A Digital Filter Design for Digital Re-Sampling in Power System Applications

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Abstract: - Digital filters are designed to provide both satisfied magnitude and linear phase responses for digital re-sampling in power system applications in this paper. This design of filters consists of two steps: magnitude approximation and group delay compensation. It also compromises between the amount of calculation needed and the performance of the filters, which makes the filters good candidates for both real-time and offline digital re-sampling applications especially for the synchronous data acquisition in Wide Area Measurement Systems (WAMS).

Key-Words: - Digital Signal Processing, Digital Re-Sampling, Data Acquisition, Digital Filter, Group Delay.

1 Introduction

Reliable power system control and monitor solutions with low cost and high performance are highly demanded in deregulated power industry. The power industry restructuring is breaking the vertically integrated utility into competitive entities and introducing competition into electric power markets in order to improve economic efficiency [1]. As a result of the deregulation, cost reduction is becoming the utmost important strategy for every energy producer and the transmission service provider to survive in this new competitive industry environment. However, power industry is an intensively invested industry. Not only the electricity generation and transmission facilities, but also the substation controlling and monitoring facilities, are expensive. At present, in a substation, a large amount of expensive control and monitor equipments are installed to enhance the power system secure and reliable operation. However, with the progress made in the digital signal processing and telecommunication technologies [2], substation automation solutions with high performance can be designed with lower cost.

In power system substations, Intelligent Electronic Devices (IED) have been installed and are replacing the conventional electromechanical or solid-state equipments [2][3], such as the digital relay [2][3], the digital fault recorder [2][3] and the Remote Terminal Unit (RTU) or Phasor Measurement Unit

(PMU) [4], etc. At present, the Current Transformers (CT) and the Potential Transformers (PT) are installed in the switch yard and convert high voltage and high current signals into low voltage and low current signals. These analog voltage and current signals are sent to the equipment room a few hundred meters away through cables to be digitized and processed by the IEDs [2][3]. This conventional signal transferring and processing approach in substation automation design has several disadvantages. First, there are large amount of signals to be transmitted, so the wiring system is not only complicated but also very costly in sense of materials and maintenance. Secondly, the analog signals are easily attenuated and distorted when transmitted through the long wires. Thirdly, data acquisition module is highly redundant because each individual IED has its own dedicated acquisition module. However, a shared data acquisition system can be developed and directly placed in the switchyard. It digitizes the analog voltage and current signals on multiple channels and sends the aggregated digital data to the equipment room through high speed data communication system, such as optical fiber or even wireless [3][5]. Thus, the tremendous cost is saved on the material and maintenance in the conventional approach. In addition, if the digital communication carrier is optical fiber and wireless medium, the electrical isolation between the high voltage switchyard and low voltage control equipment are

naturally obtained. Furthermore, this data acquisition system can be shared by several IEDs in the substation so that the IEDs can be built simply on general purpose microprocessor or DSP chip and the cost will be much lower.

For different power system applications, different sample rates are used [6]. Applications, such as fault recorder, use higher sample rate; however the RTUs may use lower sample rate. In order to be shared by different IEDs, the shared data acquisition system must be able to provide data with multiple data rates. A simple approach to achieve this is to digitize the analog signal at a high sample rate in order to keep all required harmonics, and then re-sample the data into sets of data with different sample rates. In IEEE report on COMTRADE [6], a digital re-sample solution is proposed to convert the data captured by the data recorder into the data for offline study or the IEDs testing. For both real-time and offline digital re-sampling, digital filters are required to prevent the signal alias from happening [7].

The design of the digital filter for the digital re-sampling is critical. Above all, the magnitude response needs to meet the spec in order to filter out the unwanted signal components while keeping the interested components. Meanwhile, the phase response of the filter for the interested bandwidth needs to be linear so that the output signal wave-shape is not distorted from the input signal. In [4], it reported that the nonlinear phase response from the digital or analog filter has caused notable signal distortion in the PMU applications for WAMS. In addition, for the real-time application, the order of the filter needs to be kept as low as possible to minimize the computation time. A design of digital filters for digital re-sampling is presented in this paper. This design has two steps: magnitude approximation and group delay compensation. It compromises the performance of the filter and the computation effort. It can be used for both real-time and offline study.

2   A Proposed Digital Data Transmit Solution for Substation Automation

2.1   A Proposed Digital Data Transmit Solution

The IEDs are installed in power system substations to monitor, control or record power system signals (such as voltage and current) and states (such as circuit breaker on/off state). Conventionally, the voltage and the current signals from the secondary sides of the CTs and the PTs are transmitted from the switch yard to the equipment room in analog format through conducting cables as shown in Fig. 3.

The conventional data transfer solution is costly and difficult for maintenance. In the substation, there are a lot of signals to be transmitted, and large amount of long cables are installed for this data transfer purpose. Besides, this data transfer of analog signal through long cable is vulnerable to noise and no error detect scheme can be applied. In addition, the cable does not provide a good electrical isolation between the high power in the switch yard and the low power in the equipment room
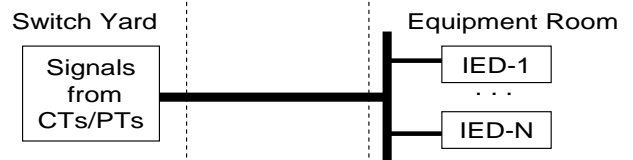

Fig.3. Conventional Analog Signal Transmit Solution

A digital data transmit solution as show in Fig. 4 is proposed in this paper. In this solution, the analog signals from CTs/PTs are digitized at a high speed. The aggregated digital data are sent to the equipment room through the local data communication network which can be high speed internet or high speed optical fiber channel [5]. The digital data received in the equipment room are then converted to digital data with different sample rate or even back to analog signal through a data format converter for the IEDs installed in the equipment room. This digital data transmit solution is not as straightforward as the conventional analog solution. However, it brings a lot more benefits. First, the quality of the data is guaranteed because the error detection scheme can be applied to reduce the error happened in the data transfer. Secondly, the optical fibers or even air in wireless solution are the preferred medium to transmit the digital data, because they provide a perfect electrical isolation between the switch yard and the equipment room. Furthermore, this high speed data acquisition and the data format converter can provide a shared data acquisition for all the IDEs in the equipment room, so the data acquisition module does not need to be designed or populated inside the IDEs. And the IDEs can be developed simply through general purpose catalog digital devices, such as microprocessor, DSP chip or FPGA. The development and manufacturing cost of the IDEs can be much cheaper.
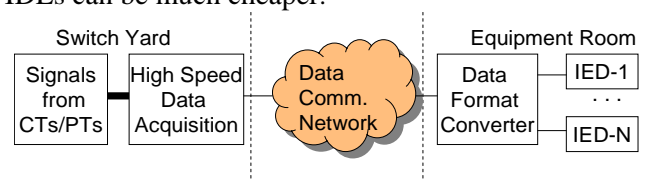

Fig.4. Proposed Digital Data Transmit Solution

2.2 Data Format Converter in the Proposed Digital Data Transmit Solution

This data format converter receives the digital data with high sample rate from the switch yard. It converts the digital data to the right format for the IEDs. At present, almost all of the IEDs have their own data acquisition module to digitize the analog input signal. But some of them may also support digital input data at a specified standard data rate [4][6], and this data format converter converts the high speed digital data into the data rates for these IEDs. For the IEDs not supporting the digital inputs, the data format converter convert digital data back to analog signal. However, the future design of IEDs is strongly recommended to support the digital input data.

In this data format converter, this digital data from the switchyard are converted into the different sets of data with lower sample rates through decimator with dividers as the power of 2 shown in Fig. 5. Low pass digital filters are used to remove the high frequency components in order to prevent signal alias from happening. On the other hand, the input digital data are also converted back to analog signal in order to be used by the IEDs only supporting analog input signals.

The sample rate of the high speed data acquisition in the switchyard is proposed to be 128 points per cycle of the fundamental frequency (60Hz in US and 50Hz in China). The digital data from this data acquisition system could contain the signal components from DC to 64th harmonics. These data are good enough for power system steady state and transient state analysis. In addition, the sample number per cycle is the power of 2, and the Fast Fourie Transform (FFT) can be applied to the frequency spectrum analysis [8].
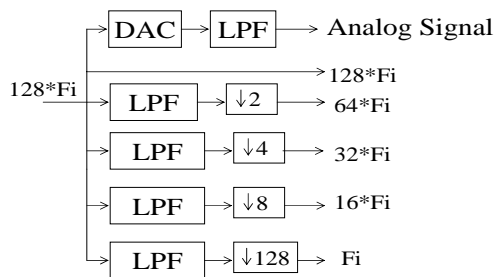

Fig.5. Data Format Converter

The presentation in this paper focuses on the design of the digital filter in the proposed data format converter.

3 Digital Re-sampling and Digital Filtering
3.1 Sampling and Digital Re-sampling

Frequency domain analysis is one of the most popular analysis tools in digital signal processing [12]. However, an improperly designed data acquisition solution may cause the frequency-domain signal meaningless and even misleading. In the sampling theorem [9], a signal with bandwidth B can be reconstructed completely and exactly from the sampled signal if the sampling frequency F_s is chosen to be greater than or equal to 2*B. A low-pass filter with cut-off frequency as B (also called anti-alias filter) is also needed in a data acquisition system. Otherwise, we cannot distinguish between the sampled values of a sine wave of F_i and a sine wave of $(F_i + k * F_s)$ if k is any positive or negative integer. The unexpected signal with frequency higher than $F_s / 2$ appearing in the frequency spectrum is called alias.

Changing sample rate of digital data is called digital re-sampling. It has two types: rate decreases (called decimation) and rate increases (called interpolation) [7]. The decimation is to reduce the sample rate by an integral factor, and the interpolation is to increase the sample rate by an integral factor. In decimation, the sample rate decreases but the bandwidth of the input signal does not decrease. An anti-alias low pass filter is needed to remove the component with frequency higher than $F_s / 2$. In interpolation, new samples need to be added into the data sequence and the new sample values need to be calculated. Zero-padding [7] method is often used for interpolation. In the zero-padding method, zeros are inserted into the original data sequence. Then the new data sequence is processed through a low-pass filter (called interpolation filter). If the sample rate change factor is not integral, the sample rate change will need to be implemented with interpolation followed by decimation as shown in Fig.6.
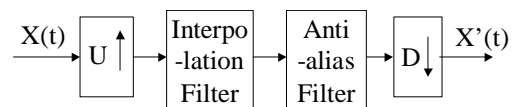

Fig.6. Digital Re-Sampling

3.2 Filtering

Filtering is the processing of a time-domain signal resulting in the reduction of some unwanted input spectral components [10]. The filters allow certain frequencies to pass while attenuating other frequencies as shown in Fig. 7.
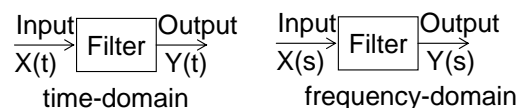

Fig.7. Filter

The transfer function of the filter in frequency domain is expressed as:

$$H(s)\big|_{s=j\omega} = \frac{Y(s)}{X(s)}\bigg|_{s=j\omega} = |H(j\omega)|e^{j\phi(\omega)} \qquad (1)$$

In frequency domain, the magnitude and the phase responses of the network function are the two main factors of designing the filter. The magnitude response is studied frequently in db through the gain function as in (2). The phase response is expressed by phase function or group delay function as in (3). The group delay function and the phase function have profound time-domain ramifications as they have a directly effect on the wave-shape of the output signals.

$$\alpha(\omega) = 20\log|H(j\omega)| \qquad (2)$$

$$T_d(\omega) = -\frac{d}{d\omega}\phi(\omega) \qquad (3)$$

An analog filter operates on a continuous signal. However, a digital filter processes a sequence of discrete sample values.

3.3 Digital Filtering

Conventional linear digital filters typically come in two types: finite impulse response (FIR) filters and infinite impulse response (IIR) filters. The FIR digital filters use only current and past input samples to obtain a current output sample value as shown in (4). The transfer function in z-plane for the FIR filter is as in (5).

$$y(n) = \sum_{k=0}^{N-1} h(k)x(n-k) \qquad (4)$$

$$H(z) = \frac{Y(z)}{X(z)} = \sum_{k=0}^{N-1} h(k)z^{-k} \qquad (5)$$

Where: N is the order of the filter; h(k) is the coefficients of the filter.

The IIR filters output sample value depends on previous input samples and previous filter output samples as shown in (6). The transfer function in z-plane for the IIR filter is as in (7).

$$y(n) = \sum_{k=0}^{N} a(k)x(n-k) + \sum_{j=1}^{M} b(j)y(n-j) \qquad (6)$$

$$H(z) = \frac{Y(z)}{X(z)} = \frac{\sum_{k=0}^{N} a(k)z^{-k}}{1 - \sum_{j=1}^{M} b(j)z^{-j}} \qquad (7)$$

Where: N is the order of the numerator of the filter; M is the order of the denominator of the filter; a(k) is the coefficients of the numerators of the filter; b(k) is the coefficients of the denominators of the filter.

4 Design of Digital Filters for Data Format Converter

4.1 Requirement of the Digital Filters

For this data format converter, low pass filters are designed to attenuate the high frequency components before applying the decimation. In general, it is desirable to design a filter to have: (1). Small passband ripple; (2). Large stopband attenuation; (3). Low transition–band ratio; (4). Constant and small group delay; (5). Lower order of the transfer function.

In the data acquisition system for the substation automation, the Analog to Digital Converter (ADC) used is generally 14 bits or 16 bits. The ADC quantization error noise [11] is:

$$SNR = (n+1)*6.02 + 1.76 \qquad (8)$$

Where: N is number of bit in the ADC.

So, the spec for the stop attenuation is set at 104 db for these filters. The passband ripple is 0.3db, and the transition-band ratio is 1/0.9. The maximal deviation of group delay (in number of samples) within the interested bandwidth is smaller than 1. The transfer functions for the filters need to be designed. And the objectives of this design are to: (1) minimize the order of the transfer function in order to reduce the amount of the calculation; (2). minimize the number of samples in group delay while keeping it as constant as possible in the interested bandwidth so that there is less wave-shape distortion and shorter delay between the input and the output signals.   The problem is formulated as a multi-objective optimization problem [1] as:

Minimize: number of total orders
Minimize: number of samples in group delay
  s.t.:
      passband ripple < 0.3 db
      stopband attenuation > 104 db
      transition-band ratio < 1/0.9
      maximal deviation on group delay < 1 sample

4.2 Design Approach for the Digital Filters

The FIR filters with symmetric coefficients have linear phase response (or constant group delay) property [7]. However, the order of the filter might be very large and the number of the samples for the delay between the input and the output signals is half of the number of the orders [7]. More calculations will be needed and the more delay will be involved if these types of FIR filters are used for this data format converter. The IIR filters have much fewer orders than the FIR for the same filter spec. However, it can not guarantee the constant group delay. In addition, there is no design approach and tool to simultaneously optimize both magnitude and group delay requirement for the IIR filters. So, the design for the IIR filters involves two steps in this paper: (1). Magnitude

approximation; (2). Group delay compensation. At the magnitude approximation step, the filters are designed with minimal order and satisfy the requirements on the passband ripple, the stopband attenuation and the transition-band ratio. At the group delay compensation step, another all pass filter is designed to make the group delay constant within the interested bandwidth.

Due to the tradeoff between the objective functions, there might not have one solution optimal for both objective functions. A value function [1] for this multi-objective optimization is defined as:

Minimize: (number of total order) + w * (group
delays)

This optimization problem works on three types of filters: the equal-ripple FIR filter, the elliptic filter and the Chebyshev II IIR filter. So the type of filter and the number of order are the decision variables for this problem.

4.3 Design Result for the Digital Filters

This filter design is for the data format converter as proposed previously. It aims for real-time applications. "w = 10" is selected in the value function to put more weight on the objective for the group delay. The selections on the filters for the decimators with rates of 2, 4 and 8 consider both magnitude and phase response. And the phase response is not considered in the filter for the decimator with rate of 128 because the main purpose for this decimation is for the application of monitoring low frequency oscillation issue in power system.

From the solution for this optimization problem, the filters for the decimator with rate of 2 and 4 are an elliptic IIR filter cascaded with an all-pass IIR filter. The total number of the orders and the total number of samples in the group delay for interested band are small and acceptable. However, the filter for decimator with rate of 8 is an equal-ripple FIR filter. The order is 585 and the number samples in group delay is 292.5, which is unacceptable for real-time application. So the architecture for this digital re-sampling in the data format converter is designed as in Fig.8.
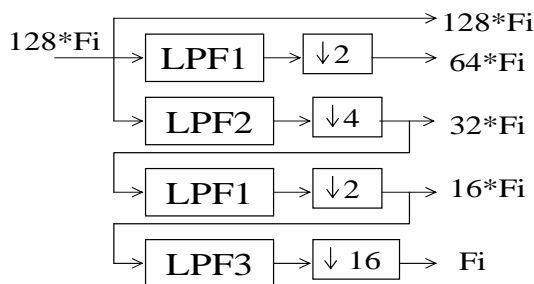

Fig.8. Digital Re-Sampling for Data Format Converter

The designed LPF1 consists of two filters: the elliptic IIR filter for the magnitude approximation and the all-pass IIR filter for the group delay compensation. The orders for the filters are 12 for the elliptic IIR filter and 6 for the all-pass IIR filter. The number of samples in the group delay of the interested bandwidth from 0 to 0.383 (normalized frequency) is about 19.4 with maximal variation of 0.90. The magnitude response and the group delay for the whole filter network are shown in Fig.9 and Fig. 10. From the figures, it is notable that the magnitude satisfies the filter spec and the group delay is constant in the interested bandwidth.
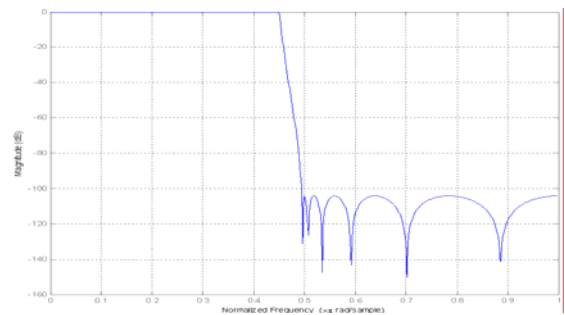

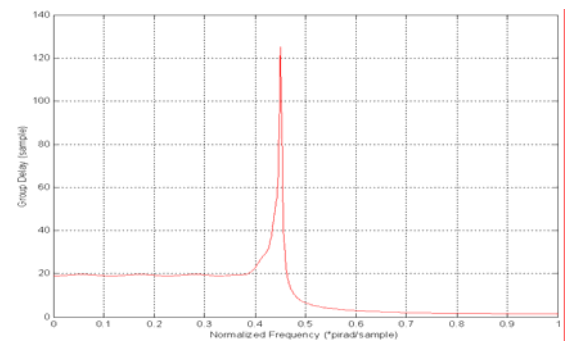Fig.9. Magnitude Response of LPF1

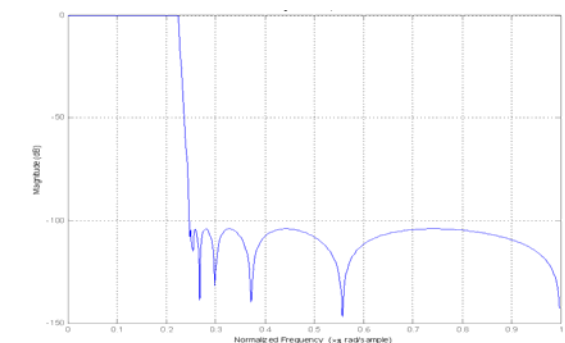Fig.10. Phase Response as Group Delay of LPF1

Fig.11. Magnitude Response of LPF2

The LPF2 also consists of two filters: the elliptic IIR filter and the all-pass IIR filter. The orders for the filters are 13 for the elliptic IIR filter and 8 for the all-pass IIR filter. The number of samples in the group delay in the interested bandwidth from 0 to

0.195 (normalized frequency) is about 48.5 with maximal variation of 0.76. The magnitude response and the group delay are shown in Fig.11 and Fig. 12. From the figures, it is notable that the magnitude satisfies the filter spec and the group delay is constant in the interested bandwidth.
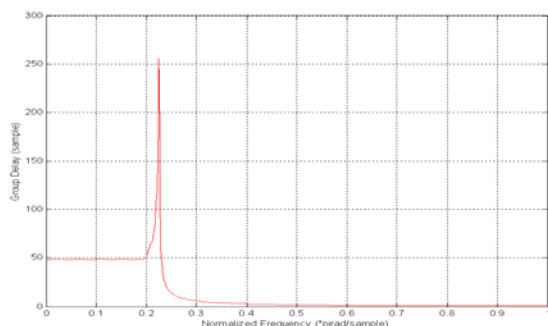

Fig.12. Phase Response as Group Delay of LPF2

The transition-band ratio of the filter for the decimator with rate of 16 is re-defined as 1/0.8 because no stable elliptic IIR filter in direct form structure was able to be obtained. The designed LPF3 is an elliptic IIR filter with order of 11. The Magnitude response is shown in Fig.13.

5   Conclusion and Future Work

A digital re-sampling solution is proposed for power system applications, especially for low cost substation automation development. Digital filters are designed with consideration of both magnitude and phase responses for this digital re-sampling solution. These digital filters have lower orders and fewer samples in group delay. They can be used for both real-time applications and offline studies. The simulation in the design demonstrate the good performance these filters.
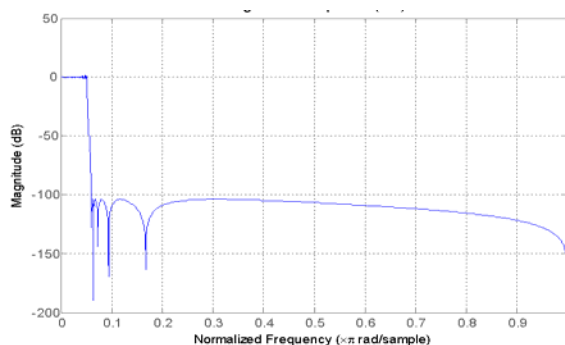

Fig.13. Magnitude Response of LPF3

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