Frequency measurement under non-sinusoidal conditions

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Abstract. Frequency measurement is an important issue in electrical engineering. Electric power systems have become increasing complex over the last decade. The use of distributed generation, the connection of non-linear loads and the presence of unexpected system faults are the main causes of frequency variations. In addition, power quality includes frequency as an important index.
From a hardware instrumentation point of view, frequency measurement has different requirements:

i.) Large power systems have slow frequency variation due to the high inertia of the overall network. This kind of problem requires frequency measuring methods capable of detecting small and also slow frequency variations.
ii.) Small power systems can have frequency variation due to their reduced short-circuit power capacity. This type of problem requires fast methods with the capacity to detect large frequency variations.

There are different groups of methods intended for frequency measurement. The methods can be compared in terms of computation and dynamic response, especially when the main voltage is disturbed.

This research work focuses on frequency measurement under non-sinusoidal conditions. The paper studies the behaviour of a modified version of Sezi's method and its hardware implementation using a microcontroller. This system can be used for frequency measurement or as a synchronized sampling source in harmonic measurement (e.g. EN 61000-4-7)

The fast time response of the system enables it to be used in almost all kinds of application: small and slow frequency variations; frequency triggering in power system protection and power quality index characterisation.

Key words

frequency meters, power quality, non-sinusoidal conditions, instrumentation.

1. Introduction

Most power system experts do not consider frequency variation as an important topic because it is rather difficult to find large frequency variation in continental power networks. However, from the point of view of the instrumentation design, some instrumentation devices used to register power quality disturbances exhibit a great sensitivity to small frequency variations. For instance, the European standard EN 61000-4-7 [1] defines the characteristics of an instrumentation device used for harmonic distortion measurement based on analog to digital conversion. This device includes a phase locked loop in order to adjust the sampling frequency dynamically but does not include any comment about the behaviour of the method used to compute the fundamental frequency.

The paper includes a review of the behaviour of frequency measurement under non-sinusoidal conditions from two points of view:

- Frequency measurement for power quality assessment according to EN 50160 [2]. This document defines the characteristics of the supplied voltage: frequency, magnitude variations, unbalance, harmonics, surges, etc.
- Frequency measurement as a reference for sampling control in analog to digital conversion. (e.g. EN 61000-4-7)

From the perspective of the type of algorithm used for frequency computation, there are several possibilities:

- Adaptive Kalman filtering methods.
- Zero-cross detection methods.
- Discrete Fourier Transform methods.
- FIR/IIR iterative methods.
- Other methods.

In order to evaluate the differences between the different methods, some algorithms have been implemented using a high performance microcontroller PIC 18F452 [3] from Microchip. The designed hardware includes a RS232 connection in order to send the computed data to a PC. Figure 1 shows the block diagram of the proposed system.
An iterative algorithm based on FIR filter has showed a good behaviour under non-sinusoidal conditions [4]. The method is not dependent upon zero-crossing of the measured voltage or current. In addition it can compute

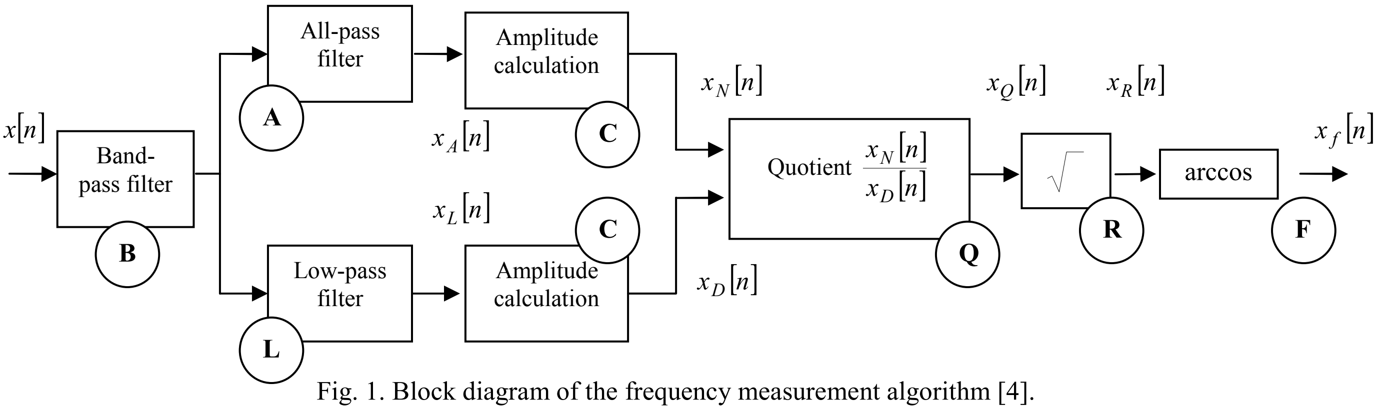

Fig. 1. Block diagram of the frequency measurement algorithm [4].

the frequency after every sample. In addition, the method only needs a 3-cycles signal-window.

The accuracy of the computed frequency is better than 10 mHz for a nominal frequency of 50 Hz.

This frequency meter is basically insensitive to steady state disturbances like harmonics or low frequency amplitude variations so it can be used as a reference method in order to obtain an accurate sampling frequency in harmonic meters

2. Frequency measurement under non-sinusoidal conditions

The above methods can be included in one of three main principles that support frequency measurement [5]:

- Period estimation based on the measurement of time interval between zero-crossing [6].
- DFT based method, which uses interpolation in frequency domain [7, 8].
- Orthogonal phasor decomposition with local determination of rotation speed [9-12].

The behaviour of each group of methods under non sinusoidal conditions depends on the kind of disturbance. In general, zeros-crossing methods can be used with harmonic distortion, especially with high-order harmonics.

DFT methods have to be implemented as block algorithms so this kind of procedures needs from two to eight cycles to compute frequency.

Orthogonal phasor decomposition, like the zero-crossing method, belongs to the group of iterative filter methods.

All methods are sensitive to transient distortion such as sags, swells and impulses. On the other hand, they are rather insensitive to white noise and harmonic periodic distortion.

3. Frequency algorithm

The proposed algorithm is based on Sezi's method [4]. This algorithm uses phase angle and amplitude information in order to compute frequency. This method avoids waiting between zero-crossing. Figure 1 shows the block diagram of Sezi's method with the stages involved.

The sampled data $x[n]$ is first filtered by a band-pass FIR filter B. The main aim of this block is to filter all harmonics, subharmonics and inter-harmonic distortion. This bandpass filter B has been designed with the following restrictions:

Equiripple FIR bandpass filter of order 40
Sampling frequency 1000 Hz
$Fstop_l$ = 10 Hz; $Fpass_l$ = 30 Hz
$Fpass_h$ = 80 Hz; $Fstop_h$ = 120 Hz
$Amplitude_{pass}$ = 1
$Amplitude_{stopl}$ = 60; $Amplitude_{stoph}$ = 80

The filter coefficients are included in Table I.

TABLE I. Bandpass filter B coefficients.

Coef.	Value	Coef.	Value
1	0.007749520989041	21	0.123420397846289
2	0.012535189342240	22	0.115871771716403
3	0.019230726379304	23	0.094319125134788
4	0.024578269196053	24	0.061865040318399
5	0.026195484608224	25	0.023205602586402
6	0.021918288148958	26	-0.016101012847003
7	0.010529708044245	27	-0.050489670521057
8	-0.007644015196679	28	-0.075333780176352
9	-0.030509573299857	29	-0.087736170069564
10	-0.054396150203906	30	-0.086999488341449
11	-0.074734365352290	31	-0.074734365352290
12	-0.086999488341449	32	-0.054396150203906
13	-0.087736170069564	33	-0.030509573299857
14	-0.075333780176352	34	-0.007644015196679
15	-0.050489670521057	35	0.010529708044245
16	-0.016101012847003	36	0.021918288148958
17	0.023205602586402	37	0.026195484608224
18	0.061865040318399	38	0.024578269196053
19	0.094319125134788	39	0.019230726379304
20	0.115871771716403	40	0.012535189342240
		41	0.007749520989041

Figure 2 shows the amplitude versus frequency response of the bandpass filter.

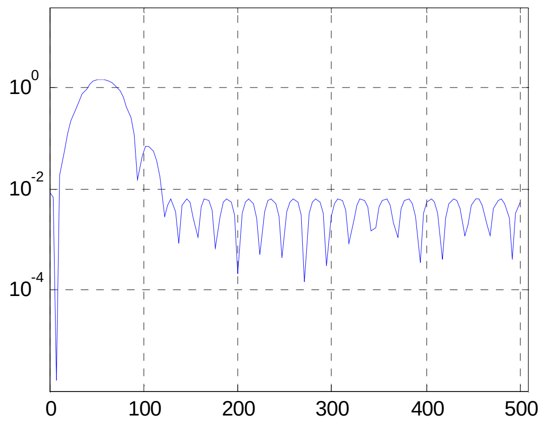

Fig. 2. Amplitude response of bandpass filter B.

The output of the bandpass filter B is connected in two different ways. The first one is an all-pass filter whereas the second is a low-pass filter. Equations (1) and (2) show the frequency transfer function of both filters,

$$A(f)=1 \quad (1)$$

$$L(f)=1+\cos(5\Omega) \quad (2)$$

$$\Omega = 2\pi\left(\frac{f}{f_s}\right) \quad (3)$$

where

Ω is the normalized frequency,
f actual frequency
f_s sampling frequency

The Z domain transfer function of filters A and L are

$$H_A(z)=z^{-5} \quad (4)$$

$$H_L(z)=0.5+z^{-5}+0.5z^{-10} \quad (5)$$

Figure 3 shows the amplitude versus frequency response of filters A and L.

The outputs of filters A and L pass through a block C that computes the amplitude of the signals $x_A[n]$ and $x_L[n]$. In fact, block C obtains a value proportional to the square of the signal amplitude. The proposed C block [13] differs from original Sezi's in some aspects: i) It is a IIR filter and ii) It has a reduced computational load.

The difference equation of this filter is,

$$y_C[n]=y_C[n-1]+0.1\left(x_C^2[n]-x_C^2[n-10]\right) \quad (6)$$

The block Q computes the quotient of the signals $x_N[n]$ and $x_D[n]$. The output of block Q is root squared in order to obtain $x_R[n]$. The amplitude frequency response of signal $x_R[n]$ is,

$$x_R[n]=[1+\cos(5\Omega)] \quad (7)$$

The last block F performs the inverse of cosine function in order to obtain Ω from equation (7),

$$\Omega=\frac{1}{5}\arccos(x_R[n]-1) \quad (8)$$

The actual frequency f can be directly obtained from expression (8),

$$f=\left(\frac{f_s}{2\pi}\right)\Omega \quad (9)$$

Another advantage of Sezi's method is that frequency dependent errors are eliminated because all the computations are done using the same numerical process.

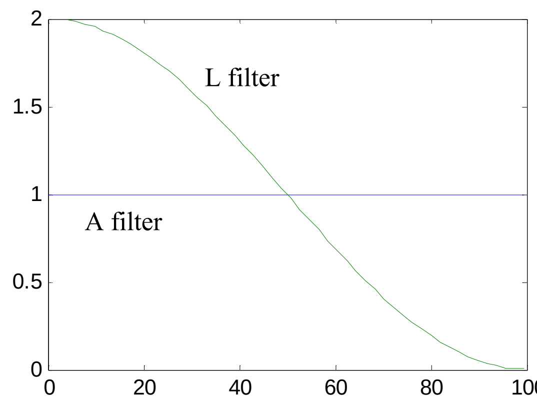

Fig. 3. Amplitude versus frequency response of filters A and L.

4. Instrumentation architecture

The algorithm has been implemented using a Microchip® microcontroller PIC 18F452 [3]. This microcontroller is a high performance RISC device with 32 kB (16.384 single-word instructions) on-chip program memory and a computation speed up to 10 MIPS with internal hardware multiplier. In spite of the fact that it has a lot of peripherals, it is important to highlight its eight 10 bits analog to digital converter.

Figure 4 shows the instrumentation basic block diagram.

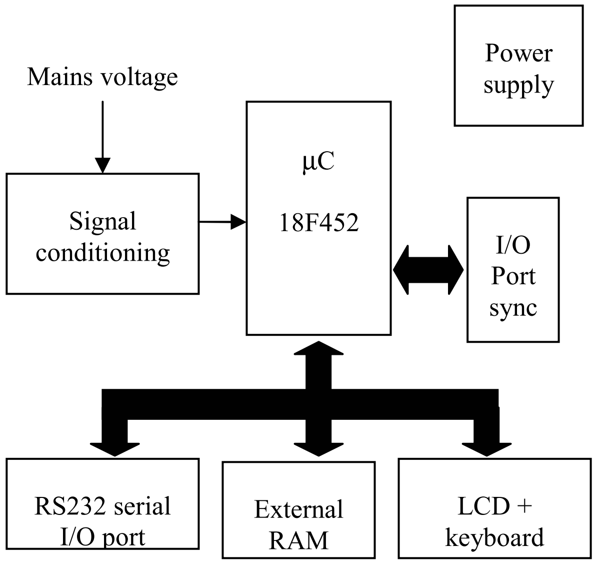

Fig. 4. Instrumentation block diagram.

The hardware designed has some advantages from the point of view of the connexion with other instrumentation:

- It is extremely simple and cheaper. In fact, all the design can be reduced to the microcontroller without the LCD, keyboard and I/O serial port.
- The synchronization I/O port can be used as a trigger source for other power quality instruments or protection devices.

5. Test results

The implemented hardware has been putting to the test with an arbitrary programmable power supply in order to know the accuracy and time response.
This programmable three phase source HP6834B [14] allows arbitrary waveforms to be generated with a maximum power of 1500 VA in every phase or 4500 VA in the case of one phase generation. The power supply is controlled using a PC and a GPIB link.
This frequency meter has been tested with two kind of signals: i) sinusoidal and ii) non-sinusoidal.
Table II summarizes statistical results obtained from sinusoidal signals with frequencies varying from 46 to 54 Hz in steady state.

TABLE II. Statistical values obtained from test signals with frequencies varying from 46 to 54 Hz.

frequency	46	48	50	52	54
Min	45.9369	47.9426	49.9257	51.9039	53.8788
Mean	46.0002	48.0002	50.0005	52.0003	54.0001
Max	46.0679	48.0606	50.0749	52.0927	54.1252
Std	0.0386	0.0383	0.0469	0.0599	0.0794

The time needed to reach the steady state is over three cycles but can be reduced if filter with shorter order are used.

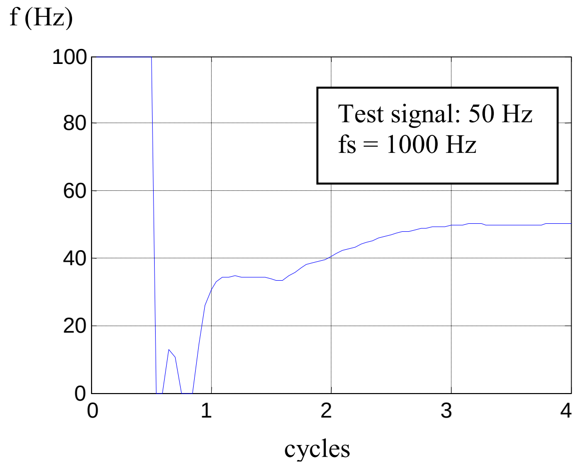

Fig 5. Time evolution of computed frequency during system initialization.

The transient response showed in Figure 5 also appears when the main signal is disturbed with some disturbances like sags, swell and impulses. Figure 6 shows the transient response during a 50% sag.

Finally, the system was tested using a non-sinusoidal signal disturbed with harmonics of orders three (30 %) and five (20 %). Figure 7 shows the steady state response.

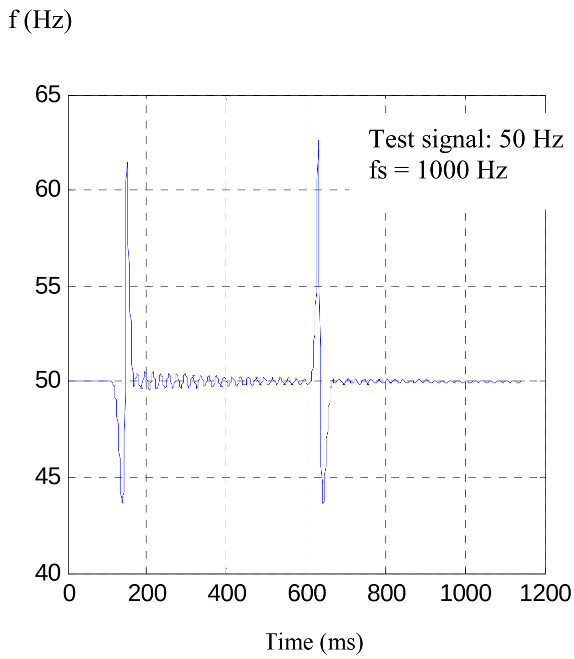

Fig. 6. Transient response of computed frequency during a 50% voltage sag.

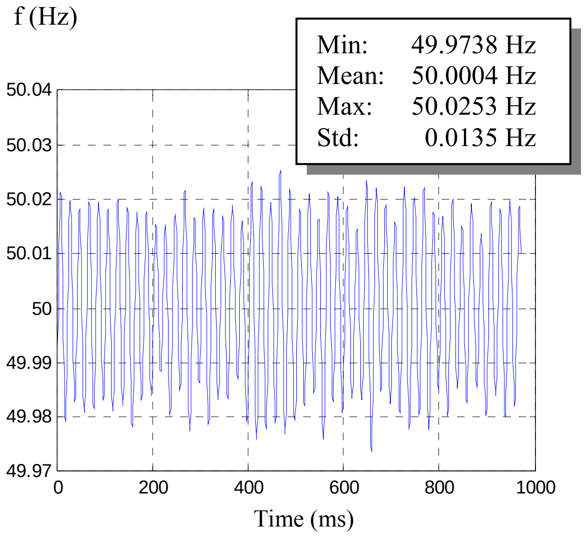

Fig. 7. System response with a 50 Hz signal disturbed with harmonics of orders three and five.

6. Conclusions

It has been probed that the modified Sezi's method has a good response under both sinusoidal and non-sinusoidal conditions.

The modification of the amplitude computation block C reduces the computational requirements. The overall system has been implemented in a real minimum hardware system based on a microcontroller.

One of the major advantages of this method is that this algorithm calculates the system frequency after every data sampling with a reduced computational load.

The initialization time is over three periods of 50 Hz (60 ms).

The system exhibits some instability when the amplitude changes suddenly, as in sags and transients. One way to solve this problem is watching the calculate frequency. If the frequency is out the range [40, 60] Hz, then the computed frequency is not valid and a warning digital output is set on.

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