Frequency-Response Masking FIR Filters

Georg Holzmann

11. Juni 2007

Frequency-Response Masking Filters

- frequency-response masking filters are a technique to design sharp low-pass, high-pass, bandpass and bandstop filters with arbitrary passband bandwidth
- furthermore linear phase FIR filters are generated, which have advantages such as guaranteed stability and are free of phase distortion
- however, the problem with FIR filters is the high complexity for sharp filters

Frequency-Response Masking Filters

- frequency-response masking filters are a technique to design sharp low-pass, high-pass, bandpass and bandstop filters with arbitrary passband bandwidth
- furthermore linear phase FIR filters are generated, which have advantages such as guaranteed stability and are free of phase distortion
- however, the problem with FIR filters is the high complexity for sharp filters

Frequency-Response Masking Filters

- frequency-response masking filters are a technique to design sharp low-pass, high-pass, bandpass and bandstop filters with arbitrary passband bandwidth
- furthermore linear phase FIR filters are generated, which have advantages such as guaranteed stability and are free of phase distortion
- however, the problem with FIR filters is the high complexity for sharp filters

Advantages

- with the frequency-response masking technique the resulting filter has very sparse coefficients
- since only a very small fraction of its coefficient values are nonzero, its complexity is very much lower than the infinite wordlength minimax optimum filter
- with an additional multiplierless design method the complexity is reduced to a minimum

Advantages

- with the frequency-response masking technique the resulting filter has very sparse coefficients
- since only a very small fraction of its coefficient values are nonzero, its complexity is very much lower than the infinite wordlength minimax optimum filter
- with an additional multiplierless design method the complexity is reduced to a minimum

Advantages

- with the frequency-response masking technique the resulting filter has very sparse coefficients
- since only a very small fraction of its coefficient values are nonzero, its complexity is very much lower than the infinite wordlength minimax optimum filter
- with an additional multiplierless design method the complexity is reduced to a minimum

Linear Phase FIR Filters

- in linear phase FIR filters phase is a linear function of frequency
- they have a symmetric impulse response
- the phase delay $(-\frac{phase}{\omega})$ is $\frac{N-1}{2}$ at every frequency
- also the group delay $(-\frac{d}{d\omega}phase)$ is $\frac{N-1}{2}$

Linear Phase FIR Filters

- in linear phase FIR filters phase is a linear function of frequency
- they have a symmetric impulse response
- the phase delay $(-\frac{phase}{\omega})$ is $\frac{N-1}{2}$ at every frequency
- also the group delay $(-\frac{d}{d\omega}phase)$ is $\frac{N-1}{2}$

Linear Phase FIR Filters

- in linear phase FIR filters phase is a linear function of frequency
- they have a symmetric impulse response
- the phase delay $(-\frac{phase}{\omega})$ is $\frac{N-1}{2}$ at every frequency
- also the group delay $(-\frac{d}{d\omega}phase)$ is $\frac{N-1}{2}$

Linear Phase FIR Filters

- in linear phase FIR filters phase is a linear function of frequency
- they have a symmetric impulse response
- the phase delay $(-\frac{phase}{\omega})$ is $\frac{N-1}{2}$ at every frequency
- also the group delay $(-\frac{d}{d\omega}phase)$ is $\frac{N-1}{2}$

Zero Phase Filters

- are a special case of linear phase filters, where the phase delay is zero
- impulse response of a zero phase filter is even about time 0:

$$h(n) = h(-n)$$

therefore this filter cannot be causal

- a real, even impulse response corresponds to a real, even frequency response
- for an odd impulse response

$$h(n) = -h(-n)$$

the frequency response is purely imaginary

Zero Phase Filters

- are a special case of linear phase filters, where the phase delay is zero
- impulse response of a zero phase filter is even about time 0:

$$h(n) = h(-n)$$

therefore this filter cannot be causal

- a real, even impulse response corresponds to a real, even frequency response
- for an odd impulse response

$$h(n) = -h(-n)$$

the frequency response is purely imaginary

Zero Phase Filters

- are a special case of linear phase filters, where the phase delay is zero
- impulse response of a zero phase filter is even about time 0:

$$h(n) = h(-n)$$

  therefore this filter cannot be causal
- a real, even impulse response corresponds to a real, even frequency response
- for an odd impulse response

$$h(n) = -h(-n)$$

  the frequency response is purely imaginary

Zero Phase Filters

- are a special case of linear phase filters, where the phase delay is zero
- impulse response of a zero phase filter is even about time 0:

$$h(n) = h(-n)$$

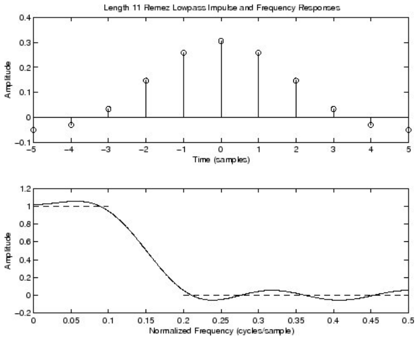
  therefore this filter cannot be causal
- a real, even impulse response corresponds to a real, even frequency response
- for an odd impulse response

$$h(n) = -h(-n)$$

  the frequency response is purely imaginary

Zero Phase Filters

Impulse and frequency response of a length 11 zero-phase FIR lowpass filter:



Symmetric Linear Phase Filters

- are derived from a delayed zero-phase filter
- are causal and symmetric about the midpoint:

$$h(n) = h(N-1-n), n = 0, 1, ..., N-1$$

- H_{ZP} is a zero-phase filter, N is odd:

$$h_{ZP}(n) = h(n - \frac{N-1}{2}), n = 0, 1, ..., N-1$$

$$H(z) = z^{-\frac{N-1}{2}} H_{ZP}(z)$$

$$H(e^{j\omega T}) = e^{-j\omega \frac{N-1}{2} T} H_{ZP}(e^{j\omega T})$$

Symmetric Linear Phase Filters

- are derived from a delayed zero-phase filter
- are causal and symmetric about the midpoint:

$$h(n) = h(N-1-n), n = 0, 1, ..., N-1$$

- H_{ZP} is a zero-phase filter, N is odd:

$$h_{ZP}(n) = h(n - \frac{N-1}{2}), n = 0, 1, ..., N-1$$

$$H(z) = z^{-\frac{N-1}{2}} H_{ZP}(z)$$

$$H(e^{j\omega T}) = e^{-j\omega \frac{N-1}{2} T} H_{ZP}(e^{j\omega T})$$

Symmetric Linear Phase Filters

- are derived from a delayed zero-phase filter
- are causal and symmetric about the midpoint:

$$h(n) = h(N-1-n), n = 0, 1, ..., N-1$$

- H_{ZP} is a zero-phase filter, N is odd:

$$h_{ZP}(n) = h(n - \frac{N-1}{2}), n = 0, 1, ..., N-1$$

$$H(z) = z^{-\frac{N-1}{2}} H_{ZP}(z)$$

$$H(e^{j\omega T}) = e^{-j\omega \frac{N-1}{2} T} H_{ZP}(e^{j\omega T})$$

Basic Principle

The basic principle of frequency masking is the following:

- in a linear phase model filter each delay is replaced by M delays
- this results in a periodic filter with much sharper transition bands
- finally a masking filter extracts the desired band

Basic Principle

The basic principle of frequency masking is the following:

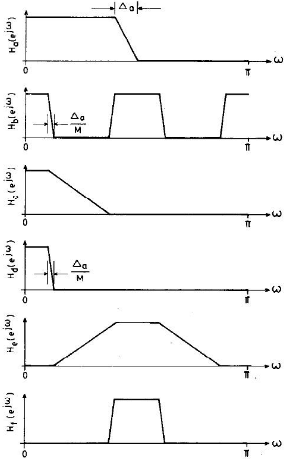
- in a linear phase model filter each delay is replaced by M delays
- this results in a periodic filter with much sharper transition bands
- finally a masking filter extracts the desired band

Basic Principle

The basic principle of frequency masking is the following:

- in a linear phase model filter each delay is replaced by M delays
- this results in a periodic filter with much sharper transition bands
- finally a masking filter extracts the desired band

Narrow Band Filter Design



Frequency Masking Principle:

- low-pass filter $H_a(z)$ with transition width Δ_a (model filter)
- replacing each delay by M delays: $H_b(z) = H_a(z^M)$
- masking filter $H_c(z)$
- resulting frequency response: $H_d(e^{j\omega}) = H_b(e^{j\omega})H_c(e^{j\omega})$ with transition width Δ_a/M
- masking filter $H_e(z)$
- resulting frequency response: $H_f(e^{j\omega}) = H_b(e^{j\omega})H_e(e^{j\omega})$

Narrow Band Filter Design

- This describes a method of deriving sharp filters (Δ_a/M) from filters with much wider transition band (Δ_a)
- Advantages: only a few coefficients in the model filter are nonzero, so the complexity is very low
- Problem: only suitable for narrow-band filters, because the passband bandwidth is reduced by the same factor

Narrow Band Filter Design

- This describes a method of deriving sharp filters (Δ_a/M) from filters with much wider transition band (Δ_a)
- Advantages: only a few coefficients in the model filter are nonzero, so the complexity is very low
- Problem: only suitable for narrow-band filters, because the passband bandwidth is reduced by the same factor

Narrow Band Filter Design

- This describes a method of deriving sharp filters (Δ_a/M) from filters with much wider transition band (Δ_a)
- Advantages: only a few coefficients in the model filter are nonzero, so the complexity is very low
- Problem: only suitable for narrow-band filters, because the passband bandwidth is reduced by the same factor

Arbitrary Bandwidth Design

Consider a filter F_c complementary to the masking filter F_a:

- z-transform of the symmetric linear phase filter F_a:

$$F_a(z) = z^{-\frac{N-1}{2}} F_{a,ZP}(z)$$

  where $F_{a,ZP}(z)$ is a zero-phase filter and N is odd

- the complementary filter F_c:

$$F_c(z) = z^{-\frac{N-1}{2}} (1 - F_{a,ZP}(z))$$

- this results in

$$F_c(z) = z^{-\frac{N-1}{2}} - F_a(z)$$

Arbitrary Bandwidth Design

Consider a filter F_c complementary to the masking filter F_a:

- z-transform of the symmetric linear phase filter F_a:

$$F_a(z) = z^{-\frac{N-1}{2}} F_{a,ZP}(z)$$

  where $F_{a,ZP}(z)$ is a zero-phase filter and N is odd

- the complementary filter F_c:

$$F_c(z) = z^{-\frac{N-1}{2}} (1 - F_{a,ZP}(z))$$

- this results in

$$F_c(z) = z^{-\frac{N-1}{2}} - F_a(z)$$

Arbitrary Bandwidth Design

Consider a filter F_c complementary to the masking filter F_a:

- z-transform of the symmetric linear phase filter F_a:

$$F_a(z) = z^{-\frac{N-1}{2}} F_{a,ZP}(z)$$

  where $F_{a,ZP}(z)$ is a zero-phase filter and N is odd

- the complementary filter F_c:

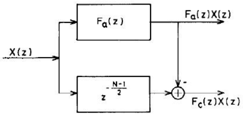
$$F_c(z) = z^{-\frac{N-1}{2}} (1 - F_{a,ZP}(z))$$

- this results in

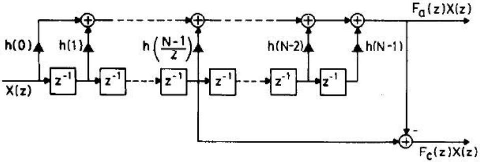
$$F_c(z) = z^{-\frac{N-1}{2}} - F_a(z)$$

Complementary Filter Pair

F_c can be implemented by subtracting the output of F_a from a delayed version of the input:

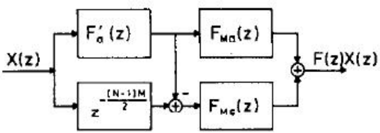


without extra delays:



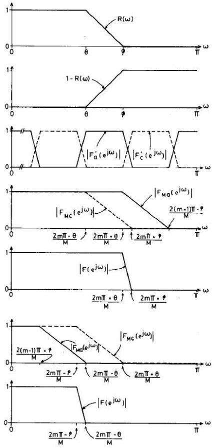
Masking Filters

If two masking filters, F_{Ma} and F_{Mc} for F_a and F_c, are used, it's possible to design wide-band sharp filters:



$$F(z) = F_a(z^M)F_{Ma}(z) + (z^{-\frac{N-1}{2}} - F_a(z^M))F_{Mc}(z)$$

Arbitrary Bandwidth Filter Design



Arbitrary Bandwidth Masking Principle:

- model filter F_a, cutoff frequencies θ and ϕ
- complementary filter F_c
- replacing
  each delay of F_a and F_c by M delays to get periodic, complementary model filters
- masking filters F_{Ma} and F_{Mc}
- resulting frequency response $F(e^{j\omega})$ with band edges ω_P and ω_S
- other masking filters F_{Ma} and F_{Mc}
- resulting frequency response $F(e^{j\omega})$

Arbitrary Bandwidth Filter Design

One can distinguish two cases:

- Case1: the frequency response of F near the transition band is determined mainly by F_a, pass- and stopband is defined by

$$\omega_P = \frac{2m\pi + \theta}{M}, \omega_S = \frac{2m\pi + \phi}{M}$$

- Case2: mainly determined by F_c, then pass- and stopband is defined by

$$\omega_P = \frac{2m\pi - \phi}{M}, \omega_S = \frac{2m\pi - \theta}{M}$$

Arbitrary Bandwidth Filter Design

One can distinguish two cases:

- Case1: the frequency response of F near the transition band is determined mainly by F_a, pass- and stopband is defined by

$$\omega_P = \frac{2m\pi + \theta}{M}, \omega_S = \frac{2m\pi + \phi}{M}$$

- Case2: mainly determined by F_c, then pass- and stopband is defined by

$$\omega_P = \frac{2m\pi - \phi}{M}, \omega_S = \frac{2m\pi - \theta}{M}$$

Synthesis Problem

In a synthesis problem the following has to be considered:

- ω_P and ω_S are given and m,M,θ,ϕ must be determined
- M should be choosen that the overall complexity of the filter is minimized
- this leads to an optimization problem:
  for increasing M the masking filters must be sharper (higher complexity) and the model filter can be broader (lower complexity)

Synthesis Problem

In a synthesis problem the following has to be considered:

- ω_P and ω_S are given and m,M,θ,ϕ must be determined
- M should be choosen that the overall complexity of the filter is minimized
- this leads to an optimization problem:
  for increasing M the masking filters must be sharper (higher complexity) and the model filter can be broader (lower complexity)

Synthesis Problem

In a synthesis problem the following has to be considered:

- ω_P and ω_S are given and m,M,θ,ϕ must be determined
- M should be choosen that the overall complexity of the filter is minimized
- this leads to an optimization problem:
  for increasing M the masking filters must be sharper (higher complexity) and the model filter can be broader (lower complexity)

Ripples of F

In this section the ripples of the overall filter $F(e^{j\omega})$ are analyzed:

- let $G(\omega)$ be the desired value and $\delta(\omega)$ the deviation from this value for each filter
- for $F(e^{j\omega})$ this leads to:

$$G(\omega) + \delta(\omega) = (G_{Ma}(\omega) + \delta_{Ma}(\omega))(G_a(\omega) + \delta_a(\omega))$$
$$+(G_{Mc}(\omega) + \delta_{Mc}(\omega))(1 - G_a(\omega) - \delta_a(\omega))$$

- we examine the effects of F_a, F_{Ma} and F_{Mc} in three frequency ranges

Ripples of F

In this section the ripples of the overall filter $F(e^{j\omega})$ are analyzed:

- let $G(\omega)$ be the desired value and $\delta(\omega)$ the deviation from this value for each filter
- for $F(e^{j\omega})$ this leads to:

$$G(\omega) + \delta(\omega) = (G_{Ma}(\omega) + \delta_{Ma}(\omega))(G_a(\omega) + \delta_a(\omega))$$
$$+(G_{Mc}(\omega) + \delta_{Mc}(\omega))(1 - G_a(\omega) - \delta_a(\omega))$$

- we examine the effects of F_a, F_{Ma} and F_{Mc} in three frequency ranges

Ripples of F

In this section the ripples of the overall filter $F(e^{j\omega})$ are analyzed:

- let $G(\omega)$ be the desired value and $\delta(\omega)$ the deviation from this value for each filter
- for $F(e^{j\omega})$ this leads to:

$$G(\omega) + \delta(\omega) = (G_{Ma}(\omega) + \delta_{Ma}(\omega))(G_a(\omega) + \delta_a(\omega))$$
$$+(G_{Mc}(\omega) + \delta_{Mc}(\omega))(1 - G_a(\omega) - \delta_a(\omega))$$

- we examine the effects of F_a, F_{Ma} and F_{Mc} in three frequency ranges

Ripples, Fr. Range 1 + 2

- *Frequency Range 1:* $G_{Ma}(\omega) = G_{Mc}(\omega) = 1$ (passband)

$$G(\omega) = 1$$

$$G_a(\omega) = 1, \delta(\omega) \approx \delta_{Ma}(\omega)$$

$$G_a(\omega) = 0, \delta(\omega) \approx \delta_{Mc}(\omega)$$

- *Frequency Range 2:* $G_{Ma}(\omega) = G_{Mc}(\omega) = 0$ (stopband)

$$G(\omega) = 0$$

$$G_a(\omega) = 1, \delta(\omega) \approx \delta_{Ma}(\omega)$$

$$G_a(\omega) = 0, \delta(\omega) \approx \delta_{Mc}(\omega)$$

Ripples, Fr. Range 1 + 2

- *Frequency Range 1:* $G_{Ma}(\omega) = G_{Mc}(\omega) = 1$ (passband)

$$G(\omega) = 1$$

$$G_a(\omega) = 1, \delta(\omega) \approx \delta_{Ma}(\omega)$$

$$G_a(\omega) = 0, \delta(\omega) \approx \delta_{Mc}(\omega)$$

- *Frequency Range 2:* $G_{Ma}(\omega) = G_{Mc}(\omega) = 0$ (stopband)

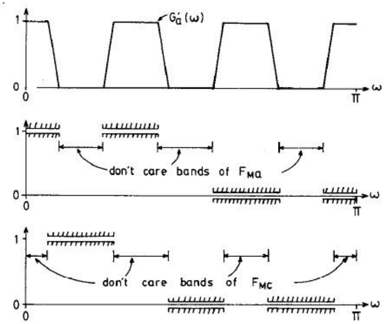
$$G(\omega) = 0$$

$$G_a(\omega) = 1, \delta(\omega) \approx \delta_{Ma}(\omega)$$

$$G_a(\omega) = 0, \delta(\omega) \approx \delta_{Mc}(\omega)$$

Don't Care Bands

Therefore F_{Ma} and F_{Mc} could be interpreted as low-pass filters with don't care bands within their pass- and stopbands:



These don't care bands help to reduce the complexity of the masking filters.

Ripples, Fr. Range 3

Frequency Range 3: $G_{Ma}(\omega) \neq G_{Mc}(\omega)$, transition band

- here $\delta(\omega)$ is a function of $\delta_a(\omega)$, $\delta_{Ma}(\omega)$ and $\delta_{Mc}(\omega)$
- however, it is possible to design the filter F_a such that $\delta_a(\omega)$ partially compensates $\delta_{Ma}(\omega)$ and $\delta_{Mc}(\omega)$

Ripples, Fr. Range 3

Frequency Range 3: $G_{Ma}(\omega) \neq G_{Mc}(\omega)$, transition band

- here $\delta(\omega)$ is a function of $\delta_a(\omega)$, $\delta_{Ma}(\omega)$ and $\delta_{Mc}(\omega)$
- however, it is possible to design the filter F_a such that $\delta_a(\omega)$ partially compensates $\delta_{Ma}(\omega)$ and $\delta_{Mc}(\omega)$

Optimization of F

F_a has to be designed to compensate for $\delta_{Ma}(\omega)$ and $\delta_{Mc}(\omega)$

- a linear equation relating $\delta(\omega)$ and F_a must be obtained:

$$\delta(\omega) = F_{a,ZP}(M\omega)(G_{Ma}(\omega) + \delta_{Ma}(\omega) - G_{Mc}(\omega) - \delta_{Mc}(\omega))$$

$$+G_{Mc}(\omega) + \delta_{Mc}(\omega) - G(\omega)$$

- now the minimization of $|\delta(\omega)|$ in the transition band is a linear programming filter design problem and can be solved by a standard mathematical programming package

Optimization of F

F_a has to be designed to compensate for $\delta_{Ma}(\omega)$ and $\delta_{Mc}(\omega)$

- a linear equation relating $\delta(\omega)$ and F_a must be obtained:

$$\delta(\omega) = F_{a,ZP}(M\omega)(G_{Ma}(\omega) + \delta_{Ma}(\omega) - G_{Mc}(\omega) - \delta_{Mc}(\omega))$$
$$+G_{Mc}(\omega) + \delta_{Mc}(\omega) - G(\omega)$$

- now the minimization of $|\delta(\omega)|$ in the transition band is a linear programming filter design problem and can be solved by a standard mathematical programming package

Optimization of M

- there is no closed-form analytic expression for finding the optimum M (in the paper from 1986)
- a good choice of M can be obtained by esimating the filter complexity for each M (nonzero multipliers) and then selecting the M which corresponds to the lowest estimate
- however, many more recent papers address the right selection of the parameter M and suggest optimized designs

Optimization of M

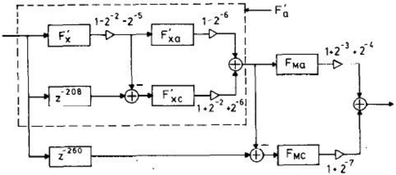
- there is no closed-form analytic expression for finding the optimum M (in the paper from 1986)
- a good choice of M can be obtained by esimating the filter complexity for each M (nonzero multipliers) and then selecting the M which corresponds to the lowest estimate
- however, many more recent papers address the right selection of the parameter M and suggest optimized designs

Optimization of M

- there is no closed-form analytic expression for finding the optimum M (in the paper from 1986)
- a good choice of M can be obtained by esimating the filter complexity for each M (nonzero multipliers) and then selecting the M which corresponds to the lowest estimate
- however, many more recent papers address the right selection of the parameter M and suggest optimized designs

Multistage Frequency Response Masking Design

The model and masking filters may again be synthesized using the frequency response masking technique, producing a multistage frequency response masking design:



Optimizing this technique is again subject of many more recent papers.

Powers-of-Two Design Technique

- the complexity of the filter may be further reduced by constraining all the coefficient values to be a sum or difference of two powers-of-two using the powers-of-two design technique
- in this case, the multiplication can be performed just by using shifts and adds

Powers-of-Two Design Technique

- the complexity of the filter may be further reduced by constraining all the coefficient values to be a sum or difference of two powers-of-two using the powers-of-two design technique
- in this case, the multiplication can be performed just by using shifts and adds

Single-Stage Design

- The single-stage FRM low-pass filter, using the powers-of-two design technique, should meet the following specifications:
- bandedges at 0.3 and 0.305 sampling frequencies, maximum passband deviation is 0.1 dB and minimum stopband attenuation is -40 dB
- this filter requires 202 shift-add operations per sampling interval, whereas the infinite precision minimax optimum design requires 383 multiply and 382 add operations

Single-Stage Design

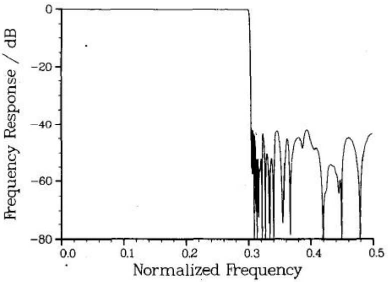
- The single-stage FRM low-pass filter, using the powers-of-two design technique, should meet the following specifications:
- bandedges at 0.3 and 0.305 sampling frequencies, maximum passband deviation is 0.1 dB and minimum stopband attenuation is -40 dB
- this filter requires 202 shift-add operations per sampling interval, whereas the infinite precision minimax optimum design requires 383 multiply and 382 add operations

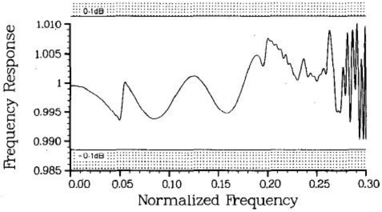
Single-Stage Design

- The single-stage FRM low-pass filter, using the powers-of-two design technique, should meet the following specifications:
- bandedges at 0.3 and 0.305 sampling frequencies, maximum passband deviation is 0.1 dB and minimum stopband attenuation is -40 dB
- this filter requires 202 shift-add operations per sampling interval, whereas the infinite precision minimax optimum design requires 383 multiply and 382 add operations

Single-Stage Design Frequency Response

Frequency Response of the single-stage FRM low-pass filter:





Multi-Stage Design

- Now a multi-stage FRM low-pass filter with the following specifications should be designed:
- bandedges at 0.2 and 0.2001 sampling frequencies, maximum passband deviation is 0.05 dB and minimum stopband attenuation is -50 dB
- a five stage design was used with $M1 = M2 = M3 = M4 = 4$ and $M5 = 3$
- the total number of multipliers is 125, whereas the infinite precision minimax optimum design requires 12055 multiplications (!)

Multi-Stage Design

- Now a multi-stage FRM low-pass filter with the following specifications should be designed:
- bandedges at 0.2 and 0.2001 sampling frequencies, maximum passband deviation is 0.05 dB and minimum stopband attenuation is -50 dB
- a five stage design was used with $M1 = M2 = M3 = M4 = 4$ and $M5 = 3$
- the total number of multipliers is 125, whereas the infinite precision minimax optimum design requires 12055 multiplications (!)

Multi-Stage Design

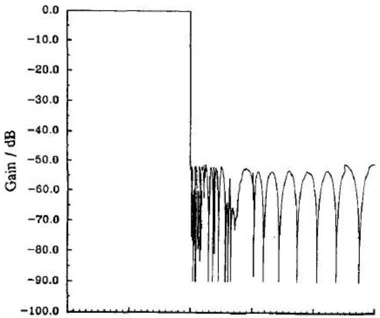
- Now a multi-stage FRM low-pass filter with the following specifications should be designed:
- bandedges at 0.2 and 0.2001 sampling frequencies, maximum passband deviation is 0.05 dB and minimum stopband attenuation is -50 dB
- a five stage design was used with $M1 = M2 = M3 = M4 = 4$ and $M5 = 3$
- the total number of multipliers is 125, whereas the infinite precision minimax optimum design requires 12055 multiplications (!)

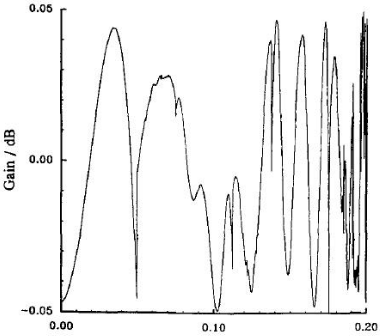
Multi-Stage Design

- Now a multi-stage FRM low-pass filter with the following specifications should be designed:
- bandedges at 0.2 and 0.2001 sampling frequencies, maximum passband deviation is 0.05 dB and minimum stopband attenuation is -50 dB
- a five stage design was used with $M1 = M2 = M3 = M4 = 4$ and $M5 = 3$
- the total number of multipliers is 125, whereas the infinite precision minimax optimum design requires 12055 multiplications (!)

Multi-Stage Design Frequency Response

Frequency Response of the multi-stage FRM low-pass filter:





Conclusion

- in the frequency-masking technique a model filter and its complementary filter is generated
- then each delay of these filters is replaced by M delays
- this results in periodic, complementary model filters with much sharper transition bands
- finally two masking filter extract the desired band

Conclusion

- in the frequency-masking technique a model filter and its complementary filter is generated
- then each delay of these filters is replaced by M delays
- this results in periodic, complementary model filters with much sharper transition bands
- finally two masking filter extract the desired band

Conclusion

- in the frequency-masking technique a model filter and its complementary filter is generated
- then each delay of these filters is replaced by M delays
- this results in periodic, complementary model filters with much sharper transition bands
- finally two masking filter extract the desired band

Conclusion

- in the frequency-masking technique a model filter and its complementary filter is generated
- then each delay of these filters is replaced by M delays
- this results in periodic, complementary model filters with much sharper transition bands
- finally two masking filter extract the desired band

Conclusion

- this technique allows one to design sharp low-pass, high-pass, bandpass and bandstop linear phase filters with arbitrary passband bandwidth
- the complexity of the resulting filter is very low, because only a small fraction of its coefficients are nonzero
- with additional optimization methods (like multiplierless or multi-stage design), which is subject of many recent papers, it is possible to reduce the complexity even more

Conclusion

- this technique allows one to design sharp low-pass, high-pass, bandpass and bandstop linear phase filters with arbitrary passband bandwidth
- the complexity of the resulting filter is very low, because only a small fraction of its coefficients are nonzero
- with additional optimization methods (like multiplierless or multi-stage design), which is subject of many recent papers, it is possible to reduce the complexity even more

Conclusion

- this technique allows one to design sharp low-pass, high-pass, bandpass and bandstop linear phase filters with arbitrary passband bandwidth
- the complexity of the resulting filter is very low, because only a small fraction of its coefficients are nonzero
- with additional optimization methods (like multiplierless or multi-stage design), which is subject of many recent papers, it is possible to reduce the complexity even more

References

- Yong Ching Lim; *Frequency-Response Masking Approach for the Synthesis of Sharp Linear Phase Digital Filters*; 1986, IEEE transactions on circuits and systems
- Julius O. Smith; *Introduction to Digital Filters*; 2006, Center for Computer Research in Music and Acoustics (CCRMA), Stanford University
- Yong Ching Lim and Yong Lian; *The Optimum Design of One- and Two-Dimensional FIR Filters Using the Frequency Response Masking Technique*; 1986, IEEE transactions on circuits and systems

Questions

Questions ... ?