Investigation of Radio Signal Reconnaissance Based on Intermediate Frequency Channel Receiver

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Abstract: The traditional radio station can only achieve a single function, the software radio broke through the limitations of this design, emphasizing the construction of open universal hardware platform, as far as possible with scalable, reconfigurable application software to achieve a variety of wireless communications.. Based on this idea, the channelized digital receiver can monitor the high-bandwidth signal to achieve full probability intercept, with multi-channel signal parallel processing and dynamic range, the advantages of low hardware complexity. According to different communication requirements and standards, the bandwidth and frequency band position distribution of each subband signal in the digital receiver monitoring bandwidth can also be different. For this situation, different channelization techniques can be used to receive the IF signal. Filter banks and other related technologies continue to develop new channels for the structure to provide more possibilities. In this work the techniques of sampling, multi-rate signal processing and digital down-conversion are studied in detail. Based on this, the channel structure based on multi-phase DFT filter bank is studied, and the channel distribution is discussed in detail. The mathematical model of the real signal channel is deduced and established, and the structure and operation complexity of each model are analyzed and compared. The correctness of the model is verified by MATLAB simulation of the selected real signal

1. Introduction

The functionality of traditional radio architectures is often determined primarily by hardware, which is almost impossible to configure by software. The hardware includes amplifiers, filters, mixers and oscillators. The software is limited to control the interface with the network, remove the packet header and error correction code, and determine the routing address of the packet based on the header information. Because the hardware dominates the entire design, upgrading the traditional radio system design essentially means completely abandoning the original design and restarting the design. Traditional radio communication architecture, more and more difficult to adapt to the ever-changing communication development situation, so software radio came into being, and attracting more and more attention. When upgrading a software radio, most of the new content is software, and the rest is an improvement in hardware component design. In general, software radio represents a transition from fixed hardware-intensive radios to multi-band, multi-mode software-intensive radios. The software radio mainly consists of three parts: one is the RF front-end processing of the antenna receiving signal, the other is the high-speed analog-to-digital conversion (A/D), the third is the digital part of the signal processing (DSP)[1,2], as shown in figure 1:

Fig.1 The Main Components of the Software Radio Receiver

Channelization technology is one of the key technologies of software radio system and broadband digital receiver for extracting single or multiple independent subband signals.Specific features include: digital down conversion, filtering, sampling rate conversion, despreading and so

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on. As the channelization process usually consumes a lot of computation and hardware resources, how to realize the channelization process has always been the hot and difficult research in the field of communication. At present, narrowband digital down conversion has a very mature technology to achieve, and for multi-channel broadband high-speed signal reception processing are generally using analog filter group channelization system, or multiple digital receivers in parallel to achieve channelization function. However, these two methods of implementation due to complex equipment, have a high cost, poor scalability and other defects. Based on the idea of software radio channel, the digital receiver of software radio is used to process the multi-channel parallel processing on a single receiver, and the receiver processing capacity and real-time processing capability are improved effectively while improving the complexity of the receiver system,in particular to enhance the receiver's full-band probability acquisition capability. Therefore, the study of digital receiver channelization technology is of great significance.

For example, when the N-way signal occupies equal bandwidth within the reception bandwidth and is arranged at equal intervals, the channel can be channeled by using the DFT filter bank based on the multi-phase decomposition. The advantage of this channel separation method is that when the number of extracted subband signals is large, the method can greatly reduce the computational complexity of the whole process, and the hardware complexity of the implementation structure is relatively low. If the N-channel signals occupy equal bandwidth, but the position distribution of each sub-band signal in the receiving bandwidth is arbitrary, you can use the Goertzel algorithm based on the filter group channelization technology.And the front two cases are not satisfied, that is, IF bandwidth contains a number of non-uniform distribution of different bandwidth sub-band signal, in the known sub-band signal state.The proposed method is based on the method of modulation filter bank. The basic idea of this method is to realize nonuniform channelized reception with uniform filter bank. The realization process includes decomposition and synthesis of two steps. The method has the advantages of small computational complexity and moderate hardware complexity compared with the parallel single channel DDC method, which makes full use of the polyphase filter structure of the modulation filter group and the equivalent exchangeability of the circuit[3,4].

But in multi standard communication system, satellite link such as multi standard, if bandwidth not only contains the subband signals, the state information of these subband signals, but also dynamic changes may occur in the receiving process, which involves the dynamic channelization problem; what is more, in some special the occasion, probably could not know the subband signals, which relates to the blind channelization problem. Obviously, the existing channelization techniques can not be applied to the above situation. Therefore, the main purpose of this work is: try to find new methods and structure of the channel, in order to efficiently implement channelization, refers to the so-called efficient hardware: low computational complexity, flexible structure and moderate complexity[5].

2. The Key Technology of the Receiver

2.1 Broadband If Bandpass Sampling Structure

Based on the broadband IF bandpass sampling digital receiver structure shown in Figure 2. This structure is similar to the conventional superheterodyne radio station. The essential difference is that the intermediate frequency bandwidth of the conventional radio station is narrowband structure, and the intermediate frequency bandwidth of figure. 2 is broadband structure.

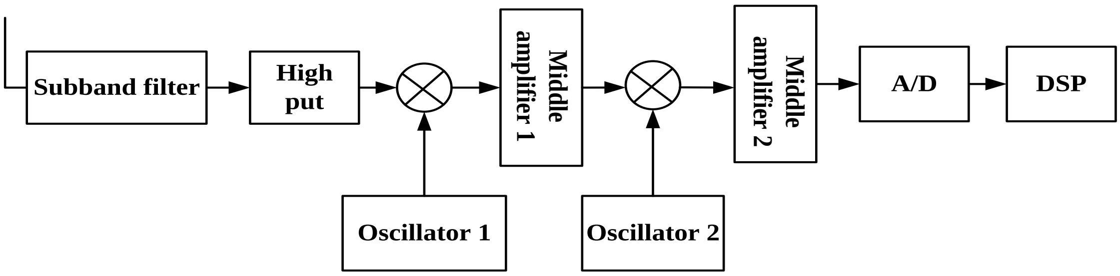

Fig.2 Digital Receiver Architecture Based on Broadband If Bandpass Sampling

The radio frequency signal received by the antenna obtains the fixed frequency intermediate frequency signal by superheterodyne technology. Because of the sampling and processing of the software radio system is a multi channel broadband signal, while the actual frequency selective filter Q value is small, the condition of superheterodyne receiver to effectively always mirror frequency interference, must use the high frequency superheterodyne structure. But IFl signal frequency is too high will make the actual level of the implementation of the system is limited by the subsequent A/D conversion device, IF2 signal receiver generally use two time mixing technology having lower center frequency, thereby reducing the difficulty of implementation of the system. The intermediate frequency signal IF2 is controlled by automatic gain control (AGC) and is converted into digital signals by A/D to process real-time digital signals. The band coverage of a digital receiver structure based on wideband IF bandpass sampling is achieved by the step change of the first order frequency of the receiver, and is easier to achieve than the RF bandpass sampling. The channelized receiver scheme of this work is based on the broadband IF bandpass sampling structure[6].

2.2 Multi-Rate Signal Processing

Multi rate signal processing is the goal of changing signal without signal distortion under sampling rate. The process of reducing the sample rate to reduce data redundancy is called signal extraction (Decimation). The process of increasing the sampling rate of data to increase data redundancy is called signal interpolation (Interpolation). Integer interpolation refers to the insertion of sampling points between the known two adjacent sampling points to integer times to improve the sampling rate, many methods such as interpolation, linear interpolation, two interpolation, Lagrange interpolation, the simple method is in between the adjacent sampling points interpolation I-1 zero, then the signal is low-pass filtering[7,8].

The signal before interpolation is $x(n_1)$ and the interpolation multiple is I, then the interpolated signal $x'(n_2)$ is:

$$x'(n_2)=\begin{cases} x(\frac{n_2}{I}) & n_2=0,\pm I,\pm 2I,... \\ 0 & others \end{cases} \tag{1}$$

Its spectrum is:

$$
\begin{aligned}
X'(e^{jw_2}) &= \sum_{n_2=-\infty}^{\infty} x'(n_2)e^{-jw_2n_2} \\
&= \sum_{n_1=-\infty}^{\infty} x'''(n_1)e^{-jw_1n_1} \\
&= \sum_{n_1=-\infty}^{\infty} \left[x(n_1)\frac{1}{D}\sum_{l=0}^{D-1} e^{j2\pi n_1 l/D} \right] e^{-jw_1n_1} \\
&= \frac{1}{D}\sum_{l=0}^{D-1}\sum_{n_1=-\infty}^{\infty} x(n_1)e^{-j(w_1-2\pi l/D)n_1} \\
&= \frac{1}{D}\sum_{l=0}^{D-1} X\left[e^{j(w_1-2\pi l/D)}\right]
\end{aligned} \tag{2}
$$

As shown by the above formula, the interpolated spectrum is the same as that before interpolation. Because the interpolation sampling rate increased I times, so the number of interpolation cycle digital spectrum can be improved by I times, the normalized frequency for interpolation in front of the spectrum in the frequency axis I times after compression results, the ideal cutoff frequency of low pass filter should satisfy the $w_c \le \pi / I$.

In order to prevent the aliasing of the signal after extraction, usually in the extraction before the signal should be anti-aliasing filter. Anti-aliasing filters often use FIR filters, set the order of N, and divided into D group (N is an integer multiple of D), then the length of each group is $L = N / D$. This decomposition is called polyphase decomposition, and its corresponding transfer function can be expressed as polyphase decomposition as equation (3).

$$
\begin{aligned}
H(z) &= \sum_{n=0}^{N-1} h(n)z^{-n} \\
&= h(0)z^0 + h(D)z^{-D} + ... + h\left[(L-1)D\right]z^{-(L-1)D} \\
&+ h(1)z^1 + h(D+1)z^{-(D+1)} + ... + h\left[(L-1)D+1\right]z^{-[(L-1)D+1]} \\
&+ ... + h(D-1)z^{-(D-1)} + h\left[2D-1\right]z^{-(2D-1)} + ... \\
&+ h\left[(L-1)D+(D-1)\right]z^{-[(L-1)D+(D-1)]} \\
&= \sum_{n=0}^{L-1} h(nD+0)(z^D)^{-n} + z^{-1}\sum_{n=0}^{L-1} h(nD+1)(z^D)^{-n} + ... \\
&+ z^{-(D-1)}\sum_{n=0}^{L-1} h(nD+D-1)(z^D)^{-n} \\
&= \sum_{k=0}^{D-1} z^{-k}\sum_{n=0}^{L-1} h(nD+k)(z^D)^{-n}
\end{aligned} \tag{3}
$$

$$
e_k(n) = h(nD+k) \tag{4}
$$

If the transfer function is $E_k(z)$, then:

$$
H(z) = \sum_{k=0}^{D-1} z^{-k} E_k(z^D) \tag{5}
$$

Prior to each phase filtering, the input data $x_1(n)$ enters each phase filter at intervals of D. From the law of each phase filter data input, the $x_1(n)$ is a one wheel to enter each phase filter, every D cycles. Therefore, for each phase filter, although its order is still L, the amount of input data is reduced by D times, and the required amount of computation is also reduced by D times, and the number of complex multiply addition is N / D. For the D filter, because the phase filter is working, so the amount of computation required to extract the equivalent single-phase branch of the complex multiplication times of N/D, the required computing speed of Nf_s/D per second complex multiplication. To sum up, the extraction system using polyphase filtering structure can effectively reduce the computational complexity of the system and improve the real-time performance of the system signal processing[9,10].

3. Modeling and Simulation of Receiver

Because the time domain signals are usually real sequences in practical applications, the operational complexity of the channelized structure of real signals is analyzed and discussed in this section, so as to provide reference for the hardware design of channelized receivers. The channel structure is based on the odd arrangement of channel division, and the output of the first K channel is[11]:

$$y_k(m) = \sum_{p=0}^{2K-1} x_p(m).e^{-j\pi(k+\frac{1}{2})\frac{p}{K}} \tag{6}$$

According to the odd type real number DFT algorithm, the N singular point DFT of the real signal $x(n)$ is defined as:

$$X_{odd}(k) = \sum_{n=0}^{N-1} x(n).e^{-j\pi(2n+1)\frac{k}{N}} \tag{7}$$

MATLAB simulation: Let $x(n)$ input signal into 6 real signal superposition, the total bandwidth of 6MHz, each sub signal bandwidth of 1MHz, f_s sampling frequency is 16MHz, the sample number is 4000. set channel number D=6, the normalized channel bandwidth is π/D, then the prototype low-pass FIR stopband starting frequency $\pi/2D$, in order to prevent 2D decimation after aliasing.

The prototype filter can give high stopband attenuation and small passband ripple at lower orders, while the squareness is also good, as shown in figure 3:

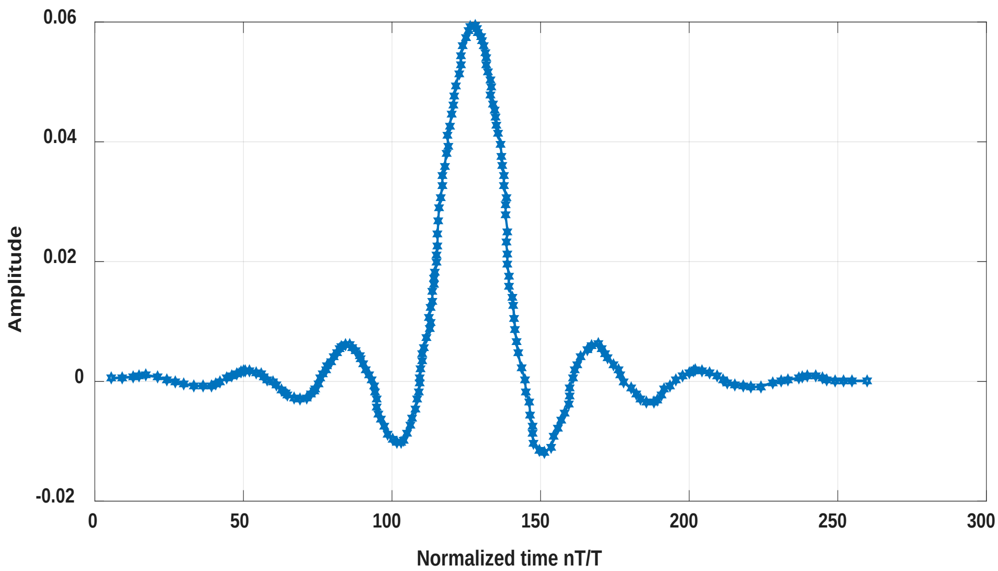

Fig.3 Prototype Low Pass Filter Characteristic Map

Set the input signal for the sinusoidal single-frequency signal, the frequency is 50.01KHz, after channelization simulation, the same results into MATLAB to draw the spectrum shown in Figure 4-6 (Here only gives the first, second and third subchannels of the spectrum output):

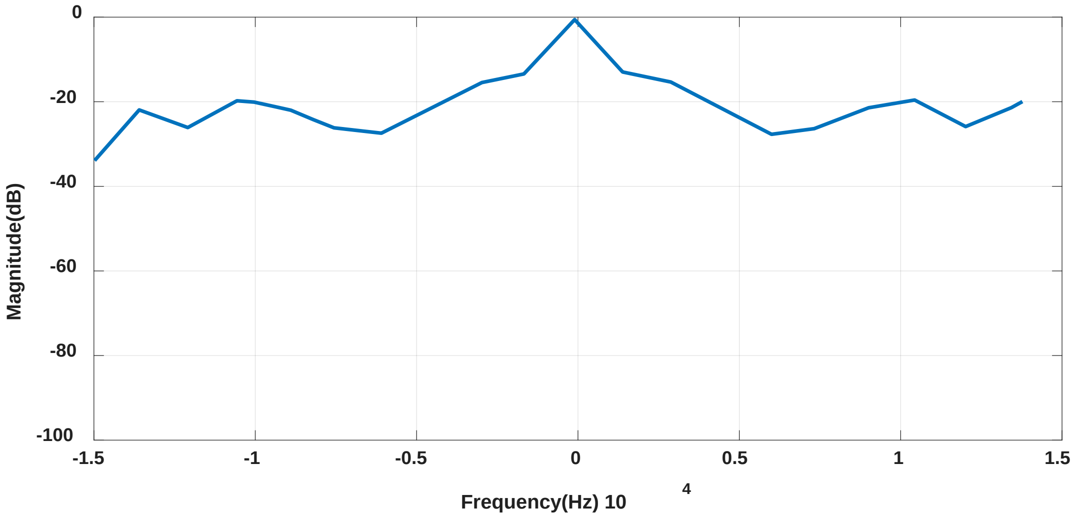

Fig.4 Domain Output of 2nd Channel Frequency

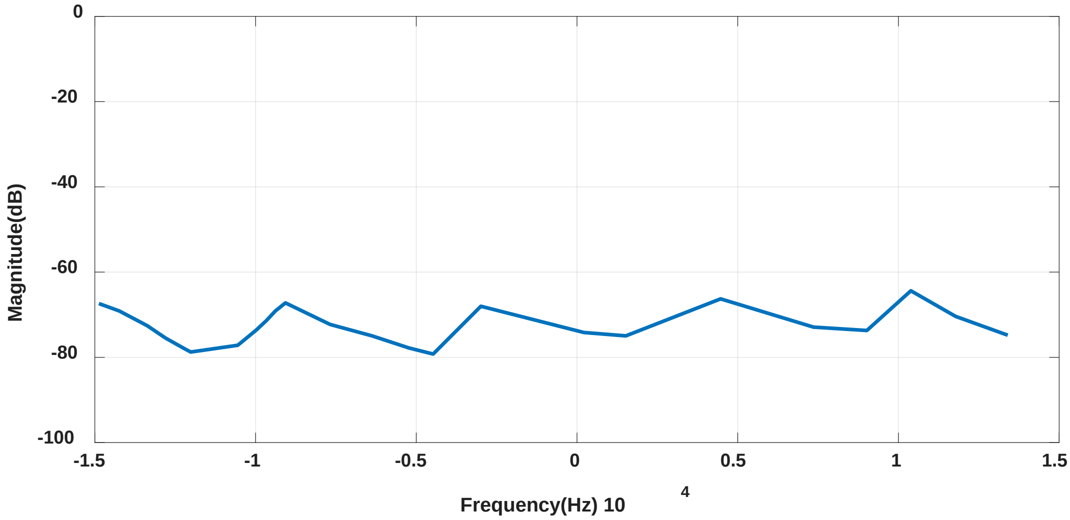

Fig.5 Domain Output of First Channel Frequency

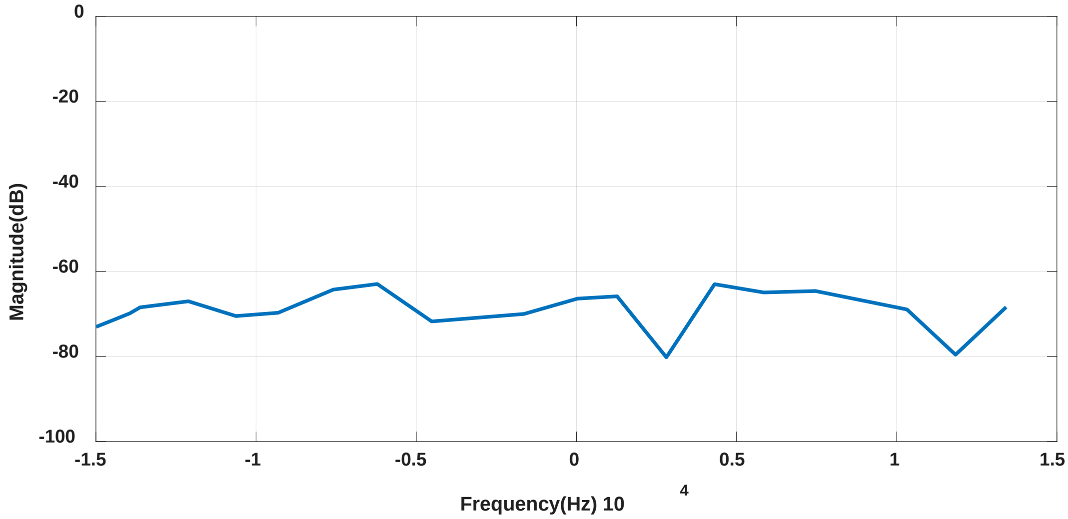

Fig.6 Domain Output of 3rd Channel Frequency

As shown in Figure 6-8, the signal appears correctly on the second sustain channel, and the adjacent subchannels have no signal output. We have done the other frequency of the relevant simulation test, the results are correct, not detailed here. The results of the digital down-conversion and multiphase channelization are basically consistent with the theoretical values of MATLAB. However, the suppression ratio of the digital down-conversion and the suppression of the adjacent channel are better than those without MATLAB. The main reason is finite word length effect.

4. Conclusion

The emergence of software radio technology has greatly expanded the research field of digital signal processing, and has opened up a potential development prospect for almost all radio electronic information systems. The digital receiver based on software radio has incomparable advantages both in real-time receiving and processing of data, and in the efficient analysis of stored data. With the development of high-speed large-scale integrated circuit, the flexibility of digital receiver will be more improved. Therefore, it is meaningful to study the related technology of digital receiver.

The modern receiver can realize the full probability intercept signal of the working frequency band, and has the real-time parallel processing ability to the multi-channel signal. Channelized digital receivers are generated specifically for these needs. In this work, we focus on the analysis of channelized receiver based on multi-phase DFT filter bank for some specific permutations of multi-channel signal neutron channel, and focus on two channel-oriented receiver schemes for specific requirements. The IFM channelized receiver combines the channelized structure with the instantaneous frequency measurement technique to complement the length of the blind in the absence of dead zone. The channel-based receiver based on the reconstruction algorithm reconstructs the channel signal after channelization, so that the original signal feature can be recovered in order to carry on the characteristic analysis and the parameter estimation to the signal in the following processing. Finally, the correctness of the two schemes is verified by MATLAB simulation.

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