Assignment EE6401 2019
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Introduction

The paper in this assignment considers sampling rate conversion technique performed in the frequency domain. This report compares the techniques preformed in the time domain and in the frequency domain in the following three aspects.

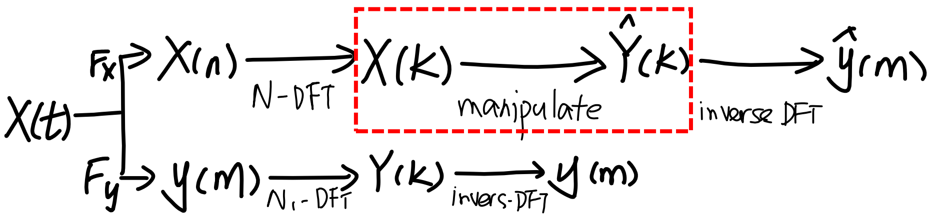
- Briefly compare the basic principles to perform the SRC in the time domain and in the frequency domain;
- Discuss the sources of distortions in the output signals in both methods;
- Compare the required computational complexities required by these methods.

Basic principles in the time domain versus in the frequency domain

Objective of SRC is to convert the sequence x(n) into another sequence y(m), which is ideally the one obtained by sampling the analog signal x(t) with another sampling frequency F_y directly, where F_y/F_x = I/D. Same information should be obtained after SRC.

The SRC in the time domain is achieved by up-sampling, filtering and down-sampling. First up-sampling is inserting I-1 zero values samples between each pair of adjacent samples. Down-sampling is outputting one sample out of every D samples. These 2 processes are both performed in time domain and the desired frequency outputs can be obtained. And a lowpass filter is needed between these 2 parts.

However, the frequency/sampling rate can be change to desired value “one step” in the frequency domain. In short term, the procedure is achieved by first dividing input signal into many small segments and getting DFT of each segment x(n), then manipulating the frequency domain sequence (X(k)) of previous step. Getting the final result by computing the inverse DFT of processed data in frequency domain. A brief structure of SRC in frequency shows in figure below. The upper part is SRC method and the lower part is direct way to get desired output, where N_1 = (I/D) N. One significant difference is that no “filter” structure in frequency domain method, which is important for reducing overlapping issues in time domain and the implements of the filter is related to the computational complexity in time domain.



The output $\hat{y}(m)$ sequence should ideally equal to $y(m)$. and the most important part is in the dotted line, which is manipulating N-point DFT $X(k)$ to N_1-point DFT $\hat{Y}(k)$. the interpolation factor I and decimation factor D are related to N and N_1. Their relation is $N_1/N = I/D$.

If sample rate decreases in a conversion ($D > I$ i.e. $N_1 < N$), which mean the original $X(k)$ give more information (magnitude and phase), so some part data in $X(k)$ will be eliminated to get $\hat{Y}(k)$. Data in the middle of $X(k)$ will usually be eliminated.

If sample rate increases in a conversion ($D < I$ i.e. $N_1 > N$), which means $\hat{Y}(k)$ need more data than those in $X(k)$, so at some points in $\hat{Y}(k)$, inserting some selected numerical values is needed. There are several values, like 0 and $X(N/2)$ (value in the middle).

Overall, to consider SRC for fractional-rate conversion. In the approach of time domain, interpolation is obtained by up-sampling the input first and the filtering, then the decimation is achieved by down-sampling. And the filtering part is carefully implemented. While the SRC is achieved by deleting or inserting some values in frequency domain to get desired result. And the matter issues related to the efficiency will be discussed in the next part.

Sources of distortions in 2 methods

As mentioned in pervious part, the filtering is important in the time domain methods, and it can prevent distortion from spectrum overlapping. But the output could be distorted by losing high frequency component. And passband ripple, transition band of the filter have something to do with the distortion.

In the approach of frequency domain, some details about dividing long signal and insertion value selection do cause different errors. In the paper said, in each segment conversion, the error between $\hat{y}(m)$ and $y(m)$ is lager towards both ends than those in middle. And, there are overlaps between 2 adjacent segments when dividing the long input signal. The paper shows increasing the overlap length in reasonable range can reduce the errors, since the errors are eliminated after final combination. Meanwhile, those inserted values can make difference in the errors between $y(m)$ and $\hat{y}(m)$ since this values has direct impact on the difference between $\hat{Y}(k)$ and $Y(k)$. The paper shows, using $X(N/2)$ for inserting is better.

Computational complexities in 2 methods

In time domain method, efficient implementation of the filter decides the computational complexities. Different from time domain, FFT algorithm required to accommodate input sequence length. The paper shows, for same fractional-rate and same error value, the multiplication per data sample in time and frequency domain are 35 and 8 respectively. Obviously, the second method is better.

Conclusion

This report compares the SRC in the time domain and in the frequency domain in some respects. The techniques in the paper gives efficient way to do SRC in frequency domain, which has 2 important parts, one is efficient FFT algorithm to get the DFT of signal, another is a reasonable way to manipulate data in frequency domain (how to delete and insert in magnitude/phase). By the way, the dividing long input method is also matter. This assignment shows solving problem or processing signal in a different domain could ignore some issues in other domain, which is very important in DSP.