New Control Algorithm for Shunt Active Filters, Based on Self-Tuned Vector Filter.

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Keywords

Active filters , Power factor correction, Power Quality

Abstract

A new, improved, method for calculating the reference of a shunt active filter is presented. This method lays on a filter, which is able to extract the main component of a vector signal. This filter acts as a Phase-Locked Loop, capturing a particular frequency. The output of this filter is in phase with the frequency isolated, and has its amplitude. Simulation and experimental results confirms the validity of the proposed algorithm.

I. Introduction

Active filters (AF) is a fast-growing field of investigation, as new regulations on power quality and polluting loads are increasing simultaneously. In recent years, some new control algorithms has been proposed, trying to measure or estimate harmonics and reactive power instantaneously [1],[2],[3],[4]. Some of this method shows a weakness when the voltage, at the point of common connection (PCC), is not as good as it should be. This method avoids this problem, as it filters both currents and voltages. The filter used for this purposes is a new self-tuned vector filter (STVF), basically composed of some rotations and low pass filtering. Simulation and experimental results confirms an improvement of the AF performance, in terms of less resulting harmonics with the same control system and load.

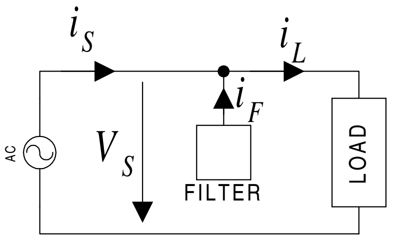
The present paper is intended to solve the problem of generating an appropriate reference to be followed, not being concerned about how to track this reference. The complementary problem, on how to follow the reference, is being studied simultaneously by the authors [6]. The paper have been structured in six sections. First, in Section II, we will discuss the two fundamental ways to implementate an Active Filter algorithm, concluding with the selection of 'sinusoidal current source'. Then, in section III we will present the filter, named Self-tuned Vector Filter, and will present some simulations about its behaviour in time and frequency domain. Later, in section IV we show how to use this filter to develop a Parallel Active Filter, presenting also some simulation results. Finally in section V we will show experimental results obtained using this filter algorithm, and some of them obtained using a classical algorithm, comparing the two methods, and in section VI will present some conclusions.

II. Parallel Active Filter Model

As is well known, a Parallel Active Filter acts injecting instantaneously the current necessary to compensate harmonics and, eventually, reactive power flowing to the load. Basically, there are two models that can respond to this behaviour: the *resistive load* model and the *sinusoidal current source* model:

Resistive load model

In this model, represented in Fig. 1,the filter acts injecting the current necessary to make the entire system (load plus filter) as a *resistive load*. Then, the current should be proportional to the voltage at every moment. This is the approach used in many of the classical methods for doing active filtration, although present some problems in certain conditions:



$$
\begin{aligned}
i_F &= i_L - K \cdot V_S \Rightarrow i_S = K \cdot V_S \\
i_S &= K \cdot V_S = K \cdot (V_S{}^1 + \sum_j V_S{}^j)
\end{aligned}
\qquad \textbf{Eq. 1}
$$

Fig. 1: Resistive Load Model

- If the voltage waveform is not purely sinusoidal, the harmonics present in the voltage will be present too in the current waveform, as could be seen in Eq. 1, where V_S is the voltage source, and V^j_S are the harmonic components of the voltage waveform.
- If there is a not negligible line impedance, there will be a feedback of the harmonics of the current to the voltage, and then from the voltage to the current, that could make the entire system unstable. This problem is represented in Fig 2, and summarised in Eq. 2

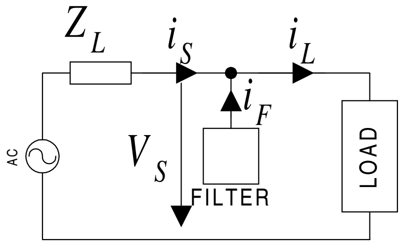


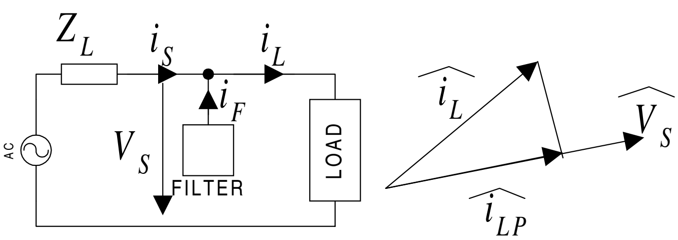
Fig.2: Resistive Load Model with source impedance

$$
\begin{aligned}
V_F &= V_S - Z_L \cdot i_S \\
i_F &= i_L - K \cdot V_F = i_L - K \cdot Vs - K \cdot Z_L \cdot (i_L - i_F)
\end{aligned}
\qquad \textbf{Eq. 2}
$$

These problems makes it suitable only where the voltage source is good enough, in terms of harmonic distortion and in terms of short circuit current.

Sinusoidal Current Source Model

In this model, what we try to do is to convert our system (load plus filter) in a sinusoidal current source, with an adequate amplitude and in phase with the fundamental harmonic of the line voltage. Ideally, this current source will have the same amplitude that the first harmonic of the current, projected over the voltage vector. In Fig. 3 is shown a representation of that, and in Eq. 3 the basic relations between the magnitudes are presented.



$$
\left.
\begin{aligned}
\hat{i}_L = i_L{}^1 ; \hat{V}_S = V_S{}^1 \\
\hat{i}_{LP} = \hat{i}_L \cdot \hat{V}_S \\
i_S = \hat{i}_{LP} \Rightarrow i_F = i_L - \hat{i}_{LP}
\end{aligned}
\right\}
\qquad \textbf{Eq. 3}
$$

Fig. 3: Sinusoidal Current Source Model

Note that Eq. 3 represent vectorial magnitudes, so the product is a dot product (projection over V).

With this approach, the voltage harmonics have no influence, as both voltage and current are being filtered. The problem lays precisely in that needing of filtering voltages and currents, because it is usually difficult to tune a filter for extracting one component only, with no delay.

This second approach presents much better results in distorted and non-ideal conditions of voltage source, and so is to be chosen.

III. Self-Tuned Vector Filter

The basis of this new control method is the Self-tuned Vector Filter developed by the authors. This filter extracts one frequency component from a vectorial signal. The equations of this filter are, in time domain, as shown in Eq.4

$$\left.\begin{aligned} \frac{d\hat{i}_\alpha}{dt} &= (i_\alpha - \hat{i}_\alpha)\cdot k_f - \varpi \cdot \hat{i}_\beta \\ \frac{d\hat{i}_\beta}{dt} &= (i_\beta - \hat{i}_\beta)\cdot k_f + \varpi \cdot \hat{i}_\alpha \end{aligned}\right\} \qquad \textbf{Eq. 4}$$

where $\hat{i}_\alpha$ and $\hat{i}_\beta$ represents the components of the filtered vector, i_α and i_β are the components of the input vector, kf is the filter constant, and ω is the centre frequency for the filter. These equations could be represented in a vectorial form, in the frequency domain (Eq. 2)

$$\begin{bmatrix} \hat{i}_\alpha \\ \hat{i}_\beta \end{bmatrix} = \frac{kf}{(s+kf)^2+\omega^2}\begin{bmatrix} s+kf & -\omega \\ \omega & s+kf \end{bmatrix}\cdot\begin{bmatrix} i_\alpha \\ i_\beta \end{bmatrix} \qquad \textbf{Eq. 5}$$

It should be noted that id, iq must be orthogonal. Figure 4 shows the Bode diagram for one output from both inputs, for four values of kf. In all cases the gain is unity for the fundamental frequency, an the phase is 0 and 90º respectively. The values of kf are 0.5, 0.1, 0.05 and 0.01, being 1Hz the fundamental frequency. An increasing value of kf decreases the effectiveness of the filter, but makes it, on the other hand, have a better dynamic response. But the better feature of this filter is that it has no delay from the source, once the permanent state is reached. Figure 5 shows the dynamic response of the filter, tracking a signal with high and low frequency harmonics, and a fundamental frequency of 50Hz.

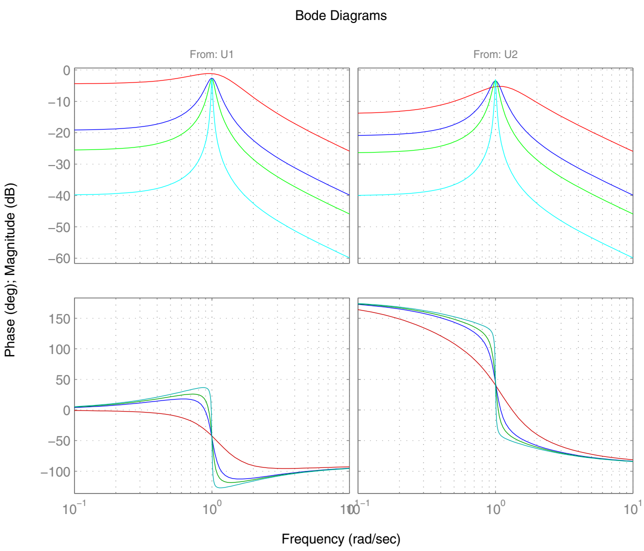


Fig. 4: Bode Diagrams for one output

Also, the central frequency could be self-tuned, using an iterative process, being ω_{est} proportional to the modulo of vectorial product of the vector filtered and the input vector, as shown in Eq. 6. That feature gives its name to this filter (self-tuned), although is not used in the present application, as the frequency is a known constant (50Hz).

$$\omega_{est} = (Kp + \frac{Ki}{s})\cdot\left|\vec{\hat{i}} \times \vec{i}\right| \qquad \textbf{Eq. 6}$$

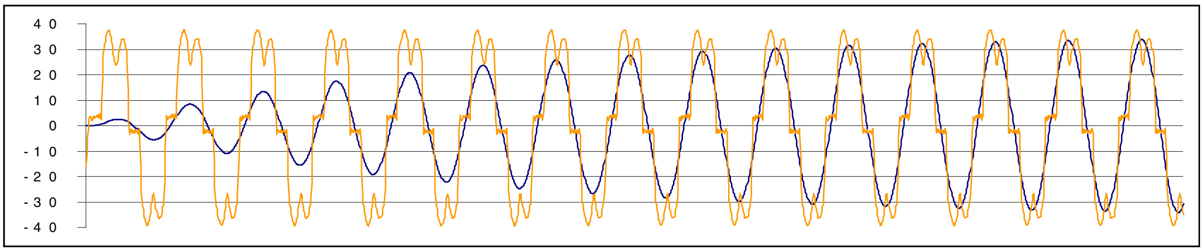


Fig. 5: Simulation results for dynamic response. ω=50Hz, kf=10

IV. Active Filter using a STVF.

Now, we present an active filter algorithm, based on a sinusoidal current source model, as defined in II, and using a filter to extract the fundamental components of both voltage and currents.

The proposed control scheme is shown in Figure 6. The currents and voltages, once measured, are passed to α-β co-ordinates using Park transformation. This components are filtered using STVF, calculating the main (50Hz) component of voltage and current. The current vector is then projected over the voltage, producing a vector in the same direction of V, and with modulo I·cos(φ), the active component of current. This vector is subtracted from the measured current to generate the current necessary to compensate harmonics and reactive power. Lately, it is added a component, in phase with voltage, thus active, which have to compensate variations on DC-Link voltage.

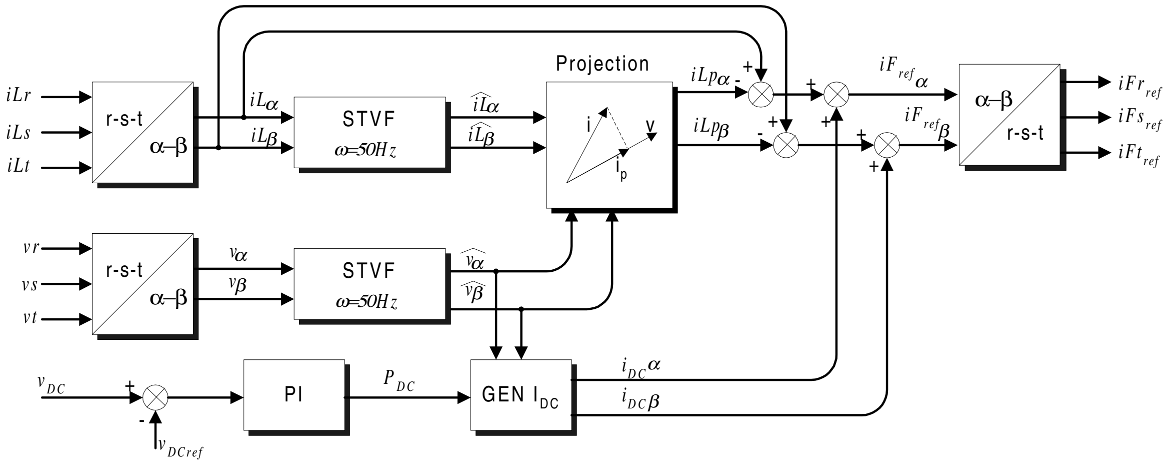


Fig. 6: Control Scheme of the Active Filter implemented

The equations for the projection and calculation of i_{DC} are shown in Eq.4. In order to implement these equations digitally, in a microprocessor, is important to notice that the term Kv is constant, as long as the line voltage is not varying too much in amplitude.

$$\begin{bmatrix} ip_\alpha \\ ip_\beta \end{bmatrix} = Kv\cdot(\hat{v}_\alpha\cdot\hat{i}_\alpha + \hat{v}_\beta\cdot\hat{i}_\beta)\cdot\begin{bmatrix} \hat{v}_\alpha \\ \hat{v}_\beta \end{bmatrix} \quad ; \quad \begin{bmatrix} i_{DC\alpha} \\ i_{DC\beta} \end{bmatrix} = Kv\cdot P_{DC}\cdot\begin{bmatrix} \hat{v}_\alpha \\ \hat{v}_\beta \end{bmatrix} \quad ; \quad Kv = \frac{1}{\hat{v}_\alpha^{\ 2} + \hat{v}_\beta^{\ 2}} \qquad \textbf{Eq. 4}$$

A model of the Active Filter has been developed using Simulink, in order to prove the validity of the method. In this model, the inverter was linearized, and the reference, once generated, was tracked using a Bang-Bang method.

In Figure 7 it is represented the load current of one phase, and source current for the same phase, when a classic, based on the p-q theory [1] method is used. It shows a poor compensation, mostly debt to he harmonics in voltage source. In Figure 8 we represent the voltage source used for simulations. The high frequency current ripple is debt to the Bang-Bang method chosen.

Then in Fig. 9 is displayed the same results, but using this new method. It should be noted the reduction of low order harmonics, due to the better response of this filter with distorted voltages.

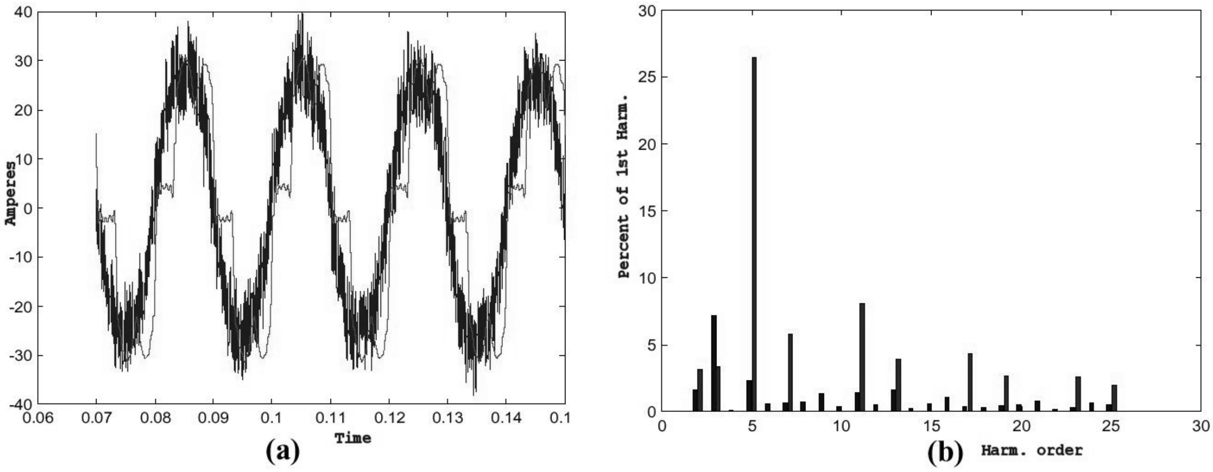


Fig. 7: Current waveform compensated using p-q method. (a): waveforms of load current and compensated current . (b) low order harmonics of load and compensated current

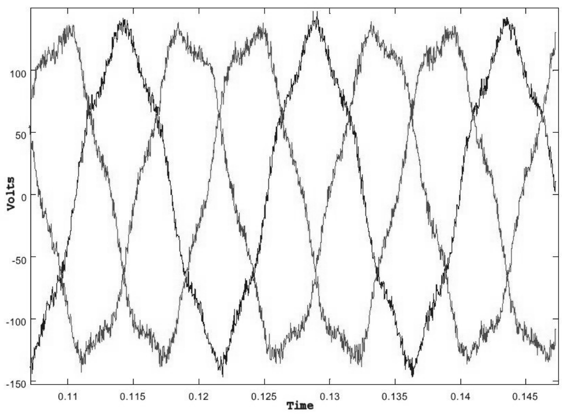


Fig. 8: Voltage waveform used in simulations

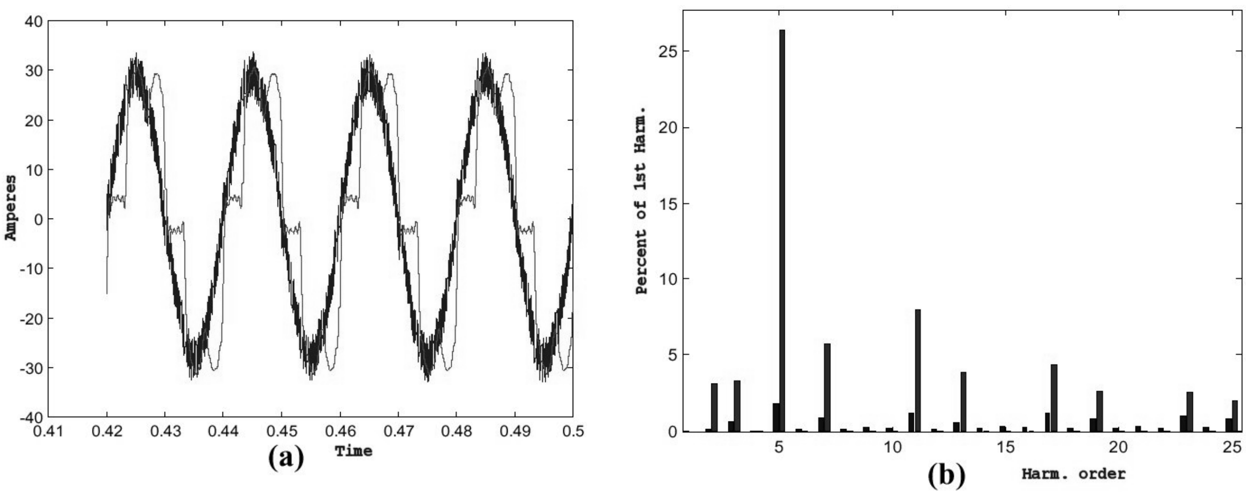


Fig 9: Current waveform compensated using STVF method. (a): waveforms of load current and compensated current . (b) low order harmonics of load and compensated current

Finally, in Table I, the most important simulation results are presented.

Table I. Simulation Results

Waveform	THD (<30Harm.)
Line Voltage	4.5% (5th harm.)
Current uncompensated	29.5%
Current compensated (p-q)	7.92%
Current compensated (STVF)	**3.22%**

V. Experimental Results

The new method was implemented on an experimental system, consisting in an Active Filter, commanded by a DSP-based board, a three phase full bridge IGBT inverter, DC-Link with 3200µF and 2.4mH inductors, connected to the net through a 320/220 Y-Y transformer. The load was a three phase 20kW thyristor rectifier. For a more detailed description, see [5]. In Figure 5 the main system is described.

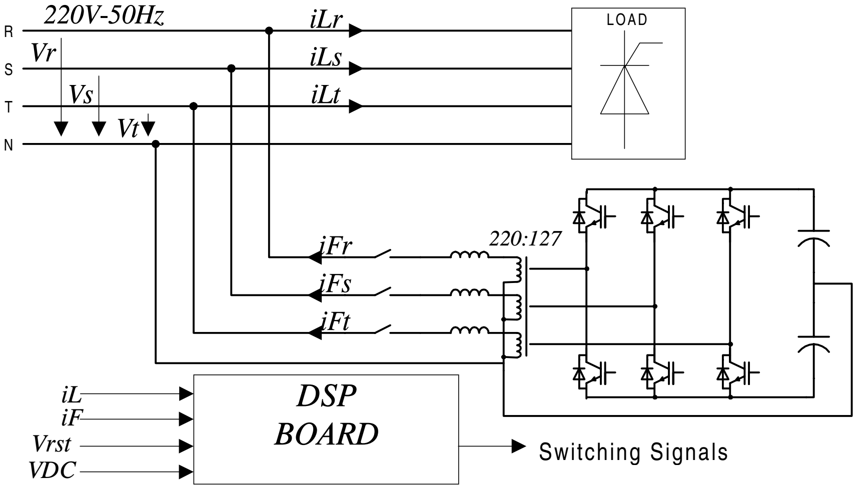


Fig. 10: Main system configuration

First, in Fig. 11 we present the results of the filter applied to voltage signal. In the figure, voltage vector, in α-β co-ordinates is represented, for the non-filtered and filtered signals. Data are collected from the DSP itself that will perform the filtering algorithm, so they include all the noise effects debt to commutations, A/D conversion, among with the 'real' voltage noise and harmonics.

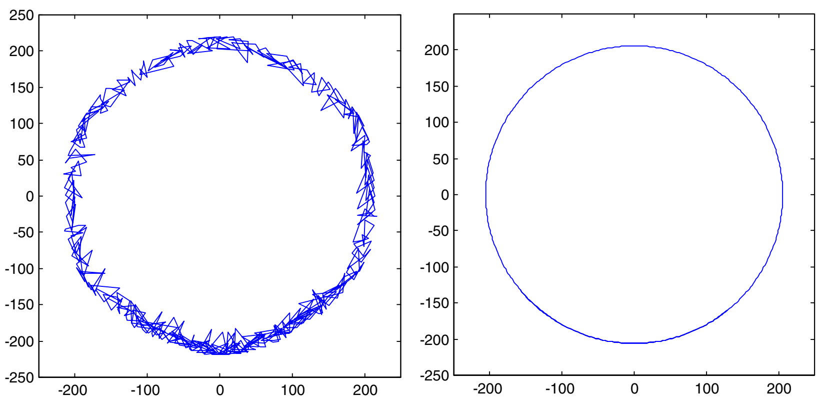


Fig. 11: Polar representation of line voltage and filtered line voltage.

It should be noted also the low order harmonics present in voltage waveform, mainly 5th and 11th, that are also the greater harmonics in load current waveform.

In Figure 12, the spectrum of the load current is displayed, acquired using a digital oscilloscope. It is noticeable the magnitude of the 5th harmonic, of almost 5A of amplitude, 28% of the fundamental harmonic. Next, in figures 13 and 14, compensation results are shown, using a classic p-q method and the new STVF method. In both cases, the high frequency ripple is important, because of the modulation method used (bang-bang). This method was chosen because of its simplicity, and affects equally to both methods of generating the reference, hence have no influence in the comparison of both methods.

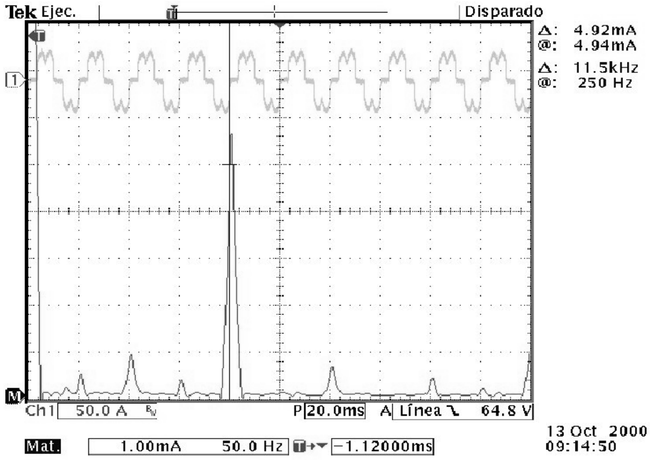
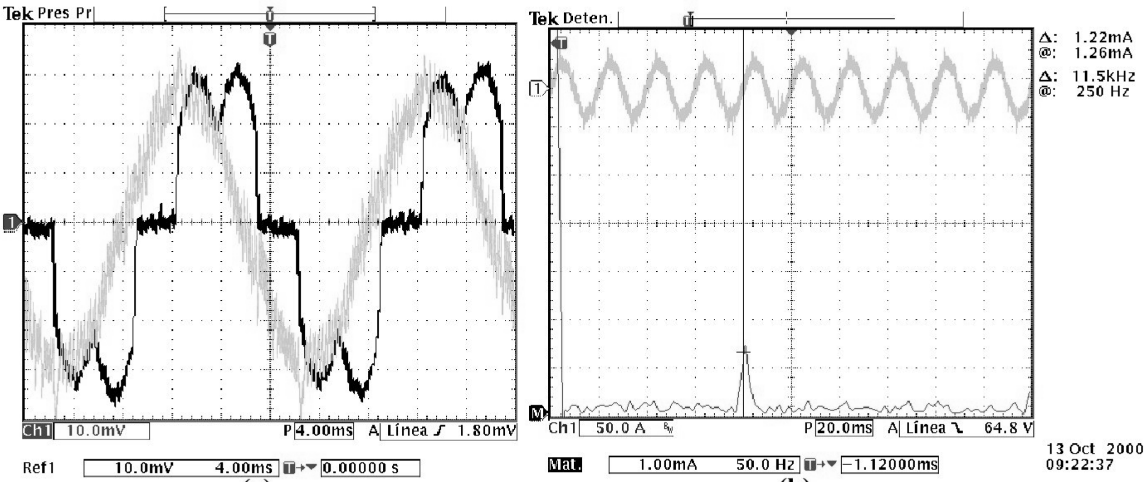
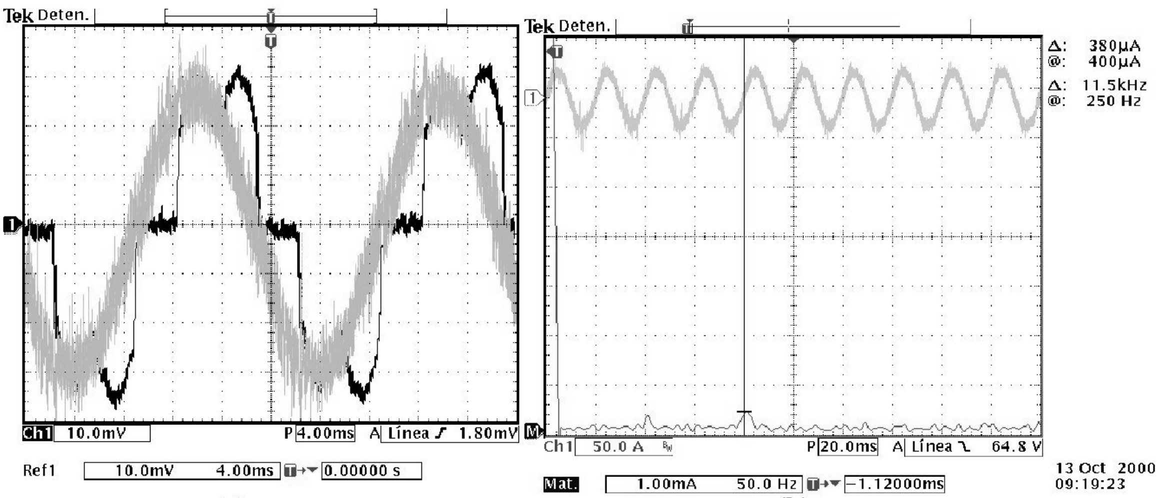


Fig. 12: FFT of the load current.



(a) (b)

Fig. 13: Compensated current Using p-q Method. (a): compensated and load current (10 A/div, 4ms/div); (b): FFT of the compensated Current. 5th harmonic is marked (1.26 A)



(a) (b)

Fig. 14: Compensated current Using new STVF Method. (a): compensated and load current (10 A/div, 4ms/div); (b): FFT of the compensated Current. 5th harmonic is marked (0.4A)

Finally, in Table II, experimental results are summarised. This results conforms with those obtained in simulations, proving the validity of the simulations and of the compensating method itself.

Table II. Experimental Results

Source Current	Cos(φ)	THD
Uncompensated	0.77	30.51 %
Compensated (p-q method)	0.98	12.4%
Compensated (STVF method)	0.99	4.85%

Comparing experimental results, is noticeable the reduction achieved, mostly on low order harmonics present in voltage waveform, with this new method. This is debt mainly to the fact of using filtered voltages, what immunises the system to the voltage harmonics.

VI. Conclusions

A novel method for calculating the reference for a Shunt Active Filter is presented, based on Self-tuned Vector Filtration. Simulations confirms the validity of the system, and its immunity from voltage harmonics. Experimental results definitely confirms expected targets.

A new modulation technique is being developed also, to avoid the high frequency harmonics generated by the inverter, based on a new pulse-width modulation technique.

In future works, the ability of self-tuning of the filter, not used in this paper, will be studied, trying to track the main frequency. This frequency, although fluctuate so little around 50 Hz, could be determinant for obtaining a perfect tracking of the signals, mostly on highly distorted systems, as wind turbine generator fields, or in isolated systems.

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