Predictive Middle Point Modulation: A New Modulation Method for Parallel Active Filters

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Active filters , Modulation strategies, Power factor correction, Power Quality

Abstract

When designing an Active filter, using a current-controlled voltage source inverter, there are two main tasks : to generate an appropiate reference, for nulling harmonic current and reactive power, ant, on the other side, to generate a switching pattern that permits to follow the reference as close as it can be done. Predictive Middle Point Modulation (PMPM) is a novel modulation technique, specially suited for Active Filters. This method is derived from predictive Dead-Beat controller, improving its dynamic response and current error. Simulation results confirm the validity of the proposed method. Experimental results will be provided in the final paper.

I. Introduction

PWM methods have became very popular on current-controlled voltage source inverters (CC_VSI). Many open-loop and closed-loop methods have been described in literature [1], [2], [3], [4]..., focused mainly on motor control, or generically control of electronic loads. When the 'load' is the public grid, special considerations had to be taken into account, as the problem is quite different:

- Stating that voltage on public grid is rigid, the CC-VSI acts purely as a current source.
- it is very easy to develop a simplified model of the entire system, and then use a predictive control method.
- When using a Space Vector method, vectors 0 and 7 (zero voltage vectors) are not interchangeable, and should be considered as the other vectors.

Also, depending on the application and topology of the inverter, a particular modulation method could be better or worse. Dealing with four wired – three legged systems, the use of Space Vector modulation instead of classic pulse width modulation is not justified in terms of simplifying algorithms, as it is necessary to use some 3D vector modulation, not a simpler 2D vector modulation. In addition, in four-wired systems each leg can be controlled separately, converting the control problem in controlling three 1-phase inverters

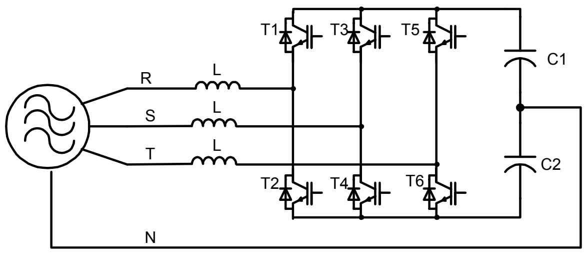


Fig. 1: 3-phase 4-wire inverter

This paper proposes a new method for calculating commutation time for each leg of a three phase, four wire inverter, commonly used in Shunt Active Filters, as shown in Figure 1.

First, in section II, we present an overview of the shunt active filter case in study, in order to state some considerations about the modulation method desired. Then, in section III we present the simplified model of the inverter connected as a shunt active filter, and based on that model, we present the basic equations for the new modulation strategy. In section IV we present some simulation results and comparison with other methods. In section V we show the experimental prototype, under development by now, in which the algorithm will be test soon. Finally, in section VI we present some conclusions about modulation strategies for active filters.

II. CC-VSI used as Shunt Active Filter

When using an CC-VSI to build a shunt active filter, there are some considerations that should be taken into account, related to modulation method. The main characteristics that make this necessary are:

- The shape of the reference current to be followed is not sinusoidal at all. It has, indeed, all the harmonics of the load, and uses to have very sharpen slopes.
- The load, as commented before, is the public grid, which is mostly a voltage source and a little series impedance.
- It is very important to track the reference as quick as it can be possible, because the compensation effectiveness depends on it.
- The effectiveness of compensation lays on the two sides -the generation of the reference and the tracking of this reference- of the problem. Therefore a balanced solution must be provided between a perfect –but slow- reference generator and a quick tracking method that makes impossible to calculate the appropriate reference.

The entire system can be represented as in figure 2. In this picture we can see the public grid, that will be considered as a non-ideal voltage source, with its series impedance; also, the load will be a thyristor rectifier, with a high 5^{th}, 7^{th} and 11^{th} harmonic, and low power factor (around 0.87); The filter itself will be implemented with the CC-VSI, divided into reference generator and reference tracking.

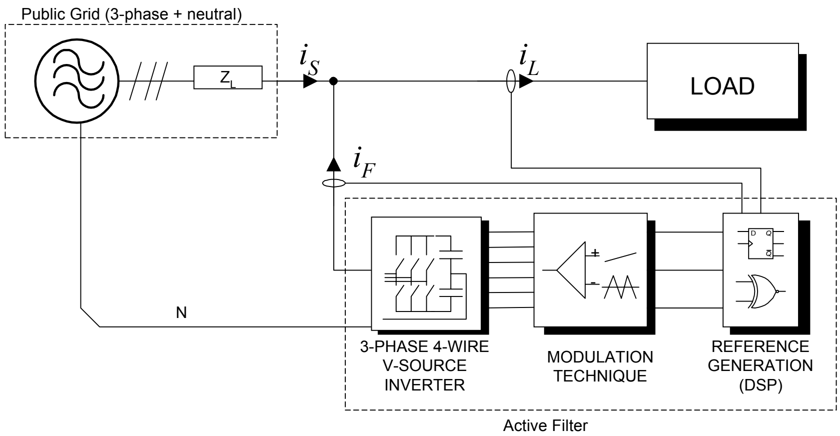


Fig. 2: General block diagram

The performance of a modulation technique can be measured in so many different ways [3], and depending on the application designed, some criteria had to be considered more important than others. For the Active Filter (AF) application, the most important criterion could be a fast response, because the shape of the reference to be followed is very sharp in certain regions and very flat in other regions. Also, it is very important to reduce high frequency harmonics generated by the inverter, as the AF is designed to reduce THD of a load.

The problem is very difficult to afford, because the reference generator is no totally decoupled from the tracking method, and there are many effects that had to be taken into account, as the effect of dead times, the effect of other systems interfering with our filter, or the public grid impedance, among others. Therefore, an approximation should be convenient, at least as a starting point.

III. Predictive Middle Point Modulation

Approximated model

In order to evaluate the performance of different modulation methods, a Simulink model of the entire system was built. The following considerations have been made:

- Source (net) impedance is much less than the series inductance of the Active Filter.
- Capacitor voltages stay constant on a sample time
- Source voltages also remains constant on a sample time
- Each phase is decoupled from the other two, and its current depends only on the position of its switches and on the voltage of the net on its phase.

Really, the fourth consideration could be derived from the second one, as the DC-LINK is the only point of connection between phases. Using this simplifications, the system was linearized per phase using the basic scheme shown in Figure 2. For this scheme, the main equations that gives current on phase r, depending on the position of switches T1 and T2 are stated in Eq.1

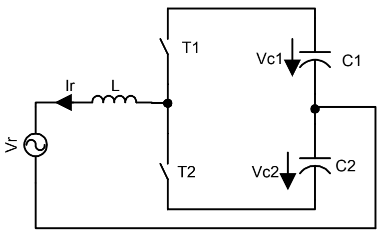


Fig. 3: Simplified System Model

$$
\begin{aligned}
T_1 on, T_2 off: &\quad L \cdot \frac{di_r}{dt} = Vc_1 - Vr \\
T_1 off, T_2 on: &\quad L \cdot \frac{di_r}{dt} = -(Vc_2 + Vr)
\end{aligned}
\qquad \textbf{Eq 1.}
$$

where Vr is the line voltage(phase to neutral) , nulling the effect of source impedance, i_r is phase current of the filter, and V_{C1} and V_{C2} are voltages across the upside and downside capacitors.

Modulation Strategy

There are many methods to modulate a signal using Pulse Width Modulation techniques. A triangular waveform is commonly used, which is compared with the signal to be modulated, and the comparison decides what switch has to be turned on. The present method is a variation of the classical suboscilation method known as predictive Dead-Beat controller [5]. In that method, the inverter voltage is chosen in order to null current error at the end of the sample period. That method presents some disadvantages for using in Active Filters:

- The reference is reached at the end of the period, so the delay is, at least, one sample time.
- The error is not distributed symmetrically, so the integral error is not zero.

Now, we will try to minimise two particular performance criteria, specially important for active filters, as stated before:

- The current error, measured as the integral of the difference between current reference and actual current.
- The delay between reference and current, measured as the time in which the current equals the reference.

From this point, we calculate commutation time, using linearized equations, in order to minimise both delay time and current integral error. The target is to get an average current, during a sample time, equal to the reference we are tracking.

In Figure 3 is shown the evolution of current on a sample time, in a typical triangular PWM suboscillation method.

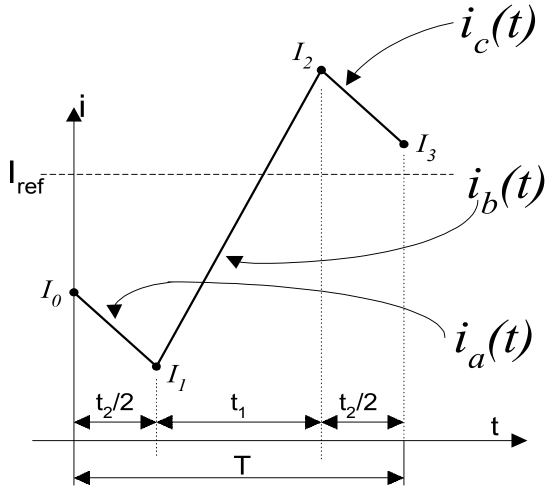


Fig. 4: Current evolution

where:
t_1, t_2 : periods when the upper and lower switch are connected, respectively.
T: Commutation period.
i_a , i_b , i_c: waveforms of the current in the three intervals of a commutation period, (not having in count dead times)
I_0: Initial current, measured at the beginning of a commutation period.
I_1 ,I_2 ,I_3: Final currents at the end of each interval.

The average current in the interval could be calculated as follows :

$$T \cdot I_{AV} = \int_0^T i \cdot dt = \int_0^{t_2/2} i_a \cdot dt + \int_0^{t_1} i_b \cdot dt + \int_0^{t_2/2} i_c \cdot dt$$

$$\begin{aligned} & i_a(t) = I_0 - \tfrac{1}{L}(Vc_2 + Vr) \cdot t; \\ & i_b(t) = I_1 + \tfrac{1}{L}(Vc_1 - Vr) \cdot t \,;\, I_1 = I_0 - \tfrac{1}{L} \cdot (Vc_2 + Vr) \cdot \tfrac{t_2}{2}; \\ & i_c(t) = I_2 - \tfrac{1}{L}(Vc_2 + Vr) \cdot t \,;\, I_2 = I_1 + \tfrac{1}{L} \cdot (Vc_1 - Vr) \cdot t_1 \end{aligned} \quad \textbf{Eq. 2}$$

where t_1 and t_2 are the time where switches T1 and T2 are turned on, respectively.
Let us consider that voltage over capacitors are balanced, so VC1 are equal to VC2. Then, average current can be calculated as:

$$I_{AV} = I_0 + \frac{1}{2 \cdot L} \cdot \left[T \cdot (Vc - Vr) - 2 \cdot t_2 \cdot Vc \right] \quad \textbf{Eq. 3}$$

Finally, if we want I_{AV} to be equal to the current reference, being I_0 actual (measured) current, then t_2 can be calculated as:

$$t_2 = \frac{L}{Vc} \cdot (I_0 - I_{AV}) + \frac{(Vc - Vr) \cdot T}{2 \cdot Vc} \quad \textbf{Eq. 4}$$

There are some advantages directly derived from the use of this modulation:

- The average current over the sample time is equal to the reference calculated.
- The reference, itself, is reached at the middle of the sampling time. That feature gives the name of the method (Predictive Middle Point Modulation, PMPM).

$$\varepsilon I = \int_0^T (i - i_{ref}) \cdot dt \quad \textbf{Eq. 5}$$

- the integral error, defined in Eq. 5, is null, supposing all magnitudes fixed over one commutation period, because is null on each period.
- It is very fast and simple to calculate t_2. In fact, we calculate $t_2 \cdot Vc$, and then compare with $T \cdot Vc$ to generate pulses, in order to avoid division in real time calculations.

IV. Simulation Results

Many simulations were performed with the linearized model of the shunt AF, in order to prove the validity of the method. Basically, a Bang-Bang method and the new PWM method are to be compared. The election of the Bang-Bang method as a reference for comparison is motivated by its simplicity and its relatively good results. It can be demonstrated that Bang-Bang method is the faster

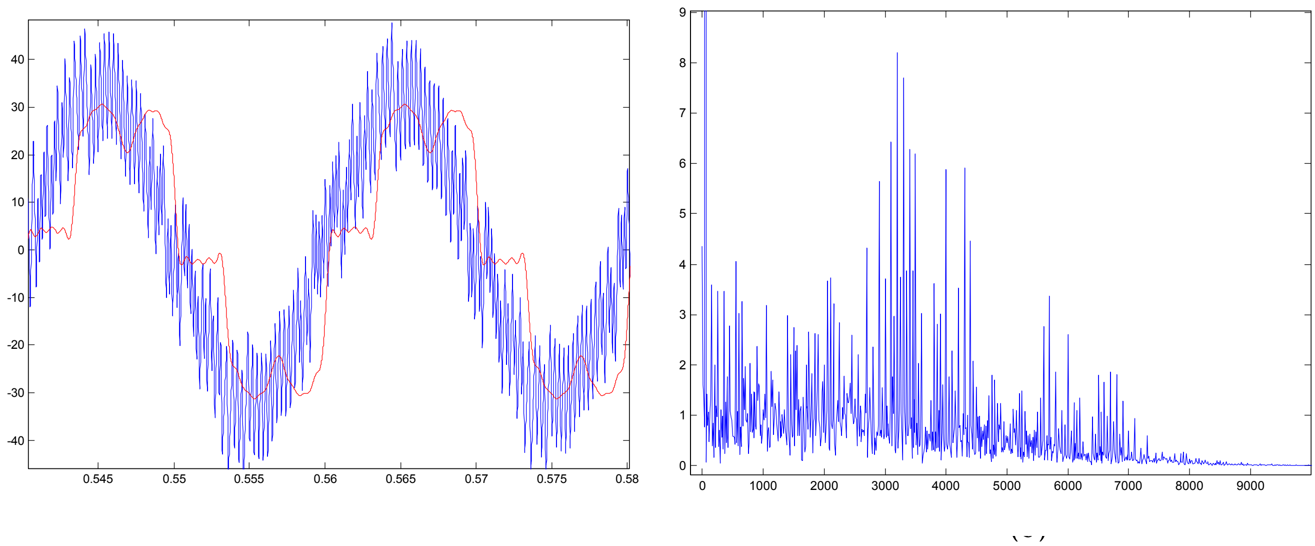


Fig 5: Simulation results for Bang-Bang at 10kHz. (a) Current waveforms of load and source side, (x axis in seconds, y-axis in amperes); (b) Source Current Spectrum. (x-axis in Hz, y-axis in percent of 1st harmonic)

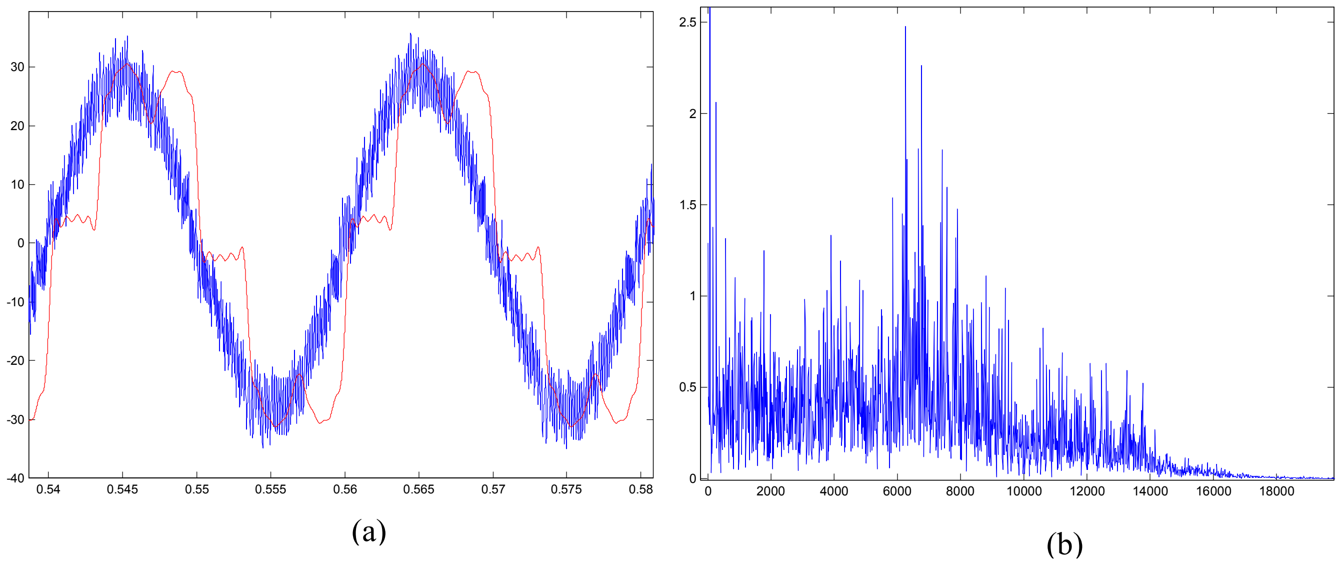
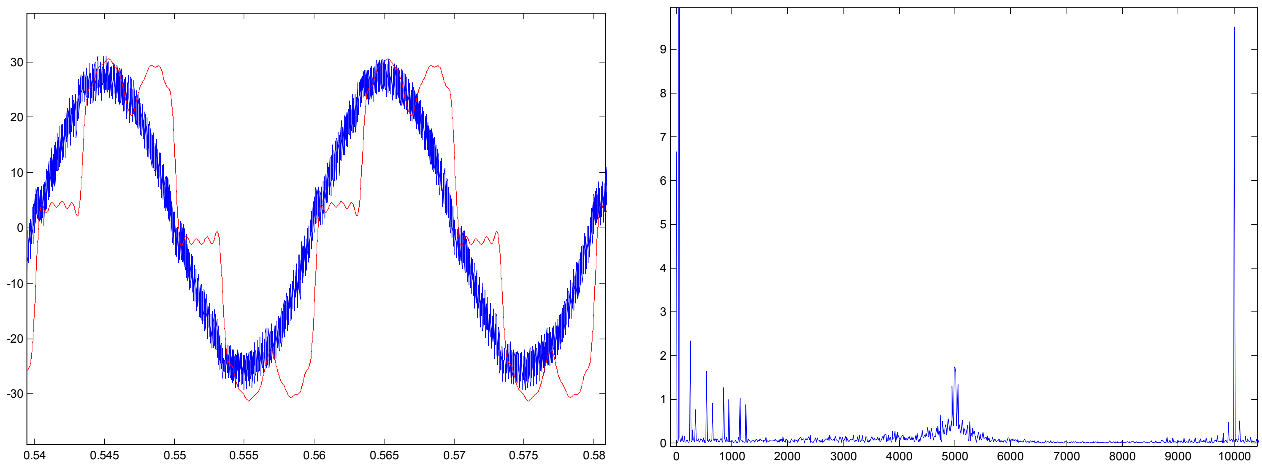


Fig. 6: Simulation results for Bang-Bang at 20kHz. (a) Current waveforms of load and source side, (x axis in seconds, y-axis in amperes); (b) Source Current Spectrum. (x-axis in Hz, y-axis in percent of 1st harmonic)

method, in term of calculation and one of the better in terms of controllability. Next, we present simulation results of an active filter applied to a thyristor rectifier, characterised using actual measures of the current.

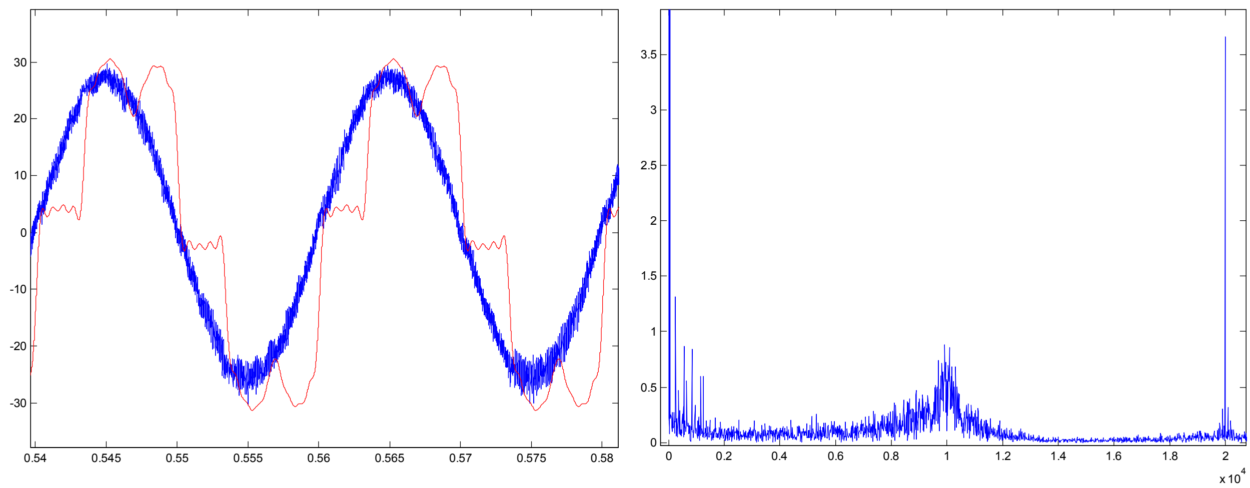
Figure 5 and 6 presents results of simulation using Bang-Bang method , at 10kHz and 20kHz of commutating frequency. Waveforms of load (uncompensated) and source (compensated) currents, and the FFT of the compensated current are shown:

Figure 7 and 8 shows the same curves, using PMPM method. It can be appreciate that ripple decreases dramatically, and Spectrum is concentrated on the commutation frequency.



(a)

Fig. 7: Simulation results for PMPM at 10kHz. (a) Current waveforms of load and source side, (x axis in seconds, y-axis in amperes); (b) Source Current Spectrum. (x-axis in Hz, y-axis in percent of 1st harmonic)



(a) (b)

Fig. 8: Simulation results for PMPM at 20kHz. (a) Current waveforms of load and source side, (x axis in seconds, y-axis in amperes); (b) Source Current Spectrum. (x-axis in Hz, y-axis in percent of 1st harmonic)

It should be noted that not all the spectrum graphics have the same scale, in order to gain precision, although this could leads to confusion.

In Table I, the main results are shown. the THD was calculated only with the first 30 harmonics, form 100Hz to 1500Hz, because the interest of an active filter is focused on low order harmonics.

Table I: Simulation Results

Method	THD (<1500 Hz)	5th Harm.
Bang Bang at 10kHz	11.96 %	3.4%
Bang Bang at 20kHz	3.96%	2.3%
PMPM at 10kHz	3.7%	2.07%
PMPM at 20kHz	2.25%	1.32%

It should be noted that the THD is always greater in Bang-Bang than in PMPM, even doubling the frequency. The 5th harmonic is shown as it is the greatest harmonic in the load, and the smaller it remains in the compensated current, the better is the method for tracking the reference.

V. Experimental Model

An experimental model is being constructed to finally validate the simulation results obtained. It is based on a previous work of the authors [6], and basically consist on a three-phase IGBT voltage Source Inverter, connected though a 380/220 autotransformer. The DSP board, in this case, was composed by two DSP-based boards. The first of them, based on the TMS320C31 from Texas Instruments, is used to calculate an appropriate reference, in real time (20kHz). The second one is based on the TMS320F243, form Texas Instruments, which is a fixed-point DSP with PWM modulators built in. In Figure 9 a generic overview of the installation is shown, and in Figure 10 a picture of the digital system is shown, marking the more important parts of it.

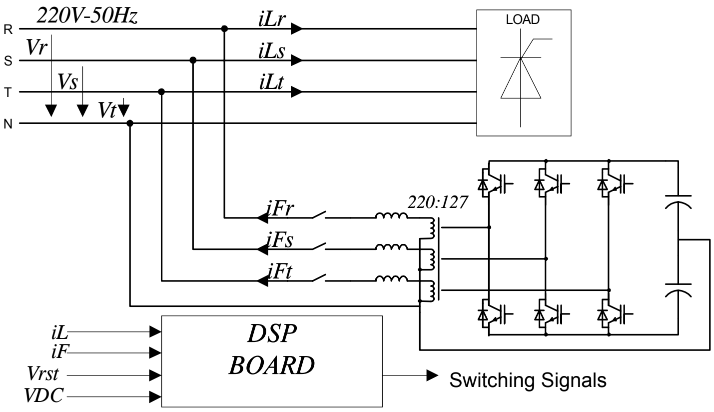


Fig. 9: Experimental model

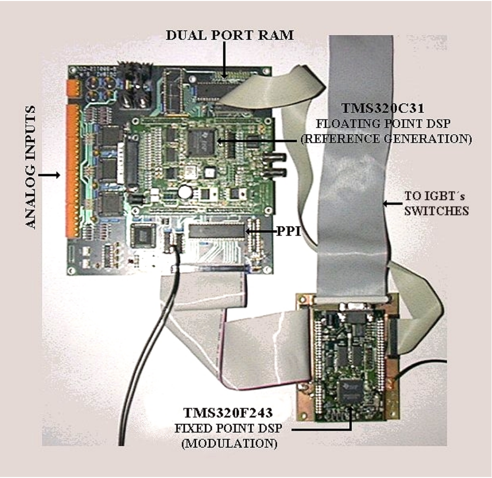


Fig. 10: Digital Control Boards

VI. Conclusions

When dealing with non sinusoidal waveforms, specially with locally high slopes, not all the modulation methods shows good results. In the particular case of active filters, an extra care have to be taken in choosing the modulation method, as it contributes dramatically to the success or not of the filtration.

The main characteristics of a modulation method to be selected, in this area, will be a fast response and a minimal error in tracking a reference, even although the electrical efficiency will be less than in other methods.

A new method for modulation, specially suited for Shunt Active filters but useful in any grid-connected application, is presented. Simulation results confirms theoretical conclusions, showing a very good and fast response.

An experimental model is being finished, and soon we will be able to present experimental results, to confirm the validity of the modulation method presented.

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