Perception of temporal response and resolution in the time domain

Workshop & Panel Discussion

142nd AES Convention, Berlin

20th May 2017

"The bottom end !"

a.k.a.

Loudspeaker time-domain response from the low frequency perspective

Michael J Turner
Nidec Motor Corporation

Nidec
All for dreams

1. Elementary recap

(Regarding enclosed electro-dynamic loudspeaker)

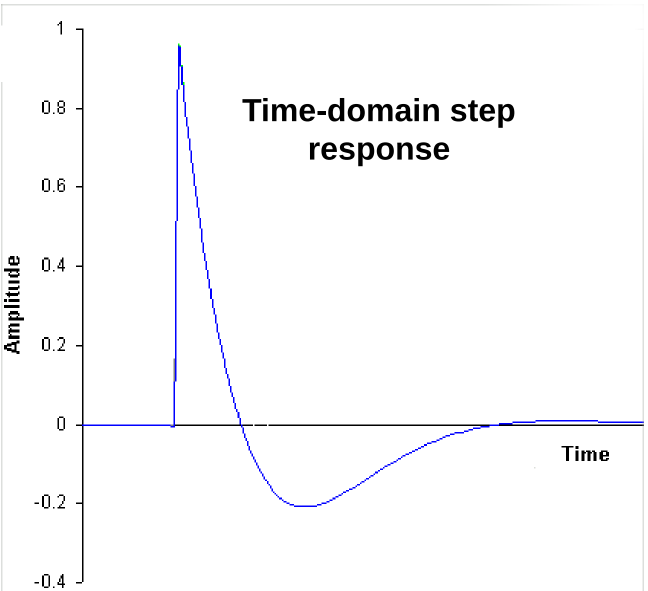
Long-established LF design paradigms

- Enclosure selection, e.g.
  - Sealed box (variants)
    - acoustic suspension (Villchur), “isobaric” …
  - Transmission line (delay or acoustical termination)
  - Open baffle
  - Vented (ported)
  - Passive radiator

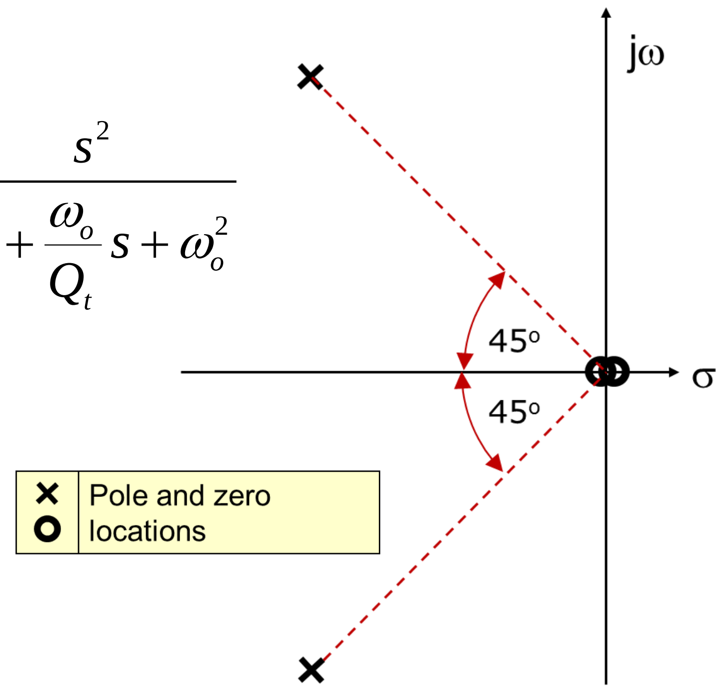
  } “reflex” systems (focus today)
- High-pass filter perspective (Thiele, Small)
  - Poles of driver (@ electrical & pneumatic loading)
  - Poles due to other acoustical components
- Infinitely many alignments possible

Second-order Butterworth (typical sealed box)

- Maximally-flat *frequency* response
- Slightly under-damped *time* response ($\zeta = Q_t = 0.707$)
- 2nd-order energy storage and exchange between the effective mass and spring stiffness of enclosed driver, damped by mechanical and (mainly) electrical losses



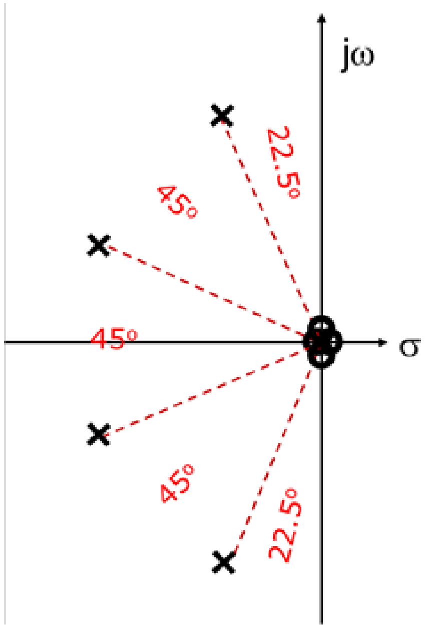
$$T(s) = \frac{s^2}{s^2 + \frac{\omega_o}{Q_t} s + \omega_o^2}$$



Fourth-order Butterworth (B4) (typical vented box)

- Maximally-flat *frequency* response
- More complex under-damped *time* response due to 4th-order energy storage and exchange between
  - Mass and stiffness of enclosed driver, as before
  - Mass of air in port and pneumatic stiffness of air in box (Helmholtz resonance)

$$T(s) = \frac{s^4}{\left[s^2 + \left(\frac{\omega_c}{0.541}\right)s + \omega_t^2\right] \cdot \left[s^2 + \left(\frac{\omega_c}{1.306}\right)s + \omega_c^2\right]}$$

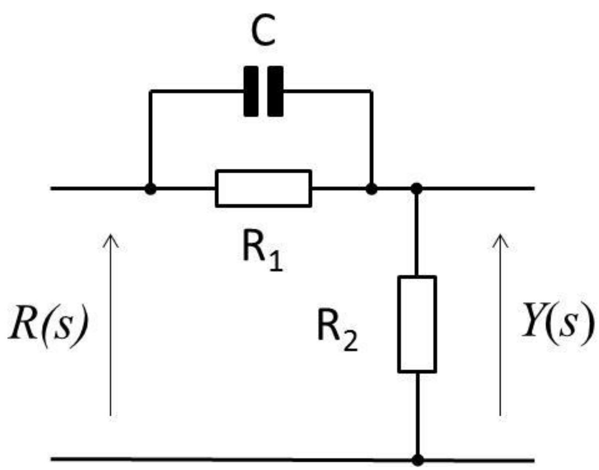


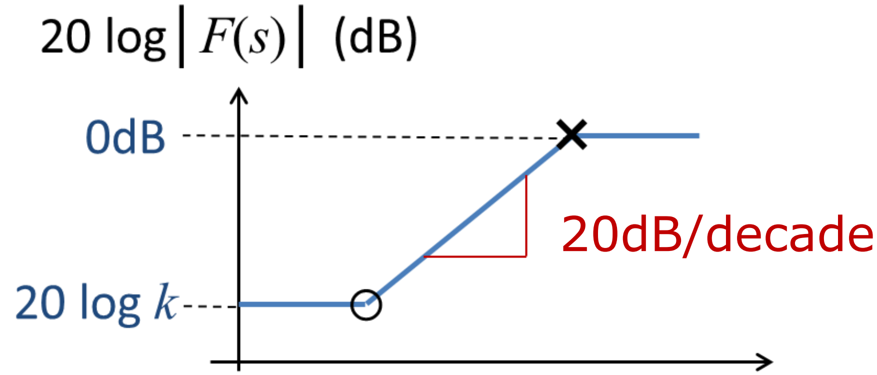
"Group delay" perspective

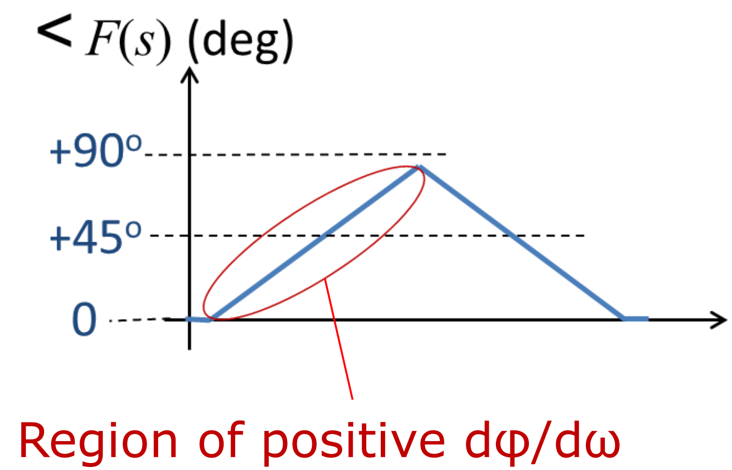
- Complex signals will be undistorted if time delay is same at all frequencies of interest
- Constant delay (across pass-band) means phase (lag) must be proportional to frequency:

  $\Phi = -\omega * \Delta t$ i.e. $-d\Phi/d\omega = \Delta t$
- Non-constant group delay with frequency → "time-smearing" of signal components
- Caution: group delay really only meaningful for the steady state response to continuous signals
- For example: regions of positive $d\Phi/d\omega$ in the phase response *don't* mean that a filter (or loudspeaker, whatever) is a time machine!

Example:
simple lead (shelving HP) network

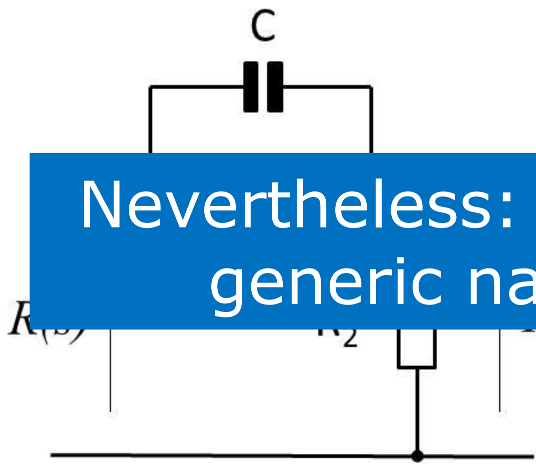


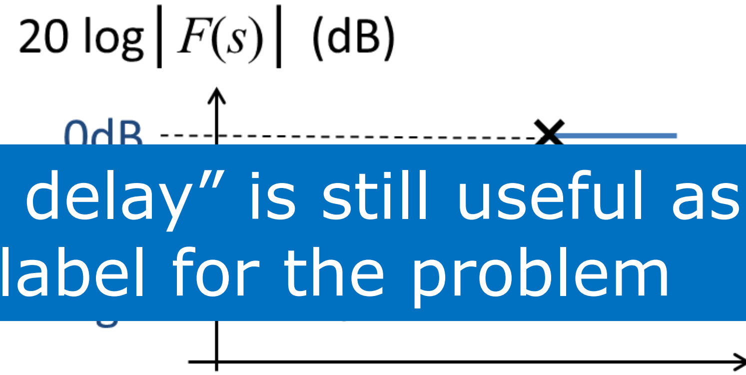




- Network acts as passive differentiator for frequencies where its phase lead approaches 90°
- For continuous sinusoids it appears to be "looking ahead" ¼ of a cycle
- But it's still a causal network!

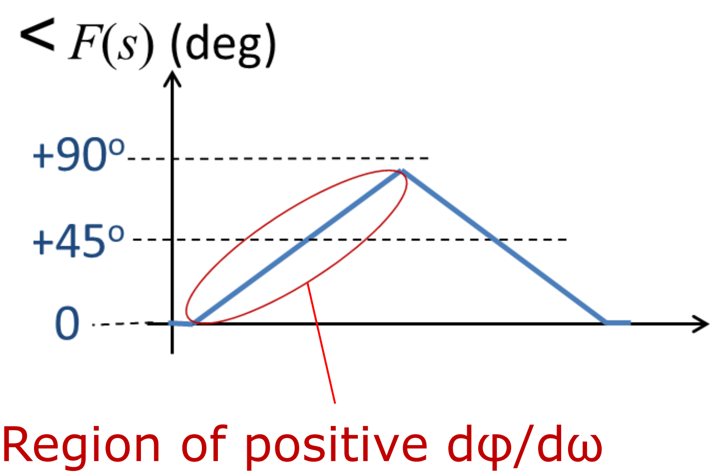
Example: simple lead (shelving HP) network



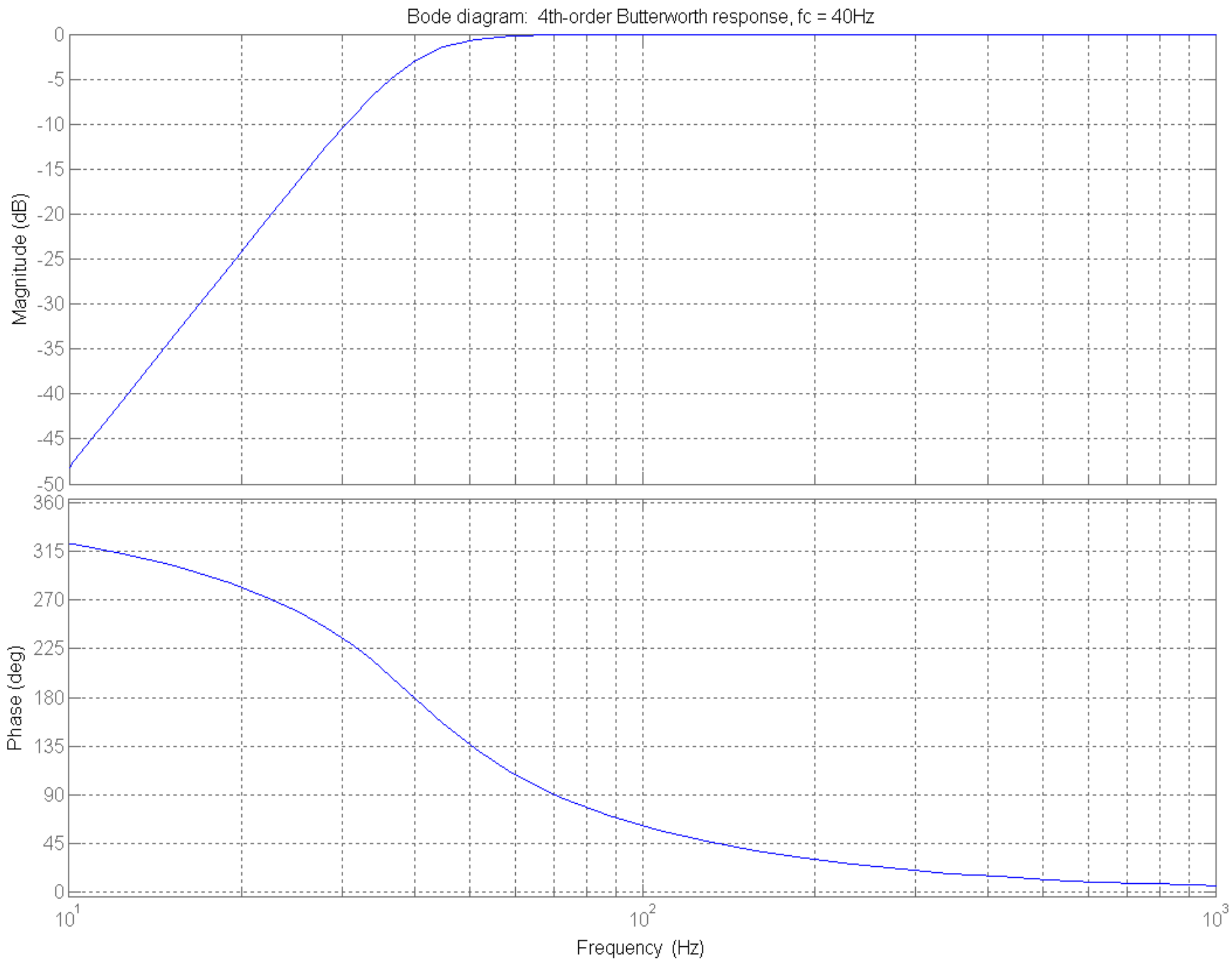


Nevertheless: "group delay" is still useful as a generic name or label for the problem

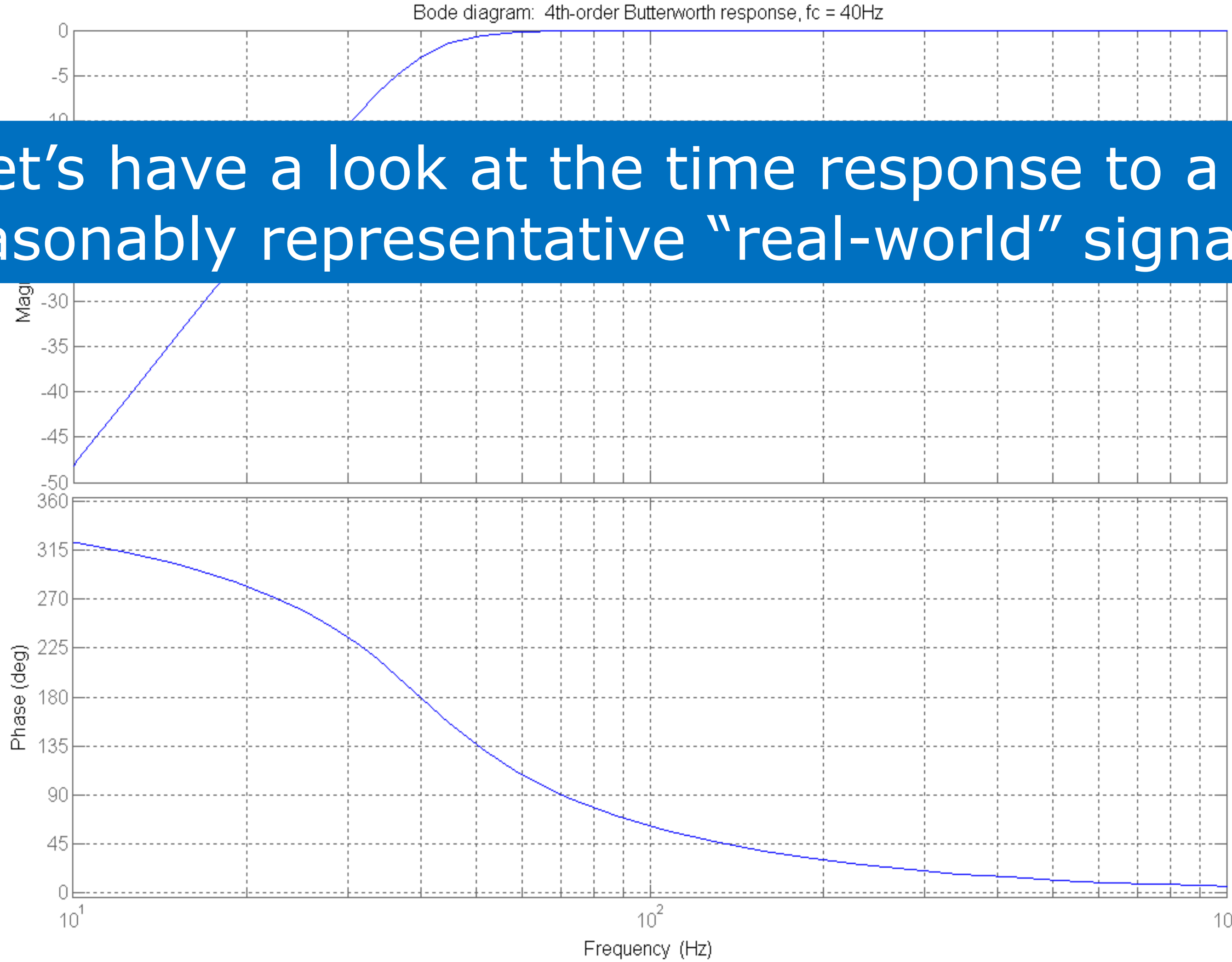
- Network acts as passive differentiator for frequencies where its phase lead approaches 90°
- For continuous sinusoids it appears to be "looking ahead" ¼ of a cycle
- But it's still a causal network!



Back to the B4 high-pass example: Frequency response (f_c = 40Hz)

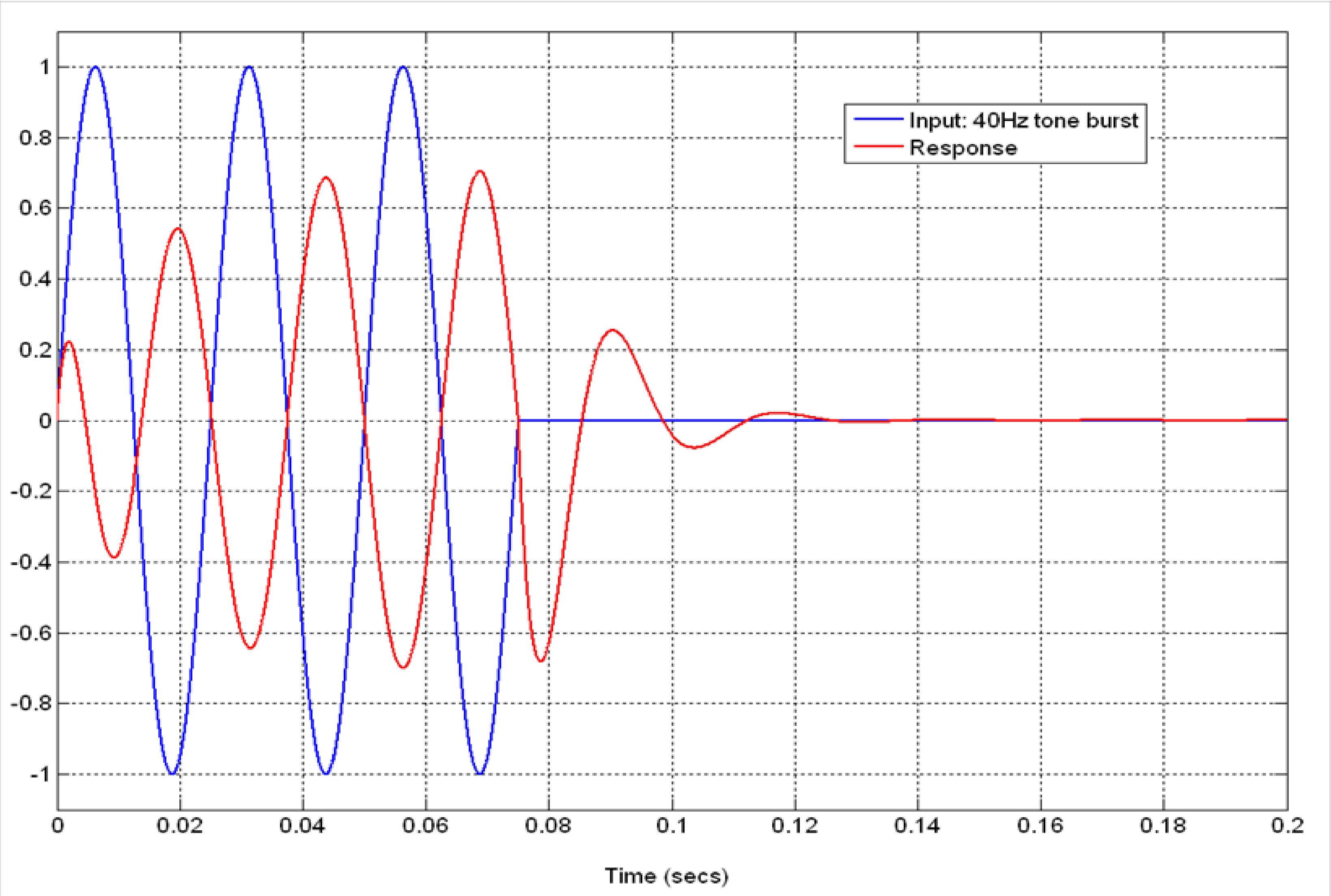


Back to the B4 high-pass example: Frequency response (f_c = 40Hz)

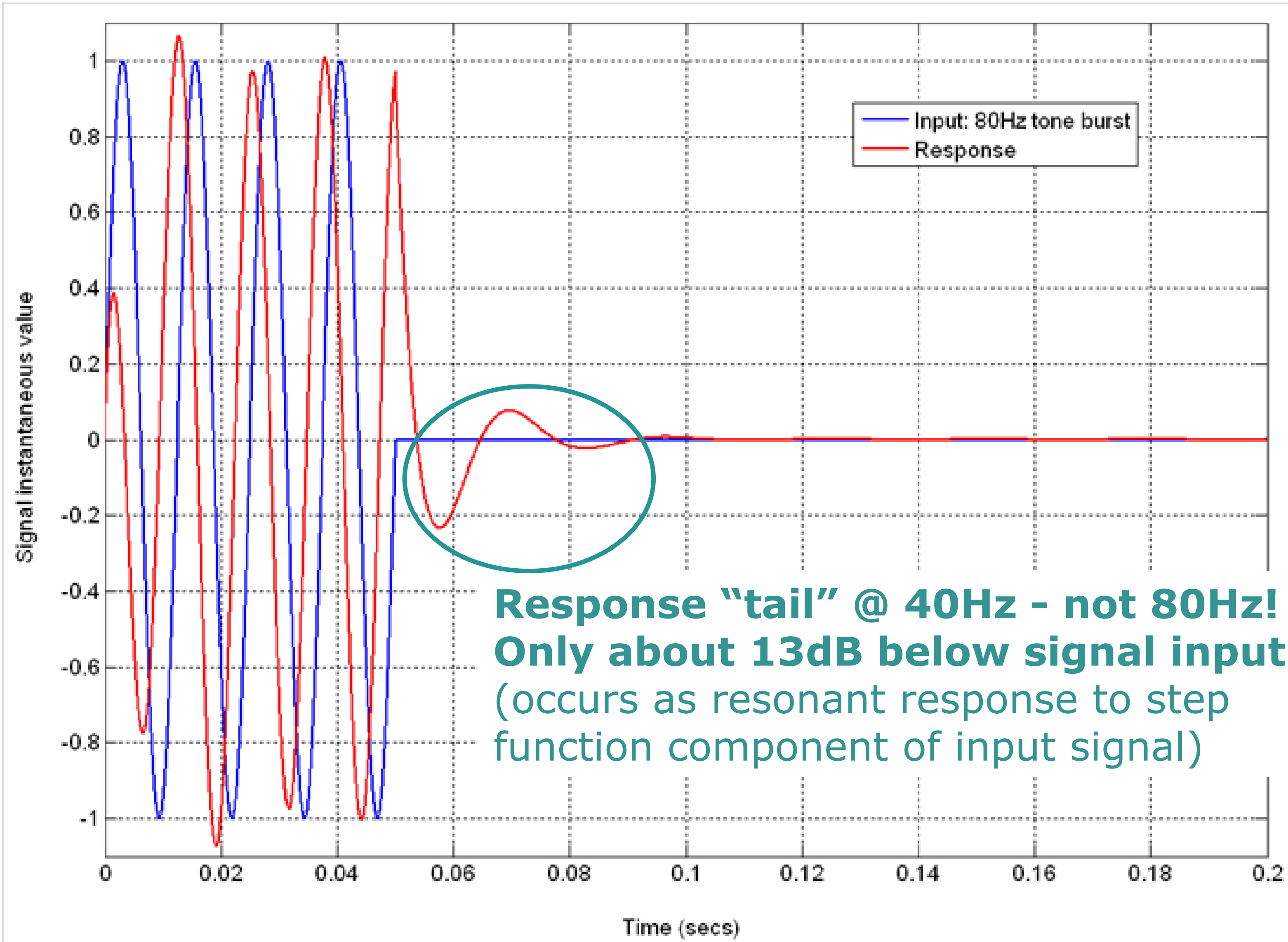


Let's have a look at the time response to a reasonably representative "real-world" signal

B4 (f_c = 40Hz) tone burst response: Input = 3 cycles @ 40Hz



B4 (f_c = 40Hz) tone burst response
Input = 4 cycles @ 80Hz



Response "tail" @ 40Hz - not 80Hz!
Only about 13dB below signal input
(occurs as resonant response to step function component of input signal)

2. Does it matter?

"What the papers say"

Points of view & experiences

Widespread current practice

Fincham, JAES vol. 33, #6, June 1985 “The Subjective Importance of Uniform Group Delay at Low Frequencies”

- Concerned with the audibility of total LF group delay in reproduced programme…
  - …including microphones, signal chain, [analogue] recorder – as well as loudspeakers
- Listening tests using programme material from custom record/replay chain
- Speakers corrected using bi-quad equaliser to f_c = 5Hz (2nd-order target response)
- Notes subjective *reduction* of bass in corrected recording / reproducing chain, also that excessive voice coil excursion was [surprisingly] not a problem
- Concludes (nevertheless):

  A reduction in [replay chain] group delay is probably worthwhile only when the recorded material is itself also free from such distortion. The effects… …are quite subtle…”

Krauss, 88th AES Convention, Montreux, 1990 “Low Frequency Transient Response Problems in Vented Boxes”

- Notes Neville Thiele postulate: *“…transient behaviour not disturbing… …at least for the standard alignments suggested”.*
- Tone burst used to simulate drum / bass signals
- Filters to emulate B2, B4 and B6 alignments
- Listening panel auditioned these via electrostatic headphones (flat down to “very low frequency”)
- B2 alignment sounded similar to the test signals
- B4 and B6 alignments clearly changed the timbre of the test signals.
- Differences still clearly audible with fundamental frequency of the tone burst one octave above the filter corner frequency.

Bech, 109th AES Convention, L.A. 2000 "Subwoofer Requirements, Part II…"

- Comprehensive test regime included auditioning of both real speaker (anechoic conditions) and of emulated loudspeaker system via headphones
- 2nd, 4th and 6th order high-pass responses with 20Hz, 35Hz and 50Hz cut-off frequencies
- Summary conclusions:

  "…lower cut-off frequency has significant influence on the perceived level of lower and upper bass reproduction, independent of reproduction levels".

  "The filter order was not found to be of significant importance for the conditions investigated".

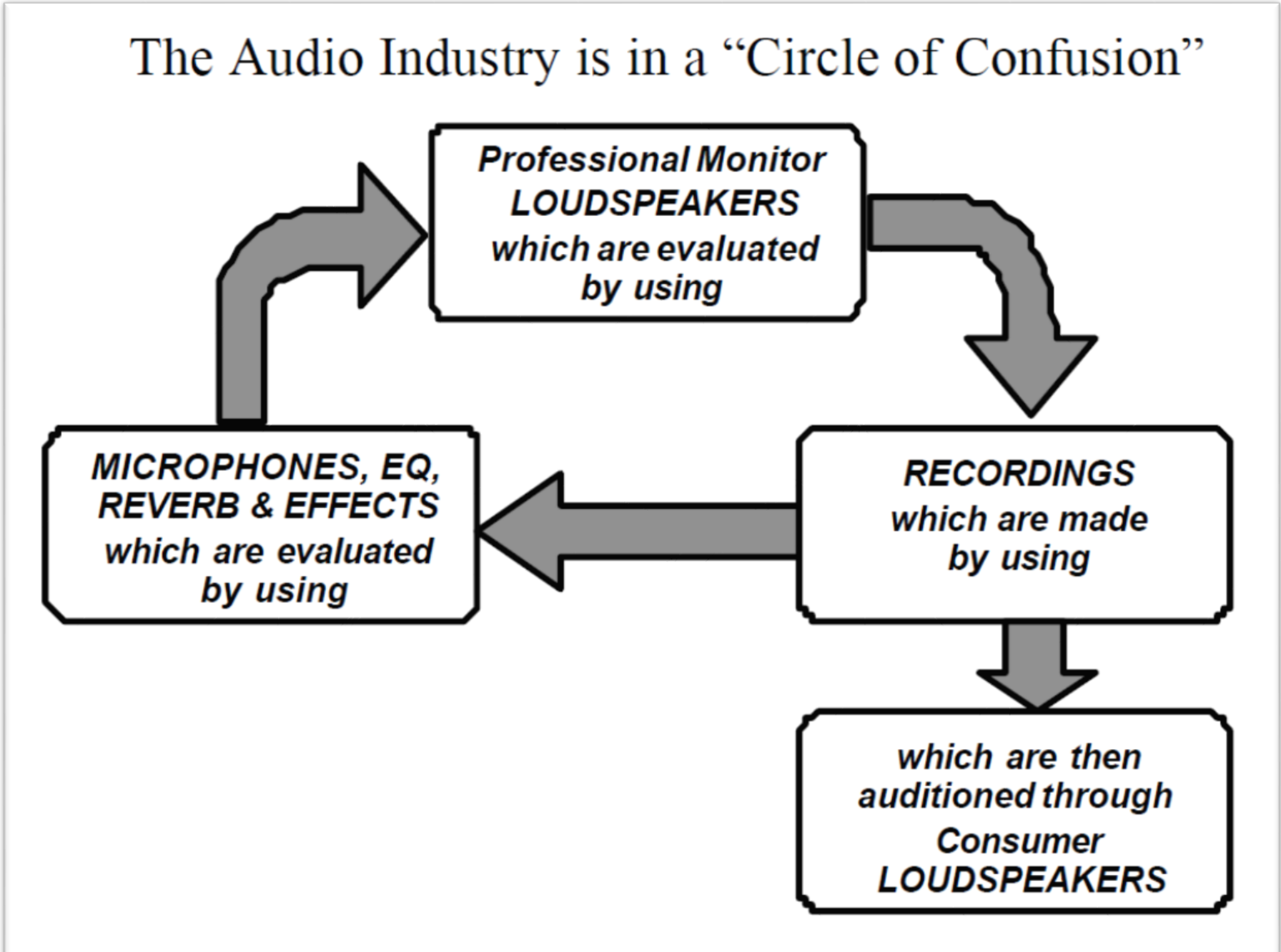
Some personal views & experiences

- Analogue broadcast signal path: a great many cascaded low-pass (and high-pass) responses
- My own pick-up cartridge (pre-amp) story
- Gradually correlated subjective preference with loudspeaker enclosure type: reflex seemed consistently inferior to sealed box…
  - LF signal content "out of time" with remainder
  - Hard to follow bass line – lack of clarity, difficulty to distinguish instruments at lowest frequencies
  - "Boom rather than bass"
- Experience of active system with extended (<20Hz) 1st-order response (not always "more bass"!)
- **My own hypothesis: LF phase response and consequent time behaviour has much bigger subjective impact than generally acknowledged**

Not to dismiss decades of industry experience… (but maybe to question it!)

- "Sweeping statements are always wrong!"
- Good and bad designs!
- Efficiency benefits of reflex designs
  - But electronics much cheaper now than historically
- Munich "High End" show: vast majority of speakers on show were reflex designs
- Big market for smaller speakers: "some" bass preferable to "none" (here I tend to agree)
- Nevertheless – if the LF time response of most loudspeakers is so bad (and it is!) then why is this deemed acceptable even in large/high end units?
- Perhaps many consumers actually [think they] like it: superficially impressive impact of "big bass"? Or at any rate are accustomed to it…

Floyd Toole: “Circle of confusion”



Is it really this bad in practice?

- Well, unfortunately, yes!
- Well-reviewed small vented speaker example

50Hz tone burst input

Acoustical output @ 1m

We *can* do things differently!

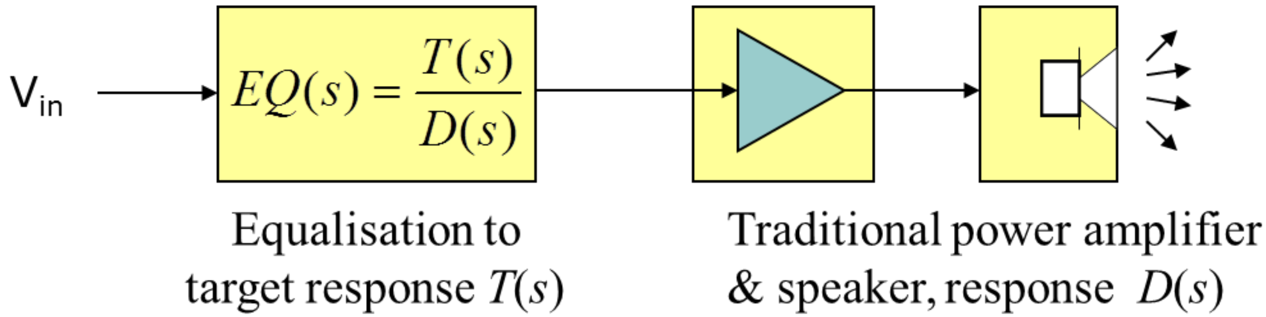
Paradigm shift needed: Consider electronics and loudspeaker together, *as a system*

Not the way much of the industry has historically operated [with notable exceptions!]

Design holistically - not as separate components

Matched equalisation

- Straightforward in principle
  - Dynamic loudspeaker motors exhibit “minimum-phase” behaviour at low frequencies
  - Enclosed drive unit “native” transfer function *D*(*s*)
  - Target system transfer function *T*(*s*)



- Then requisite equaliser transfer function is simply $EQ(s) = T(s) / D(s)$
  - Perfectly causal and realisable (but watch LF gain)
  - Accuracy depends on knowledge and stability of drive unit parameters

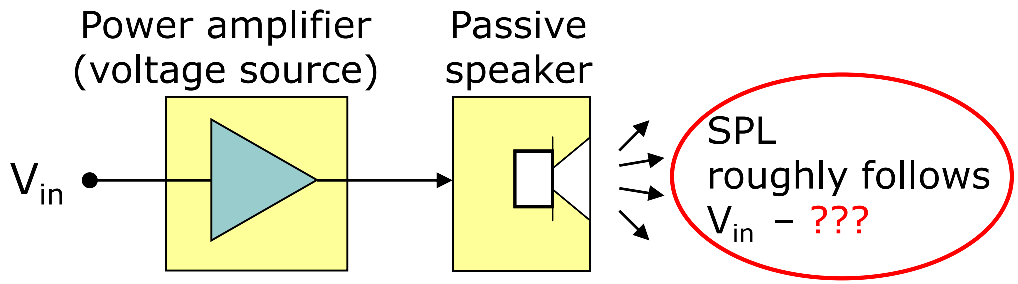
Matched equalisation

- Stiffness is notoriously ill-controlled and ill-defined (temperature, age, signal history…)
- Nonlinearity a potential issue if seeking a significant reduction in LF corner frequency and / or a lower order of response
  - Limited excursion of voice coil
  - Distortions due to $Bl(x)$, $K_S(x)$, $L_E(x)$
- But attractive possibilities offered by making the target response adaptive to operating conditions
  - E.g. small speaker equalised to full low-frequency range capability when quiet, but with raised f_c in "party mode" to maintain acceptable displacement
  - "Audio limiter" type behaviour, based upon displacement

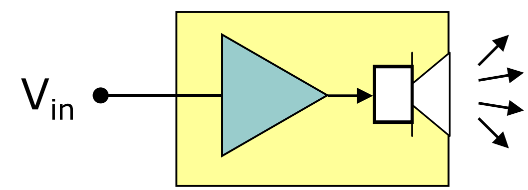
Feedback control

- The norm for decades in servo motor control
- Has been (and is) applied to speakers – but still a comparative rarity in the market place
- Apply negative feedback of voice coil motion
  - Acceleration, velocity or displacement…
  - …but remember that SPL is proportional to acceleration
- Careful tailoring of loop transfer function needed
  - But stability need NOT be an issue at low frequencies where improvements are [IMHO] most needed
- Potential increases in cost and complexity
- Rewards are extended frequency response and lower nonlinear distortion with much-reduced sensitivity to drive unit parameters
- Adaptive target response possible, as before

Open-loop versus closed-loop control

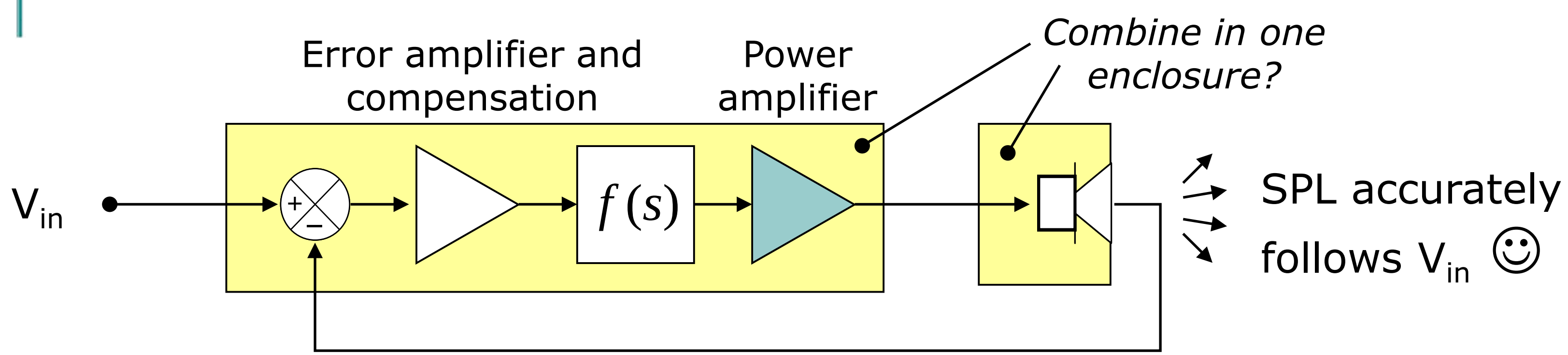


Traditional separate amplifier & speaker



“Active” loudspeaker

Conventional systems : both are essentially open-loop in operation

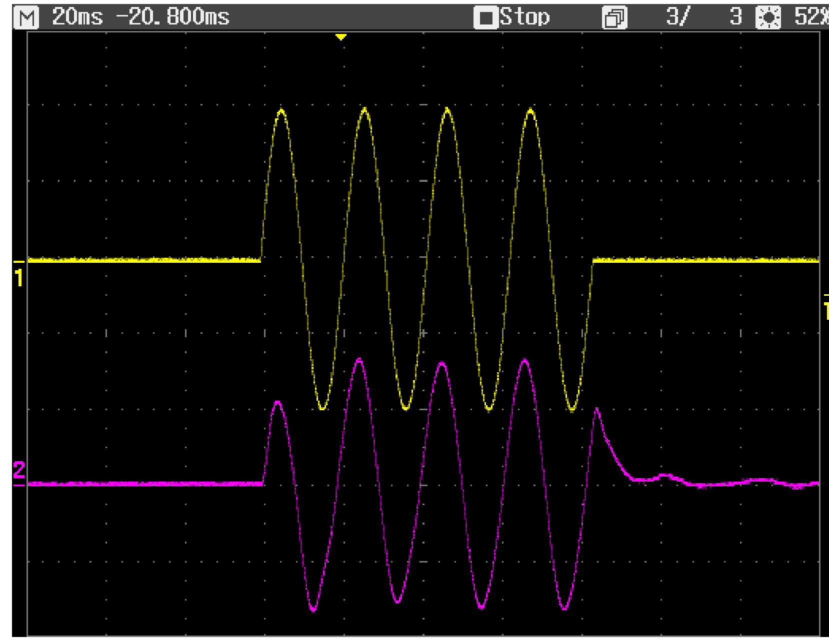


Full active servo control system

Transient response improvement:
50Hz tone burst, without and with active control

Conventional vented system

MFB system

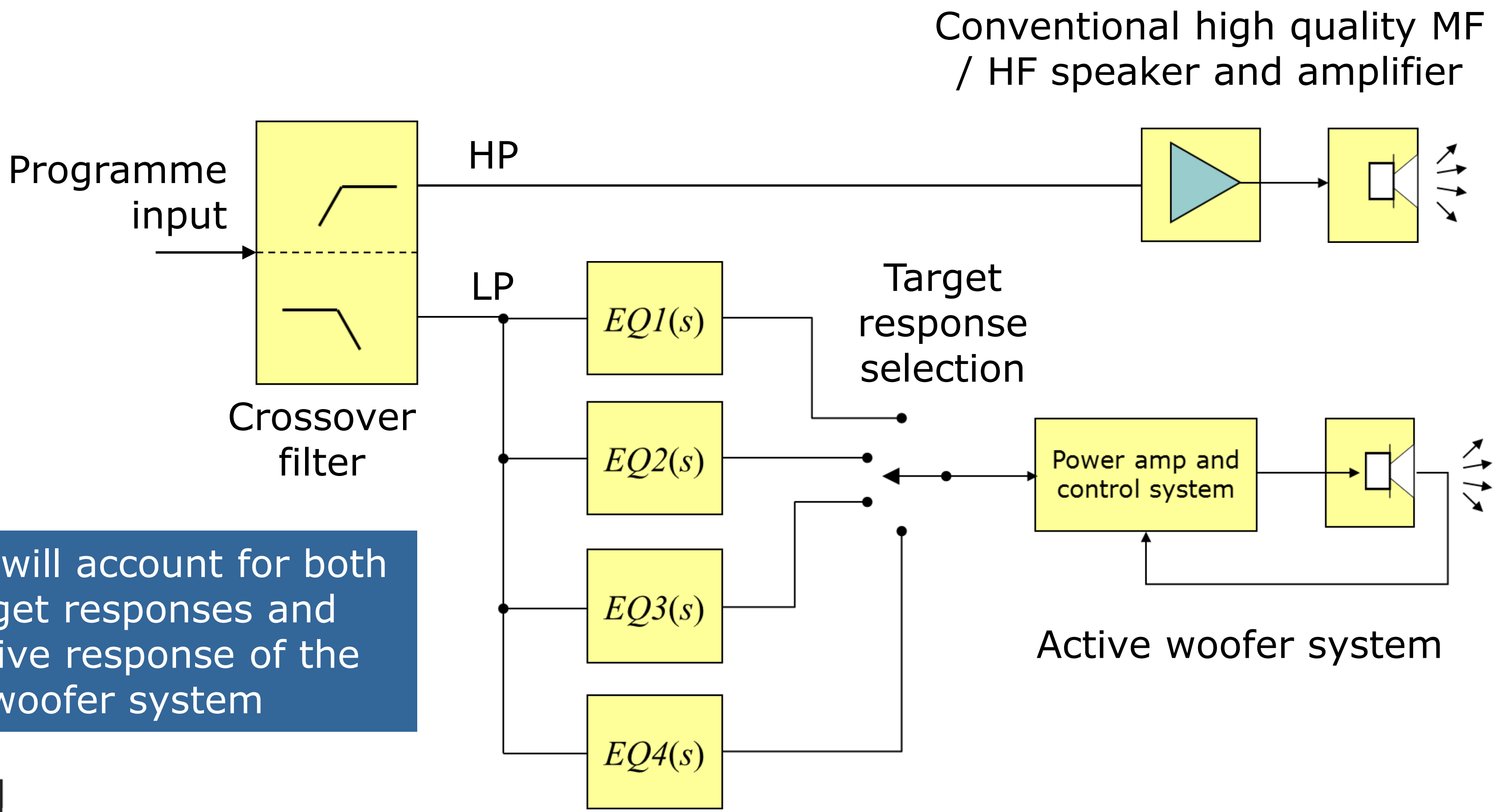


- Upper trace = input signal
- Lower trace = (near field) acoustical output

Proposal for further subjective evaluation

For presentation at a future Convention
(Spring 2018, or *maybe* Autumn this year)

Accurate emulation of a range of high-pass woofer responses



$EQ_N(s)$ will account for both the target responses and the native response of the active woofer system

Evaluation testing proposal: just how subjectively audible is all this?

- Extension of tests by others (as described earlier)
- Range of high-pass woofer responses: 1st order, B2, B4, B6 (need to decide what f_c is appropriate for each)
- Varied programme material – tone bursts, speech, various types of music
- Blind or double-blind testing in "reasonable" room
- Accurate emulation of target responses – no need for approximations
- Results not obscured by port nonlinearity etc.
- Single consistent (physically identical) test set-up for all responses – no influence of spacing etc.
- Input welcomed: defining details of procedure, and as members of the listening panel

Thank you!

Questions and discussion