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FPGA IMPLEMENTATION OF COEFFICIENT DECIMATED POLYPHASE FILTER BANK STRUCTURE FOR MULTISTANDARD COMMUNICATION RECEIVER

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ABSTRACT

Coefficient decimated polyphase FIR filter bank structure implemented for receiving narrow band channels effectively in multistandard environment. Reonfigurability in multirate filtering is required to design a prototype filter bank structure for selecting the distinct polyphase sub filters and taps for different standards. Coefficient decimation (CD) based filter bank can offer a good trade-off between reconfigurability and low complexity which satisfy most of the requirements for SDR receivers. This paper proposed a method of filter bank technique which reduces the overall complexity of the design. The proposed filter structure has been synthesized on 0.18µm CMOS technology single core Field programmable gate array. Synthesis report proved the reduced device utilization for the proposed structure.

Keywords: *FIR Filters, Reconfigurabality, Coefficient Decimation, Polyphase Filter, Multistandard, Channelizati*

1. INTRODUCTION

Software defined radio (SDR) is defined as a radio in which the receive digitization is performed at stage downstream from the antenna, typically after wideband filtering , low noise amplification and down conversion to a lower frequency in subsequent stages. It consists of a programmable communication system where functional changes can be made by merely updating software. Similar to other digital Communication systems, the transmitter of a SDR system converts digital signals to analog waveforms. These waveforms are then transmitted to the receiver. The received waveforms are down converted, sampled, and demodulated using software on a reconfigurable baseband processor. Normally, high-performance digital signal processors and/or FPGAs are used to serve as the baseband processor. Digital signal processing in flexible and reconfigurable functional blocks defines the characteristics of the radio [3]. The same architecture can be programmed or reconfigured to cope with any standard. The major applications of SDR be in mobile communication- transceivers, generic cellular base stations and military radio systems. FIR filters has extensive applications in Communication SDR.

Receiver [2], due to its inherent stability and linear phase characteristics. Filters have a symmetric impulse response. Phase delay and group delay are constant. Several methods have been proposed to implement the reconfigurable FIR filters. The magnitude response after decimation has a bandwidth and transition bandwidth M times larger than that of the modal filter. The magnitude response after interpolation has a bandwidth and transition band width are reduced by the factor of 1/D. The techniques proposed in [4]-[6] implement reconfigurable filters with variable inputs along with the variable coefficients. The reconfigurable filter proposed in [8] does not employ frequency response masking filters and interpolation techniques.

A. Channelized Receiver

In a typical SDR receiver, the channelizer extracts multiple radio channels of distinct bandwidths from a digitized wideband input

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signal using digital filter banks. [13]. The channelizer employed in a radio supporting this type of architecture must be flexible enough to accommodate all of the carrier/bandwidth combinations supported by the network architecture, and possibly allow for the dynamic reallocation of channel resources within this architecture during operation In traditional, the uniform band-pass filter banks are usually employed to achieve channelized filter in analog domain. Different signals arriving at the same time that have different carry frequency will be output from different band pass filters with different center frequency. Every band pass filter is a channel in channelized receiver. The frequency accuracy is limited to the bandwidth of the band-pass filter.

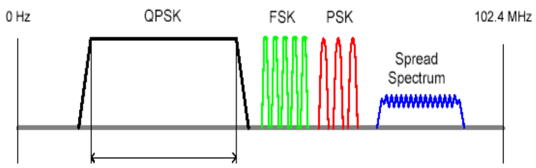

Fig. 1 Possible frequency plan

2. REVIEW OF FILTER BANKS

The per-channel (PC) approach in which each distinct channel is extracted using a dedicated filter. The PC approach can support uniform as well as nonuniform channelization. The complexity of the PC approach increases with the number of received channels DFTFB is a modulated FB consists of a single low pass filter followed by DFT operation [1]. The limitations of DFTFBs is that all the channels are of uniform BW i.e. DFTFBs cannot extract channels with distinct BWs simultaneously.

A. POLYPHASE FILTER BANK

The Polyphase filter bank consists of 2N independent FIR filters where N is the number of channels in the Polyphase-FFT system [7]. It is possible to expand H(z) in terms of M polyphase branches and it is possible to the polyphase implementation of prototype filter as mentioned in fig.2.1 makes reconfiguration tasks more tedious

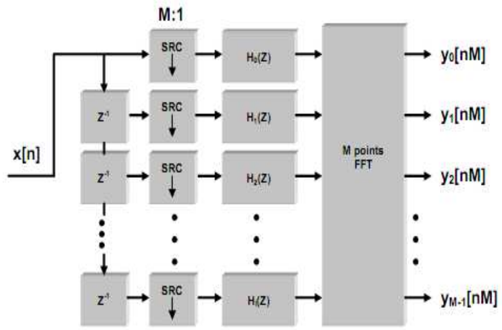

Fig. 2.1 Polyphase Filter Bank Structure

and expensive as it invokes updation of polyphase branches and the coefficients.

The polyphase filter is created through the decomposition of the low pass filter used to provide channel isolation on a per channel basis In general, the number of channels in this technique must equal the decimation rate[14], and as such the sampling rate must be a power of two times the baseband bandwidth.

B. Coefficient Decimation Method Filter Bank

The filter bank based on this approach have absolute control over the pass band width and pass band locations [1].

1. Coeffecient Decimation Method-I and II

In the CDM, N tap FIR filter in which the coefficients of a low pass FIR filter (termed as the modal filter) are decimated by M, i.e., every Mth coefficient is retained and the others replaced by zeros, to obtain a FIR filter with a multi-band frequency response. The frequency response of the resulting filter has bands with centre frequencies at $2\pi k/M$, where k is an integer ranging from 0 to $(M$-1)[1]. CDM[5] is used to obtain flexible filter banks for channelization PB widths and pass band centre frequency. the frequency response is obtained by scaling the coefficients by M = 2. the stopband attenuation reduces as M increases.the transition band width remains unaltered for any M. After performing CDM-I by decimation factor M, if all the retained coefficients are grouped together by dis-

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carding the zero coefficients in between, a low-pass frequency response with its pass band and transition band widths *M* times that of the modal filter is obtained[10]. This operation is called as CDM-II.

2. Modified Coefficient Decimation –I And II

If the modal filter is decimated by *M*, every Mth coefficient is retained and the sign of every alternate retained coefficient is reversed. All other coefficients are replaced by zeros [16]. As a result of this operation, an FIR filter with a multi-band frequency response is obtained with centre frequencies at $(2k+1)\ \pi/M$, where k is an integer ranging from 0 to $(M-1)$.

After performing MCDM-I by decimation factor *M*, if all the retained coefficients are grouped together by discarding the zero coefficients in between, a high pass filter response is obtained with its pass band and transition band widths *M* times that of the modal filter.

$$H_d(n)=H(n)\ d(n) \qquad (1)$$

This method is provided with enhanced frequency response flexibility and twice center frequency resolution compared to Coefficient decimation method.

3. Improved Coefficient Decimation Method

The combination of CDM-I and MCDM-I operations as improved coefficient decimation method I (ICDM-I), and the combination of CDM-II and MCDM-II operations as improved coefficient decimation method II (ICDM-II) respectively. Multiplexer for modal filter and complementary filter to produce different multiband frequency response. From the multiband frequency responses obtained after ICDM-I operations, individual frequency bands with identical BWs can be isolated by the use of frequency response masking filters and spectral subtraction [14].

C. Narrow band filter design

The complexity FIR filters with the filter order. FRM technique is used for the synthesis of sharp transition –band FIR filters with low complexity [9]. The advantage of FRM technique is that the bandwidths are not altered and the resulting filter will have many sparse coefficients resulting in less complex filters. A linear phase model filter whose each delay of this filter is replaced by M delays. The transition widths are a factor of M narrower than that of modal filter. This results in a periodic filter with sharper transition bands. Lower order sub filters are used to obtain the sharp filter responses. Configurations based on the use of half band masking filters achieve an improvement in the computational efficiency. The drawback of this method is that arbitrary pass band location of the resulting filters that cannot be controlled.

Interpolation by M consists of replacing each delay element of the FIR filters by M delay elements resulting in a filter with (M+I), multi-band responses having pass band width and transition bandwidth M times smaller the original filter[8]. Challenges of FRM technique can be reduced by reducing the number of multipliers.

3. RECONFIGURABILITY IN FIR FILTER

A digital FIR filter is a basic building block in any DSP system Reconfigurability is implemented in FIR filter is to realize various frequency responses using a single filter. FIR filter can be reconfigured by different coefficient word lengths, filter taps for low power architecture.. The frequency response of the filter depends on the value of its coefficient or taps LABVIEW programs [15] provide flexibility in changing the coefficients,order of the filter , window functions .Digital filters with adjustable bandwidths controlled by one or two parameters in many application Digital Filter Design Tool Kit [12] provides the multirate filter design ,analysis and implementation of Multirate filters.

The multirate filters include single-stage, half band, Nyquist, Raised cosine, and Cascaded integrator comb filters.

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A. Reconfigurability at Filter Coefficients

An N-tap FIR filter will have N multiplications whereas the filter whose coefficients are decimated by M will have only N/M multiplications. even if the initial modal filter in CDFB is designed with more taps (compared to the length of prototype filter in a DFTFB) taking into account of the stop band attenuation reduction after coefficient decimation, the effective filter length and overall multiplication complexity of the CDFB is less than that of DFTFB as coefficient decimation reduces filter length by a factor of M.

4. PROPOSED FILTER BANK TECHNIQUE

A new method for designing reconfigurable filter banks based on the multirate signal processing concept of decimation. In our design called coefficient decimation method controlled by multiplexer - with masking filters based filter bank design [15].

Architecture

FIR filtering operation performs the weighted summations of input sequences which is called as convolution sum which are used to design the frequency selective. The amount of computation and the corresponding power consumption of filter are directly proportional to the filter order.

First, design an N-tap low pass FIR filter as the modal filter. Which is decimated by D which implies that every D^{th} coefficient is retained and its remaining coefficients are replaced by zeroes. Interpolation by M which consists of replacing each delay element of the coefficient decimated FIR filters to get the multiband frequency response and design the complementary filters. The complexity will get reduced with proposed filter bank technique. Using multiplexer, the control signal can be applied to select the required frequency response. The desired channels of different bandwidths can be extracted from the identical bandwidth spectrum replicas of the decimated modal filter using one or more of following operations- Spectral subtraction, frequency masking or complementary filtering. the order of the resulting filters after coefficient decimation decreases as the decimation factor value is increased, due to elimination of filter coefficients that are not retained. As the group delays of filters with different filter orders are unequal, the corresponding group delays of the concerned filters are to be compensated before performing spectral subtraction. This is done by adding k delay elements where k is the difference between the group delays of the filters whose frequency responses are subtracted. The structure consists of sharp transtion band FIR filters are required in these systems to meet the stringent wireless communication specifications. In conventional FIR filter designs, higher order filters are required to obtain the sharp transition band. The complexity of FIR filters increaes with the filter order[15]. Adding the 2:1 multiplexers (mux) that are used to select the filter coefficients.

If S=1 for a multiplexer corresponding to a particular filter coefficient, that coefficient is retained and when S=0, that coefficient is bypassed. The adder/subtractor (add/sub) blocks are used to perform the sign reversal of the alternate retained coefficients.

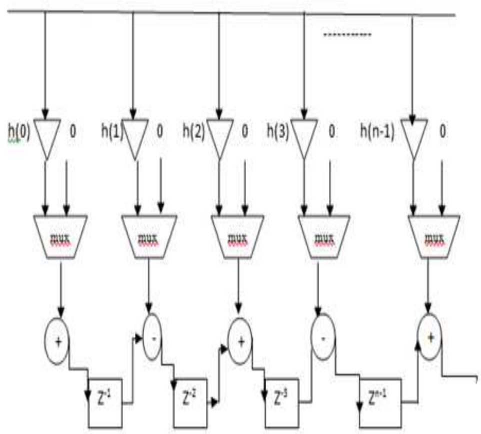

Fig.4.1 Coeffecient Decimated FIR Filter

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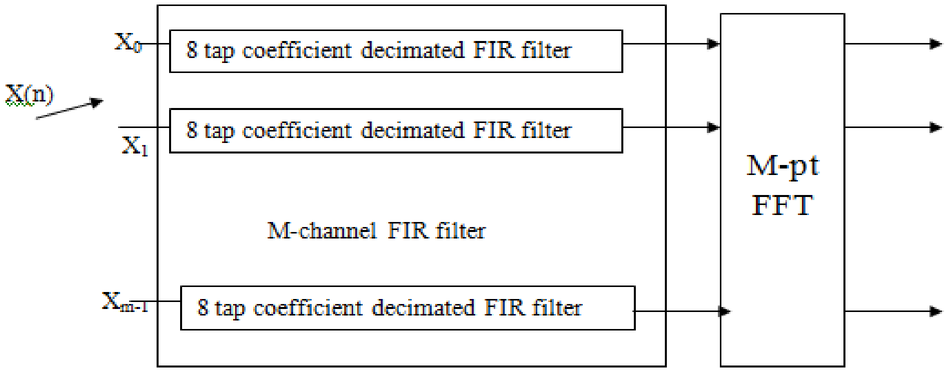

. Fig.4.2 Coeffecient Decimated Polyphase Filter Bank

A. Design Considerations and Mathematical Analysis

The reconfigurability of the receiver is accomplished by switching among different Filter banks, each designed for a particular standard. Same hardware platform is reusable for different system parameters configurations without requiring any hardware changes to Software defined radio receiver. To realize a filter bank, which can be reconfigured to accommodate multiple standards[15] with reduced hardware overhead , it is proposed that Frequency response masking based reconfigurable polyphase Filter bank architecture . A polyphase filter that uses different decimated resampling factors for maximally decimated [14] , under decimated, over decimated, and combined up and down sampled is used for reducing the area ,time power optimization efficiency. This technique consists of synthesis filters and analysis filters. Each sub band filter operates on 1/N the of the input samples Input rate has its specifications controlled by its output rate due to Nyquist sampling criterion after spectral folding due to down sample operation. The length of the finite impulse response (FIR) prototype filter s required to satisfy the filter specifications. The filter using the same hardware requires high level of flexibility in DSP[16]. Polyphase filter consists of 2N independent FIR filters , where N is the number of channels in the polyphase FFT. T tap FIR filter computes T multiplication operation x(n-i) input samples , where a_i is the filter coefficients.

$$y(n) = \sum_{i=0}^{L} a_i\, x(n-i) \qquad [1]$$

$$a_i = [a_{i,0}, a_{i,1}, \ldots, a_{i,L-i}]$$

$$H'(e^{j\omega}) = \frac{1}{M}\sum_{k=0}^{M-1} H\left(e^{j\left(\omega - \frac{2\pi k}{m}\right)}\right) \qquad [2]$$

$$H'(e^{j\omega}) = 1/M \sum_{k=0}^{M-1} H\left(e^{(j\left(\omega - \frac{\pi(2k+1)}{M}\right)}\right) \qquad [3]$$

Coeffecients of the modal filter h(n)

$$h'(n) = h(n)d_M(n) \qquad [4]$$

$d_M(n)$= 1 , n=mM , m=0,2,4,6 [5]

-1 , n=pM ,p=1,3,5,7

0 , otherwise

The filter coefficients are entirely user defined and can be loaded or updated at anytime before or during processing. The FFT core uses a decomposition of radix-4 and radix-2 butterflies for computing the DFT, ranging from 8 to 4096 points. The FFT length is a user programmable parameter and it can be changed without the need to reconfigure the FPGA. Select the appropriate values of M to get multiband frequency response containing the desired passband locations. Modified filter coefficients h(n) Frequency response

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are integral multiples of 2π/M. Designing an N-tap modal filter by replacing certain coeffecients by zeroes based on the control signal applied to

Table. 4.1 Comparison Of Filter Bank Technique

Filter bank Technique	**Transition band width**	**Stop Band attenuation**	**Bin Spacing (512 bins) KHZ**	**Reconfigu-rability**	**Non uniform Channelisation**
DFT FB	300 KHz	-45.25 dB	200	Very Poor	Uniform
Coeffecient Decimation	300 KHz	-53.72 dB	200	Good	Uniform
Frequency masking Technique	200 KHz	-54.65 dB	200	Good	Non uniform
Proposed Model using LabVIEW	200 KHZ	-60.27 dB	200	Good	Non uniform

the multiplexer. The control signal decimate the filter by the replacement of certain coefficients which leads to the modified frequency response consists on the extraction of the channels by subtracting the different versions of the decimated base filter for different decimation factors .The decimated filter can be interpolated to get the multiband frequency response by H (Z $^{M/D}$) , where D is the no. of delay elements This structure can extract multiple non uniform frequency bands. Reconfigurability can be achieved by including the specified delay elementsby reprogramming the coefficients of the filter leads to required frequency reponse characteristics. The higher order filters may be required. This architectural complexity for higher order structure structure can be replaced by the use of Frequency masking filters.The FRM technique adopted in this proposed architecture can achieve flexibility in filter filter bank technique for non uniform channelization. The control signals set to 1 for addition and 0 for subtraction. The alternate add/sub blocks lead to develop the logic with low complexity which will reduce the hardware complexity .

B. Performance Metrics

This proposed model can be used to develop the filter bank to extract multiple frequency bands simultaneously. The multiple frequency bands can be extracted using This proposed model can be used to develop the filter bank to extract multiple frequency bands simultaneously . Multiple frequency bands can be extracted using suitable masking filters and the different values of D. The masking filters can be designed with wider transition band widths as the frequency bands are located with the specifications. Table 1. Represents the comparison of various filter banks performance characteristics for the 512 frequency bins, Bin spacing is obtained as 200 KHZ.

C. Simulation Results and Analysis

Lab VIEW 2012 version can be used to build a channelizer based system. Basically, the use of LABVIEW allowed this interactive channelizer system to be built in a shorter time as compared to text based programming languages.

Multirate filtering—Multirate filters are digital filters that convert the sampling frequency of an input signal to a new sampling frequency. Multirate filters increase or decrease the sampling frequency of the input signal while minimizing pass band distortion, aliasing, and imaging in the signal[13]. Therefore, the sampling frequency of the output signal from a multirate filter is different from that of the input

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signal. Multirate filters can reduce computational complexity and data volume in one system, or multirate filters can change the frequency as necessary to be compatible with other systems. In multirate signal processing, the primary consideration is the selection and modification of the proper sampling frequency. FRM can be used to achieve sharp transition bands with fewer filter coefficients. However, in a FRM structure the sample rate remains constant unlike multi-stage filtering structures formed from interpolators and decimators Reconfigurability of the proposed model can extract the channels for multi standards.

The design specifications for the first band-select filter must have to cover the complete spectrum of the two standards, and are given by:

• Sampling freq (fs = 840MHz)

• Transition Width (_F = TW1 + TW2 = 10MHz)

TW1 : transition width of the right half of the band.

TW2 : transition width of the left half of the band,

• Passband edge frequencies (36 - 410)MHz

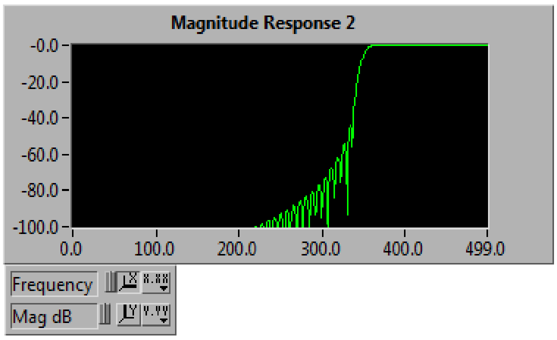
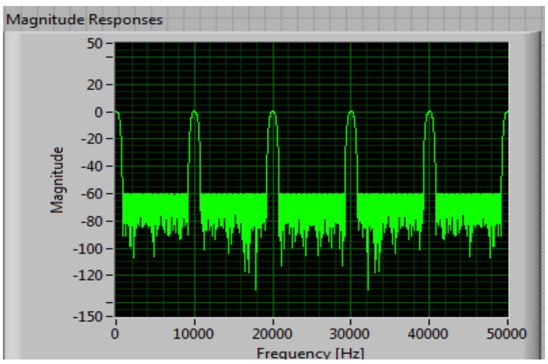


Fig 4.3.1 Modal filter



4.3.2 Interpolated FIR filter

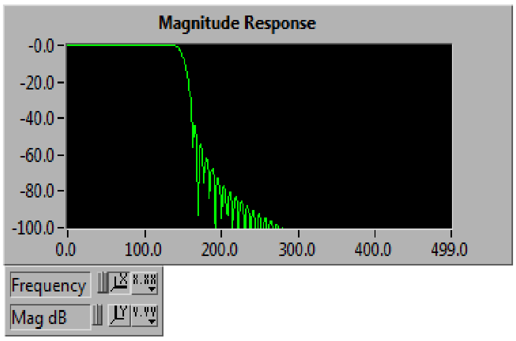


Fig.4.3.3 Frequency specifications

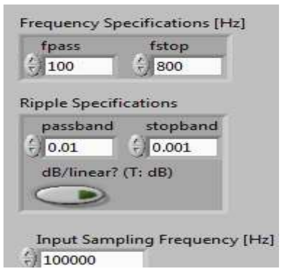


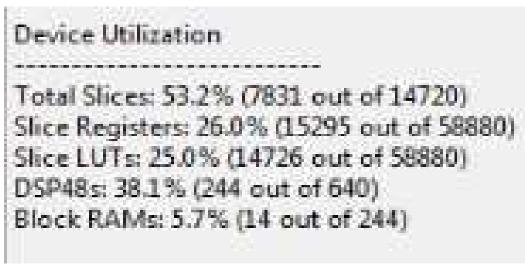
Fig.4.3.4 Complementary filter

The desired pass band and stop band attenuation can be 0.1 dB and -62.25 dB respectively. The different decimation alternatives Fig.4.3.1 Frequency specifications will form optimization constraints with standards. This filter structure consists of coefficient decimation and extracting the desired band for channel adaption. the non-uniform channelizer designs based on the FRM [9]technique showed a very high number of operations per second in Comparison with other method

Resource utilization

Single board Reconfigurable IO (SbRIO) utilizes 53.2% slices and Blok RAMs 0f 5.7 %

(a)


Device Utilization

Total Slices: 53.2% (7831 out of 14720)
Slice Registers: 26.0% (15295 out of 58880)
Slice LUTs: 25.0% (14726 out of 58880)
DSP48s: 38.1% (244 out of 640)
Block RAMs: 5.7% (14 out of 244)

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(b)

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Timing
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40 MHz Onboard Clock: 40.00 MHz (100.00 MHz maximum)
200 MHz Clock: 200.00 MHz (317.66 MHz maximum)
128MHz: 128.01 MHz (129.35 MHz maximum)
TS_AdcDataClk: 428.45 MHz (maximum)
TS_BusClk: 126.92 MHz (maximum)
TS_SlowBusClk: 147.36 MHz (maximum)
TS_DramClkDiv100: 500.25 MHz (maximum)
TS_IoRxClock: 333.33 MHz (maximum)
TS_Puma20Window_theCLIPs_IO_Module_CLIP0_NI5761Topx_NI5761AdcSamplerx_NI5761TimingEnginex_SampleClkPb: 311.33 MHz
(maximum)
TS_Puma20Window_theCLIPs_IO_Module_CLIP0_NI5761Topx_NI5761AdcSamplerx_NI5761TimingEnginex_DivSampleClkPb: 179.21 MHz
(maximum)
TS_ClockGenXilinxV5x_TxDcm_TxHighSpeedClkDcm: 298.69 MHz (maximum)
TS_ClockGenXilinxV5x_RxDcm_RxHighSpeedClkDcm: 500.25 MHz (maximum)
TS_ClockGenXilinxV5x_RxDcm_RxLowSpeedClkDcm: 189.61 MHz (maximum)
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5. CONCLUSION

A modal filter and its complementary filter is proposed , then each delay of these filters is replaced by M delays which results in much sharper transition bands .Different filter banks for multi standard software defined radio receivers were discussed. The coefficient decimated polyphase filter bank architecture is implemented in single board reconfigurable input output (SbRIO) The proposed filterbank technique offers reconfigurability in extracting the narrow band signals with low complexity compared to the conventional filter banks .. The additional optimization methods may be adopted to reduce the complexity to a minimumby the use of frequency response masking filters . Synthesis report is proving the effective resource utilization and satisfy most of the requirements for SDR receivers.

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