The Bits In-Between

An EE's Guide to Survival Between Microphone and Voice Coil

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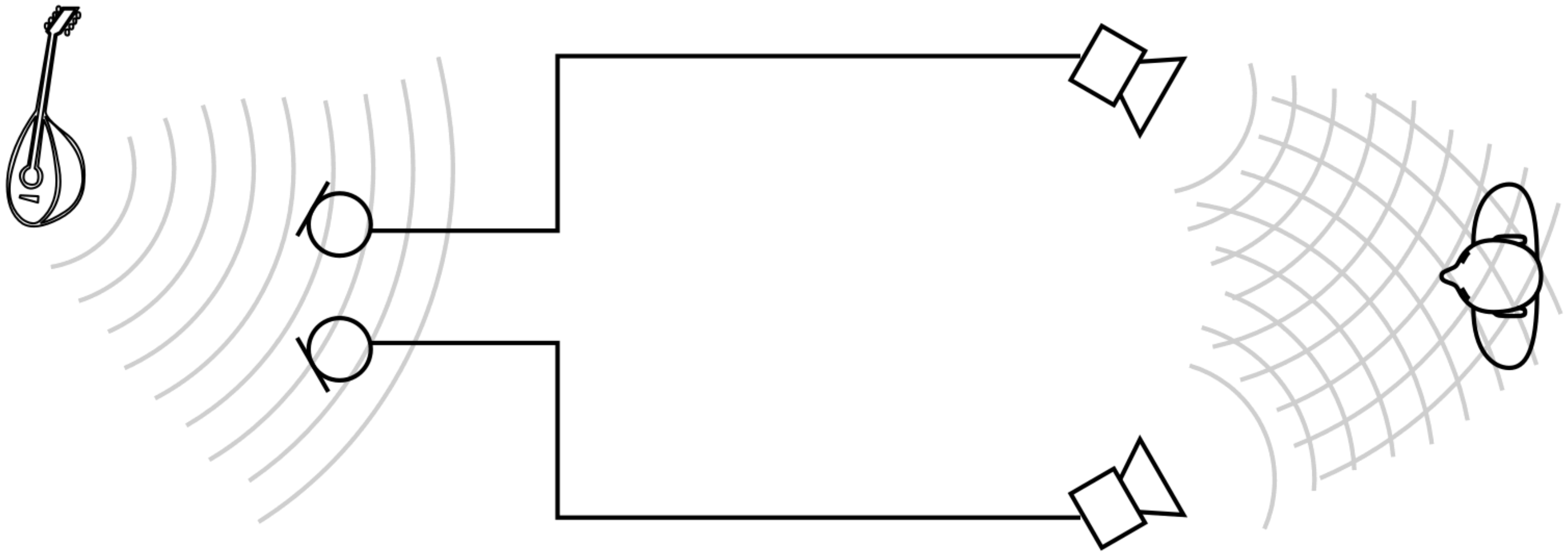
A bit of Perspective

It’s all in the head

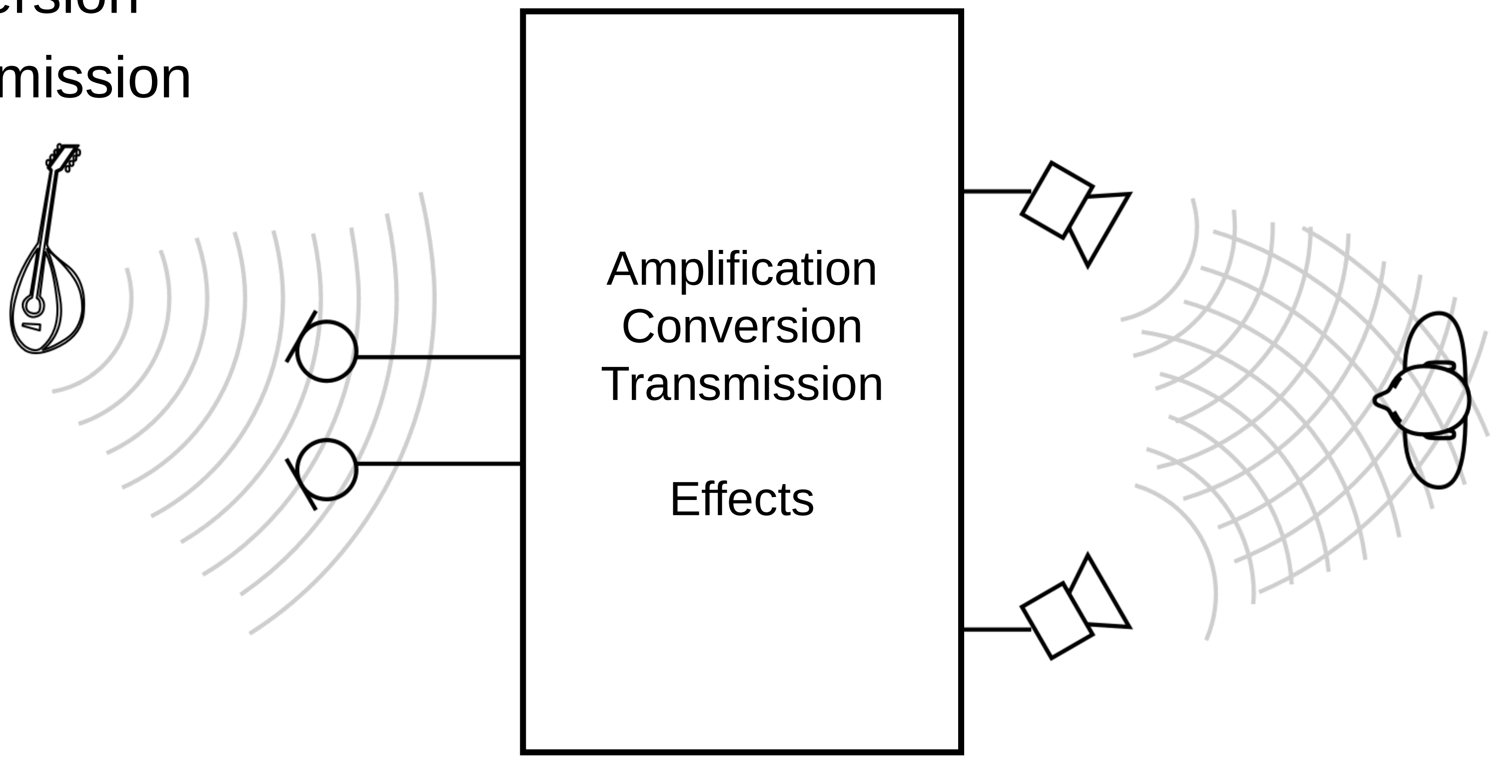
Transducer theory without psychoacoustics=rubbish



A bit of Perspective

The parts most covered in mystique are in fact the utilitarian ones:

- Amplification
- Conversion
- Transmission



- Only effects gear should modify the signal audibly

Contents

Revealed Preference: Euphonic or Transparent?

How Do You Listen?

1. “Preference Test”: compare to reference product
    - Design cycle converges on “best sounding”.

2. “Bypass Test”: compare output to input
    - Design cycle converges on “maximally transparent”.

Note different meanings of “transparent”

- Audiophile: Barrage of fine detail
- Pro: no audible difference between in and out.

Reference Listening Fallacies: Yeah but…

- “I’m listening for the most neutral sound, not for the sound I like best”
  - Result will be what you *think* is most neutral…

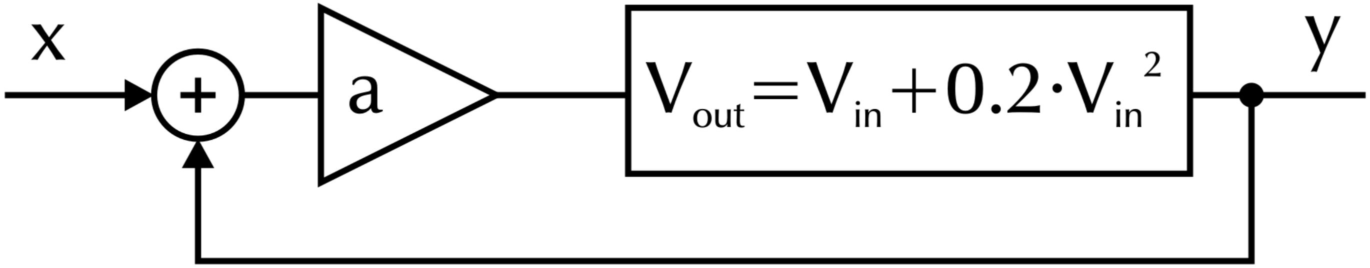
    …Not what you *really* like best
    …Not what is *truly* transparent
    } *Worst of both worlds!!!*

- “I go to live concerts regularly to recalibrate my ears”
  - No actual reproduction takes place (audio=illusion).
  - Your design will be specialised to make your favourite recordings sound realistic

Next on: Unhappy about Negative Feedback

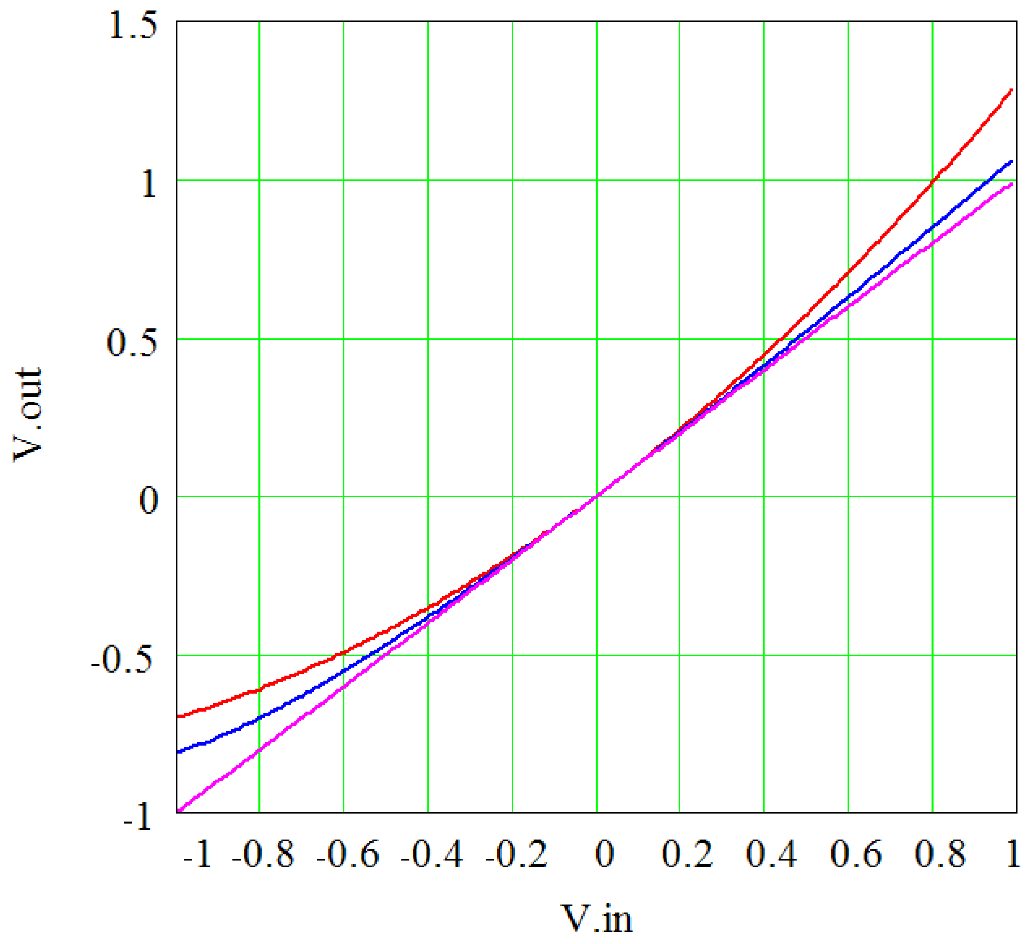
(When) Does Negative Feedback Sound Bad?

Example non-linearity: pure second order function



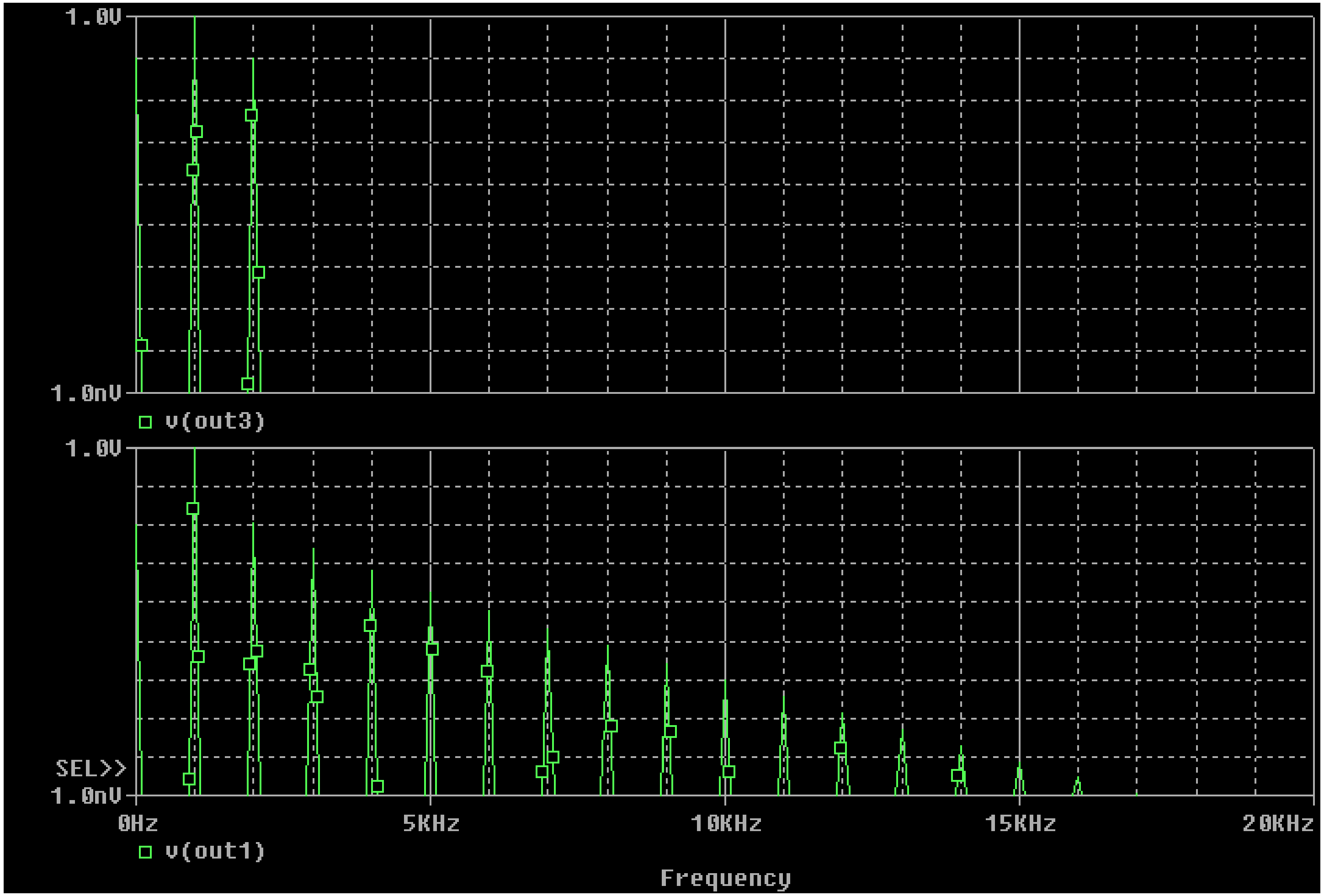
(When) Does Negative Feedback Sound Bad?

- Feedback results in “intermediate” shaped transfer (function has a square root in it)

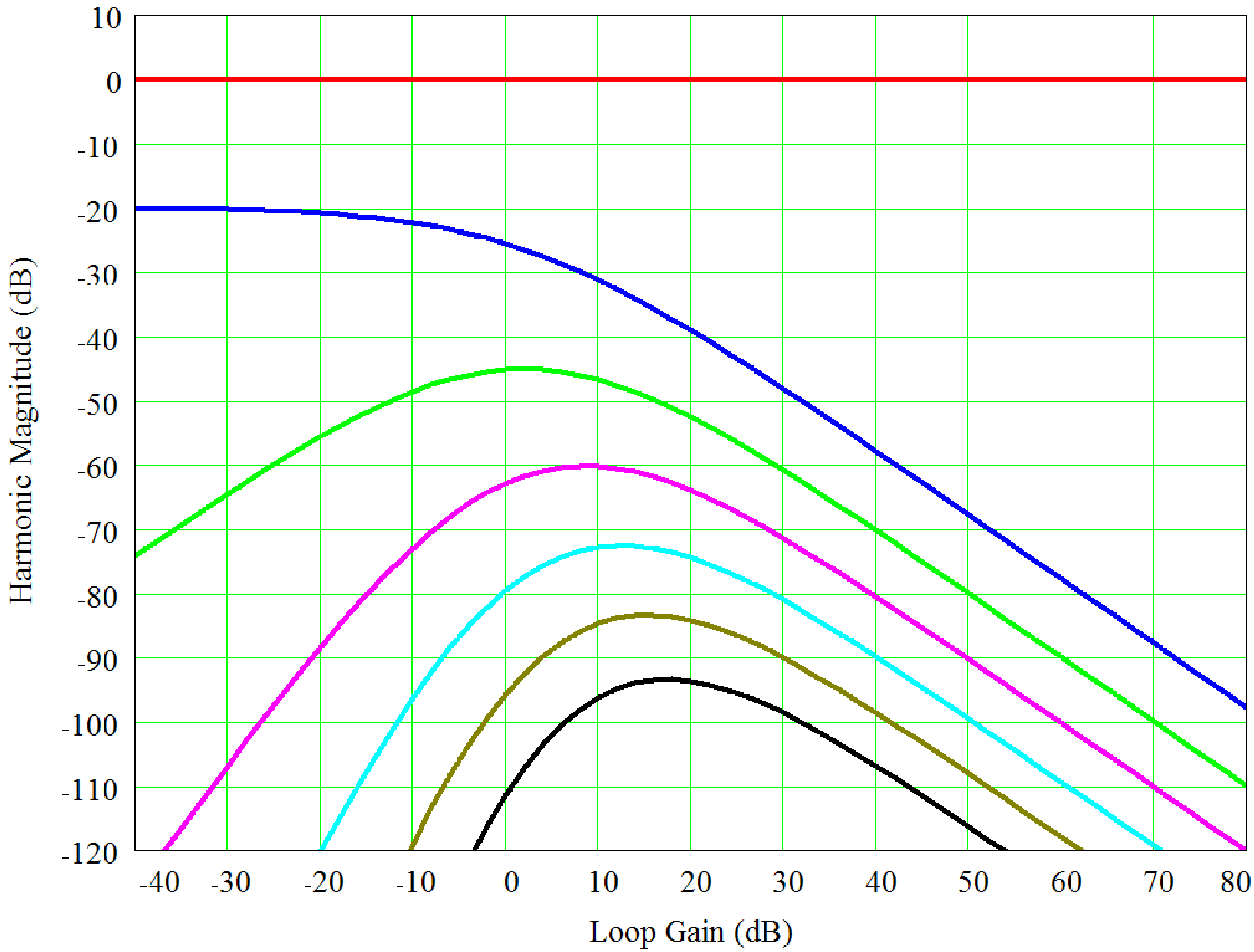


(When) Does Negative Feedback Sound Bad?

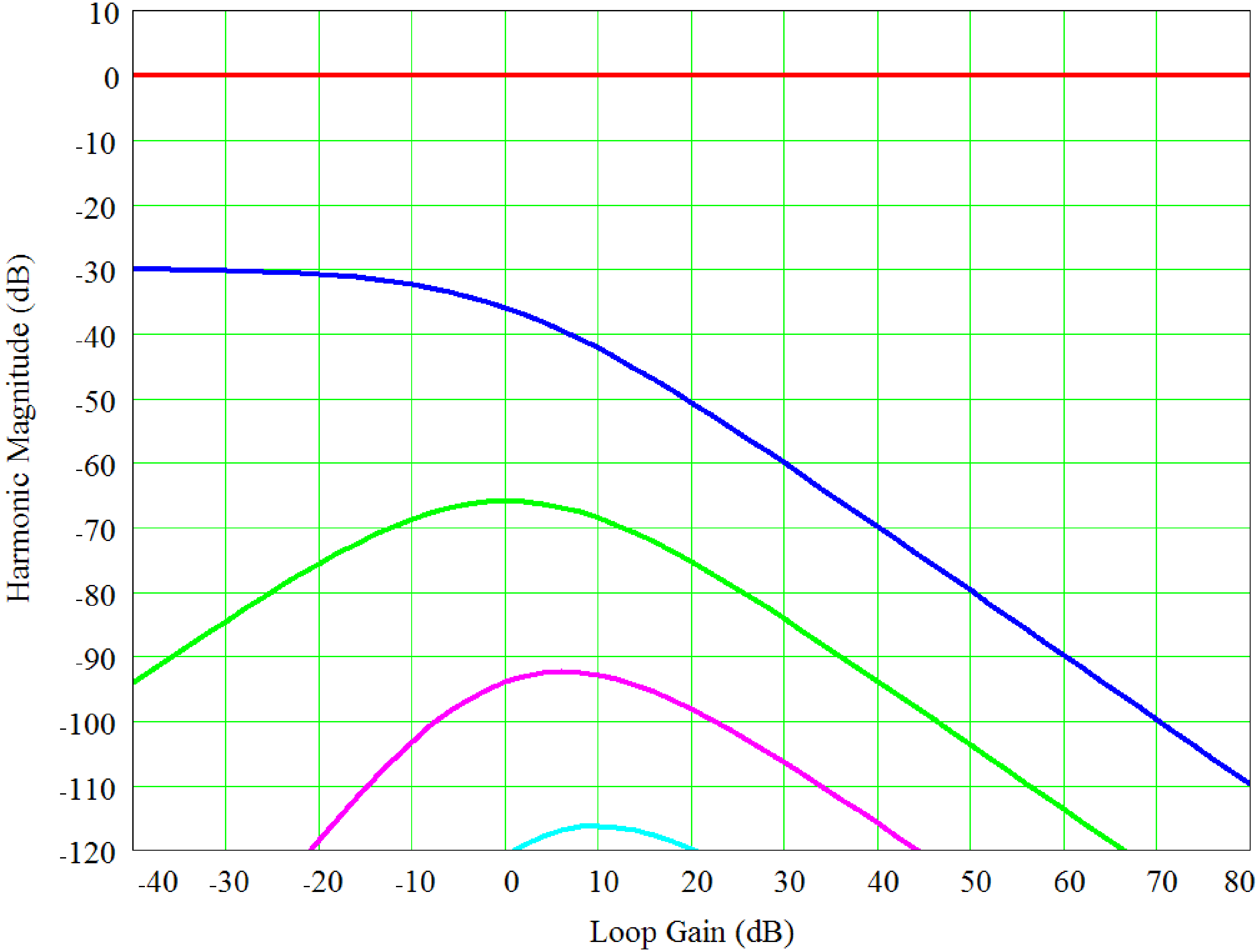
- 2nd harmonic drops.
- New harmonics appear, including odd ones!



(When) Does Negative Feedback Sound Bad?



(When) Does Negative Feedback Sound Bad?



Negative Feedback Guidelines (1)

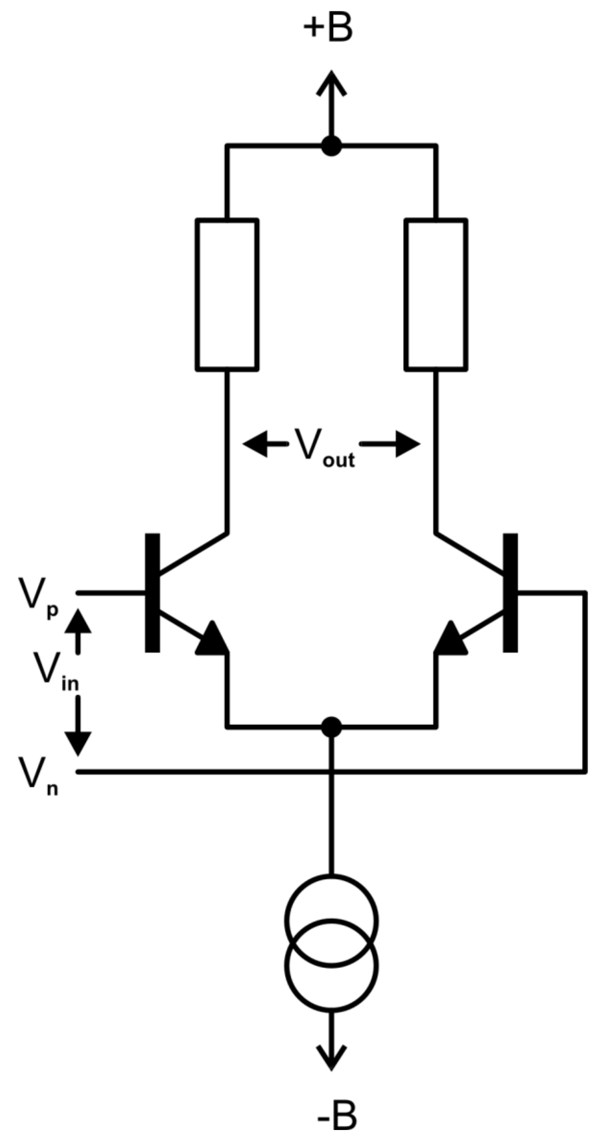
- Practical open-loop errors are too large for guaranteed transparency.
- Feedback is the most effective tool for reducing errors
- <u>Moderate loop gain does more harm than good</u> in realistic circuits.
- Improved open-loop linearity reduces NFB-related products by a greater extent.

Don’t Be a Wimp. Use NFB and use <u>tons</u> of it.

Next on: Op Amp Theory

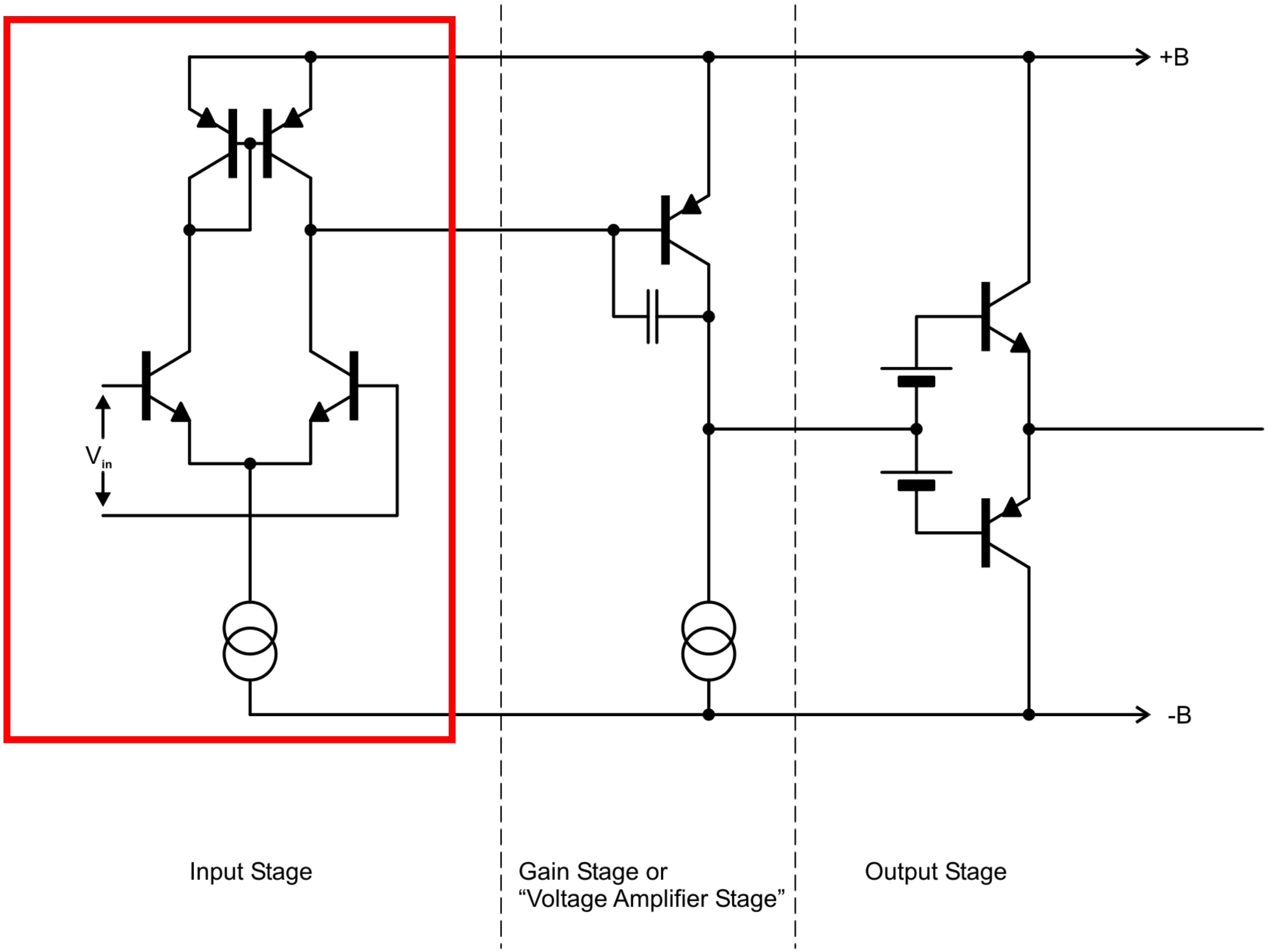
Hands-On Op Amp Theory

This is a voltage amplifier…

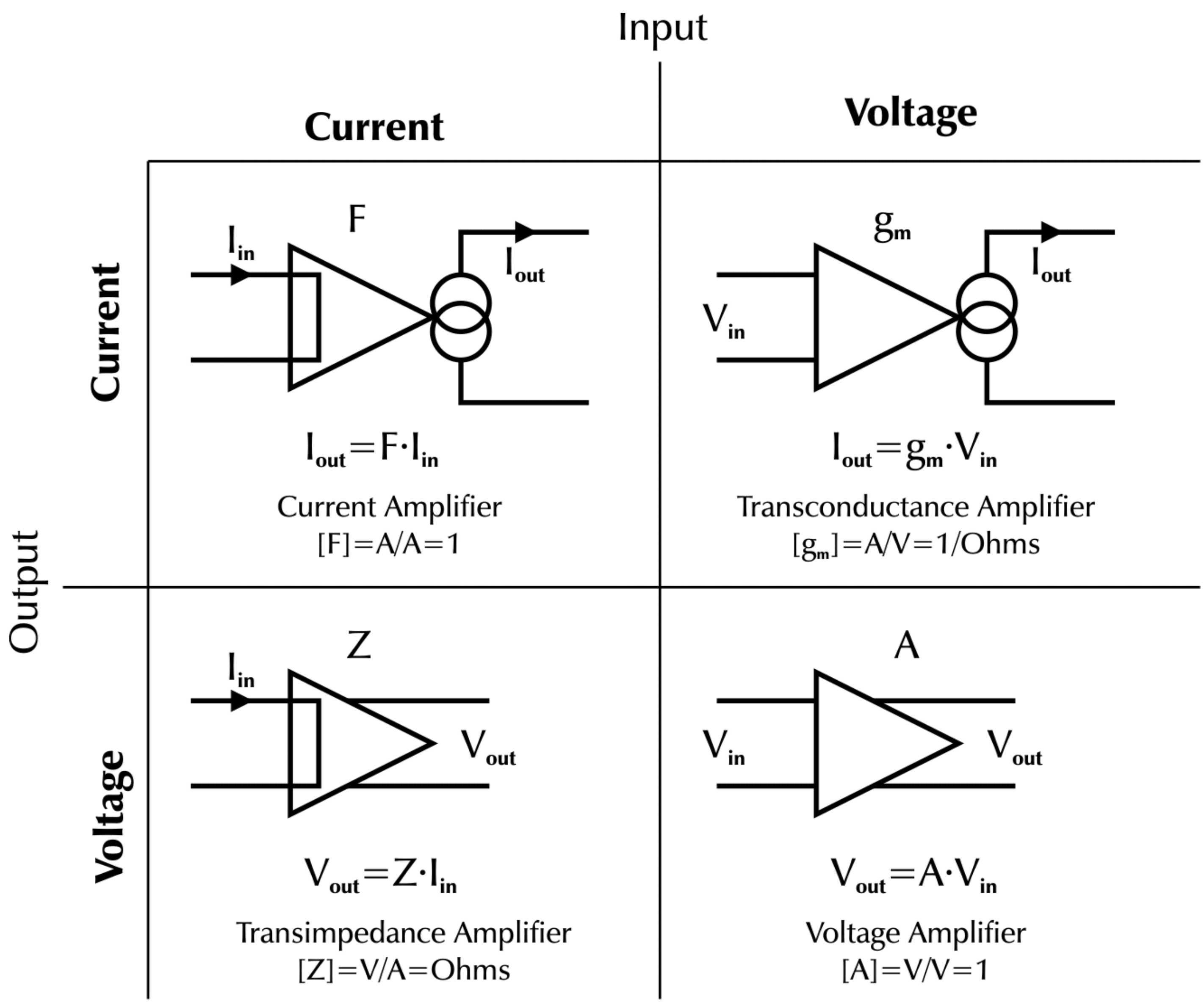


Hands-On Op Amp Theory

…This is not!

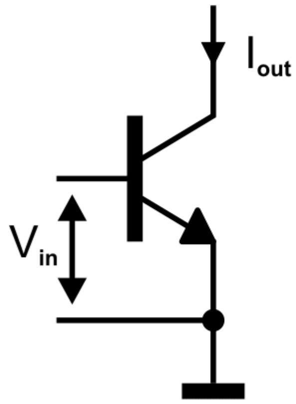


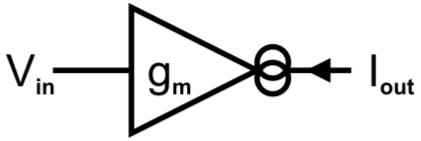
Hands-On Op Amp Theory



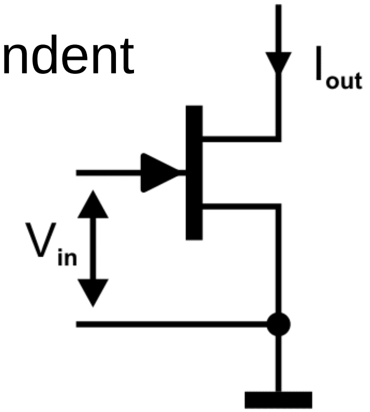
Transconductance Amplifiers

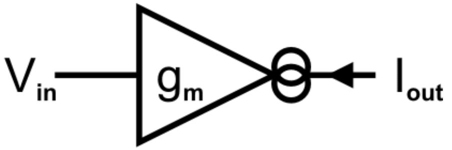
- Common Emitter Circuit
  - $g_m = I_E/26mV$
  - Moderate Z_{in}
  - High Z_{out}





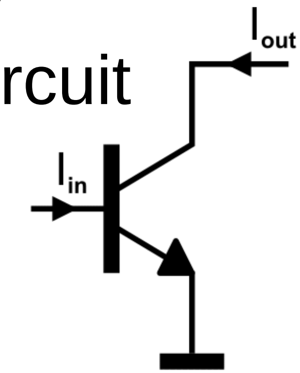
- Common Source Circuit
  - g_m=device and current dependent
  - Very high Z_{in}
  - High Z_{out}

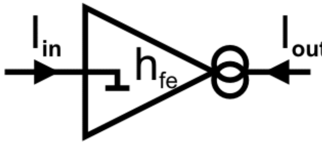




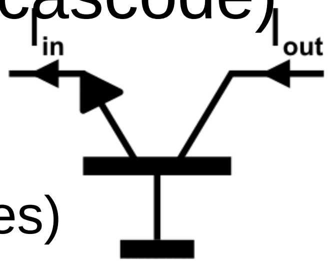
Current Amplifiers

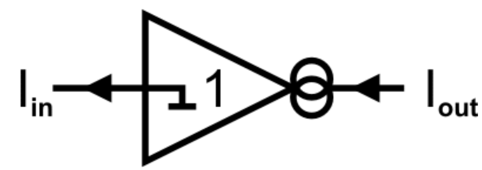
- Common Emitter Circuit
  - $A_I = h_{fe}$
  - moderate Z_{in}
  - high Z_{out}



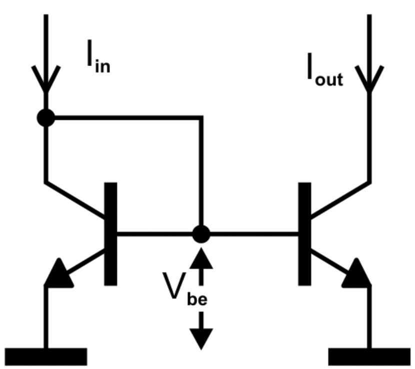


- Common Base Circuit (=cascode)
  - $A_I \approx 1$
  - low Z_{in}
  - Very high Z_{out} (Ccb dominates)





- Current Mirror
  - $A_I \approx 1$
  - low Z_{in}
  - High Z_{out}



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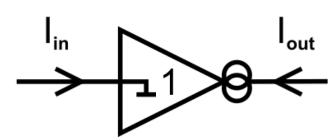
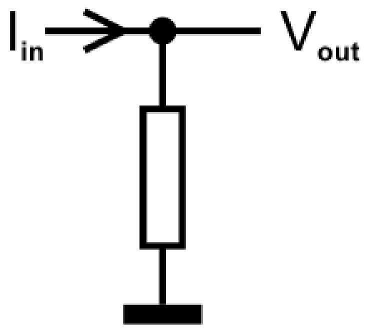
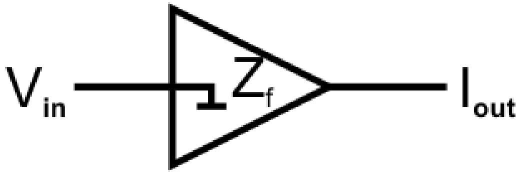


Hands-On Op Amp Theory

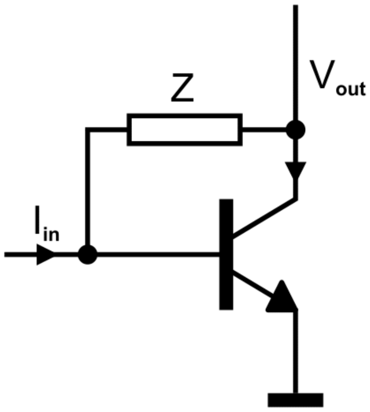
Transimpedance Amplifiers (I/V converters)

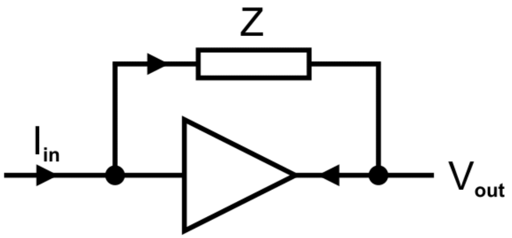
- No fundamental circuit, except perhaps:
  - Z_{in}=Z_{out}="Low" only if source and load are high-Z





- Feedback type I/V
  - Z_{in}≈1/gm
  - Z_{out}≈1/gm

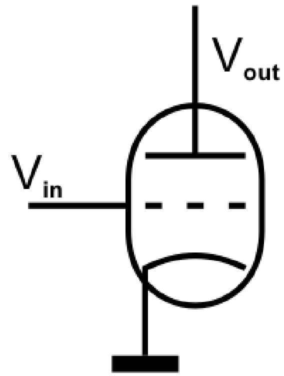


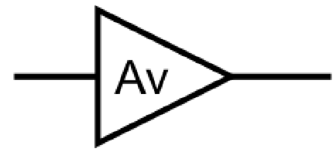


Hands-On Op Amp Theory

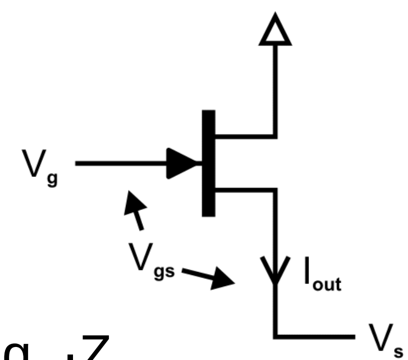
Voltage Amplifiers

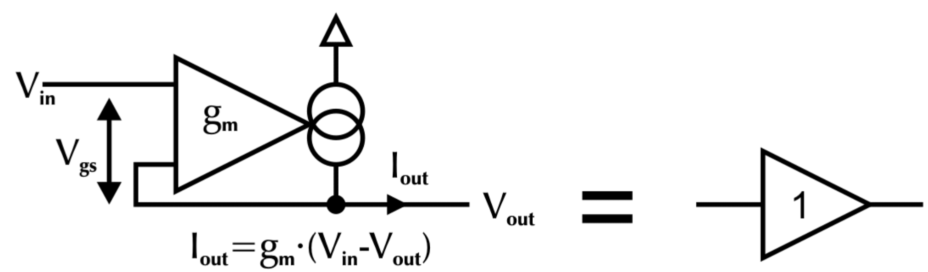
- Only one truly fundamental voltage amplifier:
  - Very High Zin
  - Low Zout





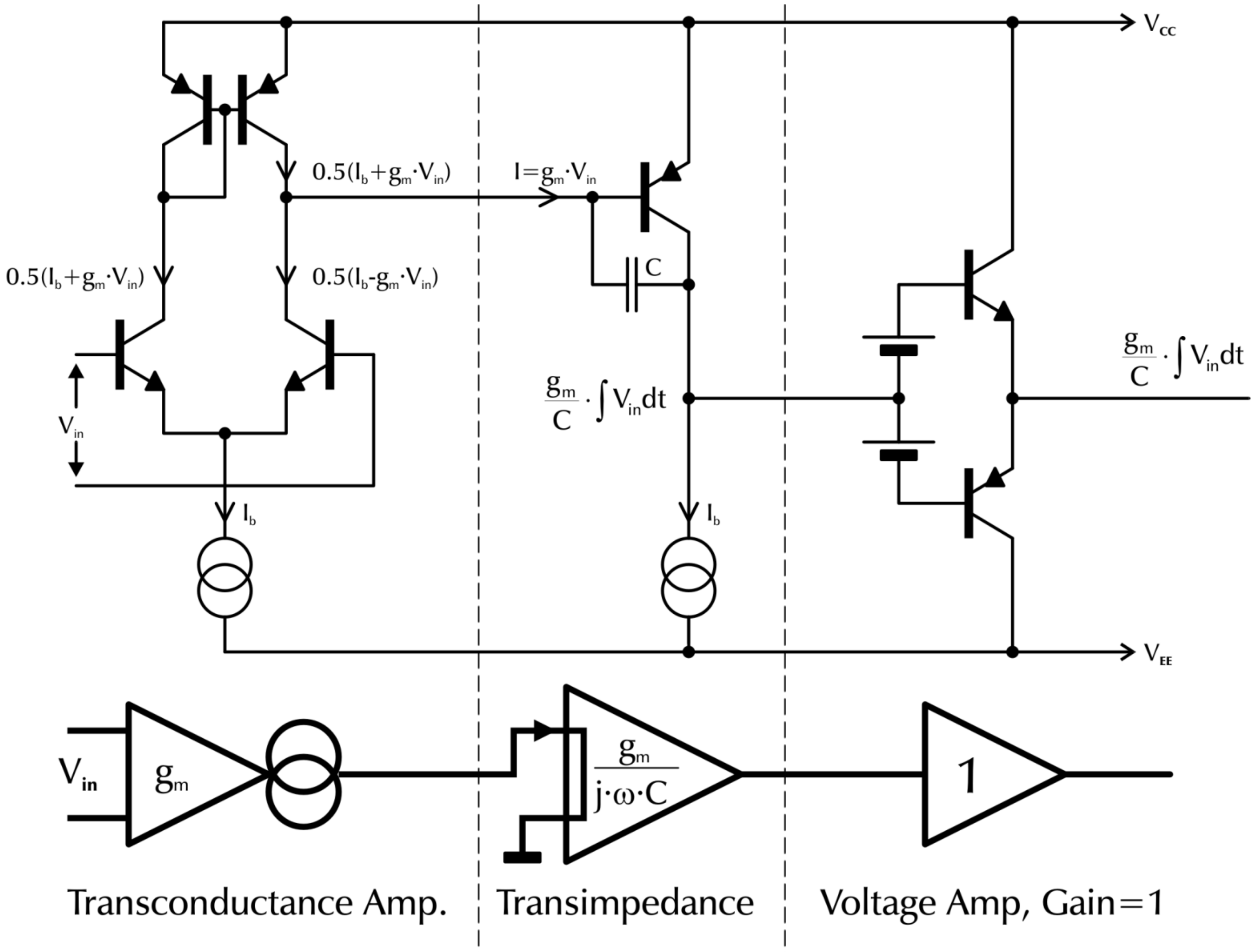
- Follower = transcond. amp with voltage feedback.
  - High Z_{in}
  - $Z_{out}=1/g_m$



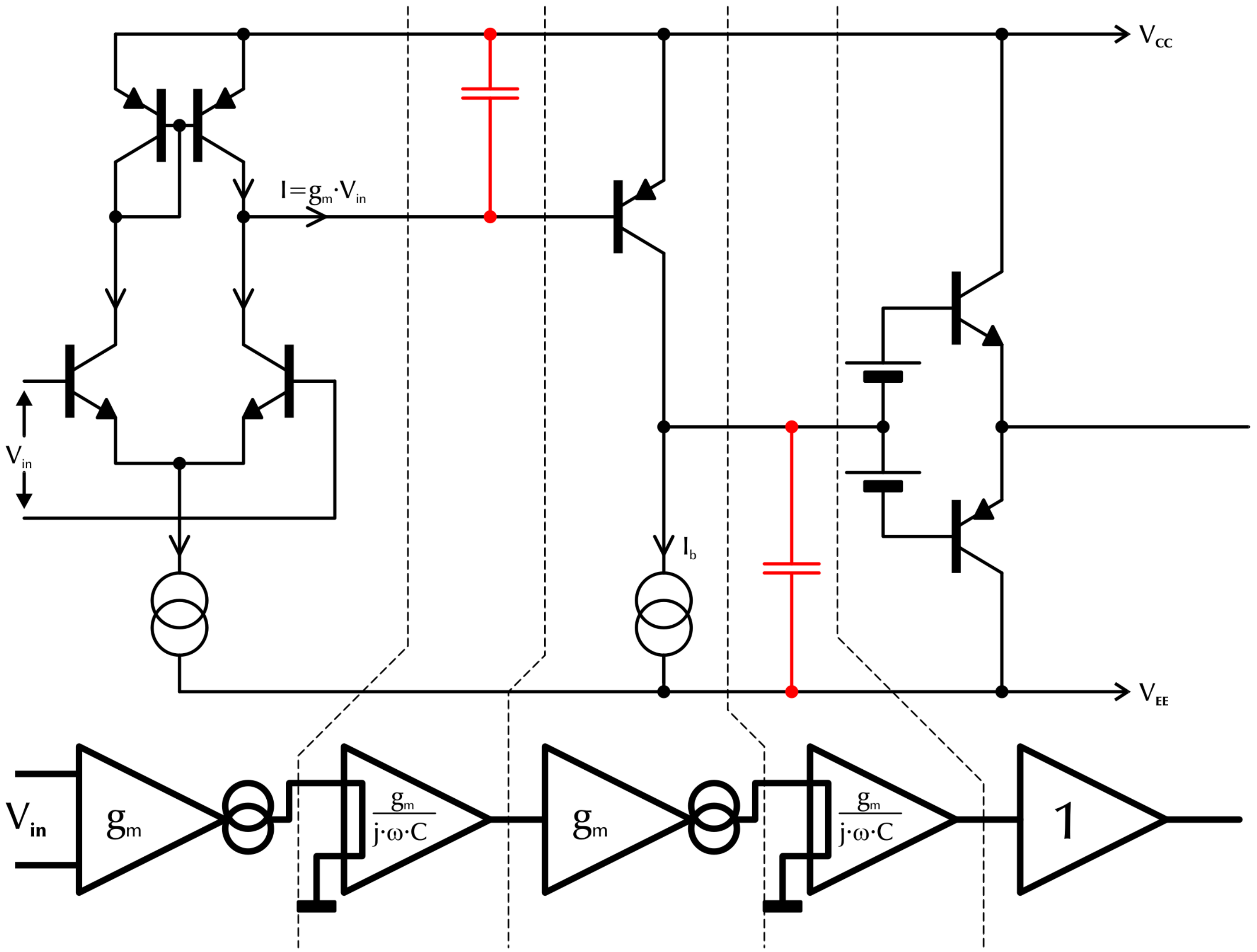


  - Loop Gain=$g_m \cdot Z_L$

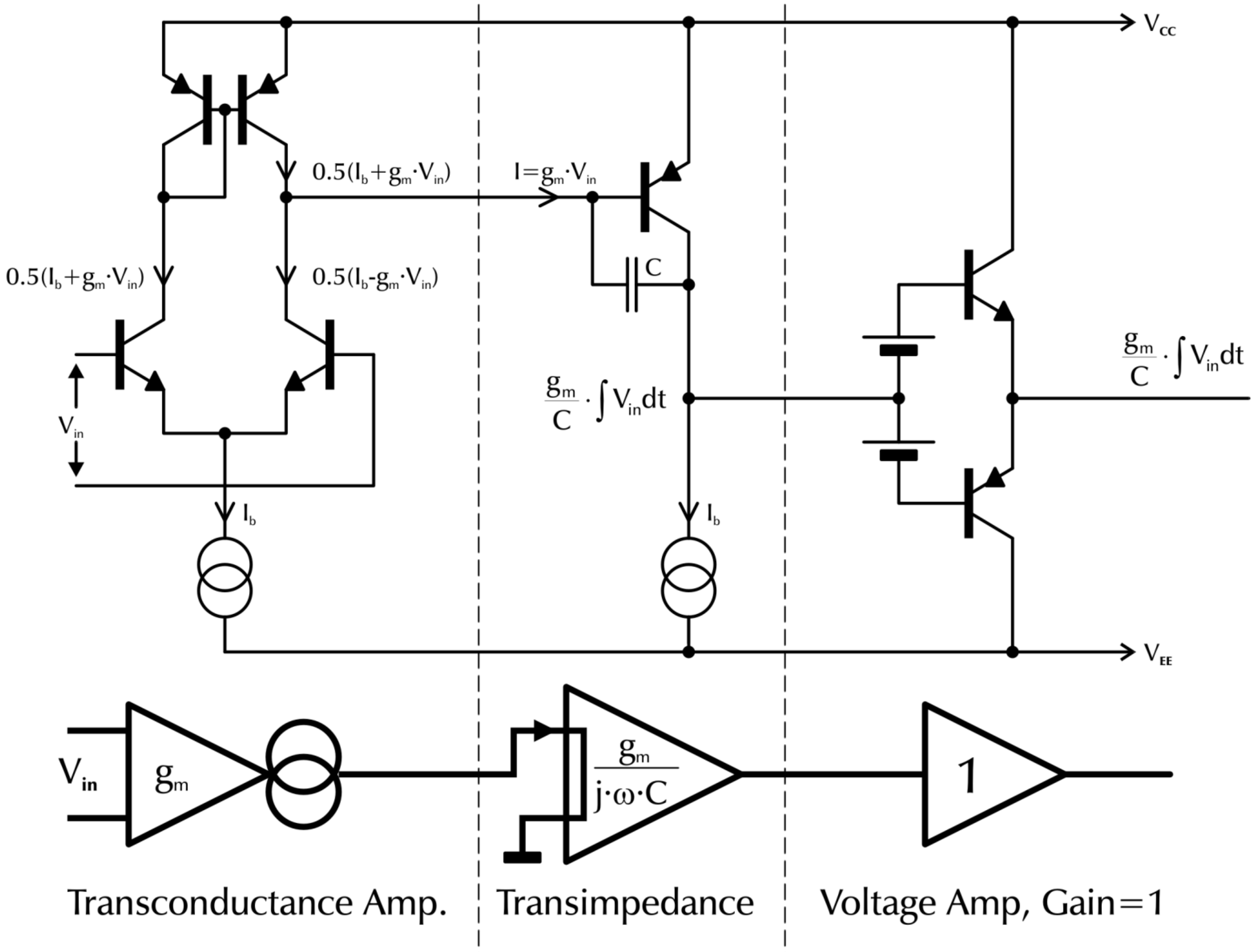
Hands-On Op Amp Theory



Hands-On Op Amp Theory



Hands-On Op Amp Theory

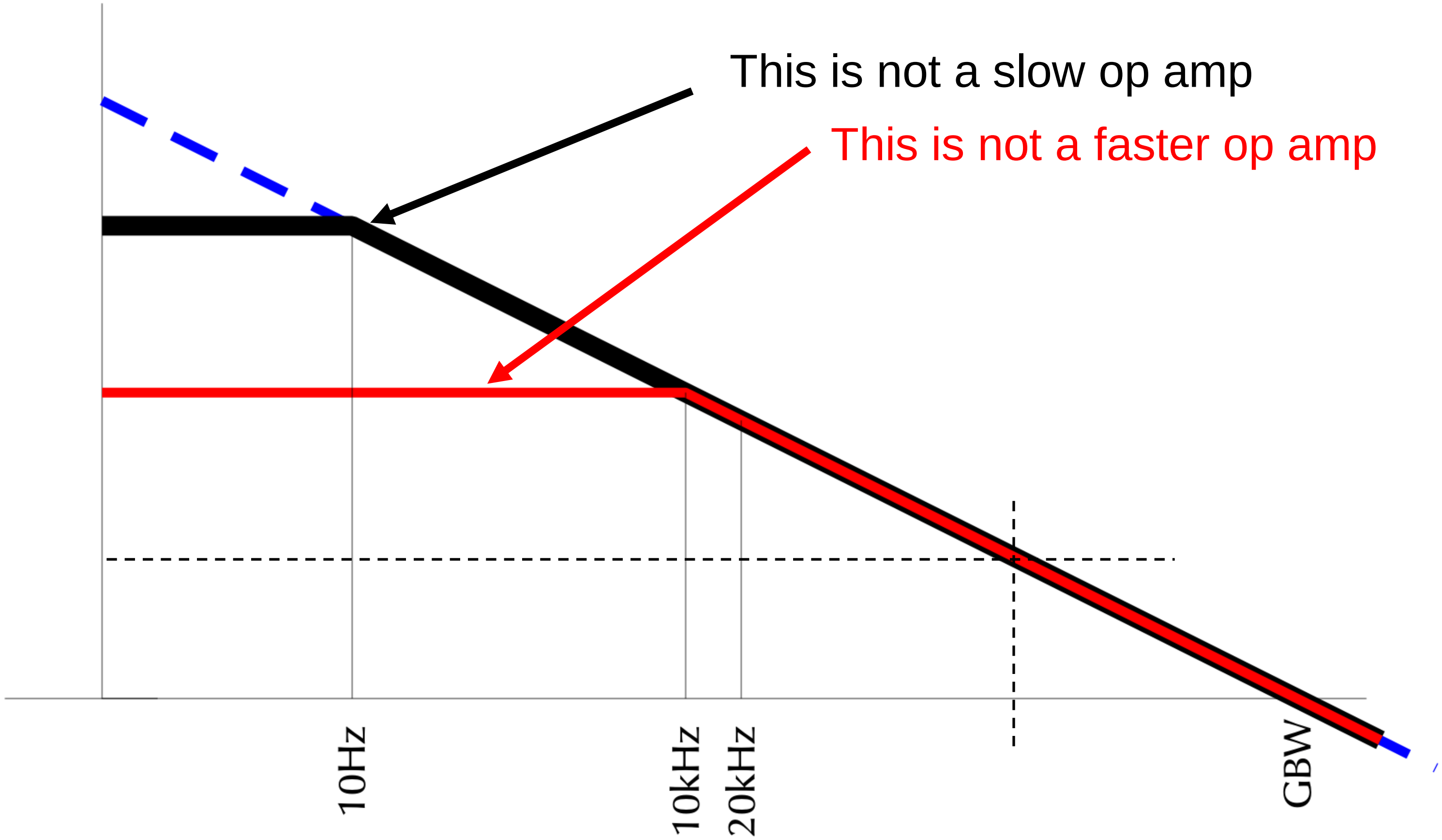


An op amp is an integrator

$$\left|A_V(f)\right| = \frac{g_m}{C} \cdot \frac{1}{2 \cdot \pi \cdot f \cdot} = \frac{g_m}{2 \cdot \pi \cdot C} \cdot \frac{1}{f}$$

$$GBW = \frac{g_m}{2 \cdot \pi \cdot C}$$

An op amp is an integrator



Negative Feedback Guidelines (2)

Impact of integrating character on sound

- Loop gain drops 20dB/decade
  - ⇨Closed-loop THD increases with frequency
  - ⇨Spectral distribution shifts towards higher frequencies

- Euphony In Action! Rising THD vs frequency profile has a recognisable sonic signature.
  - HF is only mildly affected except in very bad cases.
  - Bottom end becomes extremely “tight”, “powerful” and “controlled”.
  - Often attributed to “huge current reserve” of behemoth power stage. Really caused by HF THD of sluggish amp.
  - Propagates “Damping Factor” myth.

Negative Feedback Guidelines (2)

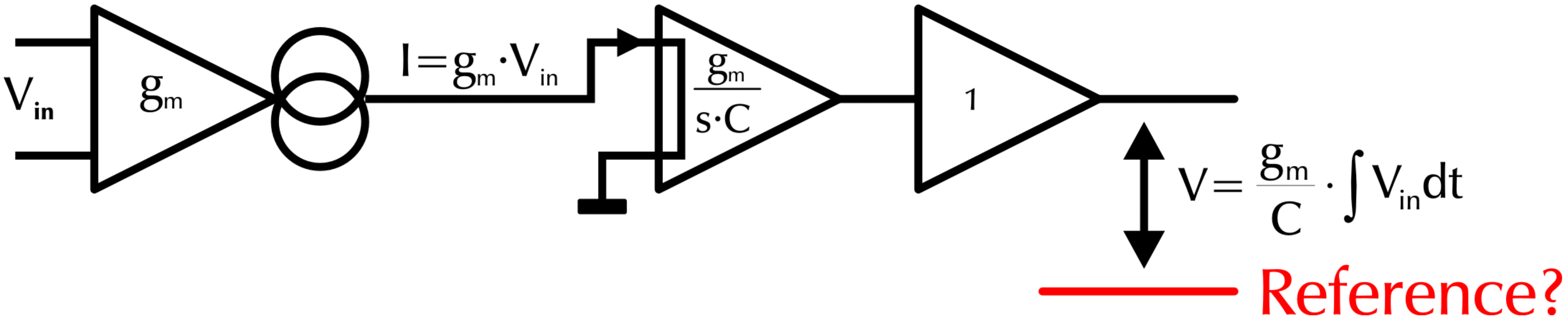
Not scientifically established but useful nonetheless:

- When you’re strapped for loop gain at 20kHz, limit low-frequency loop gain to the same value.
- THD becomes higher but constant throughout the audio band.
- Colouration becomes less obvious and less annoying.

Next on: PSRR

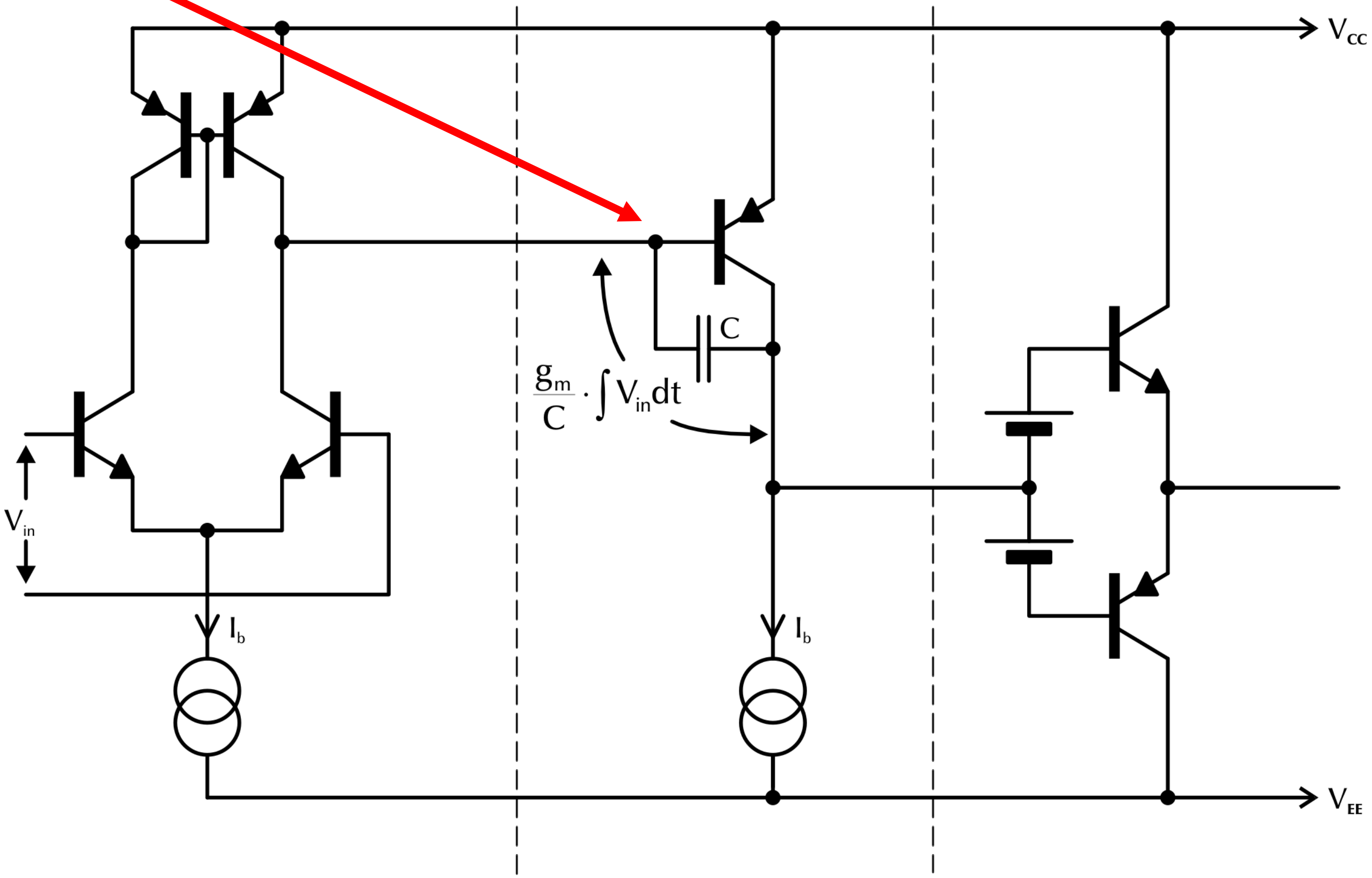
Hands-On Op Amp Theory: The PSRR Gotcha

Hang on… What's the output voltage referring to?



Hands-On Op Amp Theory: The PSRR Gotcha

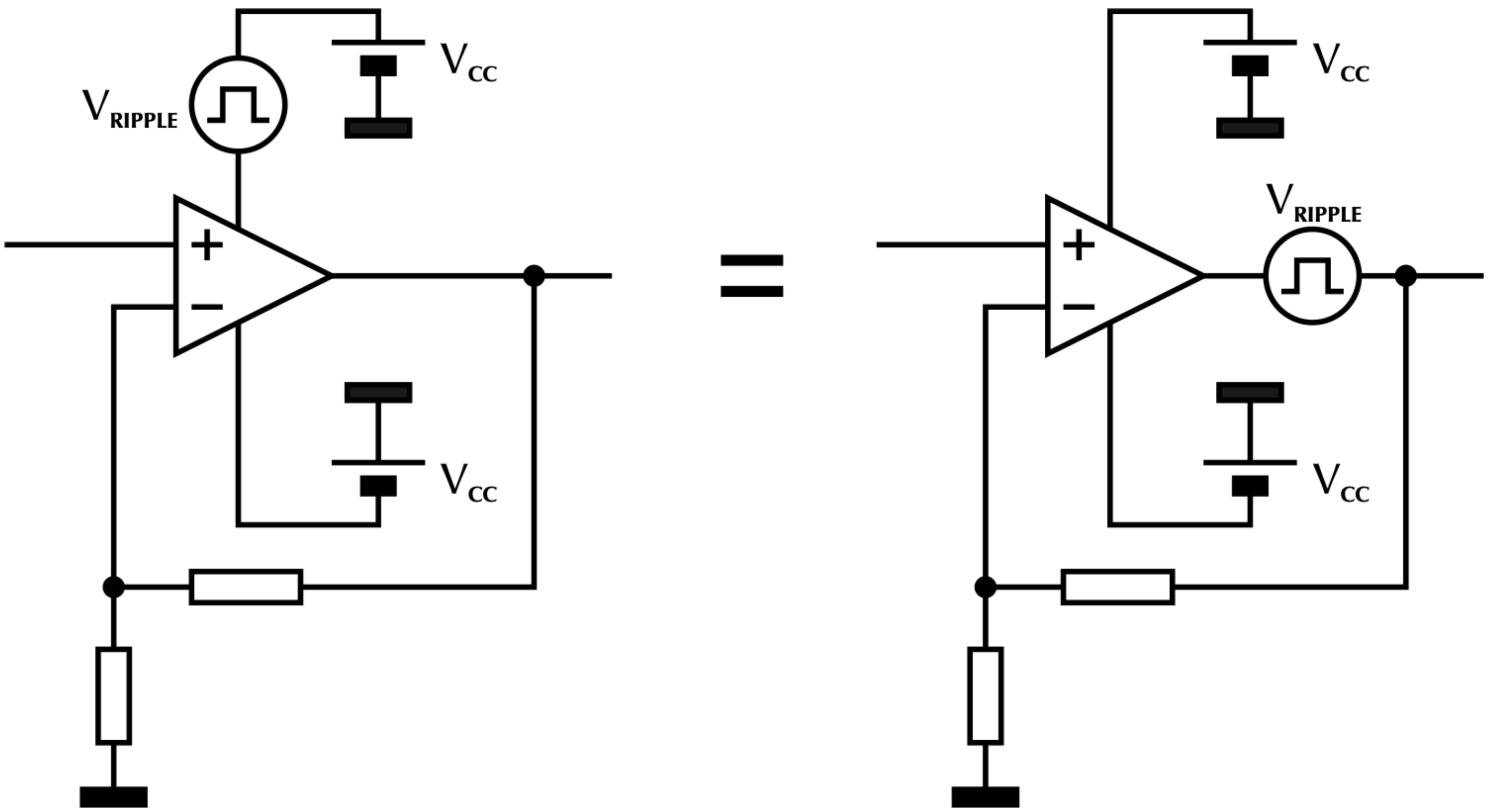
Here!!!



Hands-On Op Amp Theory: The PSRR Gotcha

So really...

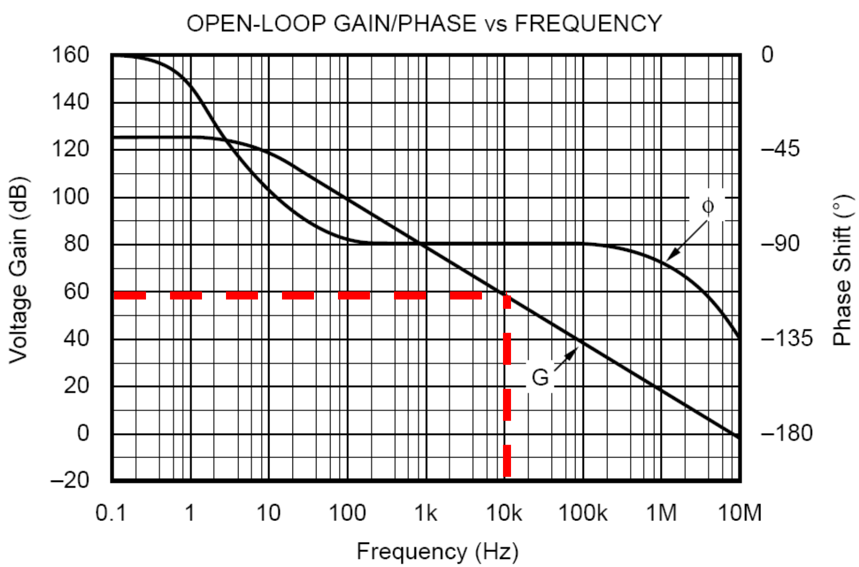
$$V_{out} = 2 \cdot \pi \cdot GBW \cdot \int V_{in} dt + \underline{\underline{V_{CC}}}$$

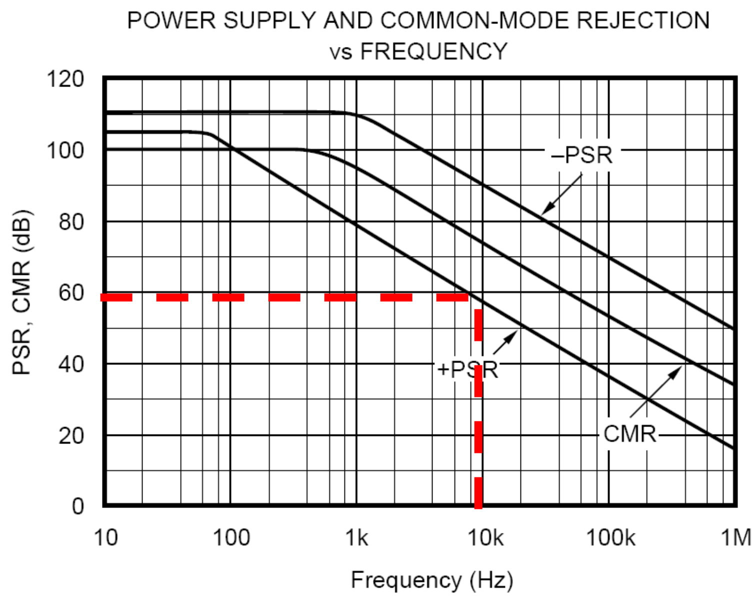


Hands-On Op Amp Theory: The PSRR Gotcha

The PSRR Gotcha:

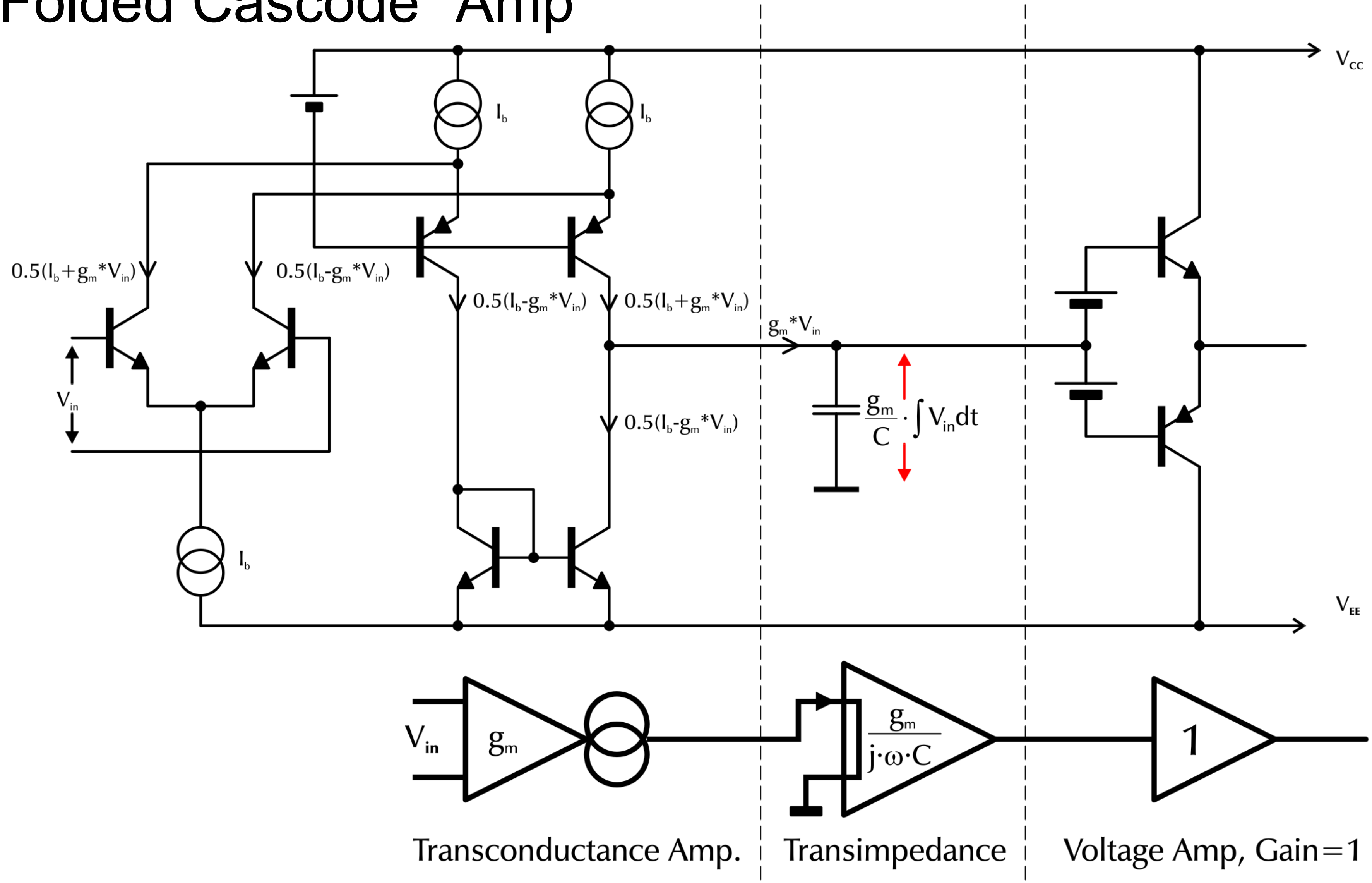
- One rail is output reference
- PSRR is essentially ZERO
- Measured PSRR = $A_L \approx A_{V,OL} - A_{V,CL}$
- PSRR in typical audio app is <u>not astronomical</u>





Hands-On Op Amp Theory: PSRR fixes

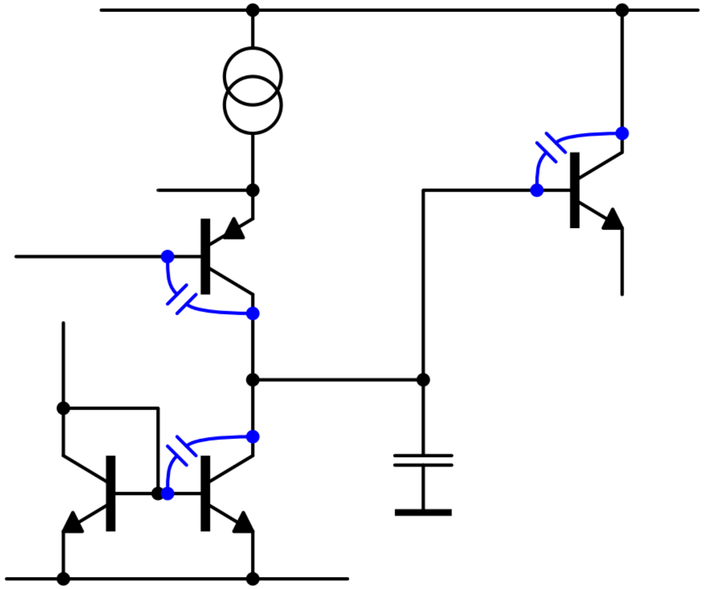
"Folded Cascode" Amp



Hands-On Op Amp Theory: PSRR fixes

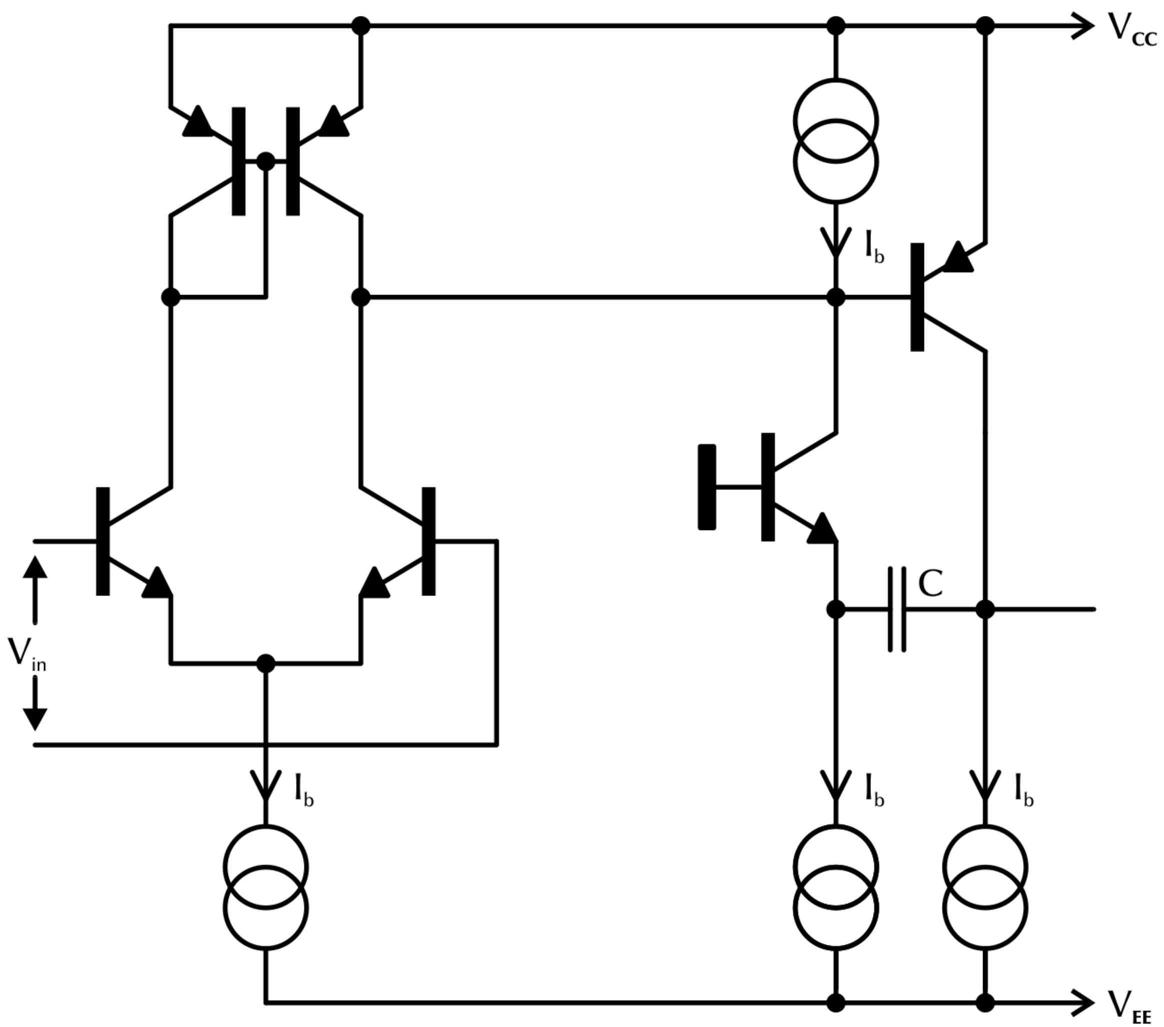
“Folded Cascode” Amp

- Pro:
  - Output Reference is Ground
- Con
  - Buffer impedance is critical
  - Bias sources add noise
  - Non-linear circuit capacitance adds to integration cap



Hands-On Op Amp Theory: PSRR fixes

Cascoding the current junction



Hands-On Op Amp Theory: PSRR fixes

Cascoding the current junction

- Pro:
  - Reference is made explicit
  - Other advantages of feedback transimp stage remain
- Con:
  - Bias sources add noise to Transcond stage output current

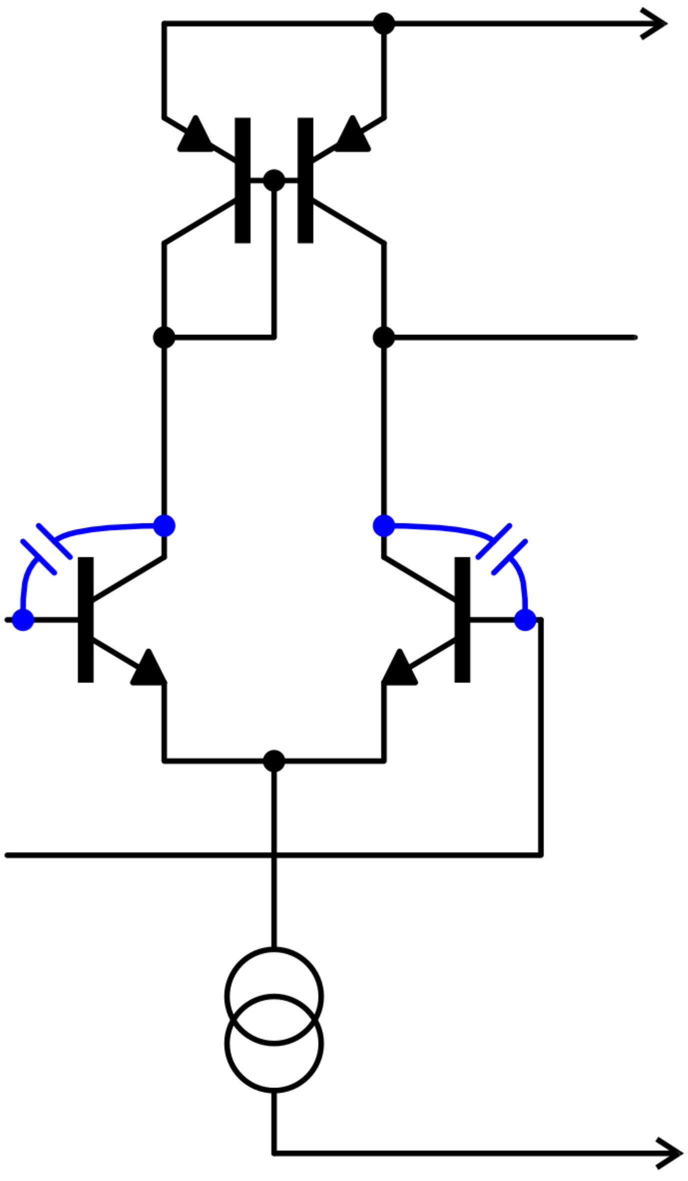
Hands-On Op Amp Theory: CM distortion

Manifestation

- 20dB/dec THD increase

Causes

- Non-linear input capacitance
  - Dominant problem
- Transistor mismatch
  - Also limits DC PSRR
- Load mismatch
  - All but negligible effect



Hands-On Op Amp Theory: CM distortion

Circuit sensitivity

- All 3 effects happen in noninverting mode
- None happen in inverting mode

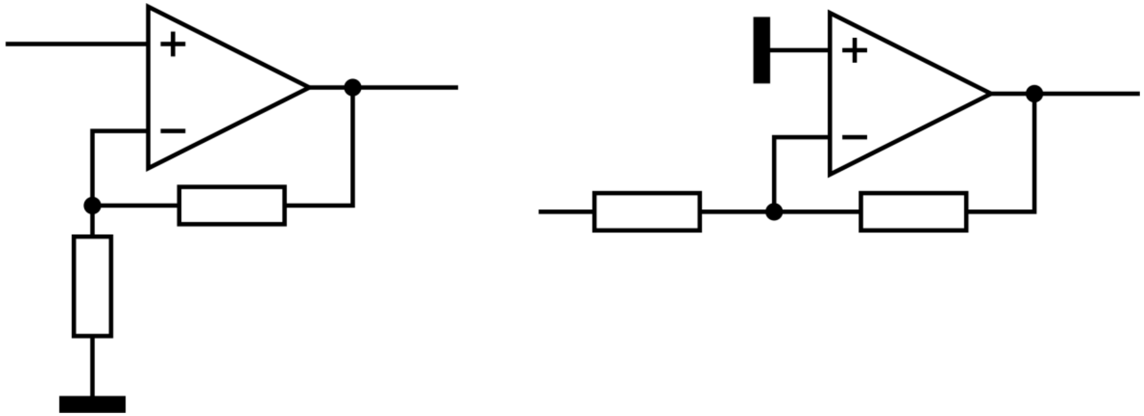
Hands-On Op Amp Theory: CM distortion

Always Invert?

- Guaranteed fix
- Useless in low-noise circuits.

Impedance Matching

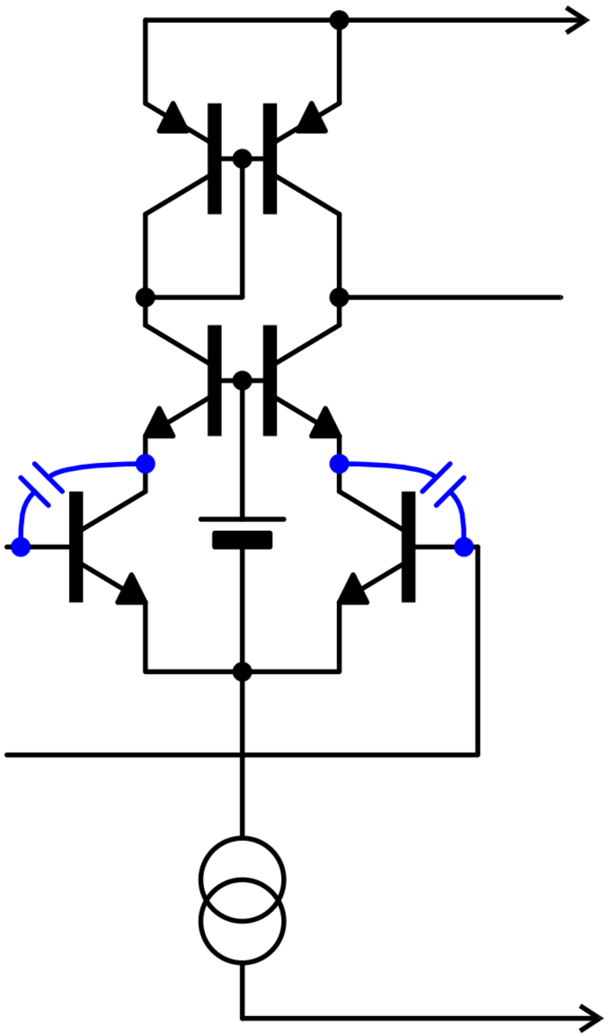
- Eliminates dominant cause
- Source impedance not always known
- Noise penalty



Hands-On Op Amp Theory: CM distortion

Input Stage Improvements

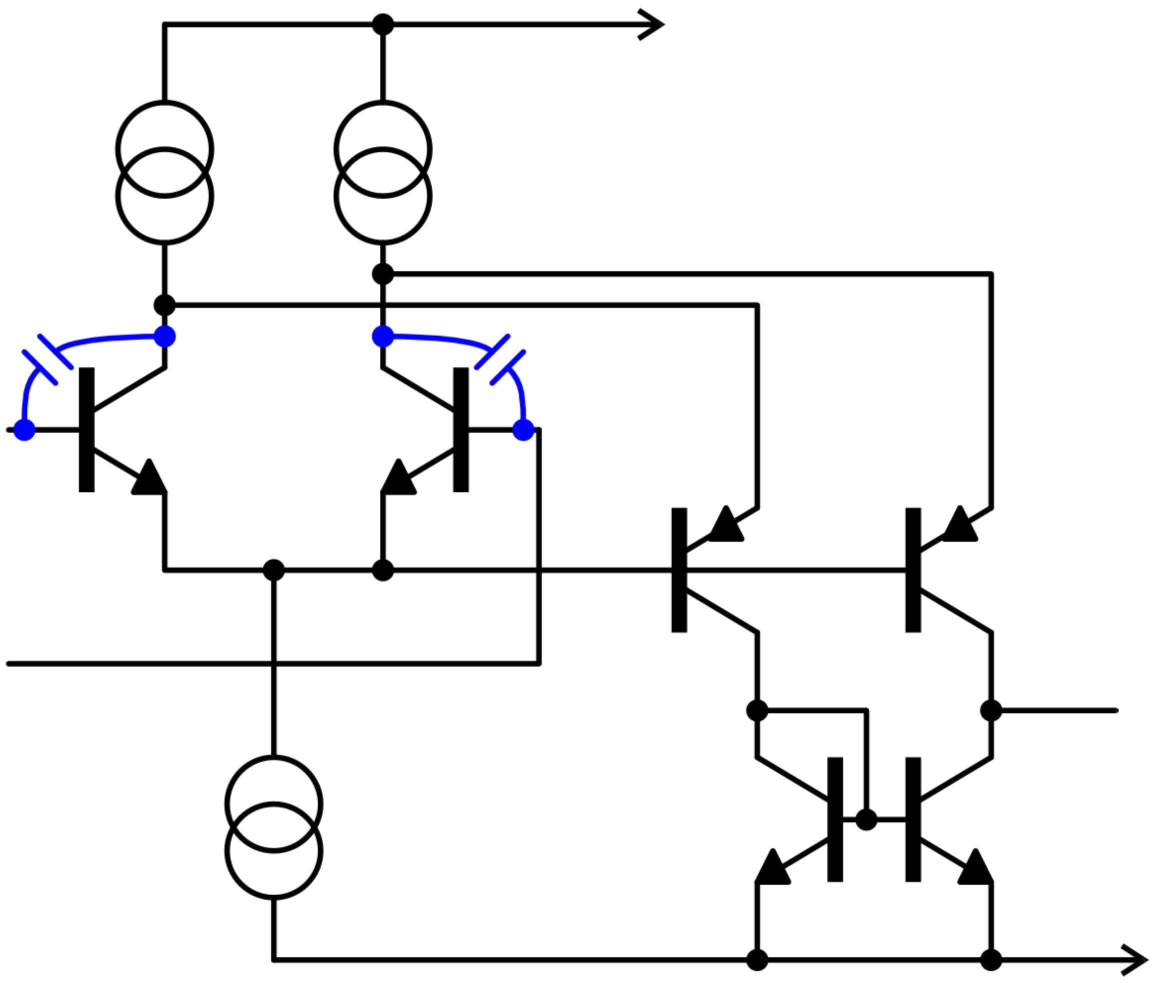
- Boot strapped cascode



Hands-On Op Amp Theory: CM distortion

Input Stage Improvements

- Boot strapped folded cascode



Hands-On Op Amp Theory: Going discrete?

Reasons for going Discrete

- Need more headroom
- Trade typical IC compromises for better performance
  - Input Common Mode range
  - Low-supply operation
  - Lack of 6th connection

Not reasons for going Discrete

- “Discrete is better”
  - Come off it, IC technology is mature
  - Discrete copy of IC op amp has the same drawbacks

Next on: Minimalism

Minimalist Design, or maybe not?

Basic premise of minimalism

- “Any component a signal passes through, degrades it”
  - Underlying assumption: the whole is always the sum of the parts

- Associated philosophy: “zero feedback”

...do I sense a self-fulfilling prophesy coming?

Minimalist Design, or maybe not?

Source of the confusion: inaccurate wording.

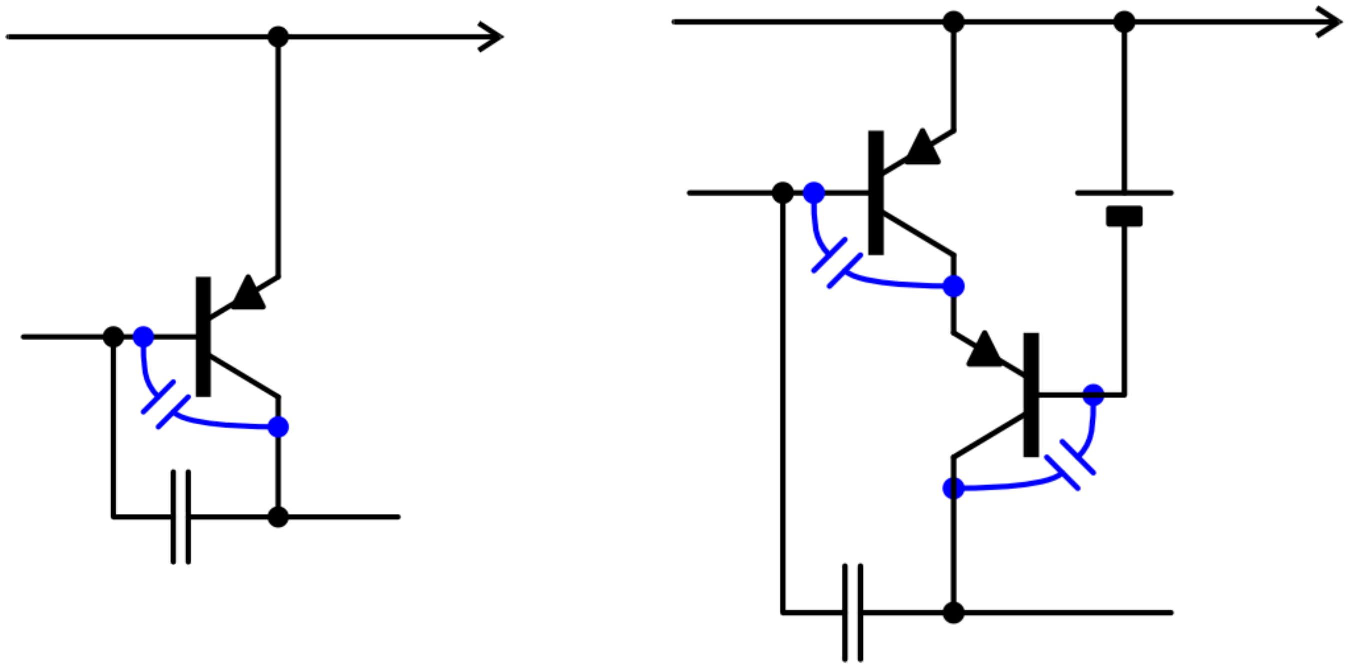
- Let’s correct this:
  “Any *process* a signal goes through, degrades it”

- A bunch of parts enclosed in a feedback loop = ONE process.
  - Result is not “sum of parts”.
  - Neither mathematically, nor sonically.

- Can a process be improved by adding parts? Yes it can!

Minimalist Design, or maybe not?

Example 1: Cascode

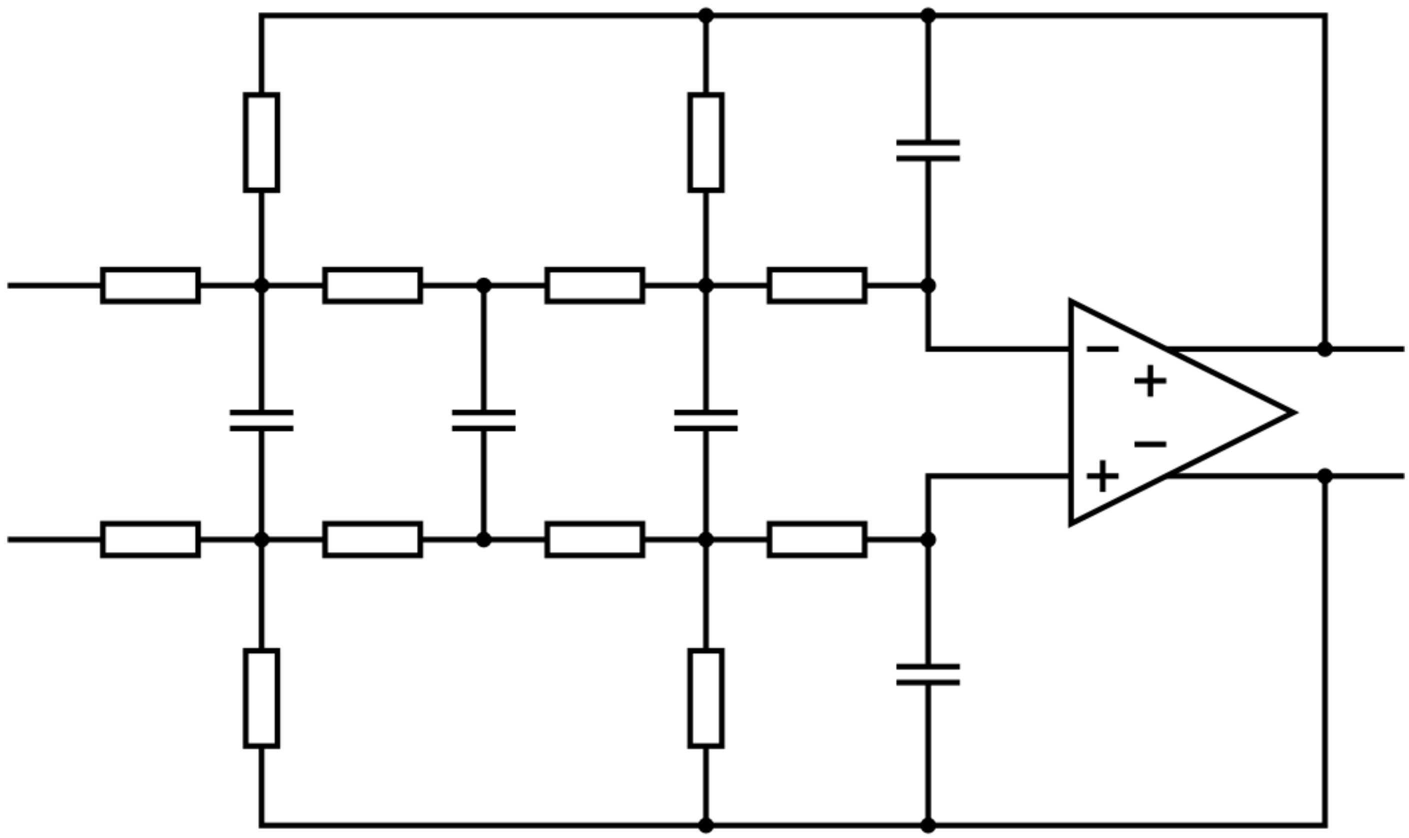


- Cascoding is also accepted by minimalists
  - Undercuts the “sum of parts” premise

Minimalist Design, or maybe not?

Example 2: 4th order low-pass filter.

- First try: Minimalist, only one op amp



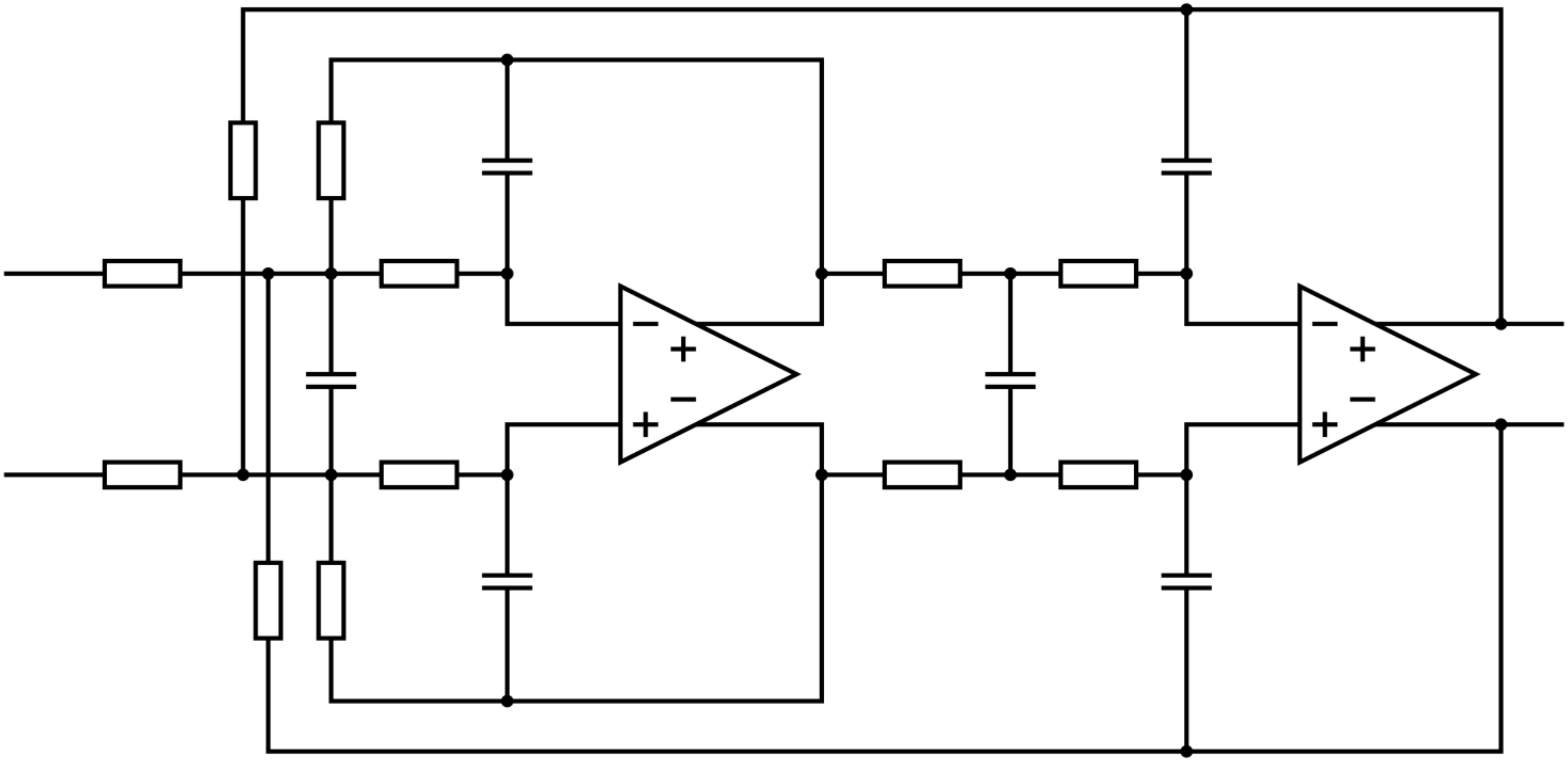
Minimalist Design, or maybe not?

Problem: Noise gain is high

- Noise outstrips DAC’s
- Op amp is starved of loop gain
- Frequency response deviates noticeably from ideal
- THD comes out of noise floor
  - Worse than 2 separate 2nd order sections
  - Sounds worse too...

Minimalist Design, or maybe not?

Second try: Two-op amp filter w/ global feedback



Minimalist Design, or maybe not?

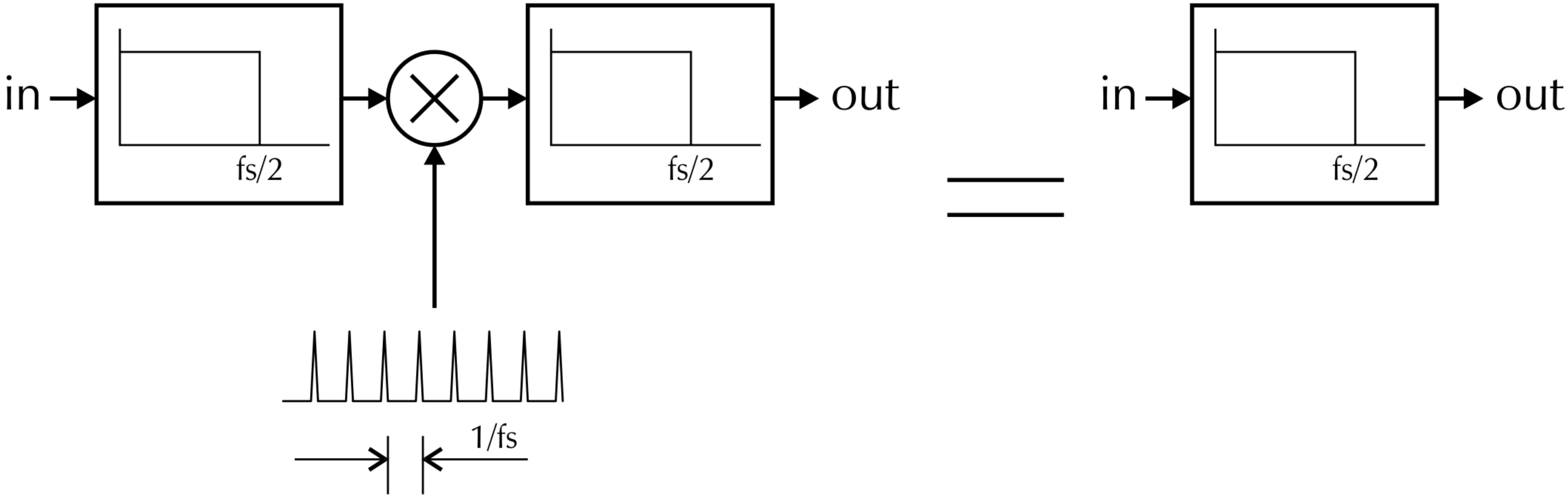
Second try: Two-op amp single stage filter

- Complexity doubles
- Either stage adds loop gain to the other.
- THD is lower than a single stage (and unmeasurable)
- Converter chain is now audibly transparent

Next on: Digital Filters

Digital Filters in AD/DA Converters

Sampling Theory's Basic Promise



A sampler flanked by low-pass filters with sufficient attenuation at fs/2 does exactly the same as the low-pass filters alone.

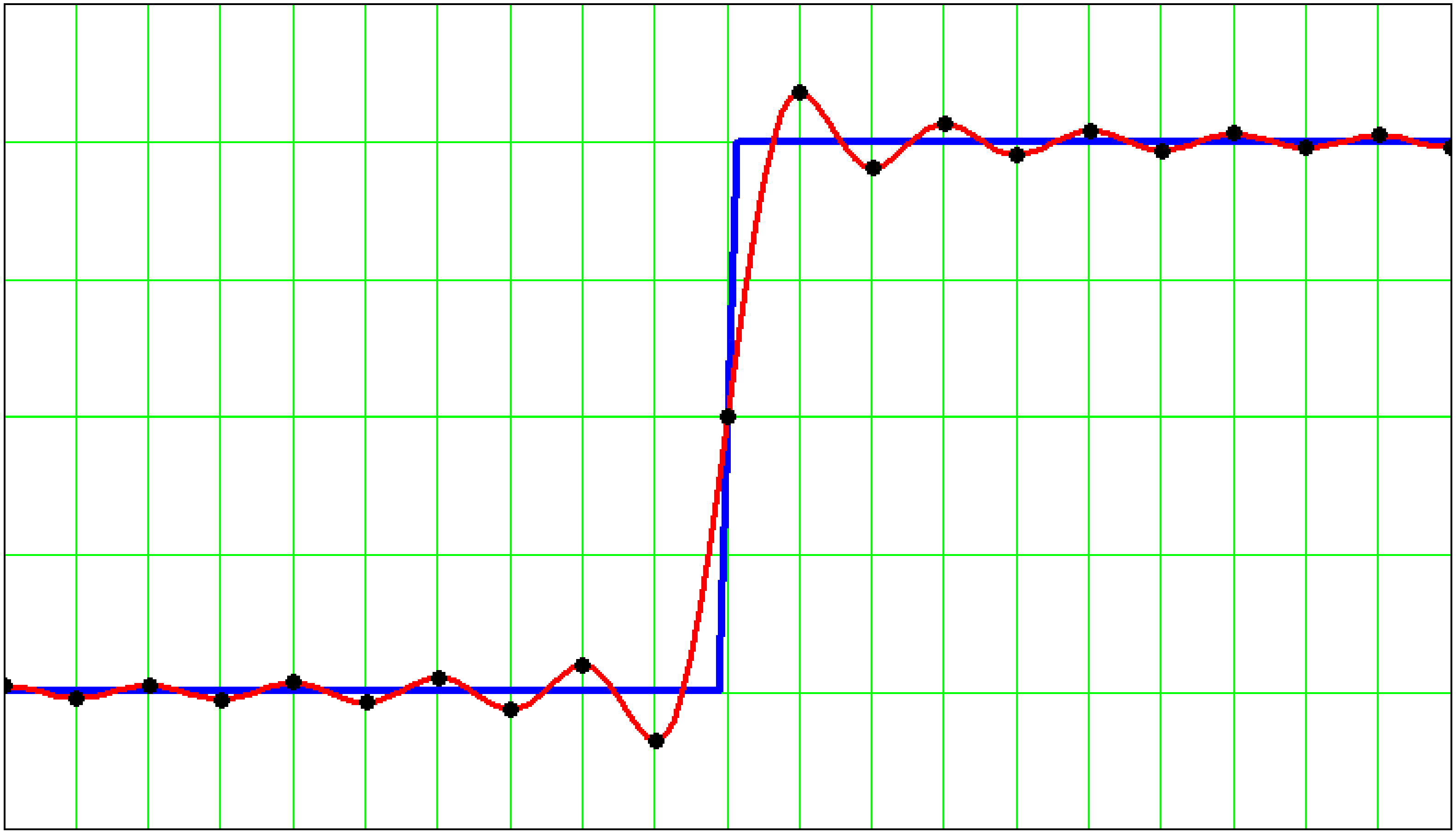
Digital Filters in AD/DA Converters

The TOA Cue Fallacy

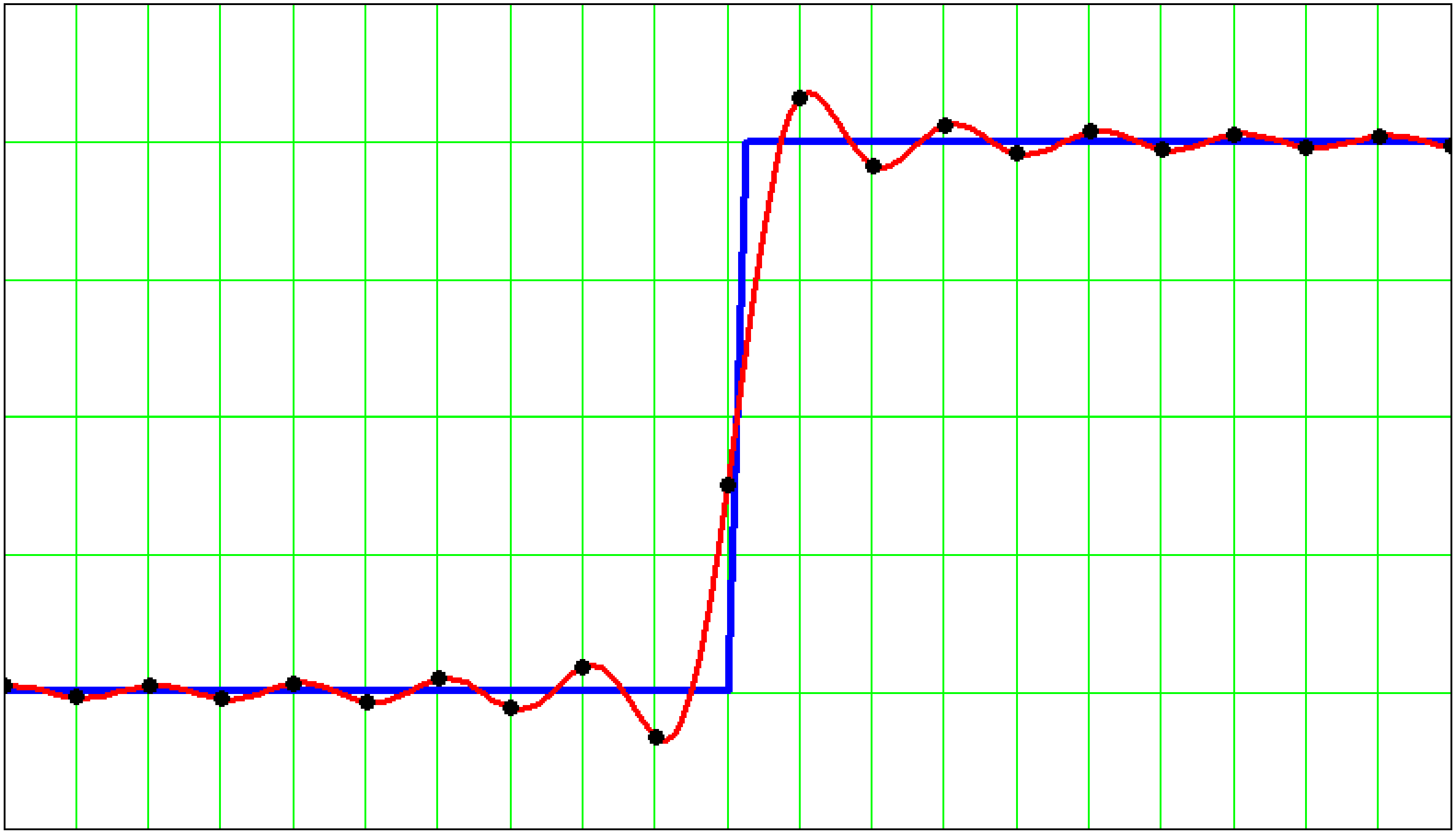
- “The ear can detect a 2us Time-Of-Arrival difference”
  - Correct! (0.2° lateral shift in stereo image)
- “So we need 500kHz sampling”
  - Uhhh not quite...

Digital Filters in AD/DA Converters

The Promise in Practice

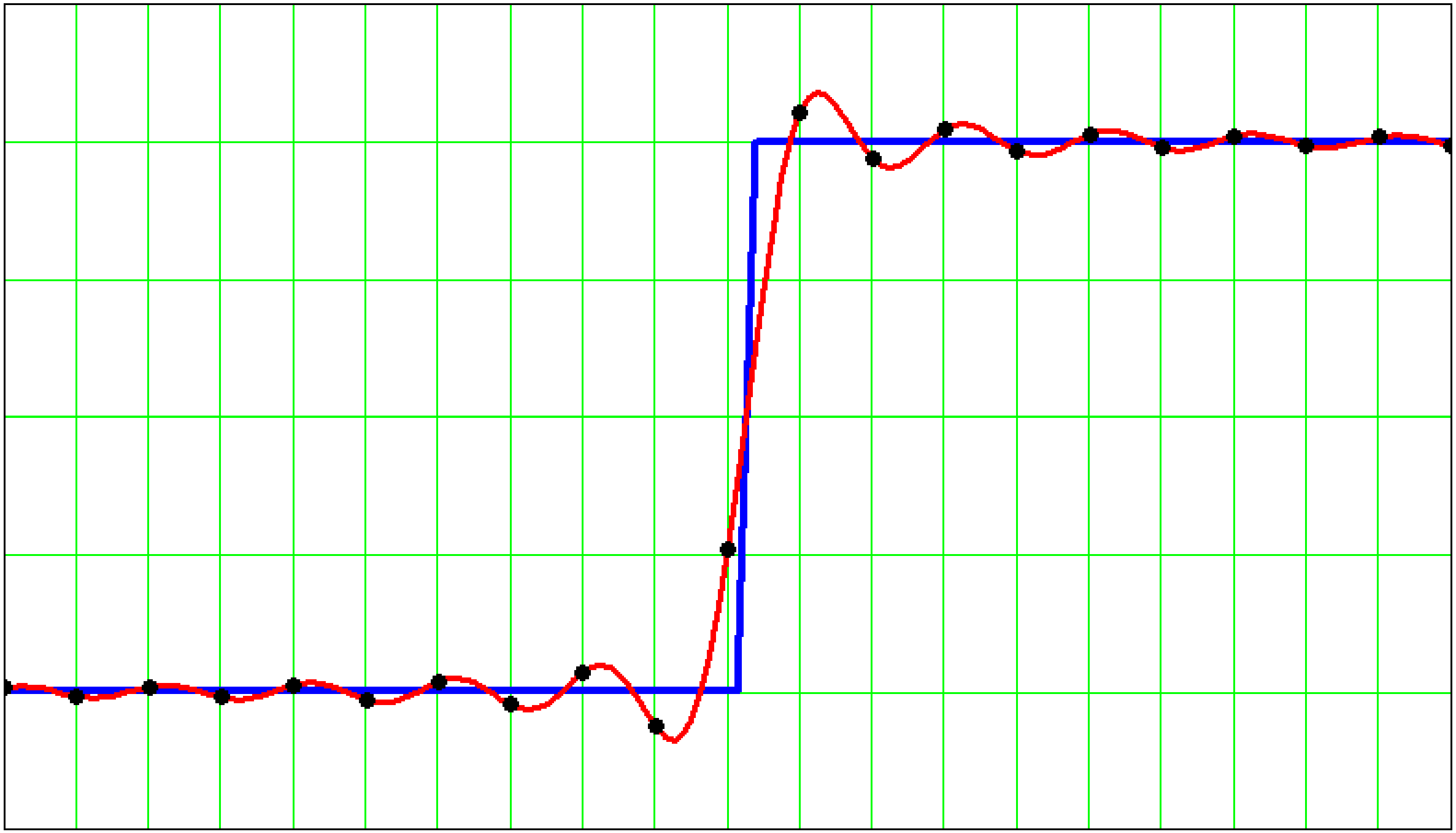


The Promise in Practice



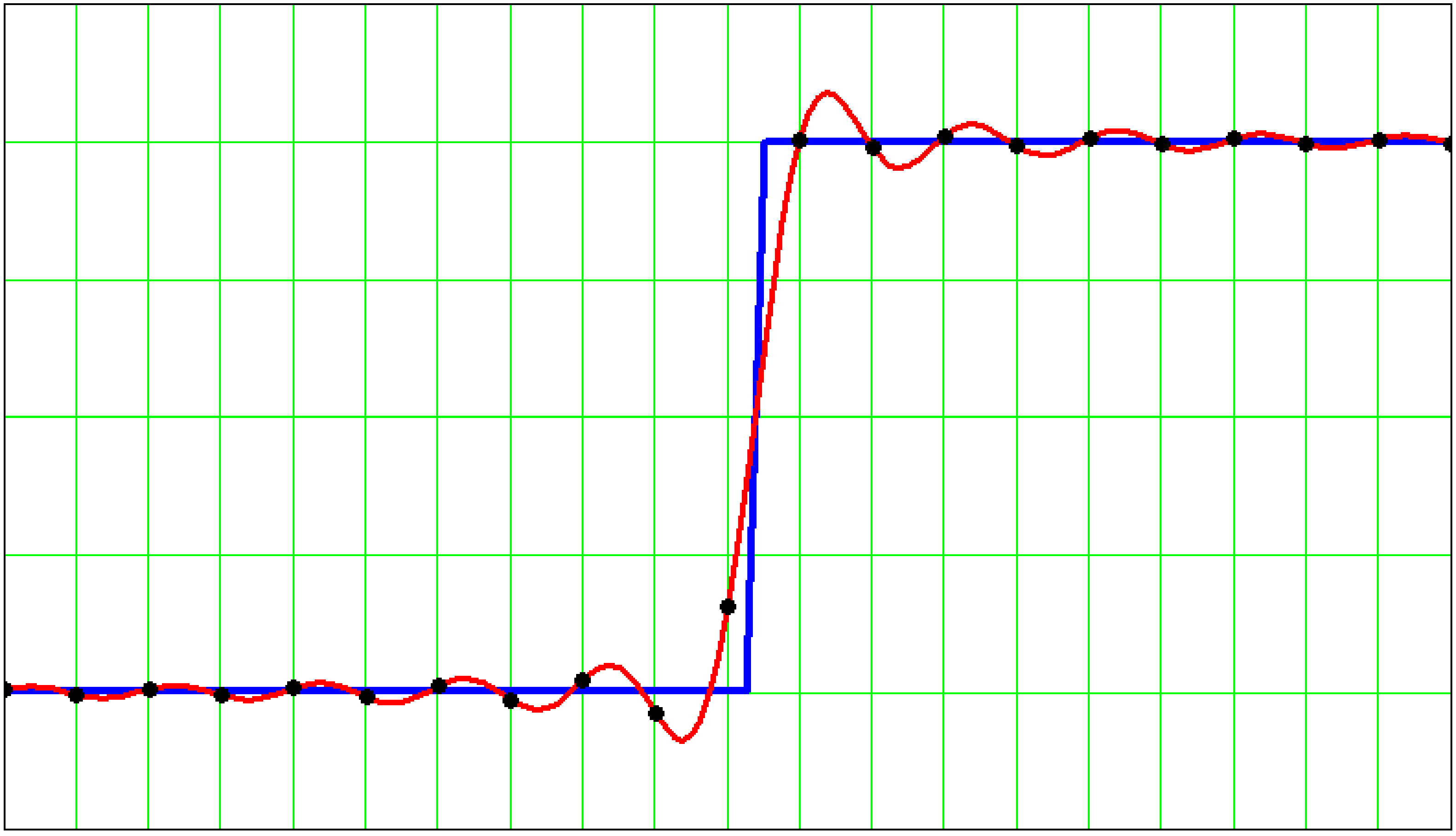
Digital Filters in AD/DA Converters

The Promise in Practice



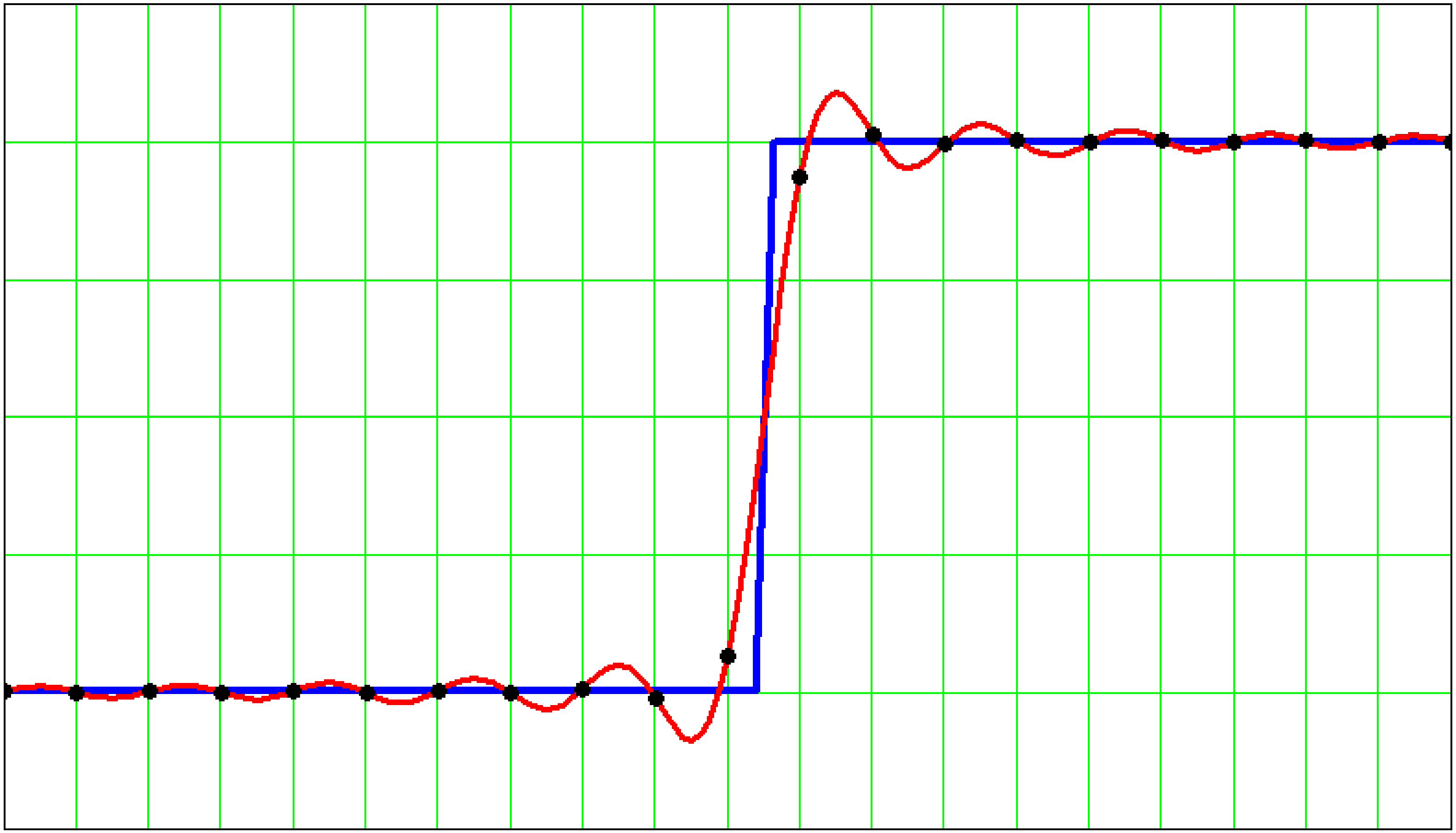
Digital Filters in AD/DA Converters

The Promise in Practice



Digital Filters in AD/DA Converters

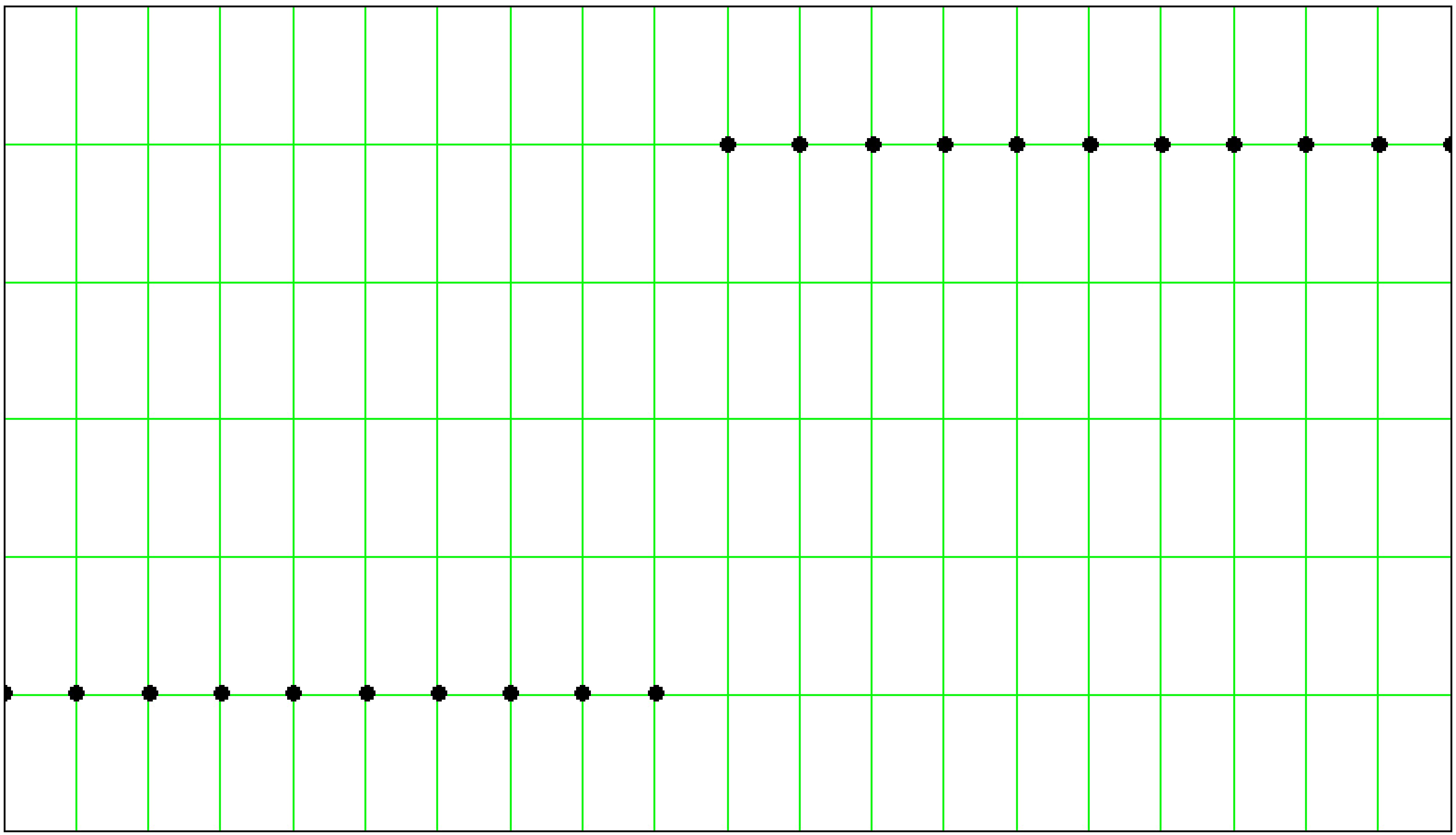
The Promise in Practice



Digital Filters in AD/DA Converters

The Nonoversampling Fallacy

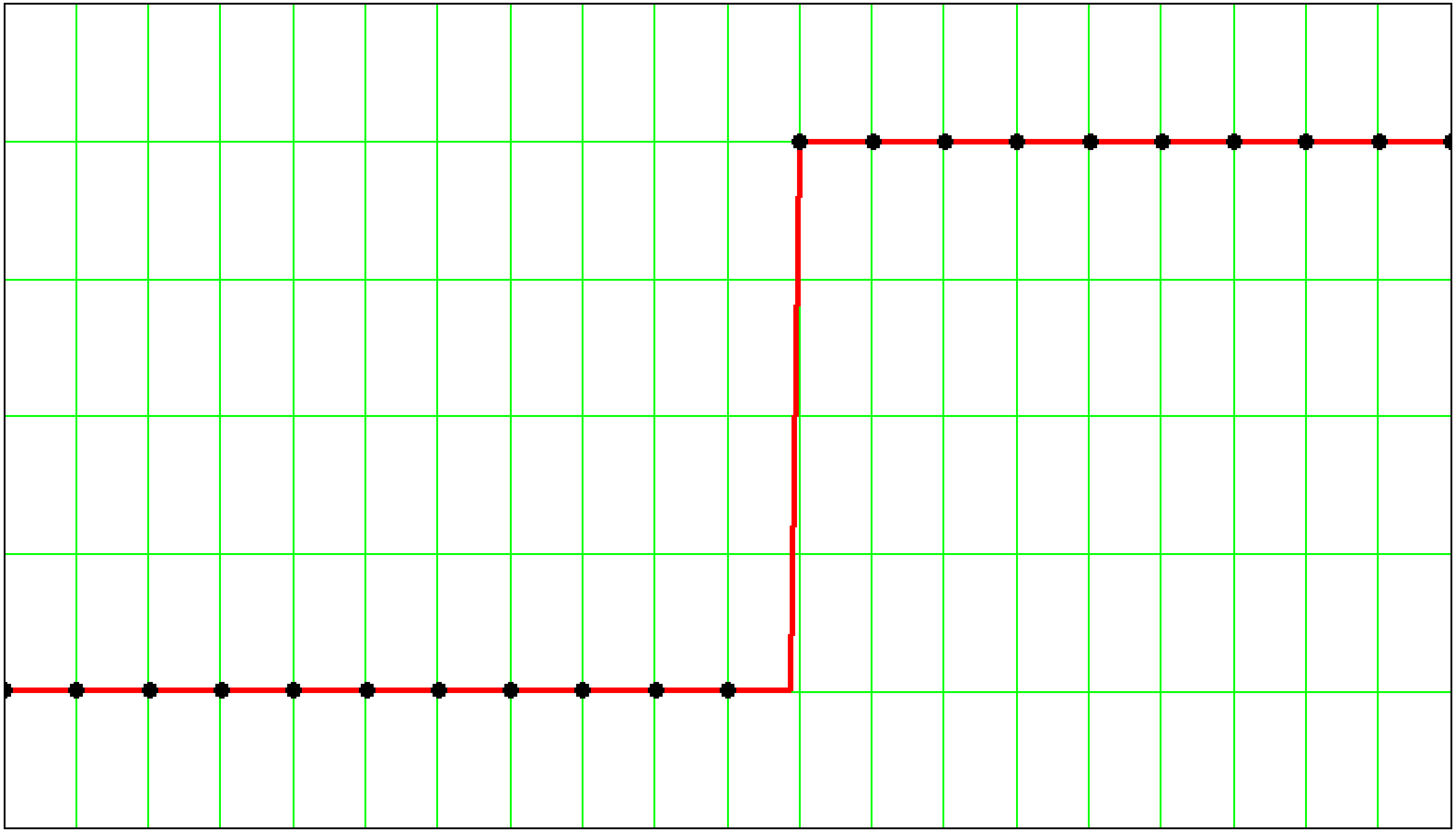
- “Digital Square Wave” test signal looks like this:



Digital Filters in AD/DA Converters

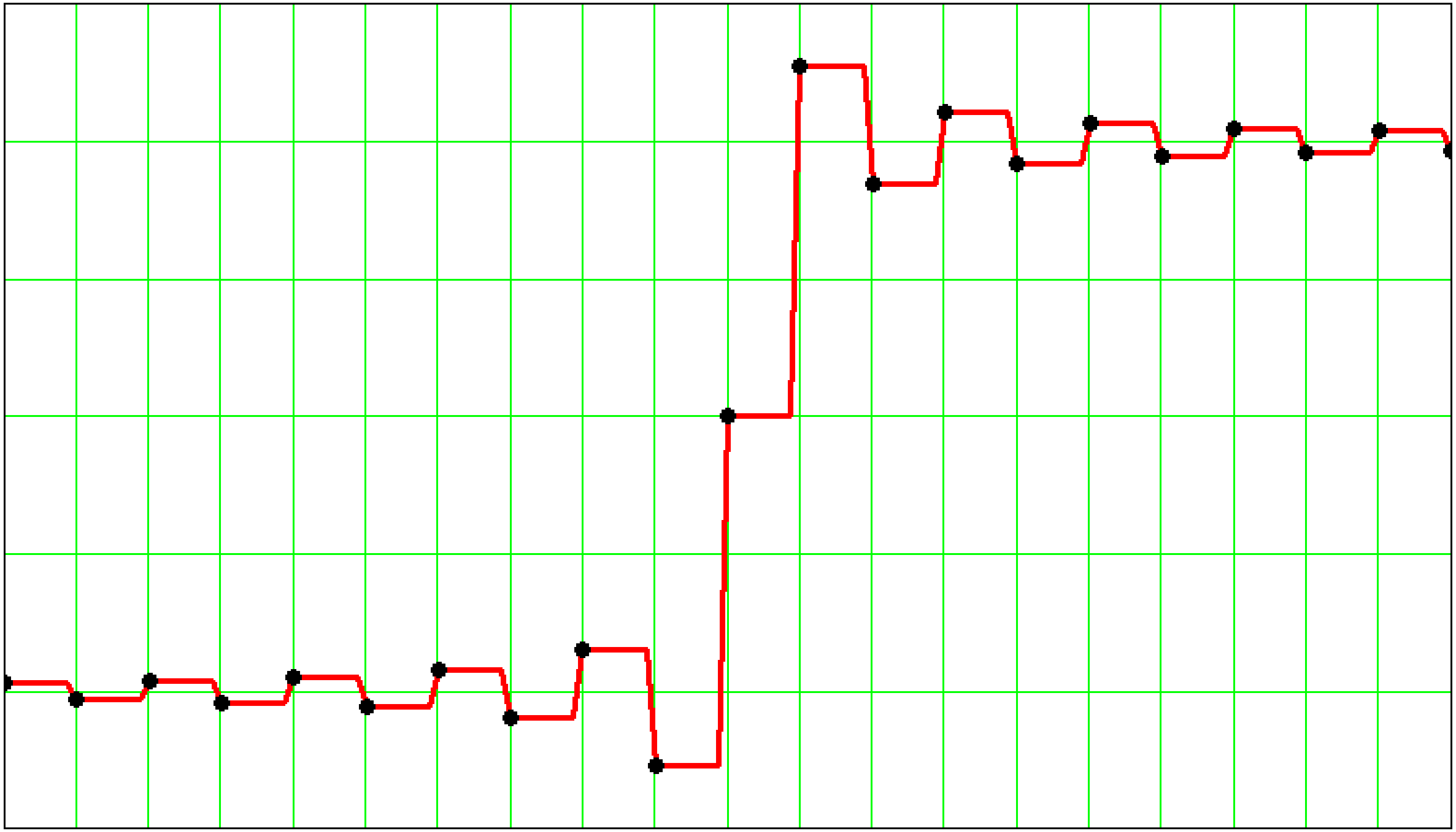
The Nonoversampling Fallacy

- After “NOS conversion” (=zero order hold) we get:



The Nonoversampling Fallacy

- Now insert a half-sample delay:



Digital Filters in AD/DA Converters

The Nonoversampling Fallacy

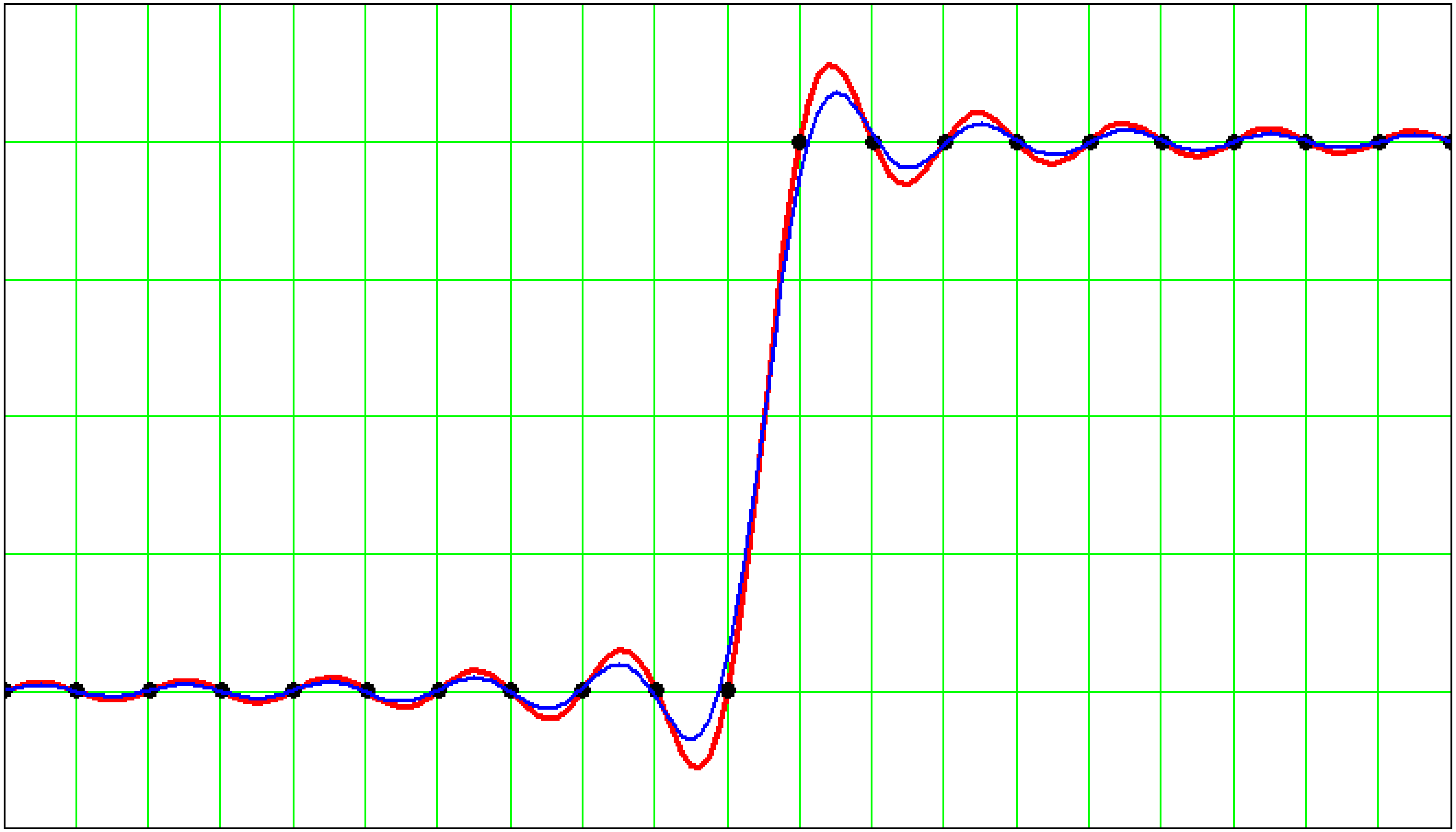
- Impulse Response becomes time-variant
- Fallacy was facilitated by the “Digital Squarewave” signals from test kit and test discs.

Digital Filters in AD/DA Converters

- The “Digital Step Function” reconstructs like this:

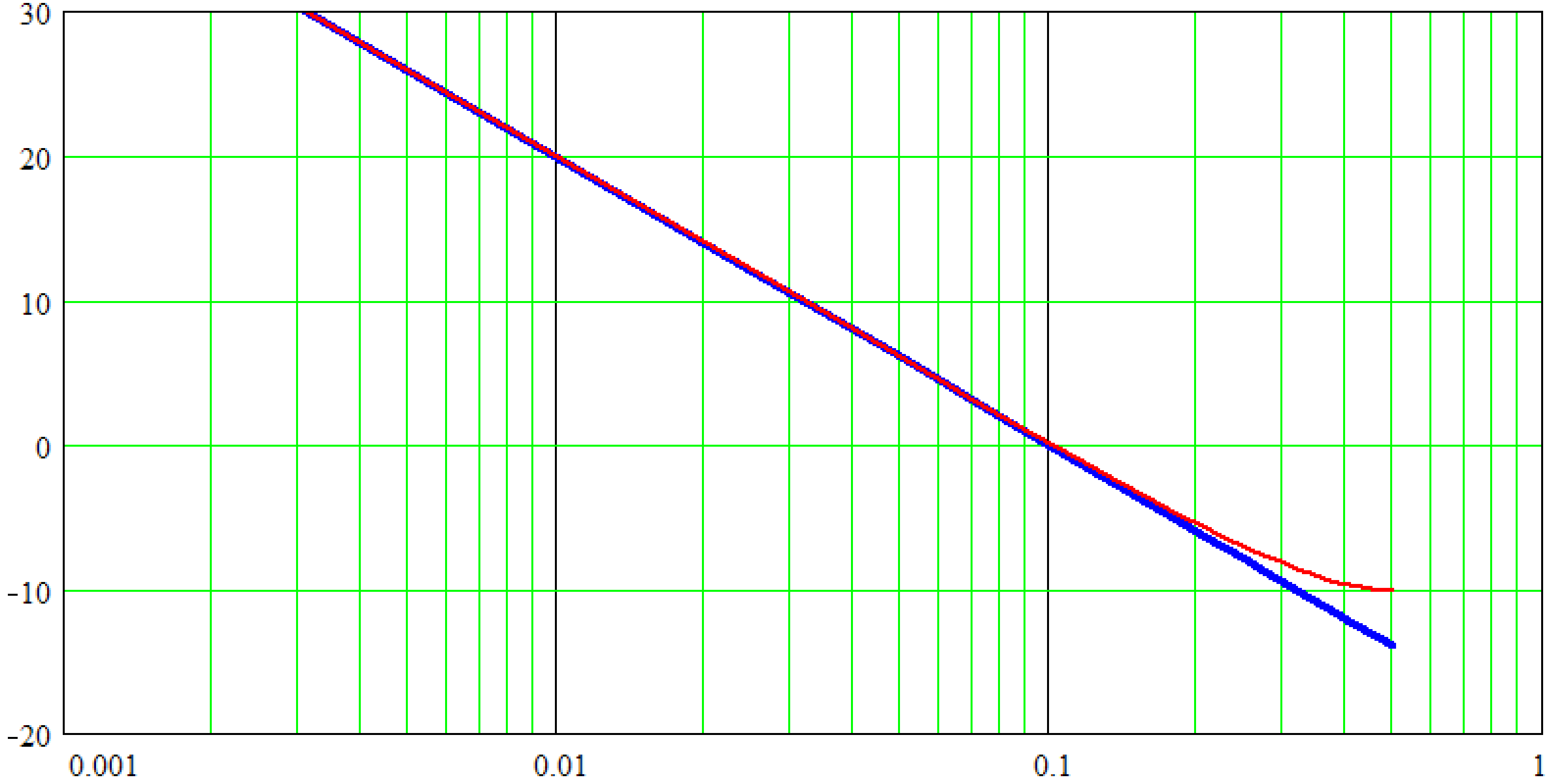
Digital Filters in AD/DA Converters

- Contrast with an actual band-limited step function



Digital Filters in AD/DA Converters

- Compare the spectra



Dear test equipment designers: please provide a “true square wave” of arbitrary frequency

NOS Rundown

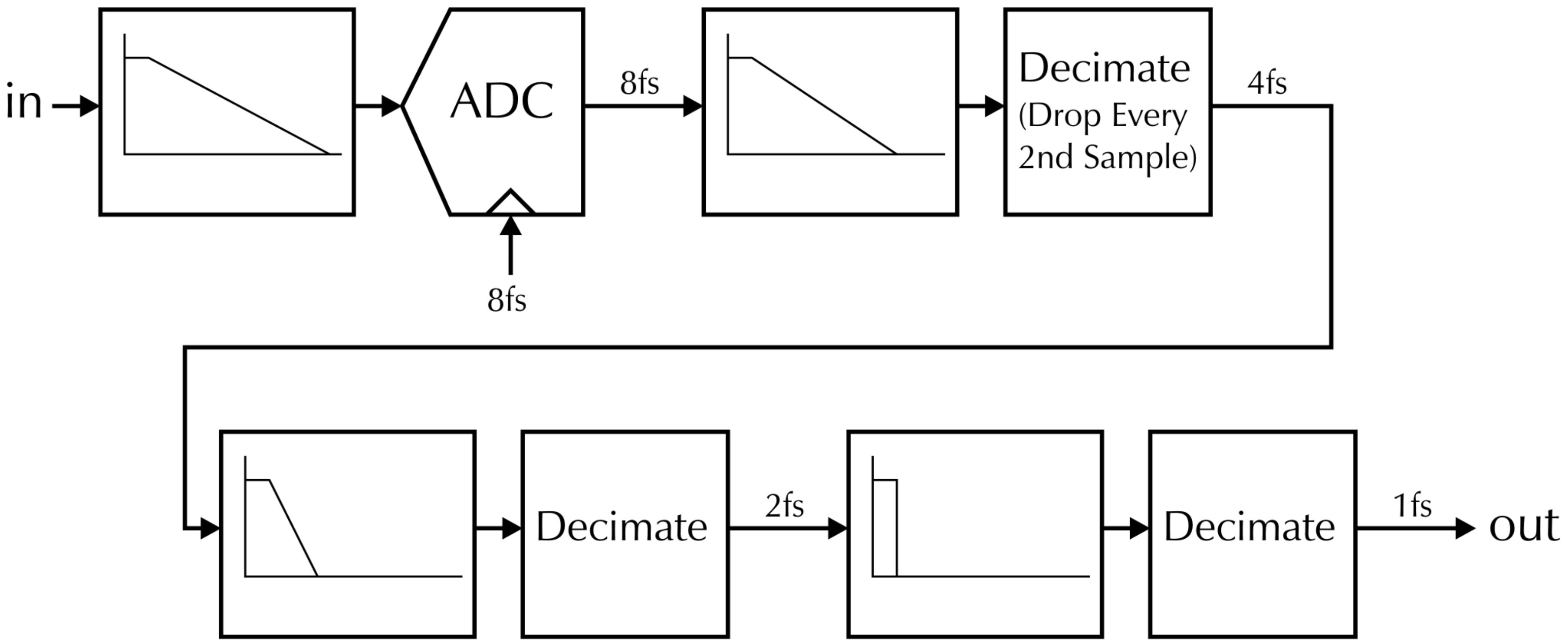
- NOS DAC may sound OK
  - We really don't notice much beyond 20k...

- NOS DAC sometimes sounds better than *same* DAC with digital filter
  - DAC in these experiments is invariably ladder type
  - Glitch contribution goes up with sampling rate
  - Latch signal passes through filter chip (increased clock jitter)

- None relate to impulse response

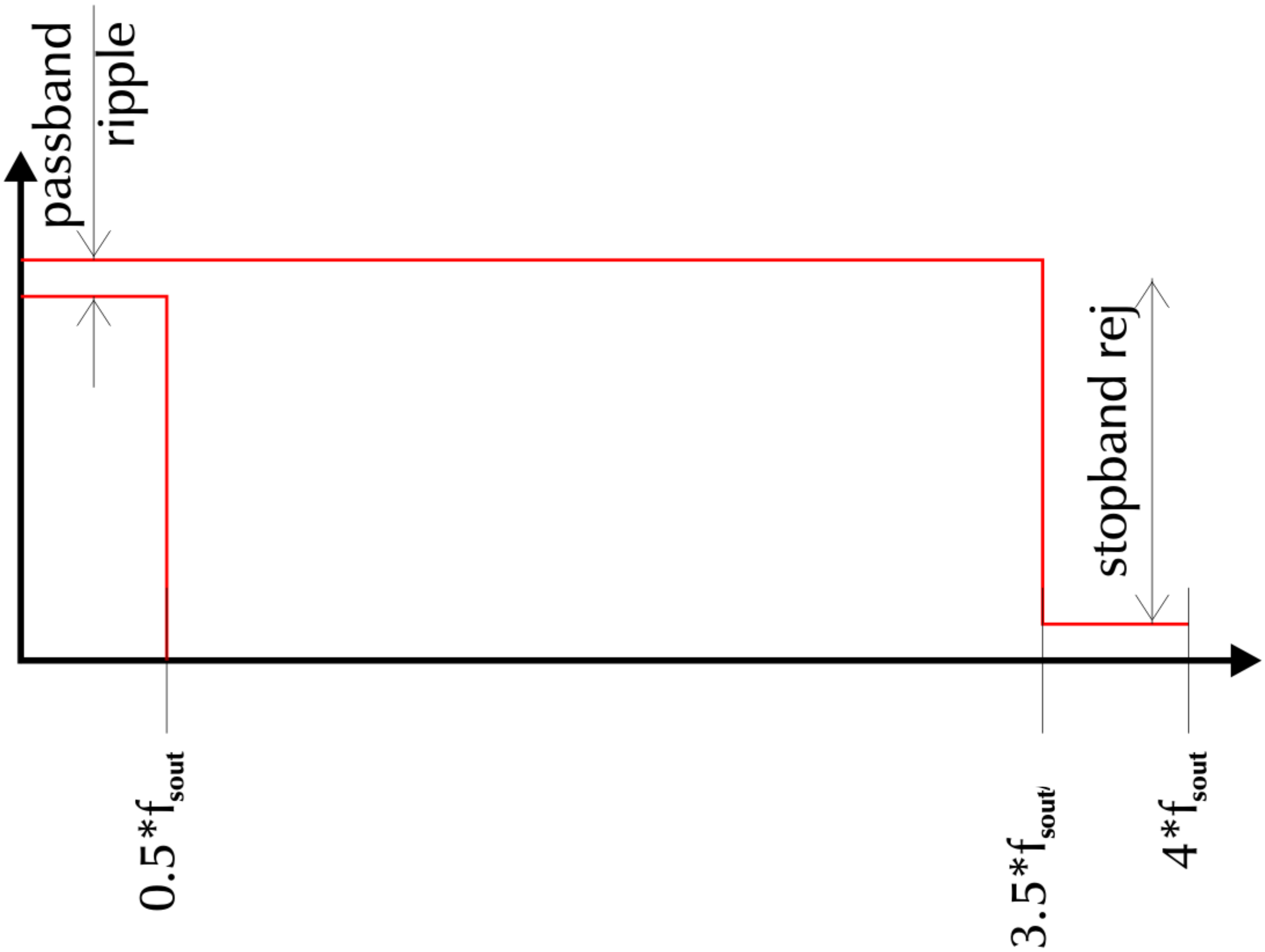
Digital Filters in AD/DA Converters

- Antialias filtering in contemporary ADC's is mostly done digitally, in a "decimation chain"



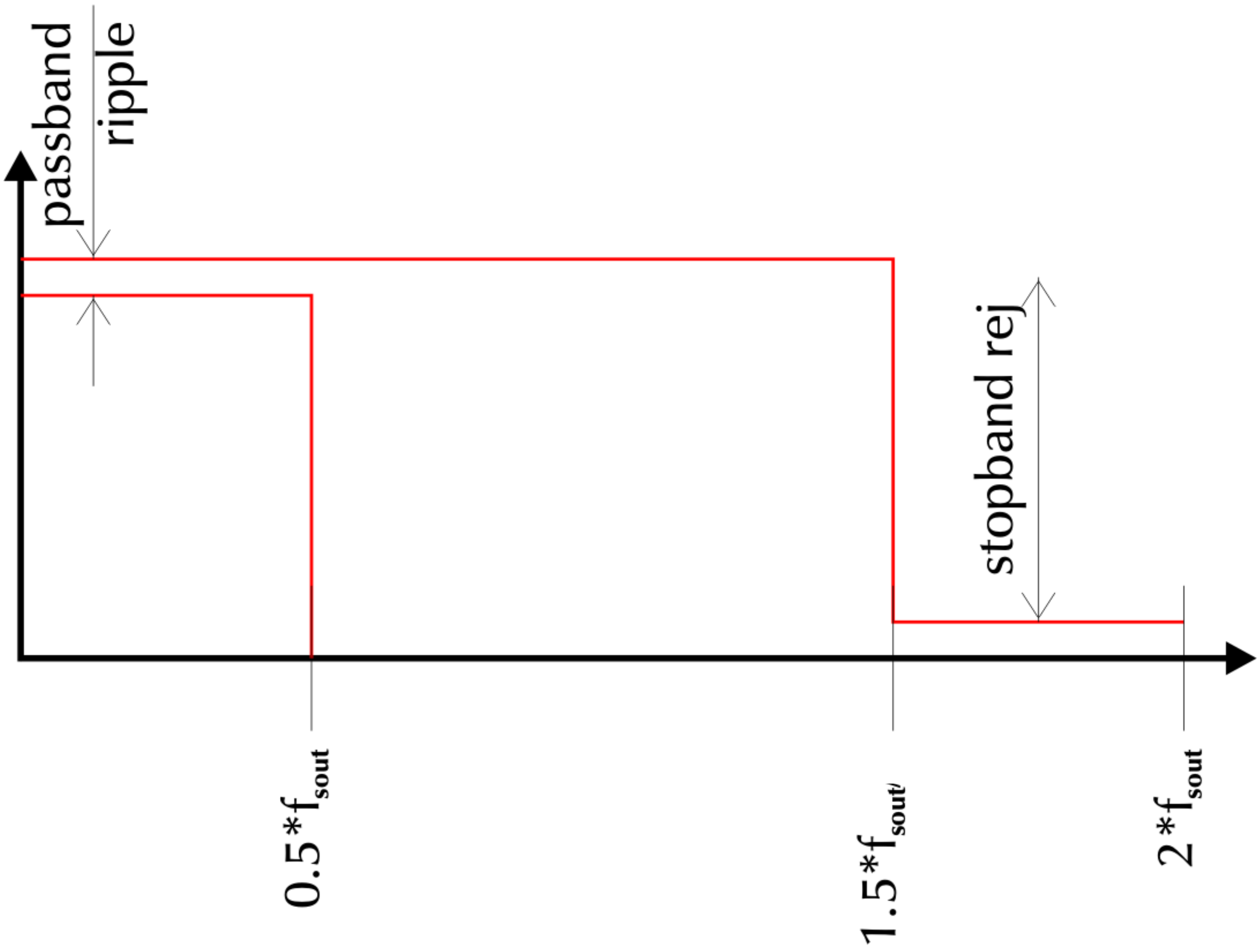
Digital Filters in AD/DA Converters

Gabarith for 8fs -> 4fs filter stage



Digital Filters in AD/DA Converters

Gabarith for 4fs -> 2fs filter stage



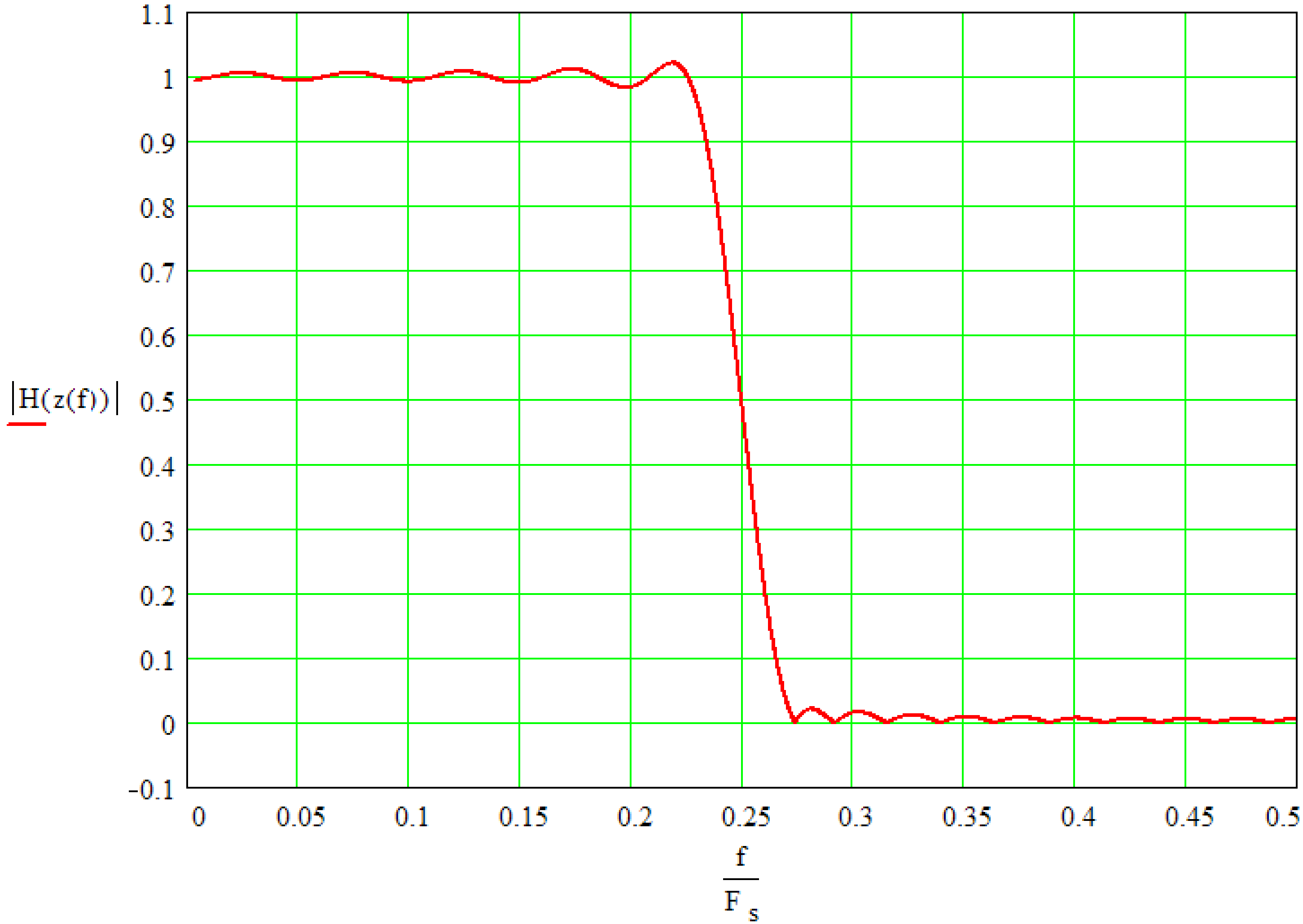
Digital Filters in AD/DA Converters

A perfect candidate: The Half-Band filter

- Magnitude response is chosen symmetrical round 0.25fs and 0.5.
  - Stop band = 0.5fs - pass band
  - Stop band rejection = stop band ripple

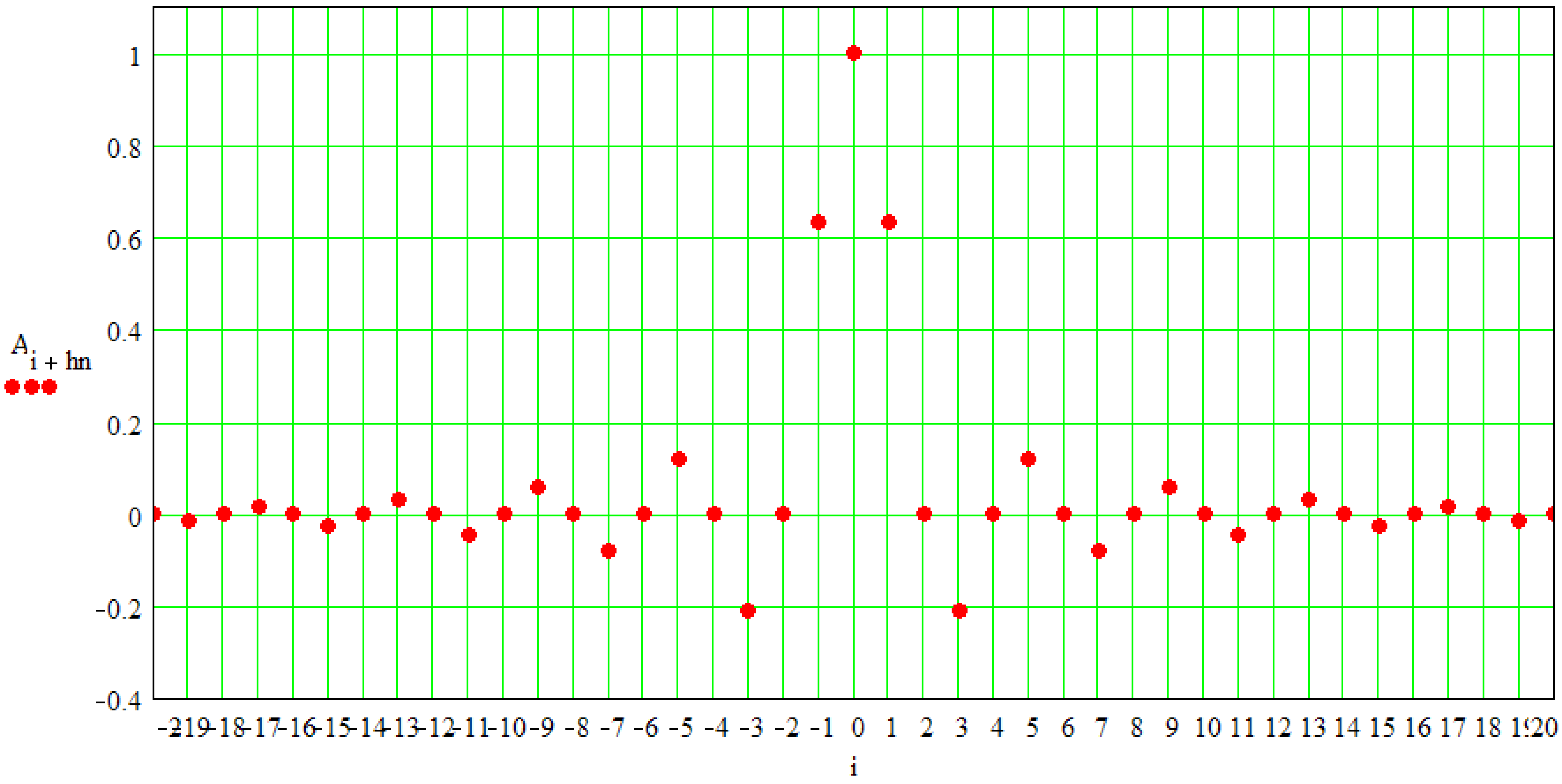
Digital Filters in AD/DA Converters

Half-Band filter, Magnitude Response



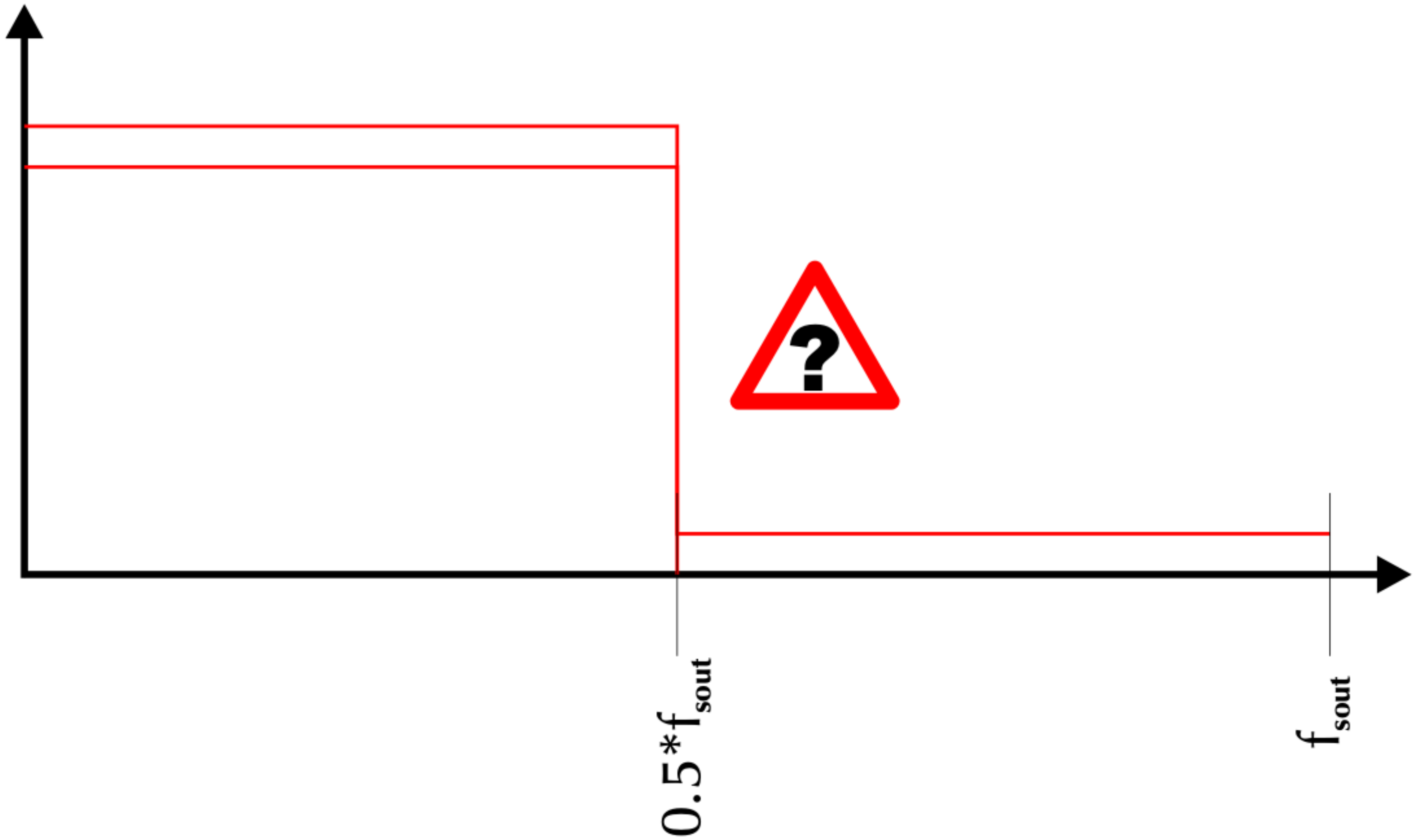
Digital Filters in AD/DA Converters

Half-Band filter, coefficients



Digital Filters: Design Compromises

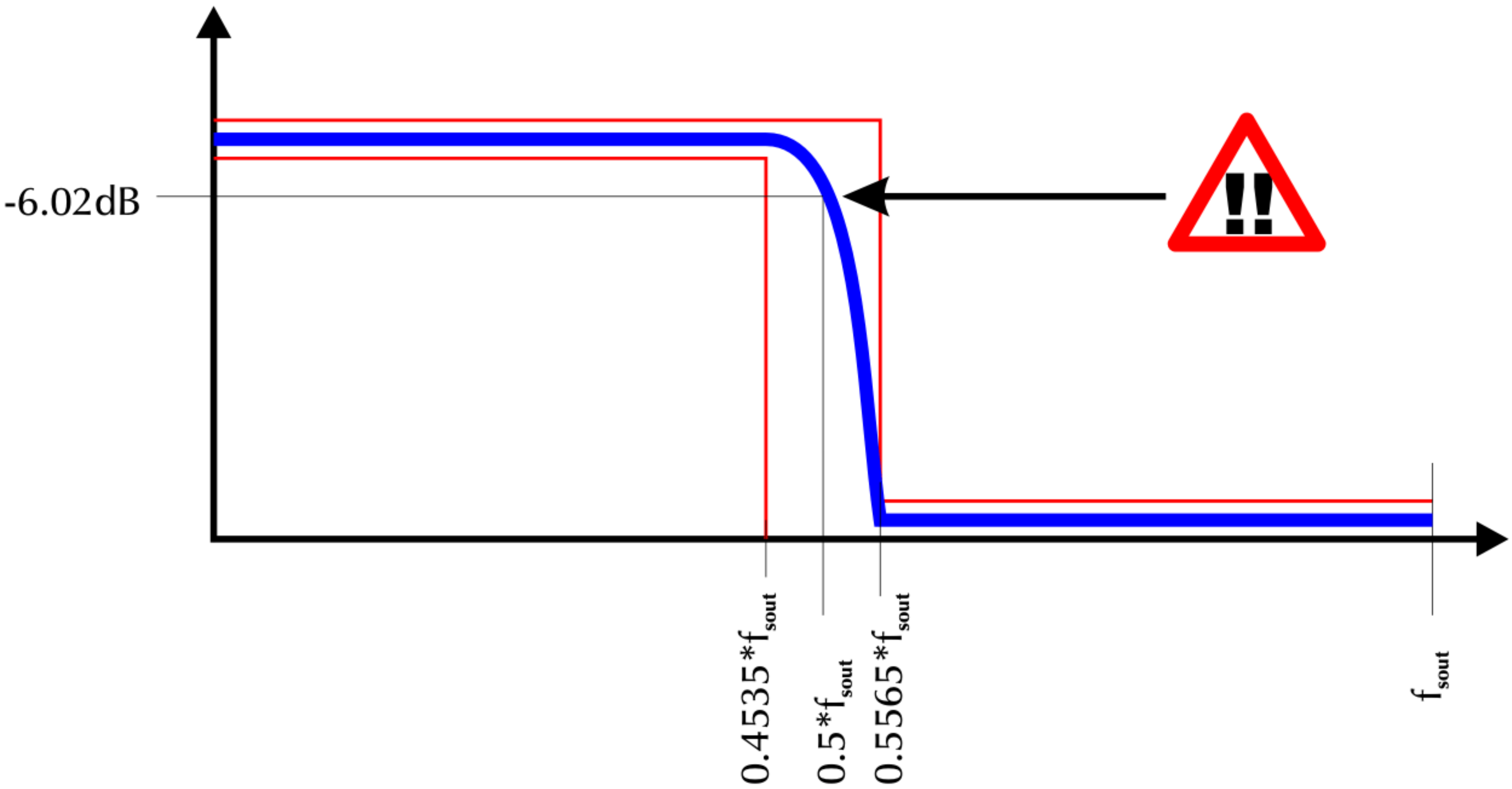
Gabarith for 2fs -> 1fs filter stage.



Oops.

Digital Filters: Design Compromises

Typical final stage in commercial converters



Cut & dried breach of Nyquist criterion!

Digital Filters: Design Compromises

0.4535*44.100kHz=20.000kHz

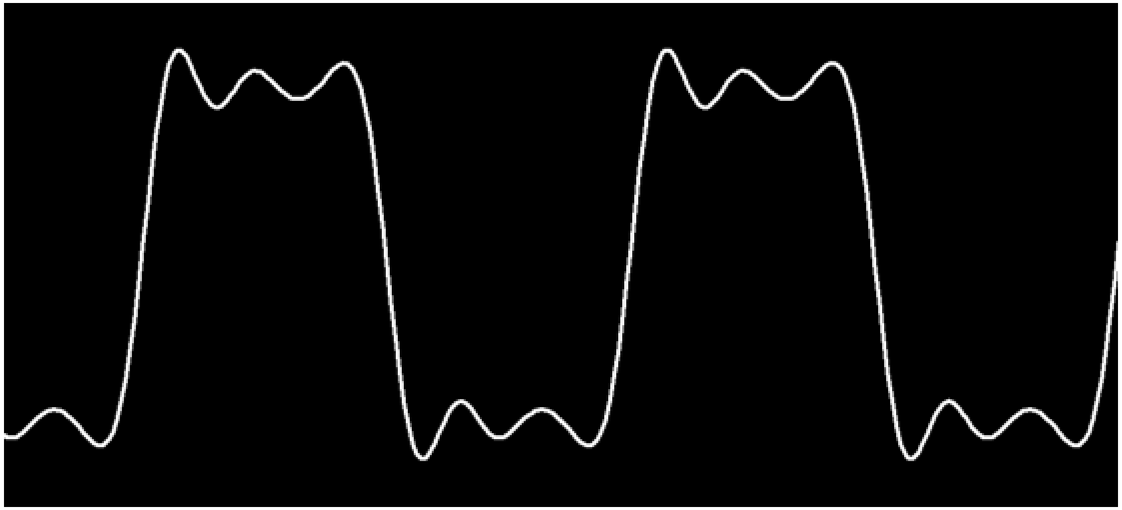
Digital Filters: Design Compromises

Result

- Band between 0.4535fs and 0.5565fs suffers aliasing.
- Only 12dB of attenuation at $f_s/2$. Signals with significant energy near $f_s/2$ are worst affected.

- Next slide: demonstration: f_s=44.1k. Square wave of f≈3150Hz is fed into ADC. 7th harmonic aliases.

Digital Filters: Design Compromises



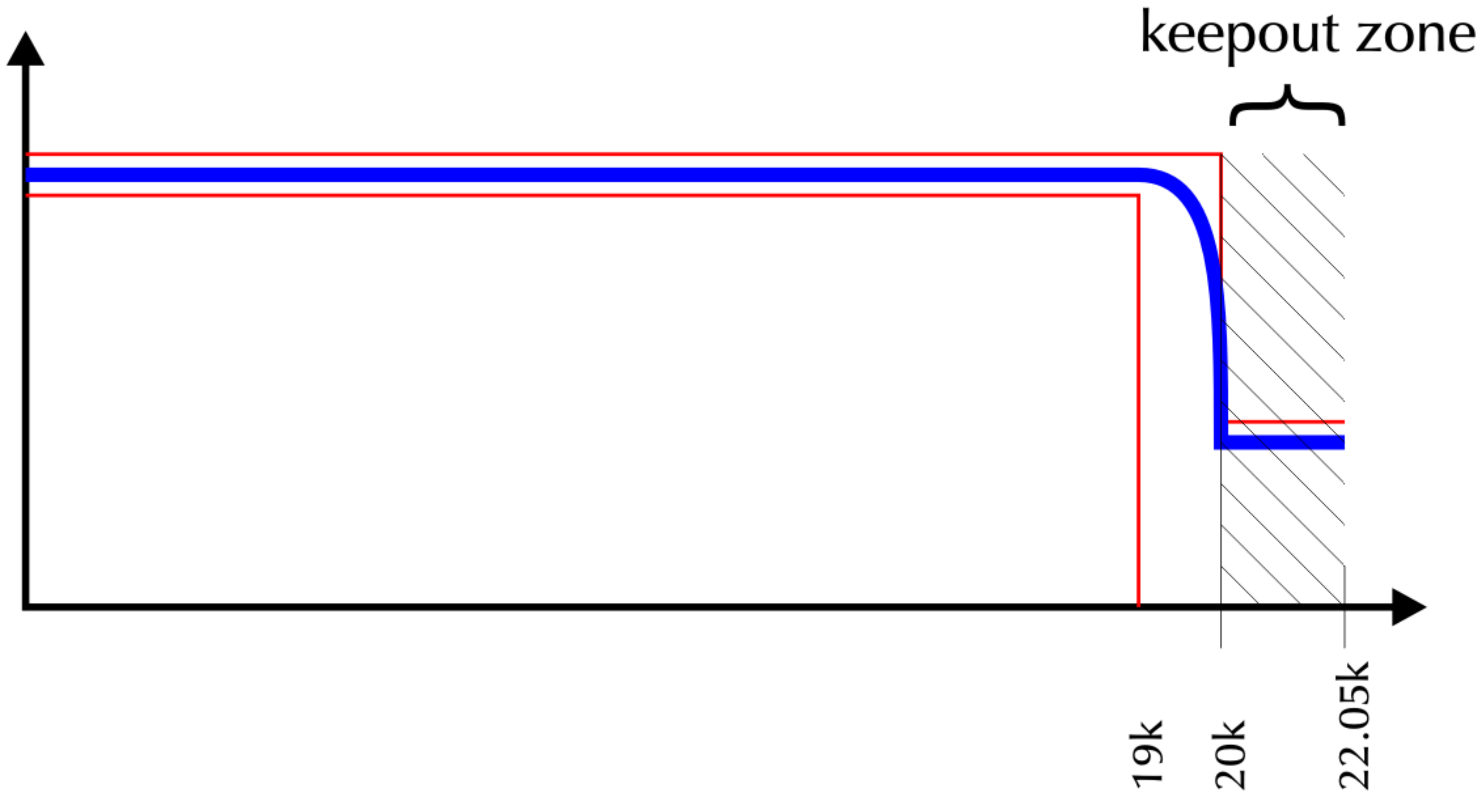
To human ears:

- TOA cues are affected for signals with significant HF.
  - Sibilants in choral music, wind and string instruments smear across the whole stereo image.
- Nearly no impact for panpot stereo.
  - Alias components are in phase across channels

Digital Filters: Design Compromises

How to Salvage a Burnt Steak

- Cut off the blackened bits.



Digital Filters: Design Compromises

Applicability of Steak-Salvaging filter.

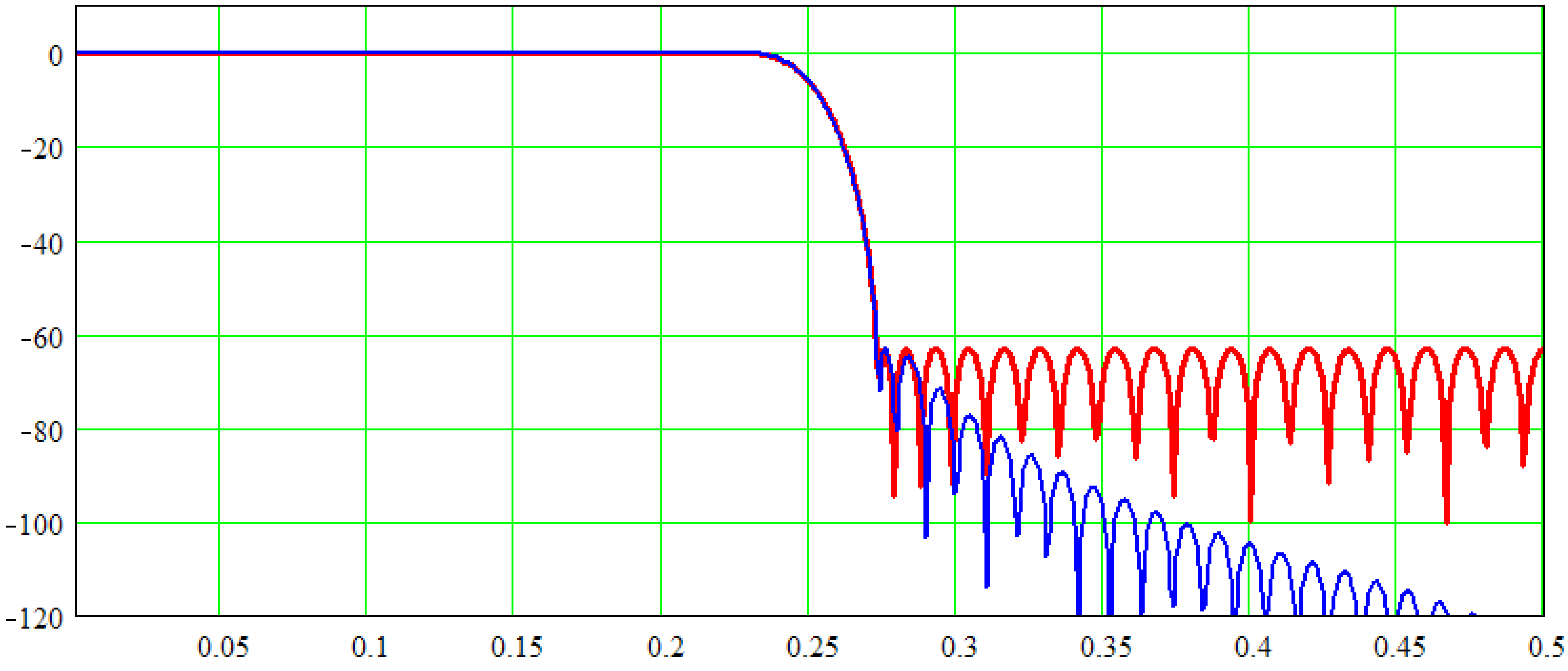
- Use once in the entire record-replay chain
  - The rest of the chain may keep using halfbands.

- Check by ear
  - The 44.1kHz version has a sonic signature.
  - Weigh against improved imaging.

Digital Filters: Design Compromises

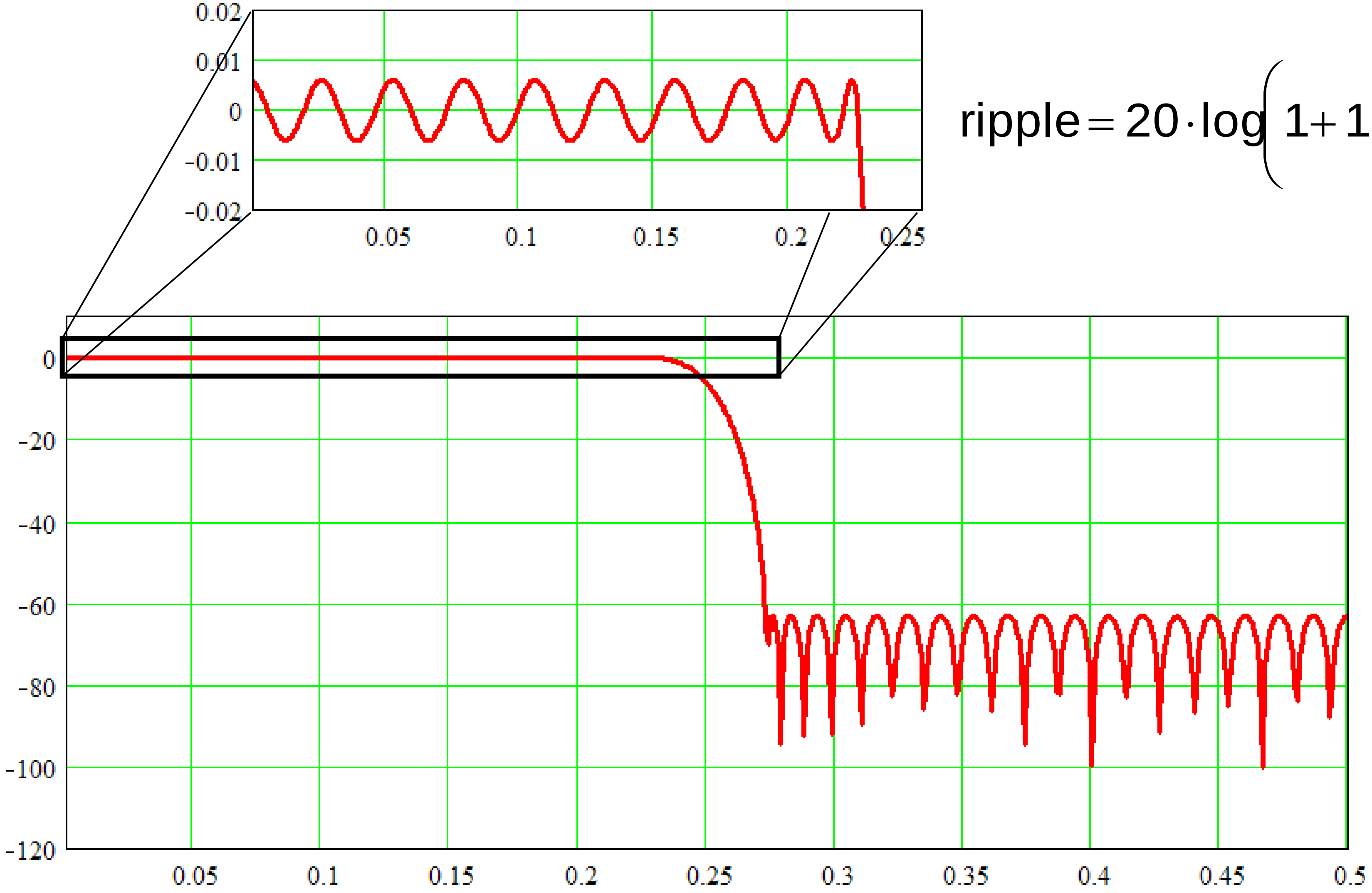
The Equiripple Filter

- “Just Enough” attenuation = minimum number of coefficients.
- Windowed Sinc filters roll off further inside the stop band. Unnecessary attenuation increases length.
- Example: Equiripple, 75 taps. Windowed, 95 taps.



Digital Filters: Design Compromises

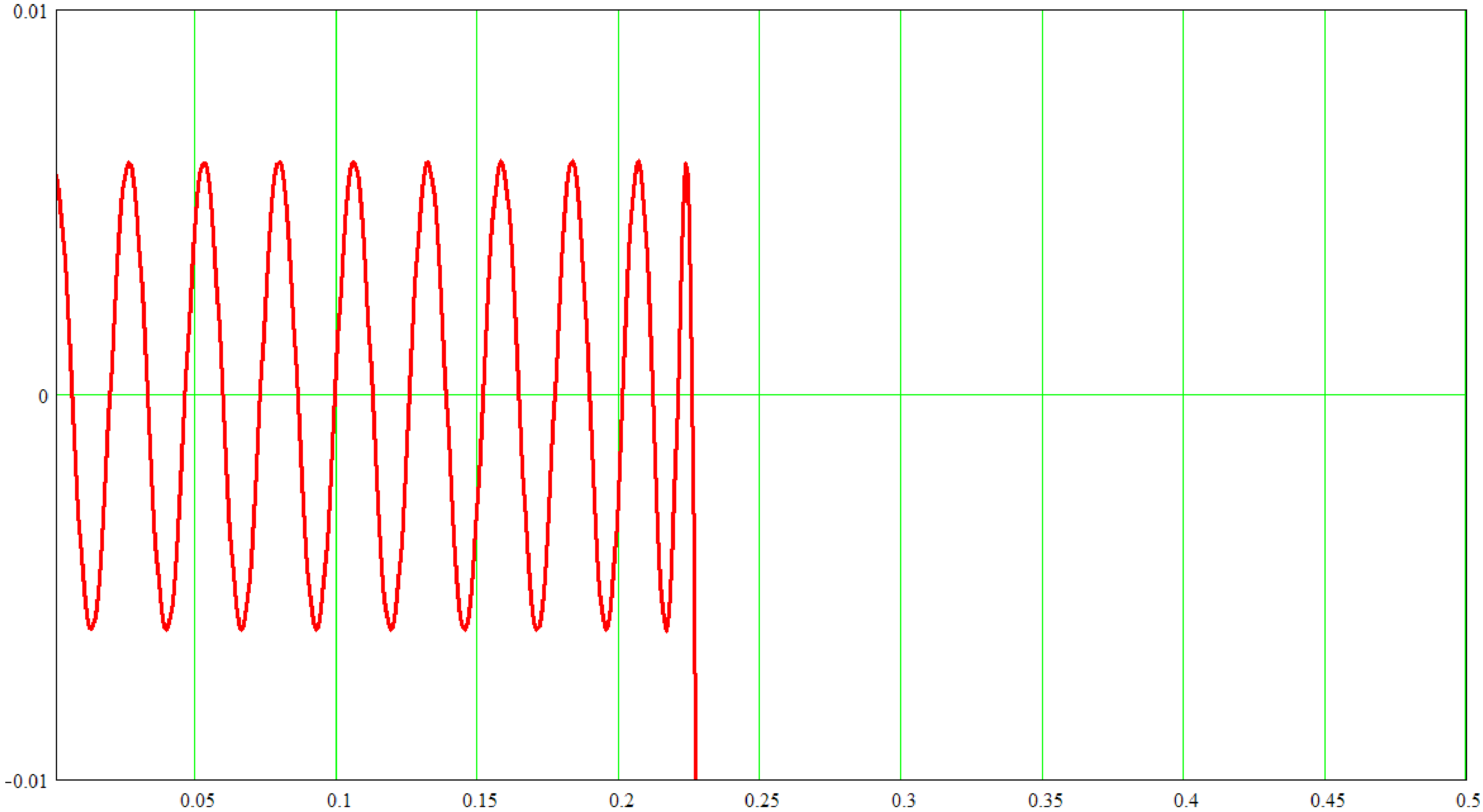
In a halfband filter, ripple and attenuation are linked



$$\text{ripple} = 20 \cdot \log\left(1 + 10^{-\frac{\text{atten}}{20}}\right)$$

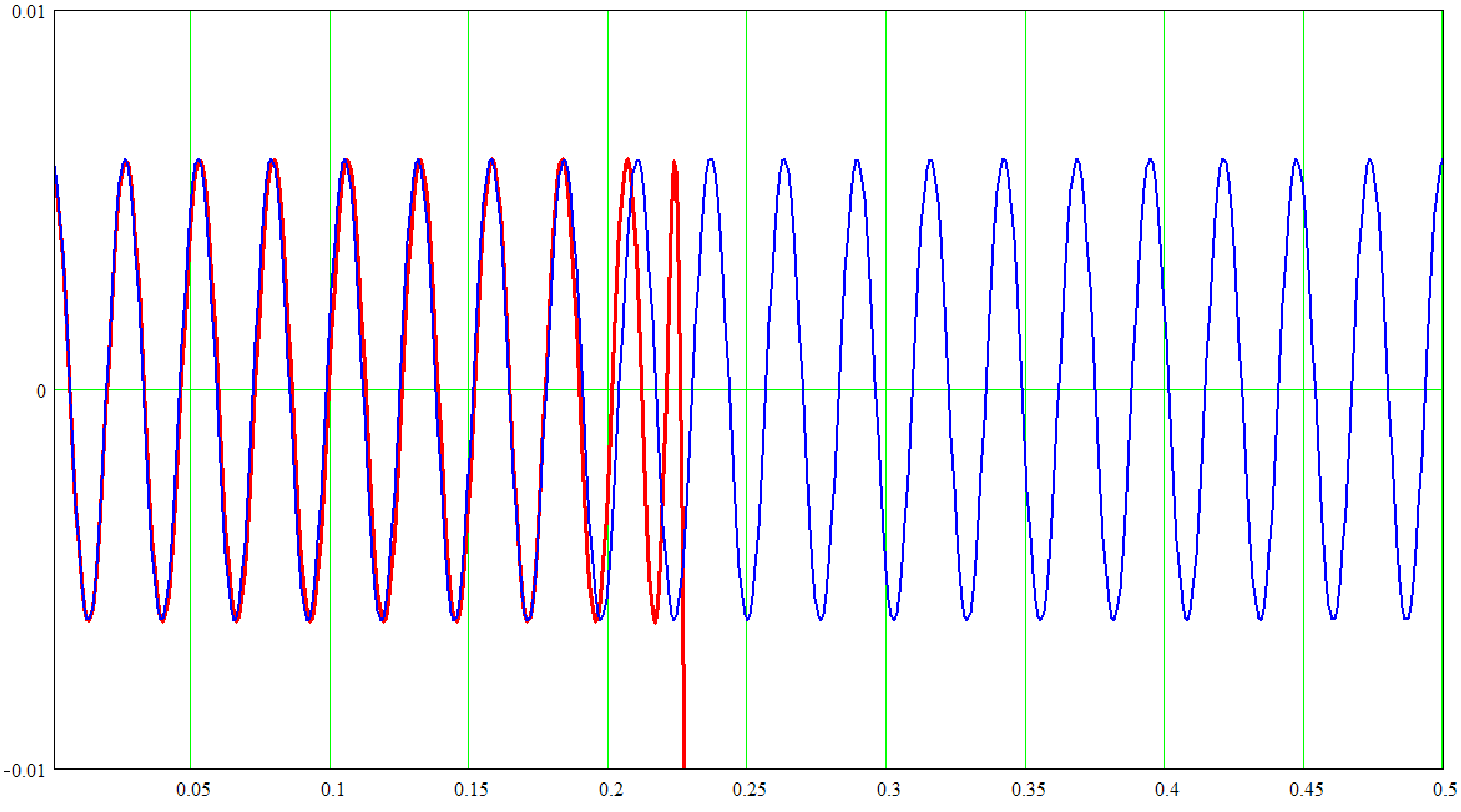
Digital Filters: Design Compromises

Ripples are equal in amplitude and nearly equally spaced. Spacing ≈ 2/(#taps+1)



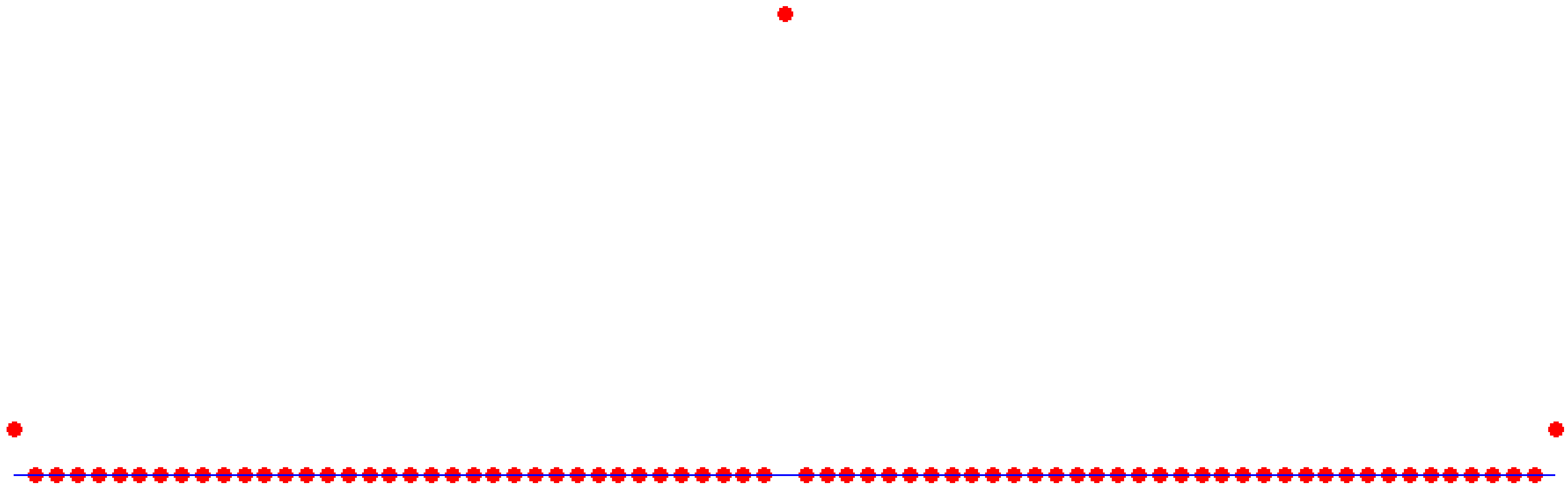
Digital Filters: Design Compromises

Let's define another linear-phase filter with nearly the same in-band response



Digital Filters: Design Compromises

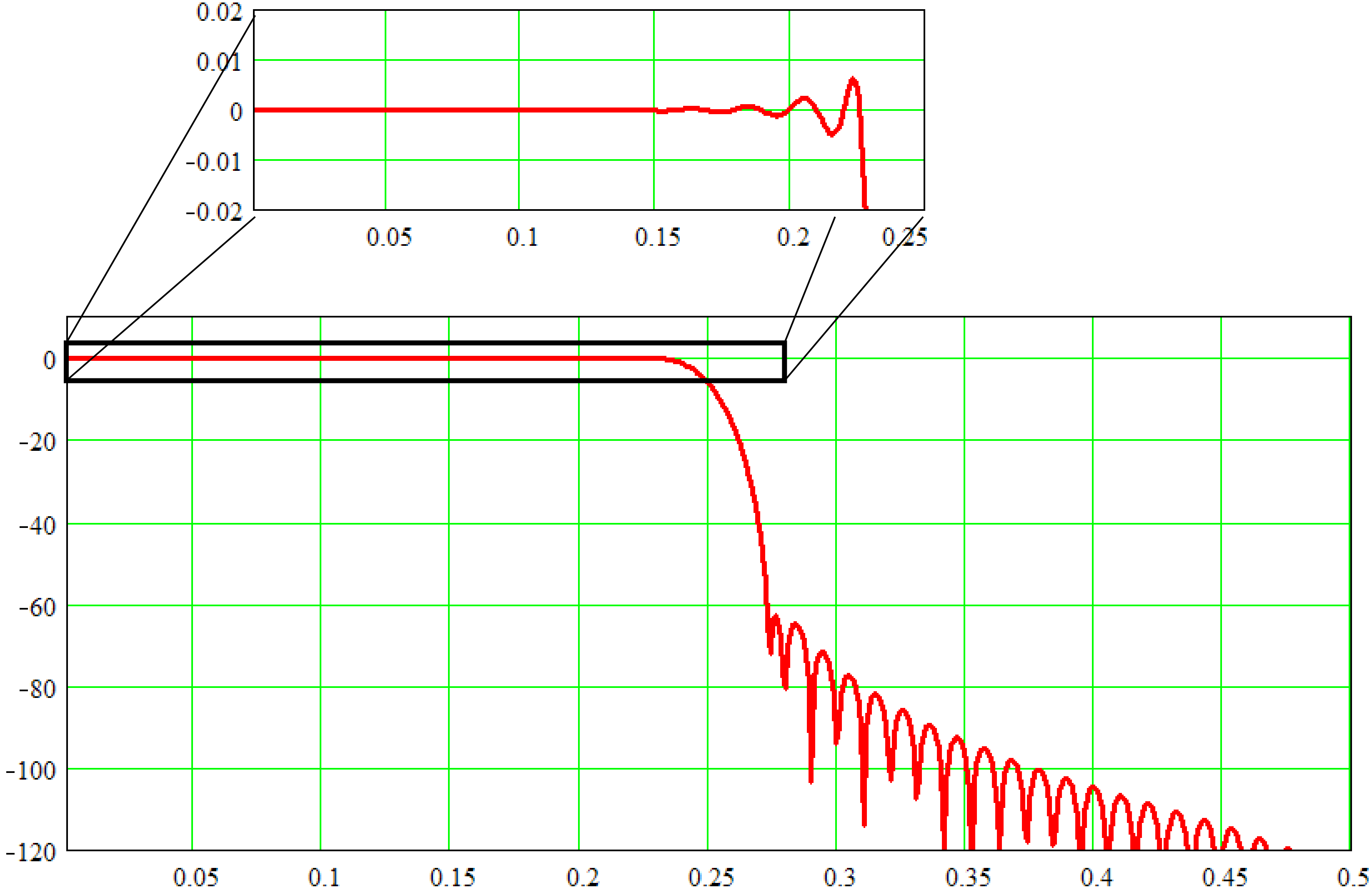
Impulse response of that filter (exaggerated):



- Constant in-band ripple equates to echos at the ends of the filter.
- Amplitude of echos = stop band attenuation – 6dB
- Post-echo is certainly masked. Pre-echo possibly not.

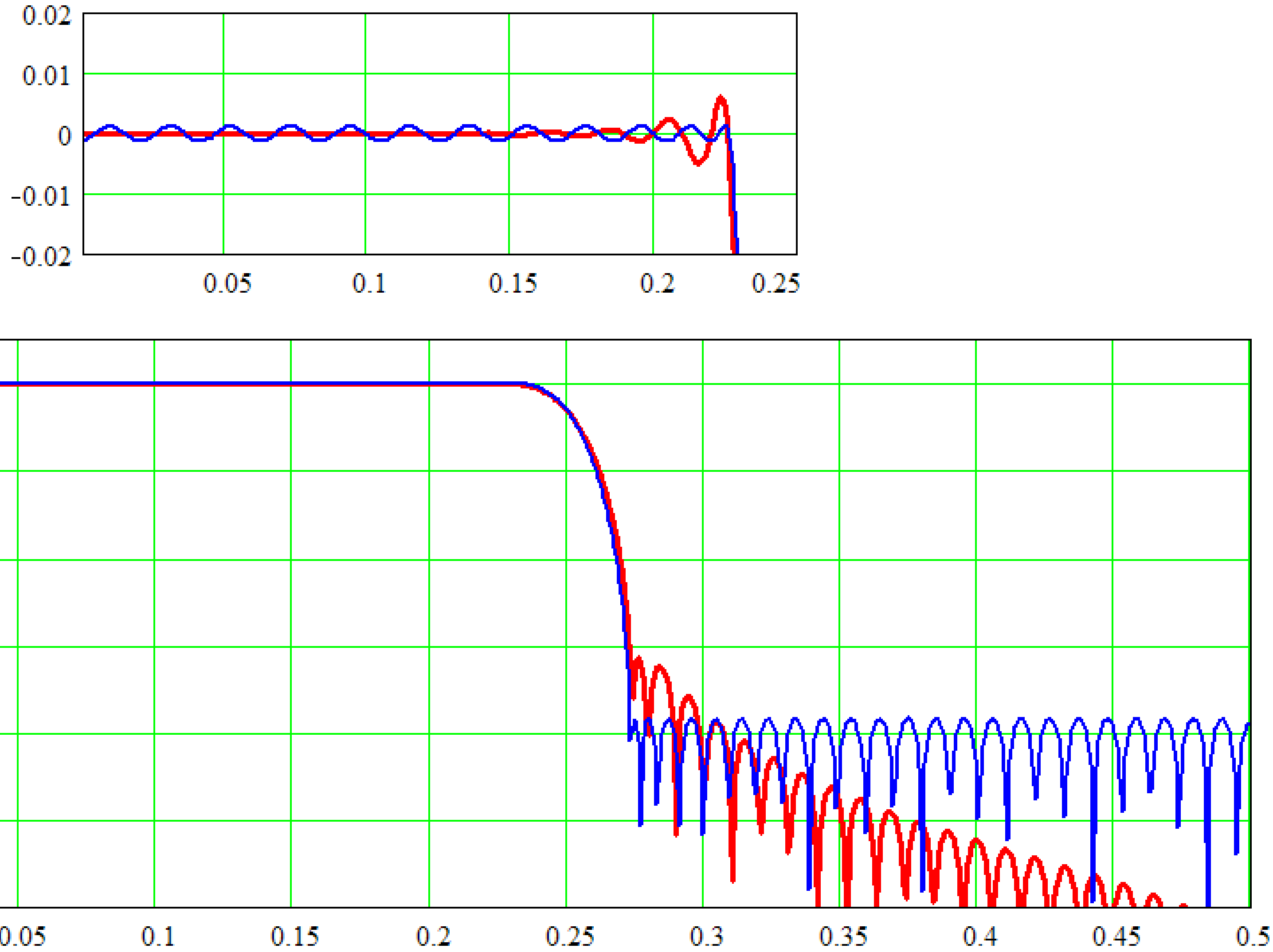
Digital Filters: Design Compromises

Close-up of ripple of windowed sinc filter



Digital Filters: Design Compromises

Compare 2 halfband filters at 95 taps



Digital Filters: Design Compromises

Import on “digiphobia”

- Classic argument against digital: “pre-ringing”
  - Little serious evidence of audibility of pre-ringing outside the audio band exists.
  - Looks like a red herring

- 2 common impementation problems were identified
  - Aliasing and Pre-Echo
  - Audible deficiencies are linked to compromising.
  - Solved by better adhering to theory, not deviating further.

- Pre-ringing hypothesis is not needed!
  - You Hear What You Hear but it’s Not What You Think.

Next on: ASRC

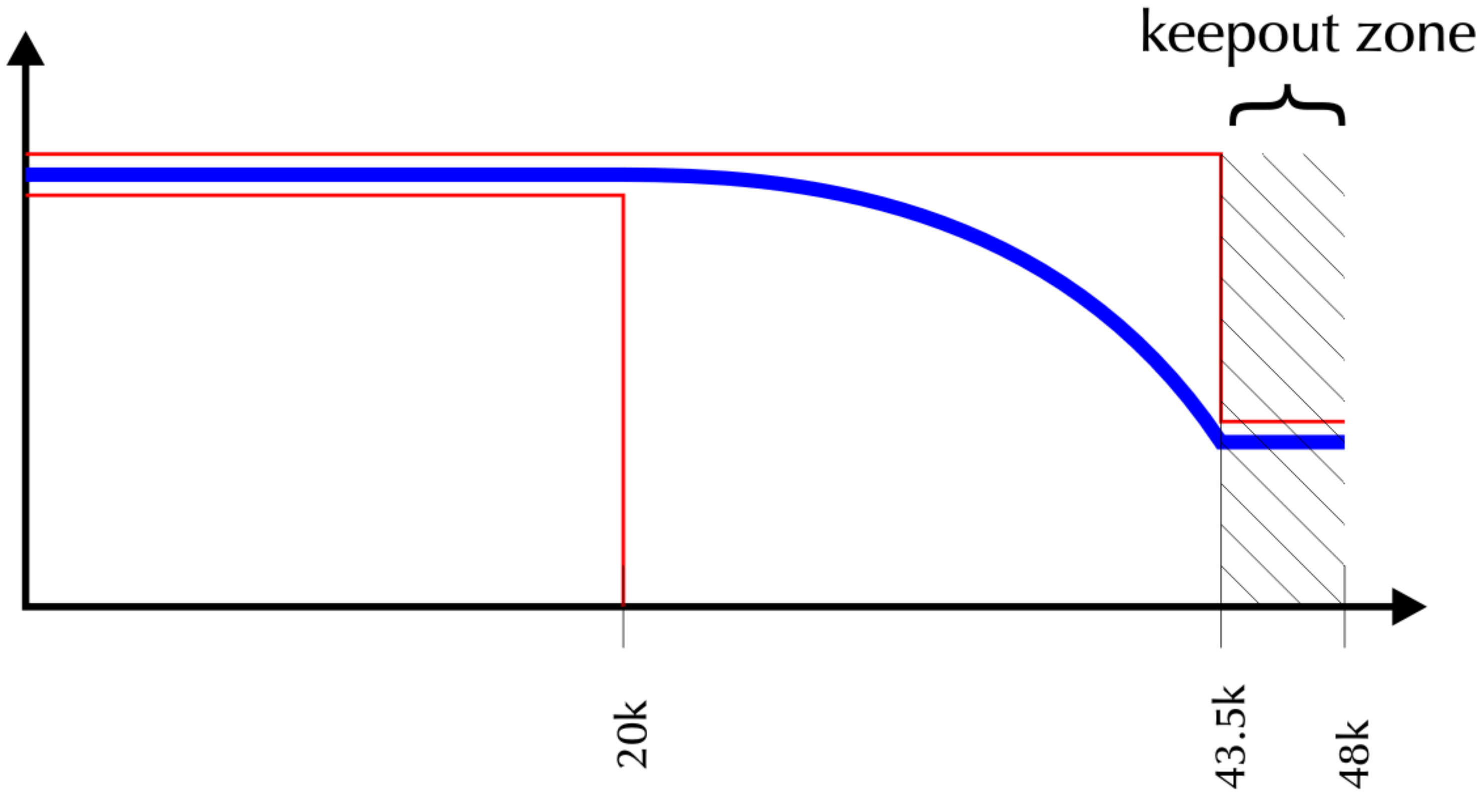
Testing the Pre-Ringing Hypothesis

Testing audibility of brick wall filtering

- Use a 96kHz or 192kHz recording.
- Slice off 0.4535-0.5fs area.
- Test the following filters (never decimate):
  - 20kHz sharp-rolloff
  - 20kHz slow-rolloff
  - 40kHz sharp-rolloff

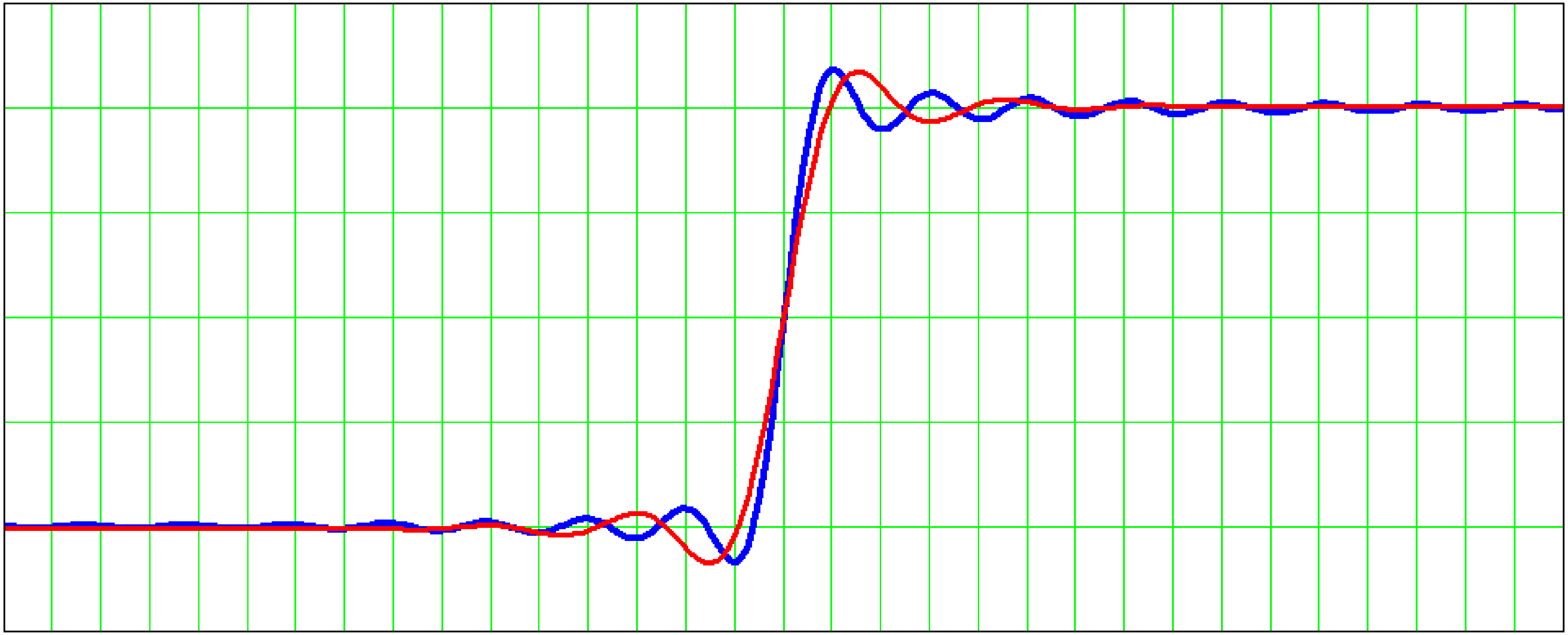
Testing the Pre-Ringing Hypothesis

Effect of Slow-Roll Filter after the fact...



Testing the Pre-Ringing Hypothesis

...reverses effect of sharp rolloff filters



(example: standard 96kHz AD/DA with slow LPF inserted)

Testing the Pre-Ringing Hypothesis

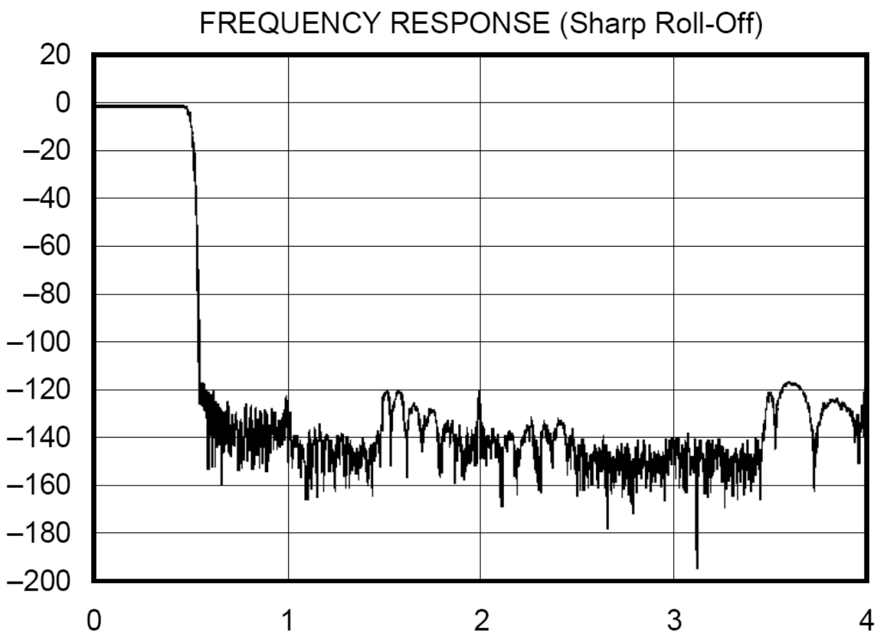
Should we put Slow-Rolloff filters in IC's?

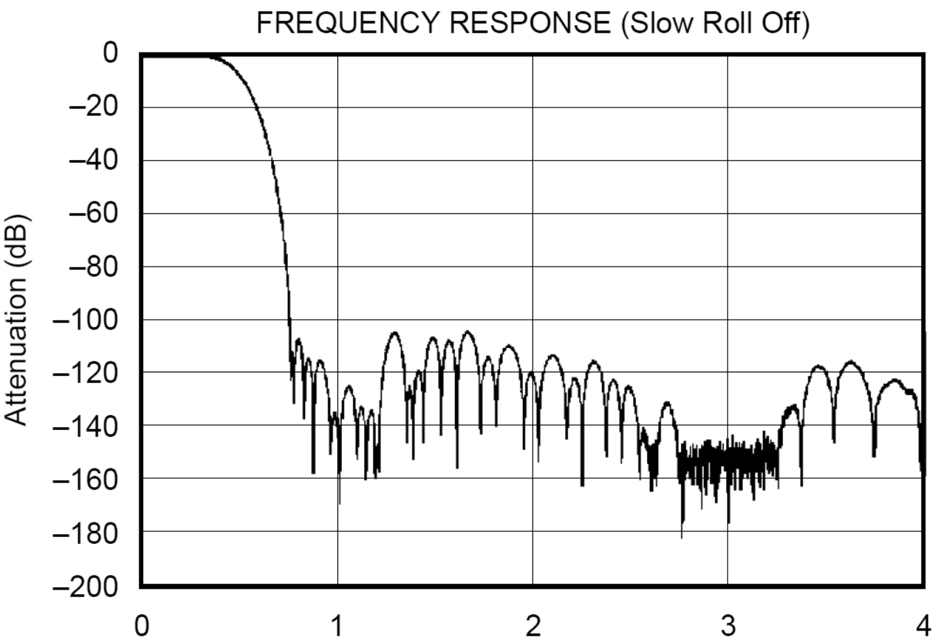
- NO! Compounded SR filters amount to a brick wall.
- Only brick-wall filters are "idempotent".
- Use brick-wall filters throughout and shape response only once.

Testing the Pre-Ringing Hypothesis

How About The Slow-Rolloff Filters in Chip XYZ?

- Intended to reduce latency, NOT improve sound quality

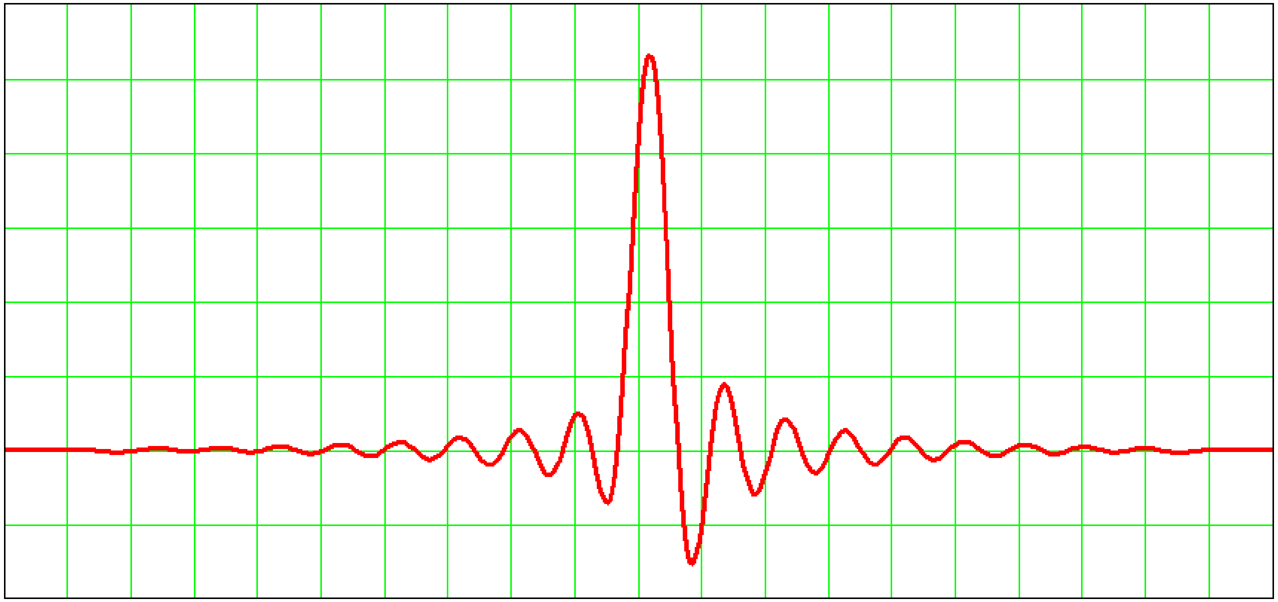




Testing the Pre-Ringing Hypothesis

The Phase-Optimised Filter

- Reduces pre-ringing at the expense of post-ringing



- Magnitude response is maintained
- Cost-effective implementation (IIR+short FIR at f_{sout})
- Reduces latency with minimal loss of sound quality

Testing the Pre-Ringing Hypothesis

Are phase-optimised filters a good thing?

- YES. Much better tradeoff between audio performance and latency.

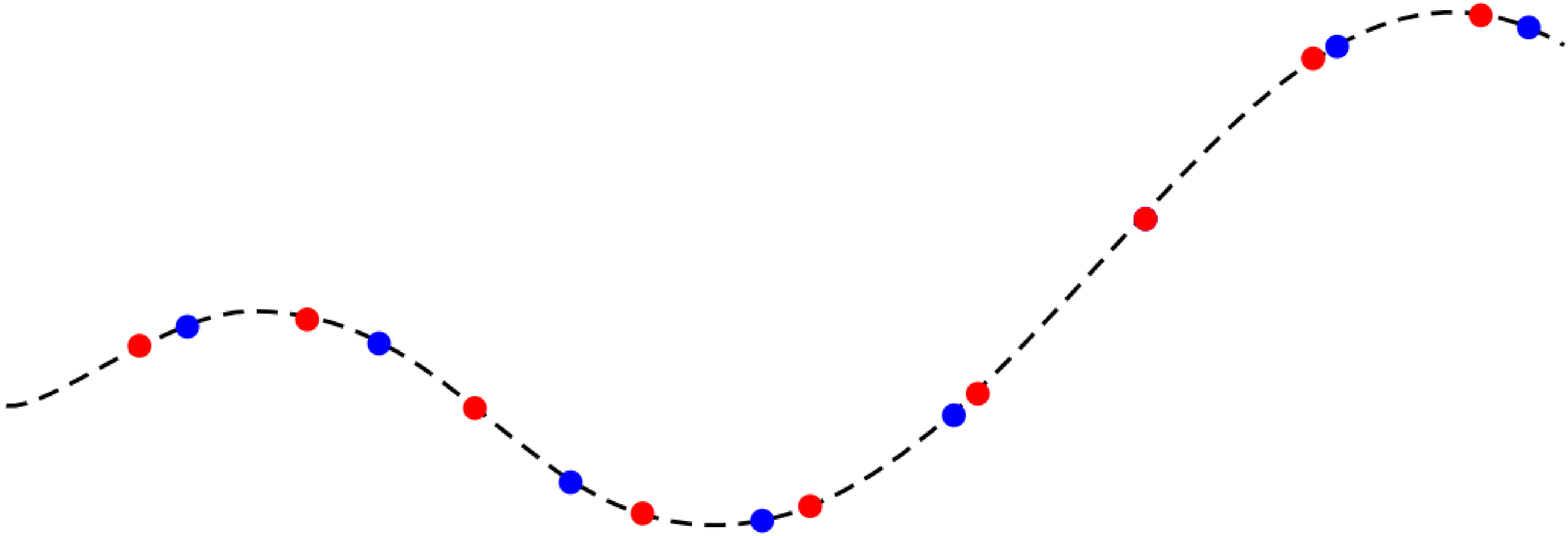
Should phase-optimised filters be standard?

- NO. One pass may be inaudible but 2 passes? 10?
- “Improved sound quality” claim is based on pre-ringing hypothesis.

Asynchronous SRC: The Fine Print

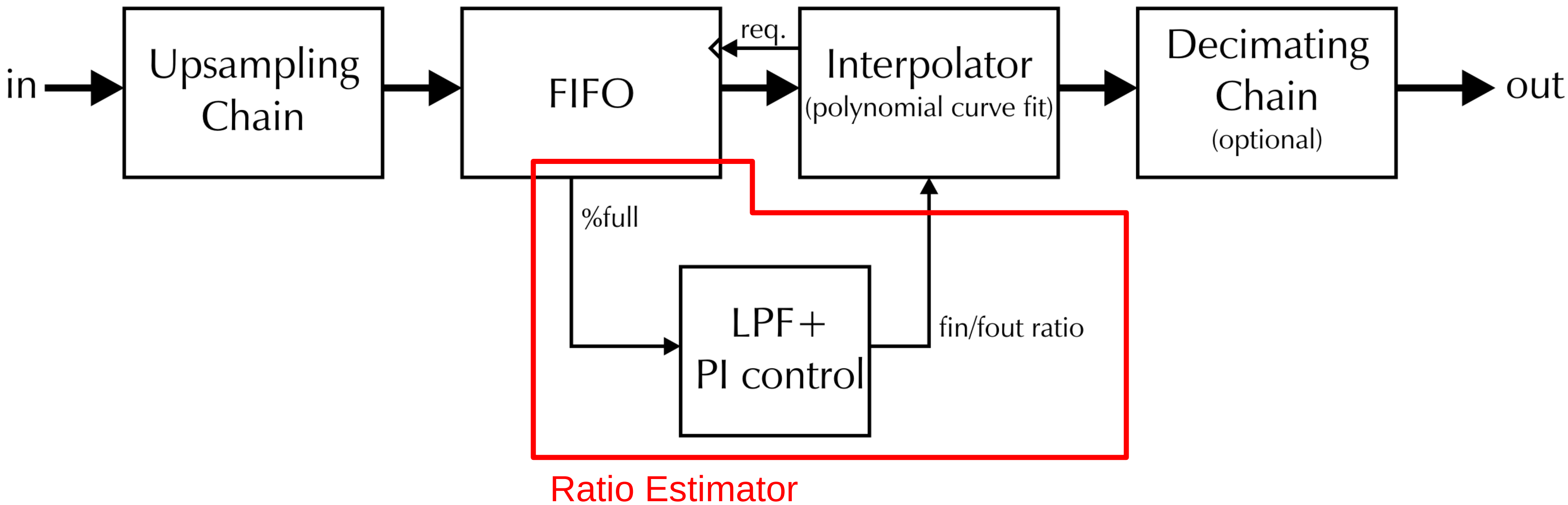
What SRC does

- Reconstruct waveform, resample at new rate
- Done by interpolation



Asynchronous SRC: The Fine Print

Basic Concept of Asynchronous Sample Rate Conversion

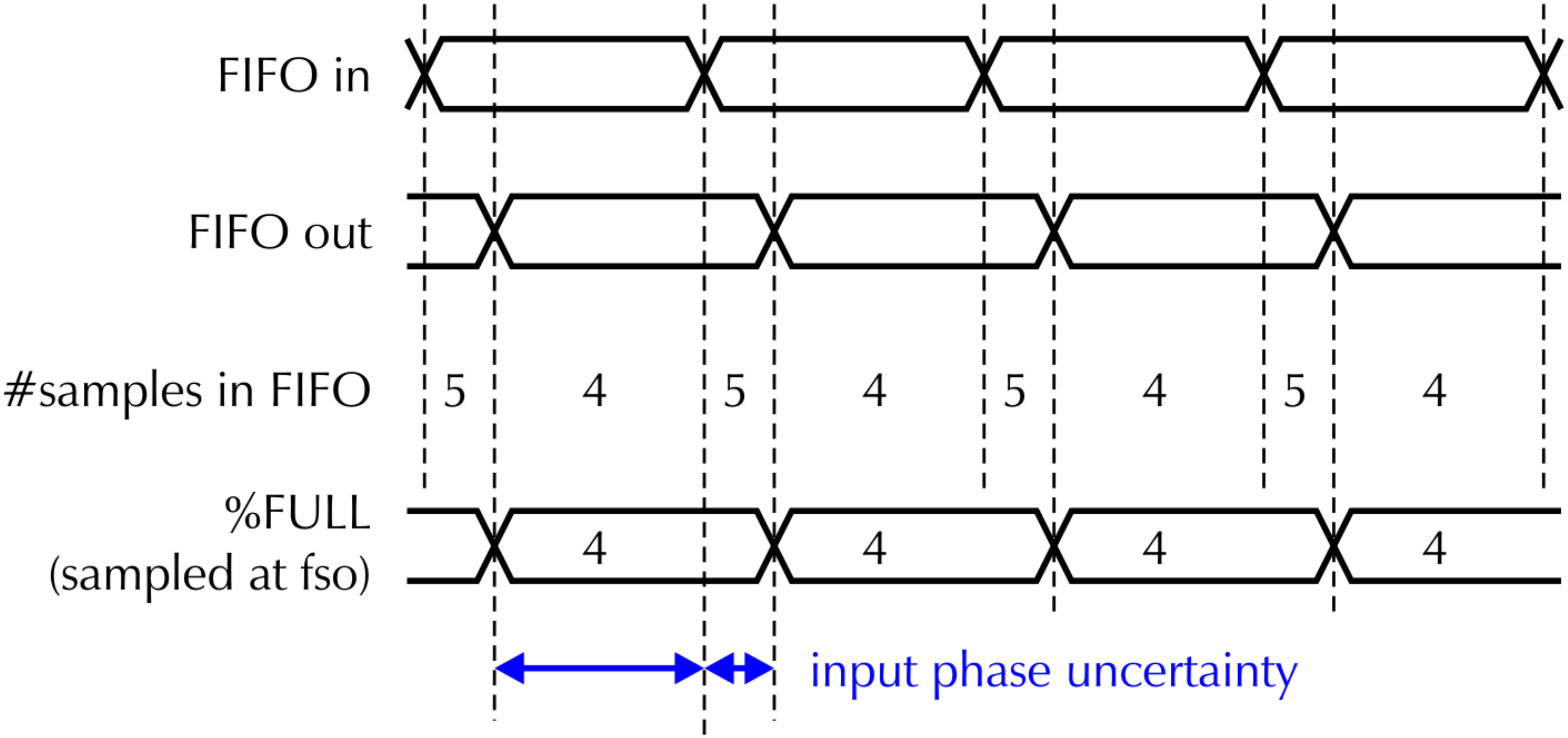


- When f_{sin}/f_{sout} ratio indication is correct, interpolator will read FIFO exactly as often as it is written.
- f_{sin}/f_{sout} ratio is updated to keep FIFO half full.
- Hardware implementations have separate Ratio Estimators

Asynchronous SRC: The Fine Print

Basic problem of ASRC: measurement accuracy

- Accuracy = sampling rate of Ratio Estimator



Asynchronous SRC: The Fine Print

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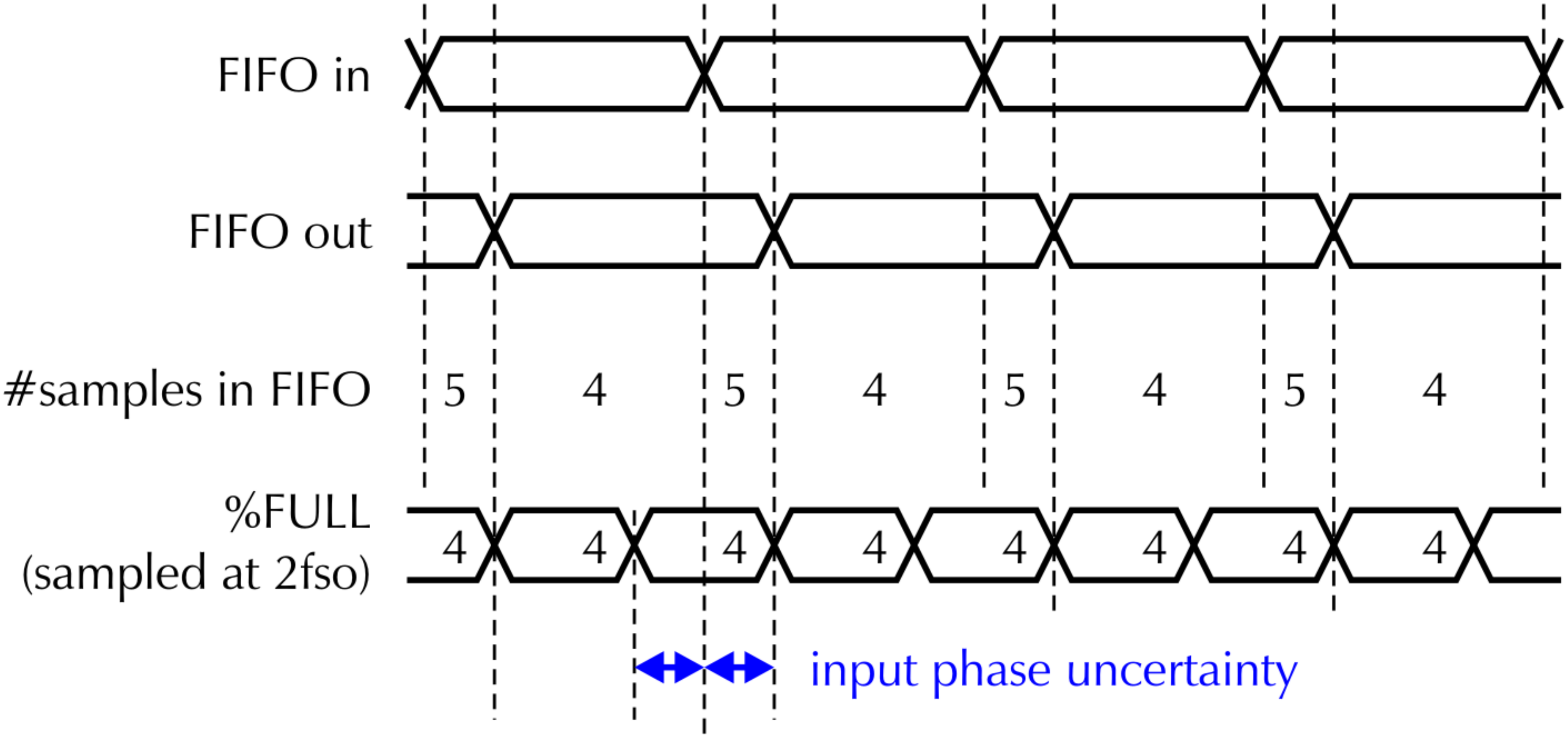
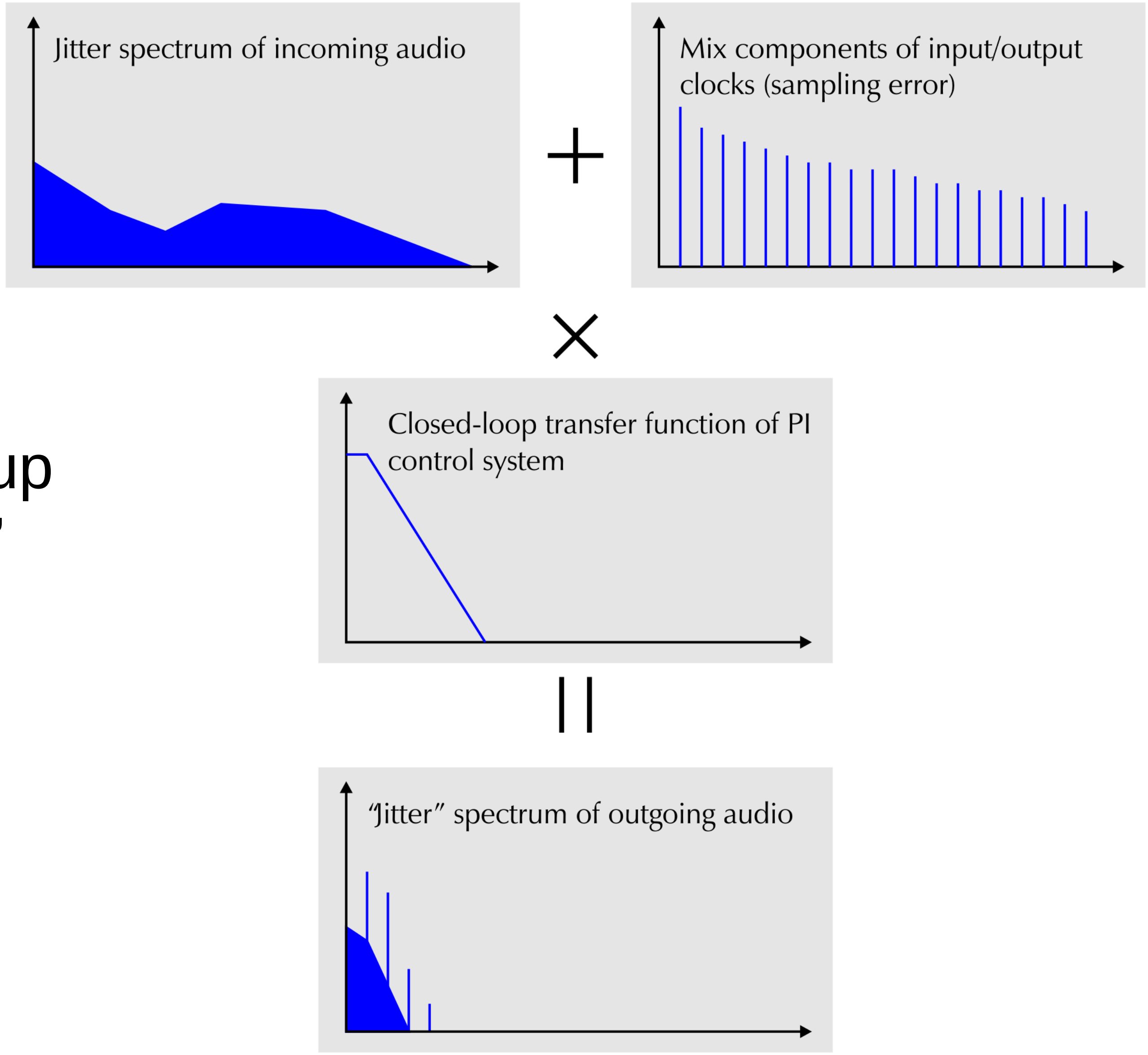


Asynchronous SRC: The Fine Print

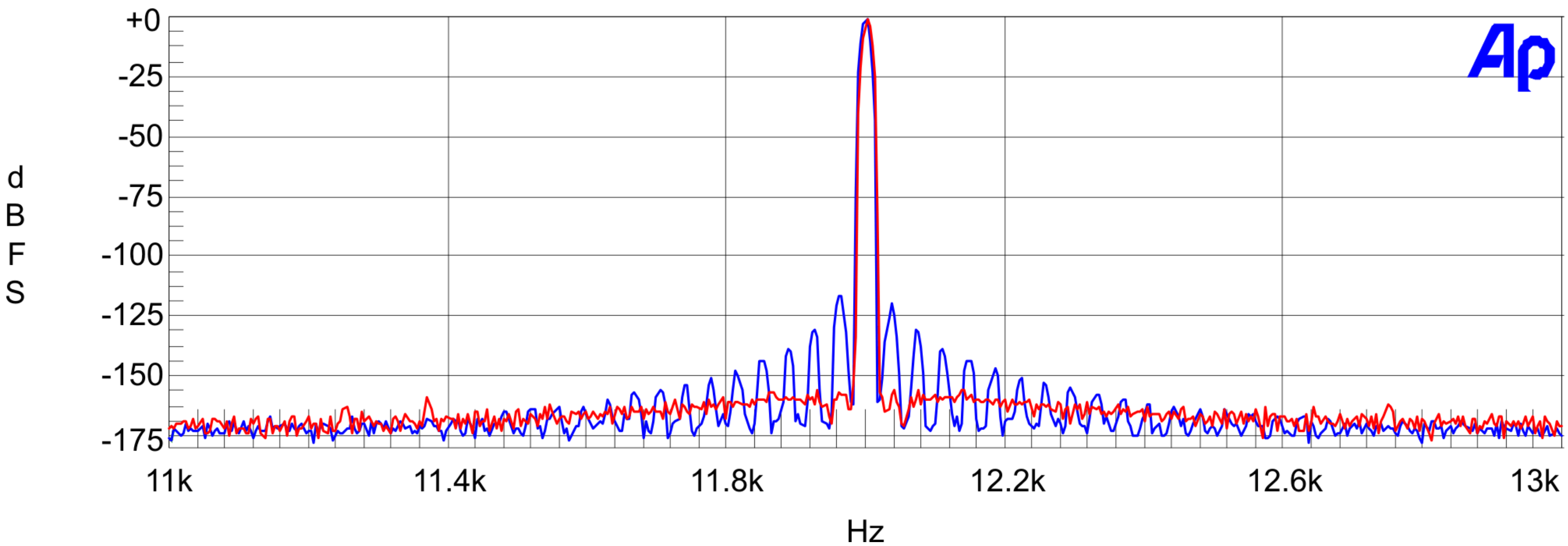
Spectral makeup
of “output jitter”



Asynchronous SRC: The Fine Print

Example IC ASRC

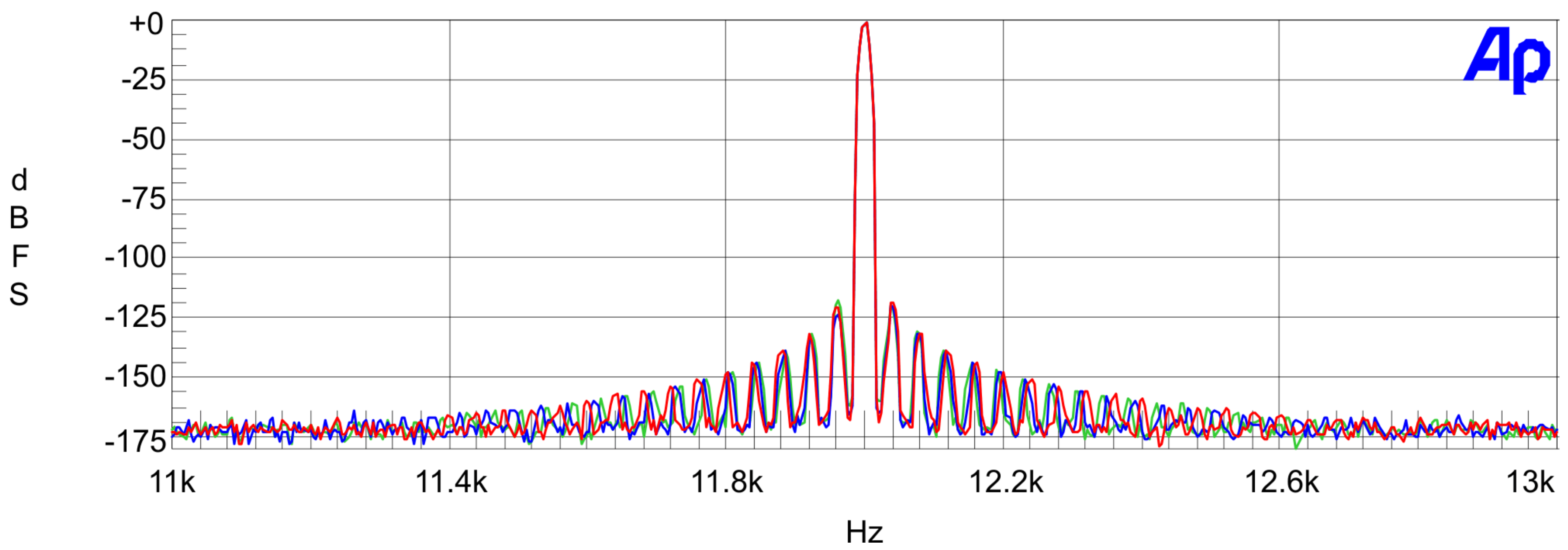
- Output rate = 47.999kHz
- Input rate = 48kHz (blue), 48.025kHz (red)
- Separate independent clock osc drives SRC process



Asynchronous SRC: The Fine Print

Example IC ASRC

- Output rate = 47.999kHz. Input rate = 48kHz (blue)
- Separate independent clock osc drives SRC process. Oscillator can temperature = 25°C 40°C 55°C



Asynchronous SRC: The Fine Print

The Headline

- ASRC's greatly attenuate input jitter...

The Fine Print

- ...but add a lot of their own before doing so!
- And encode the remainder in the data!
  - Signal degradation is irreversible
- ASRC is not a fully digital process!
  - Frequency is a physical quantity = analogue
  - Ratio between independent oscillators = analogue

Asynchronous SRC: The Fine Print

Good Uses for ASRC

- Synchronisation in a mixed-rate environment
- Jitter reduction in DAC. *Run the DAC at an odd rate!*

Not Good uses for ASRC

- Blanket synchronisation issue solver
- Mastering (use synchronous or software based SRC e.g. Barbabatch)

Utterly Repugnant uses for ASRC

- Jitter removal device in one-box players
- “Upsampler” in consumer devices

Next on: Digital Loudspeakers

DSP Filters For Loudspeakers

The Siren Song

- Perfect amplitude/phase/impulse response
- From any speaker
  - Measure speaker response, invert, apply FIR, presto!
- Ultra-steep, linear-phase cross-over

The standard approach

- Impulse inversion method.
- Corrects all linear distortions, including echo’s.

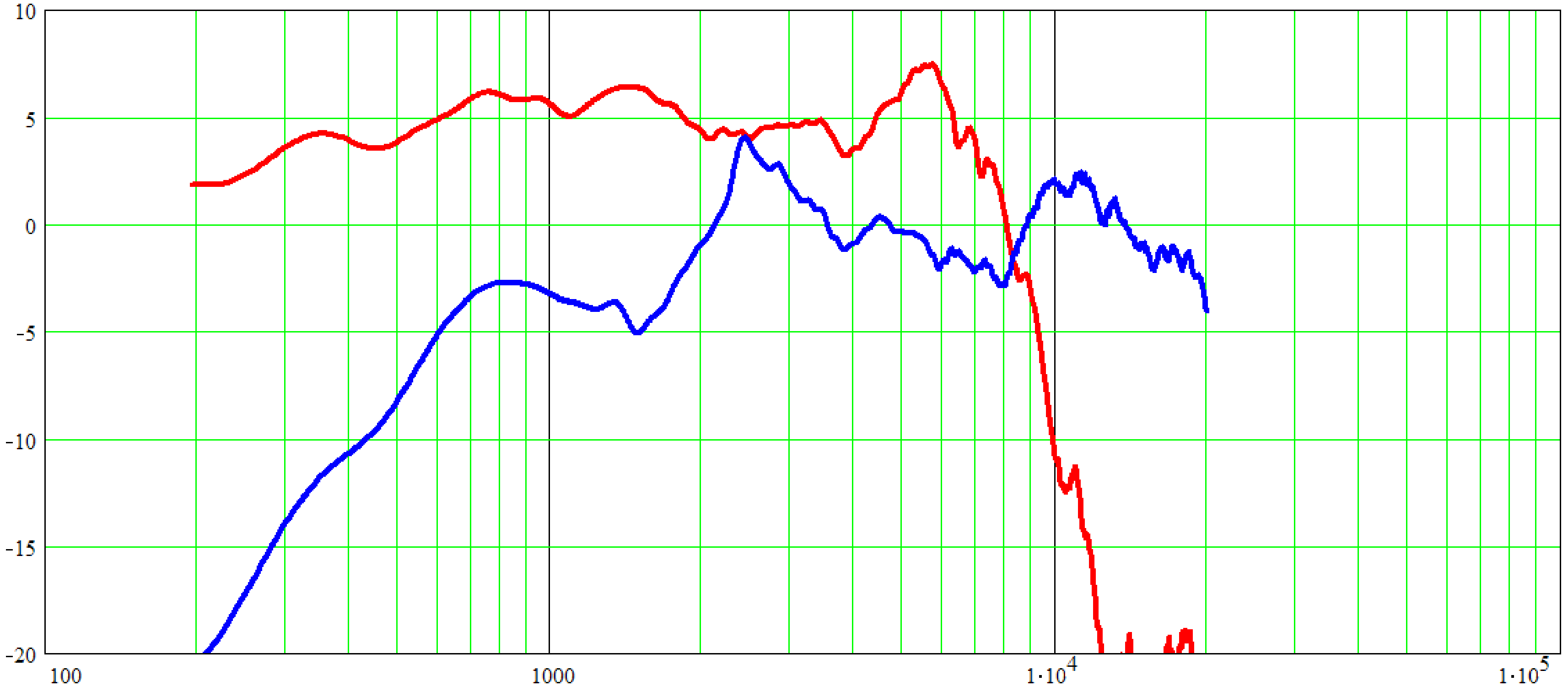
DSP Filters For Loudspeakers

OK, let’s try this!

- Test Mule:
  - 2x5” woofer (Vifa OEM)
  - 1” tweeter (Morel)
  - Classic MTM arrangement

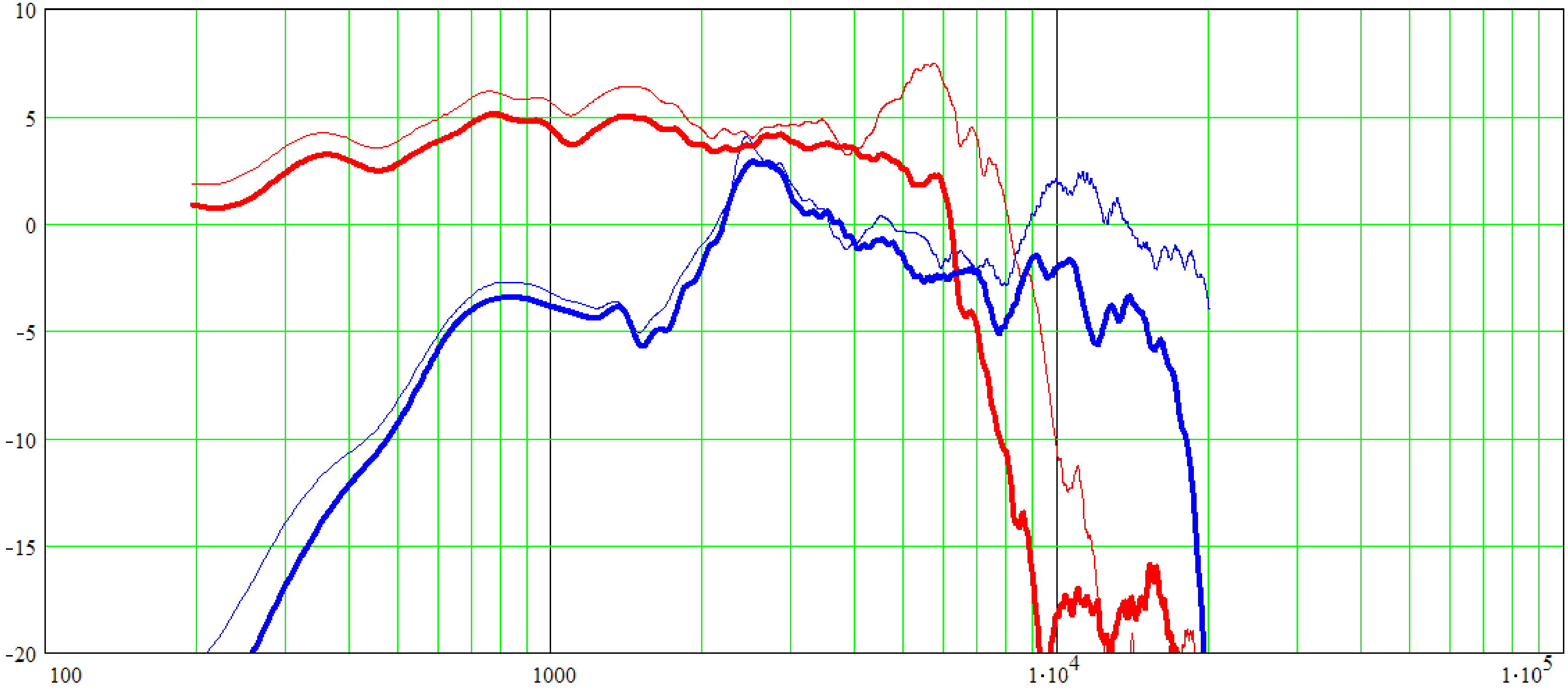
DSP Filters For Loudspeakers

On-Axis Response



DSP Filters For Loudspeakers

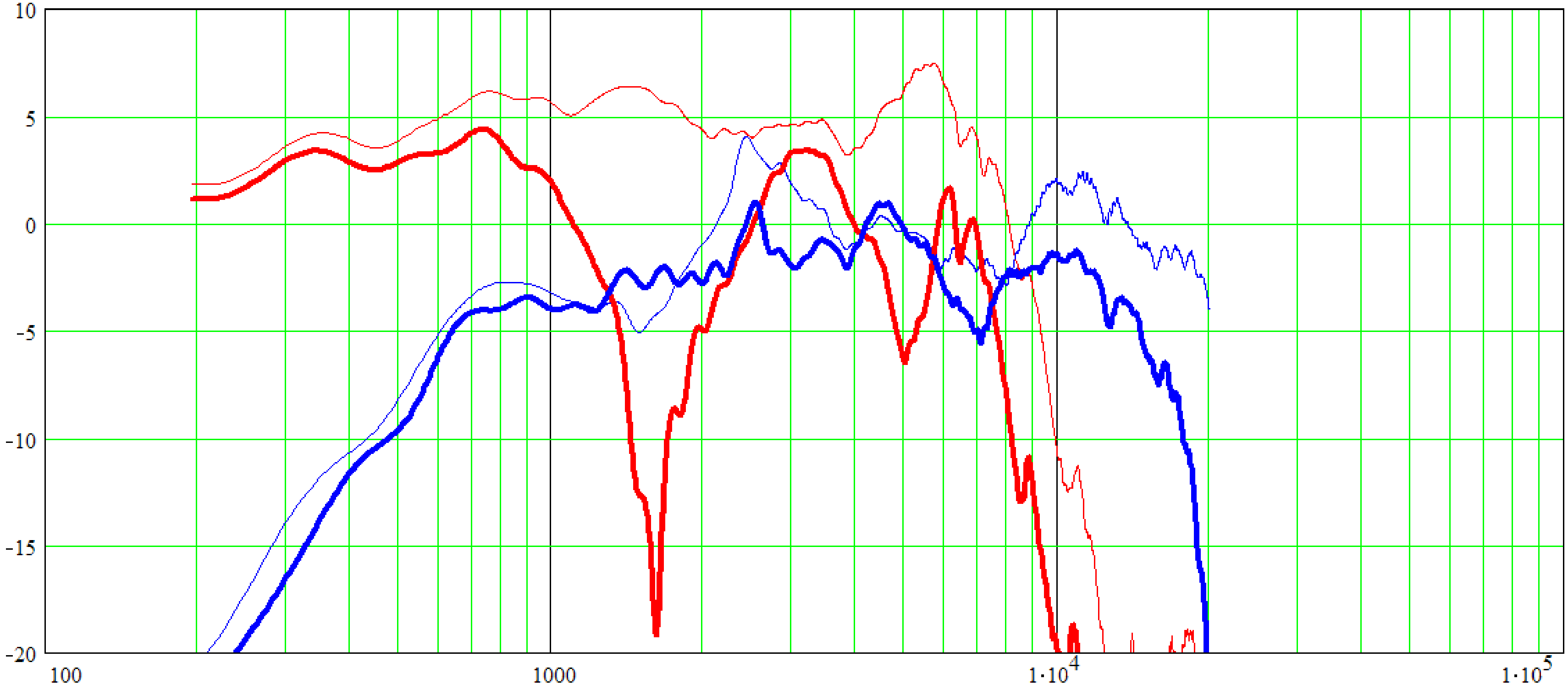
30° Horizontal Off-Axis Response



Note: some peaks/dips shift frequency!

DSP Filters For Loudspeakers

30° Vertical Off-Axis Response



- Tremendous comb filter in LF response
- Other peaks/dips shift frequency!

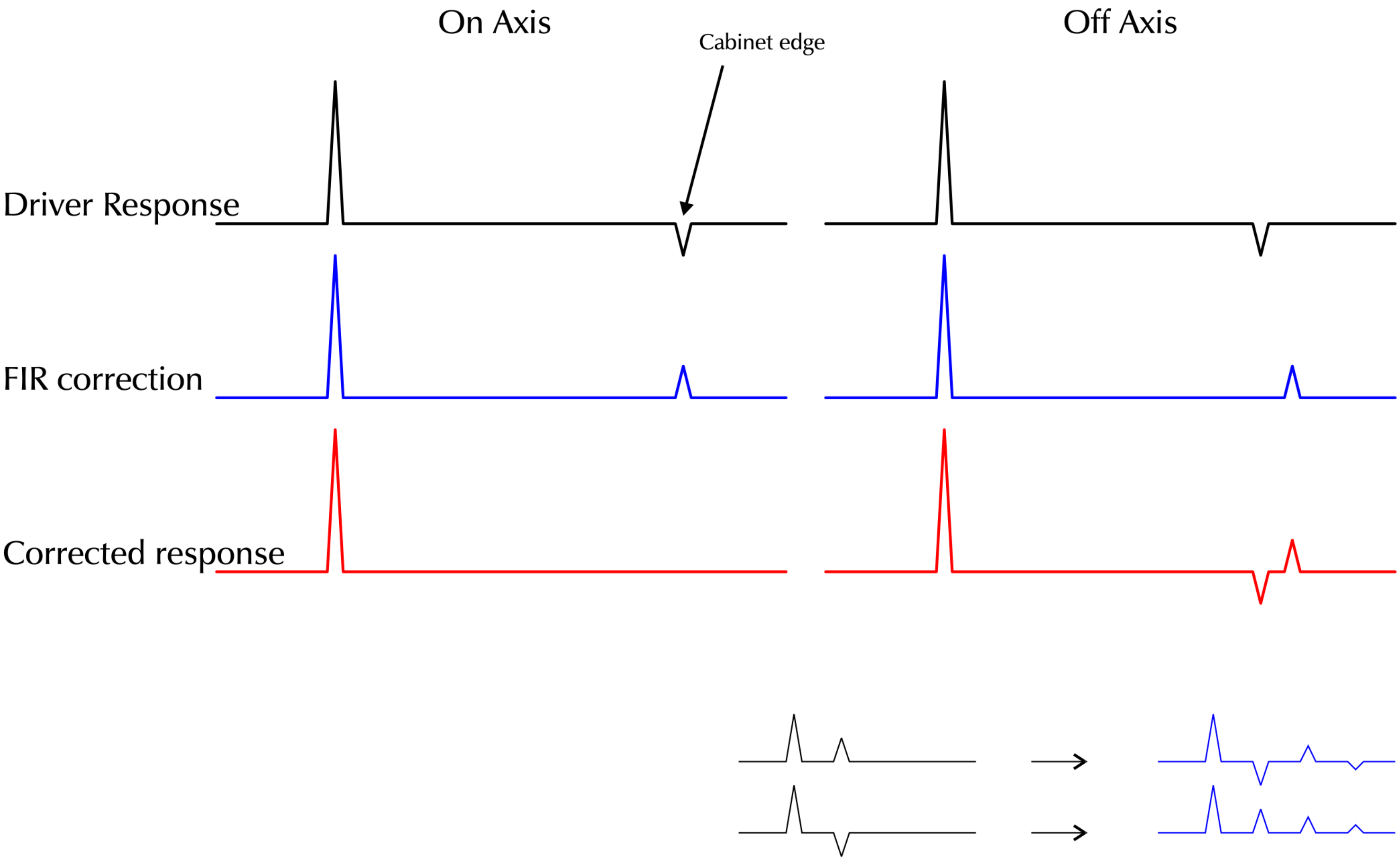
DSP Filters For Loudspeakers

Before we’ve even started...

- Worst irregularities are diffractions
  - Cabinet edges and woofer cones
- Virtual sources are far from drivers
  - Reflections change with listening position
  - Subverts response correction off axis

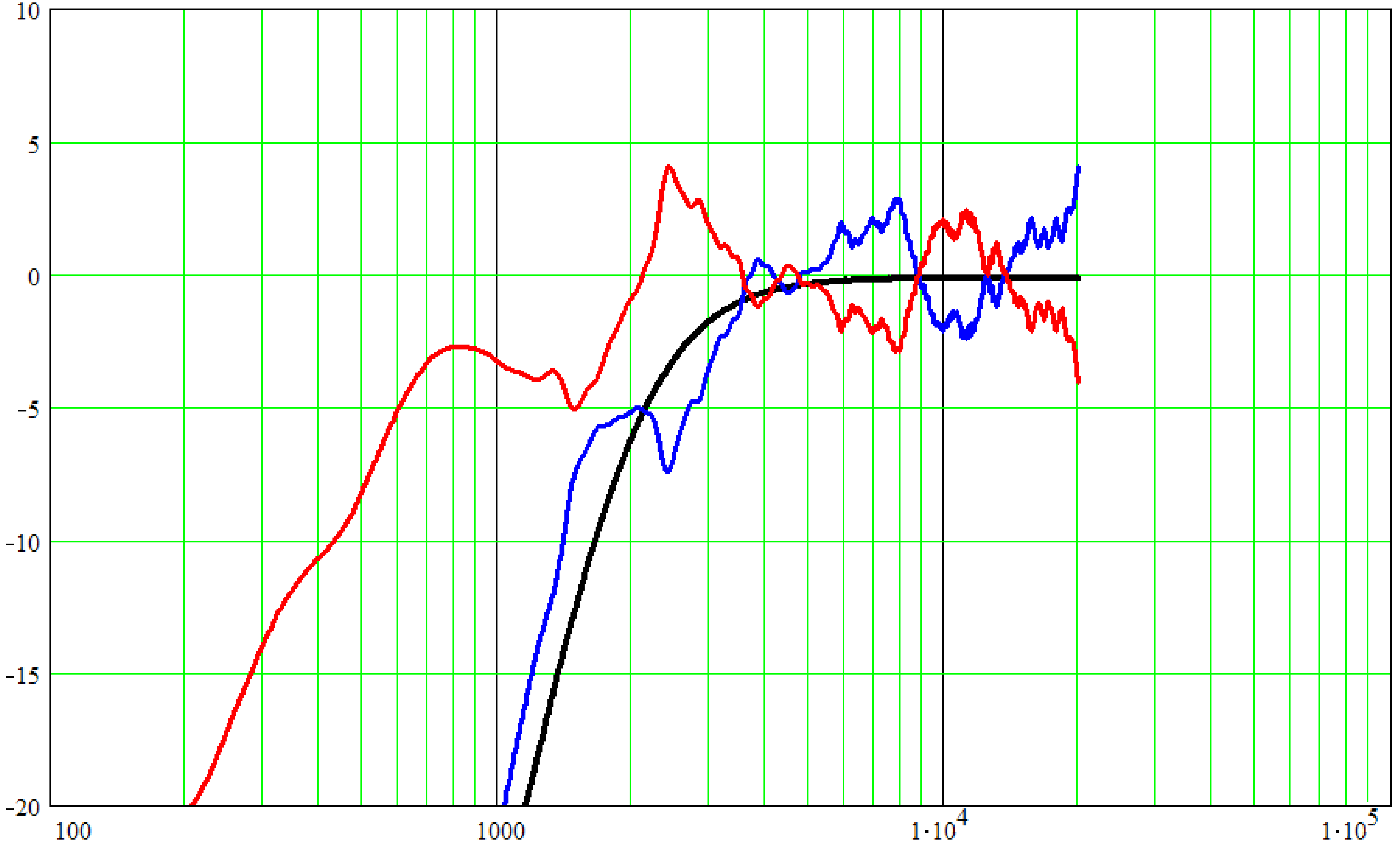
DSP Filters For Loudspeakers

Mis-correction of reflections and diffractions



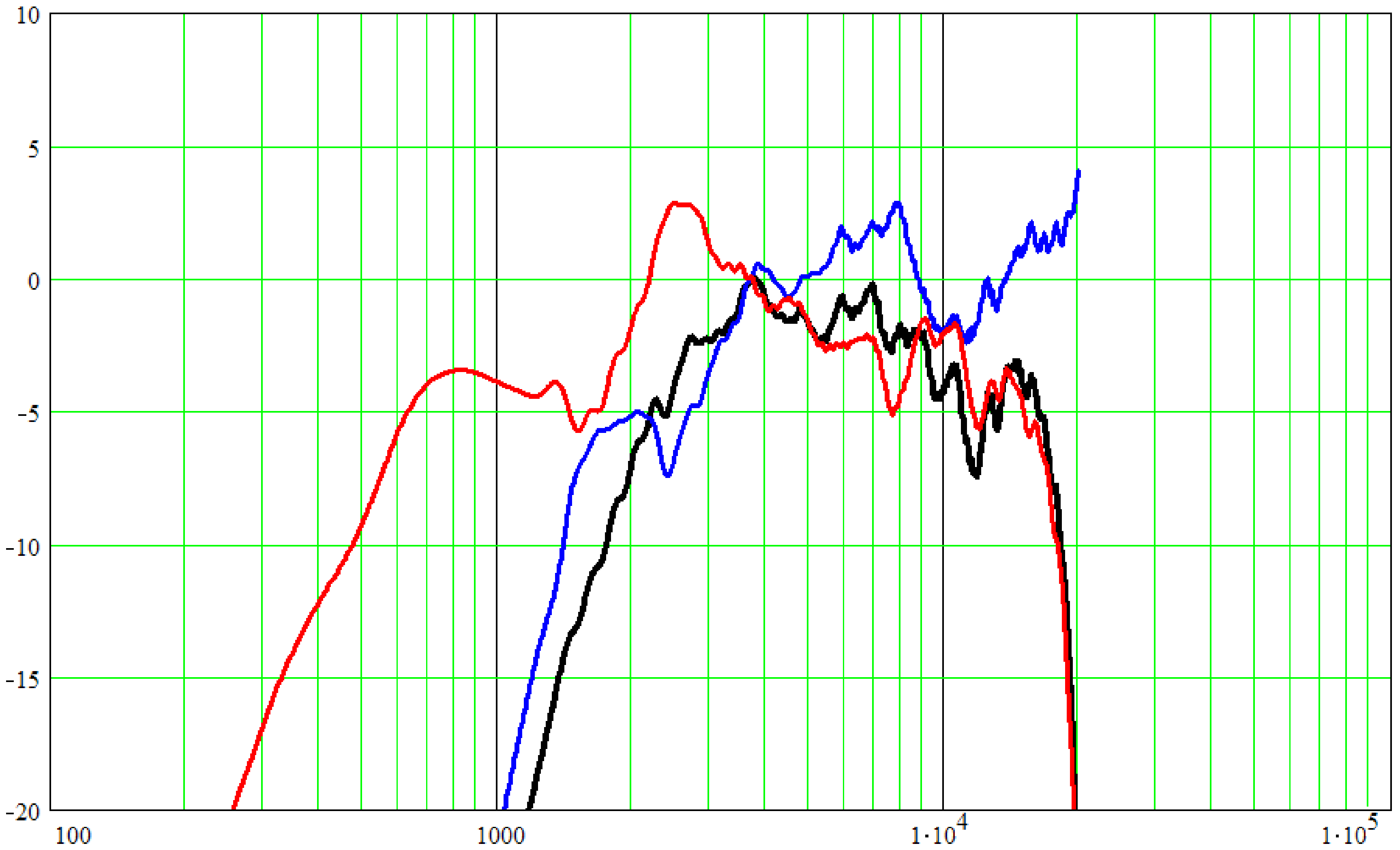
DSP Filters For Loudspeakers

Corrected and Filtered HF response (on axis)



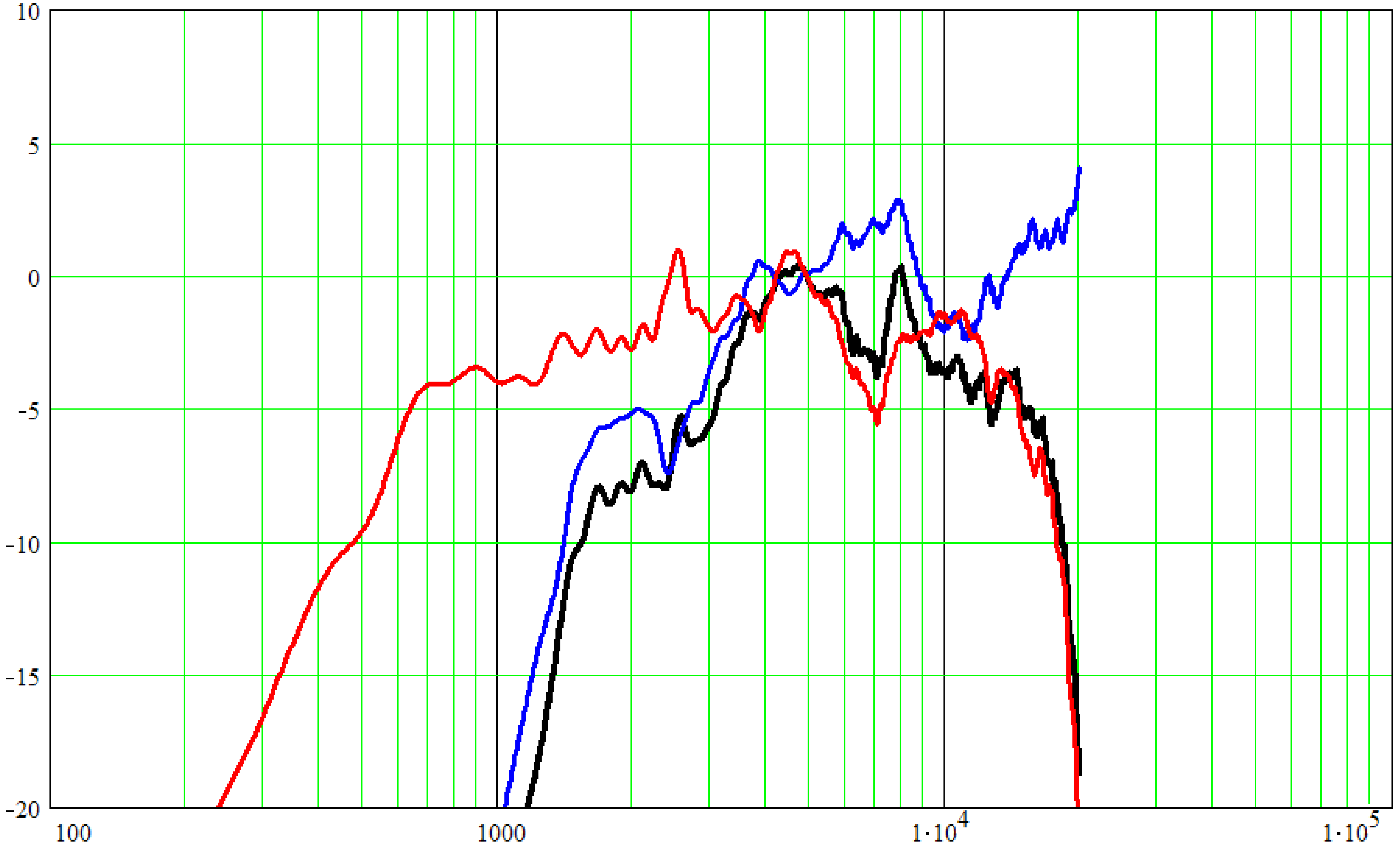
DSP Filters For Loudspeakers

“Corrected” and Filtered HF response (30°H off axis)



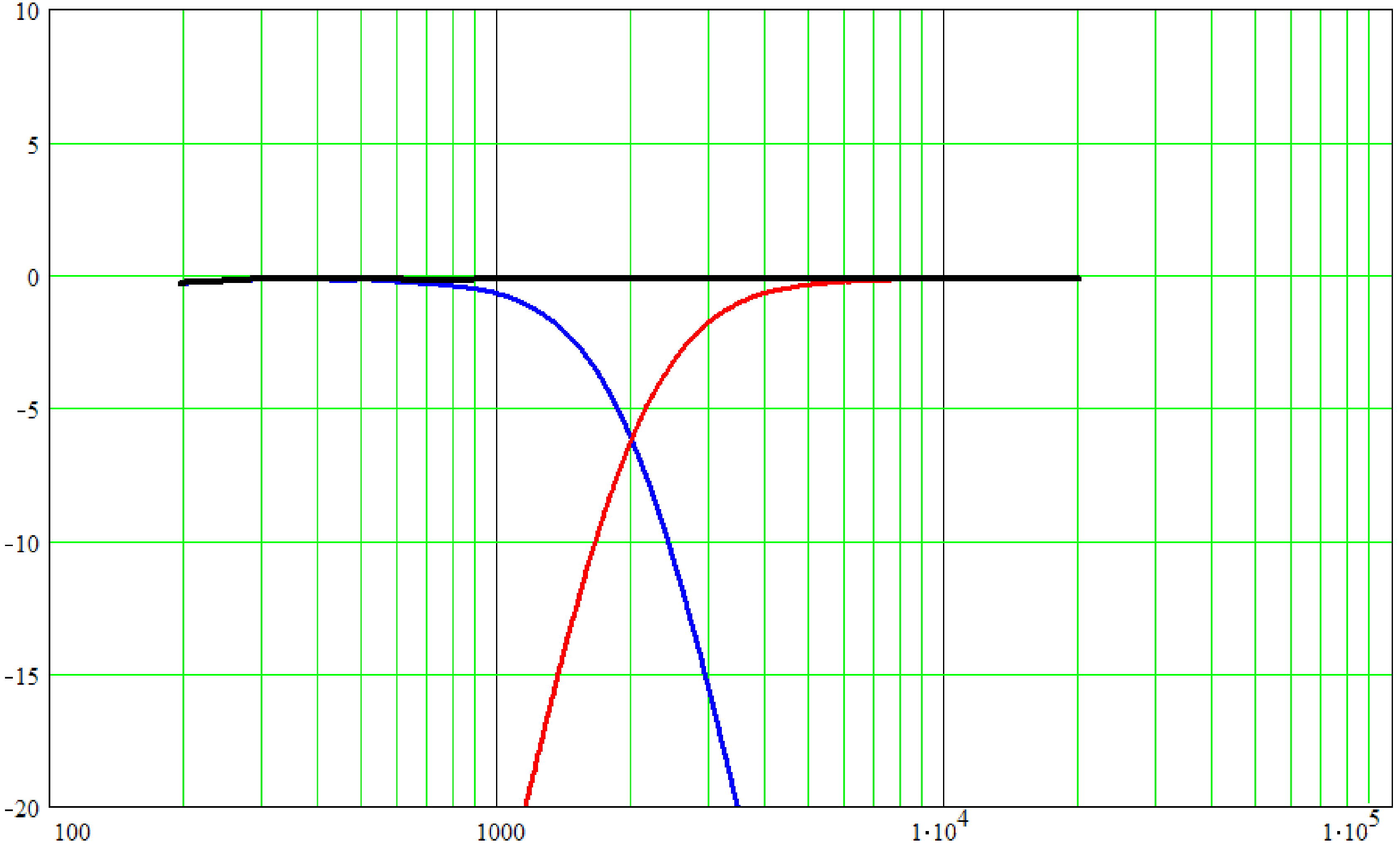
DSP Filters For Loudspeakers

“Corrected” and Filtered HF response (30°V off axis)

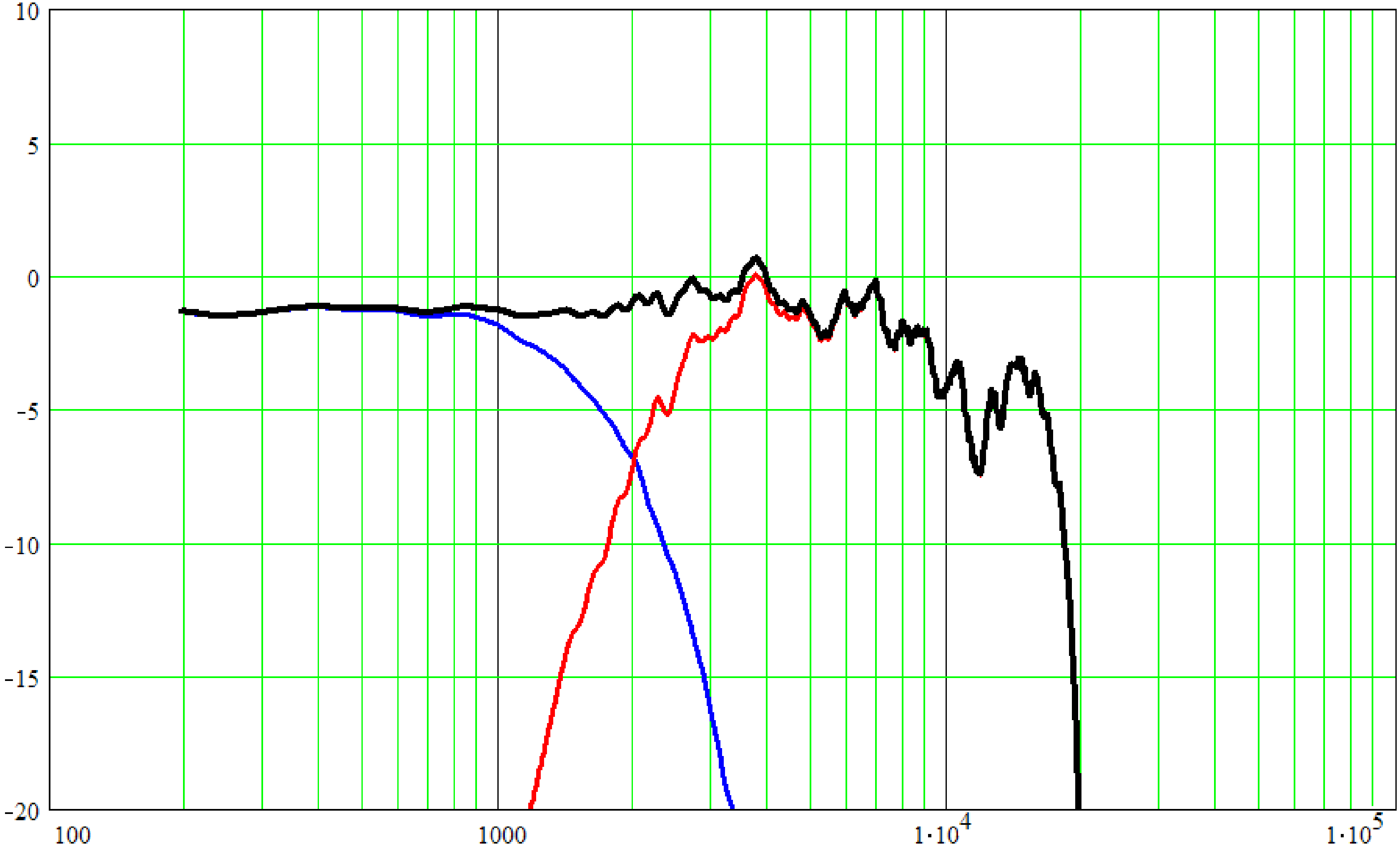


DSP Filters For Loudspeakers

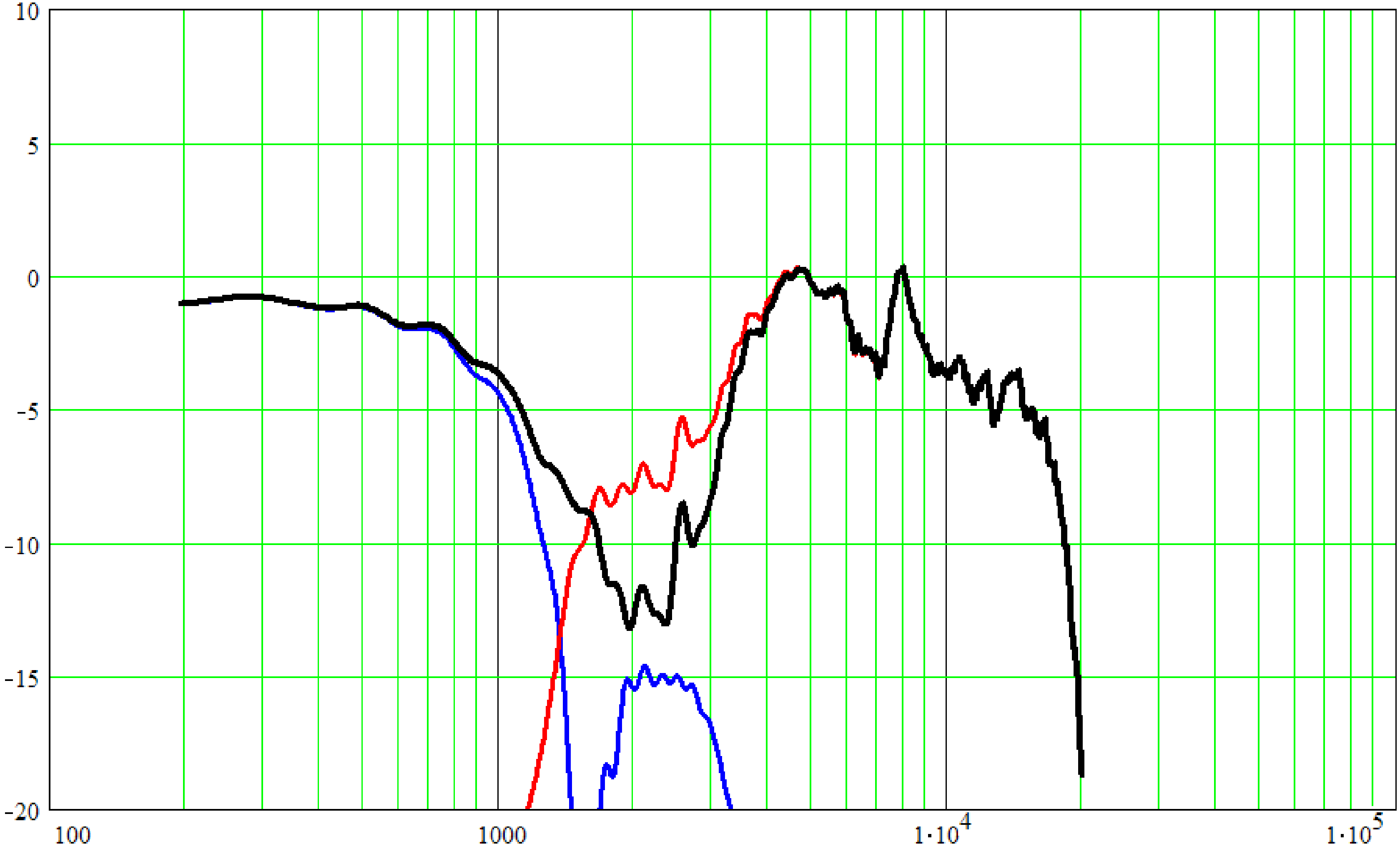
Sum Response (on axis)



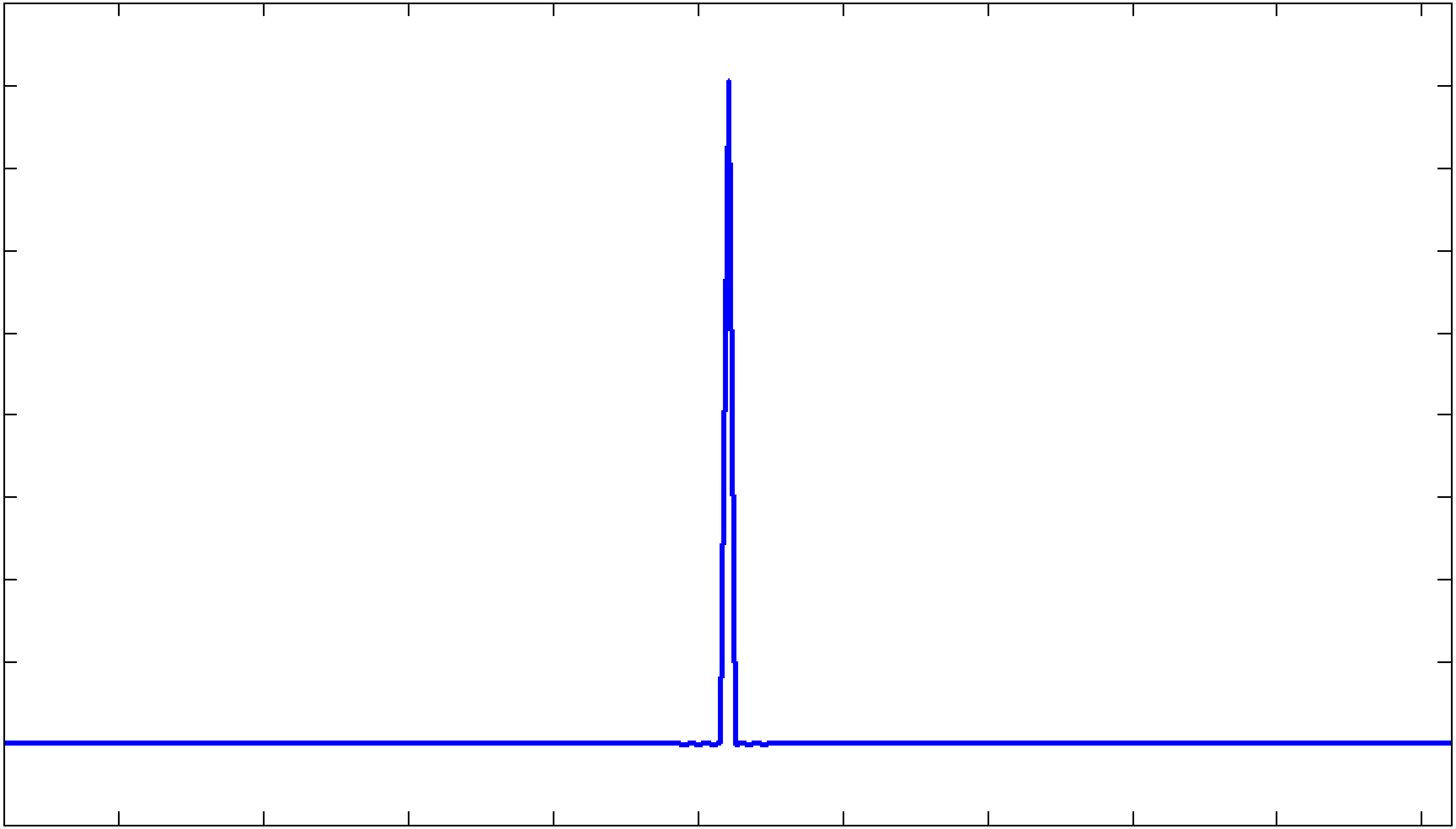
Sum Response (30°H off axis)



Sum Response (30°V off axis)

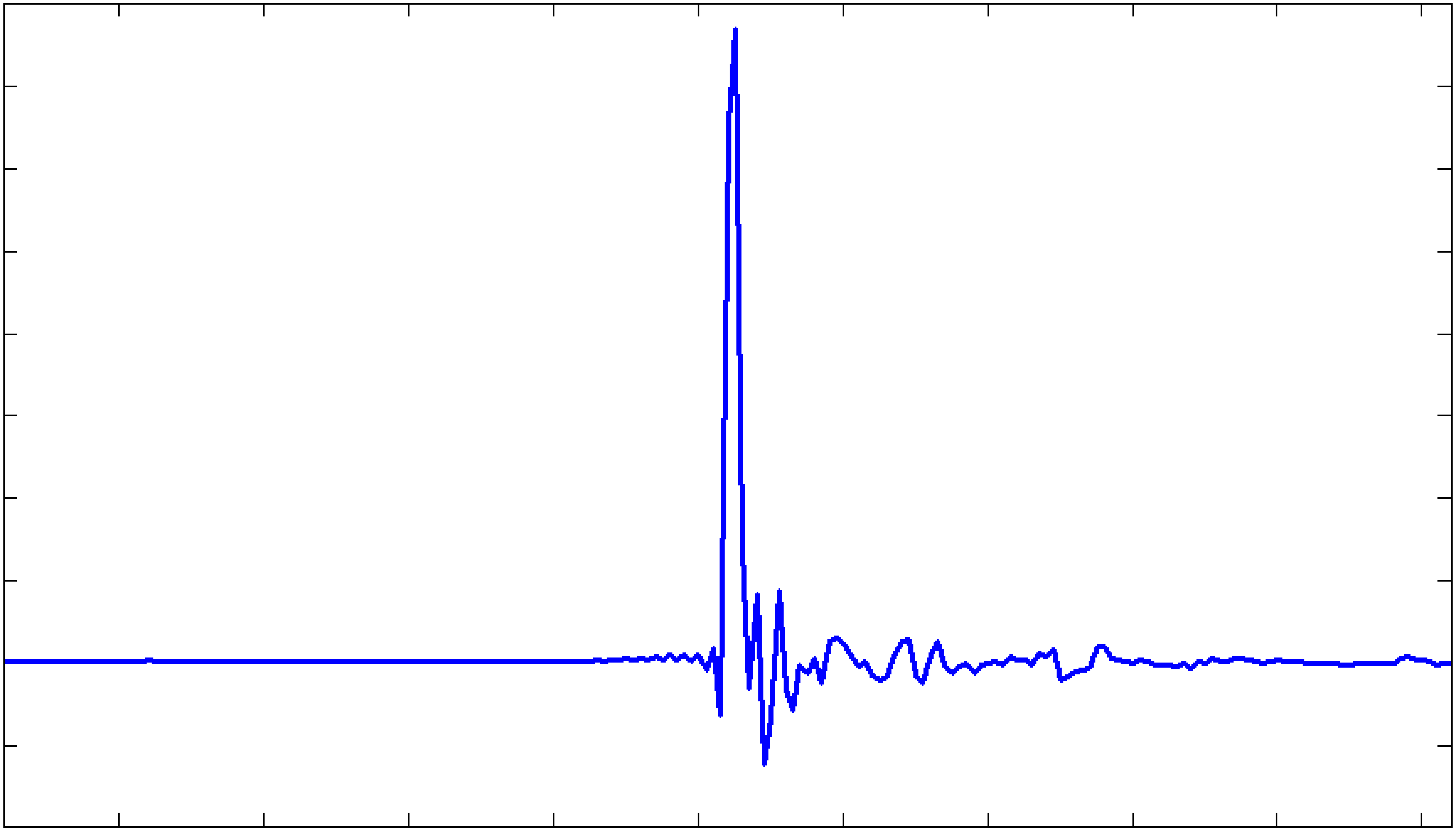


Impulse Response (on axis)



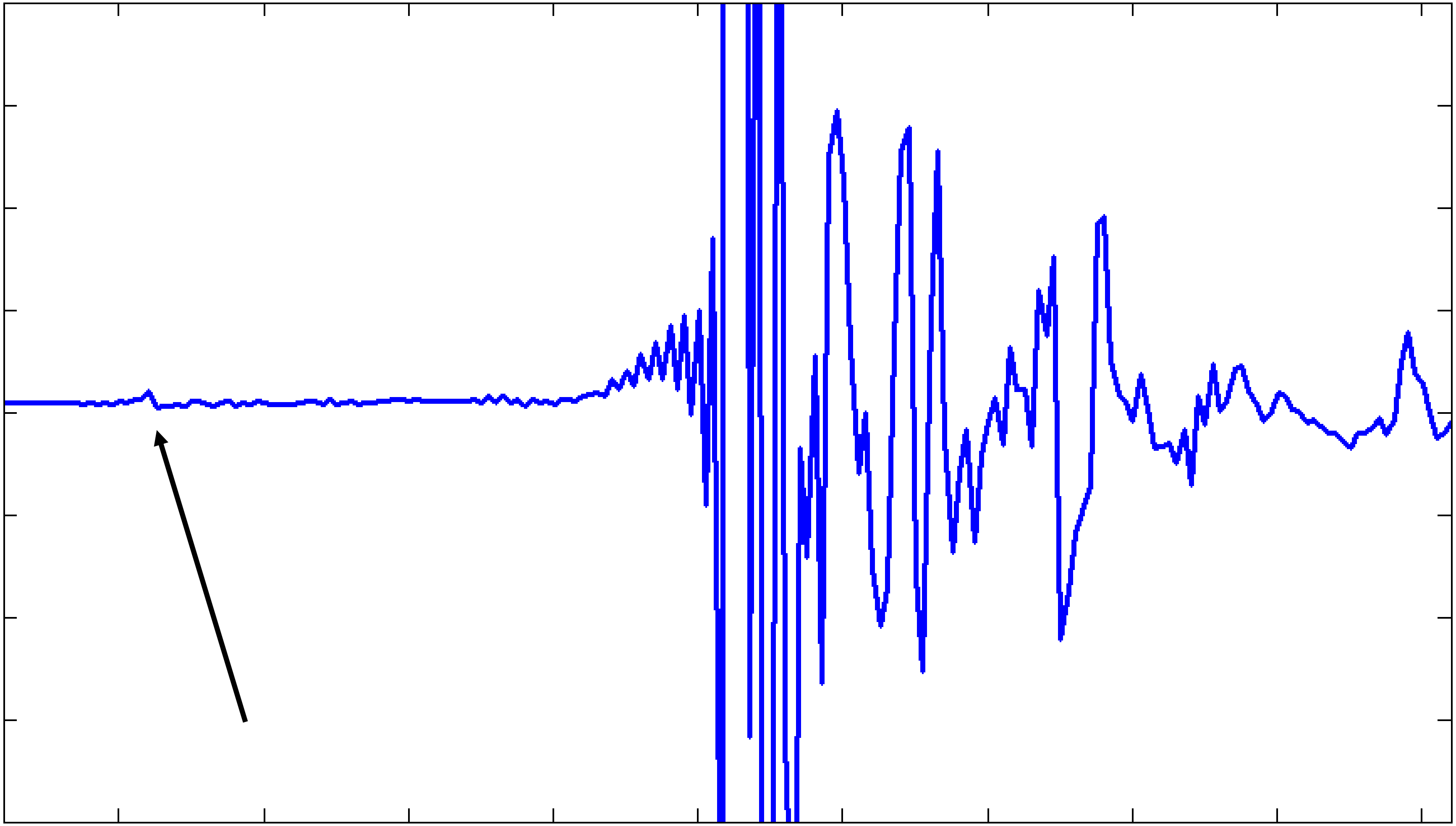
DSP Filters For Loudspeakers

Impulse Response (30°H off axis)

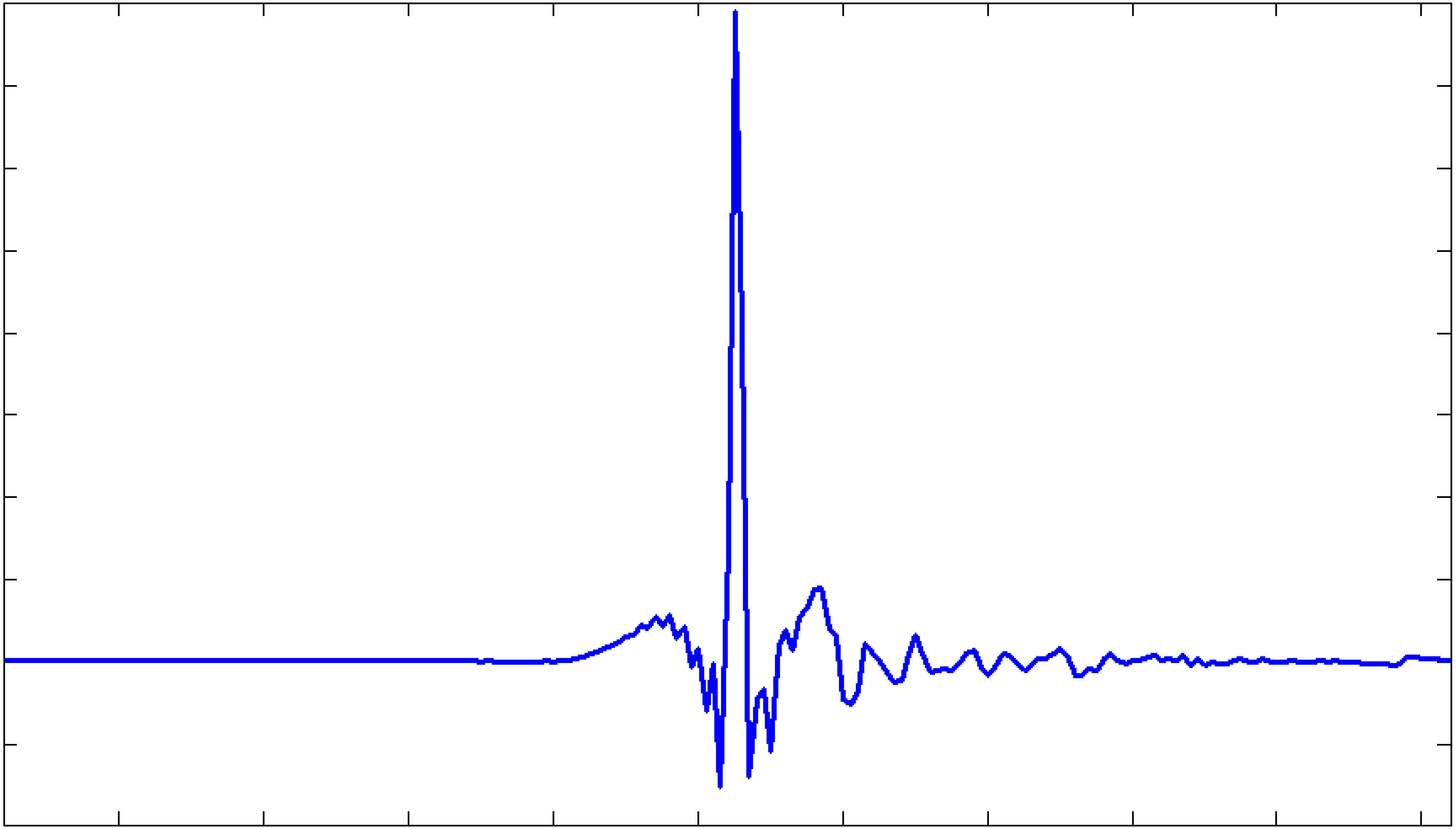


DSP Filters For Loudspeakers

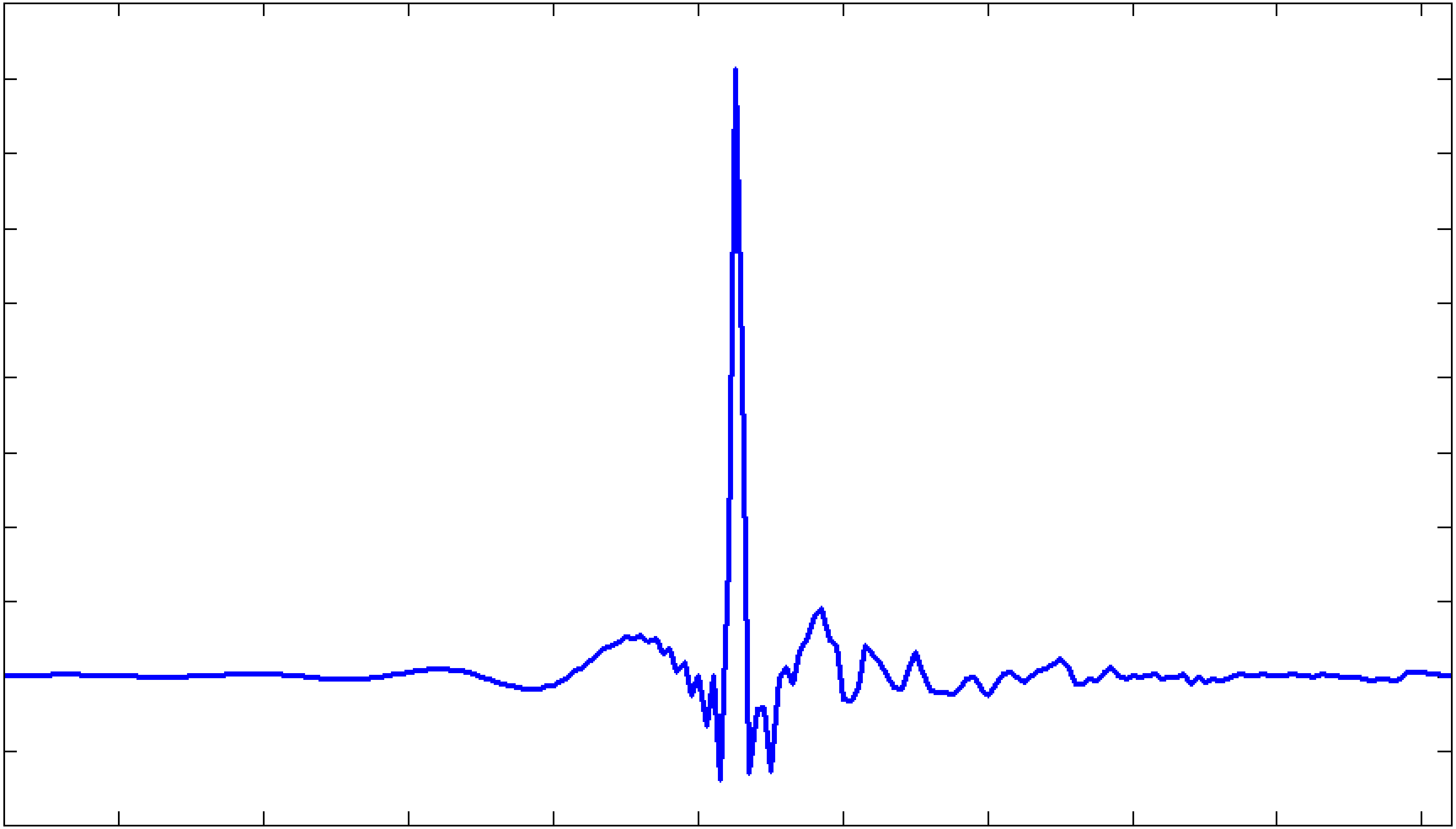
Impulse Response (30°H off axis, Y zoom)



Impulse Response (30°V off axis)



Impulse Response (30°V off axis, ultra-steep filter)



DSP Filters For Loudspeakers

Observations

- Heavy correction exacerbates acoustic problems
- Steep, linear-phase filtering causes pre-ringing in off-axis response
- Linear-phase target response invites pre-echos

⇨Brute-force correction produces ugly, smeared sound

DSP Filters For Loudspeakers

Sensible approach to correction:

- Don’t Shave Off The Hair. It’ll Grow Back.

- Limit scope of correction to a few periods

⇨ *The subtler the correction, the wider the listening angle in which it still makes some sense.*

DSP Filters For Loudspeakers

Even better approach to correction: manually!

- Forget FIR

- Of each bump and trough, find cause
  - If the driver is the source: correct ruthlessly
  - If the source is elsewhere: EQ gently

- Know your acoustics...

Cross-over Filtering

- Use shallow slopes
- Target minimum phase sum
  - We really don’t want linear phase HPF!
  - We only care about the sum, not the individual drivers.
- (...)

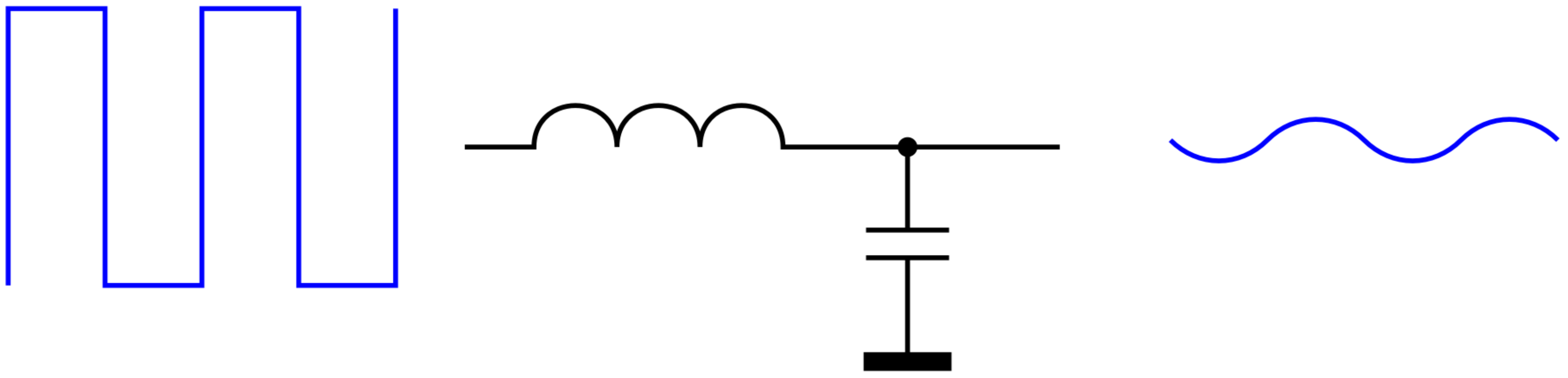
Conclusion

- DSP does not exonerate you from doing your acoustical homework
  - You might even need to work harder
  - Some acoustic concepts really are “broken”
- Automated design procedure = pipe dream
- Impulse inversion method is naïve

Class D and EMI

Low-frequency EMI: Carrier and low harmonics.

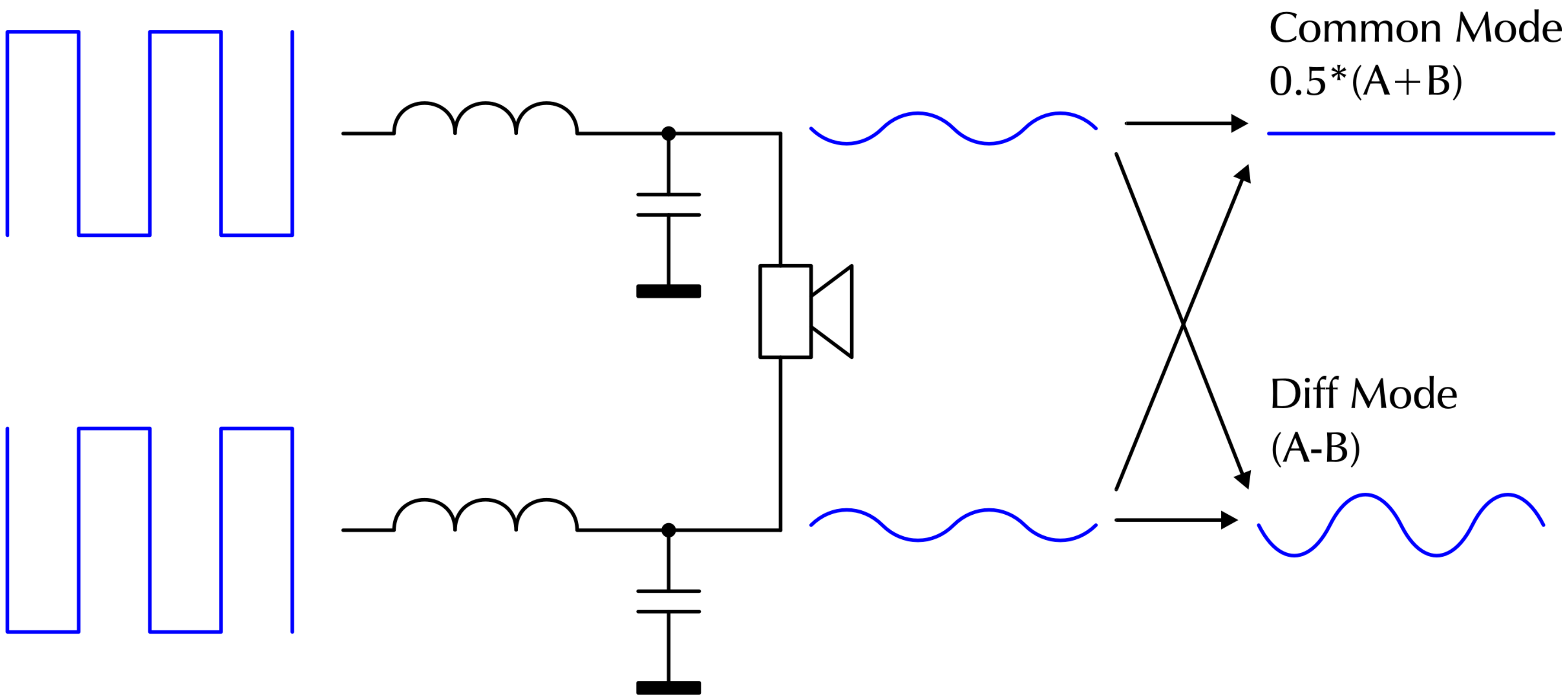
- Close match with theory.
- Ripple cancelling possible.
- Not an EMI issue except for long cables
- Not a tweeter issue (come off it!)



Class D and EMI

Common and Differential Mode in H-Bridge Class D

- “Class AD”. Carriers and modulation are out of phase

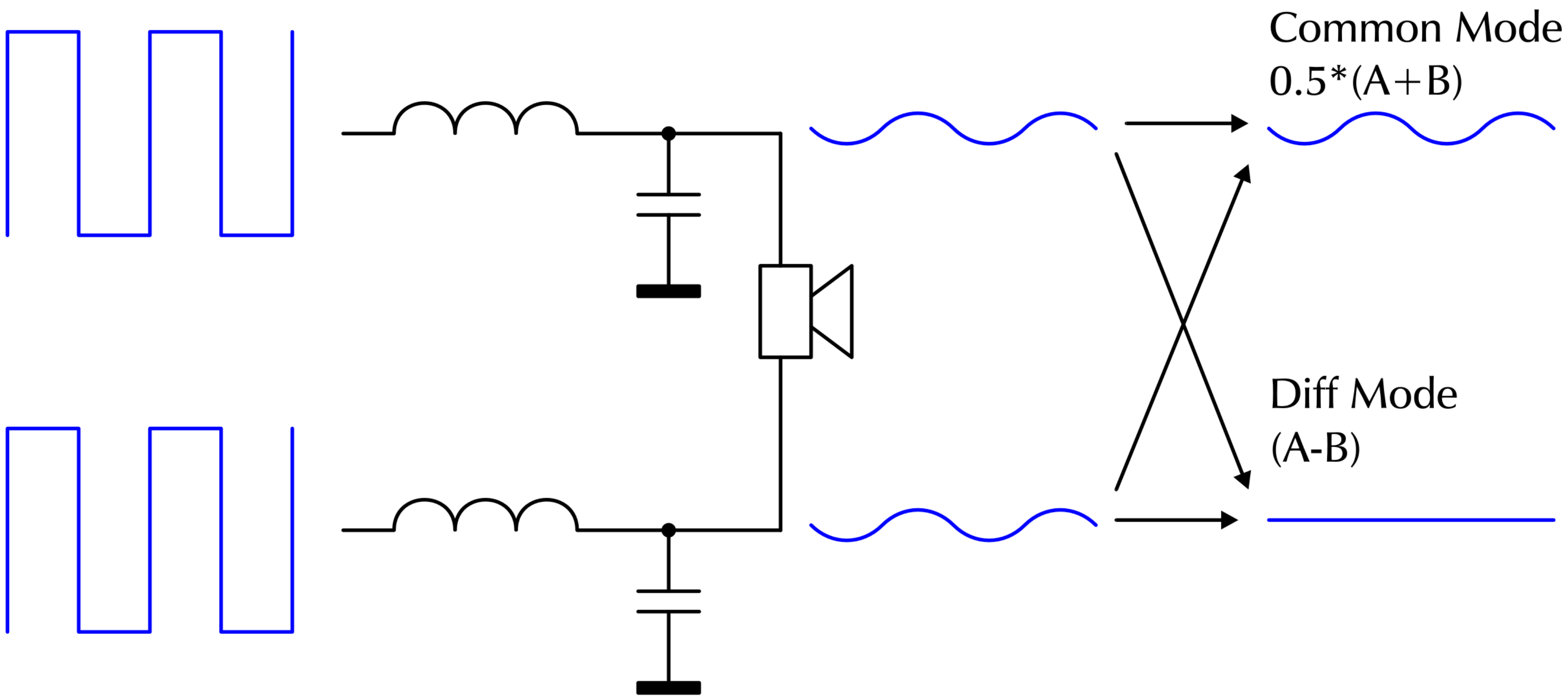


Note: Common-mode is what radiates off cables.

Class D and EMI

"Class BD".

- Carriers are in phase. Modulation is out of phase.



HF across load is reduced but CM increases.

Class D and EMI

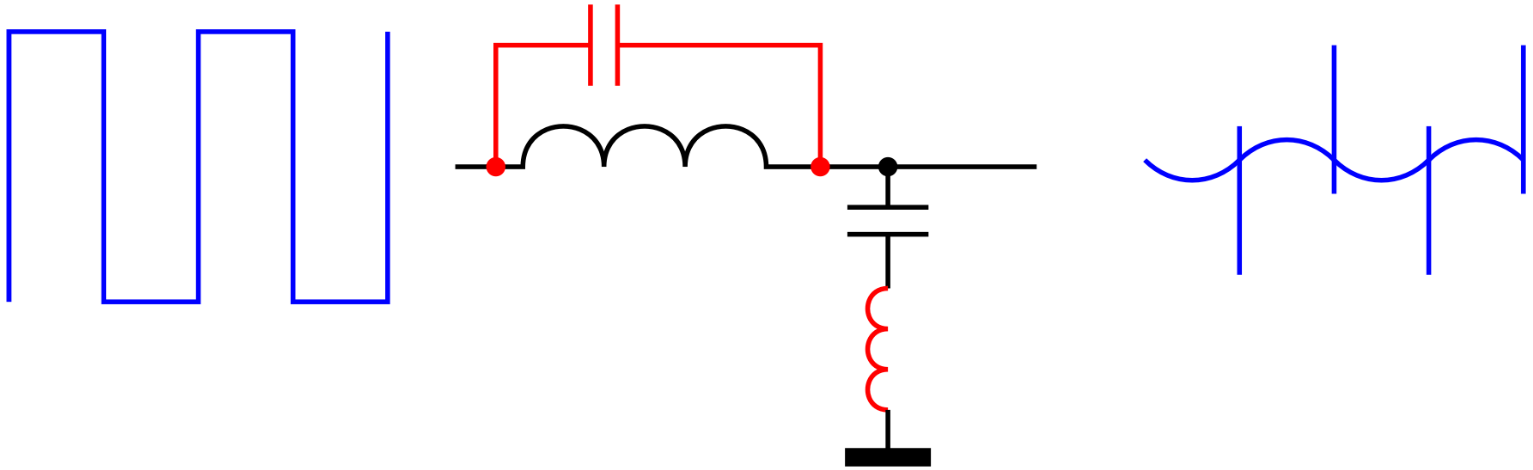
Half bridge vs Full Bridge, Class AD vs BD

- Half-bridge
  - Can’t cancel either CM or DM
  - Common-mode is half of differential mode
- AD
  - Common-mode voltage theoretically 0
  - Differential mode same as half bridge
- BD
  - Differential mode cancels at low modulation...
    ...but that was not really a problem anyway.
  - Common-mode voltage same as half bridge

Class D and EMI

High-Frequency EMI: Leaking switching transients

- Theoretical modeling is useless.
  - Capacitors become inductive
  - Inductors become capacitive
  - PCB becomes jumble of L's and C's.
- No tricks. Only good hardware design helps.
- Direct EMI problem under all circumstances.



Class D and EMI

Sensitive item 1: The capacitor.

- Myth of the “Low Inductance Capacitor”.
  (An Audiophile Favourite)
  - All modern film caps have sprayed end contacts.
  - Inductance is determined by geometry only (mostly size).

Bad.

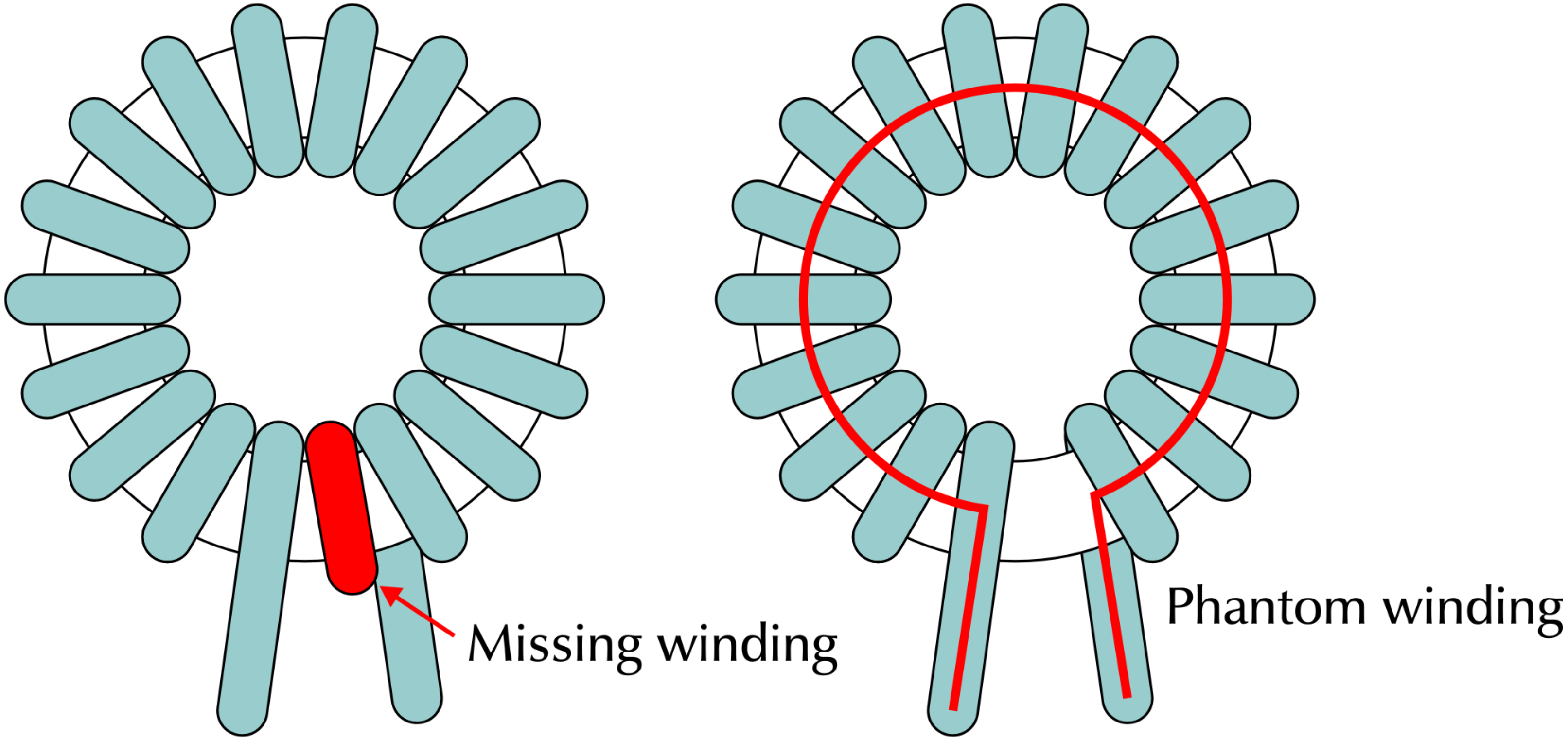
Good.

Period.

Class D and EMI

Sensitive item 2: The inductor.

- Stray fields out of toroids

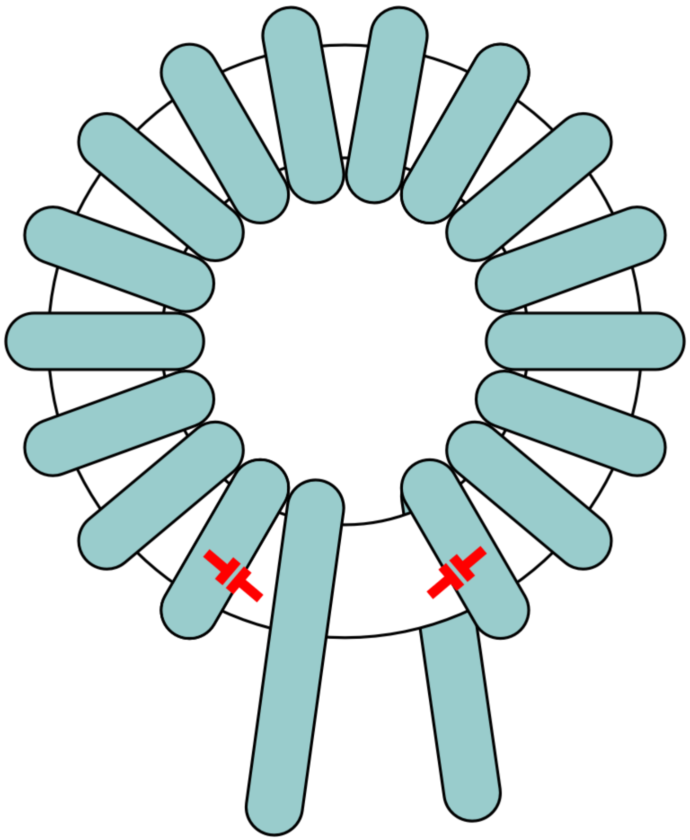


- Upright mounted toroids are worst.

Class D and EMI

Sensitive item 2: The inductor.

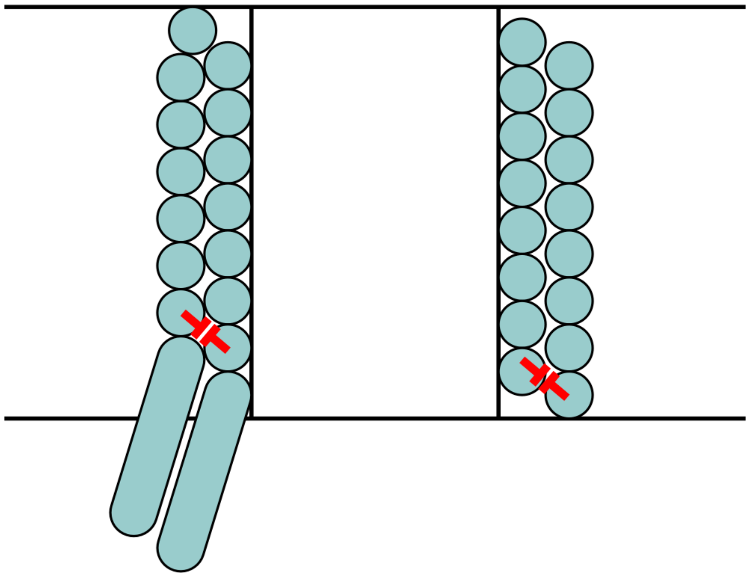
- Beware of indirect Capacitive Coupling through Core
  - Tight windings are better magnetically but worse electrostatically.
  - No external electrostatic shield: Capacitive coupling to chassis etc. can get significant.
- Toroids are not always optimal



Class D and EMI

Sensitive item 2: The inductor.

- Ferrite inductors: avoid direct capacitive coupling between windings

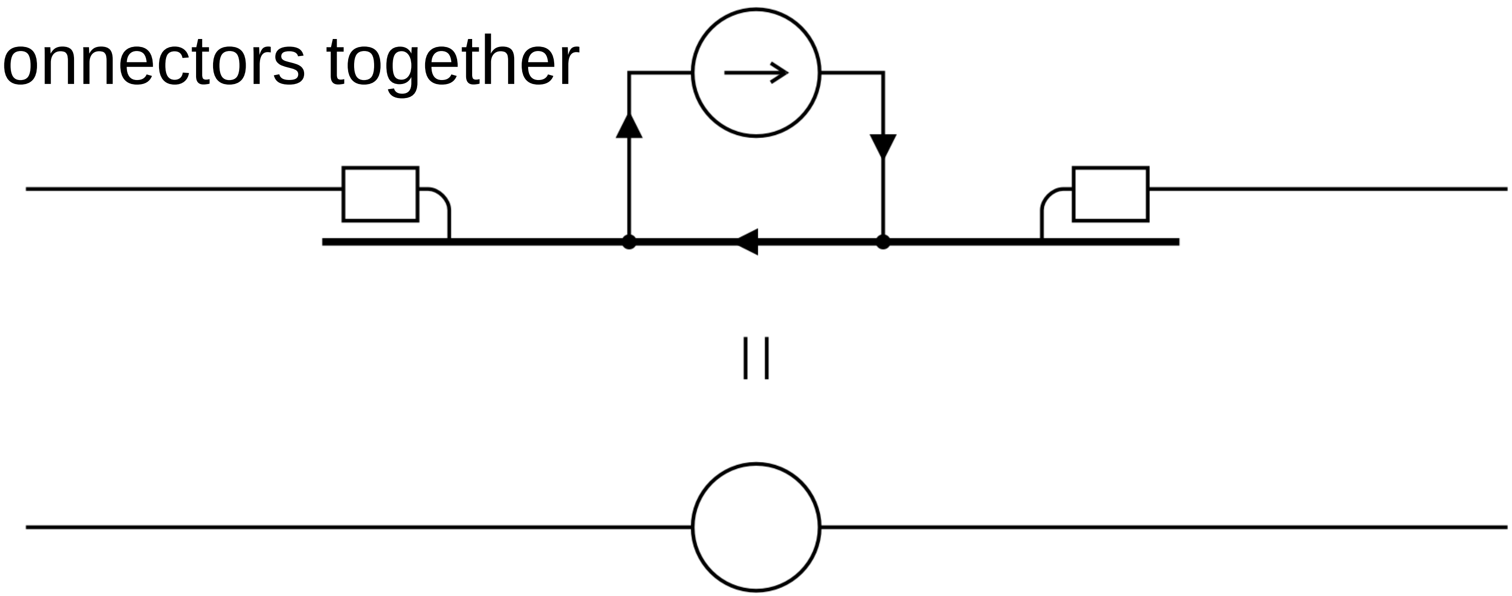


  - “Hot” end sees “Cold” end
  - 2 layers is worst case situation
  - 1 layer is best

Class D and EMI

Sensitive item 3: The PCB layout.

- Contiguous ground plane
- Keep connectors together



- Avoid capacitive coupling to external parts
- Minimize loop area (≠short traces)

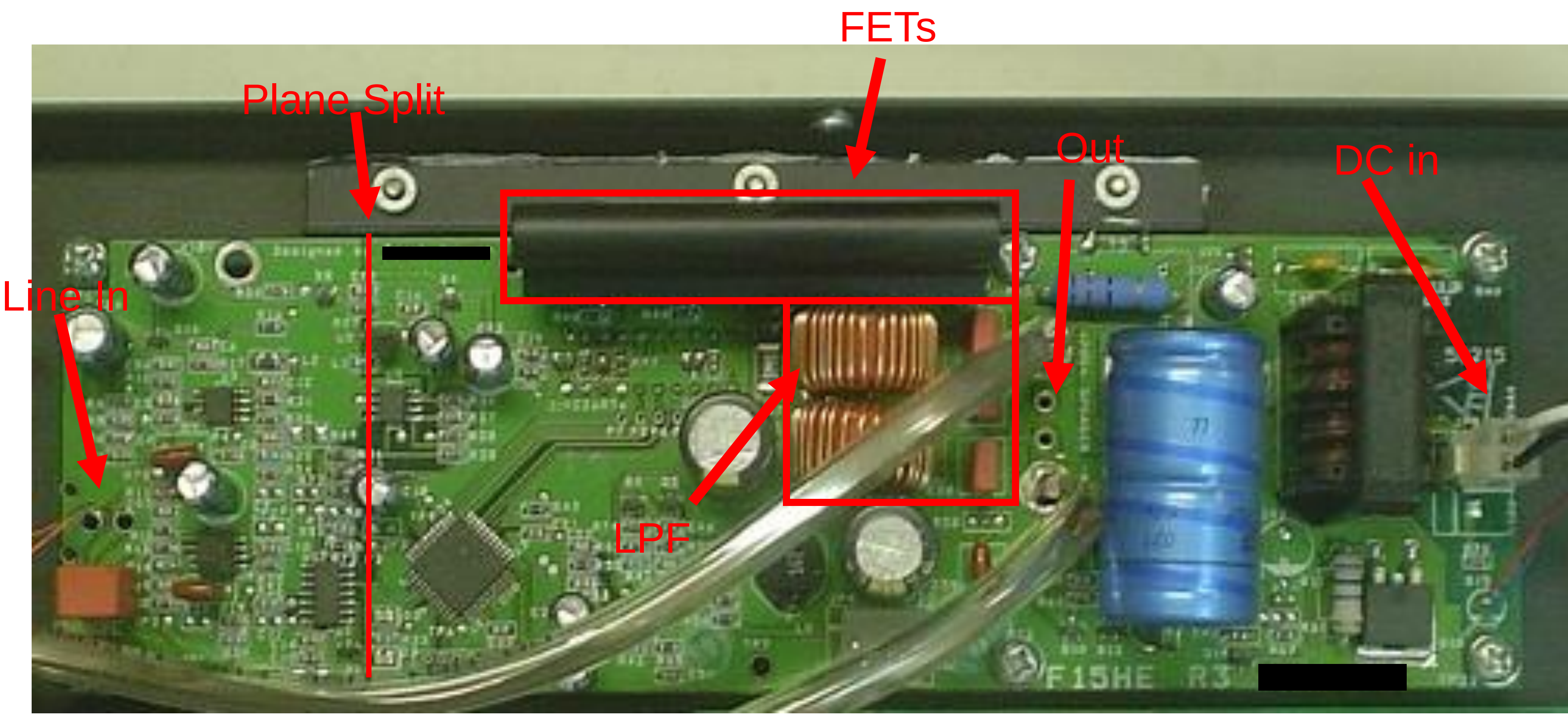
Class D and EMI

Checking for EMI without Spectrum Analyser

- Just probe around the external connections with a scope!!!
- If you see rubbish, there is rubbish
- The higher the frequency, the more you should worry

Class D and EMI

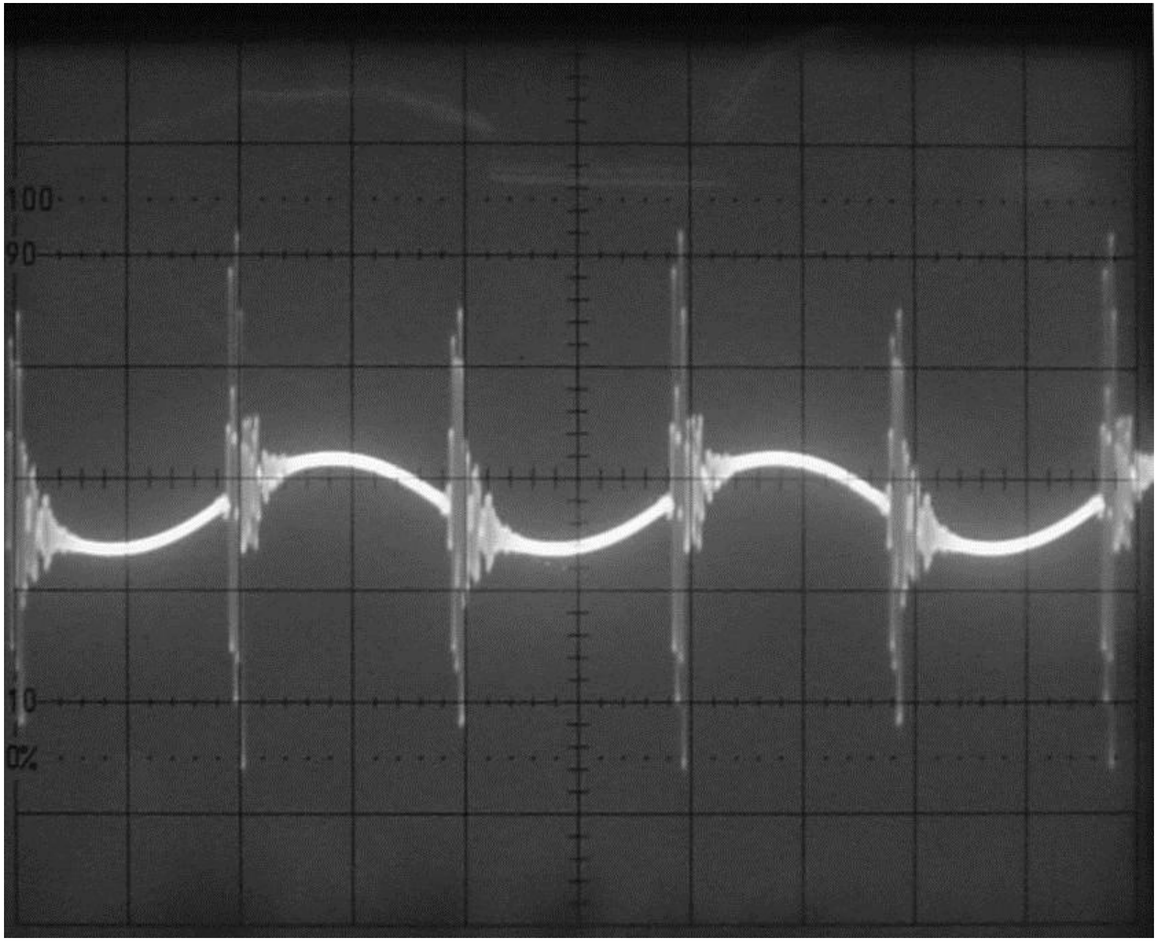
Example: Amplifier A, rated 160W



Class D and EMI

Amplifier A, one output line

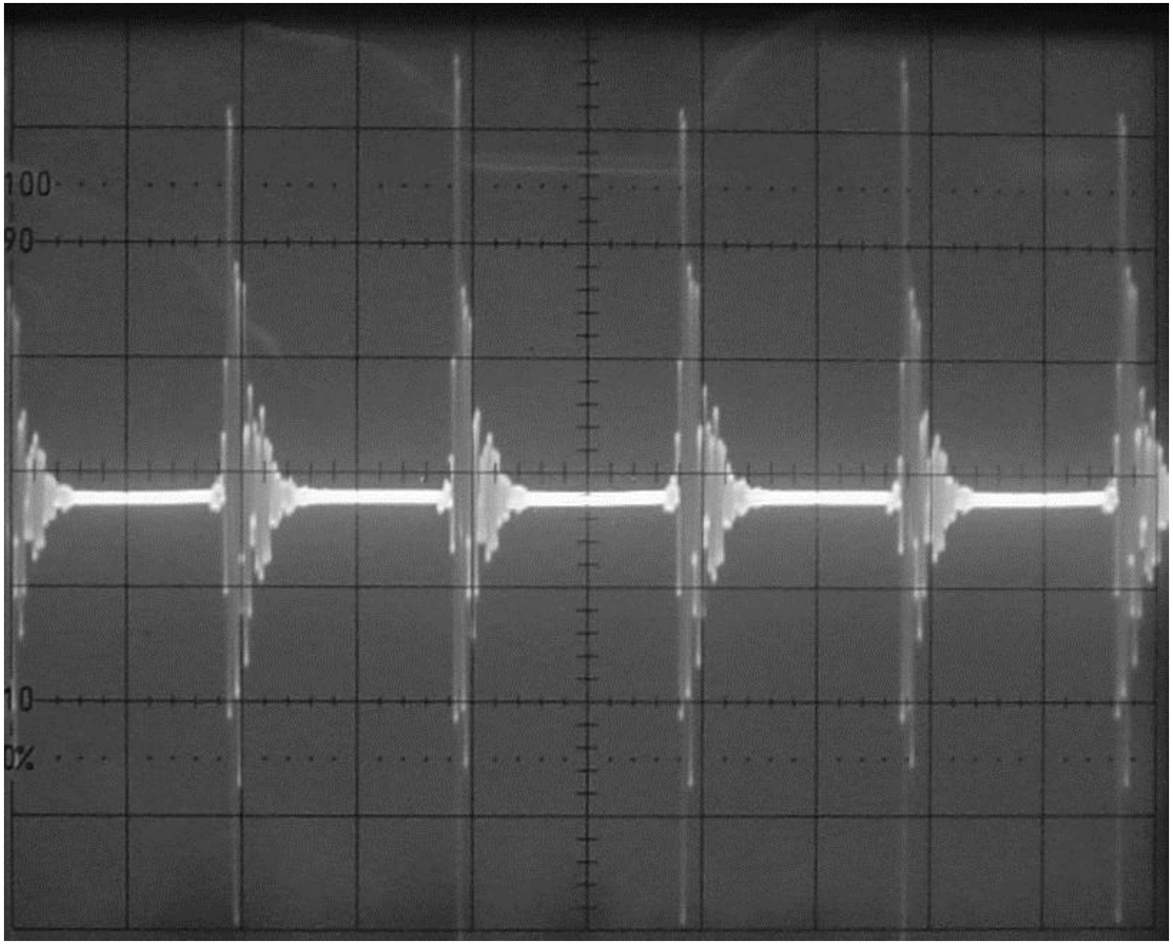
- 1V/div. Probe clip at RCA ground.



Class D and EMI

Amplifier A, common mode

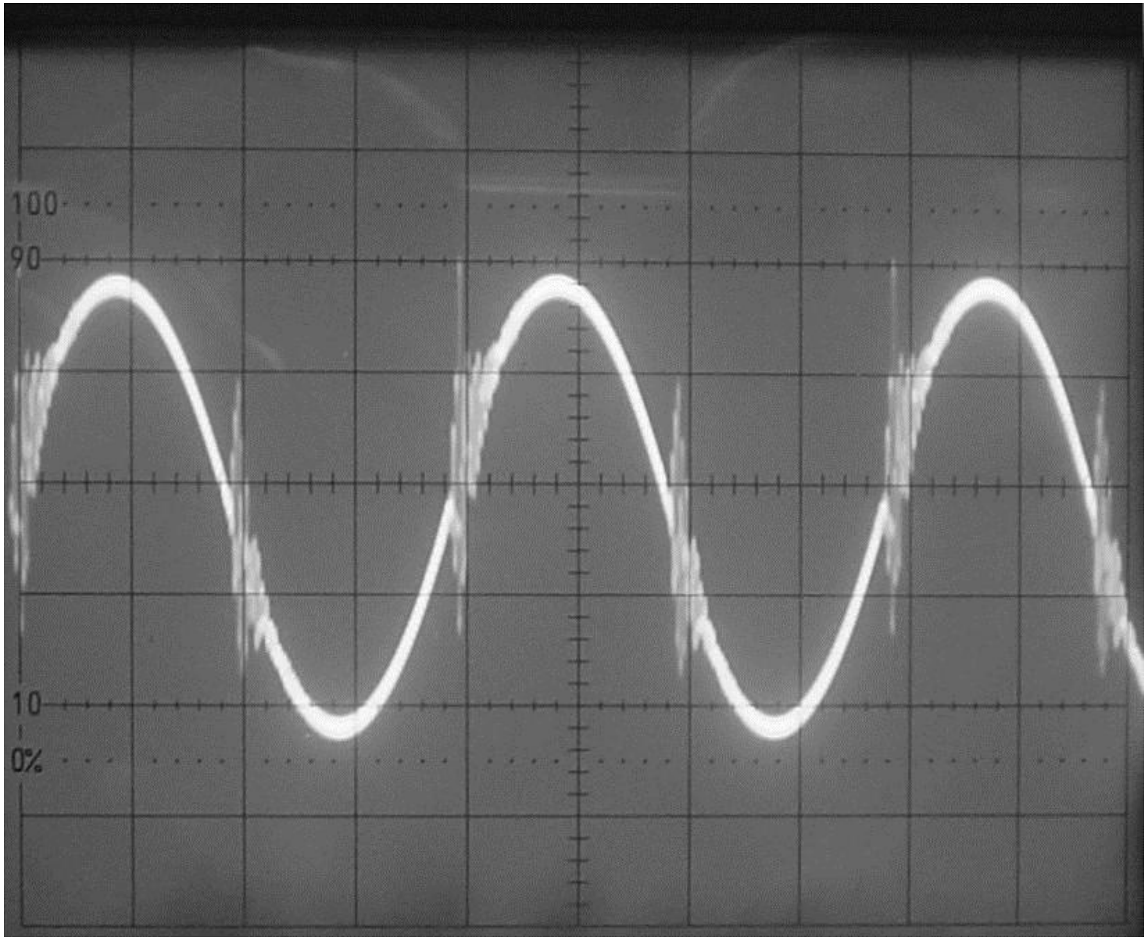
- 500mV/div. Amp is claimed to pass FCC???



Class D and EMI

Amplifier A, differential mode

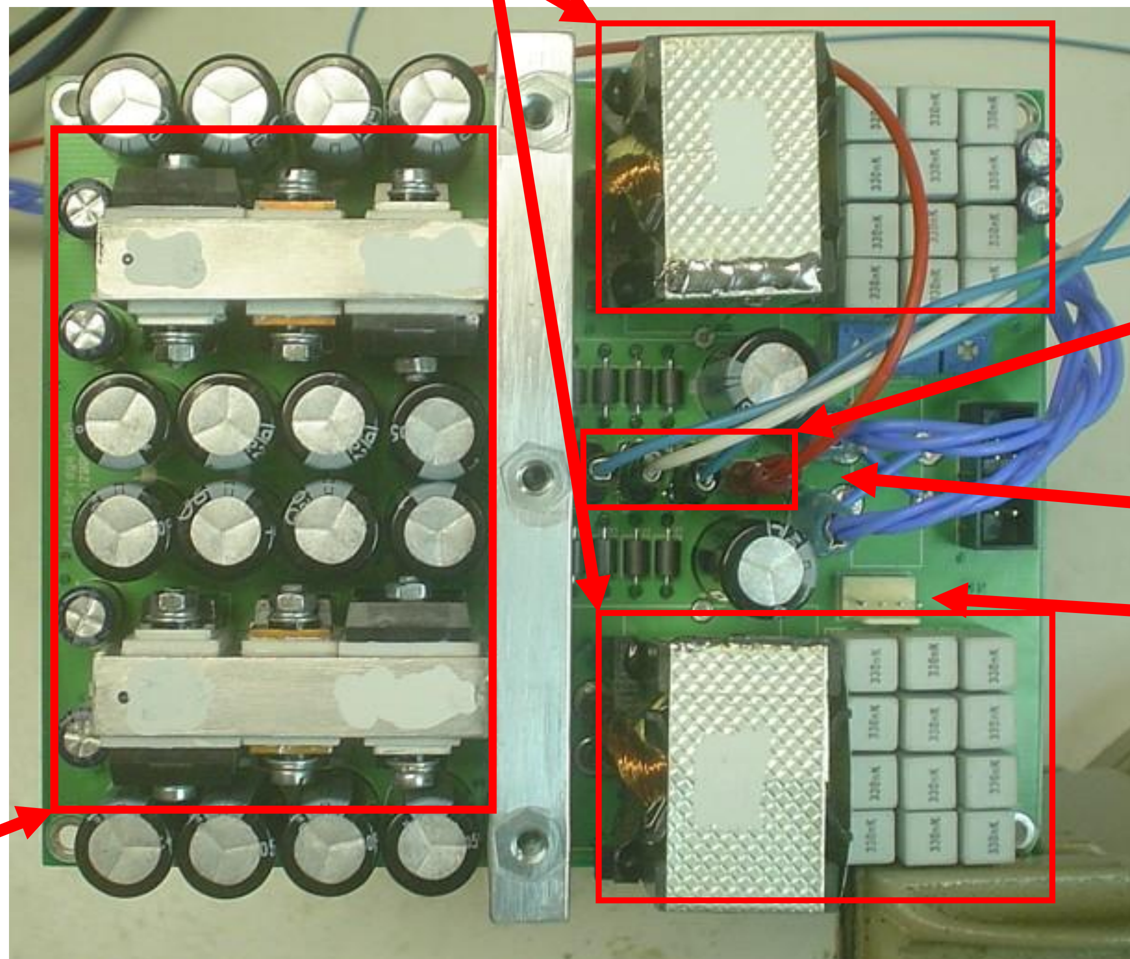
- 500mV/div. Note: relatively clean.



Class D and EMI

Example: Amplifier B, rated 2kW

LPF



DC in

Out

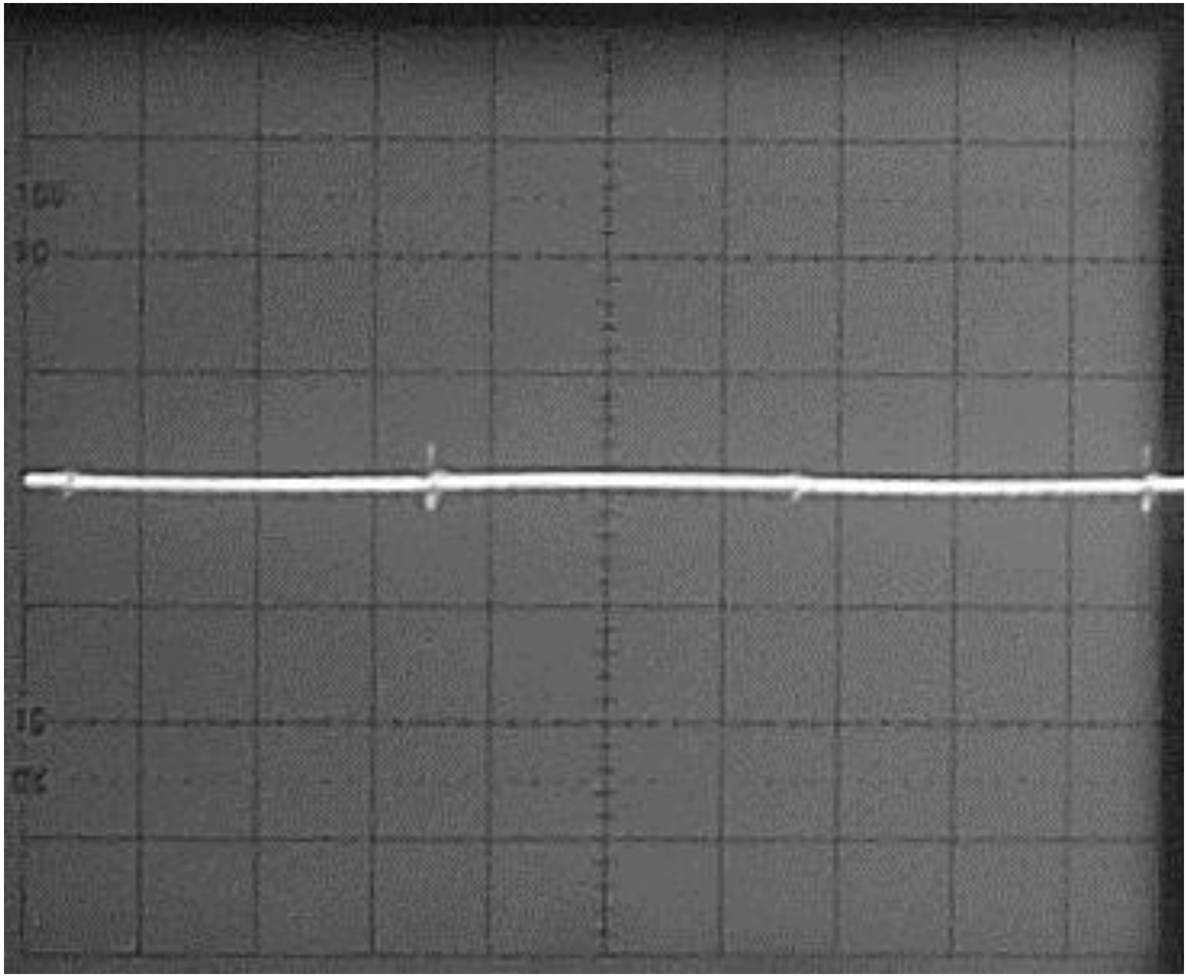
Line in

Power stage

Class D and EMI

Amplifier B, common mode

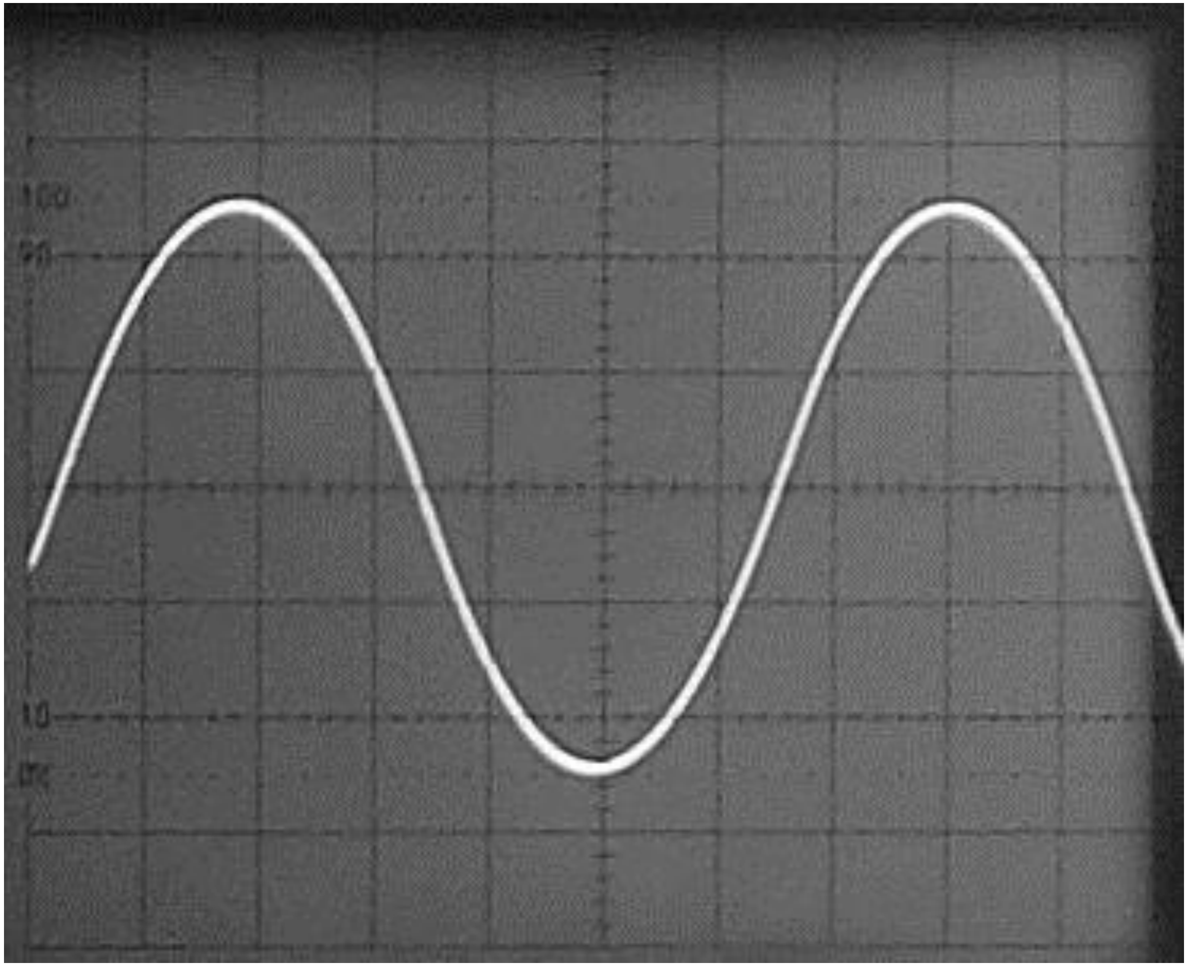
- 250mV/div. Probe clip at power GND faston tab



Class D and EMI

Amplifier B, differential mode

- 500mV/div.



Class D and EMI

Class D EMI is no mystery

- Eyeballing components and PCB gives good indication
- Invest in an analogue scope
- Don’t bother EMC testing if the scope pic isn’t squeaky clean

Next on: SMPS

Specifying SMPS for Audio

The complaint

- “I need a 2kW amp to do what a 1kW amp would do in the old days”
- “It sounds great with some sources and sux with others”

Specifying SMPS for Audio

Power Handling of COTS SMPS

- Protection limit = Peak Rating = DC rating
- Thermal design for rated output
- Protection = constant current, foldback or stop

Specifying SMPS for Audio

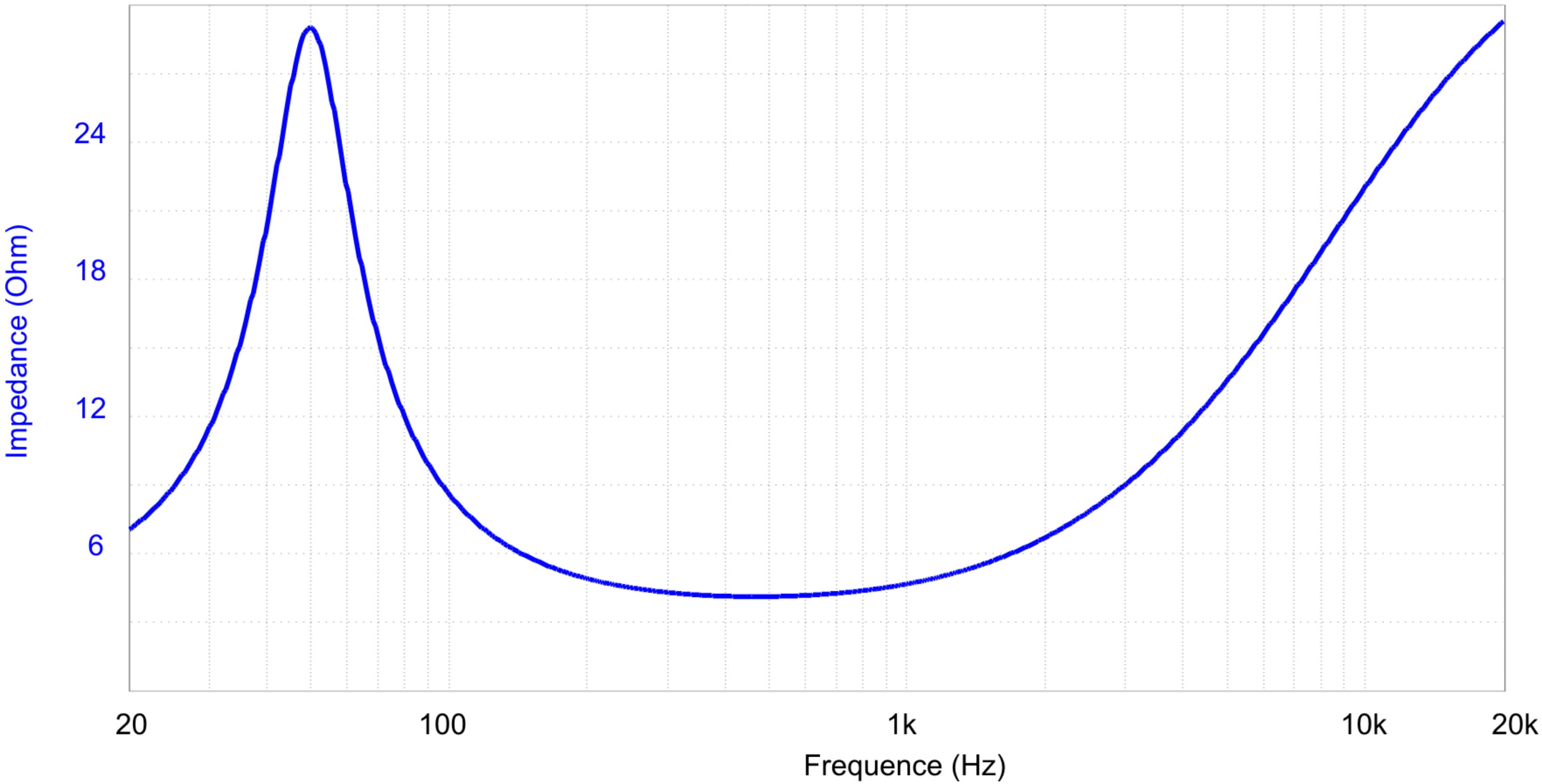
Power demand of resistive load

- Peak current/voltage = 1.414x RMS
- Peak power = 2x average (“RMS”) power

Specifying SMPS for Audio

Power demand of reactive load

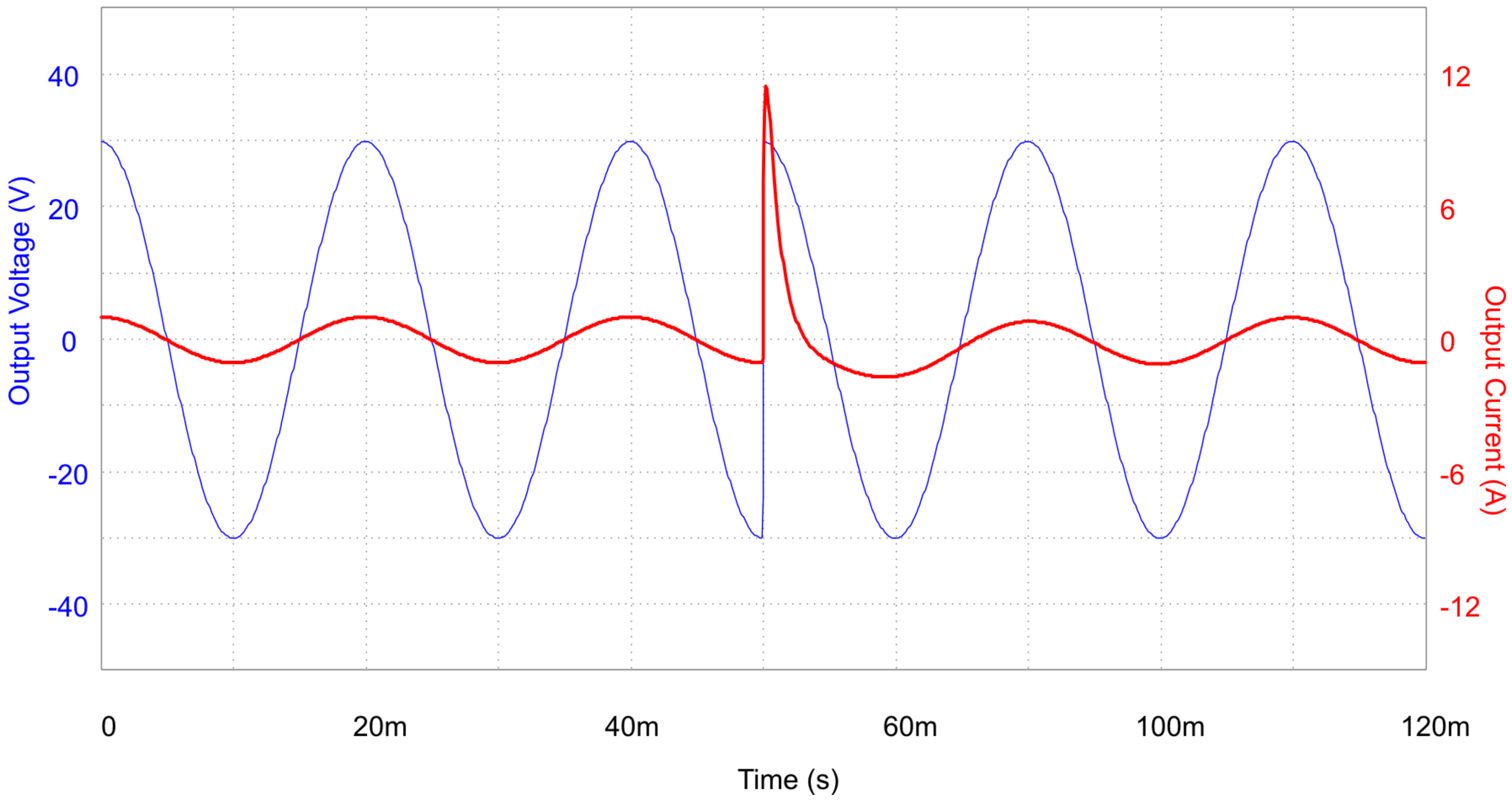
- Example impedance plot



Specifying SMPS for Audio

Power demand of reactive load

- Worst case current pulse



Specifying SMPS for Audio

Power demand of reactive load

- Maximum current pulse = 2x peak current in DC resistance!

Reactive or resistive:

- SMPS rating = amplifier rating is inadequate

Specifying SMPS for Audio

Practical set of requirements

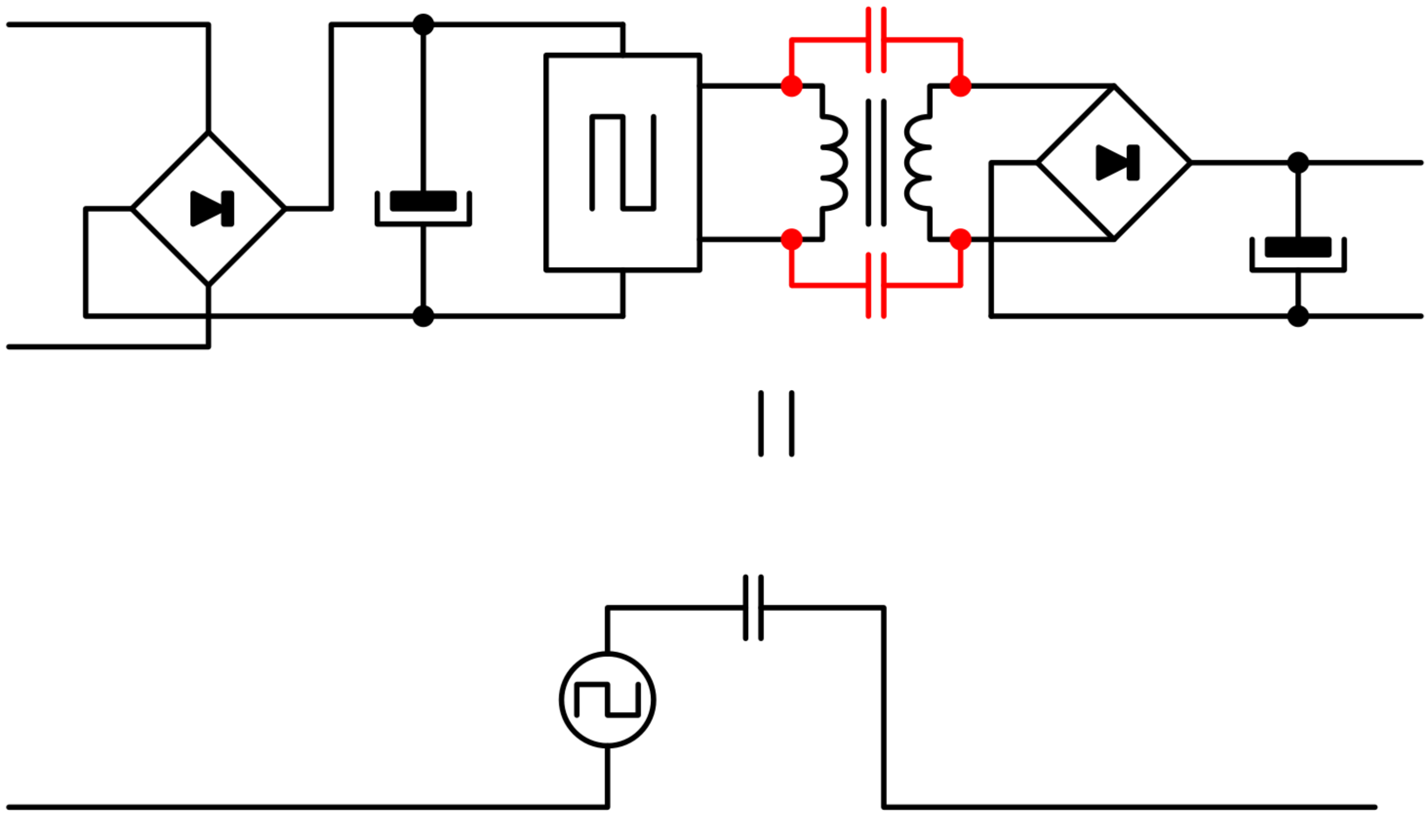
- Thermal design
  - 1/8Pr indefinitely (suggest 1/3Pr for pro)
  - Pr continuous for 5 minutes (IHF rating)

- Protection
  - Constant-Power at 2x Pr

Specifying SMPS for Audio

EMI: Injected mains current

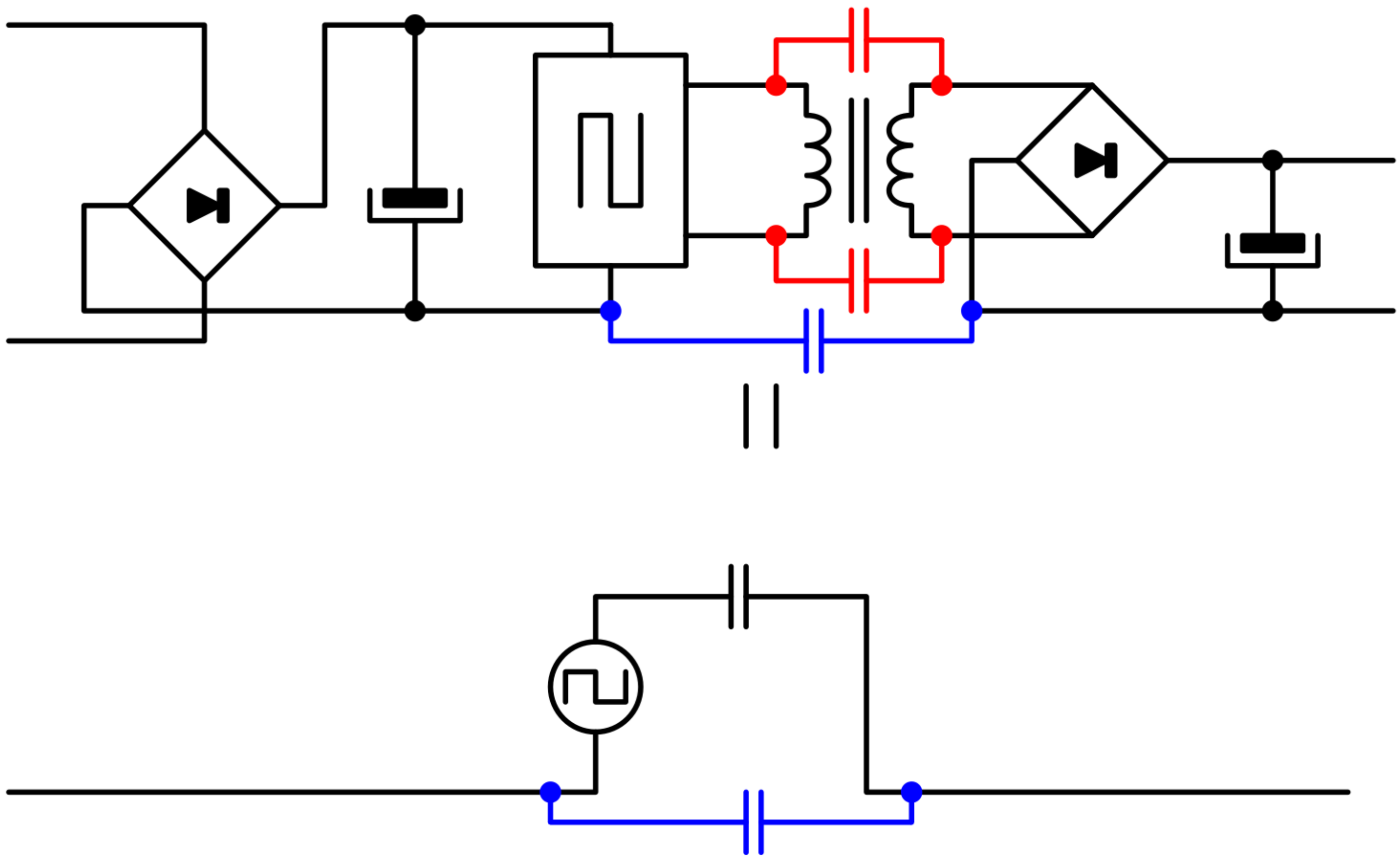
- Getting clean rails is easy (differential mode)
- Getting low CM noise is harder



Specifying SMPS for Audio

Y cap reduces CM noise voltage

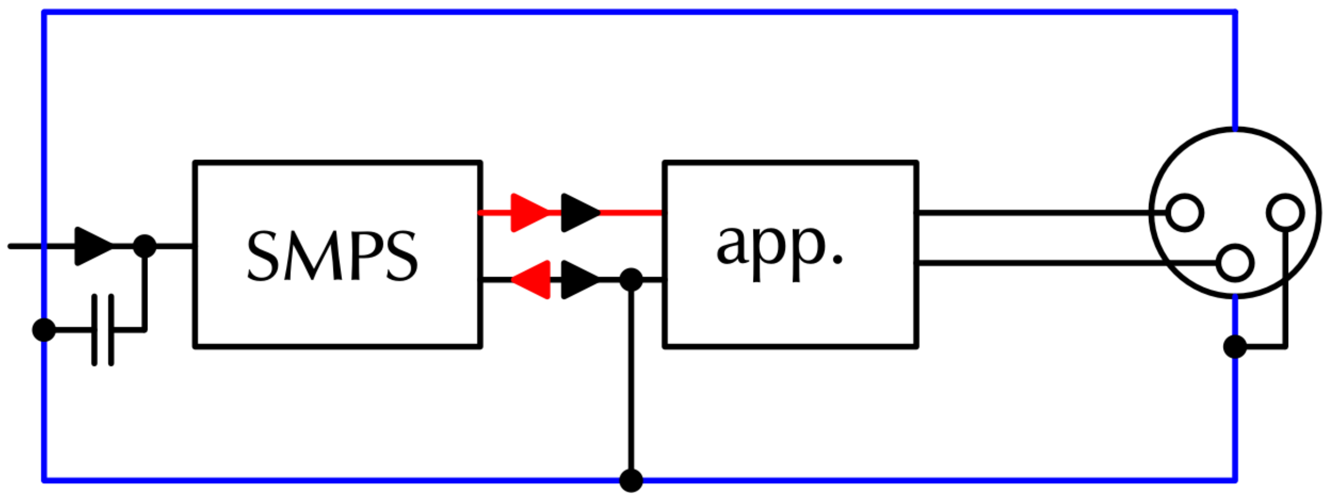
- CM current reduction is indirect
- Increases coupling of mains-borne noise



Specifying SMPS for Audio

Leakage current exacerbates Pin-1 problem

- Current enters circuit ground.
- Circuit ground current includes return
  - You can’t just disconnect it (AES48 style).
  - Requires chassis connection at PSU output.
  - Creates additional layout challenges



Specifying SMPS for Audio

Additional EMI requirements for audio SMPS

- Common mode voltage/current noise
- Primary-to-secondary impedance

Next on: Be Careful...

“ID” in Audio: Successful Co-Development

Nightmare story #1

- Customer wanted 100W class D solution
- Subcontracter had a fully working design that fit well
- C insisted on using “metal core” boards (hybrid)
- S made list of 8 technical issues that would definitely kill the project.
- C said all problems would get resolved
- All problems materialised, few got solved
- Project failed. C’s project manager resigned

What went wrong?

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Customer specified how, not what

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Customer specified how, not what

Customer could not justifiably make this judgment!

"ID" in Audio: Successful Co-Development

Nightmare Story #2

- C wants high-end class D amplifier...
  - that does not use feedback
  - that processes DSD...
  - ...with no alteration
- S produces a highly complex but working prototype
- C thanks S and starts product development cycle.
  - Layout gets changed
  - Clock distribution gets changed

  ...
- Project fails as C can't debug a buck regulator circuit...

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C specifies how, not what

C overestimates self / underestimates problem

“ID” in Audio: Successful Co-Development

Outset

- Customer has needs
- Subcontractor has capabilities

Potential problem

- <u>Perceived</u> overlap of competences
  (Real overlap of actual competences is not a problem)

Failure modes

- Customer overestimates what they can do themselves
- Customer specifies implementation details
- Subcontractor meddles in customer’s work.

“ID” in Audio: Successful Co-Development

Success Story

- C wants DSP/amplifier electronics for loudspeaker
- C and S agree “black box” spec
- S designs electronics
- C designs acoustics and filters
  - Politely refuses S’ spontaneous input (“that’s our problem”)
- Both parties finish in time, product is well received.

“ID” in Audio: Successful Co-Development

Critical steps for the Subcontractor:

- Agree and insist on responsibilities
- Avoid inept customers
- Refuse paper-only gigs
- Charge for spec changes once the design is underway

“ID” in Audio: Successful Co-Development

Critical steps for the Customer:

- Hire expertise, accept expertise.
- Write “black box” performance spec
  - Performance is judged with the box closed and the power on.
  - “Subjective sound quality” is a black box spec too.
  - Type of circuit or parts is <u>not</u> a performance spec.

“ID” in Audio: Successful Co-Development

The Two Roads

The Road To Hell:

Specify the Design, Accept the Performance.

The Road To Heaven:

Specify the Performance, Accept the Design.

Thank you!

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