

Sharp Transition Multiband Filter in Speech Processing Scheme for Hearing Impaired

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Abstract: Problem statement: A simple analytical approach to the synthesis of a sharp transition, linear phase, multiband FIR filter is presented. The filter magnitude response is modeled using trigonometric functions of frequency. **Approach:** Employing variable density of ripple cycles in passband and stopband regions with large density of ripples cycles at the sharp transition edges, reduces the abrupt discontinuities at these edges. **Results:** As a result, Gibb's phenomenon is reduced in the filter implementation thus giving a flat passband and good stopband attenuation. A closed form expression for impulse response coefficients is obtained. The filter design is easily tunable and allows for variation in transition bandwidth of each band. A speech processing scheme is implemented using a pair of the proposed sharp transition multiband FIR filters to split the speech spectrum into complementary short time spectral bands. **Conclusion:** The adjacent speech formants are fed dichotically to the two ears to reduce the effect of spectral masking and hence improve speech perception in sensorineural hearing impaired.

Key words: Sharp transition, FIR filter, multiband filter, gibb's phenomenon, sensorineural hearing impaired, speech processing scheme

INTRODUCTION

The peripheral auditory system of the human ear behaves like a bank of bandpass filters called auditory filters or critical bands (Moore, 2003; Flanagan, 1972; Zwicker, 1961). One of the hearing impairment in humans is sensorineural hearing loss due to defects in the cochlea and auditory nerve. In general, this loss is not medically curable and it becomes progressively worse with time. Sensorineural loss is associated with widening of the auditory filter bandwidths thus the filter slope becomes shallower and results in overlap of adjacent spectral bands called spectral masking (Boer and Bouwmeester, 1974; Glasberg and Moore, 1986; French and Steinberg, 1947). This leads to a decrease in frequency resolving capacity of the auditory system of the ears. One of the possible ways of improving speech perception would be to split the speech signal on the basis of its short time spectrum into two complimentary spectra for presentation to the two ears dichotically. This enhances the perception of spectral contrast of resonance peaks without adversely affecting the

perception of features cued by amplitude and duration. The two adjacent strong spectral components i.e. formants that are likely to mask each other are presented to left and right ears. Another technique to reduce effects of spectral masking is to employ multiband frequency compression (Kulkarni *et al.*, 2007). In this technique, the speech spectrum is divided into a number of analysis bands and spectral samples in each of these bands are compressed towards the band center by a constant compression factor.

Several schemes are employed in literature to split the speech signals for binaural dichotic presentation. Lunner *et al.* (1993) evaluated a scheme using synthesized vowels in which the first formant is presented to left ear and the second formant to the right ear. The scheme used comb filters with constant bandwidth of 700Hz and 40dB stopband attenuation. The listening tests were conducted on male and female subjects in age group of 39-69 years with bilateral and moderate sensorineural hearing loss. The gain of the filters was adjusted depending on the hearing loss of

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individual subject. The scheme achieved spectral splitting but the filters did not have the sharp transition bands. The results indicated an overall improvement in speech perception for dichotic conditions. Lyregaard (1982) conducted similar experiments with three values of comb filter bandwidths of 200 Hz, 500Hz and 800Hz. There was no significant improvement in speech recognition scores because of improper band separation. Chaudari and Pandey (1998) investigated a scheme for splitting the speech into two complementary spectra each with nine bands for binaural dichotic presentation using Zwicker (1961) psychophysical tuning curves. The FIR comb filters were designed by frequency sampling method using linear optimization techniques. The filter transition bandwidth was 78 to 117 Hz for various bands which resulted in maximum pass-band ripple of 1dB and stopband attenuation of 38 dB with cascade filters of order 256. The scheme was able to improve speech quality, response time, recognition scores and transmission of consonantal features as observed in the listening tests. The critical-band filter banks for use in binaural hearing aid reported in literature generally uses lowpass, bandpass and highpass filter in cascade. These filters have a wide transition band, large passband ripple, less stopband attenuation and not easily tunable. Employing filters with large transition band, the spectral components lying in the transition region are presented to both ears which impair speech intelligibility (Chaudari and Pandey, 1998).

A multiband sharp transition filter with large number of passbands and acceptable magnitude response is difficult to design and requires large filter order (Pelloni and Lagadec, 1984). Most methods do not allow independent control of band specifications mainly the transition bandwidth. Conventional methods for sharp transition FIR filters using window functions and frequency sampling method require high filter order. The numeric Remez approach, which employs optimization techniques, requires lower filter order but computational complexity is high. They do not yield an explicit expression for the filter transfer function and hence it is difficult tuning to tailor made magnitude responses desired in some applications like the one dealt in this study. Most of the numeric FIR multiband filter approaches (Rabiner *et al.*, 1974; Zahradnik and Vleek, 2005; Mintzer and Liu, 1979; Matei, 2007; Shpak and Antoniou, 1988; Selesnick *et al.*, 1998; Burrus, 1994; Estola, 1988; Samadi *et al.*, 2004; Ho *et al.*, 2008) are based on the least squares or equiripple error criteria. A Remez type exchange algorithm with an initialization strategy and a selective search for the

location of the error maxima magnitude to reduce the computation complexity is presented by Shpak and Antoniou (1988). A simple and effective multiple-exchange algorithm for the design of multiband FIR filter is presented by Selesnick *et al.* (1998) using a constrained least square approach that does not use specified transition bands. The design algorithm is a hybrid of the Parks-McClellan and least squares methods. Two methods which are a combination of numeric and analytical techniques are presented by Burrus (1994) for designing optimal linear phase multiband FIR digital filters that have well defined transition bands. They both allow explicit control of band edges and center band gains for multiple passband and independent control of each transition band edge. One is an analytical method which derives formulas for the filter coefficients but does not allow different weighting of the passbands, stopbands or transition bands. The second approach uses a mixed numericalanalytical method which allows independent weighting of all bands but requires numerical solution of a set of simultaneous equations. A general method is presented by Estola (1988) for designing computationally efficient FIR multiband filters with arbitrary frequency response and time response. These filters are constructed by cascading subfilters and solving the nonlinear approximation problem to obtain filter coefficients. A linear phase maximally flat multiband filter is presented by Samadi *et al.* (2004). The structure is easily tunable and provides a simple mechanism to vary both the centers and degrees of flatness independently at each level. A serious disadvantage of FIR filters is its complexity, which becomes acute in sharp transition filters. A computationally efficient design technique for synthesis of sharp transition FIR filters is Interpolated FIR filter which is a FIR-FIR filter cascade and employs sparse filter and an interpolator filter (Neuvo *et al.*, 1987; 1984; Jovanovic-Dolecek and Diaz-Carmona, 2002; Yang and Lian, 2002; Gudino *et al.*, 2008a; 2008b; 2008c). The interpolation designs introduce large group delays. One of the most successful techniques for the synthesis of arbitrary passband, very narrow transition width FIR filter with least filter order is the Frequency Response Masking (FRM) technique (Lim, 1986; Rodrigues and Pai, 2005a; Lim *et al.*, 2007). The sharp transition and arbitrary bandwidth is formed by using several low order FIR subfilters. Hearing aids using a bank of computationally efficient FRM filters but with wider transition band are presented by Lian and Wei (2005).

MATERIALS AND METHODS

Proposed multiband filter design: In the proposed linear phase FIR multiband filter design, the various regions of the filter magnitude response are modeled using trigonometric functions of frequency. The magnitude response model $H_{pm}(\omega)$ of the multiband filter shown in Fig. 1 has nine regions per band. BP_n is the n th bandpass component of the multiband filter.

In the stopband region from $\omega_{s,n-1} \leq \omega \leq \omega_{s2,n-1}$ the frequency response is Eq. 1:

$$H_{pm}(\omega) = \frac{\delta_s}{2} \cos k_{p2}(\omega - \omega_{s,n-1}) \quad (1)$$

where, ω is the frequency variable, δ_s is the stopband attenuation, $\omega_{s,n-1}$ is the center frequency of the stopband preceding a passband of bandpass response BP_n and $\omega_{s2,n-1}$ is the frequency at which second stopband region starts. This region is designed to possess $(n0-p)$ ripple cycles characterized by filter design parameter k_{p2} where $n0$ and p are integers to be chosen. The various filter design parameters of the passband, stopband and transition bands shape the response in the respective regions Eq.2:

$$\omega_{s2,n-1} = \omega_{s,n-1} + (n0-p) \frac{2\pi}{k_{p2}} \quad (2)$$

For the filter model, $H_{pm}(\omega_{bn} \pm \omega_{pn}) = 1$ and $H_{pm}(\omega_{bn} \pm \omega_{zn}) = 0$.

In the stopband region, $\omega_{s2,n-1} \leq \omega \leq (\omega_{bn} - \omega_{zn})$ the frequency response is Eq. 3:

$$H_{pm}(\omega) = \frac{\delta_s}{2} \cos k_p(\omega - \omega_{s2,n-1}) \quad (3)$$

where, ω_{bn} is the center frequency of passband BP_n . This region possesses $(p+3/4)$ cycles of ripple characterized by filter design parameter k_p .

For the filter model Eq. 4 and 5:

$$(\omega_{bn} - \omega_{zn}) - \omega_{s,n-1} = \left(p + \frac{3}{4}\right) \frac{2\pi}{k_p} + (n0-p) \frac{2\pi}{k_{p2}} \quad (4)$$

$$\omega_{s2,n-1} = (\omega_{bn} - \omega_{zn}) - \left(p + \frac{3}{4}\right) \frac{2\pi}{k_p} \quad (5)$$

In the transition region, $(\omega_{bn} - \omega_{zn}) \leq \omega \leq (\omega_{bn} - \omega_{pn})$, the frequency response is Eq. 6:

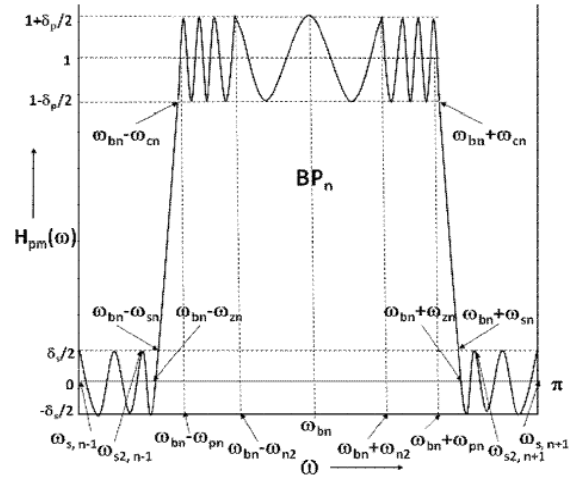


Fig. 1: Illustration of one band of the proposed ‘n’ band multiband FIR filter with equiripple magnitude response, linear transition and variable density ripple cycles

$$H_{pm}(\omega) = \frac{1}{(\omega_{zn} - \omega_{pn})} [\omega - (\omega_{bn} - \omega_{zn})] \quad (6)$$

where, $\omega_{bn} \pm \omega_{pn}$ are the passband edge frequencies and ω_{pn} is half passband width of BP_n .

In the passband region, $(\omega_{bn} - \omega_{pn}) \leq \omega \leq (\omega_{bn} - \omega_{n2})$, the frequency response is Eq. 7:

$$H_{pm}(\omega) = 1 + \frac{\delta_p}{2} \sin k_p [\omega - (\omega_{bn} - \omega_{pn})] \quad (7)$$

where, $(\omega_{bn} - \omega_{n2})$ is the frequency at which second passband region starts. This region possesses $(m+1/4)$ cycles of ripple and m is an integer to be chosen.

In the passband region, $(\omega_{bn} - \omega_{n2}) \leq \omega \leq (\omega_{bn} - \omega_{n2})$, the frequency response is Eq. 8:

$$H_{pm}(\omega) = 1 + \frac{\delta_p}{2} \cos k_{p1} [\omega - (\omega_{bn} - \omega_{n2})] \quad (8)$$

where, $(\omega_{bn} + \omega_{n2})$ is the frequency at which third passband region starts. This region possesses $(n1-2m)$ cycles of ripples and is characterized by the filter design parameter k_{p1} . $n1$ is an even integer to be chosen.

For the filter model Eq. 9:

$$\omega_{n2} = \left(\frac{n1}{2} - m\right) \frac{2\pi}{k_{p1}} \quad (9)$$

In the passband region, $(\omega_{bn} + \omega_{n2}) \leq \omega \leq (\omega_{bn} + \omega_{pn})$, the frequency response is Eq. 10:

$$H_{pm}(\omega) = 1 + \frac{\delta_p}{2} \cos k_p [\omega - (\omega_{bn} + \omega_{n2})] \quad (10)$$

This region is designed to possess $(m+1/4)$ cycles of ripple Eq. 11:

$$\omega_{n2} = \omega_{pn} - \left(m + \frac{1}{4}\right) \frac{2\pi}{k_p} \quad (11)$$

In the transition region, $(\omega_{bn} + \omega_{pn}) \leq \omega \leq (\omega_{bn} + \omega_{zn})$, the frequency response is Eq. 12:

$$H_{pm}(\omega) = 1 - \frac{1}{(\omega_{zn} - \omega_{pn})} [\omega - (\omega_{bn} + \omega_{pn})] \quad (12)$$

In the stopband region, $(\omega_{bn} + \omega_{zn}) \leq \omega \leq \omega_{s2,n-1}$, the frequency response is Eq. 13:

$$H_{pm}(\omega) = -\frac{\delta_s}{2} \sin k_p [\omega - (\omega_{bn} + \omega_{zn})] \quad (13)$$

where, $\omega_{s2,n-1}$ the frequency at which third region of the stopband starts. This region is designed to possess $(q+3/4)$ cycles of ripple characterized by k_p and q is an integer is to be chosen.

In the stopband region, $\omega_{s2,n-1} \leq \omega \leq \omega_{s,n-1}$, the frequency response is Eq. 14:

$$H_{pm}(\omega) = \frac{\delta_s}{2} \sin k_{p3} (\omega - \omega_{s2,n+1}) \quad (14)$$

$\omega_{s,n-1}$ is the center frequency of the stopband succeeding the band BPn. This region is designed to possess $(n2-q)$ cycles of ripple characterized by filter design parameter k_{p3} and $n2$ is an integer to be chosen. For the filter model Eq.15:

$$\omega_{s,n-1} - (\omega_{bn} + \omega_{zn}) = \left(q \frac{3}{4}\right) \frac{2\pi}{k_p} \quad (15)$$

$$(n_2 - q) \frac{2\pi}{k_{p3}}$$

For the filter model Eq. 16-18:

$$\omega_{s,n-1} - (\omega_{bn} + \omega_{zn}) = \left(q \frac{3}{4}\right) \frac{2\pi}{k_p} \quad (16)$$

$$\omega_{cn} = \frac{\delta_p}{2} (\omega_{zn} - \omega_{pn}) + \omega_{pn} \quad (17)$$

$$\omega_{sn} - \omega_{zn} - (\delta_s / 2)(\omega_{zn} - \omega_{pn}) \quad (18)$$

The transition bandwidth of each band is Eq. 19:

$$(\omega_{sn} - \omega_{cn}) = (\omega_{zn} - \omega_{pn}) \left(1 - \frac{\delta_2 + \delta_p}{2}\right) \quad (19)$$

For the multiband design, novel slope equalization technique (Rodrigues and Pai, 2005b) is applied which further improves the performance of the filter. The filter design parameter k_p is evaluated by equalizing the slopes of the magnitude response function at the edges of the transition region which are the points of discontinuity. This allows the function to be continuous thus reducing the effects due to Gibb's phenomenon and hence improves performance of the filter Eq. 20 and 21:

$$\text{Slope of } H_{pm}(\omega) \text{ at } \omega_{bn} \pm \omega_{pn} = -\frac{k_p \delta_p}{2} \quad (20)$$

$$\text{Slope of } H_{pm}(\omega) \text{ at } \omega_{bn} \pm \omega_{zn} = -(-1/\omega) \Big|_{\omega_{zn} - \omega_{pn}} \quad (21)$$

Equating the slopes at $(\omega_{bn} + \omega_{pn})$ and $(\omega_{bn} + \omega_{zn})$ at the points of discontinuity for each band of the multiband filter and using (20),(21) and (19) we obtain Eq. 22:

$$k_p = \frac{2 \left[1 - \left(\frac{\delta_s + \delta_p}{2} \right) \right]}{\delta_p (\omega_{sn} - \omega_{cn})} \quad (22)$$

This value of k_p further reduces ripples at the transition edges hence reduces passband ripple and improves stopband attenuation.

The expression for the impulse response coefficients for the N order multiband filter is obtained by evaluating the integral Eq. 23:

$$h(n_i) = \frac{1}{\pi} \int_0^\pi H_{pm}(\omega) \cos k\omega \, d\omega \quad (23)$$

where, $H_{pm}(\omega)$ in 23 is made up of nine regions for each passband as defined by 1, 3, 6-8, 10, 12-14.

$n_i = 0, 1, \dots, \frac{N-1}{2}$ for N odd and $k = \frac{(N-1)}{2} - n_i$.

Using Eq. 23, the symmetric impulse response coefficients for a given filter order N can be obtained.

RESULTS

Design two linear phase, sharp transition FIR multiband filters using the proposed design approach to be employed in a speech processing scheme for sensorineural hearing impaired. The speech samples are

split into two complementary short-time spectra using the multiband FIR filters with passbands based on Zwicker's critical band model (Zwicker, 1961). Each multiband filter has nine passbands with passband edge frequencies, $\omega_{bn} \pm \omega_{pn}$ in KHz being [0.07-0.2, 0.30-0.40, 0.51-0.63, 0.77-0.92, 1.08-1.27, 1.48-1.72, 2.00-2.32, 2.70-3.15, 3.70-4.40] and [0.2-0.3, 0.40-0.51, 0.63-0.77, 0.92-1.08, 1.27-1.48, 1.72-2.00, 2.32-2.70, 3.15-3.70, 4.40-5.012], sampling frequency of 11025 Hz, transition bandwidth, $\omega_{sn} - \omega_{cn}$ of 35 Hz, maximum passband ripple δ_p is ± 0.1 dB and minimum stopband attenuation δ_s is 40dB.

The specification of each band i.e., passband center frequency ω_{bn} , half passband width ω_{pn} , the center frequencies of the stopbands preceding and succeeding B_{pn} , $\omega_{s,n-1}$ and $\omega_{s,n+1}$ can be found from the passband values given above. For the given filter specifications, we compute values of k_p , ω_{zn} , ω_{n2} , k_{p1} , $\omega_{s2,n-1}$, k_{p2} , $\omega_{s2,n+1}$ and k_{p3} using Eqs. 22, 19, 11, 9, 5, 2, 16 and 15 respectively. The value of the chosen constants m , $n1$, p , q , $n0$ and $n2$ depends on the density of the ripples required in the various passbands and stopbands of the multiband filter. In the model, where sharp edges join, like the transition and first passband region, the ripples are increased. Hence a higher value of the constant is chosen in this region. The number of constants depends on the number of passbands. The approximate relation between various constants is $n1 > 2*m$, $p = q$, $n0 > p$, q , $n2 > p, q$. We choose $m = 30$, $n1=70$, $p = q = 5$, $n0 = 7$ and $n2 = 15$. The magnitude response of the multiband filter obtained is shown in Fig. 2. The peak ripple at the transition edges due to Gibb's phenomenon reduces. The proposed design yields the desired multiband FIR filter specifications for a filter order of 1025. The individual bands have a passband ripple of 0.01 dB, stopband attenuation exceeding 40 dB and a sharp transition band of 35 Hz i.e., normalized frequency of 0.00635. In a similar speech processing scheme, (Kulkarni and Pandey, 2008) employs two comb FIR filters with transition bandwidth of 78-117Hz, passband ripple < 1 dB and stopband attenuation exceeding 30 dB.

Listening tests were carried out on five subjects with bilateral moderate sensorineural hearing impairment. The test stimuli consisted of nonsense syllables formed with English consonants and vowel in vowel consonant vowel and consonant vowels contexts. Signal acquisition, administration of speech syllables and analysis was performed offline. Subjects showed an improvement in recognition of the speech syllables presented to the ears.

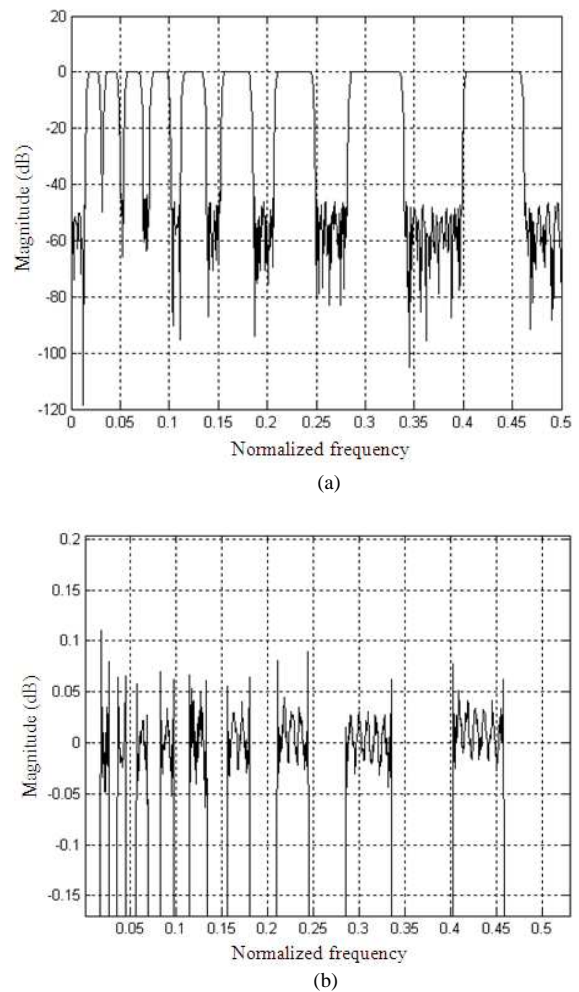


Fig. 2: (a) Magnitude response of the proposed FIR multiband filters for the design example (b) Magnified view of passband

For all the subjects the recognition scores for processed speech were higher than those for unprocessed speech. Response time to respond to speech syllables presented to the ears reduced which indicates effective speech processing. The processed speech was spectrographically analyzed to verify splitting of speech into various frequency bands. It was found that the adjacent formant frequencies of the speech syllables were presented to alternate ears thus aided in better recognition of speech using the proposed filters in the speech processing scheme.

DISCUSSION

FRM approach for a design of multiband filter with large number of passbands is complicated since many subfilters are to be designed. Also large delays

introduced are undesired in real time applications. Synthesis of sharp transition FIR lowpass filters is introduced by Rodrigues and Pai, (2005a), Rodrigues *et al.* (2006). The simple analytical methods give narrow transition band with flat passband, good stopband attenuation with less filter order. The filter designs can be extended to the design of bandpass and multiband FIR filters. In this study, we propose the synthesis of sharp transition multiband FIR filter. The design permits easy tuning of the center frequency and transition bandwidths of individual bands.

CONCLUSION

This study presents the synthesis of a sharp transition, linear phase, multiband FIR filter. The filter is easily tunable, the ripples at the transition edges due to sharp transition are reduced, the design does not employ optimization techniques and we can implement tailor made filter magnitude responses. The specifications of each band i.e. center frequency, passband width, transition width and stopband attenuation can be easily varied. Proposed multiband filters have good stopband attenuation which aids in proper band separation and low passband ripple avoids deterioration of processed speech. Employing sharp transition filters aids in proper separation of adjacent formant frequencies fed to the ears. This leads to improvement in recognition of speech for sensorineural hearing impaired subjects as observed in the experimental study conducted.

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