Dynamic Channelization Model Based on Non-maximally Decimated Filter Bank

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***Abstract*—A model is proposed, called dynamic channelization model based on non-maximally decimated filter bank, to solve the uncertainty and time-varying nature of subband signals' number, bandwidth and location distribution. The structure divided the receiving signal into several sub bands evenly based on signal reconstruction theory. We input the adjacent subbands signals which belongs to the same channel to the corresponding integrated filter banks, and the output is the base band signal for this channel. Furthermore, without changing the analysis filter bank, we just need to update the energy detection link to obtain the position information of the new signal can get the corresponding integrated filter banks to complete signal receiving and processing when the signal changes. Namely, it has a certain adaptive characteristics.**

***Index Terms*—non-maximally decimated, filter banks, dynamic channelization, reconstruction**

I. Introduction

With the complexity of modern electromagnetic environment and the LPI radar signal and the frequency agility radar appear, the bandwidth of the radar signal becoming more and more wide, which requires the digital receiver has a wide input bandwidth, high sensitivity, high resolution and large dynamic range and the ability to handle multiple signals at the same time arrived [1]. As one of the key technology of the software radio system and wideband digital receiver, and channelized technology is used to extract the single or multiple independent subband signals contained in receiving bandwidth so that the baseband processing will be more convenient [2]. The implementation of the specific functions include: digital down conversion, filtering, sampling rate conversion, channel identification and signal extraction and so on. Because the channelized processing usually spend great computational complexity and hardware resources, so how to realize the efficient digital channelized receiver has become a hot and difficult research topic. The digital channelized receiver used in the field of reconnaissance is usually divided the digital channel evenly, that is, the division of the channel is blind, so the cross channel phenomenon occurs when the signal bandwidth is greater than the bandwidth of the sub band [3]. The dynamic channelized method can reconstruct the adjacent channel when these channels belong to the same signal by channel detection, and can reconstruct the original signal, which has the characteristic of self adaptation.

In order to achieve high performance requirements and easier hardware implementation, at present, the digital down conversion technology based on polyphase filtering structure is used to realize the efficient channelized receiver [4]. At present, there is not much about the technology of channelization in the domestic and abroad. Hentschel T of the University of Dresden in Germany analysis and compared the performance of digital channelized receiver based on a signal channel, a DFT filter bank and a Goertzel filter bank [5]; P Vinod A *et al.* research on the digital channel receiver based on the efficient digital nonuniform filter banks [6]; in China, the research on the technology of the channelization has already achieved a good result, Zhang Quanpu, Liu Yanqiong use the method based on the polyphase filter to deal with the channelization, which is the most widely used method of the current uniform channelized processing [7]; then, based on the polyphase filter banks, Zhang Chunhui proposed a new efficient channelized structure by using modulated filter bank technique [8]. In this paper, the analysis and reconstruction of the wideband signal are realized by analysis and synthesis filter banks, the filter banks were designed by DFT reconstruction algorithm and meet the approximate reconstruction condition.

II. Polyphase Filter Channelized Structure Based on Non-maximally Decimated

The efficient structure of modulated filter banks based on polyphase algorithm can filtering the signal on the real-time and parallel, signal detection, making the structure more simple and efficient, with a strong engineering practicality. In order to be able to handle cross channel signal, it is required that an integrated filter bank can splice signals across several channels [9]. So this article can be from two aspects, the first aspect is the analysis part of this dynamic channelization architecture based on non-maximally decimated polyphase filter banks, the second aspect is the integrated part of this architecture, as shown in the Fig. 1 and Fig. 2.

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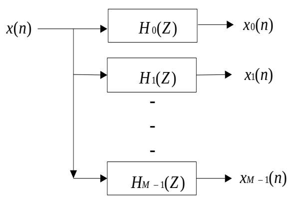

Figure 1. Analysis part

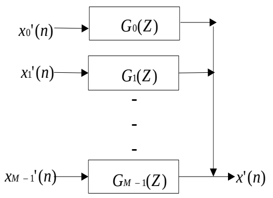

Figure 2. Analysis part

A. Channel Partition of Real Signal

In order to derive the efficient channelized receiver structure of real signal, we should division frequency band with real signal first, as shown in the Fig. 3.

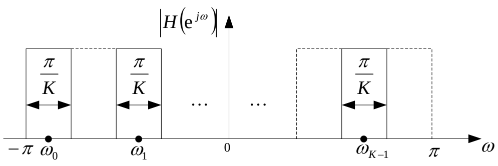

Figure 3. Channel partition of real signal

$$w_k = (k - \frac{2K-1}{4}) \cdot \frac{2\pi}{K}, \quad k = 0,1\cdots,K-1 \tag{1}$$

In the traditional channelized structure, when the general signal input channelization system, first the signal is multiplied by complex factor ($e^{j\omega_k n}$) to division the number of sub bands, the signal is converted to the zero frequency position [10], and then through the low-pass filter to filter out the signal noise outside the frequency band. And then the output of each sub band is decimated to reduce the sampling rate so that the low frequency signals can be processed further in the hardware system with lower processing speed. The tradition channelized structure is shown in the Fig. 4.

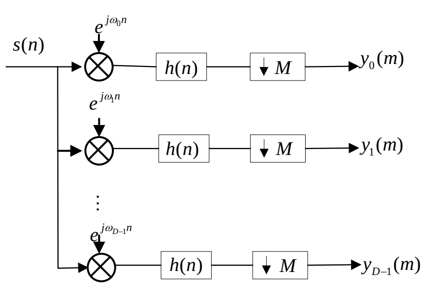

Figure 4. Channelized low-pass implementation

Set the input discrete signal is $s(n)$, the unit impulse response of the N-order FIR low pass filter is $h(n)$, the relationship between decimated ratio D and sub band number K is represented as K=DF, and the order of each sub band filter is L=N/K [11], the input signal is filtered by the filter banks and the first k sub band output is $y_k(m)$.

$$\begin{aligned} y_k(m) &= \{[s(n)\cdot e^{jw_k n}] * h(n)\}|_{n=mD} \\ &= \sum_{i=0}^{N-1} s(n-i)e^{jw_k(n-i)}\cdot h(i)|_{n=mD} \\ &= \sum_{i=0}^{N-1} s(mD-i)e^{jw_k(mD-i)}\cdot h(i) \end{aligned} \tag{2}$$

The polyphase structure expression of the signal after D times decimate and the expression of the filter polyphase component are written as:

$$s_p(m) = s(mD-p), \quad hp(m) = h(mK+p) \tag{3}$$

$$\begin{aligned} y_k(m) &= \sum_{p=0}^{K-1}\sum_{i=0}^{L-1} s(mD-iK-p)e^{jw_k(mD-iK-p)}\cdot h(iK+p) \\ &= \sum_{p=0}^{K-1}\sum_{i=0}^{L-1} s[(m-iF)D-p]e^{jw_k[(m-iF)D]}\cdot h_p(i)e^{-jw_k p} \end{aligned} \tag{4}$$

Putting $s = iF$ into (4).

$$y_k(m) = \sum_{p=0}^{K-1}\sum_{s=0}^{(s-1)F} s[(m-s)D-p]e^{jw_k[(m-s)D]}\cdot h_p(\frac{s}{F})e^{-jwkp} \tag{5}$$

Define $h_{p'}(s) = h_p(\frac{s}{F})$

$$\begin{aligned} y_k(m) &= \sum_{p=0}^{K-1}\sum_{s=0}^{(s-1)F} s_p(m-s)e^{jw_k[(m-s)D]}\cdot h'_p(s)e^{-jw_k p} \\ &= \sum_{p=0}^{K-1}\{[s_p(m)\cdot e^{jw_k mD}] * h'_p(m)\}e^{-jw_k p} \end{aligned} \tag{6}$$

B. Channelized Structure of Real Signal Based on Non-maximally Decimated Polyphase Filter

According to the band division of real signal in the 2.1 section, we put w_k (the center frequency of the K channel) into (6), then we can get the polyphase filter channelized model of real signal.

Defin e $s'_p(m) = [s_p(m)\cdot e^{jw_k mD}] * h'_p(m)$

$$y_k(m) = \sum_{p=0}^{K-1} s'_p(m)e^{-jw_k p} \tag{7}$$

Put $w_k = (k - \frac{2K-1}{4})\cdot\frac{2\pi}{K}$ into (7).

$$\begin{aligned} y_k(m) &= \sum_{p=0}^{K-1} s'_p(m)e^{-j(k-\frac{2K-1}{4})\cdot\frac{2\pi}{K}p} \\ &= \sum_{p=0}^{K-1} s'_p(m)e^{-j\frac{2k\pi}{K}p}\cdot e^{j\frac{(2K-1)\pi}{2K}p} \\ &= DFT[s'_p(m)\cdot e^{j\frac{(2K-1)\pi}{2K}p}] \end{aligned} \tag{8}$$

$$\begin{aligned} s'_p(m) &= [s_p(m)\cdot e^{jw_k mD}] * h'_p(m) \\ &= [s_p(m)\cdot e^{j(k-\frac{2K-1}{4})\frac{2\pi}{K}mD}] * h_p(m) \\ &= [s_p(m)\cdot e^{j\frac{4k+1}{2F}\pi m}] * h_p(\frac{m}{F}) \end{aligned} \tag{9}$$

The theoretical model of a generalized DFT-based non-maximally polyphase filter channelized structure is shown in Fig. 5.

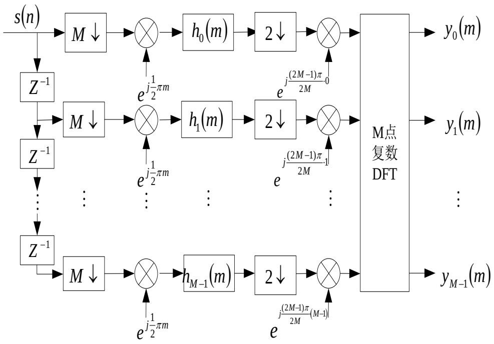

Figure 5. Non-maximally polyphase filter channelized structure

C. Structure Analysis

The generalized DFT-based non-maximally polyphase filter channelized structure is shown in Fig. 5. We can see its realized method is through a low-pass prototype filter multiplied by a multiple factor for modulation and then convert it to filter's polyphase structure. This structure split the whole frequency band into several parallel outputs, so that the signals in anyone channel can be received accurately. In addition, the use of polyphase fliter reduces the amount of calculation to the original 1/D, and improves the ability of real-time processing of channelized receiver. However, this structure is still defective in the cross channel signal receiving.

III. Dynamic Channelized Architecture

In the previous chapter, the analysis part of the dynamic channel structure based on non-maximally decimate is introduced and designed in detail. But, actually we always need to synthesize some specified signals to achieve the original after the signal through the analysis part [12]. So, in order to complete the reconstruction of the signal, the integrated part of the dynamic channel structure is designed, the overall implementation of the structure as shown below.

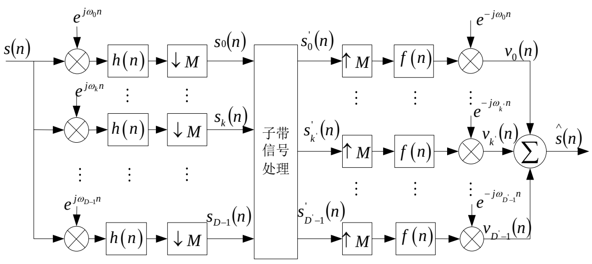

Figure 6. System design block diagram

A. Signal Approximation Precise Reconstruction Condition

As the actual received signal is always real signal, and the frequency band division of the real signal for signal reconstruction implementation structure is simple [13], the signal approximation precise reconstruction condition of this structure can be achieved based on the real signal non-maximally decimate efficient structure on chapter 2.

Select K=2D to facilitate derivation, as shown in Fig. 6, the spectrum of the decimated signal will be the original 1/D, define the frequency response of the intermediate processing between the analysis filter and the integrated filter is $K(Z)$. So the target signal after integrating can be expressed as:

$$Y(Z)=\frac{1}{D}\sum_{d=0}^{D-1}X\left(ZW_D^d\right)\times\sum_{m=0}^{M-1}K\left(Z^DW_M^{mD}\right)H\left(ZW_D^dW_M^m\right)G\left(ZW_M^m\right) \quad (10)$$

The Equation (10) can be expressed in a matrix form, and we can do the following definition:

$$G(Z)=\left[G\left(ZW_M^0\right)...G\left(ZW_M^{M-1}\right)\right]^T \quad (11)$$

$$H(Z)=\left[H\left(ZW_M^0W_D^0\right)...H\left(ZW_M^{M-1}W_D^0\right)\right]^T \quad (12)$$

$$\bar{X}(Z)=\left[X\left(ZW_D^1\right)...X\left(ZW_D^{D-1}\right)\right]^T \quad (13)$$

$$X(Z)=\left[X\left(ZW_D^0\right),\bar{X}(Z)\right]^T \quad (14)$$

$$\mathbf{H}(Z)=\begin{bmatrix} H\left(ZW_M^0W_D^0\right) & ... & H\left(ZW_M^0W_D^{D-1}\right) \\ ... & ... & ... \\ H\left(ZW_M^{M-1}W_D^0\right) & ... & H\left(ZW_M^{M-1}M_D^{D-1}\right)\end{bmatrix}_{M\times D} \quad (15)$$

$$\mathbf{K}=diag\left(K\left(Z^DW_M^{0D}\right),...,K\left(ZW_M^{(M-1)D}\right)\right) =\begin{bmatrix} K\left(Z^DW_M^{0D}\right) & ... & 0 \\ ... & ... & ... \\ 0 & ... & K\left(ZW_M^{(M-1)D}\right)\end{bmatrix} \quad (16)$$

Equation (10) is expressed as:

$$Y(Z)=\frac{1}{D}G_{1\times M}^T(Z)\mathbf{K}_{M\times M}(Z)\mathbf{H}_{M\times D}(Z)X_{D\times 1}(Z) =\frac{1}{D}\mathbf{T}_{1\times D}^{\mathbf{K}}(Z)\mathbf{X}_{D\times 1}(Z) \quad (17)$$

$$\mathbf{T}_{1\times D}^{\mathbf{K}}(Z)=G_{1\times M}^T(Z)\mathbf{K}_{M\times M}(Z)\mathbf{H}_{M\times D}(Z)=\left[\mathbf{T}_s^{\mathbf{K}}(Z)\mathbf{T}_A^{\mathbf{K}}(Z)\right] \quad (18)$$

M channels, D times decimate and the entire conversion function expression is shown in (18).

$$\mathbf{T}_s^{\mathbf{K}}(Z)=G_{1\times M}^T(Z)\mathbf{K}_{M\times M}(Z)\mathbf{H}_{M\times 1}(Z) =\begin{bmatrix} G\left(ZW_M^0\right) \\ G\left(ZW_M^1\right) \\ ... \\ G\left(ZW_M^{M-1}\right)\end{bmatrix}\begin{bmatrix} K\left(Z^DW_M^{0D}\right) & ... & 0 \\ ... & ... & ... \\ 0 & ... & K\left(ZW_M^{(M-1)D}\right)\end{bmatrix}\begin{bmatrix} H\left(ZW_M^0\right) \\ H\left(ZW_M^1\right) \\ ... \\ H\left(ZW_M^{M-1}\right)\end{bmatrix} \quad (19)$$

Equation (18) is the conversion function for the useful signal.

$$\mathbf{T}_A^{\mathbf{K}}(Z)=G_{1\times M}^T(Z)\mathbf{K}_{M\times M}(Z)\mathbf{H}_{M\times(D-1)}(Z) =\left[G\left(ZW_M^0\right)...G\left(ZW_M^{M-1}\right)\right]\begin{bmatrix} K\left(Z^DW_M^{0D}\right) & ... & 0 \\ ... & ... & ... \\ 0 & ... & K\left(ZW_M^{(M-1)D}\right)\end{bmatrix}\begin{bmatrix} H\left(ZW_M^0W_D^0\right) & ... & H\left(ZW_M^0W_{D-1}^{D-2}\right) \\ ... & ... & ... \\ H\left(ZW_M^{M-1}W_D^0\right) & ... & H\left(ZW_M^{M-1}M_{D-1}^{D-2}\right)\end{bmatrix} \quad (20)$$

Equation (20) is the conversion function for the useless aliasing signal. Equation (18) can be expressed as below based on (19) and (20).

$$Y(Z)=\frac{1}{D}\mathbf{T}_s^{\mathbf{K}}(Z)X(Z)+\frac{1}{D}\mathbf{T}_A^{\mathbf{K}}(Z)\bar{X}(Z) \quad (21)$$

When (20) is equal to zero, that is:

$$\sum_{m=0}^{M-1}K\left(Z^D W_M^{mD}\right)H\left(ZW_D^d W_M^m\right)G\left(ZW_M^m\right)=0, d=1,...,D-1 \quad (22)$$

$$\sum_{m=0}^{M-1}H\left(ZW_D^d W_M^m\right)G\left(ZW_M^m\right)=0, d=1,...,D-1 \quad (23)$$

By the modulation of the related theory, the (23) can be simplified as:

$$H\left(ZW_D^d\right)G(Z)=0, d=0,...,D-1 \quad (24)$$

In order to realize the effective transmission of the signal, (19) must be integer times delay between the analysis filter bank and the integrated filter bank, and the requirement of frequency response between the analysis filter bank and the integrated filter bank is shown below:

$$\mathbf{K}_{M\times M}=\mathbf{E}_{M\times M} \quad (25)$$

$\mathbf{E}_{M\times M}$ is M-order unit matrix in the (25).

In summary, the transmission function of the useful signal can be simplified as (26).

$$\sum_{m=0}^{M-1}H\left(ZW_M^m\right)G\left(ZW_M^m\right)=Z^{-nD} \quad (26)$$

In order to achieve almost complete reconstruction of the signal, the analysis filter banks $h(n)$ and the integrated filter banks $g(n)$ need to meet the conditions of (24), (25).

In summary, the key to the design of the reconfigurable channelized system is the design of FIR prototype low pass filter $f(n)$ and $h'(n)$. Among them, (24) requires the filter has a linear phase, all-pass characteristic and minor amplitude distortion. The (25) requires less aliasing distortion of the reconstructed signal.

B. Design of the Dynamic Channel Structure Reconstruction Filter Bank

The design process of the whole reconstruction filter bank system can be summed up as follows by the analysis of the 3.1 section:

1) Determining the number M of sub bands and the decimated times D of the analysis part [14]. And the integrated sub bands number D' and interpolation multiples M' of the integrated part according to the input signal character detecting.

2) The design of the prototype filter unit impulse response corresponding to the analysis filter banks and the integrated filter banks is determined by the precise reconstruction condition.

3) The prototype low-pass FIR filter is decomposed into polyphase structure by using the polyphase decomposition algorithm.

The dynamic channelized structure based on non-maximally decimated filter banks is shown in the following Fig. 7.

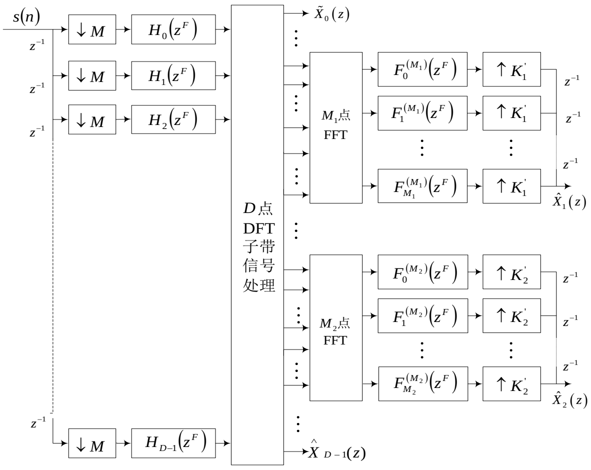

Figure 7. Dynamic channelized structure

C. Filter Design and Iterative Optimization

The design of the prototype filter bank is based on the design method of the iterative algorithm, and the design process is as follows:

1) The initial value of the iterative algorithm is the FIR filter based on square root ascending cosine function, and the window function is added to the constraint [15].

$$h_0(n)=r\cos ine(1,M,'sqrt',a,L)\cdot w(n) \quad (27)$$

$w(n)$ is a window function, and the prototype filter order $N=L\times 2M+1$.

2) Iterative optimization $h_0(n)$

$$h_t(n)=h_0(n)*h_0(n) \quad (28)$$

$$b_\lambda=(-1)^\lambda h_t(\lambda M) \quad (29)$$

$$h_{corr,\lambda}(n)=b_\lambda\frac{\sin\pi n/M}{\pi n/M}\cdot\frac{1}{1-(\frac{n}{\lambda M})^2} \quad (30)$$

$$h_{corr}(n)=\sum_{\lambda-1}^{L}h_{corr,\lambda}(n) \quad (31)$$

$$h_0(n)=h_0(n)+h_{corr}(n)\cdot w(n) \quad (32)$$

The advantages of the prototype filter method are that the filter order is low, the stop band attenuation is large, the amplitude and the aliasing error are small, compared to the traditional optimization design method is simple.

IV. Simulation Results

Simulation of the polyphase filter channelized structure based on non-maximally decimated in Section 2.2 is shown below.

Simulation conditions: Taking F=1, the number of the subband divided by the analysis filter banks is 8, the system sampling frequency is 960MHz, and the bandwidth of each sub band is 60MHz, the passband of the prototype low-pass FIR filter is set to 30MHz,the stopband is set to 40MHz, and the stopband attenuation is set to 120dB. Input signals: The first input signal is the linear frequency modulation signal, the starting frequency, termination frequency and pulse width are 20MHz, 80MHz, and 5us. The second input signal is a sinusoidal signal with 260MHz. The third input signal is the linear frequency modulation signal, the starting frequency, termination frequency and pulse width are 340MHz, 30MHz, and 5us.

The order of the prototype low-pass FIR filter is 423, and the spectrum analysis of the three input signals and the distribution characteristics of the filter banks are shown in Fig. 8.

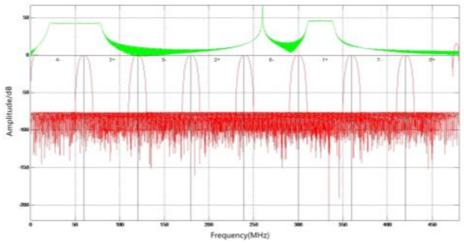

Figure 8. Spectrum distribution characteristics of input signal and filter banks

The each subband output signal time frequency feature is shown in Fig. 9.

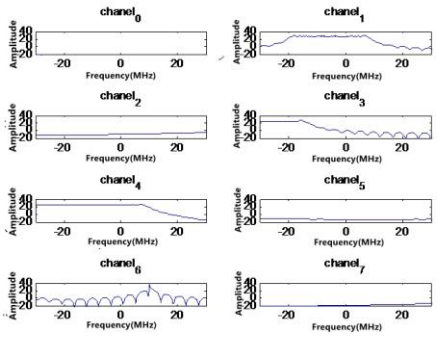

Figure 9. Each subband output signal time frequency feature

According to the spectrum characteristics of the input signal and each subband filter groups, after the analysis part filtering the main information of the first input signal is distributed in the sub band 3 and 4;the main information of the second input signal is distributed in the sub band 6; the main information of the third input signal is distributed in the sub band 1, the starting and ending frequency is -20MHz and 10MHz, the bandwidth is 30MHz, it is the result of the input signal mixing with the center frequency 330MHz then after a filter. The simulation results shown in Fig. 10 and 11 are consistent with the theoretical analysis, it can be verified that the design of this channelized structure based on non-maximum decimated is correct. However, the cross channel signal has not been solved perfectly.

Simulation of the reconstruction part in Chapter 3 is shown below.

Simulation conditions: 15 MHZ to 190 MHZ bandwidth for linear frequency modulation signal. The simulation time is 5e-6s.

The square root raised cosine function A is used to design the FIR filter, and the Blackman window is used to increase the attenuation of the stop band. The passband cutoff frequency of the Blackman window is 30MHz, the stop band cutoff frequency is 32MHz, the filter is 1889 order after the iterative optimization processing, and the number of sub bands M=16, and the length of each polyphase branch filter is about 59 order. The performance requirements of the filter has been significantly improved, the filter optimization iteration number is 60 times.

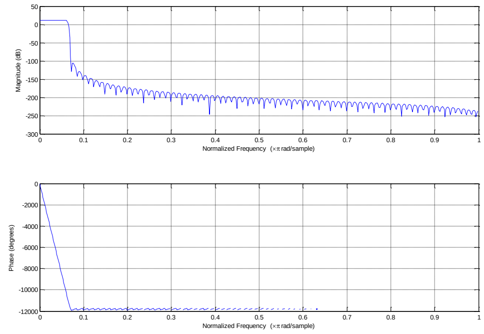

Figure 10. Feature of the reconstruction optimized prototype filter

Frequency spectrum of the input signal and reconstruction signal:

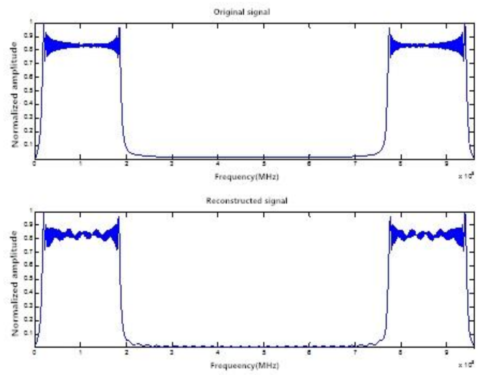

Figure 11. Feature of original signal and reconstructed signal

Increasing the order of the filter, and the amplitude frequency response of the filter is similar to rectangle, then the reconstruction system has been greatly improved. The original signal covers 0 to 6 bands, a total of 7 sub bands, and the interpolation factor is 8. It can be seen from the simulation results that the reconstructed signal start frequency is 15MHz and the stop frequency is 190MHz, remodeling the original signal accurately, and the signal reconstruction frequency error is not more than 2MHz. The reconstructed signal is interpolated by 8 times, and then the output signal frequency is 480MHz. The seventh sub band channel is zero-padding by the input part of the integrated filter banks, and the sampling rate of the output signal is improved. At the same time, the FFT can be used to reconstruct the signal efficiently.

V. CONCLUSION

Channelized technology is the key technology of the channelized technology and the wideband digital receiver, decimated a single or multiple independent subband signals contained in the receiver bandwidth for easy back-end baseband processing. How to efficiently implement the channelization is a hot and difficult research topic in the field of communication.

This paper discusses the dynamic channelized structure based on the non-maximally decimated filter banks, when the signal bandwidth is greater than the subband channel width he broadband signal is divided into several signals by the analysis filter banks, through different sub band channel output, detecting the channels with signal input by energy detecting link, finally input to the synthesis filter banks for signal reconstruction. Due to the different decimated and interpolation time, the bandwidth of the reconstructed part of the channel is reduced, and the sampling frequency is reduced, the later signal processing is more convenient. And the correctness of this method is verified by MATLAB.

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