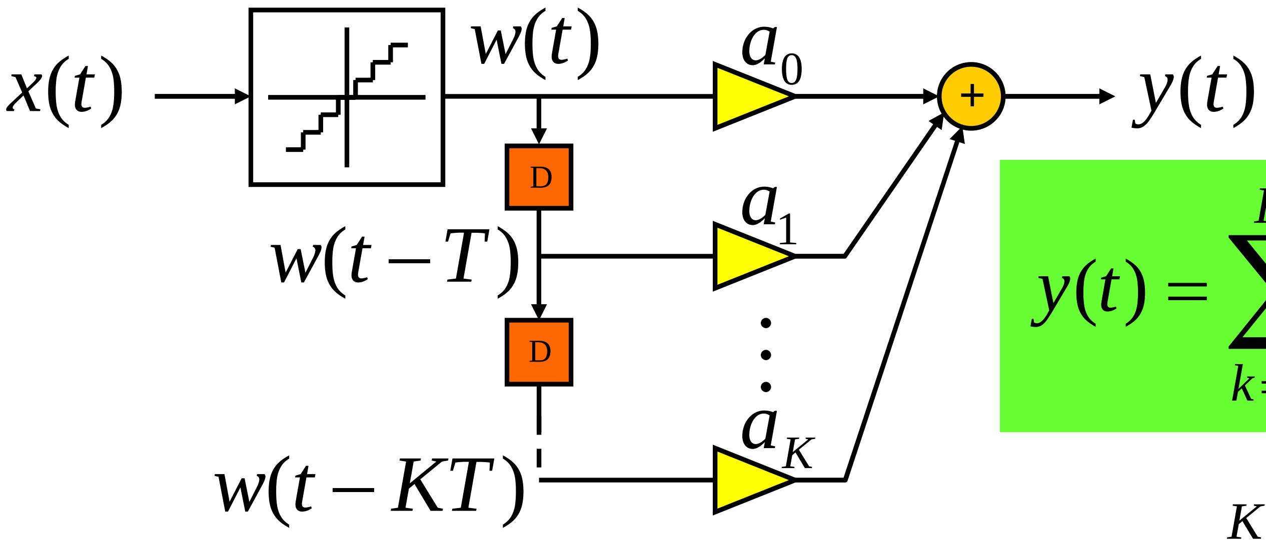
The swithing times t_i correspond to *time code modulation* [Inose et al., 1966; Foster and Wang, 1991].



Continuous-time binary waveforms have been used for a long time; e.g. in PWM (Schwartz, 1959) or in auditory feature extraction (Kumar et al., 1998).

Filter development

Start with classical c.t. delay-line filter:

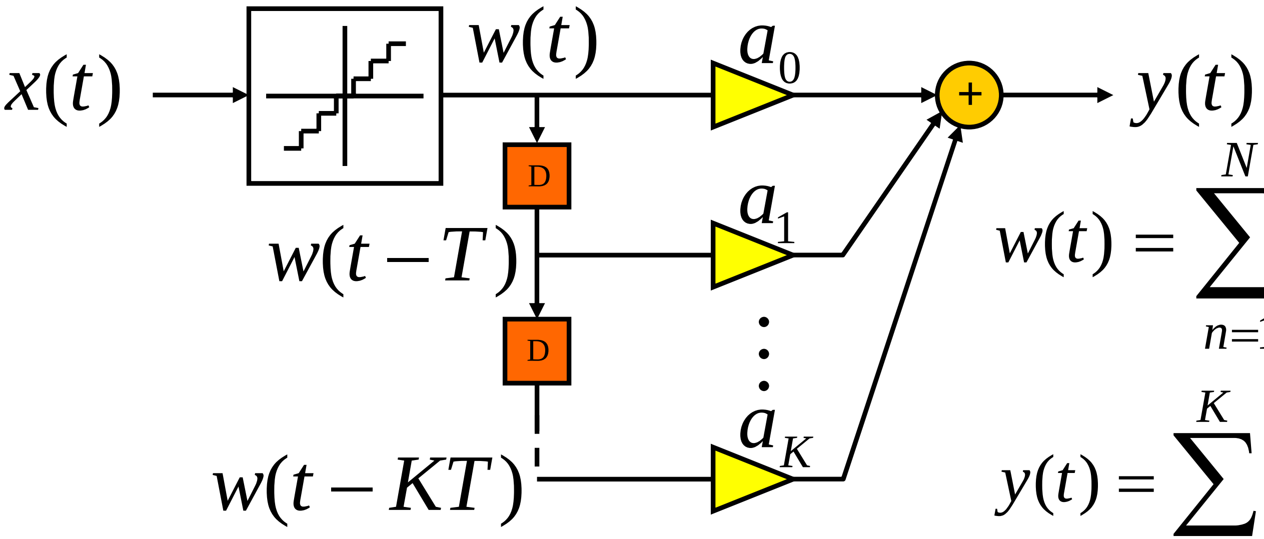


$$y(t)=\sum_{k=0}^{K} a_k w(t-kT) \qquad (2)$$

$$Y(s)=\sum_{k=0}^{K} a_k e^{-skT} W(s)$$

Thus:

$$Y(s)=H(e^{sT})W(s)$$ with: $$H(e^{sT})=\sum_{k=0}^{K} a_k (e^{sT})^{-k}$$



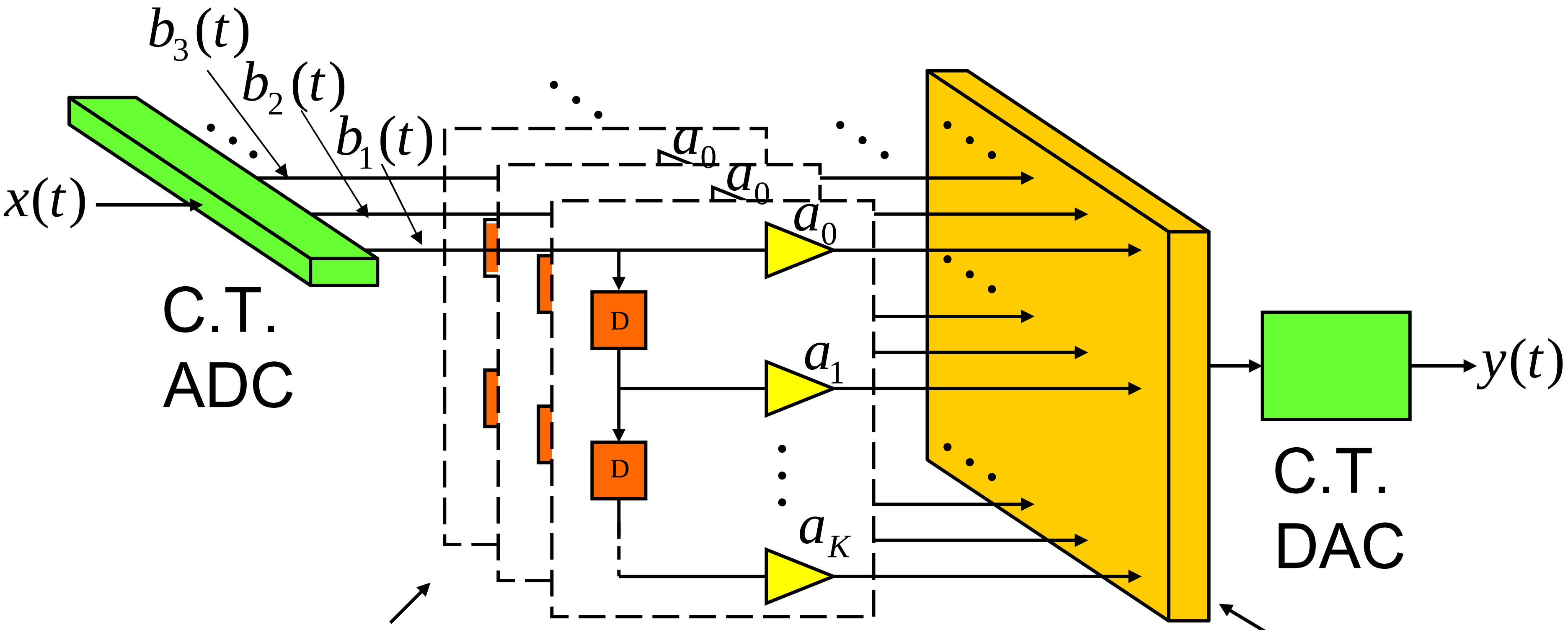
$$w(t)=\sum_{n=1}^{N}2^{-n}b_n(t) \quad (1)$$

$$y(t)=\sum_{k=0}^{K}a_k w(t-kT) \quad (2)$$

$$(1),(2)\Rightarrow y(t)=\sum_{k=0}^{K}\sum_{n=1}^{N}a_k 2^{-n}b_n(t-kT)$$

or $$y(t)=\sum_{n=1}^{N}2^{-n}\underbrace{\left[\sum_{k=0}^{K}a_k b_n(t-kT)\right]}_{\textit{Same processing as in (2)}}$$

This leads to:



C.T.
ADC

C.T.
DAC

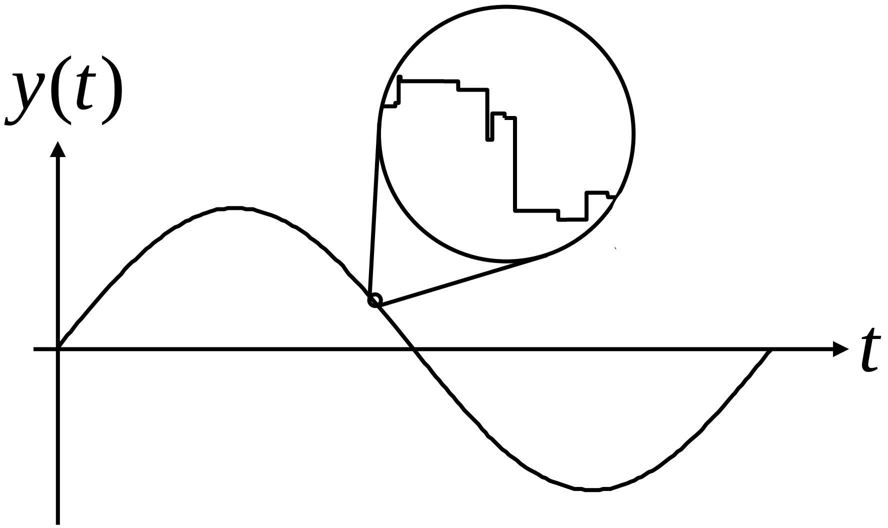
C.T. single-bit
digital delays

C.T. digital
multipliers

C.T. weighted
digital adder

CONTINUOUS-TIME DSP

Typical output waveform:



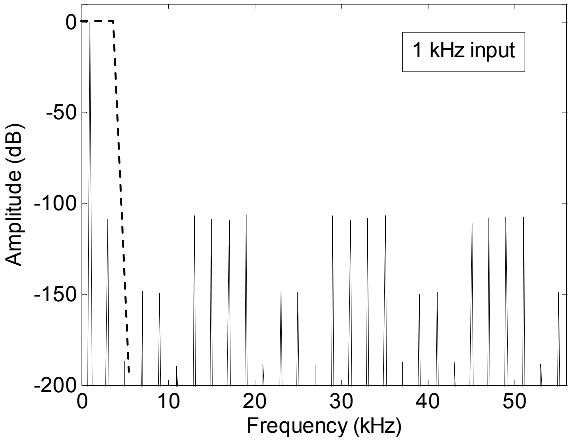
- Output time transitions can differ by arbitrarily small amount.
- Glitches must be carefully considered.
- Requires high speed non-clocked comb. logic, ADC, and DAC.
- Graceful degradation at high signal frequencies.
- For hardware considerations, see [5,6].

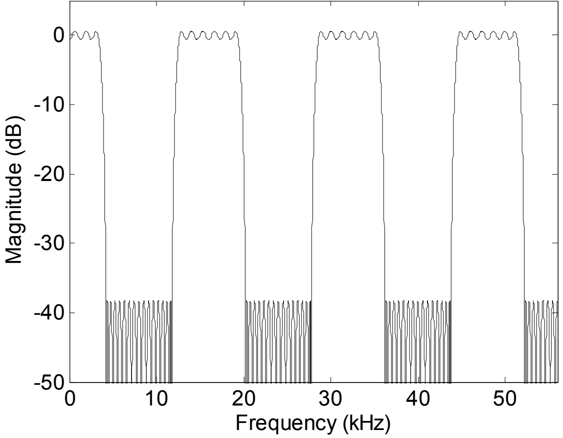
Simulations

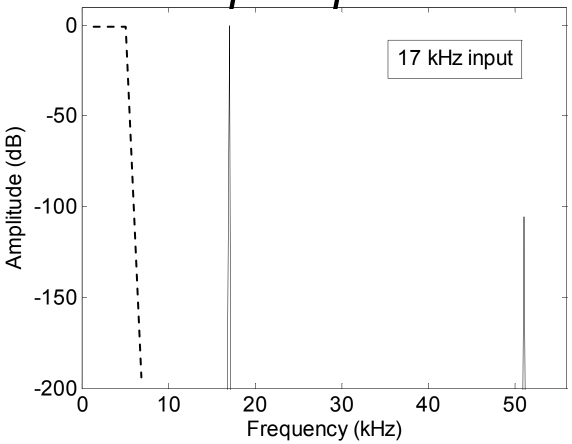
12-bit quantization

3.3 kHz low-pass
28th order FIR
125 μs unit element delay

(Matlab/Simulink)


Output spectrum


Frequency response


Output spectrum

No aliasing into baseband

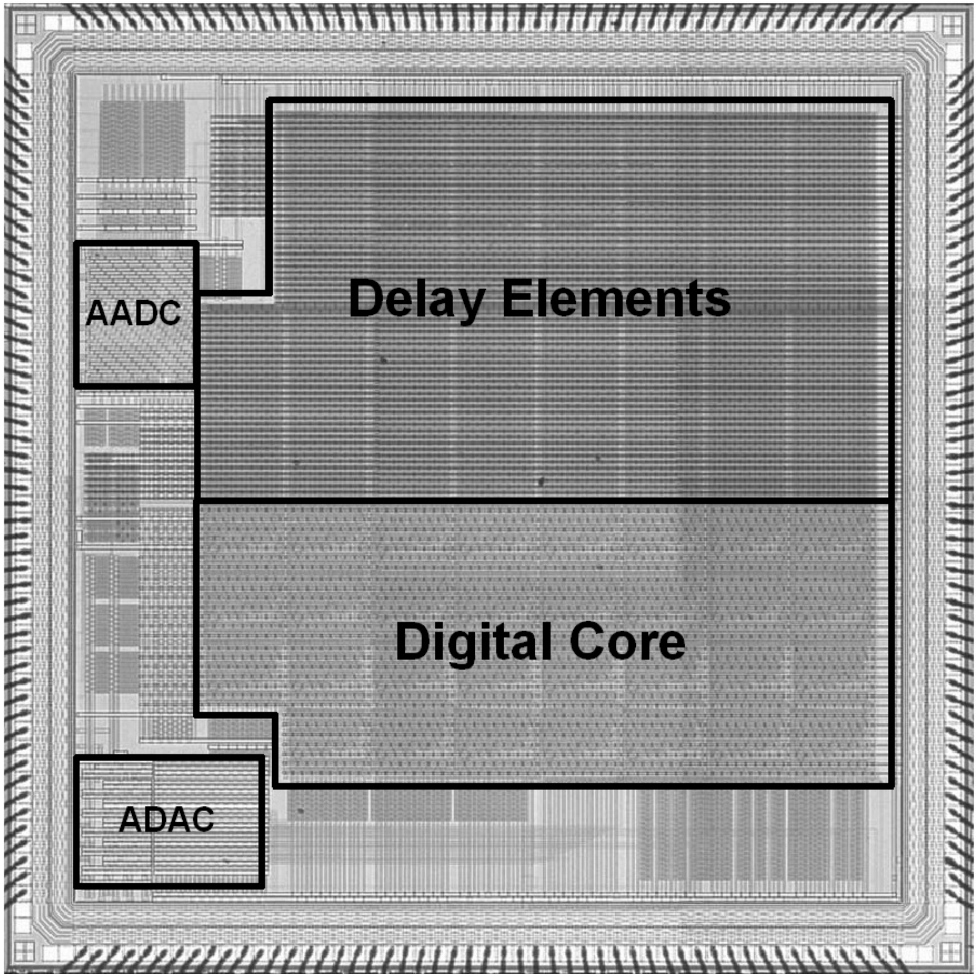
Preliminary test chip [Li et al., CICC'05]

- Delta mod
- 6-bit, 16-tap, 6-kHz LP
- Verified:

  -No signal aliasing

  -14 dB advantage compared to classical

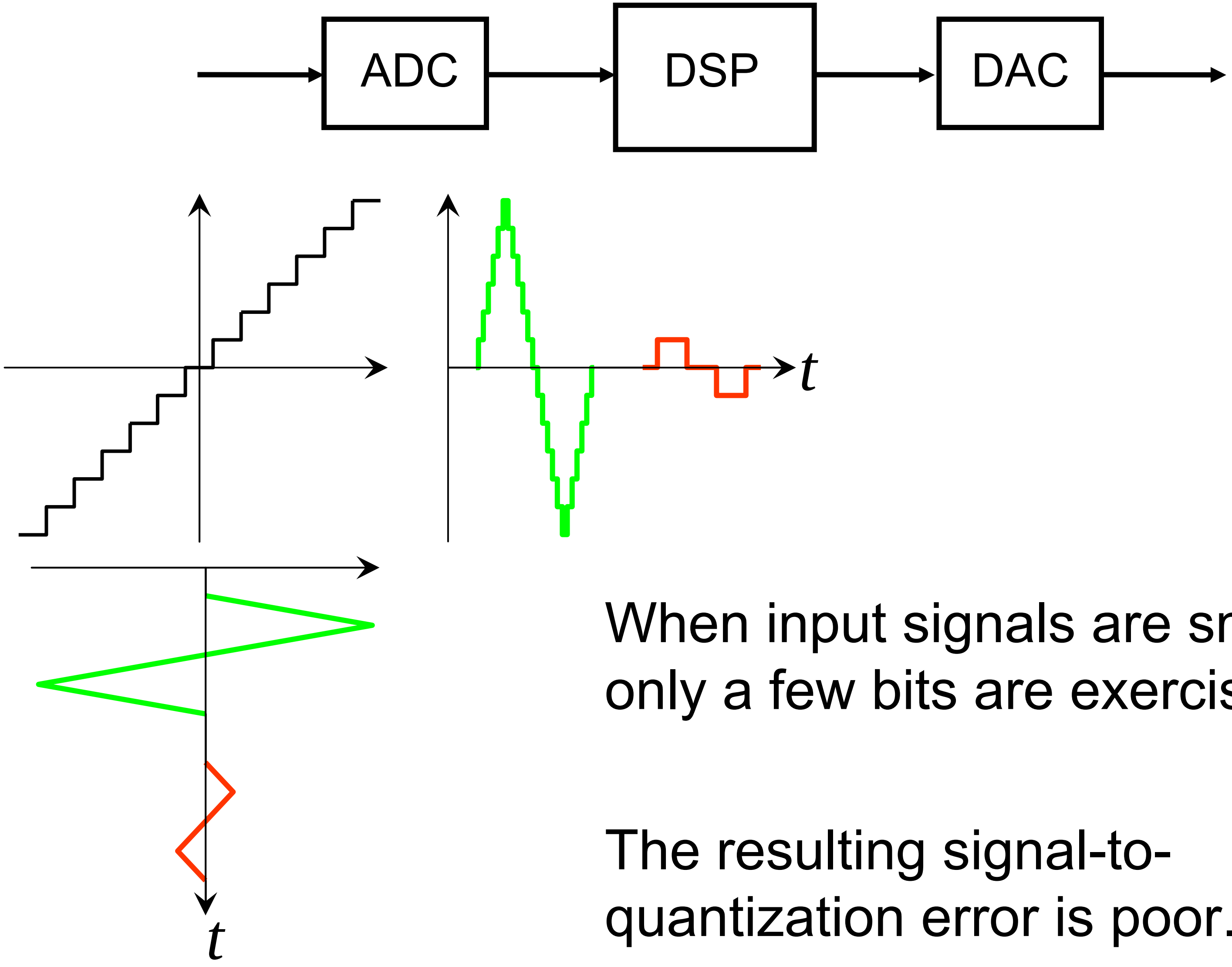
  -Power decreases with decreasing input frequency.



3. Externally linear (but internally nonlinear) systems

Tsividis, Trans. CAS II, 1997;
Klein and Tsividis, ICASSP 2006

Consider a fixed-point signal-processing system:

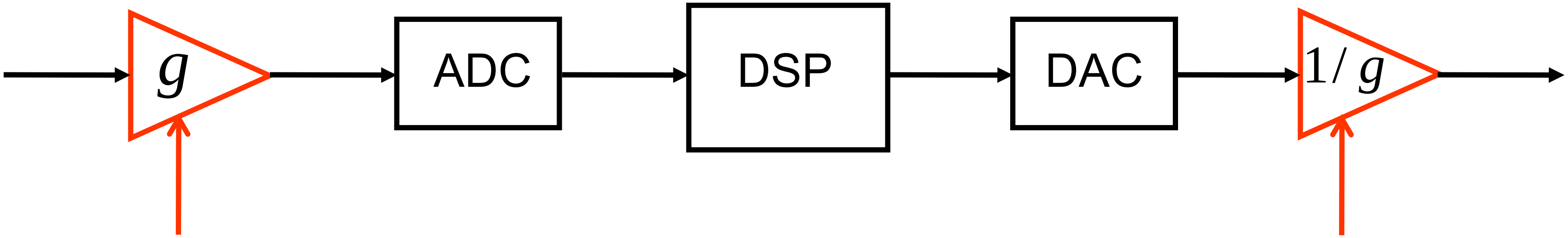


When input signals are small, only a few bits are exercised;

The resulting signal-to-quantization error is poor.

To correct this, we need to keep the internal signals always large, without affecting the final output.

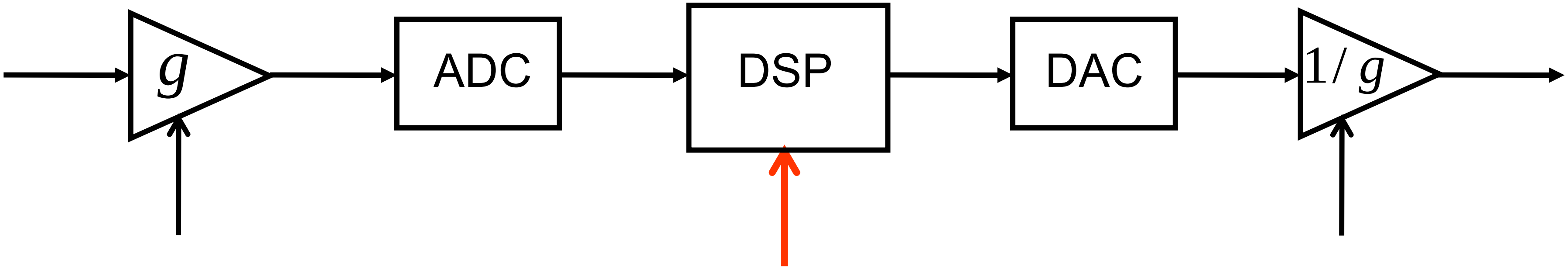
Try classical “companding” (compressing and expanding):



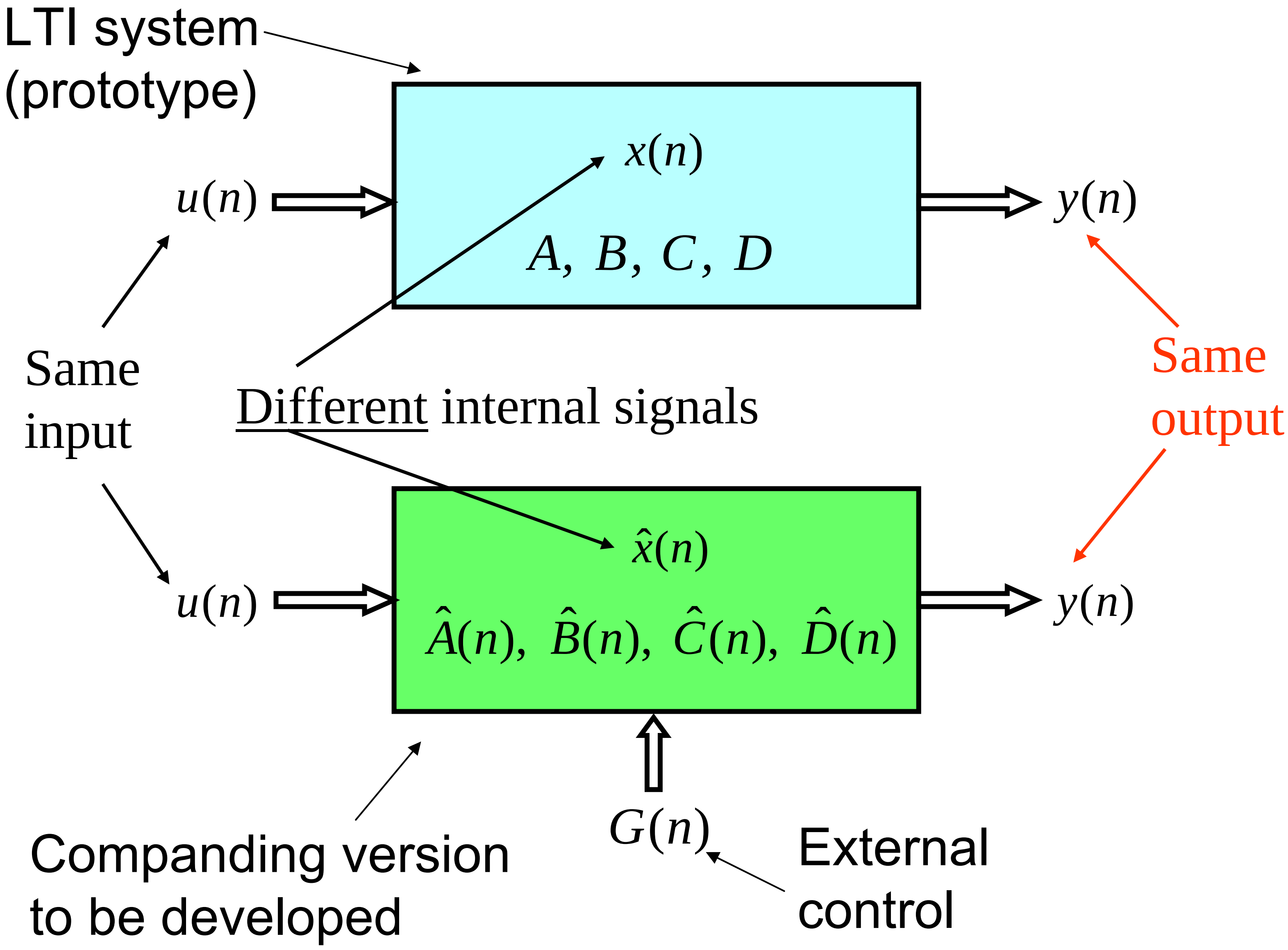
This does not work! The dynamical nature of the processor interferes.

In analog companded filters, this problem is solved by also controlling the state variables [Blumenkrantz, 1995].

Thus, let us try [Klein and Tsividis, 2006]:



To accomplish the task correctly, use a state-space approach (Tsividis, 1997):



LTI system (prototype)

$u(n)$

$x(n)$

$A,\ B,\ C,\ D$

$y(n)$

Same input

Different internal signals

Same output

$\hat{x}(n)$

$u(n)$

$\hat{A}(n),\ \hat{B}(n),\ \hat{C}(n),\ \hat{D}(n)$

$y(n)$

$G(n)$

Companding version to be developed

External control

How can this be accomplished?

Start with the state equations
of LTI prototype:

$$x(n+1) = Ax(n) + Bu(n)$$
$$y(n) = Cx(n) + Du(n)$$

Develop a companding version:

$$\hat{x}(n+1) = \hat{A}(n)\hat{x}(n) + \hat{B}(n)u(n)$$
$$\hat{y}(n) = \hat{C}(n)\hat{x}(n) + \hat{D}(n)u(n)$$

with state variables modified
by a control signal $G(n)$:

$$\hat{x}(n) = G(n)x(n)$$

Require this system to have,
for the same input (and
corresponding initial conditions),
the same output as the prototype:

$$\hat{y}(n) = y(n), \quad \text{all } n$$

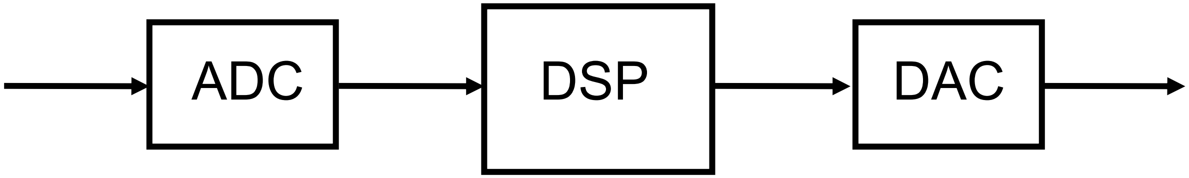
Direct substitution gives:

$$
\begin{aligned}
\hat{A}(n) &= G(n+1)AG^{-1}(n) \\
\hat{B}(n) &= G(n+1)B \\
\hat{C}(n) &= CG^{-1}(n) \\
\hat{D}(n) &= D
\end{aligned}
$$

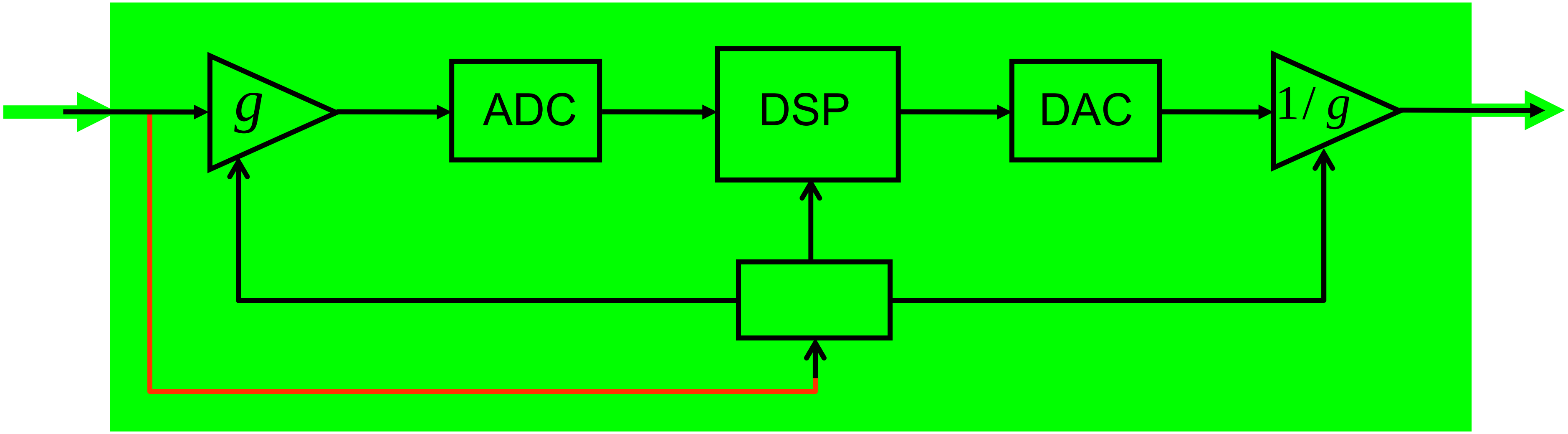
Thus, this linear transformation allows us to optimize the internal waveforms <u>without any disturbances at the output.</u>

The control $\boldsymbol{G(n)}$ can be developed from the input envelope, to make sure that all internal signals are large.

Thus, corresponding to the original system:



We have an input-output equivalent companding version:



The new system is:

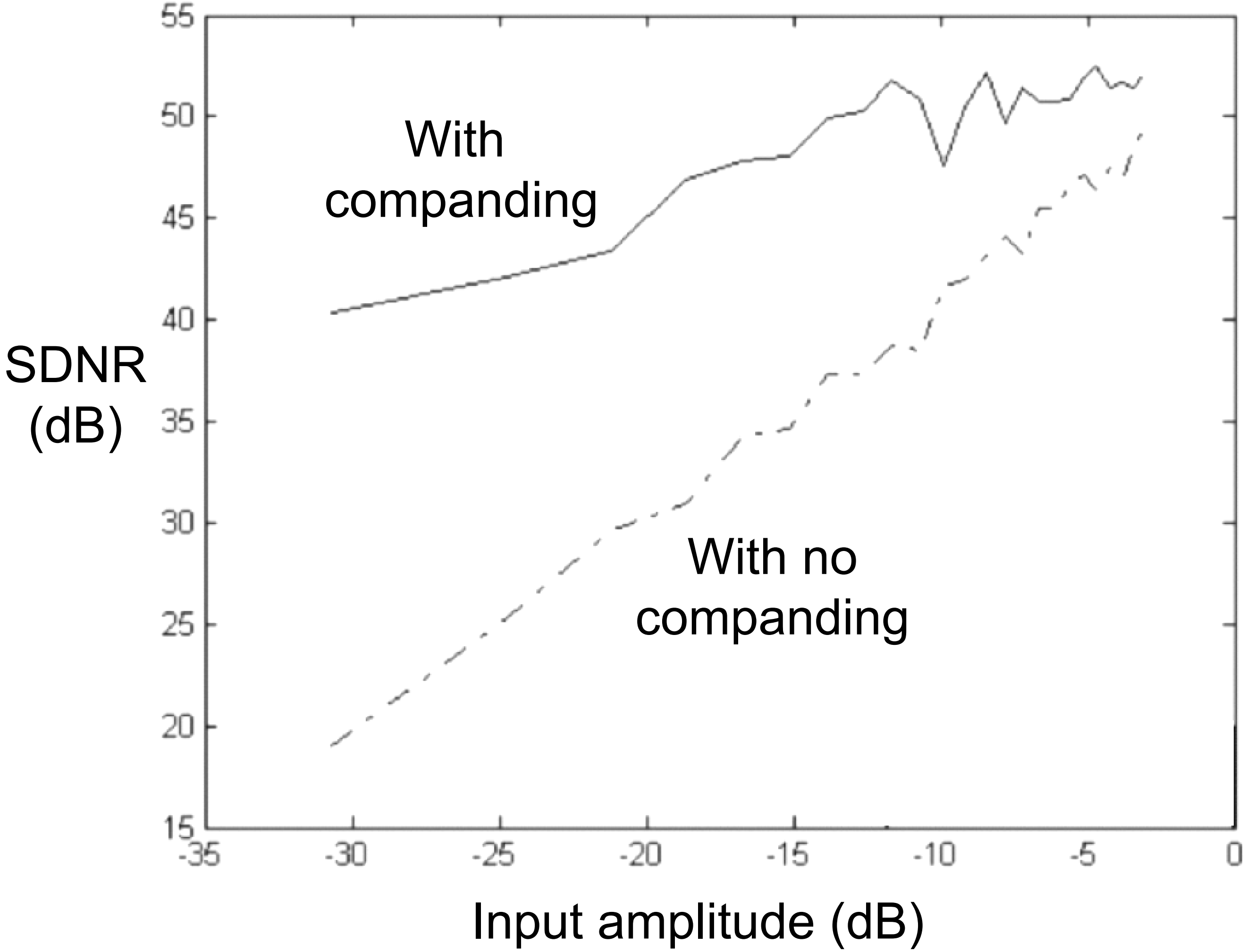
- Externally linear
- Internally nonlinear.

- The companding system has a larger usable dynamic range for a given number of bits.

- Or, it can attain a given usable dynamic range with a smaller number of bits, thus with smaller power dissipation.

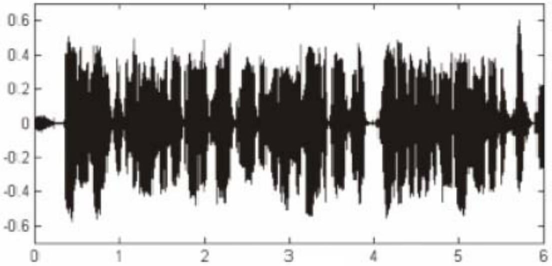
- For a continuous-time version of this approach, see Refs. 13, 16.

Example: 8-bit reverberator [14] Steady-state case

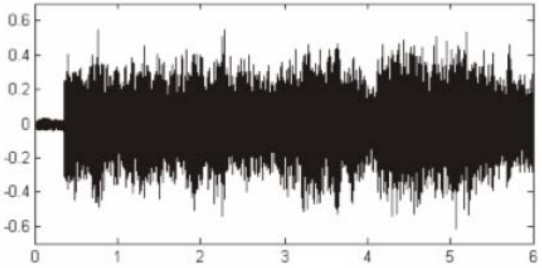


8-bit reverberator; transient case

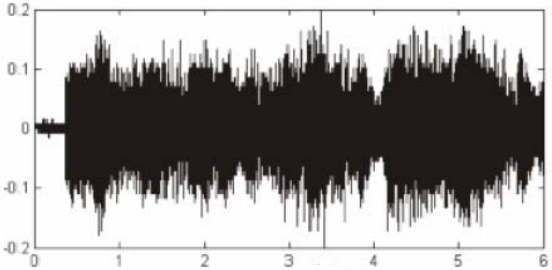
6 seconds of a sample input



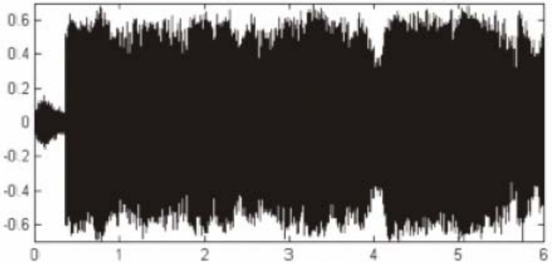
Output of the original system



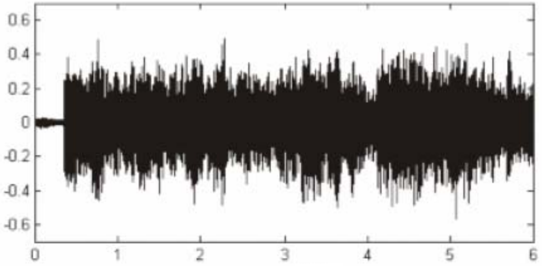
A state in the original system



Corresponding state in the companding system



Output of properly companding system

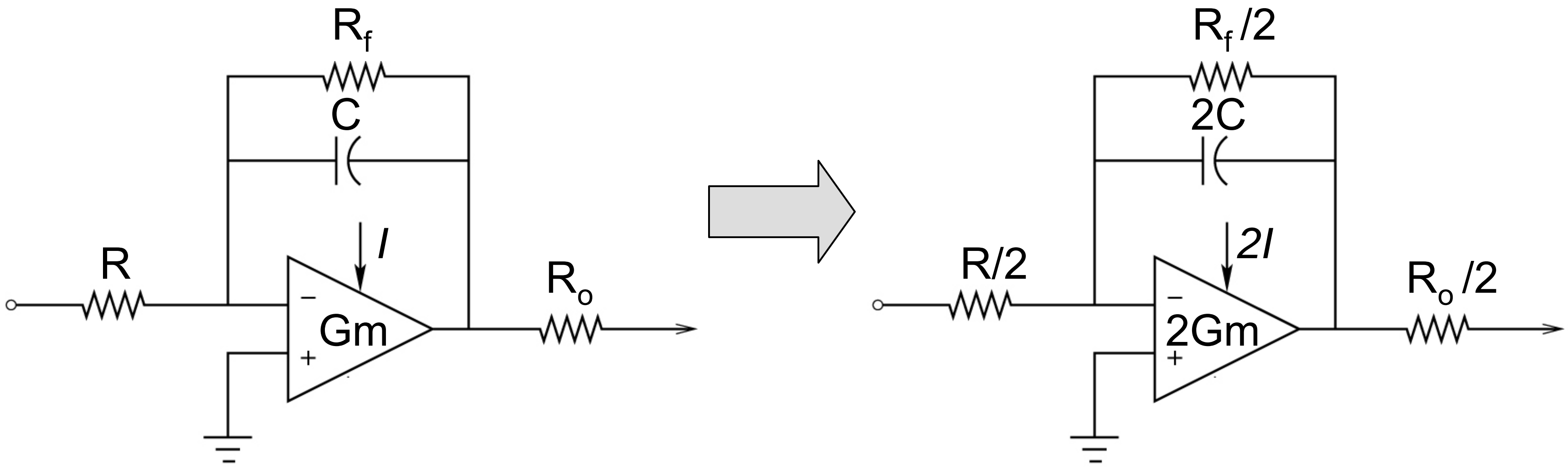


For sound clips: http://www.cisl.columbia.edu/~aek84/spconf2006.html [14]

4. Externally time-invariant (but internally time-varying) circuits

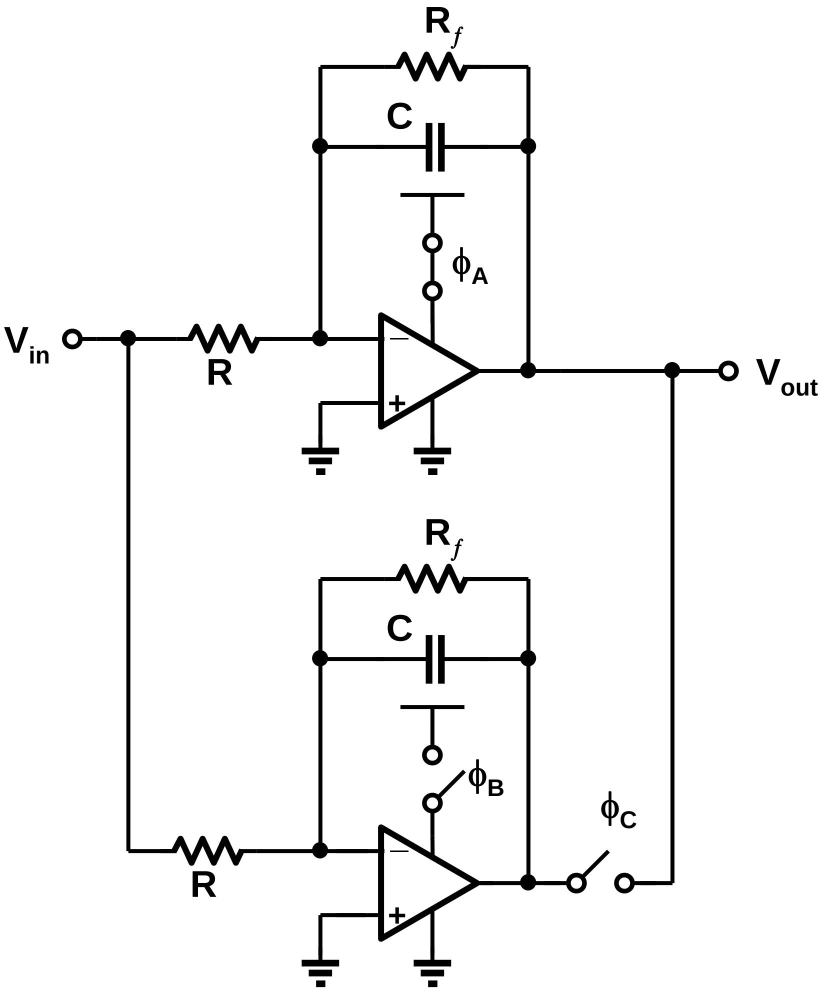
Ozgun et al., ISSCC 2005

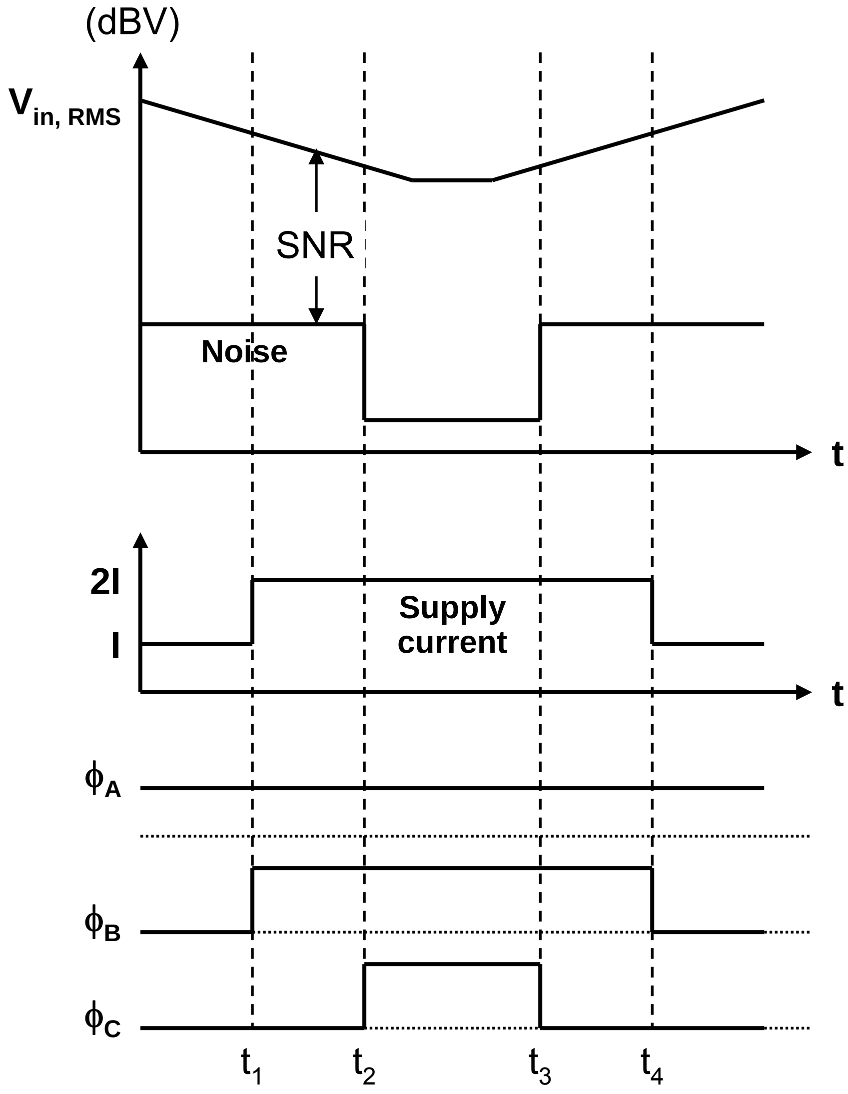
Trading off power for noise



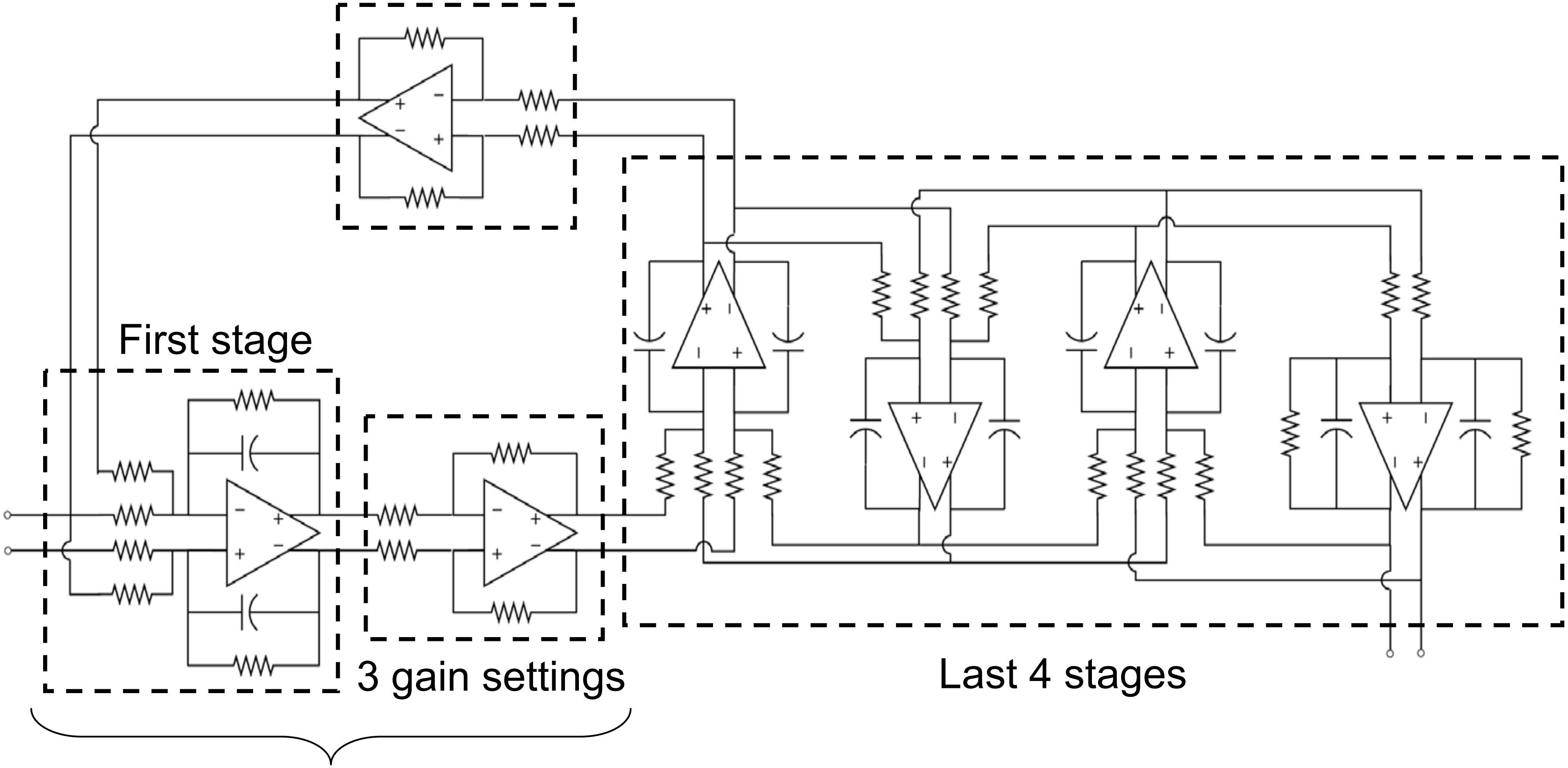
Noise Reduction: 3 dB
Supply Current: × 2

Dynamic impedance scaling with no output transients





Internally varying filter



3 gain settings

First stage

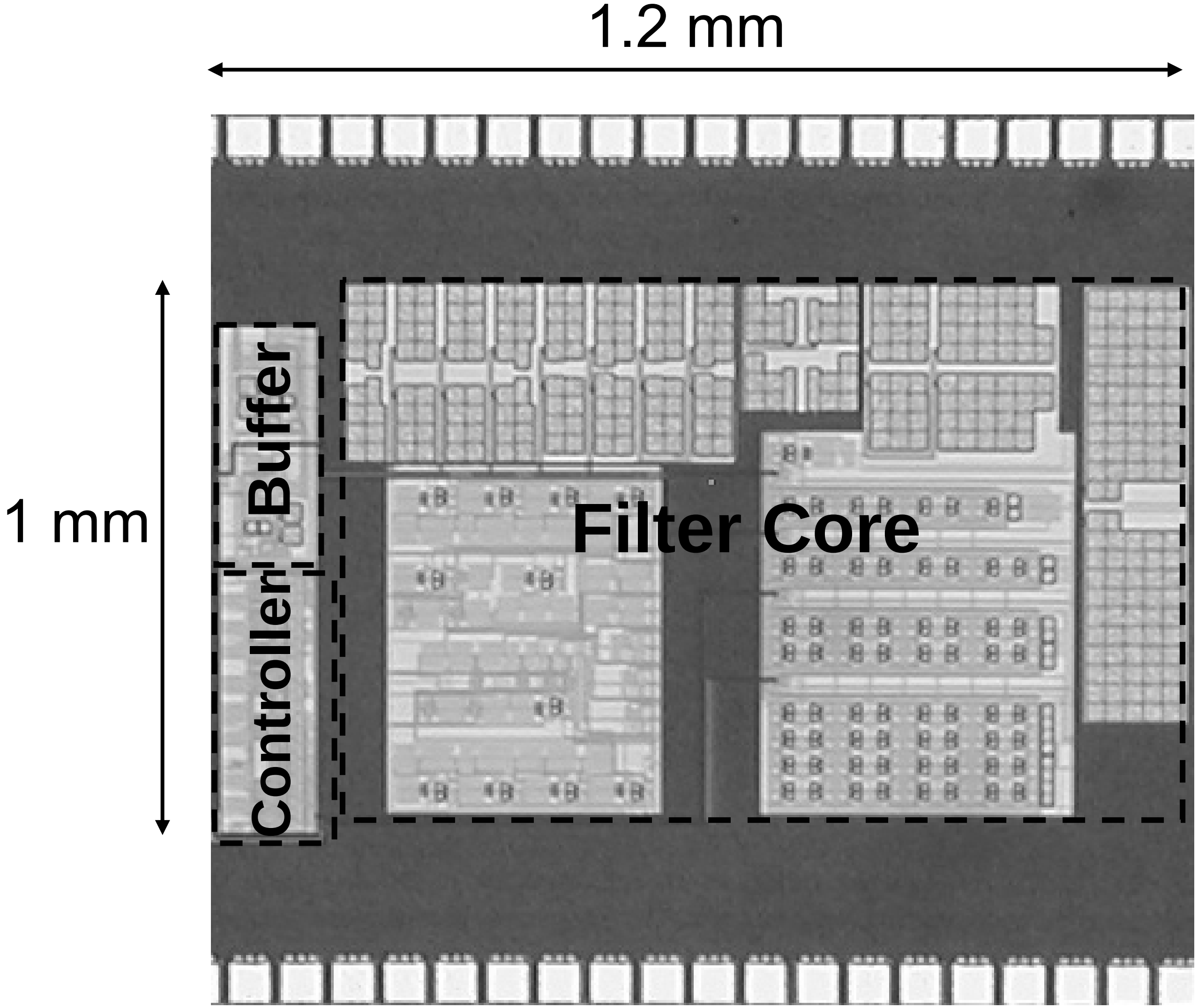
3 gain settings

Last 4 stages

Impedance-scaled; 5 settings

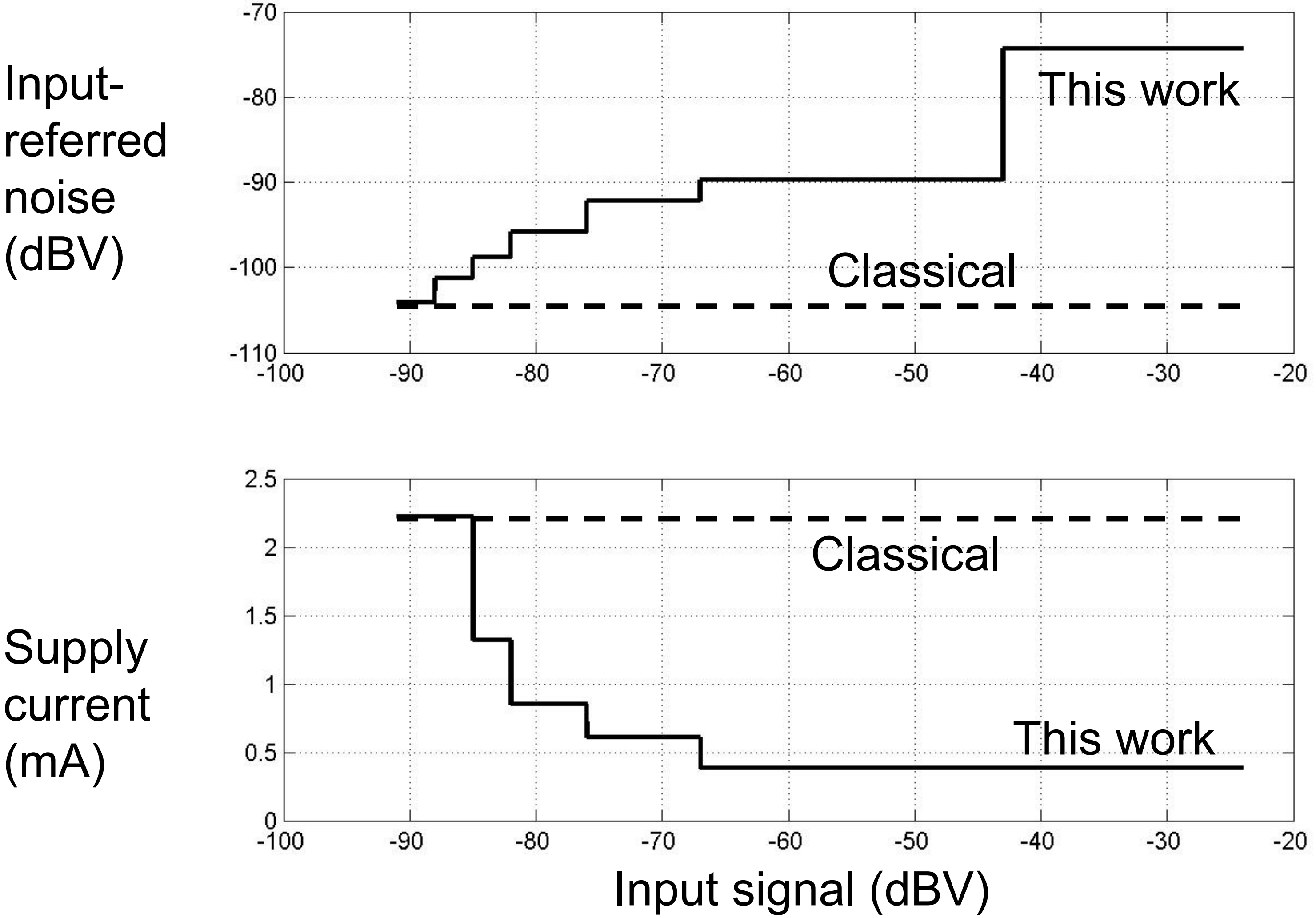
Controlled from output envelope;
Designed to withstand worst-case blocker.

Chip photograph



(Ozgun, 2005)

Measured performance



5. VLSI analog computers / analog math co-processors for digital computers

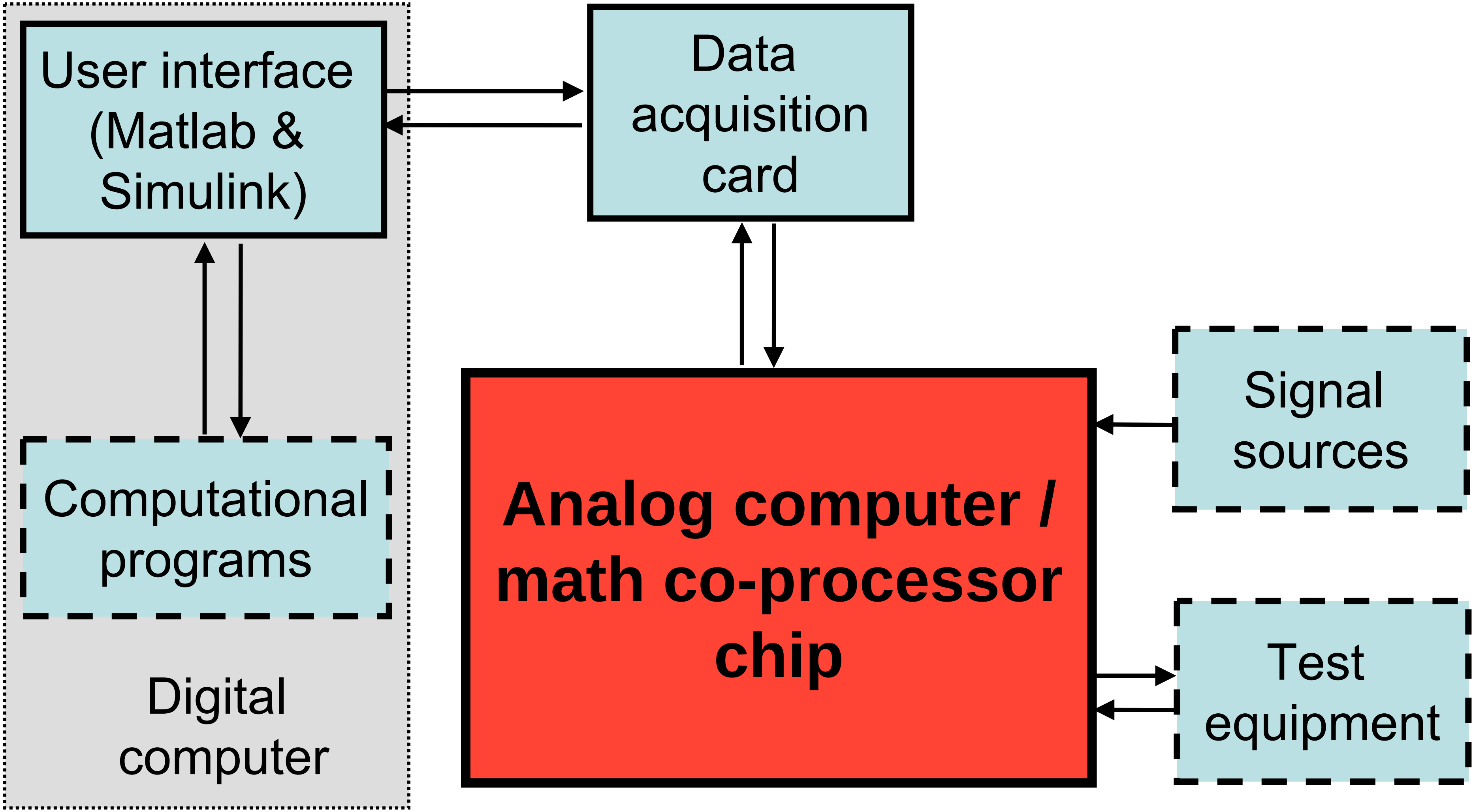
Cowan et al., JSCC 2006

- People abandoned classical analog computers back in the SSI era and never looked back…

- Could analog computers have something to offer if implemented in ULSI?

- The answer may be yes, *for special purposes*, if they operate *in synergy with digital computers*.

Hybrid computation environment


User interface (Matlab & Simulink)

Data acquisition card

Computational programs

Digital computer

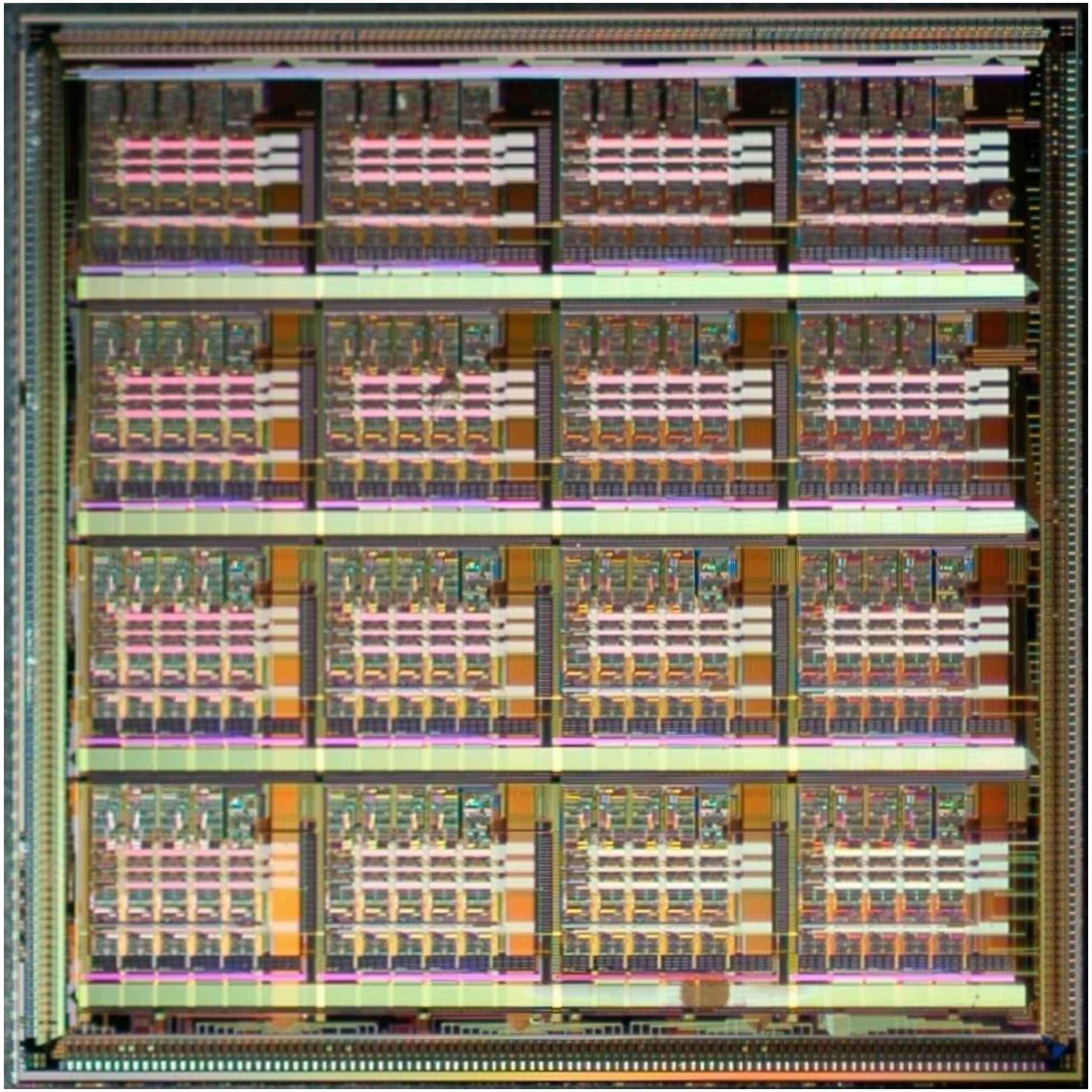
Analog computer / math co-processor chip

Signal sources

Test equipment

Analog computer chip

- CMOS, 0.25 μm
- 100 mm^2 , 300mW
- 10^6 transistors
- Current mode
- 80 integrators
- 80 VGA / multipliers
- 16 log blocks
- 16 exp blocks
- 64 programmable blocks (sign, sat, abs, ramp, min, max, >, <, chop, T/H, S/H).



Features

- Digital programmability through a standard on-screen Simulink interface.
- Pre-computation calibration process.
- Continuous-time operation; no discretization artifacts.

Features, cont’d

- Inherent parallelism.
- Real-time simulations; real-time observation of the effects of mathematical parameter changes.
- Easily includes continuous-time random inputs.

Target applications

- Nonlinear ordinary differential equations
  - e.g., system simulation, circuit simulation.
- Stochastic differential equations
  - e.g., molecular dynamics, transient noise analysis of circuits.
- Nonlinear and linear partial differential equations
  - e.g., chemical process control, nonlinear wave propagation, heat transfer.

Best-suited to *special-purpose* computation.
It can:

- Obtain fast solutions to tough numerical problems, in special-purpose applications that do not demand high accuracy.

- Provide fast approximate solution as a start to the digital computer, thus enabling quick convergence to accurate results.

Preliminary results

A limited number of benchmarks run so far.
Some encouraging results:

- Up to 400 times higher speed than a modern workstation running Matlab.
- Accelerated solution of some ODEs on a digital computer by more than 10 times.
- Up to 6.4 GFLOP/s, 20 μJ/MFLOP.

Conclusions

- This talk has argued that mixing domains within circuits and systems can result in new possibilities.

- We have presented several examples of mixed-domain systems.

- Continuous-time DSP may offer certain advantages of digital technology without its drawbacks:
  - Fully digital (noise immunity, programmability)
  - No sampling; thus no signal aliasing
  - Smaller in-band quantization error
  - Power goes down with decreasing input activity.

- Internally nonlinear systems:
  - Can be designed to be input-output linear
  - Can keep their internal signal strength large, even for small-strength inputs
  - Can maximize the signal-to-error ratio for a large range of input signals.

- Internally time-varying circuits:
  - Can be designed to be input-output time-invariant
  - Can save power when signal conditions allow.

- VLSI/ULSI special-purpose analog computers may be promising for:
  - computing fast approximate solutions to tough numerical problems;
  - helping digital computers achieve accurate results faster.

- Thorough experimental validation and applications of some of the principles presented is being pursued.

- Much remains to be done before the practical feasibility of the systems presented is confirmed.

- Mixed-domain circuits and systems are a largely unexplored area that offers several interesting research possibilities.

References

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