MASTER

Digital beacon receiver for propagation experiments: optimization and testing of a Digital Frequency-Locked Loop/Phase-Locked Loop

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Digital Beacon Receiver
for propagation experiments

Optimization and testing of a Digital Frequency-Locked Loop/
Phase-Locked Loop

by D.J. Aalberts

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Summary

Propagation experiments at Eindhoven University are carried out using beacon signals of satellites. The equipment used to receive the beacon signals consists of an analog PLL and a digital detector, based on a Z80 microprocessor. The processing power of this processor is poor. Therefore the telecommunications division started the development of a digital receiver, based on a Digital Signal Processor (DSP). The receiver consists of a digital Frequency-Locked Loop / Phase-Locked Loop and a Decimator/Detector.

The digital Frequency-Locked Loop / Phase-Locked Loop (FLL/PLL) was already built in a previous project but this design was reconsidered. The hardware is designed using a Texas Instruments TMS 320C25 DSP. The digital FLL is used to realize that the incoming analog signals are mixed to a fixed intermediate frequency. At this intermediate frequency, subsampling of the signal is performed and the All Digital PLL is used to generate coherent in-phase and quadrature components derived from of the received harmonic beacon signals. This is necessary to perform coherent detection of the received signals.

Unfortunately, the first prototype receiver did not function very well. The main problems encountered in the software of the FLL/PLL.

The purpose of this study for graduation is to optimize and measure the characteristics of the FLL/PLL. All parts of the software were studied and tested. The general setup of the software proved to be good. However, several bugs were found and debugged in the implementation.

For characterization of the FLL/PLL several measurements have been executed. The FLL is tested and proved to be able to track a beacon signal of the "ITALSAT" satellite with a C/N ratio at the input of the FLL/PLL of at least -16.5 dB. The PLL is tested and will lock on a received beacon signal with a C/N ratio at the input of at least -14.5 dB. These figures compare very good with respect to measured figures of analog Phase-Locked Loops of proven design.

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Symbols

\( \Delta f \)  
bin size  
[Hz]

\( \varepsilon_{ss} \)  
Steady state error  
-

\( \theta \)  
angle between 0° and 90°  
[degrees]

\( \theta_{\text{co}-x} \)  
angle between \( x_{\text{co}} \) and \( x_x \)  
[degrees]

\( \theta_0 \)  
detection angle  
[degrees]

\( \xi \)  
damping ratio of the PLL-filter  
-

\( \phi \)  
Angle  
[degrees]

\( \phi_d \)  
phase angle difference  
[rad]

\( \phi_i \)  
Input phase angle  
[rad]

\( \phi_o \)  
Output phase angle  
[rad]

\( \omega \)  
Angular frequency  
[rad/s]

\( \Omega \)  
Harmonic angular frequency  
[rad/s]

\( \omega_0 \)  
Angular centre frequency at the output  
[rad/s]

\( \omega_i \)  
Angular centre frequency at the input  
[rad/s]

\( \omega_n \)  
Natural angular frequency of the PLL loopfilter  
[rad/s]

\( A_{\text{co}}, A_x \)  
co-polar amplitude, cross-polar amplitude  
-

\( B_{00}, B_{11}, B_2 \)  
Beacon Signals of 12.5, 20 and 30 GHz  
-

\( B_{W} \)  
Bandwidth  
[Hz]

\( C_0 \)  
DVC0 Constant (free running angular frequency)  
-

\( C_1 \)  
Constant used in the ADPLL  
-

\( C_2 \)  
Constant used in the ADPLL  
[rad²/s⁴]

\( f \)  
Oscillator frequency  
[Hz]

\( f_s \)  
Sample frequency  
[Hz]

\( G \)  
Gain  
-

\( h(n) \)  
discrete impulse response  
-

\( H_{CL} \)  
Closed Loop Transfer function  
-

\( H_{OL} \)  
Open Loop Transfer function  
-

\( I_{\text{co}}, I_x \)  
in phase, co-polar and x-polar  
-

\( k \)  
Sample counter  
-

\( M \)  
\(^2\log N\)  
-

\( N \)  
Number of FFT inputs  
-

\( n_{\text{co}}, n_x \)  
Noise amplitude  
-

\( Q_{\text{co}}, I_{\text{co}} \)  
Quadrature, co-polar and x-polar  
-

\( s \)  
Laplace Transform variable  
-

\( t \)  
Time  
[s]

\( T \)  
time between samples  
[s]

\( t_i \)  
Clock signal low-time  
[sec]
<table>
<thead>
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<th>Symbol</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>$W_N$</td>
<td>Twiddle Factor</td>
</tr>
<tr>
<td>$X(\omega)$</td>
<td>Fourier transformed input signal</td>
</tr>
<tr>
<td>$X(k),X_1(k),X_{11}(k),X_{12}(k)$</td>
<td>Discrete Fourier transform of $x(t)$</td>
</tr>
<tr>
<td>$x(nt)$</td>
<td>Sampled signal</td>
</tr>
<tr>
<td>$x(t)$</td>
<td>Analog signal</td>
</tr>
<tr>
<td>$x_{co}, x_x$</td>
<td>Co-polar input signal</td>
</tr>
<tr>
<td>$X_i(k)$</td>
<td>Delayed input signal</td>
</tr>
<tr>
<td>$x_i(t)$</td>
<td>Input signal</td>
</tr>
<tr>
<td>$x_o(t)$</td>
<td>Output signal</td>
</tr>
<tr>
<td>$Y_i(k)$</td>
<td>Hilbert transformed input signal</td>
</tr>
<tr>
<td>$z$</td>
<td>$z$-transform variable</td>
</tr>
</tbody>
</table>
Abbreviations

<table>
<thead>
<tr>
<th>Abbreviation</th>
<th>Description</th>
</tr>
</thead>
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<tr>
<td>A/D, D/A</td>
<td>Analog/Digital, Digital/Analog</td>
</tr>
<tr>
<td>BW</td>
<td>Bandwidth</td>
</tr>
<tr>
<td>C/N</td>
<td>Carrier to Noise ratio</td>
</tr>
<tr>
<td>CL</td>
<td>Closed Loop</td>
</tr>
<tr>
<td>Co,X</td>
<td>Co-polar, Cross-polar</td>
</tr>
<tr>
<td>CRC</td>
<td>Cyclic Redundancy Check</td>
</tr>
<tr>
<td>DC</td>
<td>Direct Current</td>
</tr>
<tr>
<td>DFT</td>
<td>Discrete Fourier Transform</td>
</tr>
<tr>
<td>DPLL, ADPLL</td>
<td>(All) Digital Phase-Locked Loop</td>
</tr>
<tr>
<td>DSP</td>
<td>Digital Signal Processor</td>
</tr>
<tr>
<td>DUT</td>
<td>Delft University of Technology</td>
</tr>
<tr>
<td>DVCO</td>
<td>Digital Voltage Controlled Oscillator</td>
</tr>
<tr>
<td>EOT</td>
<td>End Of Transmission</td>
</tr>
<tr>
<td>EEPROM</td>
<td>Electrically Erasable Programmable ROM</td>
</tr>
<tr>
<td>EUT</td>
<td>Eindhoven University of Technology</td>
</tr>
<tr>
<td>FFT</td>
<td>Fast Fourier Transform</td>
</tr>
<tr>
<td>FIFO</td>
<td>First In, First Out</td>
</tr>
<tr>
<td>FLL</td>
<td>Frequency-Locked Loop</td>
</tr>
<tr>
<td>I/Q</td>
<td>In Phase/Quadrature</td>
</tr>
<tr>
<td>IC</td>
<td>Integrated Circuit</td>
</tr>
<tr>
<td>IF</td>
<td>Intermediate Frequency</td>
</tr>
<tr>
<td>I/O</td>
<td>Input Output</td>
</tr>
<tr>
<td>IRQ</td>
<td>Interrupt ReQuest</td>
</tr>
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<td>L-ADPLL</td>
<td>Linear All Digital Phase-Locked Loop</td>
</tr>
<tr>
<td>LEN</td>
<td>Length</td>
</tr>
<tr>
<td>LF</td>
<td>Loop Filter</td>
</tr>
<tr>
<td>LNB</td>
<td>Low Noise Block converter</td>
</tr>
<tr>
<td>LO</td>
<td>Local Oscillator</td>
</tr>
<tr>
<td>OL</td>
<td>Open Loop</td>
</tr>
<tr>
<td>OTS</td>
<td>Orbital Test Satellite</td>
</tr>
<tr>
<td>PC</td>
<td>Personal Computer</td>
</tr>
<tr>
<td>PD</td>
<td>Phase Detector</td>
</tr>
<tr>
<td>PLL</td>
<td>Phase-Locked Loop</td>
</tr>
<tr>
<td>RAM</td>
<td>Random Access Memory</td>
</tr>
<tr>
<td>RMS</td>
<td>Root Mean Square</td>
</tr>
<tr>
<td>ROM</td>
<td>Read Only Memory</td>
</tr>
<tr>
<td>SOH</td>
<td>Start of Header</td>
</tr>
<tr>
<td>VCO</td>
<td>Voltage Controlled Oscillator</td>
</tr>
</tbody>
</table>
Introduction

1.1 Propagation research at Eindhoven University of Technology

Research on radiowave propagation forms an important contribution to the design of telecommunications applications. Eindhoven University of Technology (EUT) and PTT Research started propagation experiments with the Orbital Test Satellite (OTS), which carries a propagation beacon and transponder. Extensive tests were carried out at 11 and 14 GHz. The Olympus satellite, which carries three beacon transmitters operating at 12.5, 20 and 30 GHz, was launched and commissioned in 1989. This provides the opportunity to characterize the atmospheric transmission channel. EUT carries out the complete set of measurements possible with the Olympus beacons. Radiometers at all three frequencies and meteorological instruments are used at the earth station. The comparison of the 12.5 GHz, 20 and 30 GHz results will give insight in the frequency dependence of the propagation effects. These results will be applied to the prediction of signal impairments on satellite earth links [10].

The beacon receiver is divided into five different functional parts as shown in Figure 1.1. The first stage is the antenna which is capable of receiving three beacon signals simultaneously. The Co- and Cross-polar components of these three frequencies are provided at six separate antenna ports.
The Low Noise Block (LNB) downconverts the received signals to 10 MHz. The converted signals are band filtered and amplified. The Phase-Locked Loop (PLL) will maintain the input frequency at a stable 125 kHz by measuring the Phase Difference in a Phase Detector (PD). The signal representing the phase difference controls a Voltage Controlled Oscillator (VCO), which is used for mixing to the 10 MHz signal. With the stable 125 kHz signal the digital detector determines the amplitude of the Co- and Cross-polar signals and phasedifference between them.

**Figure 1.1 Block diagram of the receiver**

Currently, analog receivers and digital detectors are used to process the converted signals. The analog receiver splits the incoming co- and cross-polar signals to in-phase and quadrature components. The digital detector, developed by PTT Research, determines the amplitude and the phase of the incoming satellite signals. The detector was built using a Z80 microprocessor. Unfortunately, this processor does not have enough processing power to process complicated algorithms; consequently, the algorithms have been simplified.

The Telecommunications Division started to design a new detector, using the TMS320C25 Digital Signal Processor (DSP). This processor has more processing power; therefore, and filter characteristics and communication possibilities were improved. In 1991 this resulted in a dual channel digital detector with a serial communications link to a PC [11].

The next step was the design of a digital Phase-Locked Loop (PLL), combined with a frequency tracking facility, indicated as the Frequency-Locked Loop (FLL). The detector is improved by adding a decimator, which reduces the input data. The whole path from receiver to data acquisition is digital in this setup. In this way the main disadvantage of analog receivers, i.e. the non-constant behaviour in time, will be avoided.

The FLL/PLL was implemented using the same DSP. However, a new board was designed for both the FLL/PLL and the Decimator/Detector. The two boards are based on the same architecture. Some differences exist in the number of A/D and D/A converters, communication possibilities etc.
1.2 The problem definition

The software for the FLL/PLL was designed and implemented by Op den Camp [1]. Strik [2] implemented the software for the Decimator/Detector. Together they wrote several parts of the programs. Unfortunately the software for the FLL/PLL did not function very well. Franken [9] has improved the FLL/PLL-software. However, this software still did not function satisfactorily when Franken graduated.

The purpose of this study for graduation is to optimize and characterize the FLL/PLL. All parts of the software were studied and tested. The general design of the software proved to be good. However, several bugs were found. For characterization of the FLL/PLL several measurements have been done.

This report describes the FLL/PLL hard- and software, the improvements made and the measurements done. Chapter 2 describes the functional configuration of the FLL/PLL. In Chapter 3 the FLL/PLL hardware is described. Chapter 4 deals with algorithms used for frequency tracking. In Chapter 5 the PLL is described. The configuration and algorithms used for the communication are described in Chapter 6. In Chapter 7 general configuration problems, programming problems etc. are described. Finally, conclusions and some recommendations are presented in Chapter 8.
The Frequency-Locked Loop and Phase-Locked Loop

The FLL/PLL software developed at the Telecommunications Division of EUT consists of four functional blocks [1] as shown in Figure 2.1:
- The Frequency-Locked Loop, which determines the carrier frequency of the co-polar signal;
- the Phase-Locked Loop, which has the possibility to recover the carrier of the co-polar signal with a poor Signal to Noise ratio;
- the coherent detector, which resolves the in-phase and quadrature components from the co- and crosspolar input signals with the use of the recovered carrier;
- a communications module, which is able to send data to the Decimator/Detector and to the data logger PC.

![Figure 2.1 Block diagram of the EUT digital FLL/PLL satellite receiver](image)

Optimization and testing of a digital FLL/PLL
2.1 The Frequency-Locked Loop

The Frequency-Locked Loop maintains the frequency of the input signal constant by determining the frequency of the carrier, calculating the error between the frequency of the input signal and the frequency of the output signal and correcting the difference in frequency with a Voltage Controlled Oscillator (VCO).

![Figure 2.2 The Frequency Locked Loop](image)

The carrier frequency is found by calculating the discrete frequency spectrum of the input signal. The frequency bin with the highest signal power should be the carrier frequency. Calculating the difference between the desired frequency and the measured frequency will result in an error signal. The error signal controls the Voltage Controlled Oscillator (VCO). The signal generated by the VCO is mixed with the input signal.

Op den Camp [1] has formulated several constraints for the FLL:
- maintain a carrier frequency of 124.516 kHz with the use of a VCO;
- output voltage of -5V to 5V to control VCO;
- possibility to transmit the data that represents the frequency spectrum to the PC;
- possibility to transmit the FLL-status to the Decimator/Detector.

2.2 The Phase-Locked Loop

The function of a Phase-Locked Loop is to recover the carrier from a signal with a poor Carrier/Noise (C/N) ratio. In this application the PLL is implemented using a Digital Signal Processor. The implemented algorithm is the digital representation of the analog version. The algorithm is described in Chapter 4.
Op den Camp [1] has formulated several constraints for the PLL:
- An optimum natural frequency of 17 Hz, which equals a noise bandwidth of 56 Hz. This results in an input Signal to Noise ratio that may reduce to 25 dB/Hz before loss of lock occurs;
- Constant loop bandwidth because an adaptively controlled loop bandwidth results in additional software complexity;
- Possibility to transmit the Lock-Status to the Decimator/Detector.

2.3 Coherent detection of the signals to be measured

Coherent detection is used to determine the in-phase and quadrature component of the copolar and crosspolar signals. Detection is achieved by mixing the input signal with the recovered carrier, which is the VCO output signal, shifted 90° in phase [1].

The Co-polar signal $x_{co}$ is represented by:

$$x_{co}(t) = A_{co} \cos(\omega t + \theta_{osc}(t)) + n_{co}(t) \cos(\omega t) - n_{co}(t) \sin(\omega t)$$

(2.1)

The Cross-polar signal $x_c$ is represented by:

$$x_{c}(t) = [A_x+n_{x,c}(t)]\cos(\omega t + \theta_{co,x}) - n_{x,c}(t)\sin(\omega t + \theta_{co,x})$$

(2.2)
The signal received at the co-polar input is given in (2.1). Assuming that the satellite and ground station local oscillator are ideal, the co-polar input signal is represented by [1]:

\[ x_{co}(t) = [A_{co} + n_{co,c}(t)]\cos(\omega_s t) - n_{co,i}(t)\sin(\omega_s t) \]  

(2.3)

Applying coherent detection to equation 2.3 and removing the high frequency components will result in an in-phase and quadrature-component.

\[ I_{co}(t) = [A_{co} + n_{co,c}(t)]\cos(\omega_s t) - n_{co,i}(t)\sin(\omega_s t) - 2\cos(\omega_s t + \theta_0(t)) \]

(2.4)

\[ Q_{co}(t) = [A_{co} + n_{co,c}(t)]\cos(\omega_s t) - n_{co,i}(t)\sin(\omega_s t) - 2\sin(\omega_s t + \theta_0(t)) \]

In case of ideal detection, \( \theta_0(t) = 0 \), equation 2.4 will reduce to:

\[ I_{co}(t) = A_{co} + n_{co,c}(t) \]

(2.5)

\[ Q_{co}(t) = n_{co,i}(t) \]

Averaging equation 2.5 will result in:

\[ <I_{co}(t)> = A_{co} \]

(2.6)

\[ <Q_{co}(t)> = 0 \]

Demodulation of the co-polar signal results in an in-phase and quadrature component. The in-phase component will be equal to the amplitude of the co-polar input signal.

Assuming the satellite and ground station local oscillator are ideal, the cross-polar signal (2.2) can also be demodulated using coherent detection, which results in:

\[ I_x(t) = [A_x + n_{x,c}(t)]\cos(\omega_s t + \theta_{co-x}) - n_{x,i}(t)\sin(\omega_s t + \theta_{co-x}) \cdot 2\cos(\omega_s t + \theta_0(t)) \]

(2.7)

\[ Q_x(t) = [A_x + n_{x,c}(t)]\cos(\omega_s t + \theta_{co-x}) - n_{x,i}(t)\sin(\omega_s t + \theta_{co-x}) \cdot 2\sin(\omega_s t + \theta_0(t)) \]

In case of ideal detection, \( \theta_0(t) = 0 \), equation 2.7 will reduce to:

\[ I_x(t) = [A_x + n_{x,c}(t)]\cos(\theta_{co-x}) - n_{x,i}(t)\sin(\theta_{co-x}) \]

(2.8)

\[ Q_x(t) = [A_x + n_{x,c}(t)]\sin(\theta_{co-x}) + n_{x,i}(t)\cos(\theta_{co-x}) \]

Averaging Equation 2.8 will result in:

\[ <I_x(t)> = A_x \cos(\theta_{co-x}) \]

(2.9)

\[ <Q_x(t)> = A_x \sin(\theta_{co-x}) \]

Both signals are a measure of the phase difference between the co- and cross-polar.

The phase difference can be determined by:
\[
\theta_{co-x} = \arctan \frac{Q_x}{I_x} = \arctan \frac{A_x \sin \theta_{co-x}}{A_x \cos \theta_{co-x}}
\]

Moreover, it is possible to calculate \(A_x\) from the detected co- and cross-polar by true envelope detection:

\[
A_x = \sqrt{(I_x(t))^2 + (Q_x(t))^2} = A_x [\sin^2 \theta_{co-x} + \cos^2 \theta_{co-x}]
\]

2.4 Communication between the different devices

There must be a communication link between the FLL/PLL detector and the decimator/detector board. Data which have to be transmitted consists of the in-phase and quadrature components. The communication link is based on a First In, First Out (FIFO) principle. All data are sent to the other board without handshaking or error correction. This is done because the transfer rate must be high.

Additional data have to be sent to a PC via a parallel data link. Data to be transmitted are the frequency spectrum calculated by the FLL and the status of the FLL. The PC-link is based on a Centronics interface. The redefined Centronics interface operates in a master slave configuration where the PC is the master. It is possible to connect 8 slaves. The principles of the communication between the different devices will be extensively discussed in Chapter 6.
Hardware for the digital FLL/PLL

The hardware of the FLL/PLL is designed by Op den Camp [1] and Strik [2]. The global design consists of a Digital Signal Processor (DSP), memory, a waitstate generator, an I/O-section and a reset and oscillator circuit. A block diagram is shown in Figure 3.1.

Figure 3.1 Hardware block diagram of the FLL/PLL

3.1 The Digital Signal Processor

The used DSP is a Texas Instruments TMS320C25-50. This DSP has the following characteristics:
- 50 MHz clock frequency,
- 544 words of internal RAM,
- 64k words addressable RAM and 64k words addressable ROM,
- 16 bit address and data bus,
- 16 bit address and data bus,
- 3 external interrupts,
- 40 MIPS.

### 3.2 Memory

The total memory space of a TMS320C25 DSP is divided into 64 k words program memory and 64 k words data memory. The program memory contains the software, the data memory contains the variables.

In this hardware design the program memory is divided into 32 k words EEPROM for code storage and 32 k words of RAM to hold the program code while executing. The program code copies itself into RAM where program execution is continued at triple speed. The data memory is divided into on-chip and external RAM. The address space of the on-chip RAM is 1024 words of which 544 words are available for data memory, divided into 3 blocks B0, B1 and B2. The other words are for internal use and not available for the user.

![Program and Data memory configuration](image)

**Figure 3.2 Program and Data memory configuration**

The memory configuration used is shown in Figure 3.2. To address and select the different chips, an address decoder is used which uses the signals *PS* (Program Memory Select), *DS* (Data Memory Select) and A15 (address line 15). *PS* is low when program memory is selected and *DS* is low when Data memory is selected. The truth table of the address decoder is shown in Table 3.1.

---

1. *X should be implemented as NOT(X), where X is the variable mentioned after the asterisk.
### Table 3.1 The truth table of the address decoder

<table>
<thead>
<tr>
<th>A15</th>
<th>DS</th>
<th>PS</th>
<th>active output</th>
<th>selected memory</th>
</tr>
</thead>
<tbody>
<tr>
<td>L</td>
<td>L</td>
<td>L</td>
<td>Y0</td>
<td>none</td>
</tr>
<tr>
<td>L</td>
<td>L</td>
<td>H</td>
<td>Y1</td>
<td>none</td>
</tr>
<tr>
<td>L</td>
<td>H</td>
<td>L</td>
<td>Y2</td>
<td>EPROM</td>
</tr>
<tr>
<td>L</td>
<td>H</td>
<td>H</td>
<td>Y3</td>
<td>none</td>
</tr>
<tr>
<td>H</td>
<td>L</td>
<td>L</td>
<td>Y4</td>
<td>none</td>
</tr>
<tr>
<td>H</td>
<td>L</td>
<td>H</td>
<td>Y5</td>
<td>Data RAM</td>
</tr>
<tr>
<td>H</td>
<td>H</td>
<td>L</td>
<td>Y6</td>
<td>Program RAM</td>
</tr>
<tr>
<td>H</td>
<td>H</td>
<td>H</td>
<td>Y7</td>
<td>none</td>
</tr>
</tbody>
</table>

### 3.3 The waitstate generator

The waitstate generator controls the instruction cycle length. If a device (memory, I/O) is acting slow, the processor is halted for one or two clock cycles. In this way the slower acting device is able to finish the task that is being processed. All I/O operations and EPROM reads are implemented with two waitstates, RAM access is performed without waitstates.

The waitstate generator is implemented using two J-K flip-flops. Whenever EEPROM Select or I/O is chosen, two extra clock cycles are needed before READY goes high and the processor is able to continue its task.

### 3.4 The Input/Output-section

The Input/Output-section (I/O) consists of a selector, two A/D-converters, a D/A converter, a micro switch block and several I/O buses for the parallel and the FIFO communication.

The selector selects 1 of the 7 I/O sections. The signals used are A0, A1, Read/*Write, *I/O Space and *STROBE. The first three signals are used to select a device. The other signals are used for timing. The selected devices are shown in Table 3.2.

#### Table 3.2 The selected devices

<table>
<thead>
<tr>
<th>Port address</th>
<th>IN-operation</th>
<th>OUT-operation</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Read A/D converter 1</td>
<td>Send to FIFO</td>
</tr>
<tr>
<td>1</td>
<td>Read A/D converter 2</td>
<td>Send to parallel port</td>
</tr>
<tr>
<td>2</td>
<td>Read parallel port</td>
<td>Write to D/A converter</td>
</tr>
<tr>
<td>3</td>
<td>Read data from dip-switches</td>
<td>-</td>
</tr>
</tbody>
</table>

### The A/D Converter

A/D-converters are used to convert an analog input signal to a digital value, suitable for further processing by the DSP. In this application two AD779 A/D converters are used.
The specifications of these converters configured for this application are:
- Output: 14 bit, 2s complement
- Input: -5V to +5V, sample and hold
- Conversion time: 100k conversions per second

No extra chips are needed to apply this device in a DSP-board because all necessary hardware like voltage references, amplifiers etc. are implemented on chip. A high pass filter (\(\tau^1=340\) Hz) is implemented to filter the input signal.

**The D/A Converter**

D/A converters are used to transform digital signals into analog signals. One D/A converter (AD569) is implemented on the board. The specifications of the converter configured for this application are:
- Input: 16 bit
- Output: -5V to +5V
- Settling time: 3\(\mu\)s

Latches etc. are built in the chip. The only extra device needed is an external voltage reference. In this application an AD588 is used. A second order low pass filter is designed on the printed circuit board. This filter has not been implemented.

**The First In, First Out buffer**

For communication with the Decimator/Detector board a FIFO communication bus is used. The data is sent to the other board and stored in a buffer, waiting for further processing. The FIFO communication is discussed in Chapter 6.

**Dip switches**

The FLL/PLL board is equipped with a block of 8 dip switches. The board has a communication bus address specified by the setting of 3 dip switches. One extra dip switch is used to enable or disable the communication with the acquisition computer. Four dip switches are for future use. The use of the switches is shown in Table 3.3:

<table>
<thead>
<tr>
<th>Switch</th>
<th>Function</th>
<th>default</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Parallel address LSB</td>
<td>0</td>
</tr>
<tr>
<td>2</td>
<td>Parallel address</td>
<td>1</td>
</tr>
<tr>
<td>3</td>
<td>Parallel address MSB</td>
<td>0</td>
</tr>
<tr>
<td>4</td>
<td>Parallel I/O enable</td>
<td>1</td>
</tr>
</tbody>
</table>

To avoid unnecessary interrupts of the running program, addressing of the device is checked by hardware. The address specified by the acquisition computer and the FLL/PLL board, specified by switches 1, 2 and 3, are compared with hardware. If both addresses are found to be equal, an interrupt will be generated.
The frequency of the carrier will change in time because of Doppler shift, frequency drift and oscillator phase noise. Keeping track on the carrier frequency is important as the whole system is tuned for one carrier frequency. Tracking is done by calculating the frequency spectrum with a Discrete Fourier Transform. The frequency bin with the highest signal power should be the carrier frequency. The difference between the desired frequency and the measured carrier frequency is used to control the voltage controlled oscillator (VCO).

4.1 The Fourier Transform

The Fourier transform converts information between the time domain and the frequency domain. The Fourier transform of an analog signal \( x(t) \) is defined as:

\[
X(\omega) = \int_{-\infty}^{+\infty} x(t) e^{-j\omega t} dt \tag{4.1}
\]

where \( X(\omega) \) is a complex function. The integral exists if \( x(t) \) is piece wise continuous and if the absolute value of \( x(t) \) can be integrated.

In practice the analog signal \( x(t) \) is sampled at regular time intervals. For \( N \) samples \( x(t) \) is represented by \( x(nT) \), where \( n \) is the sample number and \( T \) is the sample period.

By assuming \( x(t)=0 \) for \( t<0 \) and for \( t>(N-1)T \) and by replacing \( x(t) \) by \( x(nT) \), \( \omega \) by \( k\Omega \) (\( 0<k<N \)) and \( t \) by \( nT \), Formula 4.1 becomes:

\[
X(k) = \sum_{n=0}^{N-1} x(nT) e^{-j2\pi kn/T} \tag{4.2}
\]

where \( \Omega=2\pi/NT \) is the fundamental frequency and \( X(k) \) is understood to represent \( X(k\Omega) \). Equation (4.2) is generally known as the Discrete Fourier Transform (DFT). Rewriting equation (4.2) gives:

\[
X(k) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi nk/N} = \sum_{n=0}^{N-1} x(n)W_N^{nk} \tag{4.3}
\]
$W_N = e^{j2\pi/N}$ is known as the twiddle factor. It can be seen as a $N$-th part of the unity circle in the complex plane.

### 4.2 The Fast Fourier Transform

Equation 4.3 is an $N$-point Discrete Fourier Transform (DFT). Because of the large quantity of multiplications and additions (Order $N^2$), this method is not generally used. If $N$ is a power of 2, a more sophisticated way of calculating a Fourier transform can be performed by decomposing equation 4.3 into two DFTs of length $N/2$ by splitting the input samples into even and odd samples.

$$X(k) = \sum_{n=0}^{N-1} x(n)W_N^{nk} = \sum_{n=0}^{N/2-1} x(2n)W_N^{2nk} + \sum_{n=0}^{N/2-1} x(2n+1)W_N^{(2n+1)k} \quad (4.4)$$

Identifying the even samples $x_1(m)$ and the odd samples $x_2(m)$, results in

$$X(k) = \sum_{m=0}^{N/2-1} x_1(m)W_N^{mk} + W_N^{Nk/2} \sum_{m=0}^{N/2-1} x_2(m)W_N^{mk} \quad (4.5)$$

where $W_N = W_N^{-N/2}$. Equation 4.5 may be written as follows:

$$X_1(k) = X_{11}(k) + W_N^{k}X_{12}(k) \quad (4.6)$$

By comparing equation 4.6 with equation 4.4 it is seen that $X_{11}$ and $X_{12}$ are reduced to $N/2$-point DFTs. Repeating this rewriting, eventually results in $N/2$ 2-point DFTs. The 2 point DFT is a basic operation, called butterfly because of its shape in graphical representation (Figure 4.3).

![Figure 4.3 Representation of the Decimation in Time (DIT) radix 2 butterfly.](image)

The formulas to perform a 2-point DFT are:

$$X(k) = X_1(k) + W_N^{k}X_2(k)$$

$$X(k+N/2) = X_1(k) + W_N^{k+N/2}X_2(k) \quad (4.7)$$

$$X(k) = X_1(k) - W_N^{k}X_2(k)$$

This butterfly can be used to recompose the Fourier Transform of $N$ discrete signals. A decomposition is called a stage, and the total numbers of stages is given by $M = \log N$.
Calculating a FFT results in $N$ values which represent the power of the signal $x(t)$ in a frequency interval which is called a bin. The size of one bin is:

$$\Delta f = \frac{1}{NT} = \frac{f_s}{N} \quad (4.8)$$

The spectrum will cover a frequency interval of $\Delta f \cdot N = f_s$ Hz.

The input data in Figure 4.4 are placed in bit reversed order, e.g. the sample $x(1)$ is placed on the location of sample $x(4)$ because 4 (binary representation: 001) is the bit reversed number of 1 (binary representation: 001).

### 4.3 Window functions to be used

A run of $N$ samples is used for a FFT. When this run is not adapted at the beginning and at the end, the sharp edges in the time domain will return as high frequency components in the frequency domain. Sharp edges are the result of multiplying an infinite run of samples by a square window function in time. The high frequency components will result in leakage of information to other bins in the frequency domain. For this reason sharp edges in the window function should be avoided.

One of the most common solutions to solve the problem of sharp edges is the Hamming window:

$$h(n) = 0.54 - 0.46 \cos\left(\frac{2\pi n}{N}\right), \quad n=0,1,2,\ldots,N-1 \quad (4.9)$$

Almost the same result can be obtained by using the Hanning window:

$$h(n) = 0.5 - 0.5 \cos\left(\frac{2\pi n}{N}\right), \quad n=0,1,2,\ldots,N-1 \quad (4.10)$$

The time and frequency response of the Hamming, Hanning and square-shaped windows are shown in Figure 4.5 - 4.10.
In these figures can be seen that a square window has worse characteristics than a Hanning
or Hamming window. A Hanning window has lower first lobes, but these remain at the same level. This window is chosen because the side lobes of the Hamming window decrease with increasing frequency distance from the center frequency.

4.4 Choosing the number of samples for the FFT.

Choosing a number of samples for the FFT is done by trading off different contributions to the carrier phase drift. Choosing a large N will implement a large sample acquisition time and a larger calculation time. Moreover, the Doppler shift may eventually cause the carrier to shift through several bins. However, the frequency bins will be smaller. An optimum N must be determined, taking these factors into consideration. Table 4.5 shows the various contributions to the carrier frequency drift for the different beacons.

Table 4.5 Contributions to carrier frequency drift

<table>
<thead>
<tr>
<th>Beacon</th>
<th>B₀</th>
<th>B₁</th>
<th>B₂</th>
<th>Hz/sec</th>
</tr>
</thead>
<tbody>
<tr>
<td>Doppler Shift</td>
<td>0.0013</td>
<td>0.0022</td>
<td>0.0031</td>
<td></td>
</tr>
<tr>
<td>Satellite &amp; LO frequency drift</td>
<td>0.17</td>
<td>0.29</td>
<td>0.44</td>
<td>Hz/sec</td>
</tr>
</tbody>
</table>

Using a sample rate of 12.288 kHz and different number of samples the Table 4.6 can be drawn up for beacon B₀:

Table 4.6 Effect of the number of samples to the acquisition time and frequency drift

<table>
<thead>
<tr>
<th>Bin size</th>
<th>12</th>
<th>6</th>
<th>3</th>
<th>1.5</th>
<th>Hz</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sample acquisition time</td>
<td>83</td>
<td>166</td>
<td>333</td>
<td>670</td>
<td>ms</td>
</tr>
<tr>
<td>Total drift over acquisition time</td>
<td>0.014</td>
<td>0.028</td>
<td>0.057</td>
<td>0.114</td>
<td>Hz</td>
</tr>
</tbody>
</table>

The frequency drift of the signal is small compared to the bin size. The bin size is important, because a stable input frequency will result in a better performance of the PLL. The smallest possible bin size should be the best. However, at a sample rate of 12.288 kHz, this would result in 8192 samples, leading to a huge number of multiplications and high memory usage. Changing the sample rate to 6144 Hz will result in 4096 samples. By reducing the sample rate, the covered part of the spectrum will decrease.

Using a bin size of 3 Hz and a sample rate of 6144 Hz will result in an equilibrium between calculation time and accuracy of the FLL.

While testing the programs with the development system, the available memory is limited to 4000h. This results in a maximum number of samples of 1024.
5.1 A conventional analog PLL

A Phase-Locked Loop (PLL) is a special kind of feedback control system that keeps the phase difference between the input and output signal as small as possible. A basic analog PLL looks like Figure 5.1. It consists of a phase detector (PD), a loop filter (LF) and a voltage controlled oscillator (VCO).

![Figure 5.1 A basic analog PLL](image)

The analog phase detector compares the phase $\theta_i(t)$ of a periodic input signal with the phase $\theta_o(t)$ of the VCO output signal $x_o(t)$. The output of the phase detector is a signal which is proportional to the phase difference between the two phase detector input harmonic signals. The output of the PD is filtered by the loop filter. The loop filter output signal will adjust the frequency of the VCO in a direction that reduces the phase difference between the input signal and the VCO output signal. The basic principles of a PLL have been extensively discussed in Gardner [5] and in Op Den Camp [1].

The advantage of a digital PLL (DPLL) are the parameters that can easily be controlled. Basically DPLLs can be divided into two categories:

* Those which are built with gates, counters, registers etc. These are available in IC's such as 4046 and LM565.
* Those which uses an A/D converter and an algorithm running on a microprocessor.

To implement a PLL on a DSP it has to be implemented in software.

Two different algorithms are discussed in Op Den Camp [1]. Concluded is that the best PLL for this application is a Linear All Digital Phase-Locked Loop. This PLL is the
digital equivalent of the model of the analog Phase-Locked Loop.

5.2 The All Digital Phase-Locked Loop

At first, Digital Phase-Locked Loops were derived from analog PLL by replacing the different components by their digital equivalent. In this way some disadvantages of the analog PLL remained present. The most important disadvantages are:
- Generation of high order frequency components by the phase detector
- non-linear behaviour
- time-variant behaviour

Hagiwara [3] proposed a Linear All Digital Phase-Locked Loop (L-ADPLL), based on a zero-order loop filter which results in a first-order PLL. In 1989 Shayan and Le-Ngoc [4] proposed a second-order L-ADPLL, shown in Figure 5.2. The implemented loop filter is a first-order filter which results in a second-order PLL. A second-order PLL has better characteristics.

![L-ADPLL proposed by Shayan [4]](image)

The digitized input signal is split and one part transformed by a Hilbert transform (H(z)), which shifts all frequencies 90°. Because a Hilbert transform takes some time, the other part has to be delayed by $\Delta T$.

Assume $x_i(t)$ is defined as:

$$x_i(t) = A \sin(\omega_i t + \theta_i(t)) \quad (5.1)$$

The sampled (and delayed) signal $X_i(k)$ will be:

$$X_i(k) = A \sin[\omega_i kT + \theta_i(kT)] \quad (5.2)$$

and the Hilbert transformed signal $Y_i(k)$ will be

$$Y_i(k) = A \cos[\omega_i kT + \theta_i(kT)] \quad (5.3)$$

From (5.1) and (5.2) the input signal argument $\phi_i(k)$ can be calculated:

$$\phi_i(k) = \tan^{-1} \frac{X_i(k)}{Y_i(k)} = [\omega_i kT + \theta_i(kT)] \text{mod} 2\pi \quad (5.4)$$

The phase detector subtracts $\phi_o(k)$ from $\phi_i(k)$. The Difference is called $\phi_d(k)$. Since $\phi(k)$
and $\phi_o(k)$ are periodic functions, the substraction does not always produce the right phase difference. Correction is performed by the mod $[-\pi, \pi]$ block. Figure 5.3 shows two situations, a phase lead (situation I) and a phase lag (situation II).

\[ H_{LF}(z) = \frac{C_1}{z-1} + C_2 \quad (5.5) \]

The transfer function of the DVCO is:

\[ H_{DVCO}(z) = \frac{1}{z-1} \quad (5.6) \]

The open loop transfer function is:

\[ H_{OL}(z) = H_{LF}(z) \cdot H_{DVCO}(z) = \frac{C_2(z-1) + C_1}{(z-1)^2} \quad (5.7) \]

Thus the Closed loop transfer function $H(z)$ is:

\[ H(z) = \frac{\phi_o}{\phi_i} = \frac{H_{OL}(z)}{1 + H_{OL}(z)} = \frac{C_2(z-1) + C_1}{(z-1)^2 + C_2(z-1) + C_1} \quad (5.8) \]

This is a second order transfer function of the Linear A-DPLL.

Using the forward difference method $(z-1)$ can be replaced by $sT$ [12]. Using this method Equation 5.8 can be transformed to the transfer function of an analog second order PLL:

---

Figure 5.3 Behaviour of the mod$[-\pi, \pi]$ circuit.

---

22 Optimization and testing of a digital FLL/PLL
\[ H(s) = \frac{2\xi \omega_n s + \omega_n^2}{s^2 + 2\xi \omega_n s + \omega_n^2} \] (5.9)

If \( f_n \ll f_s \), \( C_1 \) and \( C_2 \) can be calculated from 5.8 and 5.9:

\[
C_1 = 2\xi \omega_n T \\
C_2 = \omega_n^2 T^2
\] (5.10)

The steady state error for different inputs can be calculated using the final value theorem [12]:

\[
e_{ss} = \lim_{z \to 1} \frac{(1-z)}{(1-H(z))} \phi_i(z)
\] (5.11)

The \( z \)-transforms of the different inputs are:

\[
\phi_i(z) = \frac{z}{z-1} \Delta_1 \quad \text{Phase step of } \Delta_1 \text{ rad}
\]

\[
\phi_i(z) = \frac{zT}{(z-1)^2} \Delta_2 \quad \text{Phase ramp of } \Delta_2 \text{ rad/s}
\]

\[
\phi_i(z) = \frac{zT^2}{(z-1)^3} \Delta_3 \quad \text{Phase acceleration of } \Delta_3 \text{ rad/s}^2
\]

Using equation 5.11 to calculate the steady state error gives respectively:

\[
e_{ss} = 0 \quad \text{Phase step of } \Delta_1 \text{ rad}
\]

\[
e_{ss} = 0 \quad \text{Phase ramp of } \Delta_2 \text{ rad/s}
\]

\[
e_{ss} = \frac{\Delta_3}{\omega_n^2} \quad \text{Phase acceleration of } \Delta_3 \text{ rad/s}^2
\]

These results corresponds to the steady state errors of an analog second order PLL.

Because the steady state error of a phase ramp (frequency step) is 0, the hold-in-range of the PLL will be large. This can be verified by calculating the DC Loop gain:

\[
G_{DCloop} = \lim_{z \to 1} H_{OL}(z) = \lim_{z \to 1} \frac{C_2(z-1) + C_1}{(z-1)^2} \to \infty
\] (5.12)

For a linear phase detector the hold-in-range is proportional to the DC Loop Gain.
Stability can be reached by placing the poles of the closed loop inside the unit circle in the z-plane. The poles of $H_{\text{CL}}$ are:

$$z_{1,2} = \frac{2 - C_2 \pm \sqrt{(C_2 - 2)^2 - 4(C_1 - C_2 + 1)}}{2}$$

(5.13)

Positioning of $z_{1,2}$ inside the unit circle gives:

$$2C_2 - 4 < C_1 < C_2 , \quad C_1 > 0$$

(5.14)

These constants can be calculated with Formula 5.10. Using $\xi = 0.707$ and $\omega_n = 17.2 \pi \text{ rad/s}$ (17 Hz) will result in $C_1 = 75.56 \cdot 10^{-6}$ and $C_2 = 12.29 \cdot 10^{-3}$. These values appear to meet the constraints from 5.14.

The free-running frequency should be 124.952 kHz. Using the subsampling theorem, this results in a free running frequency of 1536 Hz. This frequency is represented by $C_0$ and is determined by the number of samples per period (8 samples/period).

$$C_0 = \frac{2^{16}}{f_s/1536}$$

(5.15)

### 5.3 Measurement results of the ADPLL

Several characteristics of the ADPLL are measured using the HP-3325A oscillator. Measurements with noisy signals are carried out using the satellite signals of the beacon receiver at 40 GHz.

**Measurements without noise**

The response of the phase error $\phi_e$ is simulated with MatLab. Using an input step of 100 Hz resulted in Figure 5.4. The phase error is reduced to zero in 13 $\mu$s. The maximum phase error is 31 degrees. The step response is measured and matches the simulated step.

**Figure 5.4 Phase error $\phi_e$ as a result of a frequency step of 100 Hz, using $\xi = 0.707$ and $f_n = 100$ Hz**
response. Assuming the PLL is locked, coherent detection is performed correctly. The data shown in Figure 5.5 is measured by varying the amplitude of the co-polar input signal. As shown in Equation 2.6, $I_c$ should be proportional to the amplitude of the co-polar input signal, $Q_c$ should equal 0. The high frequency components, which are the result of the multiplications in the coherent detection procedures, are not removed in the FLL/PLL software because a digital filter is not implemented. A Low-Pass RC filter ($\tau=550\text{Hz}$) will remove the high frequency components after the D/A-converter has converted the digital output to an analog signal.

As shown in Figure 5.5, $I_c$ is linear with the co-polar input signal. Varying the phase angle between the co- and cross-polar input signal results in varying $I_x$ and $Q_x$ signals. The phase angle between the co- and crosspolar is determined (formula 2.8 and 2.9) by the calculation of the arctangent of the quotient of the $I$ and $Q$ signals. The result is shown in Figure 5.6

Figure 5.5 $I_c$ and $Q_c$ as a function of the amplitude of the Co-polar input signal
As shown in Figure 5.6, the phase angle equals the input phase difference. The output range is limited by the arctangent circuit to $[-90^\circ, 90^\circ]$.

**Measurements with noise**

For measurements with noise the ITALSAT-satellite beacon signal received at 40 GHz is used because the noisy signal is difficult to simulate. The S/N ratio is dependent on rain attenuation etc. It is also possible to change the S/N ratio manually by using an attenuator in the wave guide.

**Figure 5.7 Measurement setup**

---

1 Attention should be paid to unwanted noise in the system. Unwanted noise in the digital FLL/PLL is added by the digital part to the analog part. Moreover, the FLL/PLL and the PC-clock signal add noise at a high level to the analog PLL receivers. Keeping a distance of about 3 meter is a solution to the last problem. The first problem might be solved by redesigning the digital FLL/PLL printed circuit board.
The 40 GHz signal is down converted to 10 MHz. This signal is used as input for the analog PLL, which down converts the co-polar to 125 kHz. The analog PLL is used because a VCO for use with the FLL was not available.

Kamperman [13] added an amplifier and a bandpass filter to the analog PLL. The amplifier adapts the 125 kHz signal amplitude to the input range (-5V/5V) of the A/D converter. The bandpass filter prevents aliasing. The transfer characteristic of the amplifier/bandpass filter in the down converter is shown in Figure 5.8.

### Figure 5.8 The amplitude transfer-characteristic of the amplifier / bandpass filter section

The power spectrum of the 10 MHz signal with noise looks like Figure 5.9. The mainlobe is the noise which is mainly generated by the amplifiers and local oscillators. The smaller lobe on top of the main lobe is the received satellite signal. Whenever the satellite signal is attenuated by rain etc., the small lobe will shrink. The main lobe will remain the same because this lobe is the result of noise generated by the receiversystem.

### Figure 5.9 An impression of the shape of the powerspectrum at 10 MHz of a satellite signal

The C/N ratio of the signal at 10 MHz is determined using a power meter. The power meter is able to determine the signal power with noise of the powerspectrum shown in
Figure 5.9. When the satellite signal is removed using a piece of absorbing material mounted on the antenna, only the noise will remain.

Table 5.1 The measured power densities

<table>
<thead>
<tr>
<th>Signal</th>
<th>Power [dBm]</th>
<th>Power [W]</th>
</tr>
</thead>
<tbody>
<tr>
<td>(S+N)</td>
<td>-3.55</td>
<td>441 \times 10^{-6}</td>
</tr>
<tr>
<td>(N)</td>
<td>-5.3</td>
<td>295 \times 10^{-6}</td>
</tr>
</tbody>
</table>

This will result in a C/N ratio of:

\[
\frac{C}{N} = \frac{(S+N) - (N)}{(N)} = 0.5 \approx -3\,dB
\]  

(5.16)

The C/kT, which is constant in the whole system, is determined using the noise bandwidth of the noise signal at 10 MHz. The noise bandwidth by the bandpass filter in the 10 MHz IF-strip. The amplitude-transfer function of the filter is shown in Figure 5.10.

![Figure 5.10 The amplitude-transfer characteristic of the bandpass filter in the 10 MHz IF-strip](image)

The noise bandwidth of the bandpass filter is calculated using the following Formula [13]:

\[
B_N = \frac{\int_{0}^{\infty} |H(f)|^2 \, df}{|H(f)|_{\text{max}}^2}
\]  

(5.17)

Unfortunately only the transfer characteristic is known, so Formula 5.17 has to be evaluated numerically, which results in a noise bandwidth of 44.3 kHz.
Using this bandwidth the C/kT can be calculated:

\[
\frac{C}{kT} = \frac{C}{N} \cdot \frac{B_N}{N} = 0.5 \cdot 44.3 \text{kHz} = 43.5 \text{dBHz}
\]  (5.18)

The noise bandwidth of the bandpass filter/attenuator is also evaluated numerically. The noise bandwidth of this filter is 1530 Hz. Using the C/kT and the noise bandwidth of the bandpass filter/attenuator, the C/N at the input of the digital FLL/PLL can be calculated:

\[
\frac{C}{N} = \frac{C}{kT} \cdot \frac{1}{B_N} = 43.5 \text{dBHz} - 10 \log(1530) = 11.6 \text{dB}
\]

The analog PLL satellite receiver, which is currently used for the propagation experiments, will be used as a reference. The output of these receivers is registered on a chart. Calibration is done using a HP-synthesizer which generates a 10 MHz signal. This signal from the synthesizer is added to the 10 MHz signal as shown in Figure 5.11. In this way the synthesizer is able to simulate the satellite carrier, assuming that the noise is only added by the circuits after the attenuator in the wave guide. The received satellite signals with the dish antenna are removed with a piece of absorbing material.

![Figure 5.11 The setup of the receiver for calibration experiments](image)

The chart can be calibrated by changing the amplitude of the simulated satellite signal with the synthesizer.

The dynamic inputrange of the DPLL is calculated using the resolution of the A/D converter:

\[
\text{Dynamic inputrange} = 10 \log(2^{14}) = 42.1 \text{ dB}
\]  (5.20)

Due to crosstalk and non-linearity, the resolution of the A/D converter is smaller. The linearity is determined using the output of the analog PLL as a reference. This resulted in the following characteristic.

From Figure 5.12 it is shown that the characteristic of the digital FLL/PLL remains linear from a C/N ratio at the input of the digital FLL/PLL of 11.6 to -10.5 dB. At lower C/N ratios the analog PLL compresses the signal.

If the input frequency is varied without changing the input level, the detected I_c will change. This should not happen as shown in equation (2.4) and (2.5). An explanation is the non-ideal transfer function of the amplifier/bandpass filter in the 10 MHz IF-strip.
non-ideal transfer function of the amplifier/bandpass filter in the 10 MHz IF-strip.

Figure 5.12 The linearity of the FLL/PLL calibrated using the analog PLL

The noise behaviour of the ADPLL is determined by measuring the phase error after the loopfilter as a function of the C/N ratio at the input of the digital FLL/PLL. For simulating a worse signal to noise ratio, an attenuator in the waveguide is used.

Figure 5.13 Phase angle $\phi_e$ as function of the input C/N ratio

The phase error is an indication for the lock status. A phase error of 9 degrees is an indication the loop might be out of lock. Using this criterion, the loop falls out of lock when the C/N is about -15 dB. The analog PLL goes out of lock at -12 dB.
The FLL/PLL is able to communicate with a Personal Computer and with the decimator/detector. For communication with the PC, a modified Centronics interface is used. Transmitting data to the Decimator/detector is done with a single direction communication buffer.

### 6.1 The communication between the FLL/PLL and the PC

The parallel communication is based on a standard Centronics parallel interface. This interface is a standard device on every personal computer. The Centronics interface consists of 12 digital outputs and 5 digital inputs. The Centronics interface on a PC has a 25 pin female sub-D connector. The pinning shown in Figure 6.1

<table>
<thead>
<tr>
<th>pin</th>
<th>I/O</th>
<th>original function</th>
<th>pin</th>
<th>I/O</th>
<th>original function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>O</td>
<td>Strobe</td>
<td>10</td>
<td>I</td>
<td>Acknowledge</td>
</tr>
<tr>
<td>2</td>
<td>O</td>
<td>Data 0</td>
<td>11</td>
<td>I</td>
<td>Busy</td>
</tr>
<tr>
<td>3</td>
<td>O</td>
<td>Data 1</td>
<td>12</td>
<td>I</td>
<td>Out of paper</td>
</tr>
<tr>
<td>4</td>
<td>O</td>
<td>Data 2</td>
<td>13</td>
<td>I</td>
<td>Select</td>
</tr>
<tr>
<td>5</td>
<td>O</td>
<td>Data 3</td>
<td>14</td>
<td>O</td>
<td>Auto Feed</td>
</tr>
<tr>
<td>6</td>
<td>O</td>
<td>Data 4</td>
<td>15</td>
<td>I</td>
<td>Error</td>
</tr>
<tr>
<td>7</td>
<td>O</td>
<td>Data 5</td>
<td>16</td>
<td>O</td>
<td>Reset</td>
</tr>
<tr>
<td>8</td>
<td>O</td>
<td>Data 6</td>
<td>17</td>
<td>O</td>
<td>Select</td>
</tr>
<tr>
<td>9</td>
<td>O</td>
<td>Data 7</td>
<td>18-25</td>
<td></td>
<td>Ground</td>
</tr>
</tbody>
</table>

The 12 available output lines are used for communication from the PC to the device. Three lines are used to select a device. One line is used to generate an interrupt for the device and eight lines are used to transmit data. The five available input lines are used to receive
and eight lines are used to transmit data. The five available input lines are used to receive 
data from the device. One line is used to get an interrupt from the device, the other 4 lines 
are used to receive data. The redefined Centronics port is shown in Figure 6.2.
Redefining the Centronics interface in this way will create the possibility for the master 
(the PC) to communicate with 8 different slaves.

Table 6.2 Pinout of the redefined Centronics interface.

<table>
<thead>
<tr>
<th>pin</th>
<th>I/O function</th>
<th>pin</th>
<th>I/O function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>O IRQ on device</td>
<td>10</td>
<td>I IRQ on PC</td>
</tr>
<tr>
<td>2</td>
<td>O Data Out 0</td>
<td>11</td>
<td>I Data In 3</td>
</tr>
<tr>
<td>3</td>
<td>O Data Out 1</td>
<td>12</td>
<td>I Data In 2</td>
</tr>
<tr>
<td>4</td>
<td>O Data Out 2</td>
<td>13</td>
<td>I Data In 1</td>
</tr>
<tr>
<td>5</td>
<td>O Data Out 3</td>
<td>14</td>
<td>O Device Select 0</td>
</tr>
<tr>
<td>6</td>
<td>O Data Out 4</td>
<td>15</td>
<td>I Data In 0</td>
</tr>
<tr>
<td>7</td>
<td>O Data Out 5</td>
<td>16</td>
<td>O Device Select 1</td>
</tr>
<tr>
<td>8</td>
<td>O Data Out 6</td>
<td>17</td>
<td>O Device Select 2</td>
</tr>
<tr>
<td>9</td>
<td>O Data Out 7</td>
<td>18-25</td>
<td>Ground</td>
</tr>
</tbody>
</table>

The communication protocol

Besides a good hardware definition of the communication system, a protocol has to be 
designed. The used protocol is the result of the following constraints:
- Minimum processor time occupation
- ASCII commands and answers must be possible
- transmission error detection and recovery
Minimum processor occupation is achieved by using interrupt-driven procedures. By using 
Cyclic Redundancy Check (CRC) for error detection and timers for time-out detection,
errors can be detected and recovered.

The designed communication packet sent by the PC has the following structure:

| SOH | LEN | D₁  | D₂  | ... | D_{len-4} | CRCₜ | CRC₁ | EOT |

The meaning of the fields are:
SOH: Start of Header
LEN: Length of the packet in bytes (not including SOH)
D₁,...,Dₙ: Data fields
CRCₜ, CRC₁: 16 bit CRC-word from SOH to D_{len-4}
EOT: End of Transmission
Every character transmitted by the PC is coded as an 8 bit wide word.
A Communication packet send by a device has the following structure:

| SOH | LEN₂ | LEN₁ | LEN₀ | D₁₈ | D₁₁ | D₂₈ | D₂₁ | .... | D LEN₆ | D LEN₅ | CRC₂ | CRC₁ | CRC₀ | EOT |

The meaning of the fields are:
- **SOH**: Start of Header
- **LEN₂..LEN₁**: Length (LEN₂*256+LEN₁*16+LEN₀) of the packet in nibbles (not including SOH)
- **D₁₈..D₁₁..D₂₈..D₂₁..**: Low and High data fields
- **CRC₂, CRC₁**: 16 bit CRC-word from SOH to D LEN₆..D LEN₅..
- **EOT**: End of Transmission

Every character transmitted is 4 bit wide.

### 6.2 Solutions for Error Detecting

Error detection and recovery is done by using time-outs and Cyclic Redundancy Check (CRC) on both the master and the slave.

Using a Time-Out signal will prevent a device waiting for an answer.

Cyclic redundancy check is one of the most common error detecting codes. It is based on the use of polynomials. At both sides a generator polynomial has been agreed upon. The algorithm used in this program is based on a lookup table and was published in Byte [8].
6.3 The communication between the FLL/PLL and the decimator/detector

The First In, First Out communication passes the in-phase and quadrature demodulated signals of both channels from the FLL/PLL to the decimator/detector. This one way communication is based on a FIFO buffer. The data are transmitted in parallel. Handshaking is done with a clock and ready signal. The specification of the FIFO data format is as follows:

<table>
<thead>
<tr>
<th>word</th>
<th>bit 0</th>
<th>bit 1</th>
<th>bit 2</th>
<th>bit 3</th>
<th>bit 4</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>$I_{co}$ bit 12</td>
<td>$I_{co}$ bit 13</td>
<td>$I_{co}$ bit 14</td>
<td>$I_{co}$ bit 15</td>
<td>status data available</td>
</tr>
<tr>
<td>2</td>
<td>$I_{co}$ bit 8</td>
<td>$I_{co}$ bit 9</td>
<td>$I_{co}$ bit 10</td>
<td>$I_{co}$ bit 11</td>
<td>status PLL</td>
</tr>
<tr>
<td>3</td>
<td>$I_{co}$ bit 4</td>
<td>$I_{co}$ bit 5</td>
<td>$I_{co}$ bit 6</td>
<td>$I_{co}$ bit 7</td>
<td>status FLL</td>
</tr>
<tr>
<td>4</td>
<td>$I_{co}$ bit 0</td>
<td>$I_{co}$ bit 1</td>
<td>$I_{co}$ bit 2</td>
<td>$I_{co}$ bit 3</td>
<td>undefined</td>
</tr>
<tr>
<td>5</td>
<td>$Q_{co}$ bit 12</td>
<td>$Q_{co}$ bit 13</td>
<td>$Q_{co}$ bit 14</td>
<td>$Q_{co}$ bit 15</td>
<td>undefined</td>
</tr>
<tr>
<td>6</td>
<td>$Q_{co}$ bit 8</td>
<td>$Q_{co}$ bit 9</td>
<td>$Q_{co}$ bit 10</td>
<td>$Q_{co}$ bit 11</td>
<td>undefined</td>
</tr>
<tr>
<td>7</td>
<td>$Q_{co}$ bit 4</td>
<td>$Q_{co}$ bit 5</td>
<td>$Q_{co}$ bit 6</td>
<td>$Q_{co}$ bit 7</td>
<td>undefined</td>
</tr>
<tr>
<td>8</td>
<td>$Q_{co}$ bit 0</td>
<td>$Q_{co}$ bit 1</td>
<td>$Q_{co}$ bit 2</td>
<td>$Q_{co}$ bit 3</td>
<td>undefined</td>
</tr>
<tr>
<td>9</td>
<td>$I_{x}$ bit 12</td>
<td>$I_{x}$ bit 13</td>
<td>$I_{x}$ bit 14</td>
<td>$I_{x}$ bit 15</td>
<td>undefined</td>
</tr>
<tr>
<td>10</td>
<td>$I_{x}$ bit 8</td>
<td>$I_{x}$ bit 9</td>
<td>$I_{x}$ bit 10</td>
<td>$I_{x}$ bit 11</td>
<td>undefined</td>
</tr>
<tr>
<td>11</td>
<td>$I_{x}$ bit 4</td>
<td>$I_{x}$ bit 5</td>
<td>$I_{x}$ bit 6</td>
<td>$I_{x}$ bit 7</td>
<td>undefined</td>
</tr>
<tr>
<td>12</td>
<td>$I_{x}$ bit 0</td>
<td>$I_{x}$ bit 1</td>
<td>$I_{x}$ bit 2</td>
<td>$I_{x}$ bit 3</td>
<td>undefined</td>
</tr>
<tr>
<td>13</td>
<td>$Q_{x}$ bit 12</td>
<td>$Q_{x}$ bit 13</td>
<td>$Q_{x}$ bit 14</td>
<td>$Q_{x}$ bit 15</td>
<td>undefined</td>
</tr>
<tr>
<td>14</td>
<td>$Q_{x}$ bit 8</td>
<td>$Q_{x}$ bit 9</td>
<td>$Q_{x}$ bit 10</td>
<td>$Q_{x}$ bit 11</td>
<td>undefined</td>
</tr>
<tr>
<td>15</td>
<td>$Q_{x}$ bit 4</td>
<td>$Q_{x}$ bit 5</td>
<td>$Q_{x}$ bit 6</td>
<td>$Q_{x}$ bit 7</td>
<td>undefined</td>
</tr>
<tr>
<td>16</td>
<td>$Q_{x}$ bit 0</td>
<td>$Q_{x}$ bit 1</td>
<td>$Q_{x}$ bit 2</td>
<td>$Q_{x}$ bit 3</td>
<td>undefined</td>
</tr>
</tbody>
</table>

The hardware at the transmitter side consists of a 74F245 output buffer. At the receiver, a First In, First out (FIFO) buffer (74F225) is used. This buffer is 5 bit wide and is able to contain 16 words. The use of the FIFO eliminates the need for special synchronisation because the FLL/PLL is able to fill the FIFO independent of what the decimator/detector is doing. The Decimator/Detector receives an interrupt from the FIFO if the buffer is full (all data are available).
Implementation of the FLL/PLL software

7.1 Function description of and improvements made in the FLL/PLL program

The function description of the procedures and the improvements made in the program of Franken [9] are discussed in this paragraph. The FLL/PLL program consists of a main loop program, 2 interrupt driven procedures, a parallel command interpreter and the FLL algorithm. When not executing any (interrupt)procedure, the main loop program is executed.

The improvements are discussed in the following chapters, using the flow diagrams of the FLL/PLL program.

The interrupt 0 routine

This routine is called on every interrupt 0. The A/D converters, which are initiated by the 12.288kHz clock, generate this interrupt when the analog signals are converted. The samples are read and adapted to Q15 format. The PLL procedures are called for. Every sample is evaluated by the PLL algorithm.

The PLL algorithm that is used is an exact implementation of the block diagram shown in Figure 5.2. An ideal Hilbert transform will shift every frequency component 90°, however, an ideal Hilbert transform takes an infinite number of samples. To prevent this, a finite Hilbert transform is implemented which uses 35 samples. This will result in a delay equal to 17 samples. A table in memory is used for storage of the Hilbert constants. The time delay is compensated with a delay of the \( X_i(k) \) signal. The two signals, which represent a sine \( X_i(k) \) and a cosine \( Y_i(k) \), are divided. The result is an index to the arctangent table, which is only present from 0° to 45°. Values from 45° to 90° are calculated by its mirror value. By using the signs of \( X(k) \) and \( Y(k) \), angles in other quadrants can be calculated (table 7.1).
Table 7.1 Determination of the angle between X(k) and Y(k)

<table>
<thead>
<tr>
<th>Sign X(k)</th>
<th>Sign Y(k)</th>
<th>Argument</th>
</tr>
</thead>
<tbody>
<tr>
<td>+</td>
<td>+</td>
<td>$\phi=\theta$</td>
</tr>
<tr>
<td>+</td>
<td>-</td>
<td>$\phi=180^\circ - \theta$</td>
</tr>
<tr>
<td>-</td>
<td>-</td>
<td>$\phi=180^\circ + \theta$</td>
</tr>
<tr>
<td>-</td>
<td>+</td>
<td>$\phi= - \theta$</td>
</tr>
</tbody>
</table>

The result of the calculation represents a phase angle. The Phase detector is implemented by subtracting the desired angle from the measured angle. This implementation will result in a linear phasedetector characteristic.

The loop filter is implemented using a normal digital first-order filter with two constants $C_1$ and $C_2$. These constants are normalized to 16 bits by multiplying them with $2^{16}$.

The Digital Voltage Controlled Oscillator adds $C_0$ to the filtered signal. The calculation of the cosine of the output angle $\phi_o$ is done using a cosine-table. The sine of the phase angle $\phi_o$ can also be calculated with the cosine-table.

The interrupt routine also collects the samples for use with the FFT algorithm and stores every n-th (n=2) sample in bit reversed order in the memory. If a memory bank is full, a flag is set and another memorybank is filled. There are two memory banks for the storage of the samples.

Changes made to the interrupt 0 procedures and the called procedures are:

- The sequence of the PLL procedures is changed. The sequence was: output routines, Hilbertransform, delaylines, demodulation, lowpass filters, phasdetection, Phase-Locked Loop algorithm and is changed to: delaylines, Hilberttransform, phase detection, Phase-Locked Loop algorithm, demodulation, low pass filter, output routines.

- An extra procedure for testing is added (OUTPUT2). This procedure transfers the values of a specified memory location to the D/A converter. This procedure is not shown in the flowchart because it is for testing purposes only. In the finite program the D/A converter should be used for VCO control.

- The possibility to change the sample rate of the data stored in the FFT memory bank is added.

- A PLL lock detection is added. The absolute value of the phase error is added to a register. Every 1024 samples this register is read and compared with a threshold value. If this value is exceeded, the PLL is said to be out of lock. The PLL is reset.

- The interrupt 0 procedure is placed in block B0. This improves the performance of execution.

Figure 7.1 The flowchart of the interrupt 0 procedure
The Main loop

If the program is started, all variables are initialized. The main program consists of a loop which controls whether the FFT-memory bank is full and whether a parallel command is received. If the FFT-memory bank is full the FFT algorithm is called, the spectrum of the signal is calculated and the carrier is detected. If a parallel command is received the command interpreter is called.

The FFT algorithm consists of two separate parts. In the first part the memory banks are filled, as described in the previous chapter. In the second part of the FFT algorithm is started, all data are collected. In this case the memory bank is full and the calculation of the FFT will start. The FFT algorithm is implemented using the Butterfly algorithm [7] described in Chapter 4.2. A procedure which calls for the butterfly routine arranges the decomposition of the data.

Calculation of the FFT is done in place. Extra memory locations are required for the imaginary part of the calculated results.

When finished calculating the FFT, all values are squared and stored in a separate memory bank which represents the frequency spectrum. While squaring, the largest value and its location are searched. The location of this value determines the frequency of the carrier.

Dependent on the type of VCO, the calculated carrier frequency can be the input for a control loop with P or PI characteristic and with a desired nominal frequency of 124.519 kHz. A control loop for use with the VCO is not yet implemented.

Changes made to the main loop and the called procedures are:
- The Decimation in Time(DIT) Radix-2 Butterfly is replaced by the butterfly algorithm from [7].
- The procedures to calculate the spectrum and to search the carrier are optimized by rewriting several parts.
- The parallel command interpreter is replaced by the interpreter designed by Kooistra [6] because this procedure is more flexible.

Figure 7.2 The flowchart of the mainloop

The interrupt 2 routine

This procedure is called when a parallel command is sent or received. A status variable is used. At first the PAR_status variable is 0, during receiving PAR_status equals 1 and after the successful completion of PAR_status equals 2. During transmission PAR_status equals 3, as initialized by the procedure transmit. After transmission PAR_status is set to zero. There are no changes made to this procedures.
Some improvements are made to the variable declaration and look-up tables. The tables, which remain constant while debugging the code, are stored in a separate file which is linked with the sourcecode. This improves readability and compilation time. The variables are sorted and placed in one memory block. Due to this improvement the memory usage is more efficient and the usage of memory pages is prevented.

### 7.2 Unsolved problems

When separated, the software parts (FLL/communication and PLL) function correctly. When the parts of the software are combined, the FFT is not correctly calculated. This is the result of lack of processing power. When the sample frequency is decreased to 5 kHz, the combined software parts function correctly. The solution might be found in two possible explanations of the problem:

- The development system is slowing down the execution of the program. This problem can be solved using the FLL/PLL without development system by programming the EPROMs with the FLL/PLL software.
- The interrupt 0 procedure is too large. By optimizing the code or by increasing the processing power by using a faster version of the TMS320C25, this problem can be solved.

### 7.3 Memory Allocation

As shown in Chapter 3, the RAM of the TMS320C25 is divided into on-chip and off-chip memory. On-chip memory is addressed by 0h-3FFh, off-chip is addressed by 400h-2000h. 544 words of the 1000 addressable words are available to the user. These words are divided into three memory blocks: B0 (200h-2FFh), B1(300h-3FFh) and B2(60h-7Fh). Other memory blocks are for internal use by the processor. It is possible to execute a procedure from block B0. The advantage of this configuration is the fast execution of the procedure placed in this block, disadvantage is the loss of 100h memory space.

In the FLL/PLL software, block B0 is configured as program memory. The INTO routines copied into block B0 when the program is started. Block B1 is used for storing variables. The total number of variables is 88 (58h) and will fit into the lower half of block B1 (300h-37Fh). The upper half (380h-3FFh) is used for the three delaylines and the Hilberttransform constants.

Block B2 is used for the stack. The function of the stack is saving variables and the processor state when executing an interrupt.

The memory allocation is defined in a .CMD file (Appendix D). The assembler source code contains only names which represent the memory allocations. The command file is used by the linker to connect these names to memory locations.
Conclusions and recommendations

8.1 Conclusions

- The Frequency-Locked Loop for a digital beacon receiver is implemented using a 1024-point FFT. The reduced size of the number of points is the result of the lack of memory in the DSP development system.

- The Phase-Locked Loop is implemented using an All-Digital Phase-Locked Loop. The characteristics of this type of Phase-Locked Loop proved to be superior to those of the analog counterpart. Using $f_n=100$ Hz and $\zeta=0.707$ results in a hold-in range of 2500 Hz and a lock-in range of 600 Hz. The C/N ratio at the input of the FLL/PLL may decrease to -14.5 dB before loss of lock occurs.

- Communication between the PC and the FLL/PLL was tested and improved. Another communication procedure allows free programming of the command set. Using the current setup, data representing the discrete spectrum of the signal are sent every second to the PC.

- Coherent detection of the co- and crosspolar signals results in the in-phase and quadrature signals. It is possible to measure the phase difference between the co and crosspolar signal and the amplitudes of these signals.

- An accurate oscillator has been built, which produces the sampling clock.

- The software has been annotated, in order to improve accessibility.
# 8.2 Recommendations

- The exact characteristics of the FLL and PLL have to be determined using input signals with a well defined Carrier to Noise ratio.

- A computer program has to be written which logs the data. This program must be able to use the DCF-77 signal for time alignment.

- The interrupt routine calculating the PLL has to be optimized or a faster processor has to be used, to accomplish that the separate parts of the software cooperate. When functioning, this software has to be programmed in an EPROM.

- A voltage controlled oscillator has to be designed and built for converting the 10 MHz signal down to 124.516 kHz.

- Modern personal computers have high processing power. Using a 16-bit A/D-converter card for sampling the data, a fast computer and a C program may result in a simple and well organized system, which combines all advantages of a digital implementation of the receiver.
References

[1] Camp, Op Den, R.H.C.M.  
*Digital Beacon receiver for propagation experiments, Part I: Design and implementation of an FLL/PLL for coherent detection.*  
Eindhoven University of Technology, Department of electrical Engineering, Graduation Report, October 1992

[2] Strik, M.  
*Digital Beacon receiver for propagation experiments, Part II: A Decimator/Detector for Phase and Amplitude Measurements*  
Eindhoven University of Technology, Department of electrical Engineering, Graduation Report, October 1992

)*DSP-type first order digital phase-locked loop using linear phase detector.*  
Electronics and Communications in Japan, Part 1, Vol. 69, No. 6, 1986

*All digital phase-locked loop: concepts, design and applications.*  

*Phaselock Techniques.*  
New York, John Wiley & Sons, 1979

*On the development of a spread spectrum transceiver based on a digital signal processor.*  
Eindhoven University of Technology, IVO, IVO-report, 1995

[8] Le Van, J.

Digital beacon receiver for propagation experiments: Design, testing and optimization of a Frequency-Locked Loop / Phase-Locked Loop Eindhoven University of Technology, Department of electrical Engineering, Graduation Report, April 1994.

[10] Propagation experiments in the Netherlands in the 12,20 and 30 GHz band with the Olympus-satellite. PTT research Dr. Neher laboratories, Delft University of Technology, Eindhoven University of Technology.

Fase- en amplitude metingen met behulp van een digitale signaal processor Eindhoven University of Technology, Department of electrical Engineering, Graduation Report, July 1991.


[13] Shanmugam, K. Sam
Digital and analog communication systems John Wiley and Sons, 1985
Appendix

The 12.288 kHz Oscillator

The clock signal of the A/D converters need to be very accurate to ensure that further calculations are accurate. The demands on the clock signal are:

- Square wave TTL-signal
- Frequency = 12.288 kHz ± 0.09Hz
- \( T_1 = 5 \mu s \)

Crystal oscillators are very suitable for high accuracy clock signals. Unfortunately oscillators at the desired frequency are not available. To obtain the desired frequency a frequency divider should be used. Several are available, but IC 74HCT4059 is ideal because all natural dividers can be implemented. A suitable crystal oscillator is 19.6608 MHz, which can be divided by 1600, which results in 12288 Hz.

The duty cycle however is 50%. To reach the demands this should be adapted by a one-shot multivibrator 74HCT4538. The low time is adaptable with a RC-combination and can be calculated using \( 0.7 \times R \times C \). C is chosen 680 pF, so R should be 5147 \( \Omega \). To adjust the timing, a potential meter of 10K is used.

Figure 7.3 The 12.288 kHz oscillator
Appendix

The calculation of the FFT constants

B.1 Hanning Window factors

The Hanning window is calculated using Formula B.1.

\[ h(n) = 0.5 + 0.5 \cdot \cos(2\pi n/N) \]  \hspace{1cm} (B.1)

The Quick Basic program listed below will calculate these factors and will generate a table which contains all these factors. The table can be copied into the TMS320 program.

```
' ***************************************************************
' ** HAMMING.BAS  Written in Quick Basic
' ** D.J. Aalberts  april 1995
' ***************************************************************
'** calculates the Hamming Factors for a N-point FFT
'** for use with the FLL/PLL program for the TMS320C25
'** The file generated by this program has to be copied
'** FLL/PLL software
' ***************************************************************
'** initialization
' INPUT "N=", N
' teller = 0
' PI = 3.141592654#
'** Open file Hamming.dat
' OPEN "hamming.dat" FOR OUTPUT AS 1
'** calculation of N Hamming Factors
' PRINT "Calculation started"
' PRINT #1, "".word ";
' FOR i = (-N / 2) TO (N / 2 - 1)
'   h = .5 + .5 * COS(2 * PI * (i + .5) / N)
'   h = INT(h * ((2 ^ 15) - 1))
'   IF teller = 10 THEN PRINT #1, : teller = 0: PRINT #1, "", .word
'   PRINT #1, USING "####"; h;
'   IF teller <= 9 THEN PRINT #1, ","; : PRINT ".";
'   teller = teller + 1
' NEXT i
' PRINT "Calculation finished"
'** end of calculation, Close file
' CLOSE (1)
```
B.2 Twiddle factors

To calculate the twiddle factors the following formula is used:

\[ W_N^k = e^{-j2\pi k/N} = \cos(2\pi k/N) + j\sin(2\pi k/N) \]  
\( (B.2) \)

This formula is implemented in the Quick Basic program listed below. It generates a table TWIDDLE.DAT which can be copied into the software. The tables are filled with N/2 real and imaginary parts of equation B.1.

```
'*******************************
   * TWIDDLE.BAS   Written in Quick Basic
   * D.J. Aalberts  April 1995
   * calculates the Twiddle Factors for a N-point FFT
   * for use with the FLL/PLL program for the TMS320C25
   * The file generated by this program has to be copied
   * into the FLL/PLL software
'*******************************

'*** initialization
PI = 3.1415927#
INPUT "N =", N
Q15 = (2 ^ 15) - 1

'*** Open file Hanning.dat
OPEN "TWIDDLE.DAT" FOR OUTPUT AS 1

'*** calculation of N Twiddle factors
PRINT #1, " .word ";
FOR k = 0 TO N / 2
   X = 2 * PI * (k / N)
   RE = INT(COS(X) * Q15 + .5)
   IM = INT(SIN(X) * Q15 + .5)
   PRINT #1, USING "#######"; RE, " ";
   PRINT #1, USING "#######"; IM;
   IF INT((k + 1) / 5) = (k + 1) / 5 THEN
      PRINT #1, : PRINT #1, " .word ";
   ELSE PRINT #1, ", ";
NEXT k

'*** End of calculation, close file
CLOSE (1)
```
Appendix

The (sub)sampling theorem

Analog signals have to be sampled at certain moments to convert these signals to a digital value. The Nyquist theorem states that a signal with frequency $f$ should be sampled with a sample frequency $2f$. Signals with a frequency spectrum with frequency components higher then $f$ should be filtered by a lowpass filter. If the sample rate is lower than $2f$, the higher frequency components will return in the baseband. This effect is called aliasing.

If a signal is bandlimited, sampling at a lower frequency as stated by the Nyquist theorem is possible. This is called subsampling.

$$\frac{mf_s}{2} < f < \frac{(m+1)f_s}{2}, \quad m \in \mathbb{N}$$  \hspace{1cm} (B.1)

The principle of aliasing is used in the subsampling theorem. The bandpassed signal is copied to the baseband.

![Diagram of subsampling at 12.288 kHz of the 124.952 kHz signal](image)

The subsampling theorem is used in the FLL/PLL hardware. A carrier at 124.516 kHz is sampled at 12.288 kHz. The carrier will appear in the baseband at 1636 Hz. Attention should be payed to the fact that a shift of 100 Hz of the carrier will result in a shift of the subsampled signal of 100Hz. This principle is used in the Frequency Tracking Algorithm.
A program for the TMS320C25 consists of a file with extension .ASM contains the code and a file with extension .CMD is needed by the compiler to place all data block in the appropriate memory blocks.
تاريخ: 28/12/1985

---

/**
   * Linker Command File
   */

---

TMSPRG.OBJ
TABLES.OBJ
-c TMSPRG.out
-m TMSPRG.MAP

MEMORY

{ PAGE 0 : VECTORS : origin = 00H, length = 020H
  CODE : origin = 020H, length = 03F00H
  ProgB0 : ORIGIN = OFF00H, LENGTH = 100H

  PAGE 1 : RAMB2 : origin = 060H, length = 020H
  RAMB1P6 : origin = 0300H, length = 080H
  RAMB1P7 : origin = 0380H, length = 080H
  PARRAM : origin = 400H, length = 700H
  FFTRAM : origin = 1000H, length = 1000H
}

SECTIONS

{ vectors : > VECTORS PAGE = 0
  .text : > CODE PAGE = 0
  Strings : > CODE PAGE = 0
  tables : > CODE PAGE = 0
  
  { TABLES.OBJ
    ProgB0 : LOAD = CODE RUN=OFF00H
    page6 : > RAMB1P6 PAGE = 1
    plIRAM : > RAMB1P7 PAGE = 1
    global : > RAMB2 PAGE = 1
    parRAM : > PARRAM PAGE = 1
    fftram : > FFTRAM PAGE = 1
  }
}
The PLL functionality is added but does not function.

The PLL functionality is activated and the FLL still functions.

The necessary tables are separated from the main program.

All delay lines are added.

Two delay lines are added.

Delay = -1 used.

V PLL05, by Daniel Aalberts, November 1995

Lock detection and Reset functionality for the PLL is added.

The PLL functionality is added but does not function.

The FLL software still functions.

The PLL functionality is added but does not function.

The PLL functionality is added but does not function.

Two delay lines are added.

Delay = -1 used.

The PLL functionality is added but does not function.

The PLL functionality is added but does not function.

The PLL functionality is added but does not function.

The PLL functionality is added but does not function.

Two delay lines are added.

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The PLL functionality is added but does not function.

The PLL functionality is added but does not function.

Two delay lines are added.

Delay = -1 used.
PLL variables

CD .usect "page6",1 ;del co-polar sample after bandp filter
HD .usect "page6",1 ;del x-polar sample after bandp filter
BD .usect "page6",1 ;co-polar with hilb delay
CSHIFT .usect "page6",1 ;90 degrees shift co-polar sample after bpf
CDelay .usect "page6",1 ;delay line counter
MIDelay .usect "page6",1 ;Hilbert delay line counter
NUMER .usect "page6",1 ;numerator
DENOM .usect "page6",1 ;denominator
QUOT .usect "page6",1 ;quotient
PHI0 .usect "page6",1 ;primary phase
PHI1IN .usect "page6",1 ;PLL input phase
PHIVO .usect "page6",1 ;VCO output phase
PHIBID .usect "page6",1 ;phase detector output phase
PHILF .usect "page6",1 ;loop filter output phase
C0 .usect "page6",1 ;VCO restfreq phase accumulation constant
C1 .usect "page6",1 ;loop filter coefficient 1
C2 .usect "page6",1 ;loop filter coefficient 2
LFREGL .usect "page6",1 ;loop filter register (bit 0-15)
LFREGH .usect "page6",1 ;loop filter register (bit 16-31)
VCOREG .usect "page6",1 ;VCO phase accumulator
INDEX .usect "page6",1 ;sine table index
IDENOD .usect "page6",1 ;in-phase demodulation signal
QDENOD .usect "page6",1 ;quadrature demodulation signal
CI .usect "page6",1 ;co-polar in-phase demodulated signal
CQ .usect "page6",1 ;co-polar quadrature demodulated signal
XI .usect "page6",1 ;x-polar in-phase demodulated signal
XQ .usect "page6",1 ;x-polar quadrature demodulated signal
ONE .usect "page6",1 ;First Adress Co-polar Delay Line
SIGNEXT .usect "page6",1 ;First Adress Co-polar Delay Line
AR4BCUP .usect "page6",1 ;First Adress X-Polar delay Line?
AR6BCUP .usect "page6",1 ;Adres variabele die naar de DA
DVAR .usect "page6",1 ;; converter gescheurd worde
INV_FLAG .usect "page6",1 ;flag: 0 DA signal niet complement
SHIFT .usect "page6",1 ;beapel AF DA converter
AR0BCUP .usect "page6",1 ;tijdelijk copy ARO
LOCKSTAT .usect "page6",1 ;lock status
LOCKCTR .usect "page6",1 ;lock counter Counts number of samples
LOCKCRIT .usect "page6",1 ;lock criterium
DELAYC .usect "pllRAM",NDelay ;Co-polar delay line buffer
DELAYX .usect "pllRAM",NDelay ;X-polar delay line buffer
DELAYY .usect "pllRAM",NDelay ;Co-polar delay line buffer
HTFONCH .usect "pllRAM",LHILB ;Hilbert coefficient onchip RAM

Post detection LP variables, Memory Req: 50h

*** Hilbert delay line counter
** co-polar with hilb delay
\* delay line counter
** Hilbert delay line counter
** numerator
** denominator
** quotient
\* primary phase
** PLL input phase
** VCO output phase
** phase detector output phase
** loop filter output phase
** VCO restfreq phase accumulation constant
** loop filter coefficient 1
** loop filter coefficient 2
** loop filter register (bit 0-15)
** loop filter register (bit 16-31)
** VCO phase accumulator
** sine table index
** in-phase demodulation signal
** quadrature demodulation signal
** co-polar in-phase demodulated signal
** co-polar quadrature demodulated signal
** x-polar in-phase demodulated signal
** x-polar quadrature demodulated signal
** First Adress Co-polar Delay Line
** First Adress Co-polar Delay Line
** First Adress X-Polar delay Line?
; converter gescheurd worde
\* flag: 0 DA signal niet complement
; beapel AF DA converter
; tijdelijk copy ARO
** lock status
** lock counter Counts number of samples
** lock criterium
; Co-polar delay line buffer
; x-polar delay line buffer
; Co-polar delay line buffer
; Hilbert coefficient onchip RAM

Post detection LP variables, Memory Req: 50h
PARInt A  SMACRO
SA2L  TRDATA
RPTK  255
NOP
OUT  TRDATA, PA1
RPTK  255
NOP
ORK  10h
SACL  TRDATA
ZAC
; Set flag that timer int is not timeout
SACL  TimeOut
LPDK  0
LALK  WAITLEN
SACL  TIM
LAC  1 MR
ORK  8
SACL  1 MR
LPDK  6
SEND

LptWrtC  SMACRO
Char
LACK  ;Char:
LptWrtA  SEND

- --------- CopyStrToAnswer -------
* This macro copies a measurement reply message for PC in AnswerStr
   with format as used by Transmit. Possible messages are defined in
   "Strings" and must have an even length. Afterwards Temp2 contains the
   string length so far.

COPYSTRTOANSWerp MACRO  p_answer
LARP  AR7
MAR
*+
; Send 4 bit character (called a nibble) to the parallel communications
; port 1h. If the character is selected by the PC then the character will be put
; on the centronics bus. Both macros perform the same, however:
;  - LptWrtA expects the character to be already in the accumulator
;  - LptWrtC expects a constant as parameter
; On the centronics bus. Both macros perform the same, however:
;  - LptWrtA expects the character to be already in the accumulator
;  - LptWrtC expects a constant as parameter

- ------ LptWrtC, LptWrtA -----------
* The interrupt service routine PARlnt2
* Sends 4 bit character (called a nibble) to the parallel communications output
* port 1h. If the character is selected by the PC then the character will be put
* on the centronics bus. Both macros perform the same, however:
;  - LptWrtA expects the character to be already in the accumulator
;  - LptWrtC expects a constant as parameter

- This macro is used in:
;  - the interrupt service routine PARlnt2
;  - the procedure Transmit
; In Transmit it only sends the SOH to start a communication. The macro does not
; yet generate an IRQ-on-PC signal. However it does load the data nibble with bit
; 4 in IRQ-on-PC set to 1 (active) into the variable TRDATA and it (re)enables/
; starts the timer with period WAITLEN = 1875 = 0.15 msec (at 12.5 MIPS). Later
; on after the first deliberate timeout of the timer TRDATA is then put to the
; centronics bus and will generate IRQ-on-PC. The period WAITLEN is probably
; necessary to ensure that the data nibble signal values have settled on the
; centronics bus. (See also the PARlnt interrupt service routine.)
Hello.string 0, 41, "Hello, I am a digital PLL/PLL version 1.2"
NotInitStr.string 0, 40, "This function is not initiated by the PC"
Spectrum.string 0, 6, "Data2 *"
UnknwCmdStr.string 0, 15, "Unknown command"
Ok.string 0, 2, "Ok"
.label ParStrEnd

; Parallel port possible answers

*** Parallel port possible answers ***
Hello.string 0, 41, "Hello, I am a digital PLL/PLL version 1.2"
NotInitStr.string 0, 40, "This function is not initiated by the PC"
Spectrum.string 0, 6, "Data2 *"
UnknwCmdStr.string 0, 15, "Unknown command"
Ok.string 0, 2, "Ok"
.label ParStrEnd

; Initialize stackpointer

LRK AR7,07CH ;use block 82 as stack, top address 7CH

; Initialize interrupt mask register

LAC IMR ;copy register to accumulator
LALK 0FFC5H ;reset bit 0, 1, 2, 3, 4 and 5
LDPK 6 ;datapage = 6

; Initialize FFT constants

ZAC SACL LALK FFTPOWER ;set power of FFT to FFTPOWER
LALK 1;FFTPOWER ;calculate number of FFT input points N
SACL N ;and store
SLK 1;calculate N-1
SACL NMIN1 ;and store
ADLK 1;back to N
SFR ;calculate N/2
SAACL NBY2 ;and store
SBLK 1 ;calculate N/2-1
SAACL NBY2MIN1 ;and store
LALK 6 ;FFT size
LAC SACL FFTCOUNT

; Initialize auxiliary registers

LRLK AR1,FFTBANK0 ;initialize input bank start address and FFT bank start address
SAR AR1,AR1BCUCK

; Initialize input bank start address and FFT bank start address

ZAC SACL INFBNK
LALK FFBANK1 ;set FFT bank start addr to FFBANK1
SaACL FFBANK

; Initialize PLL

ZAC SACL VCOREG ;reset VCO register
SACL PHIVCO ;reset PHIVCO
SACL HDELAY ;reset delay line counter
SACL CDELAY ;reset delay line counter
LALK 2C2Bh ;load CO with VCO rest frequency phase
SAACL C0 ;accumulate: 16384 (90 degrees)
SAACL C1 ;accumulation: 16384 (90 degrees)
LALK 171 ;load C1 with 15 and C2 with 1422 for
SAACL C2 ;zeta = 0.707 and Bn = 100 Hz
LALK 1738 ;AR5BCKUP = 1st addr of co-p delay line
SAACL C2 ;AR5BCKUP = 1st addr of co-p delay line
initialize LPF

LALK AR1BCKUP ; AR1BCKUP = 1st addr of X-P delay line
LACL AR1
LACL AR4BCKUP ; AR4BCKUP = 1st addr of co-P delay line
LACL AR4
LACL AR560h ;init address of the variable used for D/A
SACL AR560h
SACL AR56h
SACL AR6BCKUP
SACL LOCKCRIT ; init lockcriterion

* Initialize LPF

LALK 1 ; load ONE with 1
SACL ONE
SACL ONE,15 ; load SIGNEXT with 8000h
SACL SIGNEXT
LARP 1
LRLK AR1,FltOutRam ; Copy filt data from ROM to FltDataRam
RPTK FltDataEnd-FltData
BLKP FltData,*

* Initialize bit reversal counter

ZAC
SACL EVENODD ; reset even/odd counter

Copy IntO routine into Block 130
Hilbert coefficients in Block 20
X Samples are sorted out and stored in the appropriate BANK for FFT
X and C samples are stored in a delay line.

* Initialize parallel port

LARP AR1
LRLK AR1, RAMB0
RPTK PrgLenBlkJ0
BLKP PrgBjBlkJ0,**
LRLK AR1, HILBERT
RPTK LHILB
BLKP HILCOF,**
CNFP

* Initialize parallel port

LRLK AR1.PAR_Answer ; copy parallel port strings
RPTK ParStrEnd-ParStrBeg-1
BLKP ParStrBeg,**
IN VAR1, PA1 ; Clear par port D:PF and read dippw
LD PK 0
LALK TMOUTL
SACL TIM
SACL PRD
LD PK 6

* End of initialize

EINT ; interrupts allowed

* Main program

MAIN
ZALS DATRDY
BZ MAIN_L1 ; FFT memory bank full?

ZAC
SACL DATRDY ; reset data already
CALL FFT
CALL POWER ; Calc Power spectrum
CALL OUTPUT ; VCO control output routine
MAIN_L1 LAC PAR.Status-2 ; par command received?
SUBK 2
BZ PARCommand
B MAIN

* Interrupt routines

* Interrupt 0
* New sample is arrived?
* This block is placed in internal RAM, address FF00
* X samples are sorted out and stored in the appropriate BANK for FFT
* X and C samples are stored in a delay line.

 sect "ProcB0"
.label PrgBgBlk0
SaveEnv ; save STO, ST1, P, T in Acc registers
LDPK 6 ; select data page 6
SSEX
IN C, PA1 ; get sample from receive register
IN X, PA0 ; get sample from receive register
LAC C, 2 ; scale input samples to 015 and store in C
SACL CBP ; for X BandPass
LAC X, 2 ;
SACL XBP ;
CALL DELAY ; Delay routine IN: CBP.XBP
CALL WAITH ; Hilbert Transf on CBP.OUT: CHSHIFT
CALL PHASE ; IN HD, CHSHIFT OUT: PHIN
CALL Pll ; IN PHIN, PHIVCO OUT: PHIVCO
CALL DECOD ; Demodulation routine bereken I en Q
/ CALL LPFDEC ; Roep de filters van I en Q aan.
CALL OUTPUTZ ; universele uitslagen routine
LAC EVNODD ; Determine if this is even or odd sample
BZ EVEN ; If EVENODD equals 1 (ODD), EVENODD is reset and
O DD SUBK 1 ; then the procedure stops; if EVENODD equals
SACL EVNODD ; 0 (EVEN), EVENODD is incremented and the procedure

RestoreEnv ; continues, At this way the samplefreq is halved.

 sect "ProcE0"
.label PrgBgBlk0
EVEN
LALK 7 ; 12288 Hz / 4 = 3072 Hz
SACL EVNODD ;
LARP ARl,AR1BCKUP ; set register AR1
LARP AR1 ; AR1 is active
SSEX
LAC NOSAMP ; copy NOSAMP to accumulator and
LAC RAMWIN ; add start address of Nanning table
TBLR HANCOP ; read Hanning coefficient
LT CBP ; copy SAMPLE into T register
MY HANCOP ; multiply by HANCOP (result in P register)
SIM 1 ; Set product Mode
where new

1995 PLLOS.ASM

FfTBNK branch to BLFULL branch BOFULL

RetryTrm for transmitting data to the PC (via PAR_Status~O

from the parallel communications port. For

Recei veCount 1?

2. received data bytes must be stored (PAR RecData) or where the to

and PAR L2.). During a transmission

order Byte of CRC received

m

new

with

FFT

calc

with

INT2

input sample storage address

PAR :m

calculated CRC

Ll

new

with

SPH

"*

;Store P-register (HANCOF * CBP) in the current

SPM 0 ;storage adress | Real Part| and reset the

ZAC ;imaginary part in the next storage adress

SLC

;-

LAR AR0, N ;set register AR0 to N

ADD "BRO+ ;calculate new input sample storage address

SAR AR1, AR1BCUP

LAC NOSAMP ;increment NOSAMP by 1

ADDX 1

SAI NOSAMP ;and store result

SUB N ;check if N samples collected

RSAM ;enable sign extension

BGEZ BLFULL ;N samples collected -> branch to BLFULL

RestoreEnv ;n Samples collected, return to main

BLFULL ZALS INPBNK ;determine current input bank

BGFULL ;current input bank is bank 0 -> branch BFULL

ZAC

SACL NOSAMP ;reset NOSAMP (number of samples collected)

SACL INPBNK ;set current input bank to 0

LARK ARl, FFTBankl ;set FFT bank start address to FFTBankl

SAI FFTBnk

LRLK ARl, FFTBank0 ;set input bank start address to FFTBank0

SAR ARl, ARlBCUP

RestoreEnv ;return to main program

BFULL SACL NOSAMP ;reset NOSAMP (number of samples collected)

LACK 1 ;

SAI DARYD to 1 -> calc new FFT

LACK INPBNK ;set current input bank to 0

LARK ARl, FFTBank0 ;set FFT bank start address to FFTBank0

SACL FFTBnk

LRLK ARl, FFTBankl ;set input bank start address to FFTBankl

SAR ARl, ARlBCUP

RestoreEnv ;return to main program

.LABEL PrgEndBlk0 .SET PrgEndBlk0 - PrgBgBlk0 - 1 .TEXT

**********************************************************************

PAR_interupt interrupt service routine (INT2)----------

interrupt service routine for INT2 from the parallel communications port. For

a description of the protocol and data formats see [2].

- PAR L2 up to PAR L9 for receiving data from the PC. At the first beginning

PAR_Status = 0, during the receiving PAR_Status = 1 and after the

successfull completion PAR_Status = 2.

- PAR L10 up to PAR L14 and PAR HerryMn for transmitting data to the PC (via

conditional branches in PAR LI and PAR L2.). During a transmission

PAR_Status = 3 as initialized by the routine Transmit. After completion,

whether successfull or not. PAR_Status = 0.

Auxiliary registers:

- AR points to AR1 in the whole routine

- ARI is restored by means of AR1VAR and contains the current position in the

area where received data bytes must be stored (PAR RecData) or where the to

be transmitted data nibbles are stored (PAR_TrmData).

PAR Interrupt SaveEnv

LDPIK 0 ;Switch to register space

LAC IMR

ANDX OFFFFh ;Disable parallel port and timer interrupt

SAI IMR

EINT ;Enable DCF en ADC int dep on prev status

LDPIK 6

LARP AR1

LAR AR1, AR1VAR ;Restore AR1

IN DATA, PA2 ;Get Data from parallel interface

PAR Interrupt ParRecCnt

LAC DATA

ANDX OFFFFh

SACL DATA

BLNZ PAR L1

LAC PAR_ZeroCnt

ADDX 1

SACL PAR_ZeroCnt

SUBL 5

BLZ PAR L2

SACL PAR_ZeroCnt

SACI PAR_Status = 0

SACI PAR_Status = 0

PAR L1 ZAC

SACL PAR_ZeroCnt

PAR L2 ZAC

PAR_Status = 0

PAR L2 DATA ;Get last received character

PAR EndOfInt

PAR_L6 SUBK 1 ;ReceiveCount = 17

PAR L10 PAR RecCnt

SUBK 1 ;ReceiveCount = 17

PAR L4 PAR L4

CALL CRCVal

LAC DATA

SACL PAR RecLen

LACK 2

SACL PAR RecCnt

LptWrtC DACk

B PAR EndOfInt

PAR L3 SUBK 1

PAR L10 PAR_RecCnt

PAR_RecCnt

PAR EndOfInt

PAR L5 PAR_CRC ;Initialize CRC

LRLK AR1, PAR_RecData ;Initialize receive buffer pointer

LptWrtC DACk

B PAR EndOfInt

PAR L4 ADDX 3

PAR RecLen

SUB PAR RecLen

BGEZ PAR L5

CALL CRCVal

LAC DATA

SACL PAR RecLen

LACK DACk

B PAR EndOfInt

PAR L6 SUBK 1

PAR L10 PAR_RecCnt

ADDX 1

SACL PAR_RecCnt

LptWrtC DACk

B PAR EndOfInt
**A message is send by the this device to the PC**

- **LAC PAR TrmCnt**; Status = 3
- **SUBK 1**
- **BNZ PAR_L12**; TransmitCount = 1
- **LAC DATA**
- **SUBK AACK**
- **LAC PAR_L11**; Is the received character AACK
- **ADDX 1**
- **SACL PAR_TrmCnt**
- **LARP AR1**
- **LRLK AR1, PAR_TrmData+1**
- **LAC**
- **LptWrtC B**
- **PAR_EndOfInt**

**PAR_L11**

- **MAR**
- **ADDX 1**
- **SACL**
- **SUBK 3**
- **PAR_RetryTrm**
- **LptWrtC SOK**
- **B**
- **PAR_EndOfInt**

**PAR_L12**

- **ADDX 1**
- **SUB Par_TrmLen**
- **BEZ PAR_L13**
- **TransmitCount < TransmitLen**
- **LAC DATA 1**
- **SUBK DACK**
- **BNZ PAR_RetryTrm**
- **LAC PAR_TrmCnt**
- **ADDX 1**
- **SACL PAR_TrmCnt**
- **LAC**
- **LptWrtA**
- **B**
- **PAR_EndOfInt**

**PAR_L13**

- **LAC DATA**
- **SUBK EOT**
- **BNZ PAR_RetryTrm**
- **if data = NACK the CRC not accepted.**
- **No EOT received, any case retry!**
- **ZAC**
- **SACL PAR_Status**
- **LptWrtC EOT**
- **B**
- **PAR_EndOfInt**

---

**Timer**

- **SaveEnv**; Interrupt: Timeout occurred
  - **LDPK 6**
  - **LAC Timeout**
  - **BNZ TI_L1**
  - **OUT TRDATA, PA1**; Generate interrupt on acquis computer
  - **LACK 1**
  - **SACL TimeOut**
  - **LAC PAR_Status**
  - **LDPK 0**
  - **BNZ TI_L0**
  - **LAC IMR**
  - **ADRK OFFF7h**; Disable Time-out interrupt
  - **SACL IMR**
  - **RestoreEnv**
  - **TI_L0**
    - **LALK TMOUTL**
    - **SACL TIM**
  - **RestoreEnv**
  - **TI_L1**
    - **LDPK 0**
    - **LAC IMR**
    - **ADRK OFFF7h**; Disable Time-out interrupt
    - **SACL IMR**
    - **LDPK 6**
Performs in-place N-point radix-2 DIT FFT transform
* Data should be in bit-reversed order

**FFT**

- Performs in-place N-point radix-2 DIT FFT transform
- Data should be in bit-reversed order

---

**Subroutines**

---

**ENDFFT**

---

**Theory, Algorithms and implementations, Texas Instruments, page 77
Article: Implementation of FFT algorithms with the TMS32020.
Writers: Panos Papamichalis, John So
POWER

ZAC \( \text{MAXPOW} \) ; reset MAXPOW
ZAC \( \text{MAXPOW} \) ; reset MAXPOW
LAC \( \text{FPTBANK} \); calc start address for power calculations
LAC \( \text{Temp1} \);
LAR AR4,Temp1;
set AR4 to start address for power calc
LAR AR1,MBY2MINI;
set AR1 to N/2-1
LRLK AR5,FPTBANK2;
set AR5 to power spectrum storage address
LARP AR4;
\( 
\text{AR4} \) is active

POWBIN
ZAC \( \text{ACC}=0 \)
MPYK 0; \( P=0 \)
SQR 1; \( P=\text{Re}^2 \), \( \text{ACC}+P=0 \)
SQR 1.5; \( P=\text{Im}^2 \), \( \text{ACC}+P=\text{Re}^2 \)
APAC; \( \text{ACC}+P=\text{Re}^2+\text{Im}^2 \)
SAC \( * \); store calculated power of bin in FPTBANK2
SACL \( * \);

ADJPOW
SBRK 2
SAR AR5,MAXPADR; store maximum power bin address
LAC \( * \);
SACL \( \text{MAXPOW} \); store new maximum power high
LAC \( * \);
SACL \( \text{MAXPOW} \); store new maximum power low

NEXTBIN
LARP AR1;
\( \text{BANZ POWBIN,*.4} \); if power of < N/2 bins calc \( \rightarrow \) to POWBIN
ENDPOW
LAC \( \text{MAXPADR} \); load accumulator with max power bin addr
SBLK FPTBANK2; calculate offset from DC
SFR; divide by 2
SACL FREQOFFS; and store result in FREQOFFS
SACL FREQUENCY; and store result in FREQUENCY
RET; back to main

HTFILT (Hilbert transform filter)
- This FIR Hilbert transformer uses on-chip memory block B0 (page 5:
  * 2E7-2FF) for storage of delay-tap data samples. The newest input
  * should be in memory location CBP when called.

HTFILT
LDPK 6; get input for hilbert transf
LAC \( \text{CBP} \);
SACL HTPINP;
LARP AR4; \( \text{AR4} \) is active
LARK AR4,RAM31END-1; \( \text{1 extra word for shift.} \)
MPPY 0; \( \text{reset P-register} \)
ZAC; \( \text{reset accumulator} \)
RPTK LHILB-1; \( \text{repeat [LHILB-1] times} \)
MACD OPD00H+HILBRT,\( * \); \( \text{mult/acc, shift data word in} \)
LAC \( \text{block B1 and decrement AR4} \)
APAC; \( \text{final addition of P-reg to acc} \)
LDPK 6; \( \text{scale and store y[n]} \)
SACH CSHEIF,1; scale and store y[n]
RET

* This routine determines the co-polar signal argument. The newest input
samples should be in memory locations CD and CSHIFT. The calculated
argument will be in memory location PHI IN. The procedure uses the
variable PHIO as temporary helpvariable.

```
; Increment Lockcounter
; calc phase diff between phase
detektor inp phase & VCO outp phase
; store result into PHIERR
; copy loop filter register into acc
; add C2 * PHIERR to loop filter reg
; store result into LFREGL and LFREGHI
; copy VCOREG into PHIVCO
; store result back into VCOREG
; add CO and PHILF to VCOREG and
If Lock>lockcrit then out of lock
; Subtract criterium >0 Out of Lock
; set XF
; If Lockcounter < 1024 then cont
; If lock-lowcrit then out of lock
; Subtract criterium Out of Lock
; set XF
; reset xf
; LACK 0
; reset PLL and all variables
; LACK PHIFER
; SAACL PHIFER
; SAACL LFREGL
; SAACL LFREGHI
; SAACL VCOREG
; SAACL PHIVCO
; LACK LockCTR
; LACK Lock
; LACK -32768
; LACK PHIIN
; OUT LOCK
; EINDE
```

DEMOD

```
; Demodulate the I Q signals from the delayed input lines

DEMOD
```

PHASE

```
MPXK 0 ; set P-register op 0
SORA HD ; P-register = CD'2
SORA CSHIFT ; P-register = CSHIFT'2, Acc = CD'2
SQR S CSHIFT ; Accumulator = CD'2 - CSHIFT'2
BOS PHI45 ; branch if [CD] > [CSHIFT]
ZAC LAC CSHIFT ; align denominator
ABS SAACL DENOM ; align numerator
RPTK 14 ; 15-cycle divide loop
SUBC DENOM ; store quotient
SAACL QUOT ; store quotient
ZALS QUOT ; divide by 4
SFR ADLK ATWNB ; add start address of arctangent table
TBLR PHI0 ; read phase and
B QUAD ; 1, Phi45
ZAC LAC HD ; align denominator
ABS SAACL DENOM ; align numerator
RPTK 14 ; 15-cycle divide loop
SUBC DENOM ; store quotient
RSXM ZALS QUOT ; divide by 4
SFR ADLK ATWNB ; add start address of arctangent table
TBLR PHI0 ; read phase and
B QUAD ; 1, Phi45
```

PHI45

```
LAC PHIIN ; calce phase argument
SUB PHIVCO ; detektor inp phase & VCO outp phase
SAACL PHIFER ; and store result into PHIFER
ZALK LFREGL ; copy loop filter register into acc
LT C2 ; copy C2 into T-register
MPY PHIFER ; multiply PHIFER by C2 -> P-reg
LT C1 ; copy C1 into T-register
MPYA PHIFER ; multiply PHIFER by C1 -> P-reg
add C2 * PHIERR to accumulator
; store 16 MSBs into PHILF
; copy VCOREG into PHIVCO
; store result back into VCOREG
; Increment Lockcounter
```

PLL

```
This routine generates an argument PHIVCO that is phase locked to the
input signal argument PHIIN.

PLL:
```
PLL05.ASM

```
SUML PHIIN
; detektor inp phase & VCO outp phase
; and store result into PHIFER
```

PLL2

```
LAC PHIFER
; copy VCO output argument into accumulator
```

DEMOD

```
LAC PHIIN
; copy VCO output argument into accumulator
SAACL PHIVCO
; store result back into PHIVCO
LAC LOCKCTR
; Increment Lockcounter
ADDK 1
SAACL LOCKCTR
```

DEMOD

```
SUML 1024 ; If Lockcounter < 1024 then cont
BLZ LOCK_1 ;
EALS Lock ; If lock-lowcrit then out of lock
SUB LOCKCRIT ; Subtract criterium
BLZ OUT_LOCK ; If lock-lowcrit then out of lock
RXF B LOCK_2 ; set XF
LACK -32768
; LACK PHIIN
```

DEMOD

```
SUML 1024 ; If Lockcounter < 1024 then cont
BLZ LOCK_1 ;
EALS Lock ; If lock-lowcrit then out of lock
SUB LOCKCRIT ; Subtract criterium
BLZ OUT_LOCK ; If lock-lowcrit then out of lock
RXF B LOCK_2 ; set XF
LACK 0
; reset PLL and all variables
LACK PHIFER
; LACK PHIFER
```

DEMOD

```
SUML 1024 ; If Lockcounter < 1024 then cont
BLZ LOCK_1 ;
EALS Lock ; If lock-lowcrit then out of lock
SUB LOCKCRIT ; Subtract criterium
BLZ OUT_LOCK ; If lock-lowcrit then out of lock
RXF B LOCK_2 ; set XF
LACK 0
; reset Lockctr, lock
```
PL05.ASM

BIT PHIVCO, 0 ; test sign-bit of PHIVCO
BBZ DEMOD2 ; if PHIVCO > 0, branch to DEMOD2
BIT PHIVCO, 1 ; test if PHIVCO < 16384
BBNZ DEMOD1 ; if -16384 < PHIVCO < 0, branch to DEMOD1
ANDX 16383 ; calc index of sine table (delete 2 MSBs)
SACL INDEX ; and store result in INDEX
SFR ; Acc := Acc / 2; Shift Right
SFR ;
ADLK SINTBL ; add start address of sine table
TBLR IDEMOD ; read sine
LAC IDEMOD ; calculate in-phase demodulation value and
NEG ; store result in IDEMOD
SACL IDEMOD ; calculate index of sine table
SBLK 16384 ;
NEG ;
SFR ;
ADLK SINTBL ; add start address of sine table
TBLR QDEMOD ; read sine
LAC QDEMOD ; calculate quadrature demodulation value and
NEG ; store result in QDEMOD
SACL QDEMOD ;

DEM0D1:

B DEMOD4 ; branch to DEMOD4
ANDX 16383 ; calc index of sine table (delete 2 MSBs)
SACL INDEX ; and store result in INDEX
SFR ;
ADLK SINTBL ; add start address of sine table
TBLR QDEMOD ; read sine
LAC INDEX ; calculate index of sine table
SBLK 16384 ;
NEG ;
SFR ;
ADLK SINTBL ; add start address of sine table
TBLR IDEMOD ; read sine
LAC IDEMOD ; calculate in-phase demodulation value and
NEG ; store result in IDEMOD
SACL IDEMOD ;

DEM0D2:

BIT PHIVCO, 1 ; test if PHIVCO > 16384
BBNZ DEMOD3 ; if PHIVCO > 16384, branch to DEMOD3
ANDX 16383 ; delete 2 MSBs
SACL INDEX ; and store result in INDEX
SFR ;
ADLK SINTBL ; add start address of sine table
TBLR IDEMOD ; read sine
LAC IDEMOD ; calculate index of sine table
SBLK 16384 ;
NEG ;
SFR ;
ADLK SINTBL ; add start address of sine table
TBLR QDEMOD ; read sine
LAC QDEMOD ; calculate index of sine table
SBLK 16384 ;
NEG ;
SFR ;

DEM0D3:

B DEMOD4 ; branch to DEMOD4
ANDX 16383 ; calc index of sine table (delete 2 MSBs)
SACL INDEX ; and store result in INDEX
SFR ;
ADLK SINTBL ; add start address of sine table
TBLR QDEMOD ; read sine
LAC QDEMOD ; calculate index of sine table
SBLK 16384 ;
NEG ;
SFR ;

OUTPUT:

LAC QDEMOD ; calc quadrature demodulation value and
NEG ; store result in QDEMOD
SACL QDEMOD ;

DEM0D4:

LT CD ; load T-register with CD
MPY IDEMOD ; multiply CD by IDEMOD and store result
MPY CQ ; multiply result by QDEMOD and store result
LT XD ; load T-register with XD
MPY IDEMOD ; multiply XD by IDEMOD and store result
MPY XD ; multiply XD by QDEMOD and store result
RET ;

Output:

OUTPUT LT FREQOFFS ; calculate frequency
MPVR 32
LACK 5000H
APAC
SACL Temp1 ; and store result in FREQUENCY
OUT Temp1, PA2
RET

* This routine sends the value of the var in adress DAVAR to
* the da converter, it can be complemented by the var inv_flag
* 0 will pass the complement part

OUTPUT2 SAR AR0, AR0BCUP ; save contents of AR0
LAR 0, DAVAR ; load AR0 with DAVAR (adress of var)
LARP AR0
LAC INV_FLAG ; need a complement?
BZ COMPL_NOT
LAC * ; calculate complement
B WRT
COMPL_NOT LAC * ; calculate complement
WRT XORK 0800H ; Temp 1 to D/A converter
OUT Temp1, PA2 ; store variable in temp1
LAR AR0, AR0BCUP ; restore AR0
RET

* PARALLEL COMMUNICATIONS PORT ROUTINES
* - Transmit
* - PARCommand
This procedure starts a packet transmission from the device to the PC by sending the first character SOH (a nibble). After that the whole rest of the parallel port communication is interrupt driven and taken care of by the interrupt service routines PAR_Init2 and PAR_Tint. First however Transmit fills the packet array PAR_TrmData with an answer string.

Auxiliary registers:
- AR0 points to the to be transmitted answer string of which the first word contains the length in bytes, see for examples the section "Strings".
- AR1 points to the packet storage area PAR_TrmData.
- AR2 counts the number of bytes in the string.

Labels:
- TRM_L1 and TRM_L2 from a loop that stores the high byte, respectively the low byte of the words in the string as nibbles (two nibbles per byte) in the packet.
- TRM_L3 at the end the SOH is put to port 1h with LptWrtC and PAR_Status=3, PAR_Retry=0, PAR_TrmLen=length of packet, and PAR_TrmCnt=1. It is not necessary to initialise AR1VAR, because this will be done in PAR_L10.

Remark:
When a packet transmission does not complete successfully a retry can be performed in the following cases:
- When another nibble then AACK was received from the PC on SOH, then two additional retries can be done by immediately retransmitting the SOH in PAR_L1. For even more retries the program branches to PAR_RetryTrm.
- The part PAR_RetryTrm restarts a packet transmission over again by making PAR_TrmCnt=1, resetting AR1 and sending the SOH. This can happen up to two times, otherwise PAR_Status=0 and the transmission is aborted.
- PAR_L10, when no correct AACK was received on SOH.
- PAR_L12, when no correct DACK was received on LEN2-0, D1h..Dlen-4h.
- PAR Retransmission part can be branched from:
- PAR_L10, when no correct AACK was received on SOH.
- PAR_L12, when no correct DACK was received on LEN2-0, D1h..Dlen-4h.
- PAR_L13, when no correct EOT was received on Dlen-41.
- Just as PAR_RetryTrm the part TI_RetryTrm in the PAR_Tint interrupt service routine restarts a packet transmission all over again when a time-out of TMOUTL = 5.2 msec (at 12.5 MIPS) occurs. This indicates that no data was received from the PC.

Transmit
-----

LACK 3 ; unselect sign extension mode
SACL PAR_Status
SACL PAR_Retry
SACL PAR_CRC
SACL PAR_ZeroCnt
LARP AR1
LRLK AR1, PAR_TrmData
SACL **, AR0 ; SOH
LAR AR2, **, AR2
MAR **, AR0 ; Repeat count - 1
LAC **, AR1
ADDK 8
SACL PAR_TrmLen
LAC PAR_TrmLen, 8
SACH **
ANDK OPOOh
SACH **, 4
LAC PAR_TrmLen
ANDK OFh
SACL **, AR0
SACH **, OPOOh
ANDK **, 4, AR2
SACH **, 4, AR2 ; Store Low nibble
LAC **, 12, AR1

---

PARCommand
-----

Find subroutine for received command
on entry PAR_ReceData contains received command
if no command is recognized "Unknown command" message is transmitted
if command is recognized AR0 points to character after last checked char in receive buffer
(usefull to retrieve time information following command ID string)

LAR AR4, * ; Number of commands to check
MAR **, AR3
LRLK AR4, CmdCnt + 1 ; First command ID address
Algorithm with lookup table from BYTE magazine November 1987 page 339

**Input**: Start position

**Increment factor**

Frequentie in eerste woord

Hoogste word power 0

Hoogste word power 1

Hoogste word power 2

Hoogste word power 7

Gemiddelde truc

Length of the data

Plus de al bewaarde lengte

Help2aeindpositie

Store Length field

Initialize transmitting

Do nothing, give timer chance to generate last interrupt of "Ok" transmission

**Spectrum out**

*Transmits power density spectrum to PC*

---

**Answers and actions taken on received command**

Reset: complete reboot is executed after transmission of "Ok".

- Same as pressing reset button except that FIFO is not reset by hardware.

- **Help2**: Start position

- **RS_L1**
  - LAC: PAR Status
  - B: 0000h

- **RPTX**: 255

- **LAC**: 300h

---

**CRCVVal**

- **PAR_CRC**, 8: Get old CRC and shift 8 places
- **VAR1**: Save high order byte CRC >> 8
- **VAR2**: Save low order byte CRC << 8
- **VAR1**: Calculate (CRC>>8)"Character"
- **VAR2**: Add offset
- **VAR1**: Save new CRC
- **VAR2**: Save CRC

---

*End program*