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Asynchronous sampling rate conversion of digital audio signal

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Abstract. At present, in the digital audio processing sampling rate is respectively 32 kHz, 44.1 kHz, 48 kHz [1]. Because of the different criteria, there is much inconvenience in the process of research. Therefore, the sampling rate converter is a must, between any two kinds of sampling rate. In synchronous sampling rate conversion, you can use decimation and interpolation for sampling rate conversion, but in the asynchronous sampling rate system, due to the different input clock pulse with the output clock pulse, the above method cannot achieve. Therefore we introduce the fractional delay filter sampling rate conversion. This article introduces the principle of the sampling rate conversion and the fractional delay filter based on Farrow structure. At last, we simulate asynchronous sampling rate conversion of audio signal through the MATLAB.

Introduction

In the digital signal transmission system, the use of sampling rate conversion is very common. There are many methods of sampling rate conversion. The most common is using integral decimation and interpolation to realize fractional sampling rate conversion. [2] But in the actual implementation process, interpolation and extraction ratio cannot be too big, or the implementation of filter will be very difficult. For example, converting 44.1 kHz signal to 48 kHz signal by the above method, needing for 160/137 times of sampling rate conversion, which means that the sampling rate of the input signal is interpolated by a factor of 160, and decimated by a factor of 147, the difficulty of its realization is obvious. [3]. The above method is only applicable to simple synchronous sampling rate conversion (sender and receiver have the same clock pulse control), and when the sender and the receiver in different clock pulse control, namely for asynchronous sampling rate conversion, simple decimation and interpolation will not be able to use, so we will introduce fractional delay filter to realize asynchronous sampling rate conversion.

Asynchronous Sample Rate Converter (ASRC)

First of all, we introduce the concept of quasi continuous system, which consists of a continuous impulse response with sampling by f_o frequency. As shown in figure 1, a pure digital system using for asynchronous sample rate converter.

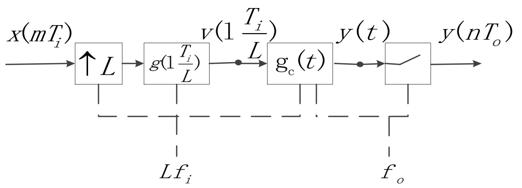


Fig. 1 The concrete structure of the pure digital Asynchronous sample rate converter

According to the system as shown in the diagram, the output signal of L – interpolation:

$$v\left(l\frac{T_i}{L}\right) = \sum_{\forall m} g\left(l\frac{T_i}{L} - mT_i\right) x(mT_i) \; = g\left(l\frac{T_i}{L}\right) \otimes x(mT_i) \tag{1}$$

Then make the convolution of $\mathrm{v}\left(\mathrm{l}\frac{\mathrm{T_i}}{\mathrm{L}}\right)$ and the impulse response of quasi continuous interpolation $\mathrm{g_c(t)}$:

$$y(t) = \sum_{\forall m} g_c\left(t - l\frac{T_i}{L}\right) v\left(l\frac{T_i}{L}\right) = g_c(t) \otimes v\left(l\frac{T_i}{L}\right) \tag{2}$$

According to the (1) (2):

$$y(t) = \sum_{\forall l}\sum_{\forall m} g_c\left(t - l\frac{T_i}{L}\right) g\left(l\frac{T_i}{L} - mT_i\right) x(mT_i) \;= g_c(t)\otimes g\left(l\frac{T_i}{L}\right)\otimes x(mT_i) \quad (3)$$

Sampling $y(t)$ by the frequency of $f_o = 1/T_o$:

$$y(nT_o) = \sum_{\forall l}\sum_{\forall m} g_c\left(nT_o - l\frac{T_i}{L}\right) g\left(l\frac{T_i}{L} - mT_i\right) x(mT_i)$$

$$= g_c(nT_o)\otimes g\left(l\frac{T_i}{L}\right)\otimes x(mT_i) \quad (4)$$

It seems that the key step in the asynchronous sample rate converter is the design of continuous interpolation. We usually use Lagrange interpolation based on Farrow structure.

Filter Based On Farrow Structure

Assumes that the sampling rate conversion for digital signal sequence $x(nT_x)$, the receiver get the signal $y(mT_y)$ which sampling period is T_y. According to the sampling rate conversion formula:

$$y(mT_y) = \sum_{n=-\infty}^{\infty} x(nT_x)h(mT_y - nT_x) \quad (5)$$

Assumes: $mT_y = (k_1 + \mu)T_x, 0 \le \mu < 1, k = (k_1 - n)$ is a positive number. We simplify (5) by the above equations:

$$y(m) = \sum_{k=-\infty}^{\infty} x(k_1 - k)h(k + \mu) \quad (6)$$

In the (6) , $\mu = \frac{mT_y}{T_x} - \left[\frac{mT_y}{T_x}\right]; k = \frac{mT_y}{T_x} - n,$

Then approximate $h(k + \mu)$ by the Taylor Series :

$$h(k + \mu) = \sum_{l=0}^{P} b_i(k)\mu^i \quad (7)$$

Finally, we get the result:

$$y(m) = \sum_{l=0}^{P}\left[\sum_{k=-\infty}^{\infty} x(k_1 - k)b_l(k)\right]\mu^l \quad (8)$$

Equation (8) is the expressions of Farrow structure filter. Figure 2 is the diagram of Farrow structure filter, composed of P groups of N-order sub-filters. [4][5][6]

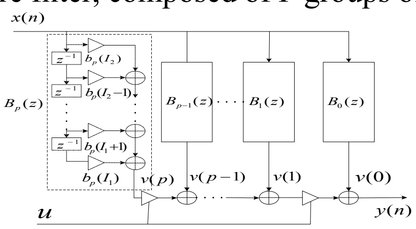


Fig. 2 Farrow structure filter

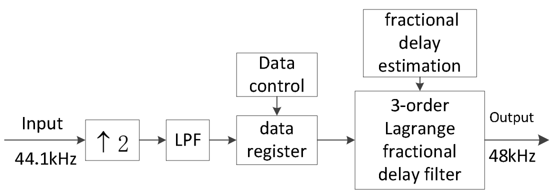


Fig. 3 Block diagram of CD-DAT sampling rate conversion system

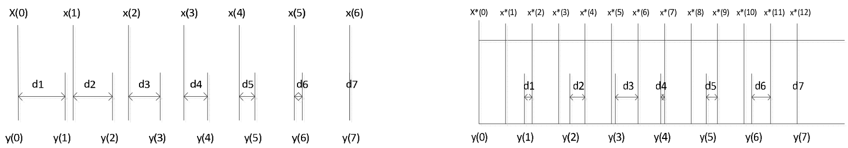
Implementation

The simulation phase, we will adopt Lagrange interpolation filter based on Farrow structure to realize asynchronous sampling rate conversion of CD (44.1 kHz) to DAT (48 kHz), the filter structure is shown in figure 4, The structure diagram of the whole process is shown in figure 3.[7]

A. Simulation of tone signal and performance evaluation

Early experiments, in order to facilitate the performance comparisons, we get the 6 kHz test signal $x(n)$ by sampling analog tone signal (frequency of 10 kHz). Then we make a sampling rate conversion of the test signal by the factor of 7/6. In the meantime, we also sample the original analog single at the sampling frequency of 7 kHz to get digital tone signal $x_2(n)$. Finally, we use respectively the third order, five order, seven order Lagrange interpolation filter based on FARROW structure to realize sample rate conversion of test signal $x(n)$ by the factor of 7/6. Then we can compare the performance of each order filter according to the quality of each output signal.

As shown in figure 4. First, 6 kHz audio signal convert into 12 kHz audio signal by a two times up-sampling and a low-pass filter. Because the value of each output sample can only be calculated through a n-order Lagrange fractional delay filter, so $(n+1)$ consecutive input sample points should be deposited into data buffer, one output sample is obtained after the fractional delay filter. In the process of sampling rate conversion, the value of output sample can be obtained by fractional delay filter, time position of output sample is determined by fractional delay d and position of reference points. Figure5 shows the timing relationship of the input and output when the output sampling rate is greater than the input sampling rate by a factor of 7/6. d_n is the fractional delay of $\mathrm{y(n)}$ relative to $\mathrm{x(n)}$. But the value of the $\mathrm{d_n}$ will be larger, therefore, making optimization, we first make a two times interpolation of input signal, through the contrast we can found that the latter d_n's value is lower than the former's. When using the MATLAB function call, we need to use forward delay, so make $d_n = 1 - d_n$, the results are shown in figure 5b. This case reference point is the forward nearest point of the $\mathrm{y(n)}$.



(a) (b)

Fig. 4 Positional relation diagram of input and output

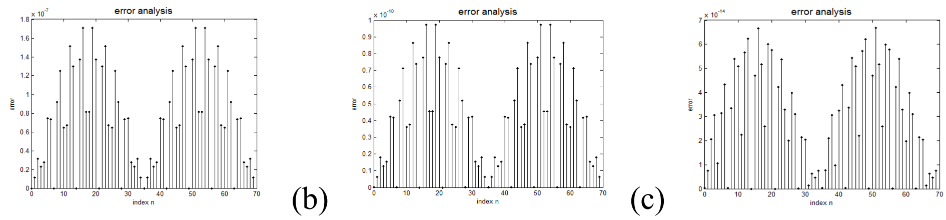
By the above method, 6 kHz tone signal can be converted into 7 kHz output $y_1(n)$, $y_2(n)$, $y_3(n)$ respectively, through 3-order, 5-order, 7-order Lagrange interpolation filter based on Farrow structure. For performance analysis, we calculate error by making subtraction between output signal $y_1(n)$, $y_2(n)$, $y_3(n)$ and tone signal $x_2(\mathrm{n})$, we define the absolute error $E(n)$:

$$E(n) = |y(n) - x_2(n)| \tag{9}$$

Figure 5a is the output absolute error of 3-order Lagrange interpolation filter based on Farrow structure.

Figure 5b is the output absolute error of 5-order Lagrange interpolation filter based on Farrow structure.

Figure 5c is the output absolute error of 7-order Lagrange interpolation filter based on Farrow structure.



(a) (b) (c)

Fig. 5 Output error of Lagrange interpolation filter based on Farrow structure

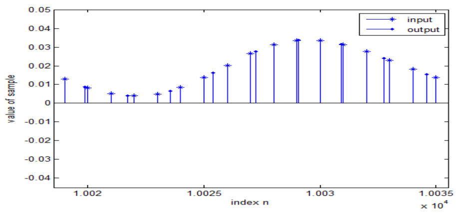


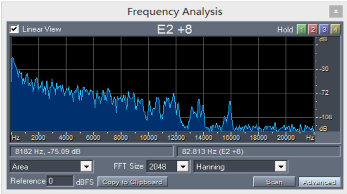
Fig. 6 The relationship between the input and output sample

B. ASRC of CD(44.1 kHz) to DAT(48 kHz)

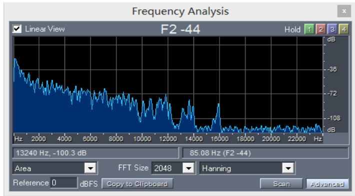
At the last of the simulation, we begin to implement the asynchronous sampling rate conversion of CD (44.1 kHz) to DAT (48 kHz), the original signal is 5 minutes CD audio, using 3-order Lagrange interpolation filter based on Farrow structure can be converted to the 5 minutes DAT audio.

Figure 6 shows a part of the relationship between the input and output sample.

Figure 7, respectively, the frequency domain waveform of the input CD audio and the output DAT audio.



(a)



(b)

Fig. 7 The frequency domain waveform of the input CD audio and the output DAT audio.

Analysis Of Simulation Result

We can found that magnitude of output error of the 3-order Lagrange interpolation filter based on FARROW structure is 10^{-7}, while that in 5-order and 7-order is 10^{-10} and 10^{-14}respectively. It is obvious that with the increase of the order of interpolation filter, its performance has a great improvement.

After that, we adopt the 5 minutes CD audio as input signal, the number of sample is 220500. After the system of sampling rate conversion through MATLAB simulation, output signal is 5 minutes DAT audio, the number of sample points is 22400. Figure 6 shows the relation of the input and output sample in amplitude and position. It can be seen that the output signal is fully comply with the trajectory of the input signal.

Finally, we analysis the input signal and output signal in frequency domain respectively through the Cool Edit software. As shown in figure 7. It can be seen that after sampling rate conversion, the audio signal's characteristics remain the same in 0-20 kHz. And we also can find that their hearing feelings are the same by audition of the two audio signals.

Conclusion

This article, through the MATLAB simulation, we firstly analyze the performance of Lagrange interpolation filter based on Farrow structure, and found that with the improvement of filter order, its performance has a significant improvement. After it, we realized the sampling rate conversion of CD audio to DAT audio. In this process, we adopted 3-order Lagrange differential filter based on Farrow structure for audio signal processing, the coefficient matrix is the structure of [4*4]. By the data analysis later, we found that this method is completely feasible, The quality of the audio signal processing is also pretty good. Here， we only used the 3-order Lagrange interpolation filter. With the higher order Lagrange interpolation filter, the effect will be better.

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