Perceptually motivated filter design with applications to loudspeaker-room equalization

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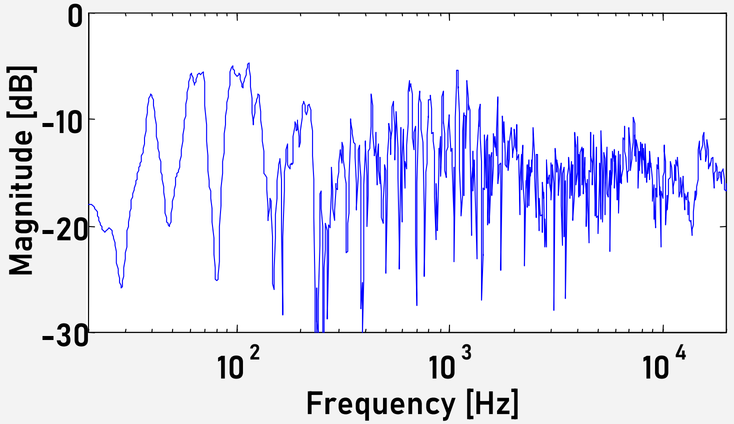
Outline

1. Motivation
2. Fractional octave smoothing
3. Traditional FIR and IIR filters
4. Automatically optimized parametric EQs
5. Warped filters
6. Kautz filters
7. Parallel filters
8. Similarities, comparison

Problem statement

Equalization of high-order systems, e.g., loudspeaker-room response

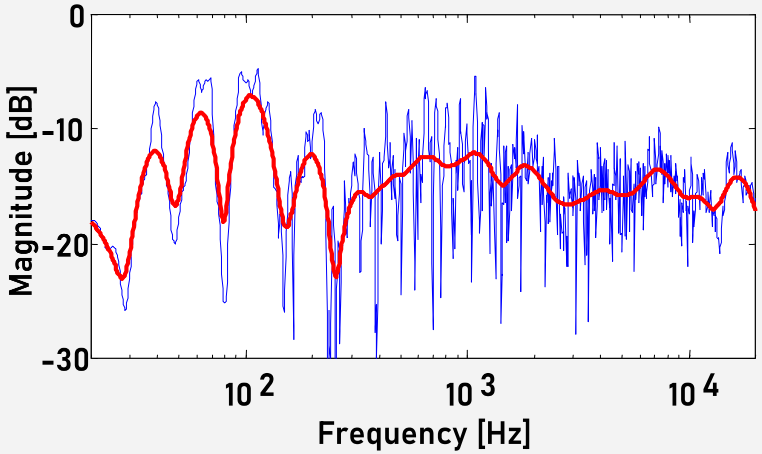
Direct inversion has problems:
- parameter sensitivity (e.g., position)
- amplifier/driver overload
- computational complexity

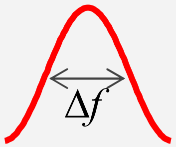


Transfer function smoothing

Traditionally: magnitude smoothing (e.g. third octave) for displaying transfer functions

Complex smoothing also possible
- impulse response can be computed
- phase can also be equalized



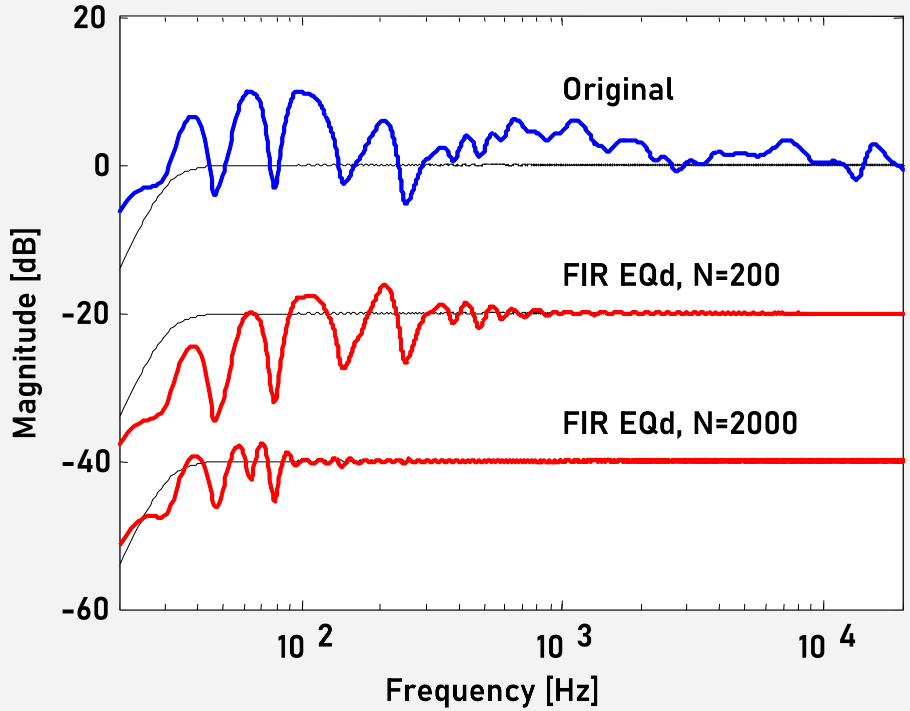


Convolution with a
Hann window

FIR EQ examples

The frequency response of the FIR filter equals the DFT (or FFT) of its impulse response

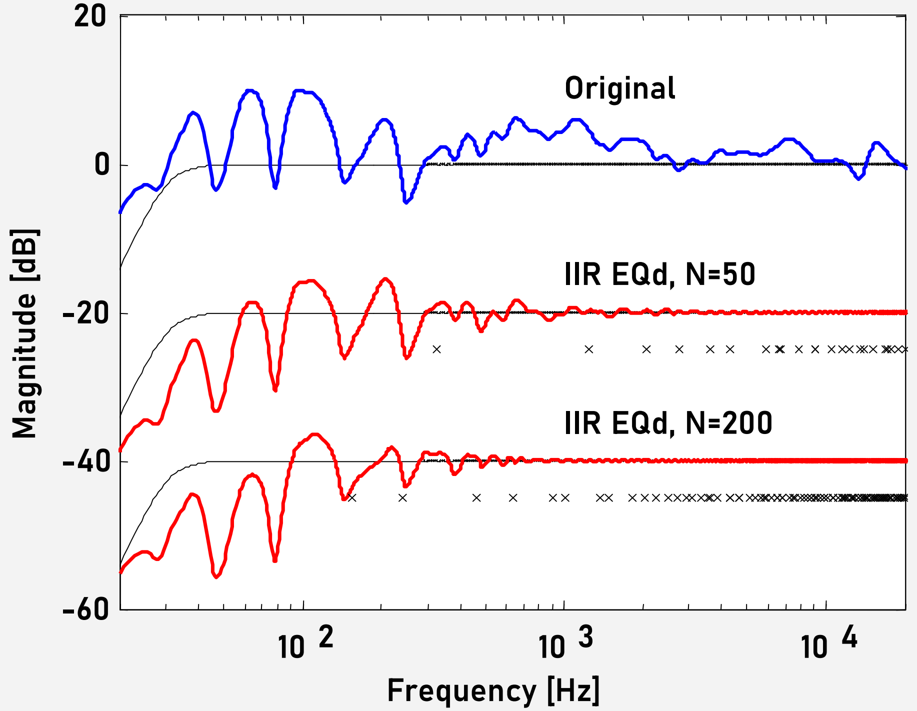
$$\Delta f = \frac{f_s}{N}$$



Time-domain EQ design based on the smoothed room response

IIR EQ examples

The high pole
density required by
log. frequency resolution
cannot be realized with
direct form IIR filters
(weighting does not help)



Time-domain EQ design based on the smoothed room response

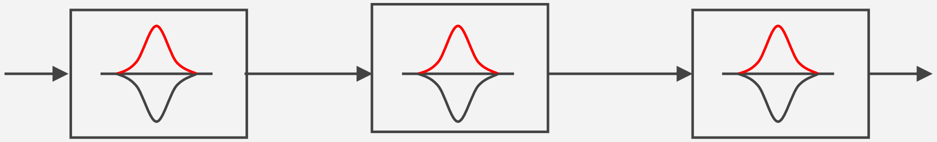
Log. frequency resolution filter design methods

Series or parallel sections + tricky design!

Examples:
- parametric EQs (automatic)
- warped filters
- Kautz filters
- fixed-pole parallel filters

Automatic calibration of parametric EQs

For each section:
- center frequency
- Q factor
- gain



Properties:

+ intuitive parameters: easy to change them manually
- only 3 free parameters/section instead of 4 of a biquad
- magnitude-only (minimum-phase) EQ
- nonlinear optimization needed (no closed from solution)

Frequency warped filter design

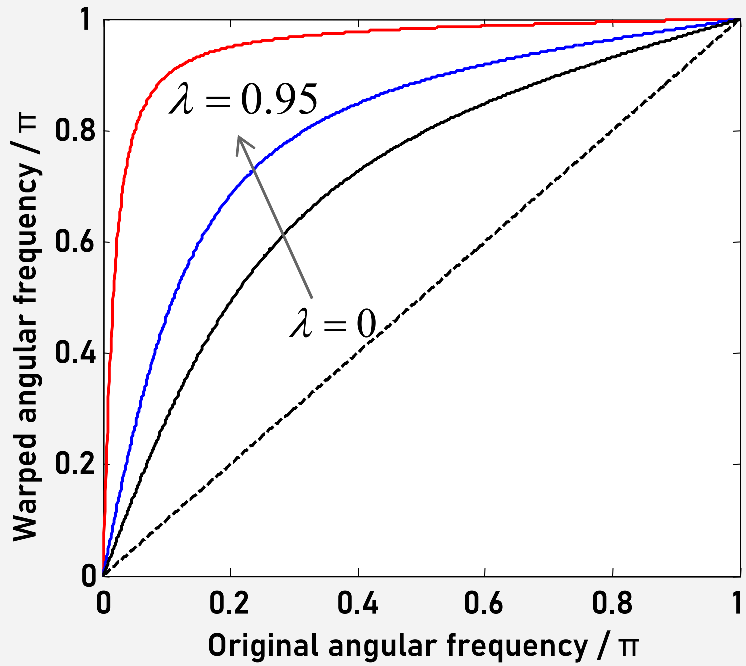
Unit delays in traditional FIR or IIR filters are replaced by a first-order allpass filter

$$z^{-1} \leftarrow D(z) = \frac{z^{-1} - \lambda}{1 - \lambda z^{-1}}$$

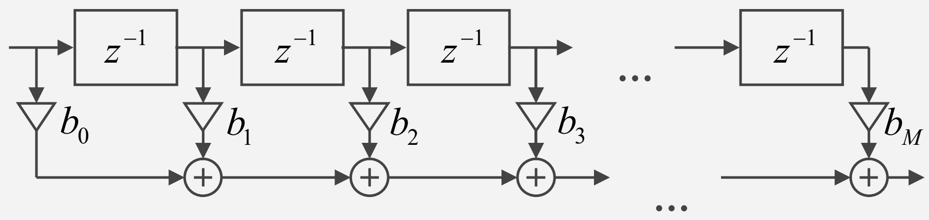
resolution is controlled by λ

Transformation of the frequency axis:

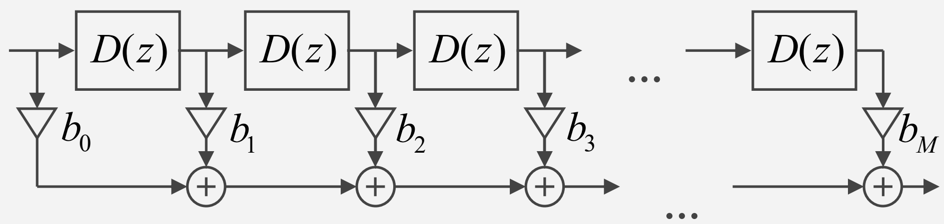
$$\tilde{\vartheta} = \arctan \frac{(1-\lambda^2)\sin(\vartheta)}{(1+\lambda^2)\cos(\vartheta) - 2\lambda}$$



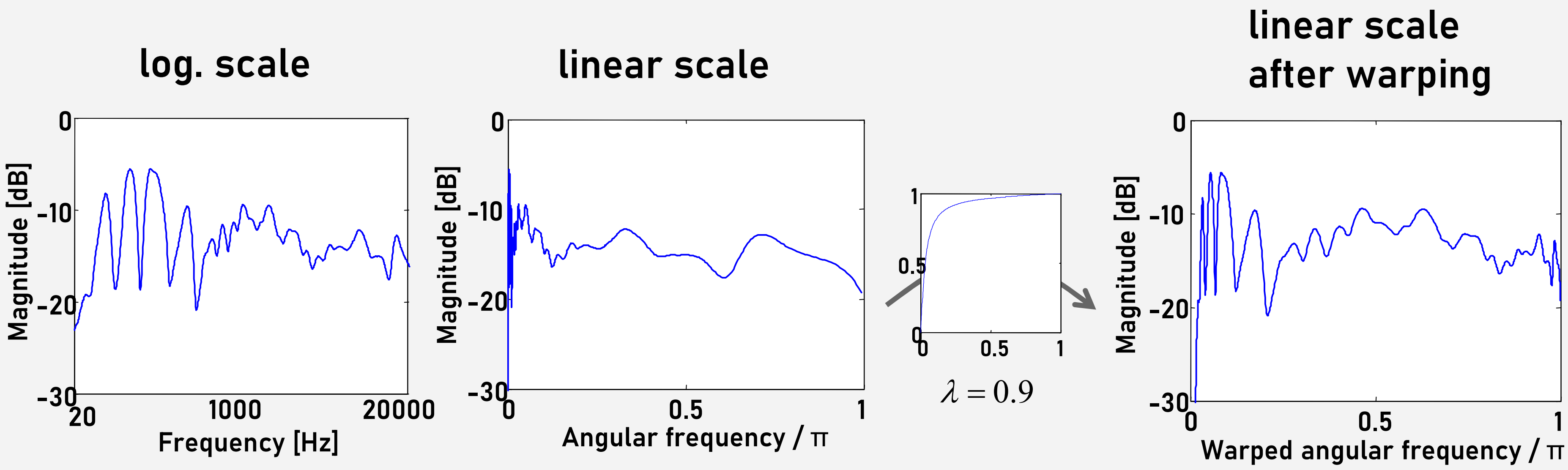
Implementation of warped FIR filters



$$z^{-1} \leftarrow D(z) = \frac{z^{-1} - \lambda}{1 - \lambda z^{-1}}$$



Warped filter design



Design an FIR or IIR filter based on the prewarped response

Implement the filter with allpasses instead of the unit delays

Warped filter properties

+ Traditional FIR and IIR filter design techniques are used after prewarping

- Relatively complicated filter structure $\longrightarrow$

- Less flexible allocation of the frequency resolution due to the single warping parameter λ $\longrightarrow$

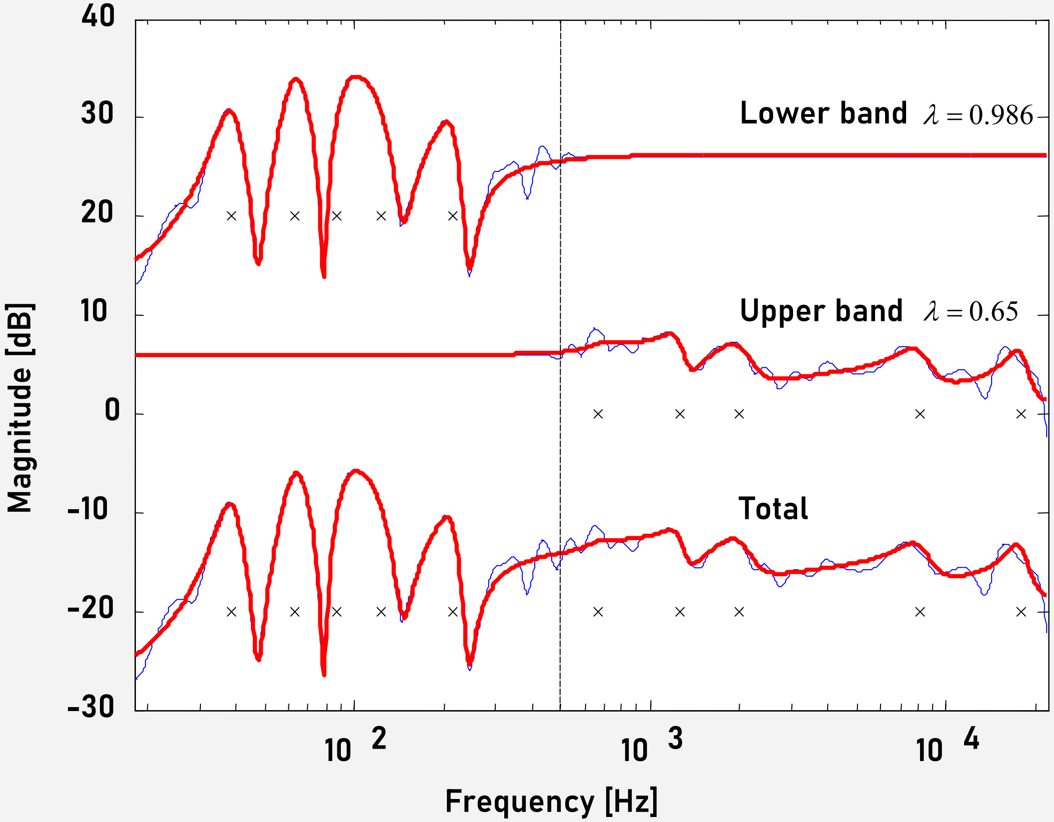
Improvements:

- Dewarping to series biquads

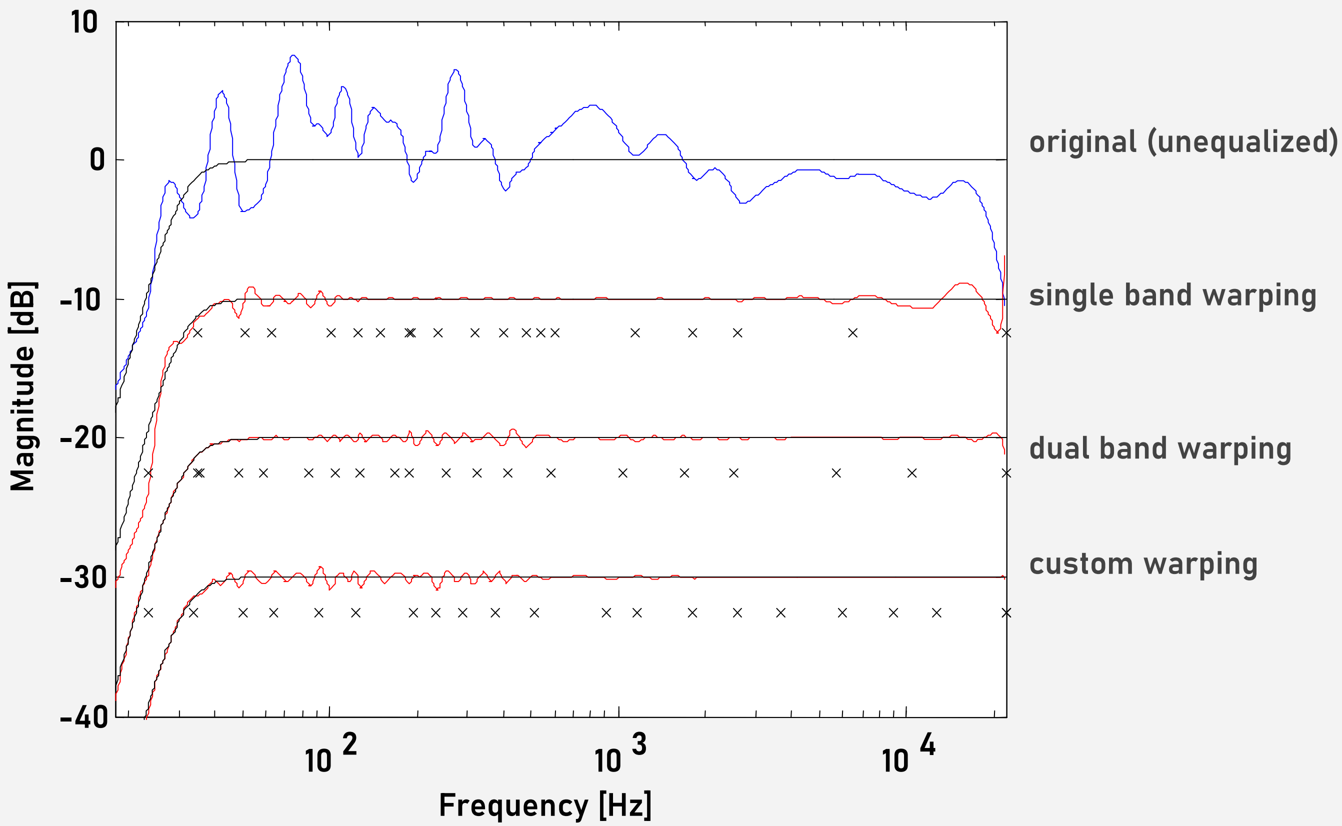
- Dual band warping (separate λ for low and high freq.)
- Custom warping (truly logarithmic warping profile)

Dual-band warping

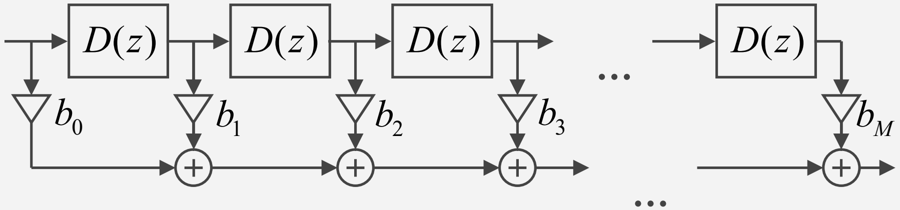
Total filter order is N=20



Warped IIR equalization (N=40)



Extension to warped FIR filters



$$D(z) = \frac{z^{-1} - \lambda}{1 - \lambda z^{-1}}$$

Could we have different λ values for each section?

Kautz filters

Can be seen as the generalization of warped FIR filters

$$H(z)=\sum_{k=1}^{K} w_k G_k(z)$$

free parameters

fixed basis functions

$G_k(z)$: Orthonormal Kautz polynomials = orthonormalized complex exponentials

Frequency resolution is controlled by the poles p_k (can also be complex)

Implementation of Kautz filters

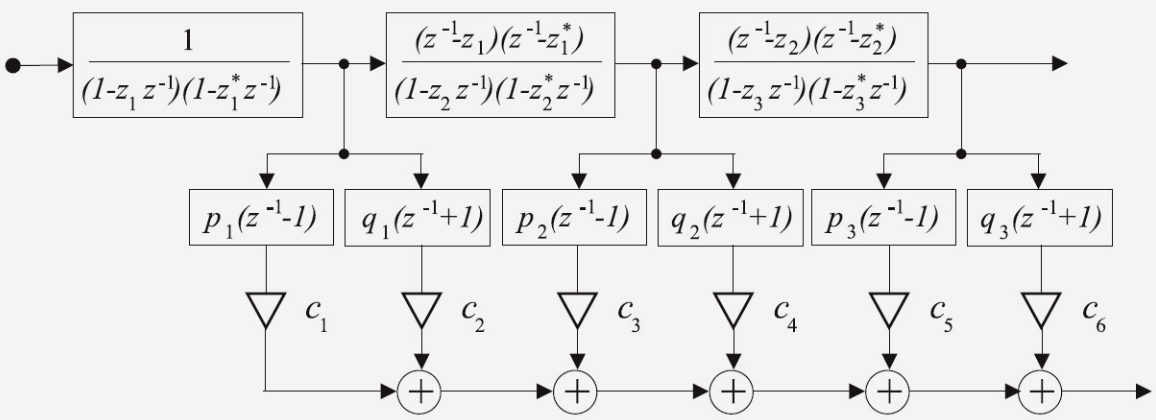


Figure by Paatero and Karjalainen (2003)

6 multiplications and 8 additions per second-order section

Parallel second-order filters

$$H(z) = \sum_{k=1}^{K} \frac{b_{k,0} + b_{k,1} z^{-1}}{1 + a_{k,1} z^{-1} + a_{k,2} z^{-2}}$$

Usual steps:

1. Design a high-order IIR filter
2. Convert to parallel form

Benefits:

- better numerical properties
- possibility for code parallelization

Fixed-pole design of parallel filters

free parameters

$$H(z) = \sum_{k=1}^{K} \frac{d_{k,0} + d_{k,1} z^{-1}}{1 + a_{k,1} z^{-1} + a_{k,2} z^{-2}} + \underbrace{\sum_{m=0}^{M} b_m z^{-m}}_{\text{optional FIR part}}$$

fixed basis functions

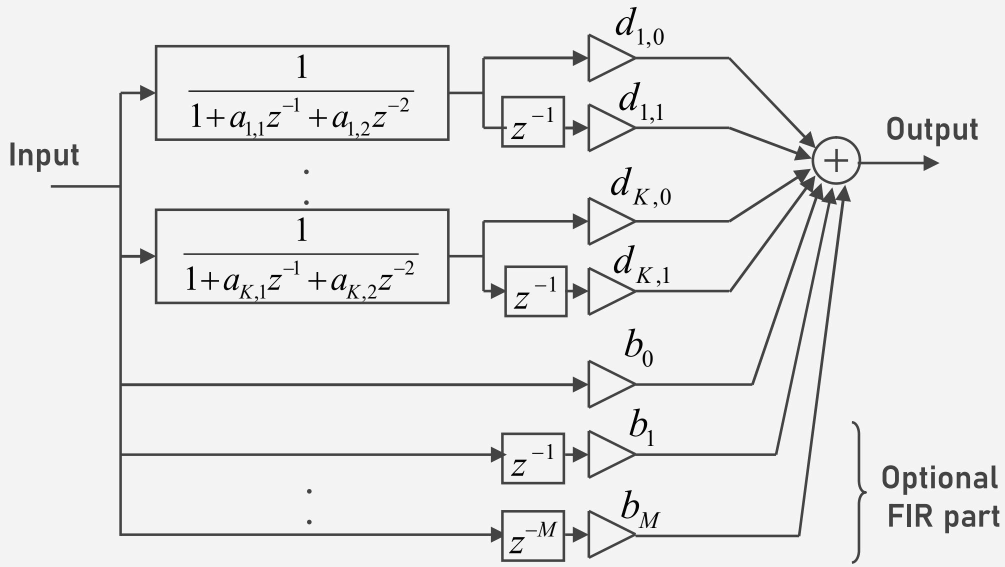
Linear-in-parameter model with exponentially decaying basis functions

↓

Least-squares solution

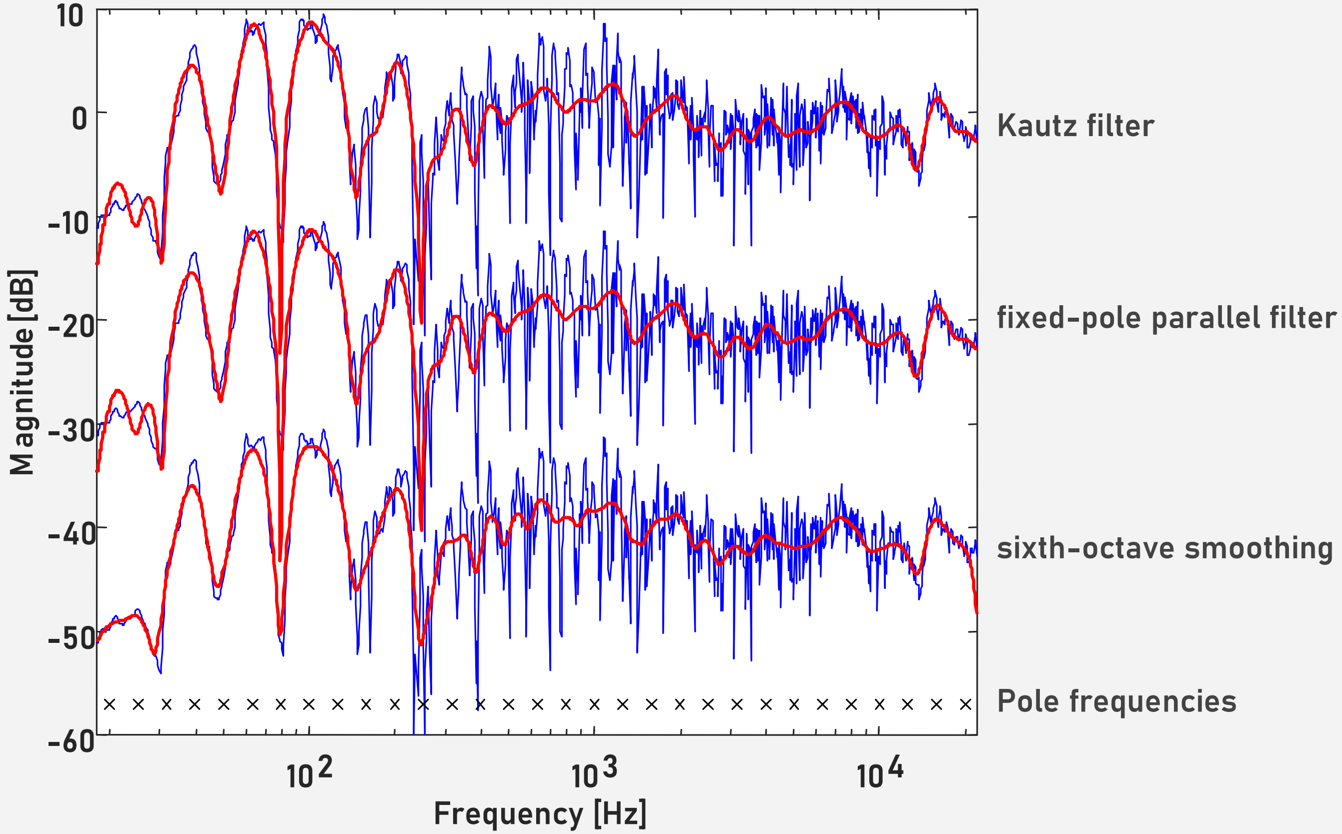
$$\mathbf{p} = \left(\mathbf{M}^{\mathrm{T}}\mathbf{M}\right)^{-1}\mathbf{M}^{\mathrm{T}}\mathbf{y}_{\mathrm{t}}$$

Filter structure

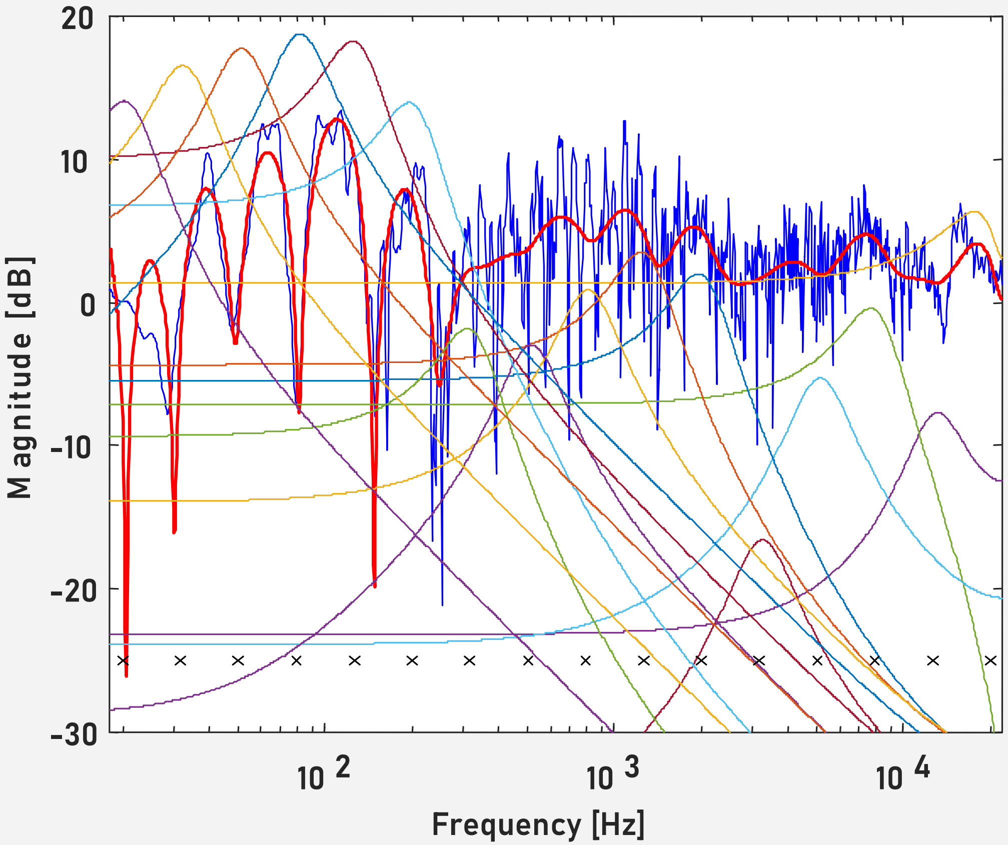


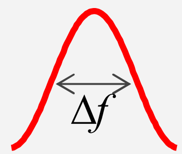
4 mul. and 4 add. per section (vs. 6 and 8 in Kautz filters)

Kautz and parallel filter example



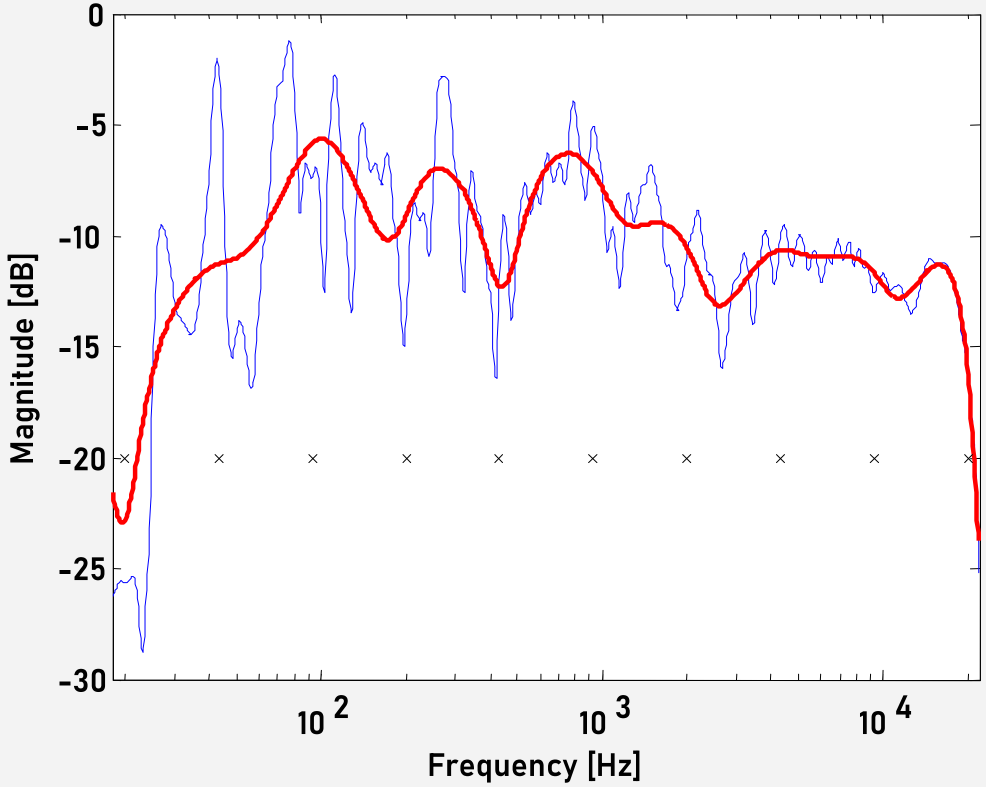
Responses of the second-order sections





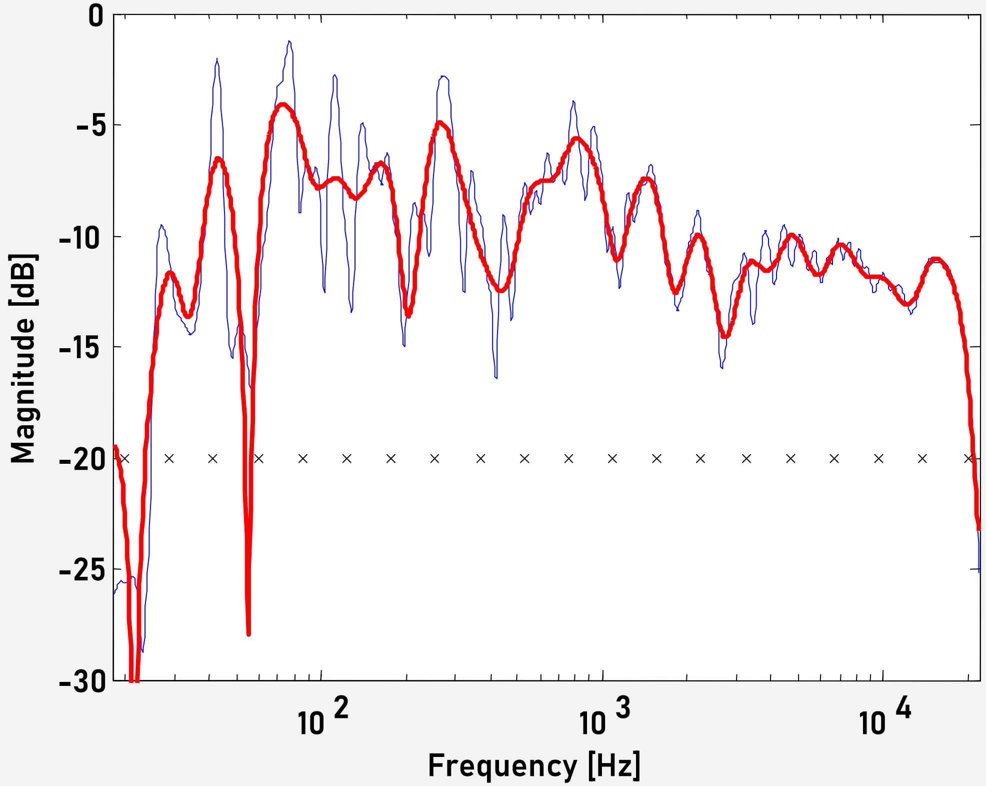
Hann window

The effect of pole frequencies



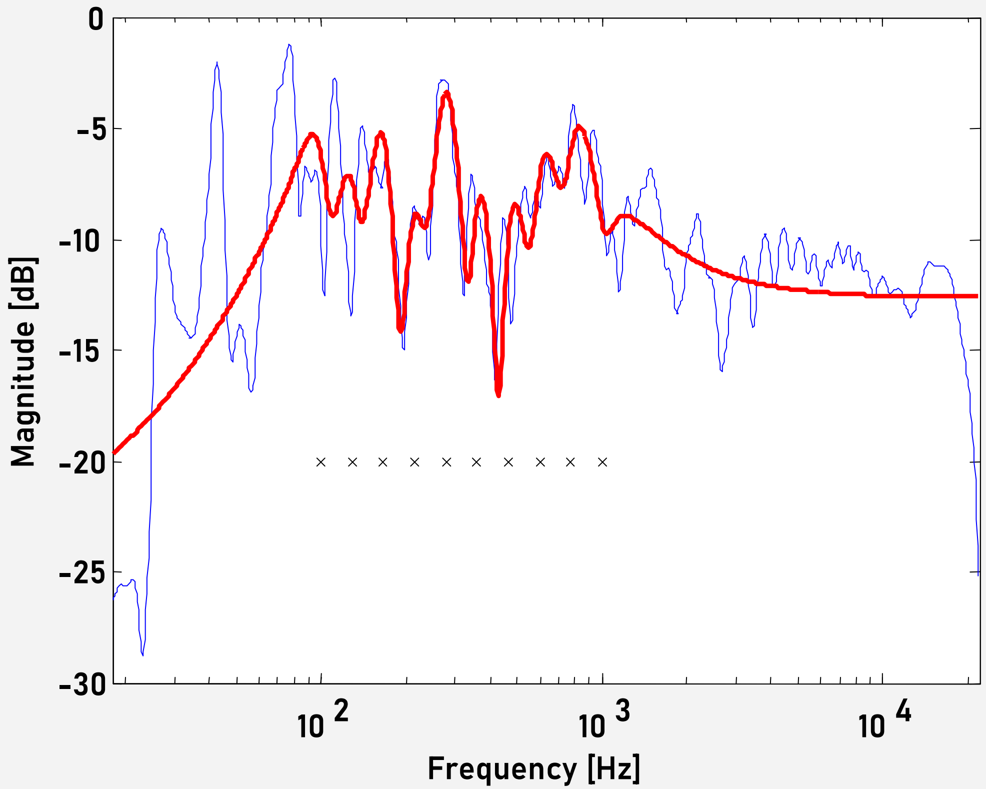
logarithmically distributed poles

The effect of pole frequencies



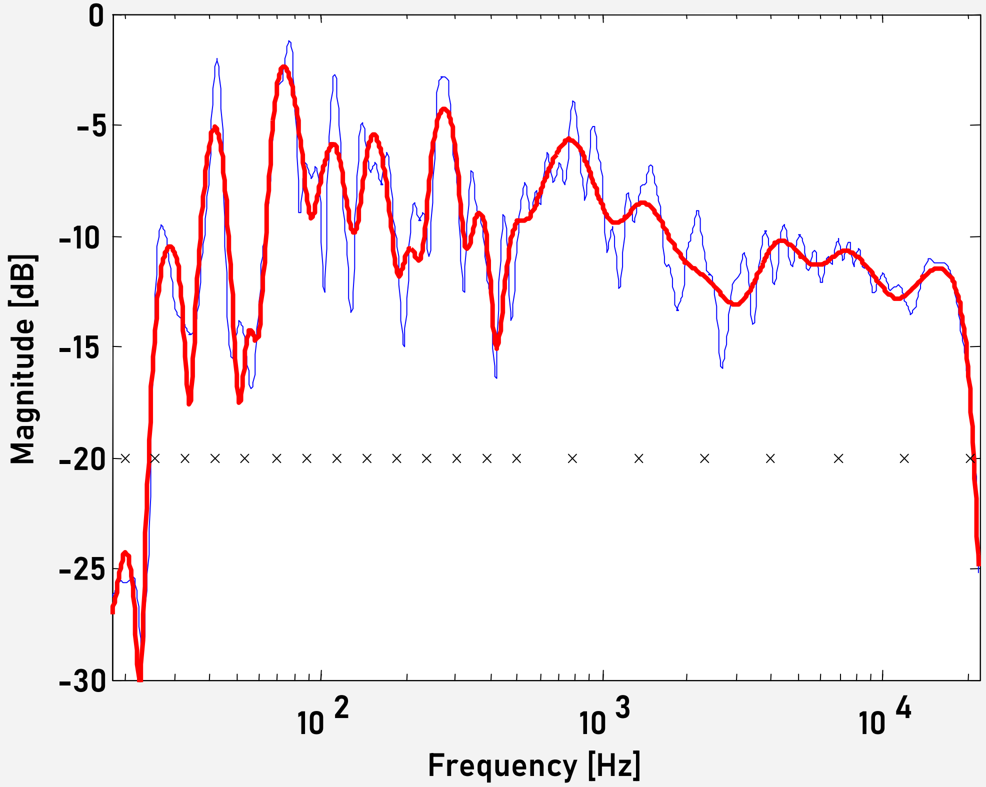
higher pole density

The effect of pole frequencies



poles only in a limited region

The effect of pole frequencies



two different densities

Pole positioning strategies

Predetermined (manual) pole positioning

- logarithmic
- stepwise logarithmic
- arbitrary

Very simple, user control Works also with unsmoothed responses	Needs manual intervention

Automatic pole positioning based on the ripple density

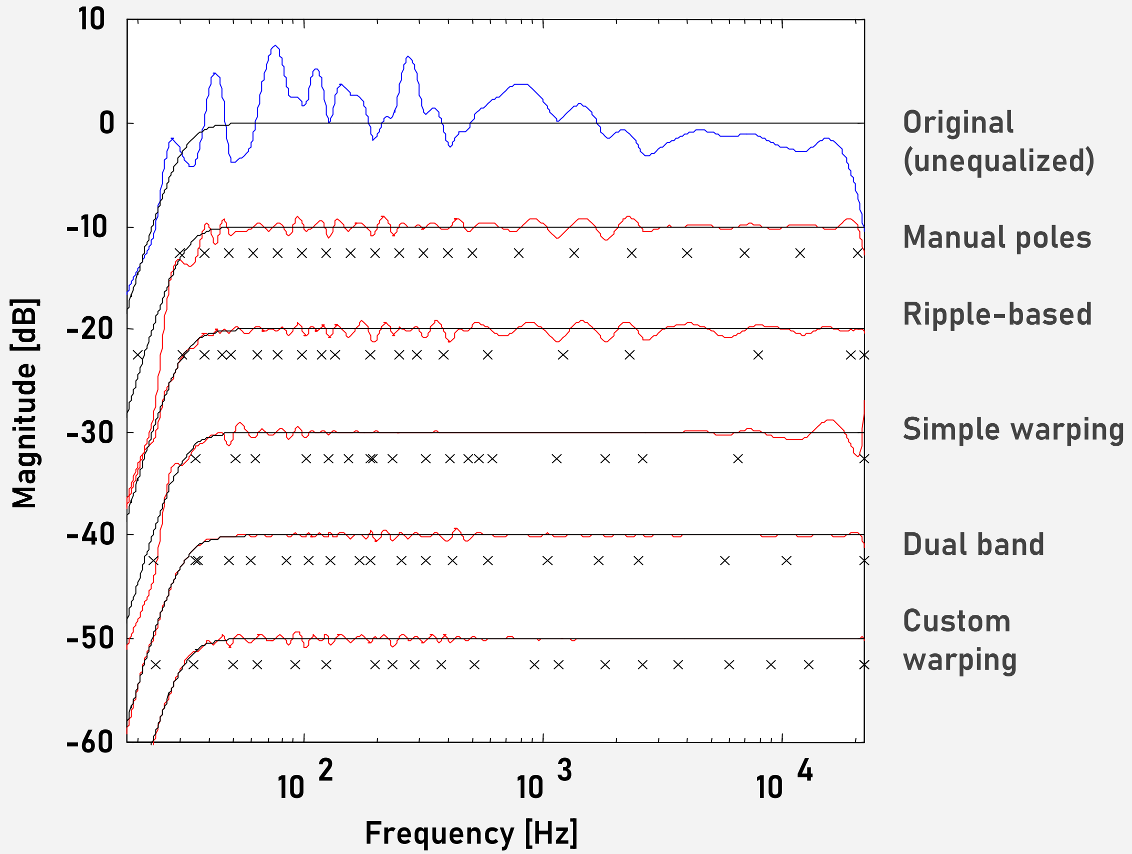
Automatic, simple	Mathematically not optimal

Warped IIR design based

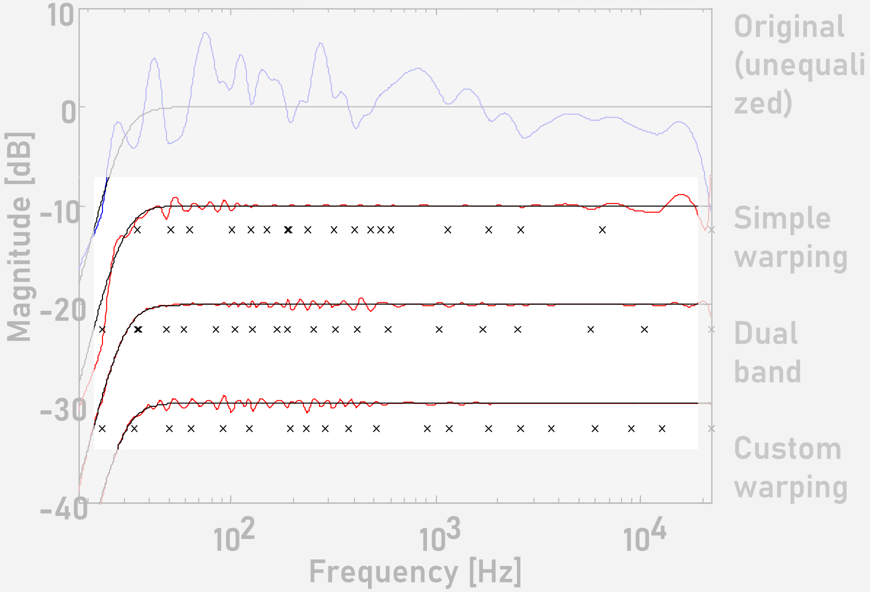
- simple warping
- dual-band
- custom warping

Most accurate results	Bit more complicated

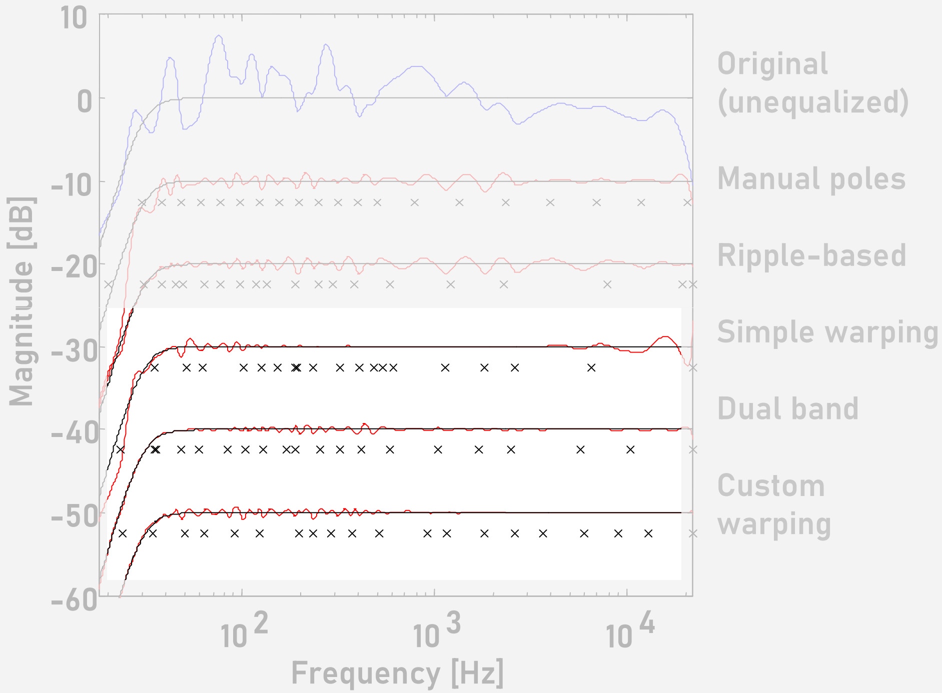
Pole positioning comparison (N=40)



Comparison to warped IIR filters



Warped IIR design



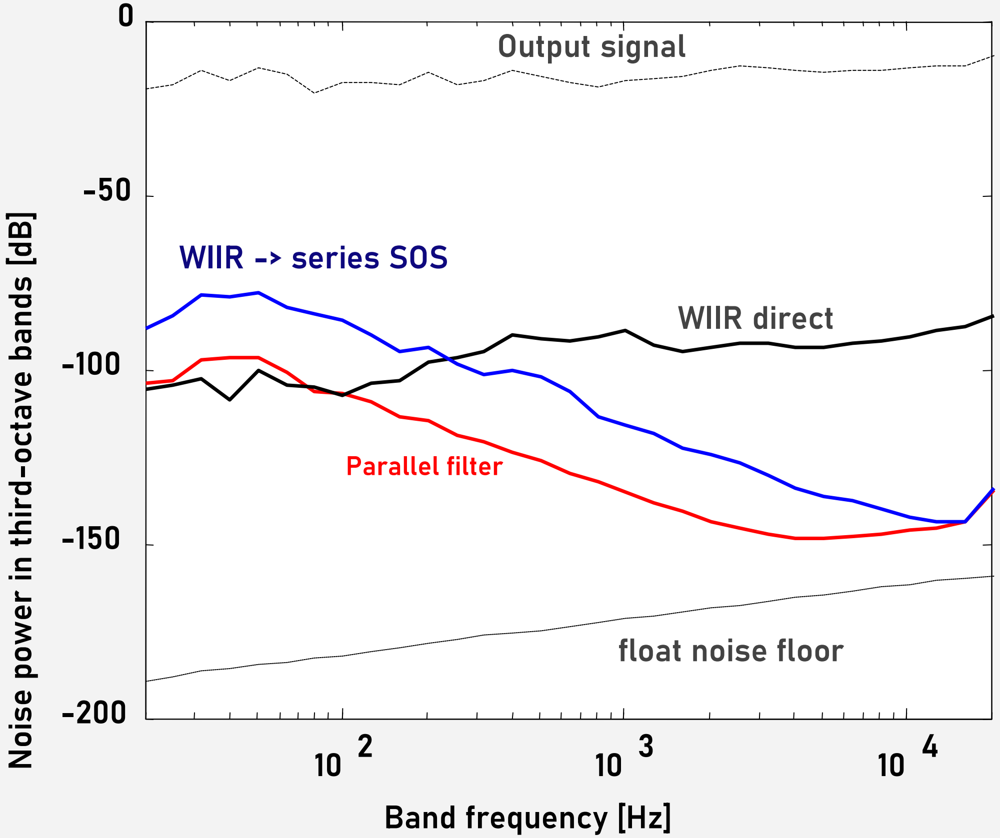
Parallel filter using warped IIR poles

Quantization noise in third-octave bands

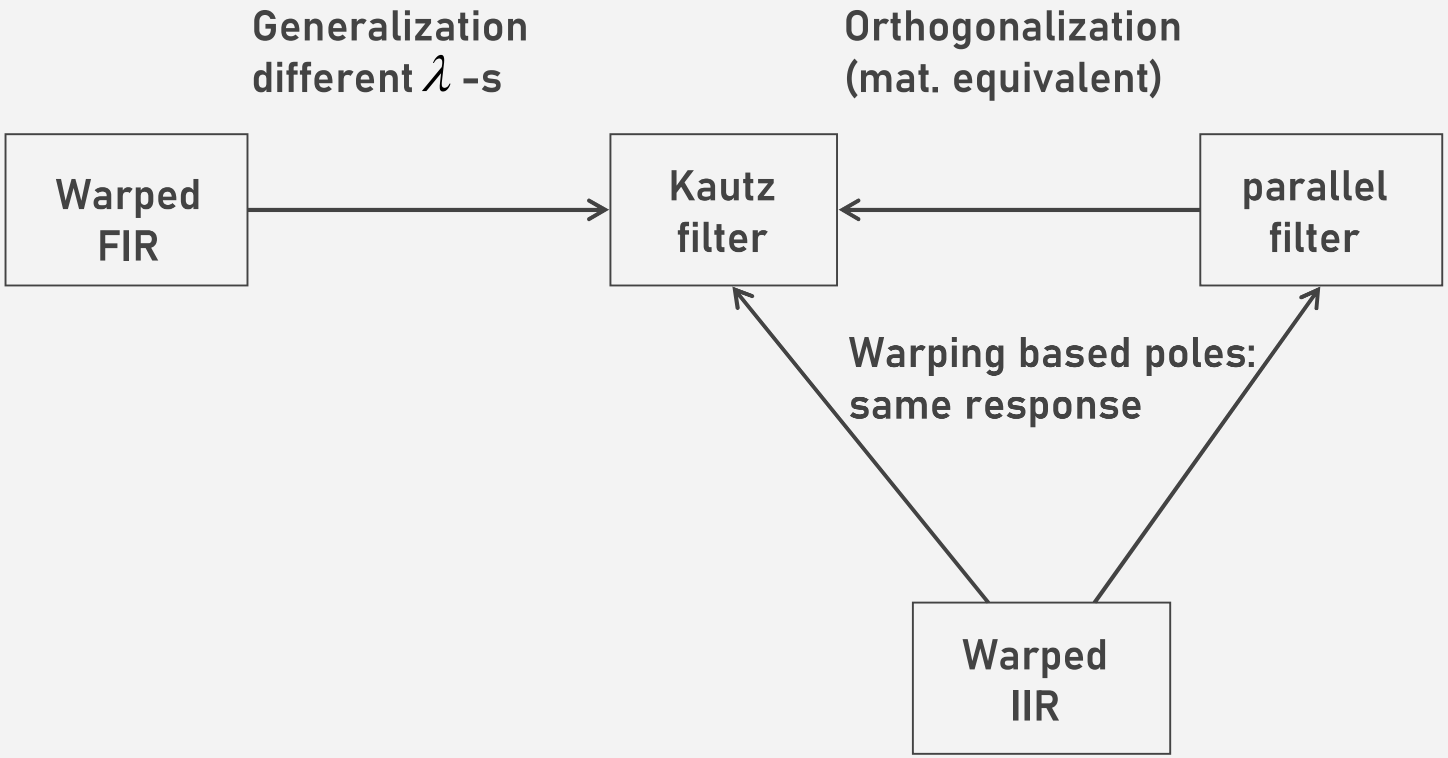
Single precision floating point (float) implementation

Filters excited with pink noise

Parallel filter has the lowest noise



Relations


Generalization
different λ -s
Orthogonalization
(mat. equivalent)
Warped
FIR
Kautz
filter
parallel
filter
Warping based poles:
same response
Warped
IIR

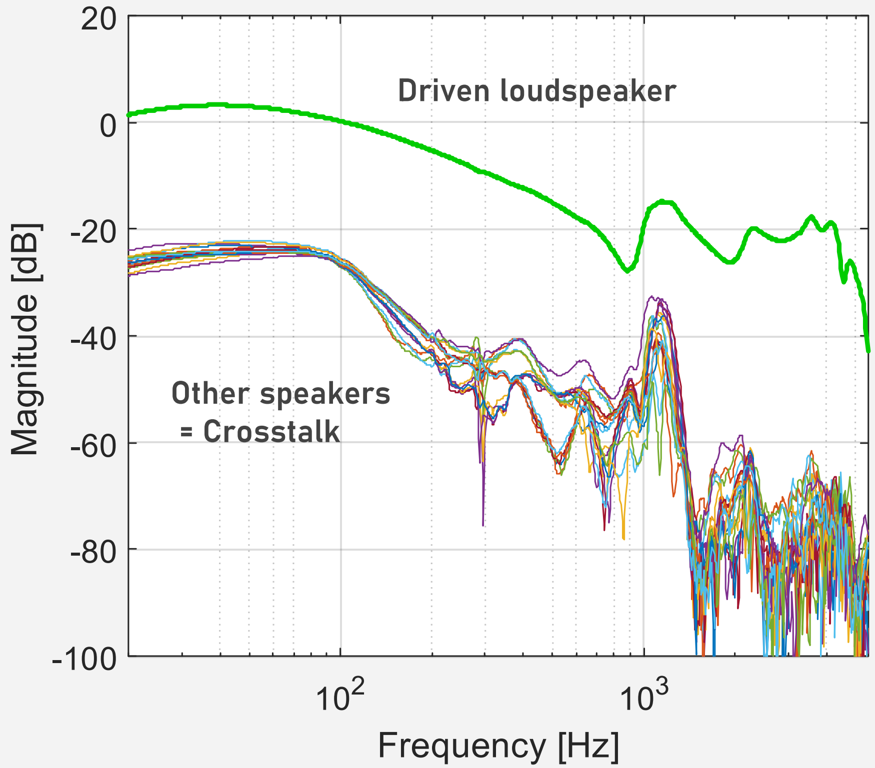
Practical differences

	Warped FIR/IIR	Kautz filter	parallel filter
Design steps	1. Warp f. resp. 2. Normal FIR/IIR design	1. Choose poles 2. Linear LS opt.	
Implementation	Special struct. or Dewarp to SOS (4 MAC)	Special struct. (6 MAC + 2 ADD)	Parallel SOS (4 MAC) Best quant. noise

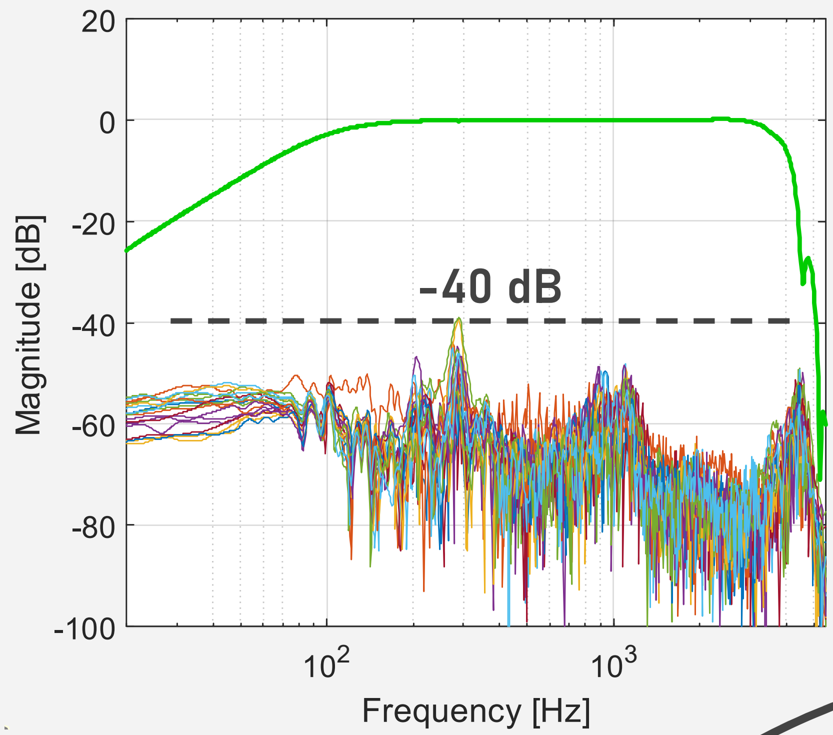
MIMO equalization example

Spherical louspeaker at IEM, Graz
20 independently driven loudspeaker elements

Significant crosstalk due to acoustic coupling

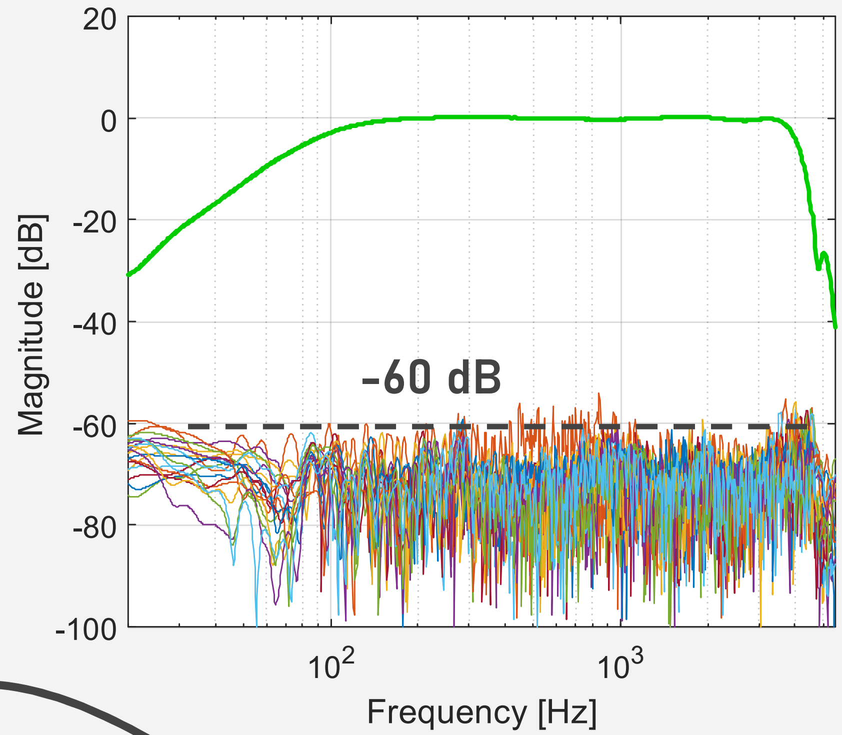


MIMO equalization example



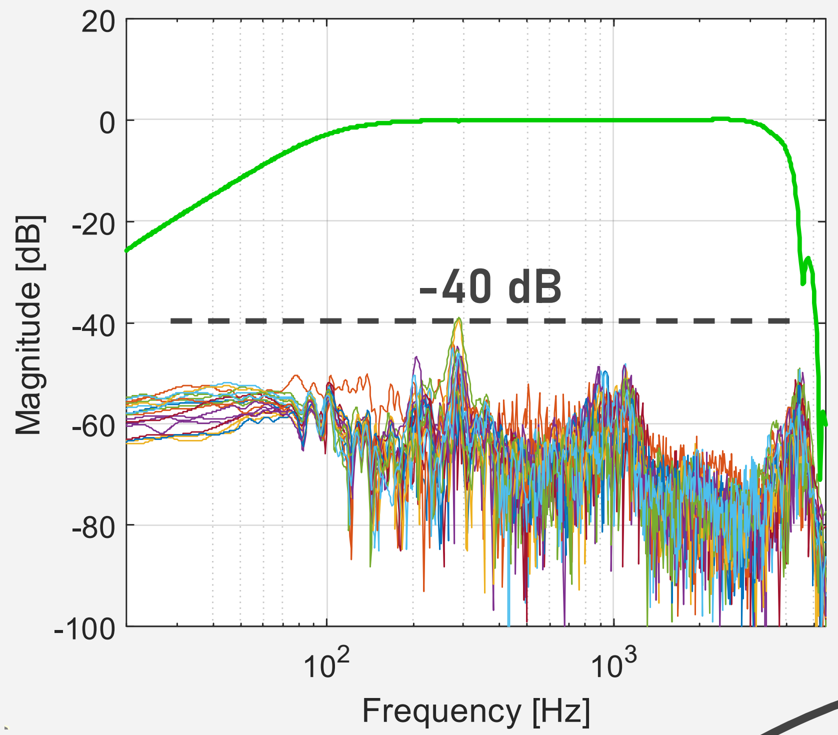
256 tap FIR filter

Same number of MAC operations



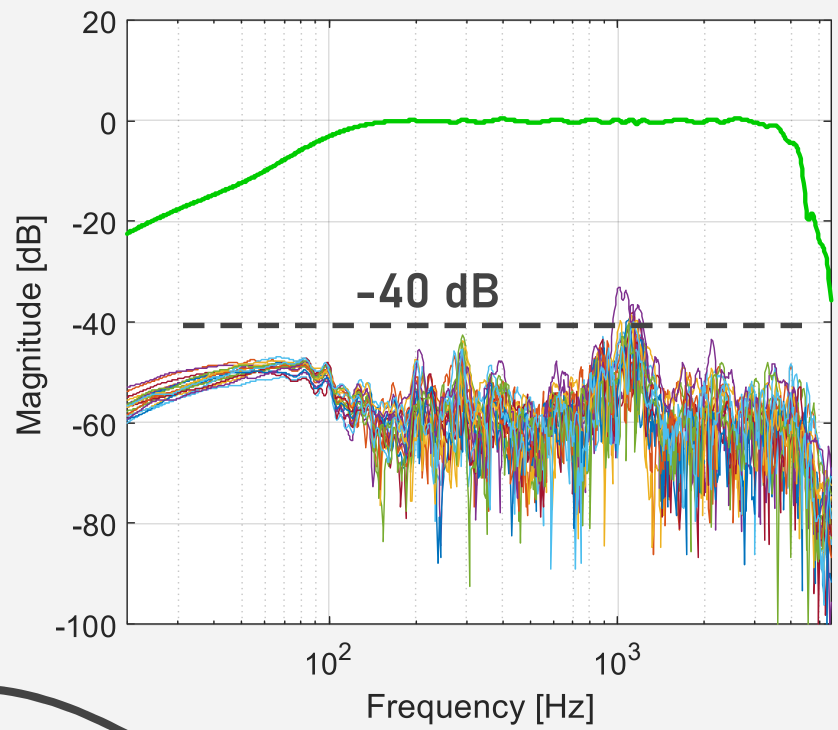
64 second-order IIR sections

MIMO equalization example



256 tap FIR filter

1/4 as much
MAC operations



16 second-order IIR
sections

Conclusion

Already the simplest methods are much better than normal FIR and IIR filters (savings in the order of 10)

- Warped FIR filter
- Parallel filter with manual (log.) poles

More sophisticated warping-based methods

- dual-band warping
- custom warping

provide the best performance,

can be implemented as

- warped IIR (special struct. or dewarped SOS)
- fixed-pole parallel filter: better quant. noise, code parallelization

Thank you!

Questions?

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MATLAB code:

Warped: http://www.acoustics.hut.fi/software/warp/

Kautz: http://www.acoustics.hut.fi/software/kautz/kautz.htm

Parallel: http://www.mit.bme.hu/~bank/parfilt

References

Room equalization (related to slide No. 3):

S. Cecchi, A. Carini, S. Spors, “Room Response Equalization – A Review,” J. Appl. Sci., 8(1), article 16, 2018.

M. Karjalainen, T. Paatero, J. N. Mourjopoulos, P. D. Hatziantoniou, “About Room Response Equalization and Dereverberation,” Proc. IEEE Workshop Appl. of Signal Process. To Audio and Acoust., New Paltz, NY, USA, pp. 183–186., 2005.

Complex smoothing (related to slide No. 4):

P. D. Hatziantoniu and J. N. Mourjopoulos, “Generalized Fractional-Octave Smoothing for Audio and Acoustic Responses,” J. Audio Eng.Soc., 48(4):259–280, Apr. 2000.

M. Karjalainen and T. Paatero, “Frequency-Dependent Signal Windowing,” Proc. IEEE Workshop Appl. of Signal Process. To Audio and Acoust., pages 35–38, New Paltz, NY, USA, Oct. 2001.

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Automatically tuned parametric equalizers (related to slide No. 8):

G. Ramos and J. J. Lopez, “Filter Design Method for Loudspeaker Equalization Based on IIR Parametric Filters,” J. Audio Eng. Soc., 54(12): 1162–1178, Dec. 2006.

H. Behrends, A. von dem Knesebeck, W. Bradinal, P. Neumann, and U. Zölzer, “Automatic Equalization Using Parametric IIR Filters,” J. Audio Eng. Soc., 59(3): 102–109, Mar. 2011.

G. Vairetti, E. De Sena, M. Catrysse, S. H. Jensen, M. Moonen, and T. van Waterschoot, “An Automatic Design Procedure for Low-Order IIR Parametric Equalizers,” J. Audio Eng. Soc. 66(11): 935–952. 2018.

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Warped filter design and implementation (related to slides No. 9-12):

A. Harma, M. Karjalainen, L. Savioja, V. Valimaki, U. K. Laine, and J. Huopaniemi, “Frequency-Warped Signal Processing for Audio Applications,” J. Audio Eng. Soc., 48(11): 1011–1031, Nov. 2000.

M. Waters and M. B. Sandler, “Least Squares IIR Filter Design on a Logarithmic Frequency Scale,” Proc. IEEE Int. Symp. on Circuits and Syst., pp. 635–638, May 1993.

M. Tyril, J. A. Pedersen, P. Rubak, Digital Filters for Low-Frequency Equalization, J. Audio Eng. Soc. 29(1–2): 36–43., 2001.

Multiband and custom warping (related to slides No. 13-14):

B. Bank and G. Ramos, “Improved Pole Positioning for Parallel Filters Based on Spectral Smoothing and Multi-Band Warping,” IEEE Signal Process. Lett., 18(5):299–302, Mar. 2011.

B. Bank, “Warped IIR Filter Design with Custom Warping Profiles and its Application to Room Equalization,” In Proc. 130th AES Conv., Preprint No. 7965, London, UK, May 2011.

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Kautz filters (related to slides No. 15-17):

T. Paatero and M. Karjalainen, “Kautz Filters and Generalized Frequency Resolution: Theory and Audio Applications,” J. Audio Eng. Soc., 51(1-2): 27–44, Jan/Feb. 2003.

M. Karjalainen and T. Paatero, “Equalization of Loudspeaker and Room Responses Using Kautz Filters: Direct Least Squares Design,” EURASIP J. on Advances in Sign. Proc., Spec. Iss. on Spatial Sound and Virtual Acoustics 2007(13), Article ID 60949, 2007.

G. Vairetti, E. De Sena,M. Catrysse, S. H. Jensen, M. Moonen, and T. van Waterschoot, “A Scalable Algorithm for Physically Motivated and Sparse Approximation of Room Impulse Responses with Orthonormal Basis Functions,” IEEE Trans. Audio, Speech, and Lang. Process. 25(7): 1547–1561. 2017.

P.W. Broome, “Discrete Orthonormal Sequences,” J. of the Association for Computing Machinery 12(2): 151–168. 1965.

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Fixed-pole parallel filters and their relation to Kautz filters (related to slides No. 18-22):

B. Bank, “Audio Equalization with Fixed-Pole Parallel Filters: An Efficient Alternative to Complex Smoothing,”. J. Audio Eng. Soc., 61(1-2):39–49, Jan. 2013.

B. Bank, "Logarithmic Frequency Scale Parallel Filter Design with Complex and Magnitude-Only Specifications," IEEE Signal Process. Lett., 18(2): 138-141, Feb. 2011.

Pole-positioning methods (related to slides No. 23-28) :

B. Bank, “Loudspeaker and Room Equalization Using Parallel Filters: Comparison of Pole Positioning Strategies,” Proc. 51st AES Conf. on Loudspeakers and Headphones, Helsinki, Finland, Aug. 2013.

E. Maestre, G. P. Scavone and J. O. Smith,"Design of Recursive Digital Filters in Parallel Form by Linearly Constrained Pole Optimization", Signal Processing Letters, 23(11): 1547-1550, Nov. 2016.

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Quantization noise in parallel filters (related to slide No. 30) :

B. Bank and K. Horváth, "Quantization Noise of Warped and Parallel Filters using Single-Precision Floating-Point Arithmetic", Proc. 142nd AES Conv., eBrief No. 337, Berlin, Germany, May 2017.

K. Horváth and B. Bank, "Optimizing the Numerical Noise of Parallel Second-Order Filters in Fixed-Point Arithmetic", J. Audio Eng. Soc., 67(10): 763-771, Oct 2019.

The delayed parallel structure (note that there was no time to talk about this, but I consider it important for practical implementations):

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B. Bank, "Converting Infinite Impulse Response Filters to Parallel Form", IEEE Signal Proc. Mag., 35(3): 124-130, May 2018.

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Multichannel equalization and crosstalk cancellation (related to slides No. 33-35):

B. Bank, "Multichannel Equalization and Crosstalk Cancellation Using Fixed-Pole IIR Filters", J. Audio Eng. Soc., 66(11): 901-909, Nov. 2018.

O. Kirkeby, P. A. Nelson, H. Hamada, and F. Orduna-Bustamante, “Fast Deconvolution of Multichannel Systems using Reguralization,” IEEE Trans. Speech Audio Process. 6(2): 189–194. 1998.

Finally, an overview about equalizer design:

V. Välimäki and J. D. Reiss, “All About Audio Equalization: Solutions and Frontiers,” J. Appl. Sci. 6(5): 129. 2016.