SIGNAL PROCESSING ALGORITHM FOR BEAM POSITION AND PHASE MONITORS AT LANSCE

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Abstract

The new beam position and phase monitors at LANSCE measure the phase of the beam relative to a reference signal from the master reference oscillator. Because of the various beam pulse formats used at LANSCE the algorithm needs to be flexible and to work well with short bursts of signals. We have developed an algorithm that provides phase resolution of better than 0.25 degrees with signal bursts one microsecond long, and also allows measurement of bursts as short as 100 nanoseconds. For beam position measurements flexibility took priority over precision; the processing scheme provides precision of less than 0.1 mm. In this paper we will present the principles of the algorithm and results of measurements.

INTRODUCTION

We are preparing to install beam position and Phase Monitors (BPPMs) in the linac at the Los Alamos Neutron Science Center (LANSCE.) The transducers are 4-electrode shorted-stripline detectors; the 201.25MHz signals from the electrodes are sampled at 240 Msamples/second, along with a 201.25MHz reference signal. These sample streams are processed using a custom algorithm in a field-programmable gate array (FPGA) to provide both the position of the beam and the phase (arrival time) with respect to the reference signal.

The signal-processing algorithm was developed to provide good precision for the phase measurement, as this is used for the Δt turn-on process [1] for the linac, where the energy of the proton beam can be inferred using phase measurements at two BPPMs. Another feature of the algorithm is its ability to provide good measurements using short bursts of beam signals. This is important because the LANSCE linac provides beams with various pulse formats to several user facilities.

In the following sections the algorithm is described.

THE ALGORITHM

The algorithm can be thought of as a fit of a sinusoid to the data. That is, an amplitude, phase, and DC offset that best fit the data are determined using a linear fitting process.

The i^{th} data sample of an electrode signal is:

$$y_i = A\cos(wi+\phi)+y_{DC} \tag{1}$$

A and ϕ are the signal amplitude and phase, y_{DC} is the DC offset, and w is the phase advance of the RF waveform per sample interval:

$$w = 2\pi f_{RF} \div f_{sample}.$$

In order to make the fit linear in the fit parameters, Eq. 1 can be written as:

$$y_i = a\cos(wi)+b\sin(wi)+y_{DC}$$
$$\text{where } A^2 = a^2+b^2 \text{ and } \tan\phi = b/a \tag{2}$$

To keep the notation compact, the series of N samples to be analyzed can be denoted by:

$$\vec{y} = a\vec{c}+b\vec{s}+y_{DC}\vec{u} \tag{3}$$

where $\vec{c}$ is the series of values of cos(wi), $\vec{s}$ is the series of values of sin(wi), and all elements of $\vec{u}$ are one. Each of these vectors has N elements.

Now equation 3 can be multiplied by each vector, $\vec{c}$, $\vec{s}$ and $\vec{u}$:

$$\begin{aligned}\vec{c}\cdot\vec{y} &= a\vec{c}\cdot\vec{c}+b\vec{c}\cdot\vec{s}+y_{DC}\vec{c}\cdot\vec{u}\\ \vec{s}\cdot\vec{y} &= a\vec{s}\cdot\vec{c}+b\vec{s}\cdot\vec{s}+y_{DC}\vec{s}\cdot\vec{u}\\ \vec{u}\cdot\vec{y} &= a\vec{u}\cdot\vec{c}+b\vec{u}\cdot\vec{s}+y_{DC}\vec{u}\cdot\vec{u}\end{aligned} \tag{4}$$

Note that each vector dot product is a scalar number. The entire data series to analyse is now represented by 9 numbers (because some of the dot products appear twice in Eq. 4.) Equations 4 can be written in matrix form:

$$\begin{pmatrix}\vec{c}\cdot\vec{y}\\ \vec{s}\cdot\vec{y}\\ \vec{u}\cdot\vec{y}\end{pmatrix} = \begin{pmatrix}\vec{c}\cdot\vec{c} & \vec{c}\cdot\vec{s} & \vec{c}\cdot\vec{u}\\ \vec{s}\cdot\vec{c} & \vec{s}\cdot\vec{s} & \vec{s}\cdot\vec{u}\\ \vec{u}\cdot\vec{c} & \vec{u}\cdot\vec{s} & \vec{u}\cdot\vec{u}\end{pmatrix}\begin{pmatrix}a\\ b\\ y_{DC}\end{pmatrix} \tag{5}$$

Now all that needs to be done is to invert the matrix in Eq. 5 and multiply the inverse onto the vector on the left-hand side. This will determine the 3 quantities to be fit.

The Vectors of Sines and Cosines

Because the frequencies of the RF and sampling are known, one could in principle compute $\vec{c}$ and $\vec{s}$. We found, however, that slight drifts of the sampling frequency caused problems with the stability of the phase

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measurements, especially with long records (series of data.)

Instead, the stream of samples of the reference signal can be used as the source of $\vec{C}$ and $\vec{S}$. The vector of cosines is taken directly from the reference samples: $c_i = r_i$ where r_i is the i^{th} sample of the reference signal.

To generate the vector of sines, two consecutive samples of the reference are used:

$$s_i = \frac{r_{i-1} - r_i \cos w}{\sin w} \quad .$$

(This can be thought of as a two-tap 90° phase shift FIR filter.)

One advantage of using the reference signal as the source of the fitting model is that the phase of the beam signal is measured directly with respect to the reference. If the beam signal and reference were fitted separately, one would then need to subtract the two phases and deal with phase-wrap issues to get an answer.

IMPLEMENTATION

The dot products in Eq. 5 are computed as data samples stream into the FPGA. The 16-bit samples are multiplied and accumulated and, when the end of an analysis record is reached, the accumulated sums are clocked into registers. This enables the accumulators to begin processing the next record as further analysis steps are executed.

In order to provide the numerical dynamic range necessary to accommodate the wide range of signal strengths and record lengths that we anticipate, much of the algorithm is implemented using floating-point arithmetic. The integer dot products are converted to floating-point representation and the matrix inversion is carried out by expansion of cofactors [2]. The inverse matrix is used to determine the fit parameters. These parameters are converted to amplitude and phase according to Eq. 2. Finally the amplitude, phase, and DC offset are converted back to integers and communicated to the external data system.

This algorithm is rather expensive in terms of FPGA fabric, especially multiplier blocks. The benefits in terms of versatility and precision of the algorithm justify this cost.

PERFORMANCE

One advantage of this approach over algorithms based on Fourier transforms (such as the Goertzel algorithm [3]) is its ability to operate stably on short records. As is typical with these techniques, short records with produce ringing effects in the frequency domain. In particular the measured phase using the Goertzel algorithm depends on the record length, while the fitting algorithm shows no such systematic dependence.

The algorithm presented here works well on beam pulses <100ns long; this is beneficial for us, as such short pulses are frequently used during linac tune-up and for delivery to one of our user facilities.

This system has shown phase resolution of better than 0.25°, meeting our requirement for the Δt turn-on procedure.

We have performed in-situ measurements on this system in order to ensure that the system was characterized in the environment in which it will operate. To perform the tests, we drove one electrode of a transducer with 201.25MHz RF, and picked up the signal on two other electrodes. We used record lengths of 100 samples (400ns.) We used a variety of signal strengths, and found that with signals of ±200 ADC counts the phase resolution was better than 0.1° and the position resolution was better than 20µm.

SUMMARY

The algorithm described above measures the amplitude and phase of a beam-induced signal with respect to a reference signal by using the reference signal as the fit function. This method works well with a wide range of record lengths and signal strengths.

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