Chapter 1

INTRODUCTION

1.1 Problem overview

The signal processing in general and digital signal processing in particular, finds a plethora of applications in the field of biomedical engineering. The application intended in this thesis is related in broader terms, to the field of medical prosthesis, the prosthetic device being a digital hearing aid. The research work reported in this thesis addresses a particular hearing impairment called sensorineural hearing impairment [1], [2] and FIR filters for speech processing scheme in a digital hearing aid.

Sensorineural hearing impairment is characterized by marked distortions or abnormalities in sound perception. Its major causes are the damaged hair cells in the cochlea and/or degeneration of auditory nerve fibers of the ear. The result is increased spectral masking, which in turn causes severe smearing of the spectral envelope of speech signals. The smearing is equivalent to artificially widening of auditory filters. The debilitating effects of this impairment are high frequency hearing loss, increase in the threshold of hearing, compression in the dynamic range, severity of temporal masking and loss of spectral resolution due to spread of masking.

A proven speech processing scheme, that improves the speech perception among the persons suffering from the sensorineural hearing impairment, consists of splitting the speech spectrum into different frequency bands based on a critical band (auditory filter) model proposed by Zwicker [3]. The splitting of speech spectrum is obviously carried out by means of a set of two filter banks generating a pair of complementary spectra. Each of these spectra is
separately presented to left and right ear. This is known as binaural dichotic presentation [1], which helps in reducing the effect of spectral masking and is likely to improve speech perception in cases of bilateral sensorineural hearing impaired persons. Numerous studies were undertaken to verify the concept of critical bands and the concept is extended to other aspects of auditory perception and the frequency resolving capabilities of the human auditory system. The subdivision of the speech signal spectrum into critical bands seems to be correlated very closely to the cochlear mechanics and to frequency discrimination [4]. Zwicker’s critical band model was the first comprehensive model of its type and has been used by many researchers. Studebaker et. al. [5] have proposed a model, which shows improvement in intelligibility at lower cutoff frequencies. These critical band models point to the need of synthesizing a multiband filter consisting of several pass and stopbands with near to ideal frequency responses in each constituent band.

Several speech processing schemes for improving speech perception in sensorineural hearing impaired are reported in literature. Lunner et. al. [1] implemented an eight channel signal processing system using TMS320C25 processor for a hearing aid. The filter bank in this scheme consisted of linear phase FIR filters with equal bandwidth of approximately 700 Hz and 40 dB stopband attenuation. For frequency response adjustments, the gain in each filter band is set individually to fit the hearing aid user. In listening tests with subjects having bilateral hearing loss, improvement in recognition score was obtained for dichotic over diotic presentation i.e. without splitting of the speech spectrum.

The hardware realization of the critical-band filter banks reported in the literature [6] uses lowpass, bandpass and highpass filter sections in cascade. However, the passband ripple of these filters is appreciable and band separation is not adequate due to less stopband
attenuation. In addition the filter transition is not sharp. As a result, the ability of the ear to binaurally receive and perceptually combine the dichotically presented speech signal is greatly curtailed. In the case of filters with finite crossovers in magnitude response, i.e. less steeper skirts in the transition region, there is an increased possibility of the spectral components belonging to crossover region being presented to both the ears. This causes an imbalance in loudness perceived at the crossovers between adjacent bands degrading the speech intelligibility.

With sharp transition filters, there is reduced possibility of spectral components presented to both ears at the transition region. Hence imbalance in loudness perceived at the crossovers between adjacent bands will reduce which increases speech intelligibility with sharp transition filters. Several methods have been proposed in the literature for the synthesis of FIR linear phase, sharp transition digital filters. Linear phase FIR digital filters have many advantages in speech processing applications such as guaranteed stability, negligible phase distortion and low coefficient sensitivity. A serious disadvantage of FIR filters is its complexity, which becomes acute in sharp transition filters. One of the most successful techniques for the synthesis of very narrow transition width, linear phase FIR filters is the frequency response masking (FRM) technique [7]. The FRM technique can be used for implementing sharp transition filters with arbitrary bandwidth with the resulting FRM filter has very sparse coefficients.

In the FRM techniques, closed form expressions for impulse response coefficients are not obtained. The subfilters in this technique are optimized to obtain final FRM filter. For a given frequency response specification its effective filter length including both zero and non-zero coefficient and delays required are longer than the infinite word length optimum minimax
filter. Resulting higher group delay is undesirable, particularly in speech processing applications. FRM is a graphical approach and transfer function for subfilters and the FRM filter are not evolved in frequency and time domain.

1.2 Research Objectives

The objective of the research is to synthesize narrow transition bandwidth, linear phase, FIR filters with multiple passbands of arbitrary width in the light of the literature cited above. Further, the synthesized filters were required to possess following desirable features:

- sharp transition with low passband ripple and large stopband attenuation.
- a well behaved function to model filter frequency response.
- closed form expressions for impulse response coefficients.
- impulse response coefficients related to transition width.
- finite transition width and well defined transition region unlike that of the conventional designs.
- a simple design procedure without optimizations involving less complex computations.
- a synthesis procedure without subfilters and their optimizations unlike FRM technique.
- design that is independent of center frequency and passband width for a bandpass/multiband filter.

1.3 Proposed FIR filter design

In pursuant to the stated objective, the method of filter synthesis proposed by us is as follows. The synthesis can be thought of as an extension of the classical methods reviewed in Appendix A1.1. In addition, a line of synthesis that also uses FRM technique is given. Several design procedures, each differing from the other in the formulation of pseudo-magnitude
response, for synthesizing low pass filters are presented. These are followed by the synthesis of band-pass filters with sharp transition, arbitrary passband width and center frequency. Each filter is specified by passband ripple $\delta_p$, stopband attenuation $\delta_s$, passband frequency $\omega_p$ (not the same as the cut-off frequency $\omega_c$ as defined in classical filter theory), the transition bandwidth $(\omega_s - \omega_c)$ where $\omega_s$ is the stopband frequency and $\omega_c$ is the cut-off frequency. In the case of bandpass filter, $\omega_b$ the center frequency is specified additionally. The specification of the transition bandwidth $(\omega_s - \omega_c)$ and its use in determining pseudo-magnitude in transition region is unique to the synthesis presented here. Hitherto, the transition region was treated as a "don't care region" and no attention was paid to the shape of its response in the methods of synthesis reported in literature.

The salient points of our synthesis are given below:

- The entire range $[0, \pi]$ of the frequency variable $\omega$ is split into three regions, namely pass-band, transition-band and stop-band. Each region of the pseudo-magnitude response is defined in terms of a cosine/sine function of $\omega$, confirming to the filter specifications.

- The pseudo-magnitude response function is assumed to be of the form given by (A 1.6a) and (A 1.6b) for N odd and N even respectively.

- Applying cosine transformation [11] to the pseudo-magnitude function $H_1(\omega)$ in (A 1.6a) and (A 1.6b), we obtain the impulse response sequence as

$$h(n) = \frac{1}{\pi} \left[ \int_{0}^{\pi} H_1(\omega) \cos k\omega \, d\omega \right]$$  \hspace{1cm} (1.1)
where \( n = 0, 1, \ldots, \frac{N-1}{2} \) for \( N \) odd,
\[
n = 0, 1, \ldots, \frac{N}{2} - 1 \quad \text{for } N \text{ even}
\]
and \( k = \frac{N-1}{2} - n \).

- The integral in (1.1) is similar to the inverse Fourier transform integral. The impulse-response sequence is extracted from (A1.6a) and (A1.6b) using the orthogonal property of the cosine/sine functions.

- Three parameters \( k_p, k_t, \) and \( k_s \) are used in the sine/cosine functions defining the passband, transition band and stopband respectively. These parameters control the shape of the pseudo-magnitude function \( H_1(\omega) \) in the three regions referred to above.

- In our synthesis, the desired response \( H_1(\omega) \) is not the ideal lowpass/bandpass response. The ideal (brick-wall) response is convenient to obtain the impulse response using inverse Fourier transform [8]. Instead, we formulate the desired response in terms of sine/cosine functions in pass, transition and stop bands which simplifies the evaluation of impulse-response sequence \( h(n) \) greatly. A closed form expression is obtained for \( h(n) \) in terms of filter design parameters \( k_p, k_t, k_s \) and filters specifications \( \omega_p, (\omega_s - \omega_c), \delta_p \) and \( \delta_s \).

- Ideal lowpass response with infinite slope at cut-off cannot be synthesized accurately in practice by finite physical functions. In our formulation of \( H_1(\omega) \) "well behaved" i.e. sine/cosine functions that are continuous and that possess continuous derivatives of all orders are used.

- The "well behaved" functions used in the formulation also reduce the ripples in \( H_1(\omega) \) that result from the "truncation" due to Gibb's phenomenon as in (A1.2). However
there is a small amount of “discontinuity” present at the boundaries of passband/transition band and transition/stopband regions. This residual discontinuity is removed by “slope-matching” i.e. equalizing slopes of pseudo-magnitude response on either side of the discontinuities. This will reduce the ripples due the Gibb’s phenomenon further.

- Narrow transition band is realized by having a large slope at cut-off or at $\omega = \omega_p$ the passband frequency, either by a large value of parameter $k_t$, or by having an equiripple passband and an equiripple stopband as in the case of the optimal filter [8]. Additionally, the stopband ripples are allowed to have negative values.

- A lowpass /bandpass filter synthesis is carried out using two values of parameter $k_p$, one giving a smaller frequency of ripples around the frequencies $\omega = 0$ and the other giving a larger ripple frequency around the passband edge $\omega_p$. Similarly two values of the parameter $k_s$ are used the one giving smaller ripple frequency is used closer to $\omega = \pi$ and the other giving larger ripple frequency is used near $\omega = \omega_s$. This technique combined with the “matching of slopes” is found to reduce the ripples due to Gibb’s phenomenon further.

- A linear transition region of constant slope is assumed in the formulation of the pseudo-magnitude response, which greatly simplifies the synthesis.

In summary, the filters proposed by us

- do not possess a flat passband or stopband which is not realizable in practice.
- do not have finite sized discontinuities except in cases without “slope matching”.
- do not have an infinite cut-off slope at band-edges.
• have finite width transition region.
• have considerably reduced band-edge ripple due to Gibb’s phenomenon.
• have closed form expression for impulse-response sequence.
• require a considerably simplified synthesis procedure that requires less algebraic computation involving trigonometric functions and also filter of less order (truncated impulse response sequence) to obtain a close degree of matching with the desired response.

An alternative interpretation of the truncation process that yields linear phase FIR filter in the window design is given in Appendix A1.2. With reference to the Appendix A1.2, we can say that the main planks on which our synthesis rests are the following:

• Deriving closed form expression for the impulse response sequence by formulating a pseudo-magnitude response that is practically realizable as in (1.1), unlike the ‘brick wall’ response.
• Using a large number of these impulse response coefficients obtaining an approximation that approaches the desired response by proposed ‘truncation’ explained in Appendix A1.2 and
• Design does not involve any optimization steps and the computations are simpler than reported in the literature.

In the window design, the cut-off amplitude is 6 dB, which is fixed independent of filter length N which is a serious disadvantage. This limitation is overcome in the proposed synthesis wherein, the cutoff amplitude is adjusted to within ± 0.1dB or 0.2dB. In FRM, the synthesis involves four subfilters. The model and mask filters should be individually synthesized and optimized which requires several cycles of computation. In frequency
sampling techniques, the desired response is the ideal response having the limitations referred to above. The technique also requires a large number of samples for accurate interpolation and does not yield a closed form expression for impulse response sequence. It requires computation of IDFT followed by DFT requiring large number of computations than that of our proposed design method. The optimal filter design requires computation of error function and its maximum in the range $[0,\pi]$. The optimization which minimizes this maximum error i.e. chebyshev criterion require several computations. The response obtained is not unique and depends on the initial response used and number of computations are large. A closed form expression for impulse-response sequence is not obtained. In optimal response, ideal lowpass response is assumed having the limitations enumerated above.

1.4 Organization of the thesis

The thesis is divided into six chapters. In chapter one, problem overview, research objectives and the proposed FIR filter design is given. Chapter two, gives a brief literature survey on sharp transition FIR filters, their features and problems encountered in implementation. Proposed slope equalization technique to reduce Gibb's phenomenon in sharp transition FIR filter is explained. Various lowpass, linear phase, sharp transition, digital filter models are formulated and their designs are proposed. Different lowpass filter models are formulated namely, nonmonotonic, monotonic, equiripple, equiripple with linear transition and finally filter model with equiripple response, linear transition with variable ripple density. Expressions for filter model parameters and impulse response coefficients are derived and coefficients of the filters obtained. Results of the various filter designs are tabulated and comparisons made between various models developed with conventional FIR filter design.
Chapter three, deals with the proposed linear phase, sharp transition FIR bandpass filter with slope equalization technique applied to the design. The bandpass design is a direct approach without need of component highpass and lowpass filters and is adaptable to any change in the center frequency and passband width as verified in this chapter. Another novel bandpass filter technique with variable density of ripple cycles in passband and stopband to further reduce Gibb's phenomenon is proposed.

In chapter four, literature survey for synthesis of very narrow transition width FIR filter using frequency response masking (FRM) technique is given. A modified frequency response masking technique for the synthesis of linear phase, sharp transition FIR filters with low arithmetic complexity is proposed.

In chapter five, functioning of human auditory system is explained and various types of hearing impairments are described. Various speech processing schemes employed in literature, to improve speech perception degraded due to loss of frequency selectivity caused by spectral masking for sensorineural hearing impaired are reviewed. A pair of multiband filters are designed, using the proposed approach to split the speech into various frequency bands, based on Zwicker's model, for binaural dichotic presentation of speech. It is experimentally proved that, this reduces the effect of spectral masking in sensorineural hearing impaired.

Chapter six gives the conclusions of the work done and future work proposed in this research area. The thesis has appendices, where conventional FIR filter design methods are reviewed, an alternative interpretation of impulse response truncation of FIR filter is given along with the implementation of speech processing scheme, listening tests and results of listening tests conducted on sensorineural impaired patients. The filter design steps for various proposed FIR filters developed are worked out and programs developed in MATLAB listed.