Measurement of EVM (Error Vector Magnitude) for 3G Receivers

Master's Thesis

by

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PREFACE

This report describes a thesis project for the International Master Program of Digital Communications Systems and Technology, at Chalmers University of Technology, Gothenburg, Sweden.

The work was done at the Base Station Radio Design Department, TB/RB, at Ericsson Microwave Systems AB, Mölndal, Sweden, from August 2001 to February 2002.

I would like to thank my family for all the support given during my studies, my friends for the good times spent in Sweden, and the people who contributed and helped in the development of this project, specially to my supervisors and examiner:

ABSTRACT

The aim of this thesis work is to measure the Error Vector Magnitude (EVM) directly at the radio receiver of the 3rd Generation WCDMA base stations, specifically at the gamma bus, and the output of the analogue to digital converters. The EVM is a figure-of-merit for the down-conversion of the modulated radio signal and is a measure of the signal to noise and distortion ratio. The error vector is obtained by subtracting the received signal from an ideal reference signal. The set-up consists of generating an RF signal using a signal generator, feeding it into the receiver, and acquiring it from the interfaces of interest using a logic analyzer. The ideal reference signal is created by means of software. The work done in this project includes an algorithm proposal to measure the received signal at the optimal position, this algorithm is based on the eye diagram, specifically, in the analysis of the variance of the signal's magnitude. In addition, another algorithm was developed to compensate for the effects of frequency offsets introduced by the RF local oscillator. The frequency offsets can be visualized as phase variations of the IQ constellation in the time domain, such phase variations were tracked and compensated using a one-tap Least Mean-Square adaptive filter. Both algorithms were implemented in software using Labview, resulting in an EVM_{RMS} of 6.6% at the gamma bus, and 4.5% at the ADCs. The signal measured at the gamma bus was affected by quantization noise, therefore, it was low-pass filtered to minimize the problem.

TABLE OF CONTENTS

Contents

Prepared (also subject responsible if other)		No.		
EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen		
Approved	Checked	Date	Rev	Reference
EMW/TB/RBC Ödgärd Andersson		February 2002	A	

TERMINOLOGY

3GPP	3rd Generation Partnership Project
ADC	Analog to Digital Converter
AGC	Automatic Gain Control
AWGN	Additive White Gaussian Noise
BS	Base Station
BTS	Base Transceiver Station
CDMA	Code Division Multiple Access
CPICH	Common Pilot Channel
DSSS	Direct Sequence Spread Spectrum
DUT	Device Under Test
EVM	Error Vector Magnitude
FDD	Frequency Division Duplex
GSM	Global System for Mobile Communications
IF	Intermediate Frequency
IMT-2000	International Mobile Telephony 2000
ITU	International Telecommunications Union
LMS	Least Mean-Square
LSB	Least Significant Bit
MSB	Most Significant Bit
MSC	Mobile Switching Center
QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency
RRC	Root Raised Cosine Filter
SCH	Synchronization Channel
SIR	Signal-to-Interference Ratio
SMS	Short Message Service
SNDR	Signal to Noise and Distortion Ratio
SS	Spread Spectrum
TDD	Time Division Duplex
TRX	Transceiver
UMTS	Universal Mobile Telecommunication Services
UTRAN	UMTS Terrestrial Radio Access Network
VI	Virtual Instrument
VSA	Vector Signal Analyzer
WBTS	WCDMA – Base Transceiver Station
WCDMA	Wideband CDMA

1 Introduction

This report describes an implementation to measure the Error Vector Magnitude (EVM), at the receiver of the third generation (3G) WCDMA base stations, so-called WBTS. The Error Vector Magnitude is defined in the base station conformance testing technical specifications of the Third Generation Partnership Program (3GPP). It is defined as a figure-of-merit for the transmit modulation, however, the goal of this project is to use the EVM as a figure-of-merit of the 3G base station's receivers. The origin of the project comes from the need to evaluate, in a laboratory environment, the performance of the receivers of the transceiver boards (TRX). In this case there are two interfaces which are of our interest, the gamma bus and the analogue to digital converters (ADCs). The gamma bus is used for the transfer of down-link and up-link user data. The ADCs interface the analogue TX/RX and the FPGAs.

EVM provides an insight into the signal's quality that other performance measurements such as the eye diagram or BER measurements do not cover. One of the advantages is the simplicity of the measurement set-up, since there is no need for an entire communication system. It possesses a direct relation with the signal to noise and distortion ratio (SNDR) and can be used to determine the physical error introduced at different stages of a communication system, helping the designer to troubleshoot specific problems. The measurement is done using the down-converted digitally modulated radio signal.

1.1 EVM Definition

3GPP standards provide the following definition of EVM: "The Error Vector Magnitude is a measure of the difference between the reference waveform and the measured waveform. This difference is called the error vector. Both waveforms pass through a matched Root Raised Cosine filter with bandwidth 3.84 MHz and roll-off α =0.22. Both waveforms are then further modified by selecting the frequency, absolute phase, absolute amplitude and chip clock timing so as to minimize the error vector. The EVM result is defined as the square root of the ratio of the mean error vector power to the mean reference power expressed as a %. The measurement interval is one timeslot as defined by the C-PICH (when present) otherwise the measurement interval is one timeslot starting with the beginning of the SCH. The requirement is valid over the total power dynamic range". [2]

Figure 1-1 shows the error vector and its components.

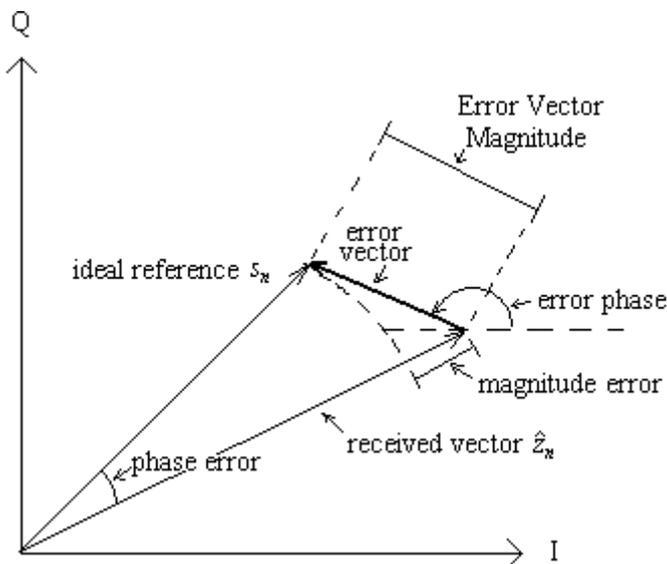| Prepared (also subject responsible if other)		No.		
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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

Figure 1-1 An illustration of the error vector and its components

A time slot has a length of 2560 chips (666.6 µs), therefore, the EVM calculation is done over bursts of 2560 elements (N=2560). In this project one chip is equivalent to one symbol, and the calculation of the error vector is done at each symbol instant, then, the signal has to be measured at the instants which best represent each symbol.

The variable k is used when the signal has not been measured yet at the optimal position, and each symbol is represented by more than one sample. Once the signal has been sampled at the optimal position, the variable n will be used. In other words, the signal contains a total of K samples and a total of N symbols, being k and n their respective variables. If we have an over-sampling of T (T points represent each symbol), then $K=NT$.

Signal	Burst before sampling	Burst after sampling	Sample at instant k	Sample at instant n
Ideal Reference	*	$S(n)$	*	s_n
Measured waveform	$Z(k)$	$Z(n)$	z_k	z_n
Modified version of the measured waveform	*	$\hat{Z}(n)$	*	$\hat{z}_n$

* Does not exist

Table 1-1 Notation of each signal

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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

The measured waveform is denoted as *Z(k)* and consists of a signal which has been corrupted by noise and frequency offsets. After this signal has been measured at the optimal point, it will be represented as *Z(n)*. The ideal reference signal is denoted as *S(n)*, it is a signal free of noise whose magnitude has been normalized to one. $\hat{Z}(n)$ is referred as the modified version of the measured signal where the frequency, absolute phase, absolute amplitude and chip clock timing have been selected so as to minimize the error vector. Each complex element of the bursts is represented as z_n, s_n, $\hat{z}_n$. See Table 1-1 for a summary of the notation.

The instantaneous error vector is obtained by subtracting the ideal reference from the modified version of the measured waveform. The root mean square EVM is defined by

$$EVM_{RMS} = \sqrt{\frac{\sum_{n \in N} \left| \hat{z}_n - s_n \right|^2}{\sum_{n \in N} \left| s_n \right|^2}}$$
(1-1)

1.2   Difficulties of the measurement

The first step to take is to find the proper interface since multiple tasks are usually integrated in a single component and it is not possible to acquire the signal at certain points of the system. Another important issue is the generation of a reference waveform. The received signal is not known and no specific synchronization signal or predetermined sequence is supplied. Thus it becomes another task to observe and detect the signal such that a reference signal can be obtained. The goal is to create the ideal reference waveform by observing the measured one. The measured signal consists of a sequence of *N* unknown symbols. It would be possible to observe a pilot signal which could help us in the generation of the ideal waveform, but this involves the creation of complex algorithms. Instead, assuming that the SNR is high enough, this ideal waveform is to be created by detecting each received symbol. In addition, two specific problems represent the main task to solve in this project: find the optimal measurement position and compensate for frequency offsets. Both difficulties are described in more detail in the next sections.

1.2.1   Optimal Measurement position

EVM is calculated at the symbol rate. The signal to be analyzed consists of a burst of *N* symbols, each symbol consists of a certain number of samples (*T*) equally distributed over the symbol interval, the number of samples representing a symbol is given by the ratio of the sampling frequency over the symbol frequency. If the sampling frequency equals 4 times the symbol frequency, then the number of samples per symbol *T* is 4.

The goal is to find the position inside the interval *T* that minimizes the EVM. After this is done, the samples must be extracted at that position, and used for the EVM calculation.

ERICSSON ≥		Open REPORT			9 (78)

Figure 1-2 shows the eye diagram of a QPSK signal divided into its I and Q components. The vertical lines indicate the instant with the smallest distortion and therefore minimizes the contribution to EVM.
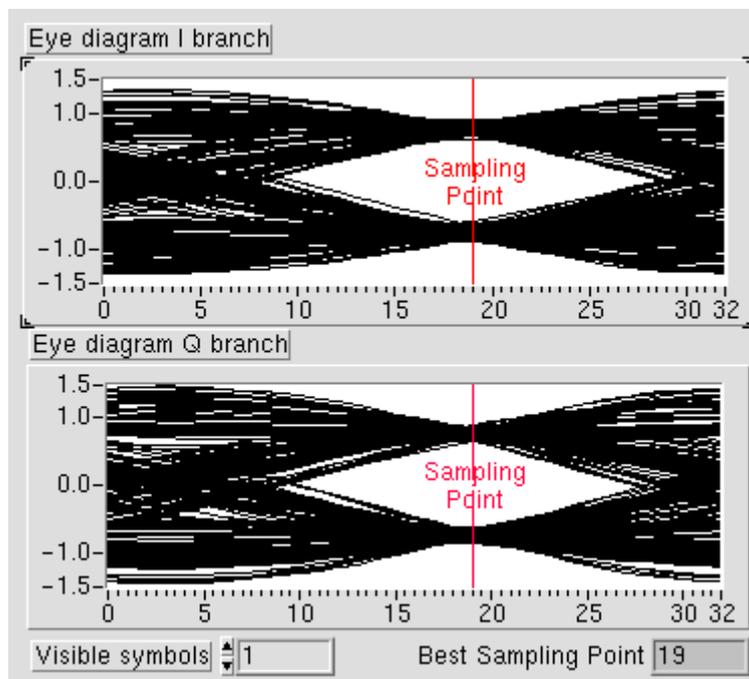

Figure 1-2 Eye diagram of a QPSK signal, divided into its I and Q branches.

But in our case, the signal is affected by frequency offsets, which makes the eye diagram more difficult to analyze and also makes it more difficult to find the optimal sampling instant. In Figure 1-3 it is possible to observe the effects of a frequency offset of 1 KHz.
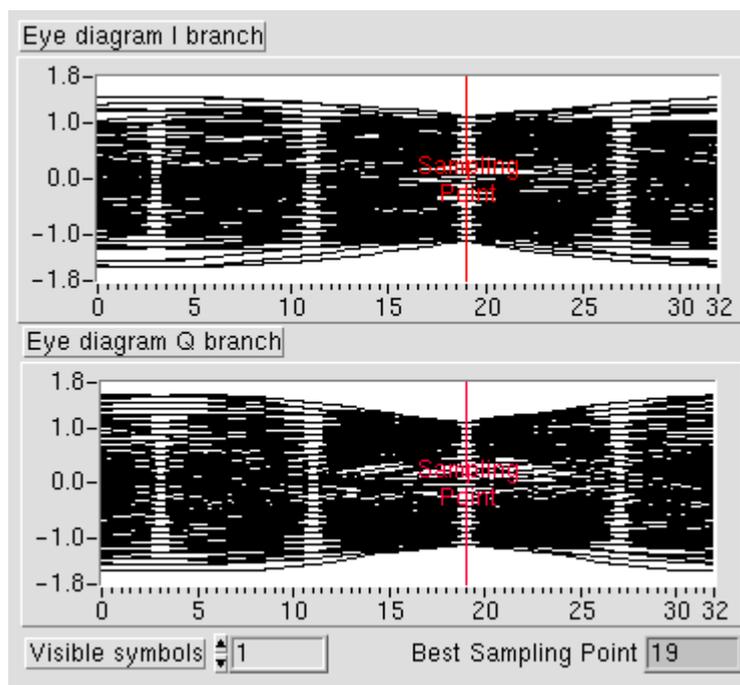
Figure 1-3 Eye diagram of a QPSK signal divided into its I and
Q branches, the signal has a frequency offset of
1KHz.

1.2.2   Frequency offsets

The RF signal sent is down-converted by an unlocked RF local oscillator (LO), which introduces frequency offsets. These frequency offsets affect the signal making it impossible to calculate the EVM without a prior compensation. To see the effects of frequency offsets in the time domain, we can make use of one of the Fourier's transform properties, the frequency shifting property. It states that if $F(\omega)$ is shifted by ω_0 its inverse transform is multiplied by $\exp(j\omega_0)$

$$\Im^{-1}\{F(\omega - \omega_0)\} = f(x)\exp(j\omega_0)$$

Therefore, compensating for frequency offsets in the frequency domain can be translated as compensating for phase rotation in the time domain. Phase rotation can be visualized if the received signal is plotted in the IQ constellation. In Figure 1-4 it is possible to recognize the 4 possible states of the QPSK signal and how have they been rotated around the origin due to a frequency offset.

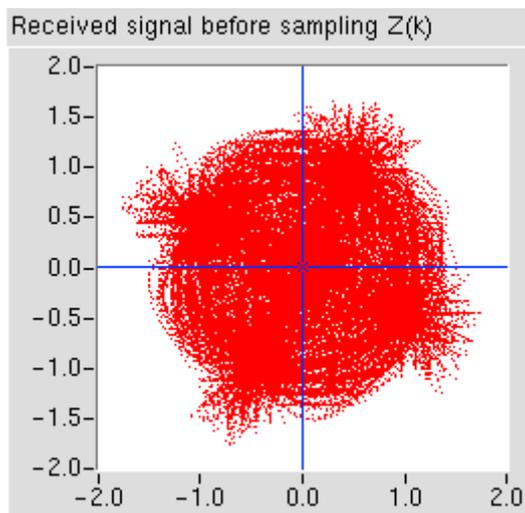| Prepared (also subject responsible if other)		No.		
EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen		
Approved	Checked	Date	Rev	Reference
EMW/TB/RBC Ödgärd Andersson		February 2002	A	

Figure 1-4 QPSK signal affected by a
frequency offset of 1Khz.

1.3 QPSK Modulation

The signal on which the EVM measurements are to be done is in Quadrature Phase Shift Keying (QPSK) format. In Phase-Shift Keying modulation, the phase of a constant amplitude carrier signal moves between different states and the information is contained in the phase of the carrier. For QPSK the phase of the carrier signal can take one of the following four equally spaced values: $\frac{\pi}{4}, 3\frac{\pi}{4}, 5\frac{\pi}{4}$, and $7\frac{\pi}{4}$, and the signal is given by

$$s_i(t) = \left\{ \sqrt{\frac{2E}{T}} \cos\left[2\pi f_c + (2i+1)\frac{\pi}{4} \right] \quad 0 \le t \le T \right. \tag{1-2}$$

for i=0,1,2,3.

E is the transmitted signal energy per symbol and f_c is the carrier frequency. Since the phase can assume four possible states, two bits (dibit) can be encoded in each symbol (2^2=4). The binary sequence is taken and divided into two different sequences, one including the even-numbered bits and the other including the odd-numbered bits, forming the in-phase (I) and the in-quadrature (Q) components respectively. See Table 1-2.

Input Dibit (IQ)	Phase
11	$\pi/4$
01	$3\pi/4$
00	$5\pi/4$
10	$7\pi/4$

Table 1-2 Signal space characterization of QPSK

The relation between the energy E carried by a symbol and the energy per bit E_b is given by

$$E = 2E_b \tag{1-3}$$

The average probability of symbol error for a QPSK signal in the additive white Gaussian noise (AWGN) channel can be expressed as

$$P_e = \text{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right) \tag{1-4}$$

where N_0 is the noise density and erfc(x) is the complementary error function defined by

$$\text{erfc}(x) = \frac{2}{\sqrt{\pi}} \int_x^\infty \exp(-y^2)\, dy \tag{1-5}$$

For more details about QPSK see Ref. [3].

2 Measurement Setup

The goal is to measure the EVM at the receiver of the base station in order to evaluate the uplink performance of the TRX, which is our device under test (DUT). Since the base station supports 2 antenna diversity, the receiver possess two identical branches A and B. The EVM measurement can be done at any of both branches.

The set-up consists basically in creating an RF signal by means of a signal generator, feeding it into the TRX, obtaining the digital modulated baseband signal using a logic analyzer, creating a reference signal, and computing the EVM by software. The set-up is shown in Figure 2-1.
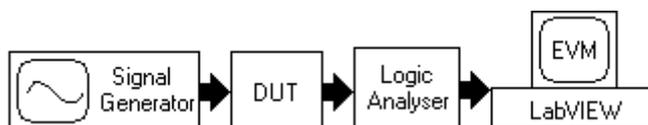

Figure 2-1 Measurement setup

The RF signal obtained from the signal generator is fed into the TRX board via the RF input. The transmitted waveform is a WCDMA QPSK signal with equal amplitude in the I and Q components. The signal has a symbol frequency of 3.84 M symbols per second and has been passed through a root raised cosine filter with a roll-off factor of 0.22. The center carrier frequency is 1950 MHz and the RF power level is –45 dBm.

At the TRX receiver, the signal is first down-converted to an intermediate frequency, then digitized by the analogue to digital converters (ADCs) before being sent to the FPGAs (so-called TRX-DIG). The FPGAs perform the down-conversion to baseband, decimation, RRC pulsed matched filtering, and automatic gain control (AGC). The RRC filters at the receiver have 77 taps.

To see a detailed list of the equipment used and the signal generator parameters refer to Appendix D.

2.1 Interfaces at the TRX

Two interfaces are to be analyzed for EVM calculation: gamma bus and the output of the ADCs. Figure 2-2 shows their location inside the WBTS system.

The gamma interface, with a rate of 30.72 M samples per second, is used for the transfer of down-link and up-link user data. The RX-gamma-bus consists of 2 high-speed pairs to send out the data from the TRX board to the TRX-IF. However, in the laboratory environment where the measurements are done, the data is sent to a test board instead to the TRX-IF. The test board function is to provide the signal to the logic analyzer.

The format of the parallel data stream consists of 10 bits in total, 5 bits for the quantized samples (bits 0-4) where bit 4 is the MSB and bit 0 is the LSB. The quantized samples are time multiplexed with respect to I and Q. The remaining bits (5-9) are used for future expansion, bit-parity, gain factor, and strobe. The format, shown in Appendix E, is the same for branches A and B. The output of each receiver branch is fed into 10-bit parallel to serial converters before being sent to the test board. For more details about the gamma bus consult ref. [11].
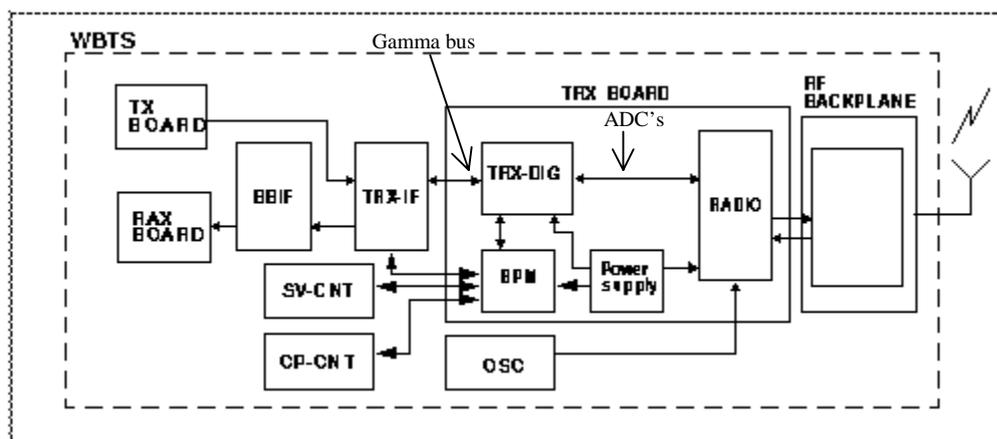

Figure 2-2 WBTS system block diagram

The other interface of interest is the output of the analogue to digital converters (ADCs). The ADCs, placed in the TRX board, interface the analogue TX/RX and the FPGAs, the output is fed into the input of each receiver branch. The signal consists of a 14 bit sample stream at a rate of 61.44 M samples per second. Both branches use 28 pins in total, which are fed into the signal analyzer using a test connector [8]. The modules for down-conversion to baseband and RRC pulse matched filtering, are implemented in Labview. The RRC filter in the software implementation has 161 taps.

The measurement done at the gamma bus gives us a figure-of-merit of the analogue RX, ADCs, and FPGAs performance. On the other hand, the measurement taken from the ADCs output gives us a figure-of-merit of the analogue RX and ADCs only, since the functions done by the FPGAs (down-conversion and RRC filtering) are implemented in software, and are considered to have an ideal behavior. We have assumed that the signal sent by the signal generator is ideal and has a negligible contribution to the EVM budget.

Based on the measured signal, a theoretical signal has to be generated by the software tool internally. First the measured signal is sampled at the optimal position and then compensated for the phase rotation generated by the frequency offsets. Simultaneously with the phase rotation compensation the signal is detected and the theoretical waveform is created. The EVM will be calculated by comparing the theoretical waveform and the modified version of the measured signal.

3   Methods

The proposed solution which finds the optimal measurement position and compensates against frequency offsets, to finally compute the EVM, can be divided into 5 different blocks:

- Normalization

- Interpolation

- Sampling

- LMS Adaptive Filtering/Detection

- EVM calculation

Figure 3-1 shows a block diagram of the proposed solution; notice that the first step is to obtain the signal at the point of interest. There are modules which were already implemented in Labview to obtain the signal in baseband form.
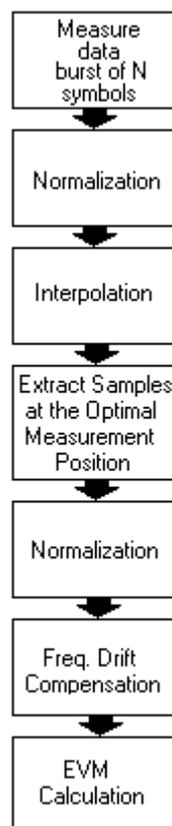

Figure 3-1 Proposed solution
block diagram

The following sections describe each block in detail.

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3.1 Normalization

Here, the constellation diagram of the measured signal is normalized, in other words the mean distance between the origin and the sampling points is set to one. The operation extracts the magnitude of the IQ samples, computes the magnitude mean over the burst and scales all the samples so the mean at the end is equal one. The normalized signal is given by

$$Z_{norm}(k) = \frac{Z(k)}{\frac{1}{K}\sum_{k=0}^{K-1}|z_k|}$$

(3-1)

where N is the number of elements in the burst.

This block is applied twice during the program execution: before interpolation, and after sampling. The reason of normalizing twice is because the total power in the burst changes after sampling. Therefore the signal has to be normalized again so the mean distance between the origin and the symbols magnitude equals one. The name of the module that performs this task in Labview is *Normalize IQ Magnitude.vi*

3.2 Interpolation

Interpolation is necessary to increase the number of samples per symbol interval *T*. Having more samples per symbol interval results in a better precision when selecting the optimal sampling point. If *T* is not large enough, then errors can be introduced, which leads to a higher EVM. The original measured signal taken from the gamma bus has *T*=4. The signal taken from the ADCs has *T*=16. It is recommended to interpolate the data in the gamma bus in order to have at least *T*=16. However, using *T*=64 in both (gamma bus and ADC's) signals, will result in a more accurate measurement.

Interpolation is done by first up-sampling the signal (filling up with zeroes between samples) and then filtering the signal in a low pass (LP) filter. The need of the LP filter comes from the fact that the zero-valued samples introduced by the up-sampler have to be converted to interpolated samples. The explanation of this effect is based on the time-frequency duality which is a property of the Fourier transformation. The spectrum of the up-sampled signal is an *L*-folded compressed version of the original signal spectrum. Multiple images of the compressed spectrum are created after the up-sampling, thus an LP filter is needed to suppress the extra images. [9]
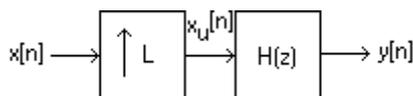

Figure 3-2 Block diagram of an *L*-interpolator

If the sequence x[n] is passed through a factor -*L* up-sampler, as shown in Figure 3-2, having x_u[n] as an output, the relation between the Fourier transform X_u(e^{jw}) and X(e^{jw}) is given by

$$X_u\left(e^{jw}\right) = X\left(e^{jwL}\right) \tag{3-2}$$

The LP filter must have a cutoff at π/L (relative to the sampling frequency, $f_s=2\pi$) and a gain of L. The effect that the up-sampler and the LP filter have over the signal spectrum is shown in Figure 3-3.
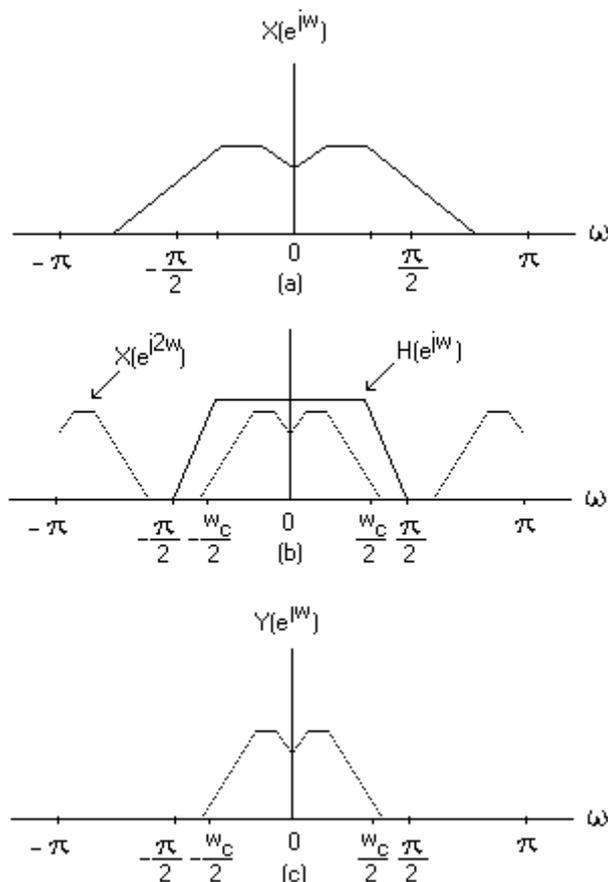

Figure 3-3. (a) spectrum of the signal before up-sampling. (b) L-folded compressed version of the original signal spectrum when L=2. (c) interpolated signal spectrum, the up-sampled signal has been passed through a low pass filter with L=2.
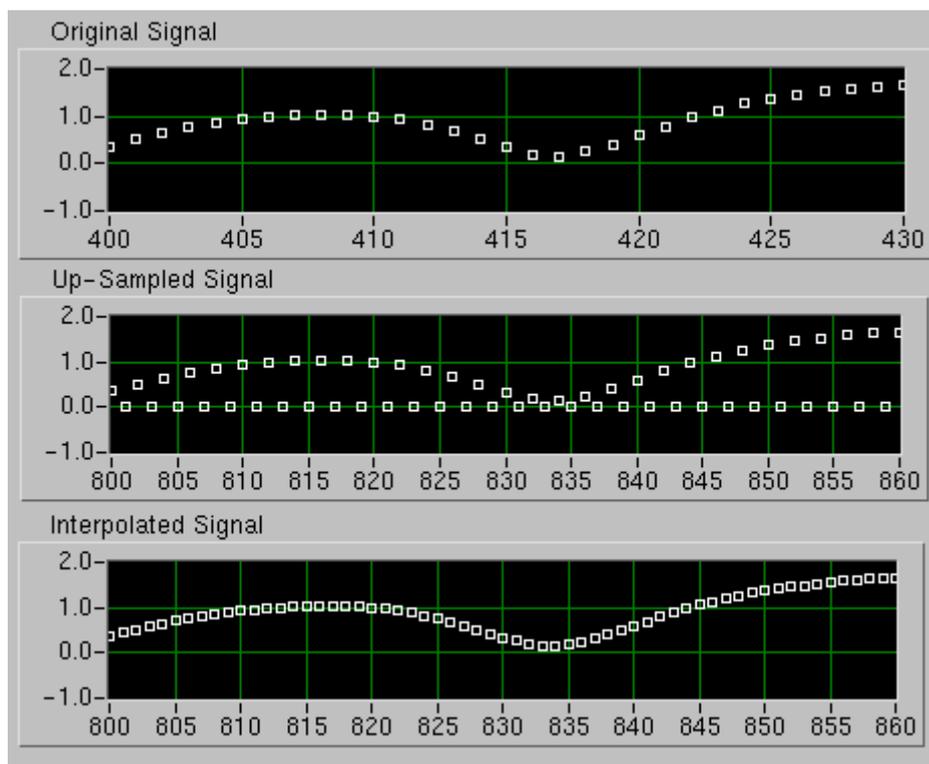
Figure 3-4 Effects of an interpolator with *L*=2. (Top) original signal. (Center) up-sampled signal. (Bottom) interpolated signal.

Finally, after applying the *L*-factor interpolator, the signal posses *L* times the number elements before interpolation. See Figure 3-4.

In this application an interpolation factor of *L*=2 is applied, the original signal is up-sampled by a factor of two and a half-band LP filter is used to suppress the *L*-folded compressed version of the original signal spectrum.

The effect of two interpolators with *L*=2 applied in cascade is equivalent to the effect of one interpolator with *L*=4. In the same way an interpolator with *L*=8 can be built using three interpolators of *L*=2, and so on. Thus, an interpolator with *L*=2 is to be used and applied repeatedly in cascade when a higher factor *L* needed.

The module created in Labview called *Interpolator Factor 2.vi* interpolates the signal by a factor *L*=2, the input is an array of *N* samples and the output consists of an array of 2*N*-1 samples. This implementation is divided in two parts: up-sampling and LP filtering. The up-sampling part interleaves the array containing the original sequence with another array of zeroes of the same size. The LP filtering part performed by the Labview module named *Halfband Filter.vi* convolves the resulting up-sampled sequence with a set of coefficients listed in Table 3-1.

Coefficient	Value
h_0	6.295593455434E-3
h_1	0.000000000000E+0
h_2	-5.009969463572E-2
h_3	0.000000000000E+0
h_4	2.938045468181E-1
h_5	5.000000000000E-1
h_6	2.938045468181E-1
h_7	0.000000000000E+0
h_8	-5.009969463572E-2
h_9	0.000000000000E+0
h_{10}	6.295593455434E-3

Table 3-1 Half band filter coefficients

The Labview module *Interpolator Multiple.vi* runs repeatedly *n* times an interpolator of *L*=2. The inputs are *n* and the original sequence with *N* samples. The outputs are the interpolated data by a factor of $L=2^n$ with $N*2^n-2^n+1$ samples, and the number of samples per symbol interval.

3.3   **Optimal measurement position**

This section describes an algorithm to find the optimal measurement position. A good way to start is taking a look at the eye diagram.

The eye diagram or eye pattern is the result of translating the waveforms in successive symbol intervals, into one interval on the display. The term *eye* comes from the similarity of this graphic tool with the human eye. [3]

The input is the received signal *Z(k).* It has a symbol interval length of *T* samples, and a total length of *N* symbols. The total number of samples in the burst equals *K=NT*.

The output displayed is given by

$$z_{t,n} = z_{k-t-nT} \qquad\qquad (3-3)$$

for *t*=0,1,...,*T*-1 and *n*=0,1,...,*N*

The display is periodic and identical for each symbol interval:

$$nT \le t \le (n+1)T \qquad\qquad (3-4)$$

where *n*=1,2,...,*N.*

At the end we observe the patterns displayed in Figure 1-2. The eye opening is the distance from the zero level to the closest trace at the optimal sampling instant (where the eye opens widest). The wider the opening the better the quality of the signal. The closing of the eye along the time axis is indicative of distortion and jitter. [3] [10].

The proposed algorithm is based on the analysis of the variance of the IQ signal magnitude at each sampling instant $0 \le t \le T-1$ when looking at the eye diagram. The lowest variance will be obtained at that instant where most of the crossings coincide, i.e. the best sampling instant.

The algorithm consists in the following steps:

1. Get burst of K samples: $Z(k)$ k=0,1,2,...,K-1

2. Divide $Z(k)$ in T vectors:
$$\mathbf{v}_0=[z_{k-0},\ z_{k-T},\ z_{k-2T},...,z_{k-(N-1)T}]^T$$
$$\mathbf{v}_1=[z_{k-1},\ z_{k-1-T},\ z_{k-1-2T},...,\ z_{k-1-(N-1)T}]^T$$
$$\mathbf{v}_2=[z_{k-2},\ z_{k-2-T},\ z_{k-2-2T},...,\ z_{k-2-(N-1)T}]^T$$
$$\vdots$$
$$\mathbf{v}_{T-1}=[z_{k-(T-1)},\ z_{k-(T-1)-T},\ z_{k-(T-1)-2T},...,\ z_{k-(T-1)-(N-1)T}]^T$$

in general
$$v_{t,n}=z_{k-t-nT}$$
for t=0,1,2,...,T-1 and n=0,1,2,...,N-1

3. Obtain the magnitude of each element of the vectors $\mathbf{v}_t$:
$$\mathbf{m}_0=[\ |v_{0,0}|\ \ |v_{0,1}|\ \ |v_{0,2}|\ ...\ |v_{0,N-1}|\ ]^T= [\ |z_{k-0}|,\ |z_{k-T}|,\ |z_{k-2T}|,...,|z_{k-(N-1)T}|]^T$$
$$\mathbf{m}_1=[\ |v_{1,0}|\ \ |v_{1,1}|\ \ |v_{1,2}|\ ...\ |v_{1,N-1}|\ ]^T= [\ |z_{k-1}|,\ |z_{k-1-T}|,\ |z_{k-1-2T}|,...,|z_{k-1-(N-1)T}|]^T$$
$$\mathbf{m}_2=[\ |v_{2,0}|\ \ |v_{2,1}|\ \ |v_{2,2}|\ ...\ |v_{2,N-1}|\ ]^T=[\ |z_{k-2}|,\ |z_{k-2-T}|,\ |z_{k-2-2T}|,...,|z_{k-2-(N-1)T}|]^T$$
$$\vdots$$
$$\mathbf{m}_{T-1}=[\ |v_{T-1,0}|\ \ |v_{T-1,1}|\ \ |v_{T-1,2}|\ ...\ |v_{T-1,N-1}|\ ]^T$$
$$=[\ |z_{k-(T-1)}|,\ |z_{k-(T-1)-T}|,\ |z_{k-(T-1)-2T}|,...,\ |z_{k-(T-1)-(N-1)T}|\ ]^T$$

4. Calculate the variance of each magnitude vector $\mathbf{m}_t$

$$\sigma_1^2 = \mathrm{var}\{\mathbf{m}_1\}$$
$$\sigma_2^2 = \mathrm{var}\{\mathbf{m}_2\}$$
$$\vdots$$
$$\sigma_{T-1}^2 = \mathrm{var}\{\mathbf{m}_{T-1}\}$$

Each of the variances σ^2_t calculated corresponds to a sampling instant t from 0 to T-1. Therefore it can be considered as a function of time $\sigma^2(t)$

5. The optimal sampling position is the one that leads to the lowest variance. In other words, the argument of the minimum of the variance function, given by

$$t_{opt} = \arg\{\min\{\sigma^2(t)\}\} \tag{3-5}$$

6. Once the optimal sampling position has been determined it is possible to extract the sample representing each symbol in the following fashion

$$Z(n) = Z(k - nT - t_{opt}) \tag{3-6}$$

for n=0,1,2,...,N-1.

Below in Figure 3-5, the magnitude of I+jQ of the burst, has been plotted in the same way as an eye diagram, observe how most of the values coincide at sampling position 60.

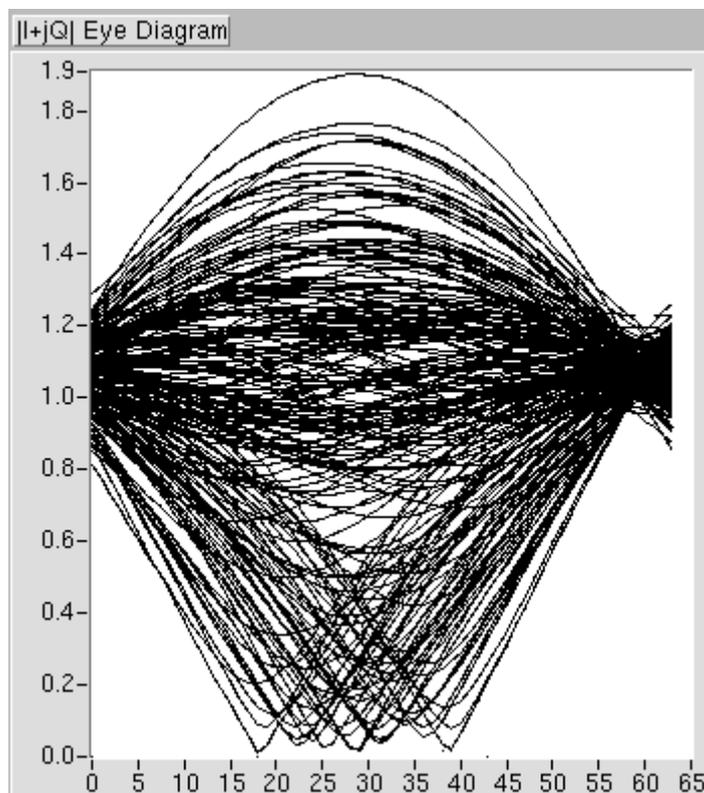

Figure 3-5 Eye diagram of $|$I+jQ$|$. N=2560. T=64.

At Figure 3-6 is possible to observe the variance at each instant corresponding to the same signal shown in Figure 3-5. Note how the variance reaches its minimum at t_{opt}=60.

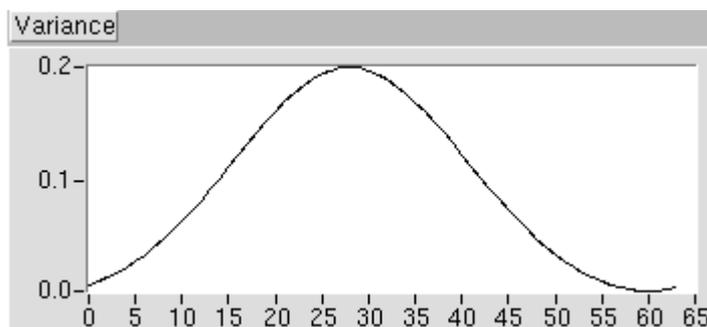| Prepared (also subject responsible if other)		No.		
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Figure 3-6 Variance observed at each instant inside the symbol interval. *N*=2560. *T*=64.

This algorithm performs well regardless of the phase rotation, since it is based on the magnitude of the samples. The larger the number of samples per symbol the more accurate solution. Interpolation represents a trade-off since as we use more samples, more data has to be processed and the algorithm becomes slower. However, in this case, it is recommended to have a larger number of samples per symbol interval which results in more exact measurements.

Since this method works based on statistical information of the signal, it is suggested to use more than 50 symbols per burst processed to ensure a good performance. Nevertheless, for this application the number of symbols to be processed will be equal or greater than 2560.
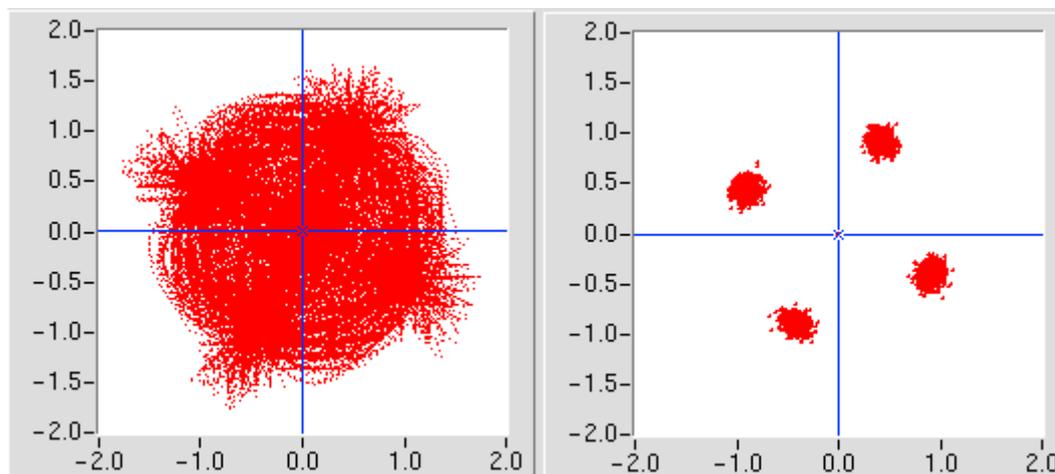
Figure 3-7 IQ constellation of the QPSK signal before and after sampling.

The proposed algorithm described is implemented in the Labview module called *Sample.vi*, this block is composed by two parts: the first one which finds the optimal measurement position is performed by the module called *Find Sampling Point.vi*, and the second one which extracts the samples at the indicated measurement position is performed by the module *Extract Samples.vi*.

The inputs for the first part are the data burst and the number of samples per symbol interval, the output is the optimal measurement position. For the second part the inputs are the data burst, the number of samples per symbol interval and the optimal measurement position. The output is the array of samples taken at the optimal measurement position.

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The Labview module *Sample.vi* has as inputs the data burst and the number of samples per symbol interval. And as outputs the array of samples taken at the optimal measuring position, and the optimal measuring position itself.

Even though we have managed to sample the signal, the effects of the frequency offsets are still present, and the EVM cannot be computed yet, otherwise its value would go to infinity. In the following section a method to compensate for these frequency offsets is described.

3.4 Frequency-offset compensation

Compensating for a frequency offset involves a tracking process for correcting the phase rotation changes and extracting the information relevant for EVM calculation which is phase noise and magnitude errors. The problem becomes more difficult since phase rotation has to be de-embedded from the phase noise. However, we take the advantage of the fact that the phase response as a function of frequency is considered as a deterministic parameter and phase noise as a random parameter.

Figure 3-8 illustrates the problem for 1000, 2000, 3000, and 4000 symbols, as more and more data is plotted the rotation of the IQ constellation becomes more remarkable.

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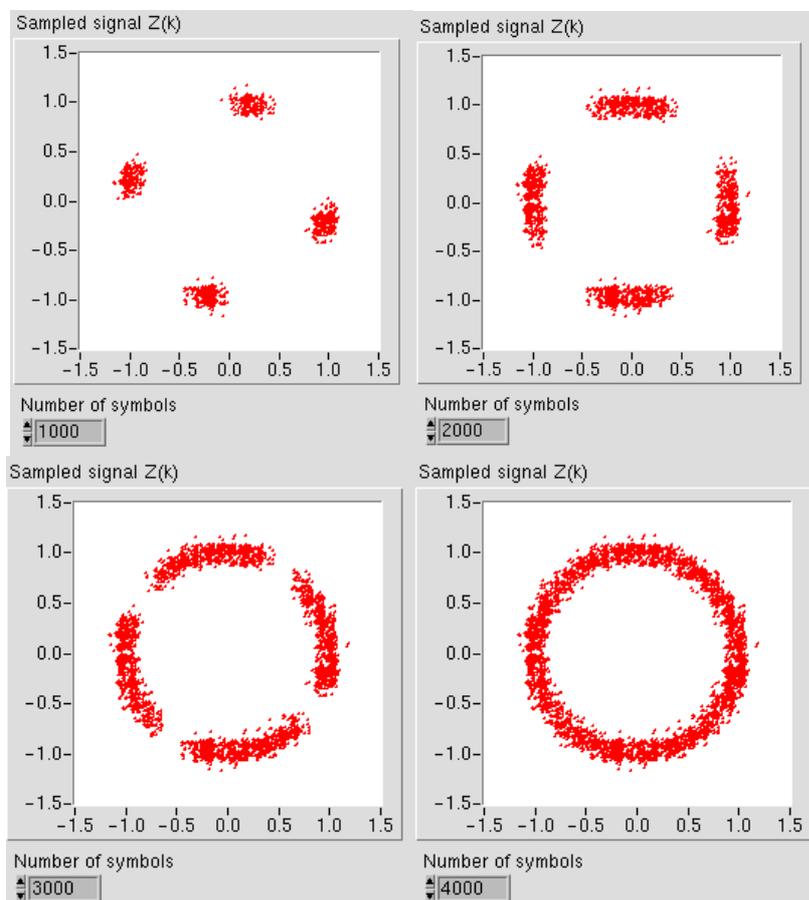
Figure 3-8 Illustration of phase rotation

Here we can create a model for the measured signal. Each received symbol has a magnitude which has been affected by AWGN, and its phase possess 3 components: the ideal phase of the transmitted symbol exp($j\theta_n$), the phase error added due to AWGN exp($j\psi_n$), and the phase component due to frequency offsets exp($j\phi_n$). Therefore the model for each symbol z_n is given by

$$z_n = A_n \exp(j(\theta_n + \psi_n + \phi_n)) \tag{3-7}$$

Our objective is to de-embed each of the phase parameters. In this case, we assume that the component due to a frequency offset has a linear behavior defined by

$$\phi_n = vn + b \tag{3-8}$$

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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

where v is the parameter which defines the phase rotation (number of radians per symbol), and b is the initial phase offset. We will make use of a Least Mean-Square adaptive filter to extract the phase component due to a frequency offset $\exp(j\phi_n)$, and during this process it is possible that part of the phase noise $\exp(j\psi_n)$, is extracted together. However the assumption that ϕ_n behaves linearly will help us to differentiate it from ψ_n which is a zero-mean Gaussian process.

The Least Mean-Square (LMS) algorithm belongs to the family of stochastic gradient-based algorithms and its main feature is the simplicity, which leads to a low complexity implementation. The LMS describes a feedback control system and consists of a combination of two basic processes [4]:

1. An adaptive process, which involves the automatic adjustment of a set of tap weights.

2. A filtering process, which involves (a) forming the inner product of a set of tap inputs and the corresponding set of tap weights emerging from the adaptive process to produce and estimate of a desired response, and (b) generating an estimation error by comparing the estimate with the actual value of the desired response. The estimation error is in turn used to actuate the adaptive process, thereby closing the feedback loop.

However, the signal at this moment is not continuous and each received symbol z_n has an arbitrary ideal phase ($\theta_n=\pi/4$, $3\pi/4$, $5\pi/4$, $7\pi/4$). Subsequent samples are considered un-correlated between each other with respect to the ideal phase, therefore, the number of filter coefficients is one for this application (w_n), and the algorithm performs analyzing sample per sample, obtaining the error at each time, and using it to correct the next incoming symbol so as to minimize the cost function. The error obtained at each instant is a function of ψ_n and ϕ_n and plays and important role to de-embed both variables, as we will see further on. Notice that the estimation error computed here is not the same as the error vector defined in section 1. The estimation error e_n is used to de-embed the phase rotation that affects the received signal.

The cost function is defined by

$$\xi_n = E\{|e_n|^2\} \tag{3-9}$$

e_n is the estimation error at instant n, given by

$$e_n = \hat{s}_n - s_n \tag{3-10}$$

where s_n is the value of the desired response and $\hat{s}_n$ is the estimate. Then we need to apply a correction update to the filter coefficient w_n, the update must lead to a minimization of the cost function and has the following form

$$w_{n+1} = w_n - \mu\nabla\xi_n \tag{3-11}$$

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The explanation of this update comes from the fact that the gradient of the cost function points towards the direction it increases. Thus, the update consists of a small correction toward the direction it decreases, and that is the reason why we use the negative sign. The parameter μ affects the rate at which the weigh coefficient moves toward the optimal solution.

The gradient corresponds to the derivative of $E\{|e_n|^2\}$ with respect to w_n^*, given by

$$
\begin{aligned}
\nabla \xi_n &= \nabla E\{|e_n|^2\} \\
&= E\{\nabla |e_n|^2\} \\
&= E\{e_n \nabla e_n^*\}
\end{aligned}
\tag{3-12}
$$

The gradient of the estimation error is given by

$$
\nabla e_n^* = -z_n^*
\tag{3-13}
$$

thus, the gradient of the cost function becomes

$$
\nabla \xi_n = -E\{e_n z_n^*\}
\tag{3-14}
$$

The simplicity of the LMS algorithm comes when the expression $E\{e_n z_n^*\}$ is replaced by a one point sample mean $e_n z_n^*$. Then the update equation assumes the following form given by

$$
w_{n+1} = w_n - \mu e_n z_n^*
\tag{3-15}
$$

this equation is the key in the adaptation process of the LMS algorithm and we have to recall that it was derived for the case when the number of taps in the filter is one.

The filter coefficient at the instant n, w_n, is updated for each new sample as defined in Eq. (3-15). The estimated signal sample $\hat{s}_n$ is defined as

$$
\hat{s}_n = w_n^* z_n
\tag{3-16}
$$

for $n \in N$.

Each estimated symbol $\hat{s}_n$ has to be compared with a reference symbol s_n to compute the error, but at that time it is unknown which symbol has been sent. Then $\hat{s}_n$ is detected to create the ideal reference.

Detection theory deals with the design and evaluation of a decision-making processor that observes the received signal and guesses which particular symbol was transmitted according to some set of rules [3].

In QPSK the signal constellation consists of 4 equally spaced symbols: m_0, m_1, m_2, m_3:

$$m_0 = \cos\left(\frac{\pi}{4}\right) + j\sin\left(\frac{\pi}{4}\right)$$

$$m_1 = \cos\left(3\frac{\pi}{4}\right) + j\sin\left(3\frac{\pi}{4}\right)$$

$$m_2 = \cos\left(5\frac{\pi}{4}\right) + j\sin\left(5\frac{\pi}{4}\right)$$

$$m_3 = \cos\left(7\frac{\pi}{4}\right) + j\sin\left(7\frac{\pi}{4}\right)$$

therefore, the observation space is partitioned in 4 regions B_0, B_1, B_2, B_3. Each region is divided by the decision boundaries, which in this case are the I and Q axis. The rule for detection consists in setting $\hat{m} = m_i$ if the distance between the received symbol z_k and m_i is the minimum for i=0,1,2,3.

This method is called maximum-likelihood detection and basically selects the message point closest to the received signal point. [3]

Each generated symbol resulting from detection has the form given by

$$s_n = A\exp(j\theta_n) \tag{3-17}$$

where the ideal amplitude A equals always 1, and θ_n can assume one of the four possible ideal angles $\pi/4$, $3\pi/4$, $5\pi/4$, $7\pi/4$.

Once we count with a reference symbol and an estimated symbol it is possible to obtain the estimation error at the instant n. The estimation error is used to close the feedback loop of the adaptive process, as shown in Figure 3-9.
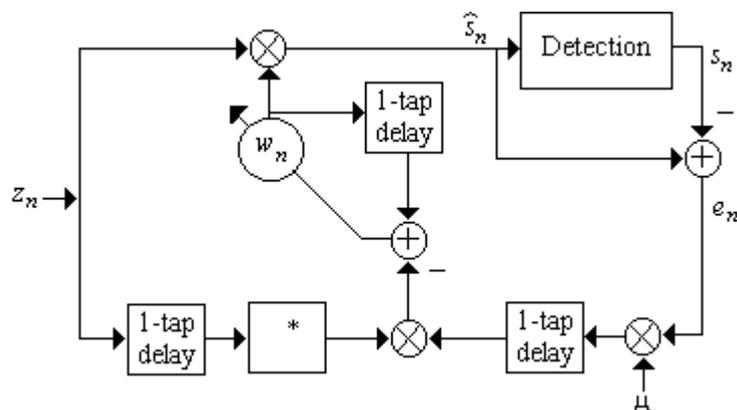

Figure 3-9 LMS adaptation process.

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The response of the algorithm can be tuned through the step size μ. The step size μ modifies the speed of convergence, if this value is small we get a more accurate response when $t \to \infty$ but the speed of convergence is slow. If μ is large then the response when $t \to \infty$ is less accurate and speed of convergence is fast. Nevertheless, for stability, μ is upper and lower bounded by

$$0 < \mu < \frac{2}{\lambda_{max}} \tag{3-18}$$

where λ_{max} is the maximum eigenvalue of the autocorrelation matrix $\mathbf{R}_z$. It is possible to use the fact that λ_{max} is upper bounded by the trace of $\mathbf{R}_z$

$$\lambda_{max} \leq \sum_{i=0}^{p} \lambda_i = \text{tr}(\mathbf{R}_z) \tag{3-19}$$

Eq. (3-19) is designed for the general case where the number of taps in the LMS filter equals *p+1*, being *p* the order of the filter (0 in our case).
If we consider *Z(n)* as a wide-sense stationary process, then the trace can be replaced by

$$\text{tr}(\mathbf{R}_z) = (\text{number of taps}) E\{|Z(n)|^2\} \tag{3-20}$$

therefore, $E\{|Z(n)|^2\}$ can be estimated from an average as

$$\hat{E}\{|Z(n)|^2\} = \frac{1}{N} \sum_{n=0}^{N-1} |z_n|^2 \tag{3-21}$$

and the boundaries for the step size now become

$$0 < \mu < \frac{2}{(\text{number of taps}) \frac{1}{N} \sum_{n=0}^{N-1} |z_n|^2} \tag{3-22}$$

However, our application requires a number of taps equal to one, then Eq. (3-22) can be simplified to

$$0 < \mu < \frac{2}{\frac{1}{N} \sum_{n=0}^{N-1} |z_n|^2} \tag{3-23}$$

The upper bound is generally considered too large to ensure stability. It might converge in the mean but probably with a large variance, which most of the times is unwanted. Common values range around one order less than the upper bound. Thus, for this application we set μ equal the upper bound and introduce a new factor called β whose default value is 0.1 and can also be adjusted by the user. Hence, we define a new step size parameter μ' defined by

$$\mu' = \beta \cdot \mu \tag{3-24}$$

μ is defined as

$$\mu = \frac{2}{\frac{1}{N}\sum_{n=0}^{N-1}|z_n|^2} \qquad (3\text{-}25)$$

finally β can be varied between the range given by

$$0 < \beta < 1 \qquad (3\text{-}26)$$

w_0 is initialized as

$$w_0 = 1 \qquad (3\text{-}27)$$

z_0 is the first incoming symbol.

The LMS algorithm can be summarized in the following steps [5]:
1. Parameters: Number of taps= 1
                        β=Step size adjustment (0.1 default)
2. Initialization:
   $$w_0 = 1$$
3. Computation: For n=0,1,2,...,N-1
   a) $\hat{s}_n = w_n^* z_n$
   b) s_n=detect($\hat{s}_n$)
   c) $e_n = \hat{s}_n - s_n$
   d) $w_{n+1} = w_n + \mu' e_n z_n^*$

After the entire sequences $S(n)$ and $\hat{S}(n)$ have been computed it is still not possible to calculate the EVM. Observe Figure 3-10 at the right
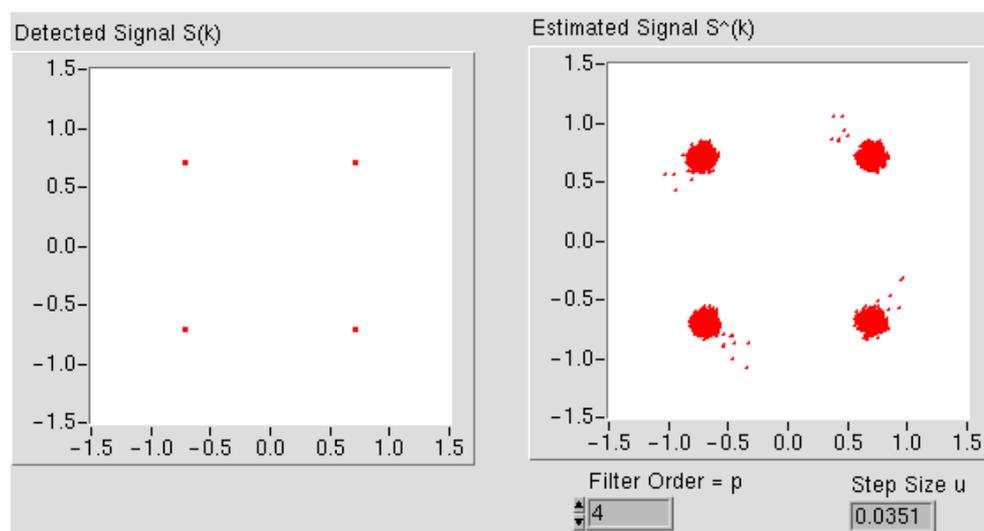

Figure 3-10 (Left) detected signal. (Right) illustration of the transient response.

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the first values of $\hat{S}(n)$ forming a wake towards the reference points belong to the transient response. They should not account for the EVM since the algorithm has not been adapted yet and might lead to an erroneous measurement. One way to avoid this problem is to calculate the time that takes for the algorithm to converge and skip the symbols included in that period of time.

Before going further lets first state the misadjustment. The misadjustment is defined as the ratio of the excess mean-square error to the minimum mean square error, and is thus a measure of how closely the adaptive process tracks the true Wiener solution, [6]. Usually a misadjustment of 10% is considered satisfactory. The misadjustment is given by

$$M = \frac{\text{Excess MSE}}{\xi_{\min}} \qquad (3\text{-}28)$$

$$\approx \mu \, \text{tr}(\mathbf{R_z})$$

$$\approx \mu (\text{number of taps}) \frac{1}{N} \sum_{n=0}^{N-1} |z_n|^2$$

The misadjustment M and the time constant τ_{MSE} are related through the following approximation

$$M \approx \frac{(\text{number of taps})}{4\tau_{MSE}} \qquad (3\text{-}29)$$

this approximation assumes that all the eigenvalues are equal, however, it has shown to be a good guess even when the eigenvalues are not equal [6]. The algorithm takes four time constants to converge, and the total transient time can be expressed as

$$t_s \approx \frac{\text{Number of taps}}{\text{Missadjustment}} \qquad (3\text{-}30)$$

for our case

$$t_s \approx \frac{1}{\text{Missadjustment}} \qquad (3\text{-}31)$$

Figure 3-11 shows the squared error after an arbitrary signal has been passed through the one-tap LMS filter, with a step size equal to one tenth of the upper bound and a misadjustment of 10%. This gives the following settling time:

$$t_s = \frac{1}{0.1} = 10 \qquad (3\text{-}32)$$

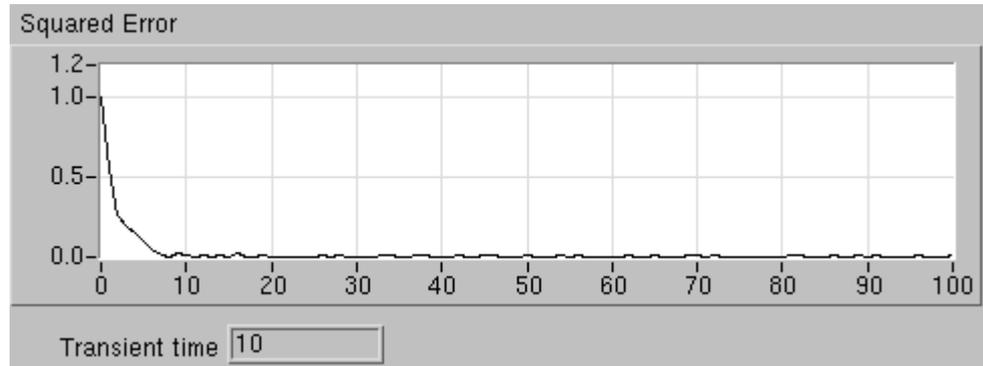| Prepared (also subject responsible if other)				
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Figure 3-11. Transient response of the squared error.

We have proven in practice that the expression derived for the settling time in Eq. (3-30) is a good estimate.

If we take a look to the phase of the filter coefficient w_n after the algorithm has converged, we observe that we have a rough estimate of ϕ_n which we can denote as $\hat{\phi}_n$. Therefore, the filter coefficient w_n is defined by

$$w_n = W_n \exp(-j\hat{\phi}_n) \tag{3-33}$$

where W_n is the magnitude of w_n. To be able to de-embed ϕ_n it is necessary to obtain the best linear fit of $\hat{\phi}_n$ along the burst, given by

$$\phi_n = \text{linear fit } \{\hat{\phi}_n\} \tag{3-34}$$

for $n=0,1,2,...,N$-1.

We can define a new coefficient c_n whose phase equals exp($-j\phi_n$). The magnitude of c_n is given by

$$|c_n| = 1 \tag{3-35}$$

Then multiplying c_n by z_n for all $n \in N$, results in an estimated signal $\hat{Z}(n)$ whose magnitude of each symbol equals the magnitude of its respective symbol in $Z(n)$. The resulting $\hat{z}_n$ is given by

$$\hat{z}_n = z_n \cdot c_n \tag{3-36}$$

Finally ψ_n is the difference between the phase angle of $\hat{z}_n$ and θ_n. The phase noise ψ_n must have a mean equal to zero for all $n \in N$, and to be sure of it the mean is calculated and removed if present. Any residual phase offset taken away in the following way

$$\hat{Z}(n) = \hat{Z}(n) \exp(-j \frac{1}{N} \sum_{n=0}^{N-1} \psi_n) \tag{3-37}$$

$\widehat{Z}(n)$ is referred as the modified version of the measured signal where the frequency, absolute phase, absolute amplitude and chip clock timing have been selected so as to minimize the error vector. The compensation done against frequency offsets involved only the phase. The magnitude of the original signal remained unaffected.

Now it is possible to compute the EVM as defined in Eq. (1-1).

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4   Results, discussion and future work

4.1   Results

Following the setup described in section 2 and the algorithm implementation described in section 3, two different results were obtained: EVM_{RMS} at the gamma bus, and EVM_{RMS} at the analogue to digital converters.

EVM_{RMS} measured at the gamma bus was 6.6%. Figure 4-1 shows $\hat{Z}(n)$ obtained from the gamma bus, the crossings of the straight grid lines represent the four ideal states.
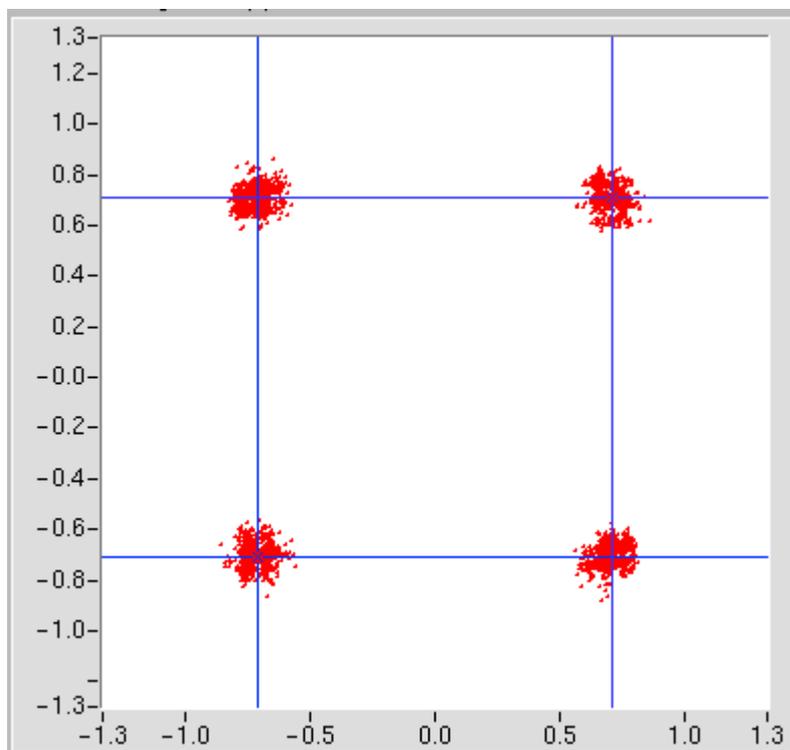

Figure 4-1. $\hat{Z}(n)$ measured at the gamma bus.

EVM_{RMS} measured at the ADCs was 4.5%. Figure 4-2 shows $\hat{Z}(n)$ taken from the ADCs. Note how in this case the received symbols are closer to the ideal references than in the signal taken at the gamma bus.

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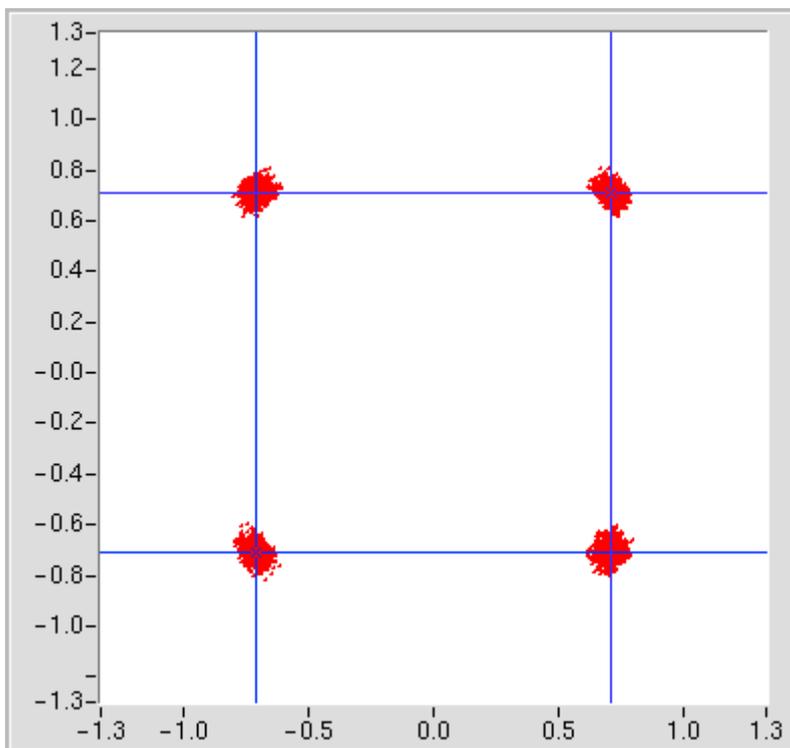| Prepared (also subject responsible if other)		No.		
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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

Figure 4-2. $\hat{Z}(n)$ measured at the ADCs.

Figure 4-3 shows a signal obtained from the gamma bus, before and after being measured at the optimal position.
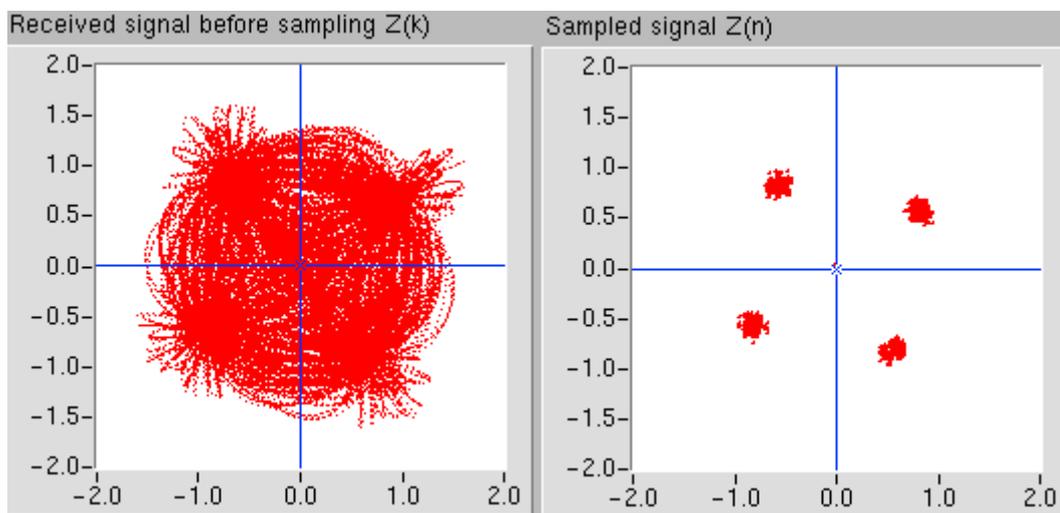

Figure 4-3. $\hat{Z}(n)$ measured at the gamma bus. (Left) before sampling. (Right) after sampling.

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The original number of samples per symbol interval at the signal shown in Figure 4-3 was 4. After being interpolated with a factor of L=16, the resulting over-sampling was T=64. For the signal taken at the ADCs the original number of samples per symbol interval was 16, and after interpolation with L=4 the resulting over-sampling is T=64.

Below in Figure 4-4, $\hat{\phi}_n$, ϕ_n, and $-\phi_n$, have been plotted. They correspond to the signal analyzed in Figure 4-3. The vertical axis is in radians and the horizontal axis represents n. Observe that the initial phase offset b is equal to 0.15 rads ($\approx$8.6 degrees), and v has a value of 1.95×10^{-5} rads/symbol.
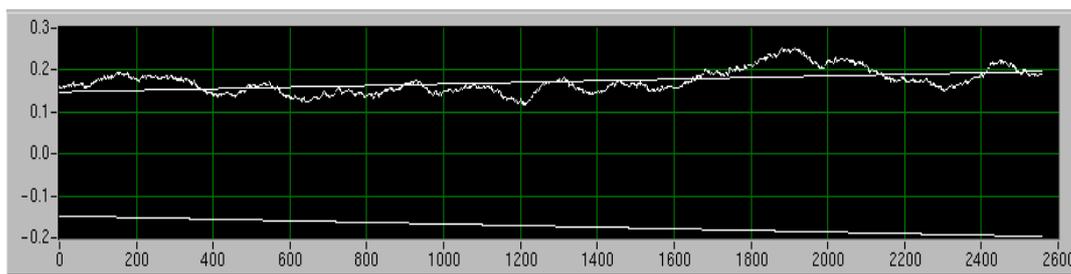

Figure 4-4. Illustration of $\hat{\phi}_n$, ϕ_n, and $-\phi_n$.

Hence, ϕ_n for this specific burst can be modeled as

$$\phi_n = 1.95 \times 10^{-5} n - 1.5 \times 10^{-1} \text{ rads}$$

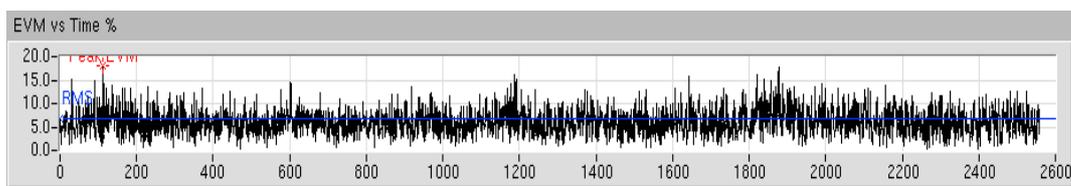Figure 4-5 shows the EVM versus time, corresponding to the same signal analyzed in Figure 4-3.

Figure 4-5. EVM vs time.

After applying the same linear compensation over the original received signal $Z(k)$, it is possible to plot the eye diagram of I and Q branches as shown in Figure 4-6.
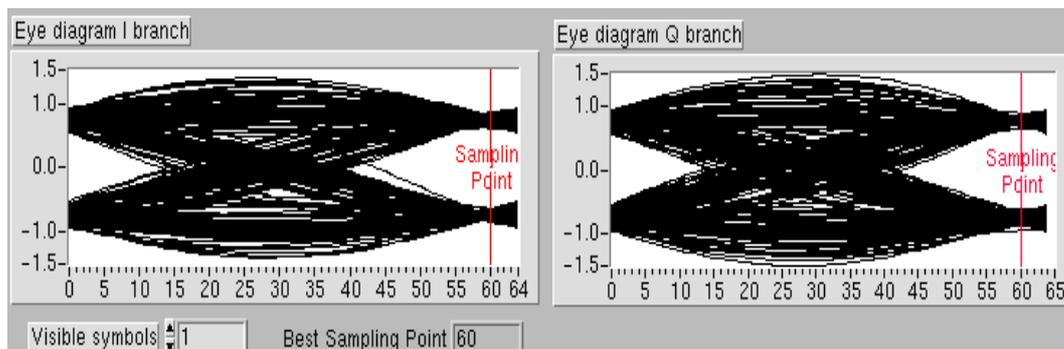
Figure 4-6. Eye diagram of *Z(k)* after removal of frequency offsets.

Observe how the widest opening of the eye occurs at t_{opt}=60. Coinciding with the results shown in Figure 3-5 and Figure 3-6.

The Labview virtual instruments (VI's) which resulted from the implementation of the algorithms are shown in Appendix A. The results presented here were obtained using such VI's, and every input and output is described in Appendix B.

4.2   Discussion

The difference between the EVM$_{RMS}$ obtained at the gamma bus and the one obtained at the ADCs is attributed to the different number of taps used in the RRC filters of the TRX and the software implementation. A larger number of taps leads to a better signal quality and lower EVM. Besides, the RRC matched filters applied to the signal observed at the ADCs are implemented in software and have an ideal behavior.

A problem was found in the signal measured at the gamma bus. The values of the original 4 samples representing each symbol fall inside certain regions and create gaps between each region. This effect is attributed to the quantization noise caused by round-off errors, since 5 bits are used to represent each sample. Figure 4-7 shows the I+jQ magnitude of a signal obtained from the gamma bus. The signal has not been interpolated. The optimal measurement position is t_{opt}=2; observe the gaps caused by quantization noise. Once the samples were extracted at t_{opt} and plotted at the IQ constellation, they follow a pattern of circular lines separated by small gaps. See Figure 4-8.

In order to solve this problem, the original signal was low-pass filtered prior to any processing. The low pass filter removes the quantization noise which is spread all over the spectrum. Since the bandwidth of the RRC filters equals one quarter of the sampling frequency ($0.25f_s$), the cut-off frequency of the low-pass filter must be higher than this quantity. For instance at $0.4f_s$. If the LP cut-off frequency gets closer to the RRF filters bandwidth then the quality of the signal is degraded and the EVM increased.
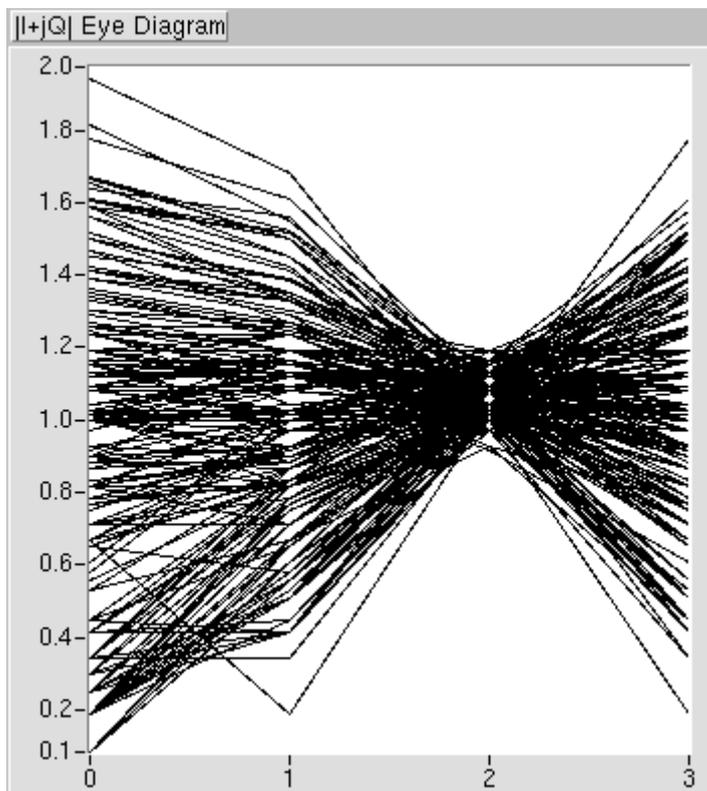
Figure 4-7. Illustration of quantization noise in the eye diagram
of the signal´s magnitude.
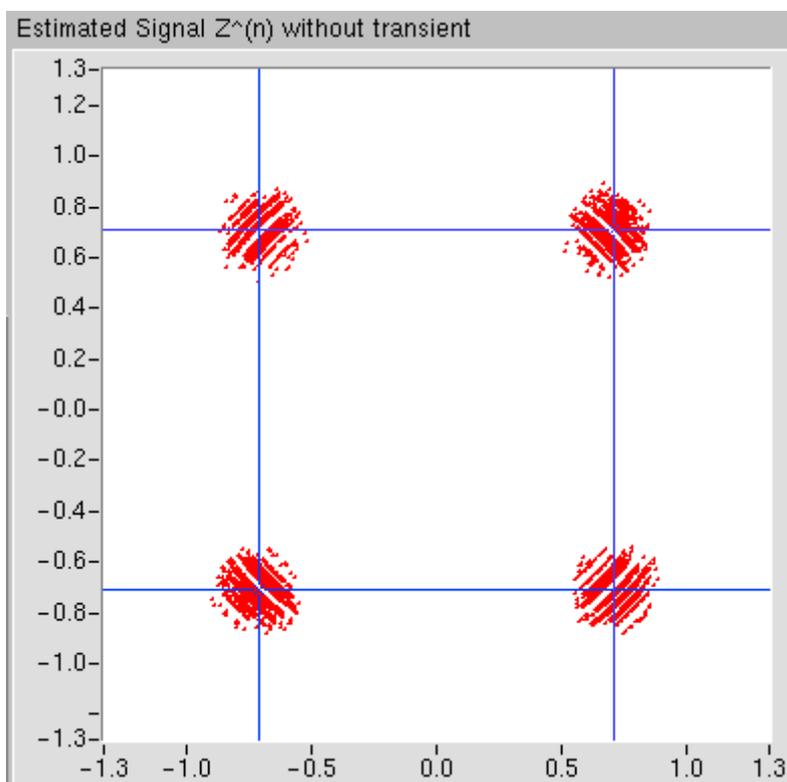
Figure 4-8. Illustration of quantization noise in the IQ constellation.

Using LP filtering the effects of quantization noise can be lowered. See Figure 4-9 which shows the same signal as in Figure 4-8 but now LP filtered.
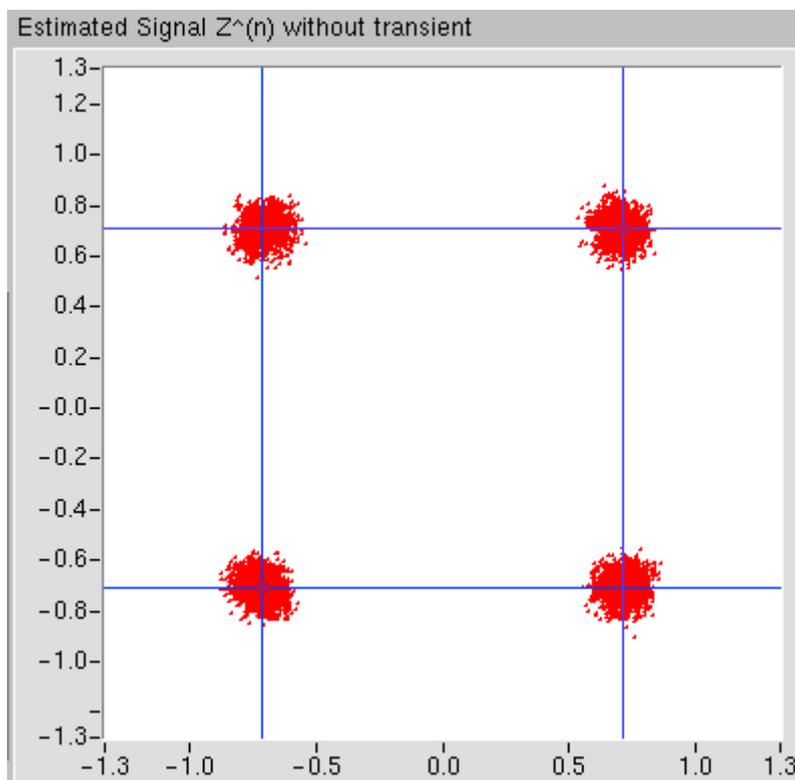

Figure 4-9. Illustration of the quantization noise removal results.

The problem becomes imperceptible at the signal measured at the ADCs where 14 bits are used to represent each sample. At the ADCs the quantization noise is negligible.

So far the results have demonstrated how the EVM measurements help the designer in the troubleshooting of signal impairments. First by showing the difference in the signal quality between two interfaces observed. Second, by making visual the effects of quantization noise. Although 3GPP standards do not specify EVM testing for the receiver side, the tool has provided a figure-of-merit of it, and its use can still be extended to other areas. We followed the definition, which refers to the transmit modulation, whenever it had applicability to the receiver side. But the reason that we used LP filtering, nevertheless it is not specified in the standard, is to reduce the effects of parameters which could lead to an erroneous measurement.

Unfortunately at this moment we do not count with another tool which could give us a comparison point of our results. However, similar outcomes are expected if any other device is used. Some interesting papers relate similar work done to measure the EVM. In Ref. [12] a method to compute EVM in GSM (8-PSK) systems is described. The device under test is a nonlinear amplifier. A steepest descent method is used to de-embed frequency offsets, amplitude changes, carrier leakage, and arbitrary phase and output power of the transmitter.

Another paper which may be of interest to the reader is Ref. [13]. This paper gives a full description of EVM as well as hints to use the measurement for troubleshooting in digital RF communications systems.

Other research papers that may also be of interest and have influenced my work are the listed in Ref. [14] to [18]. Such papers describe specific EVM applications over different scenarios.

4.3 Future work

If this project is to be continued, it is necessary to find another tool which could give us a point of a comparison. Either a commercial implementation or another algorithm so as to validate our results.

If another algorithm is to be created, it is recommended to make use of a pilot signal. This may involve some complex work in decoding but can provide aid in the compensation of frequency offsets.

It is also necessary to keep track of the evolving standards, since no method is specified to compensate against frequency offsets. This leads to an ambiguity in the interpretation of how the signal must be modified prior to EVM measurements. This leads to slight differences in the outcomes depending on the algorithms used in different measurement equipment. If EVM has been introduced to the standards, then the method to obtain it should also be standardized.

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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

5   Conclusions

We have presented the work done to measure the EVM directly at the output of the third generation WCDMA base station receivers. Algorithms for finding the optimal measurement position and frequency-offset compensation were presented.

EVM provides a figure-of-merit of the down-converted baseband signal. Although the standards refer to it as a way to measure the quality of the transmit modulation, we have proven that it can also represent a figure-of-merit for the receivers. EVM is defined as the magnitude of the vector resulting from the difference between an ideal reference signal and a measured signal. Such difference is calculated at symbol rate. The signal analyzed was a WCDMA QPSK with equal amplitude in the I and Q components. It was created by a signal generator, fed it into the RF input of the receiver and acquired at the interfaces of interest using a logic analyzer. The ideal reference signal was created by means of software.

One of the main tasks solved in this project was the implementation of an algorithm which finds the optimal measurement position. The solution presented for this problem is based on the eye diagram, specifically, in the analysis of the variance of the signal's magnitude.

Once the samples were extracted at the optimal sampling position we still faced the problem of frequency offsets introduced by the RF local oscillator. Frequency offsets can be seen as a phase rotation of the IQ constellation. The proposed solution which compensates for this effect consists of a one-tap LMS adaptive filter. It tracks the movements of the IQ constellation and provides a rough estimate of the phase rotation. Such rough estimate was used to obtain a best linear fit, considered as the compensation to be made to the signal. The reason of the linear fit is the assumption that the phase rotation is linear. Within the adaptive algorithm loop, a detector was implemented, its function is to generate a reference ideal signal which is used to compute the EVM.

Both algorithms were implemented in Labview and two interfaces were analyzed, the gamma bus and the output of the analogue to digital converters. The EVM_{RMS} measured at the gamma bus was 6.6%. The EVM_{RMS} measured at the ADCs was 4.5%. The difference is attributed to the number of taps used in the RRC filters at the receiver, which is lower than the number of taps used in the software implementation of the RRC filters applied to the ADCs signal.

A problem of quantization noise was found at the gamma bus, affecting our EVM measurement. The solution to compensate against the quantization noise effects was a LP-filter with a cutoff frequency of $0.4f_s$. Although this problem was out of the scope of project, it was necessary to correct in order to avoid errors in the measurements.

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ERICSSON ⅀	REPORT	41 (78)

Prepared (also subject responsible if other)		No.		
EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen		
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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

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Approved	Checked	Date	Rev	Reference
EMW/TB/RBC Ödgärd Andersson		February 2002	A	

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[18] Dow S., Yang J., Yen K., Matreci R., Spotted-Elk E., Pettis S., Trinh L., *"Vector Signal Characterization of 38GHz Power Amplifier with 100 Mbps QPSK Modulation,"* in Proc. IEEE MTT-S Digest, 2000, pp. 1847-1850

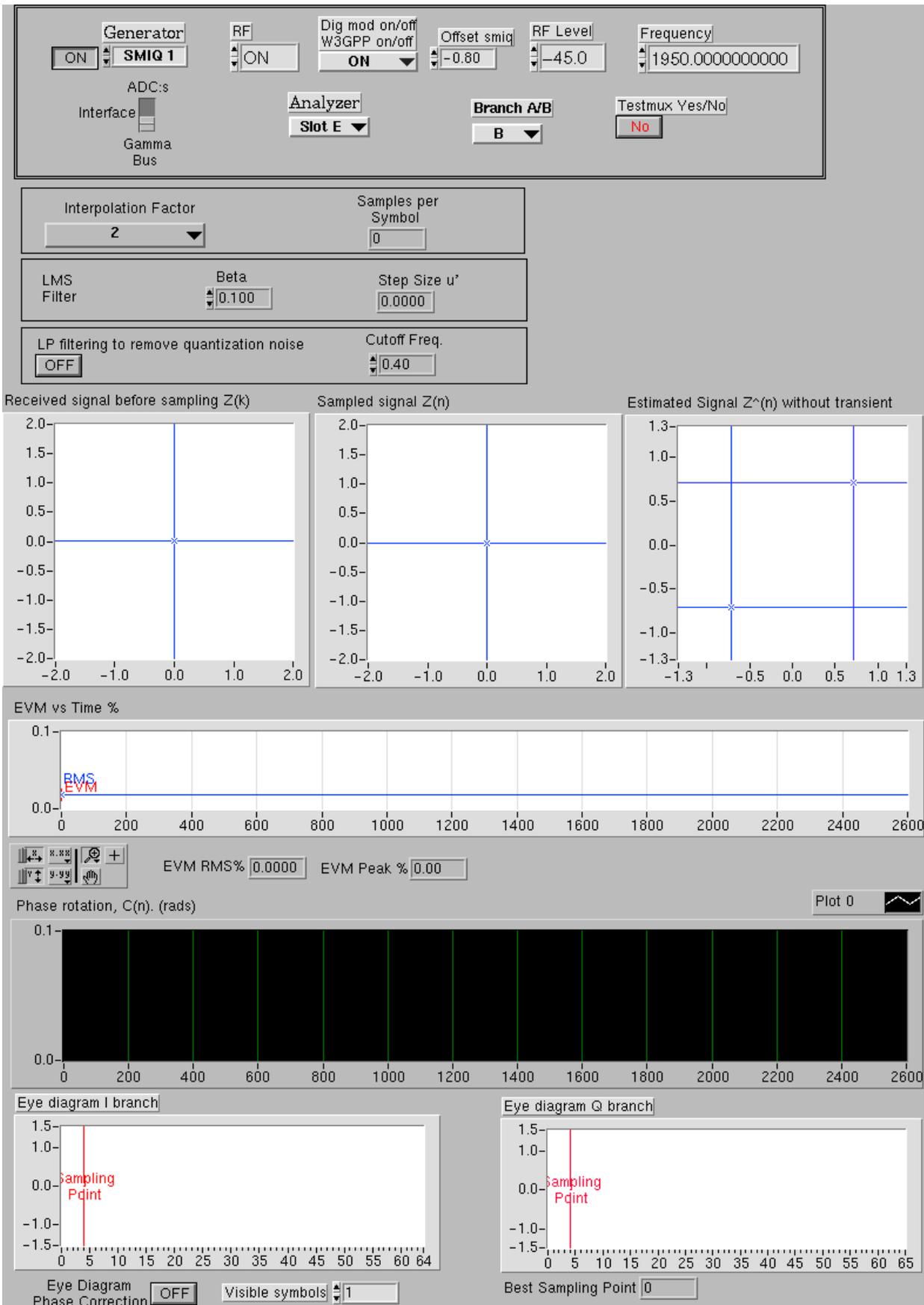| Prepared (also subject responsible if other)		No.		
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Appendix A. LabVIEW VIs

QPSK Sampling & EVM.vi
-Reads Data from: Gamma bus & ADC's at the TRX
-Measures the QPSK data at the optimal position.
-Compensates against the efects of frequency offsets.
-Computes the Error Vector Magnitude over bursts of 2560 symbols

Connector Pane

Front Panel

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Approved	Checked	Date	Rev	Reference
EMW/TB/RBC Ödgärd Andersson		February 2002	A	

Generator: ON SMIQ 1
RF: ON
Dig mod on/off W3GPP on/off: ON
Offset smiq: -0.80
RF Level: -45.0
Frequency: 1950.0000000000

ADC:s Interface — Gamma Bus

Analyzer: Slot E ▼

Branch A/B: B ▼

Testmux Yes/No: No

Interpolation Factor: 2 ▼

Samples per Symbol: 0

LMS Filter
Beta: 0.100
Step Size u': 0.0000

LP filtering to remove quantization noise: OFF
Cutoff Freq.: 0.40

Received signal before sampling Z(k)

Sampled signal Z(n)

Estimated Signal Z^(n) without transient

EVM vs Time %
RMS EVM

EVM RMS% 0.0000   EVM Peak % 0.00

Phase rotation, C(n). (rads)

Plot 0

Eye diagram I branch
Sampling Point

Eye diagram Q branch
Sampling Point

Eye Diagram Phase Correction: OFF   Visible symbols 1

Best Sampling Point 0

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		REPORT			45 (78)
Prepared (also subject responsible if other)		No.			
EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen			
Approved	Checked	Date	Rev	Reference	
EMW/TB/RBC Ödgärd Andersson		February 2002	A		

Controls and Indicators

U16 Visible symbols

TF Eye Diagram Phase Correction

TF Testmux Yes/No

U16 Branch A/B

TF Boolean

DBL Offset smiq

U16 Dig mod on/off W3GPP on/off

Generator

U16 RF

DBL RF Level

DBL Frequency

DBL Beta

I16 Interpolation Factor

TF Interface

U16 Analyzer

TF LP filtering to remove quantization noise

DBL Cutoff Freq.

I32 Best Sampling Point

Sampled signal Z(n)

Received signal before sampling Z(k)

DBL EVM Peak %

DBL EVM RMS%

Estimated Signal Z^(n) without transient

[DBL] EVM vs Time %

[SGL] Eye diagram Q branch

[SGL] Eye diagram I branch

DBL Step Size u'

[DBL] Output

[DBL]

I16 Samples per Symbol

[DBL] Phase rotation, C(n). (rads)

Block Diagram

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EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen		
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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

List of SubVIs

LMSnew.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/LMSnew.vi

Remove Mean & Normalize Magnitude.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Remove Mean & Normalize Magnitude.vi

Sample.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Sample.vi

Calculate EVM.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Calculate EVM.vi

Complex Eye Diagram.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Complex Eye Diagram.vi

Upsample phase correction.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Upsample phase correction.vi

Get_Logic_Analyzer_data.vi
/proj/gr_labview/instrument/hp/Logic_Analyzer/hp16700.llb/Get_Logic_Analyzer_data.vi

Convert_HP16700_data_testboard1.vi
/proj/gr_labview/measure/wbts/trx/RX_test/WBTS_Testboard1/BBTT_TRXctrl.llb/Convert_HP16700_data_testboard1.vi

gen_offset.vi
/proj/gr_labview/instrument/general/gen_offset.vi

Toggle_Data_Testboard.vi
/proj/gr_labview/measure/wbts/trx/RX_test/WBTS_Testboard1/BBTT_TRXctrl.llb/Toggle_Data_Testboard.vi

Convert_HP16700_data_testconn.vi
/proj/gr_labview/measure/wbts/trx/RX_test/RX measurements from test connector/RX_measure_using_HP16702.llb/Convert_HP16700_data_testconn.vi

Complex Remove mean .vi
/proj/gr_labview/measure/wbts/trx/RX_test/wbts_rxsub.llb/Complex Remove mean .vi

Normalize IQ magnitude.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Normalize IQ magnitude.vi

Complex Up and Down Converter.vi
/proj/gr_labview/measure/wictoria_rx/wic_rxsub.llb/Complex Up and Down Converter.vi

Halfband Filter.vi
/proj/gr_labview/measure/wictoria_rx/wic_rxsub.llb/Halfband Filter.vi

RRC filter.vi
/proj/gr_labview/measure/wictoria_rx/wic_rxsub.llb/RRC filter.vi

Interpolator Multiple.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Interpolator Multiple.vi

Phase_Rotation_Compensation.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Phase_Rotation_Compensation.vi

LP Filter.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/LP Filter.vi

History
"QPSK Sampling & EVM.vi History"
Current Revision: 342
rev. 1 Tue, Jun 28, 1994 01:57:21 AM Apostolos K
SubVIs of this VI were modified.
VI was converted from LabVIEW version 3.0.1 to LabVIEW version 3.1a16.

LP Filter.vi

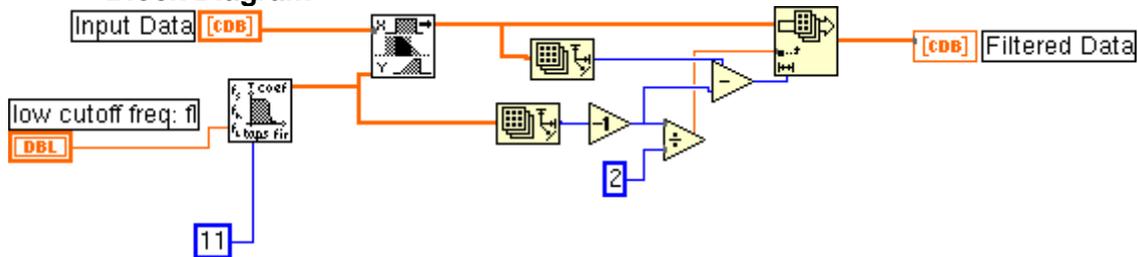Connector Pane

Front Panel

Controls and Indicators

[CDB] Input Data

[DBL] low cutoff freq: fl

[CDB] Filtered Data

Block Diagram

List of SubVIs

Complex Convolution.vi
/proj/gr_labview/measure/wictoria_rx/wic_rxsub.llb/Complex Convolution.vi
FIR Windowed Coefficients.vi
/opt/LabView/5.1/sol2/fds/lv51/vi.lib/Analysis/3filter.llb/FIR Windowed Coefficients.vi

History

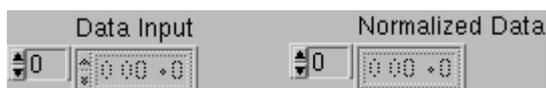"LP Filter.vi History"
Current Revision: 14

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Normalize IQ magnitude.vi

Normalizes the incomming data. The output possess absolut magnitude.

Connector Pane

Data Input —— NORM —— Normalized Data

Front Panel

Data Input	Normalized Data
0 0 00 ·0	0 0 00 ·0

Controls and Indicators

[CDB] Data Input
[CDB]

[CDB] Normalized Data
[CDB]

Block Diagram

Data Input
[CDB]

Mean

Normalized Data
[CDB]
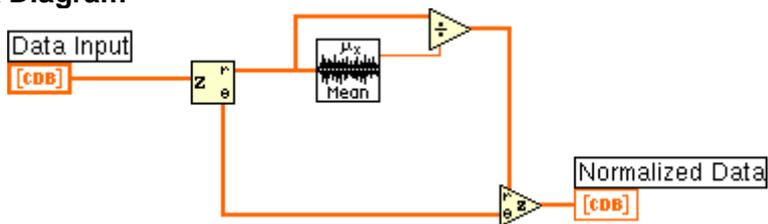
List of SubVIs

Mean.vi
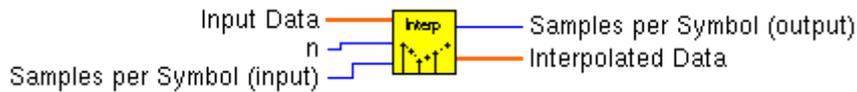/opt/LabView/5.1/sol2/fds/lv51/vi.lib/Analysis/5stat.llb/Mean.vi

History
"Normalize IQ magnitude.vi History"
Current Revision: 2

Interpolator Multiple.vi

Interpolates the Input Data by a factor of L=2^n.
n can have values equal t 1,2,3,4 (L=2,4,8,16 respectively)
n=5 turns off the interpolator

Connector Pane

Input Data ———
n —
Samples per Symbol (input) —
Interp — Samples per Symbol (output)
— Interpolated Data

Front Panel

Input Data

⇕0 ⇕0 00 ⋅0

Samples per Symbol (input)
⇕0

n
⇕0

Interpolated Data
⇕0 0 00 ⋅0

Samples per Symbol (output)
0

Plot 0 ∿

r

2.0–
1.5–
1.0–
0.5–
0.0–
0   10   20   30   40   50   60   70   80   90   100   109

Controls and Indicators

[CDB] Input Data
CDB

I16 I

I16 Samples per Symbol (input)
[CDB] Interpolated Data
CDB

I16 Samples per Symbol (output)

[DBL] I

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EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen		
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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

Block Diagram
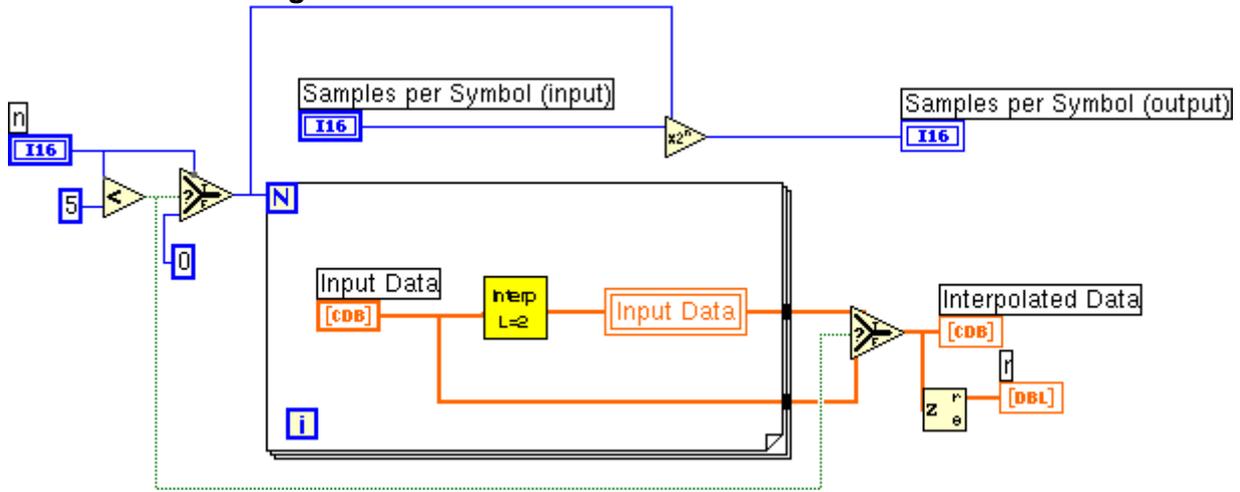
List of SubVIs

Interpolator Factor 2.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Interpolator Factor 2.vi

History

"Interpolator Multiple.vi History"
Current Revision: 12

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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

Interpolator Factor 2.vi

Interpolates the input data by a factor of two, first by up-sampling (filling up with zeroes), and then by filtering the data with a half-band LP filter. The output posses 2N-1 samples, where N is the number of samples of the original sequence.

Connector Pane

Input Data — Interp L=2 — Interpolated Data

Front Panel

Controls and Indicators

[CDB] Input Data
[CDB]
[CDB] Interpolated Data
[CDB]
[DBL] Original Signal
[DBL] Up-Sampled Signal
[DBL] Interpolated Signal

Block Diagram

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EMW/TB/RBC Ödgärd Andersson		February 2002	A	
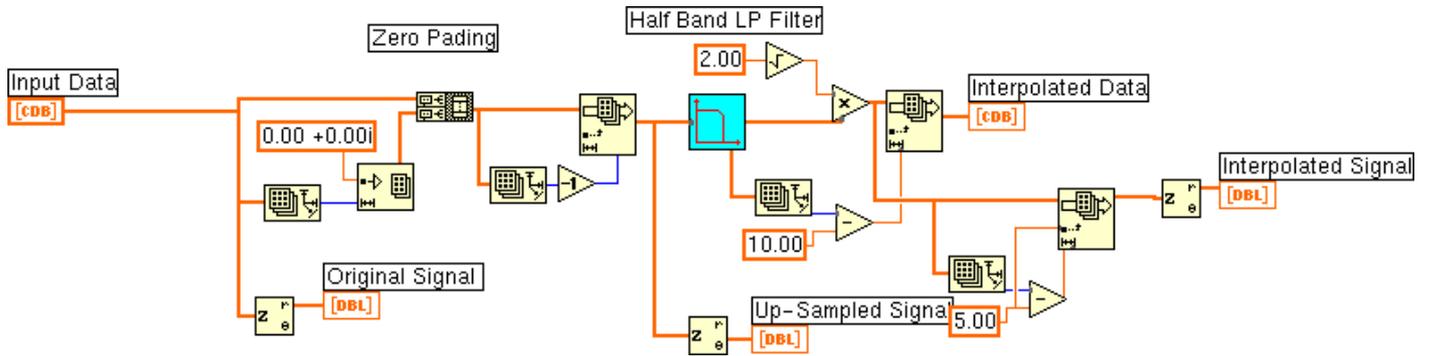
List of SubVIs

Halfband Filter.vi
/proj/gr_labview/measure/wictoria_rx/wic_rxsub.llb/Halfband Filter.vi

History
"Interpolator Factor 2.vi History"
Current Revision: 9

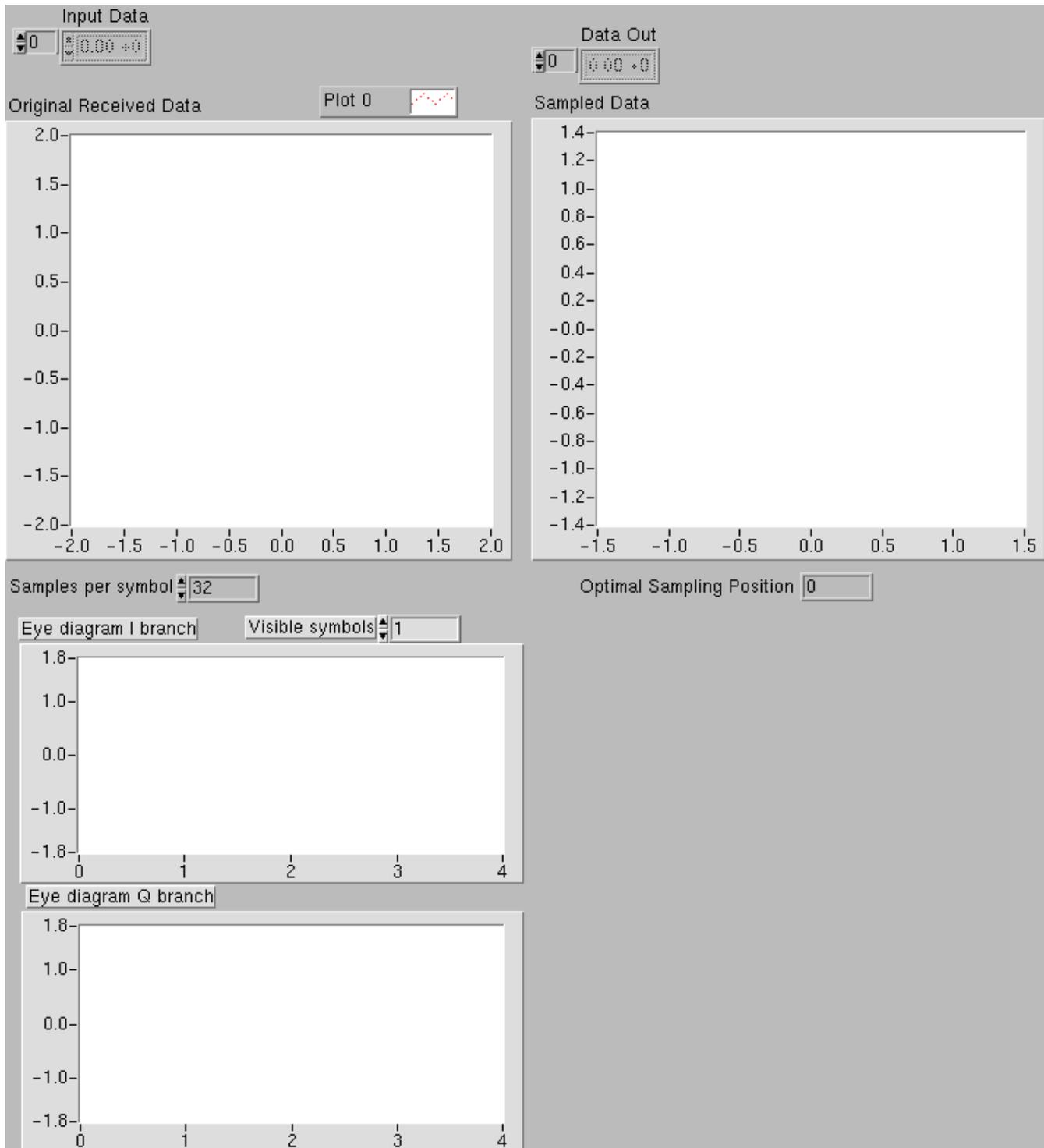| ERICSSON ≋				
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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

Sample.vi

Finds the optimal measurement position and extracts the samples.
The inputs are the number of samples per symbol, and the IQ data.
The output are the sampled data and the instant at which it was sampled.

Connector Pane

Front Panel

Input Data

0 | 0.00 +0 |

Data Out

0 | 0.00 +0 |

Original Received Data   Plot 0

Sampled Data

Samples per symbol 32

Optimal Sampling Position 0

Eye diagram I branch   Visible symbols 1

Eye diagram Q branch

Controls and Indicators

U16 Samples per symbol
Number of samples per symbol

[CDB] Input Data
[CDB]

U16 Visible symbols
U16 Optimal Sampling Position

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EMW/TB/RB Osvaldo Mendoza			No. EMW/TB/RB-02:003 Uen		
Approved	Checked		Date	Rev	Reference
EMW/TB/RBC Ödgärd Andersson			February 2002	A	

[CDB] Data Out
CDB

[▦] Original Received Data
[SGL] Eye diagram Q branch
[SGL] Eye diagram I branch
[▦] Sampled Data

Block Diagram
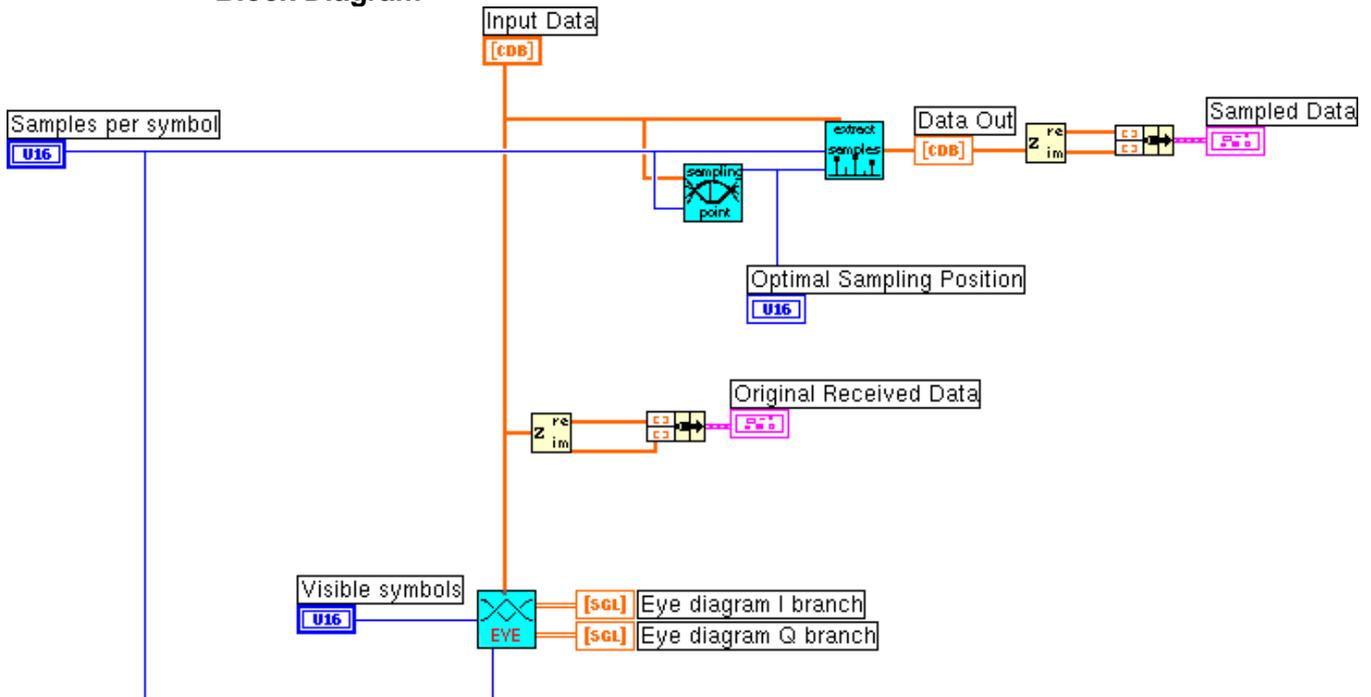
List of SubVIs

Find Sampling Point.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Find Sampling Point.vi
Extract Samples.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Extract Samples.vi
Complex Eye Diagram.vi
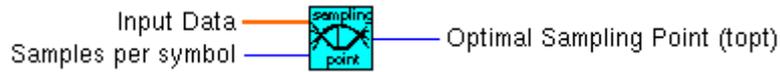/proj/gr_labview/measure/Osvaldo_temp/EVM/Complex Eye Diagram.vi

History
"Sample.vi History"
Current Revision: 27

Find Sampling Point.vi

Returns the optimal sampling point for a set of data.

Connector Pane

Input Data ————
                        sampling
                        ⧓⧓   —— Optimal Sampling Point (topt)
Samples per symbol ——   point

Front Panel

Input Data

|I+jQ| Eye Diagram

Variance

Optimal Sampling Point (topt) 0

Samples per symbol 0
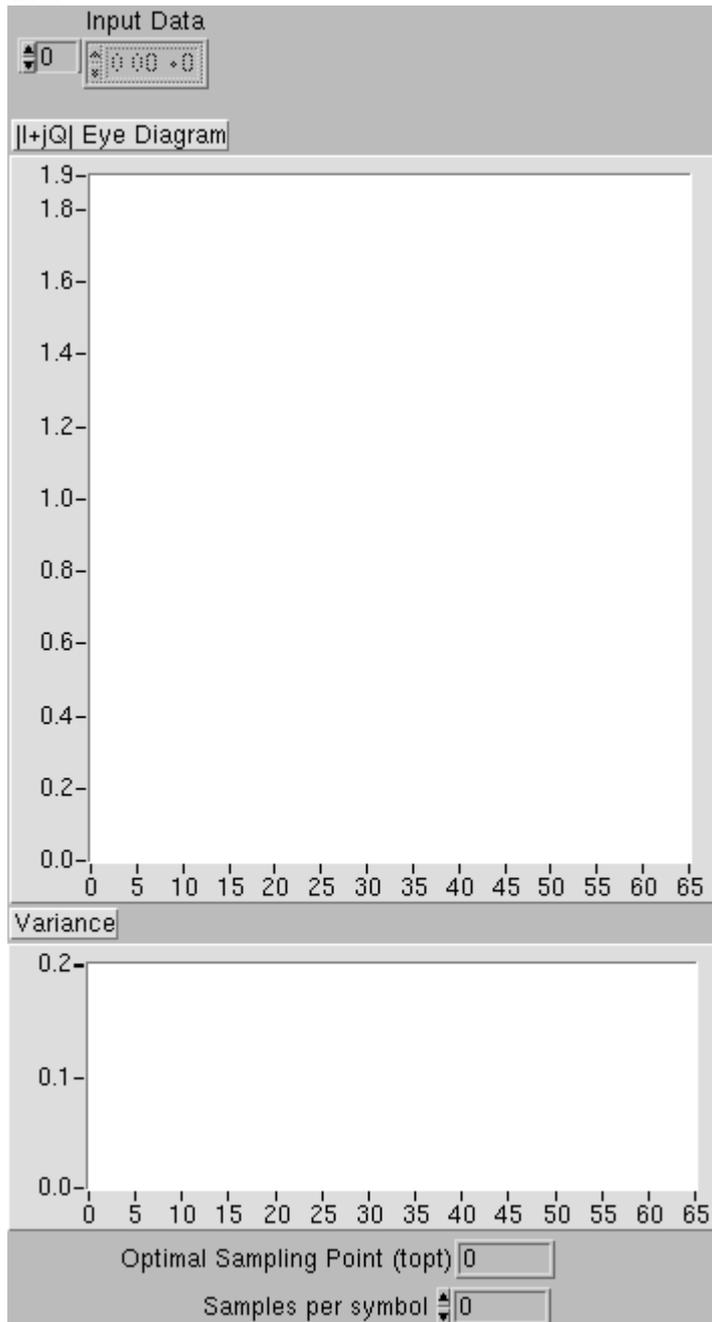
Controls and Indicators

Input Data

[CDB] Data in

ERICSSON		Open REPORT			59 (78)
Prepared (also subject responsible if other)		No.			
EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen			
Approved	Checked	Date	Rev	Reference	
EMW/TB/RBC Ödgärd Andersson		February 2002	A		

[CDB] Samples per symbol
[U16] Number of samples per symbol
[U16] Optimal Sampling Point (topt)
[DBL] |I+jQ| Eye Diagram
[DBL] Variance

Block Diagram

List of SubVIs

σ_x^2 Variance.vi
/opt/LabView/5.1/sol2/fds/lv51/vi.lib/Analysis/5stat.llb/Variance.vi

History

"Find Sampling Point.vi History"
Current Revision: 23

ERICSSON 💳		Open REPORT			60 (78)
Prepared (also subject responsible if other)		No.			
EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen			
Approved	Checked	Date	Rev	Reference	
EMW/TB/RBC Ödgärd Andersson		February 2002	A		

Extract Samples.vi
Extracts the samples at the indicated position.

Connector Pane
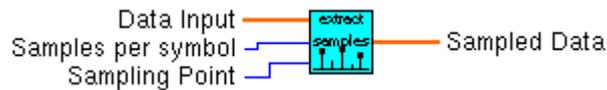
Front Panel

Controls and Indicators

U16 Samples per symbol

U16 Sampling Point

[CDB] Data Input

CDB

[CDB] Sampled Data

CDB

Block Diagram
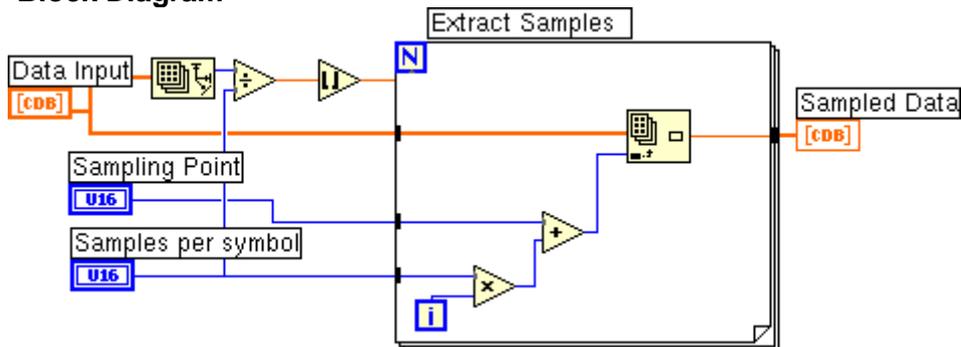
List of SubVIs

History
"Extract Samples.vi History"
Current Revision: 11

LMSnew.vi

Applies a Least Mean-Square transversal filter to the received QPSK signal Z(k).
This VI detects the received symbols and compensates for variations of phase.
The step size is determined by the Beta (0<Beta<1)
The default number of taps is 1
The outputs modiified version of the measured sigal signal Z^(n), and the detected signal S(n), both with and without the transient.
The effects of frequency offsets have been removed in Z^(n).

Connector Pane

Front Panel

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EMW/TB/RBC Ödgärd Andersson			February 2002	A		

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EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen		
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Controls and Indicators

[CDB] Received Signal
[CDB]
[U16] Number of taps
[DBL] Beta
[TF] Linear Fit of phase compensation
[CDB] W
[CDB] Element Data
[DBL] Phase Error
[CDB] Detected Signal S(n)
[CDB] Detected Symbol
[DBL] Wn Magnitude
[DBL] Wn Phase (rads)
[DBL] Squared Error
[DBL] Step Size u'
[U16] Transient time
[CDB] Ideal Reference S(n) without transient
[CDB] Detected Symbol
[CDB] Modified version of the measured waveform Z^(n) without transient
[CDB] X * Y
[DBL] Misadjustment
[DBL] EVM vs Time
[CDB] G(n)
[CDB]
[DBL] Phase Error mean
[DBL] C(n) & G(n)
[CDB] C(n)
[CDB]
[CDB] G(n) without transient
[CDB]

Block Diagram

List of SubVIs

Complex Dot Product.vi
/opt/LabView/5.1/sol2/fds/lv51/vi.lib/Analysis/7linalg.llb/Complex Dot Product.vi

Variance.vi
/opt/LabView/5.1/sol2/fds/lv51/vi.lib/Analysis/5stat.llb/Variance.vi

Linear Fit.vi
/opt/LabView/5.1/sol2/fds/lv51/vi.lib/Analysis/6fits.llb/Linear Fit.vi

Unlimited phase.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Unlimited phase.vi

Detection.vi
/proj/gr_labview/measure/Osvaldo_temp/EVM/Detection.vi

History

"LMSnew.vi History"
Current Revision: 70

ERICSSON ⚡		Open REPORT			66 (78)
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EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen			
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EMW/TB/RBC Ödgärd Andersson		February 2002	A		

Detection.vi

Maximum likelyhood detection

For every received symbol selects the ideal symbol with the closest distance

Connector Pane

Received Symbol ——— [QPSK detection] ——— Detected Symbol

Front Panel

Received Symbol

0.00 +0

Detected Symbol

0.00 +0
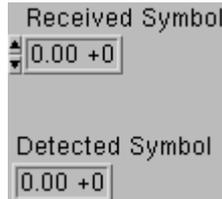
Controls and Indicators

[CDB] Received Symbol

[CDB] Detected Symbol

Block Diagram

0

| 0.70711 +0.70711i |
| -0.70711 +0.70711i |
| -0.70711 -0.70711i |
| 0.70711 -0.70711i |

Received Symbol [CDB]

Detected Symbol [CDB]
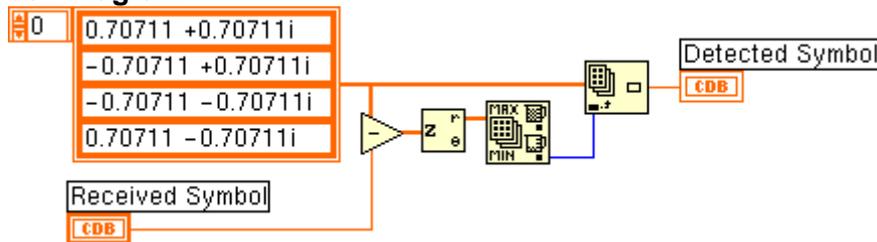
List of SubVIs

History

"Detection.vi History"

Current Revision: 15

Prepared (also subject responsible if other)			No.		
EMW/TB/RB Osvaldo Mendoza			EMW/TB/RB-02:003 Uen		
Approved	Checked		Date	Rev	Reference
EMW/TB/RBC Ödgärd Andersson			February 2002	A	

Phase_Rotation_Compensation.vi

Extracts the phase of the original data array, compares it an ideal reference and removes the mean.

Connector Pane
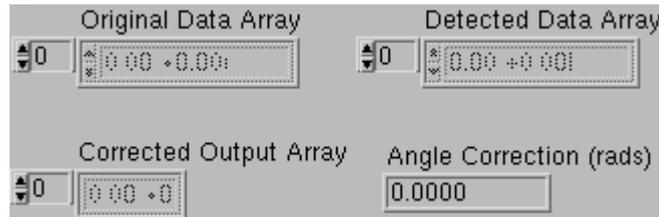
Original Data Array —— [phase rotation] —— Angle Correction (rads)
Detected Data Array —— —— Corrected Output Array

Front Panel

Original Data Array   Detected Data Array
0   0.00 +0.00i   0   0.00 +0.00i

Corrected Output Array   Angle Correction (rads)
0   0.00 +0   0.0000

Controls and Indicators

[CDB] Detected Data Array

[CDB] Original Data Array

[CDB] Corrected Output Array

[DBL] Angle Correction (rads)

Block Diagram
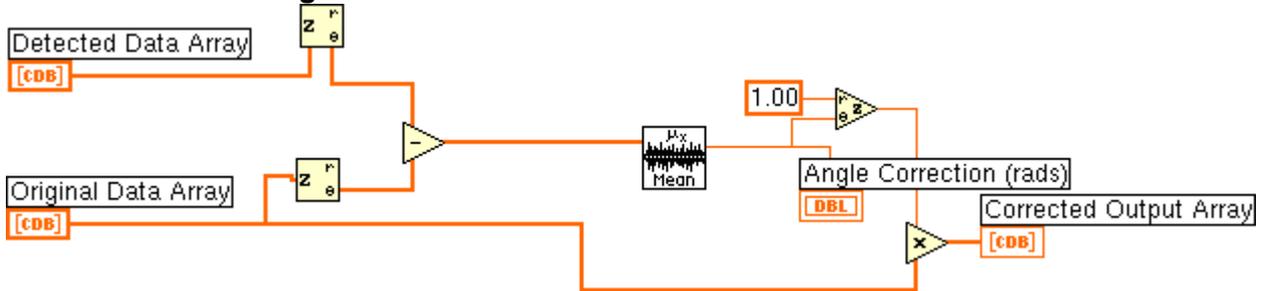
List of SubVIs

Mean.vi
/opt/LabView/5.1/sol2/fds/lv51/vi.lib/Analysis/5stat.llb/Mean.vi

---

History

"Phase_Rotation_Compensation.vi History"
Current Revision: 8

Unlimited phase.vi

Transforms the phase limited to a range of -3.14<phase<3.14 rads to a unlimited range -inf<phase<inf

Connector Pane

Front Panel
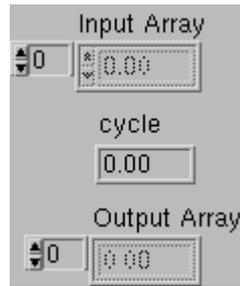
Controls and Indicators

[DBL] Input Array
[DBL]
[DBL] cycle
[DBL] Output Array
[DBL]

Block Diagram
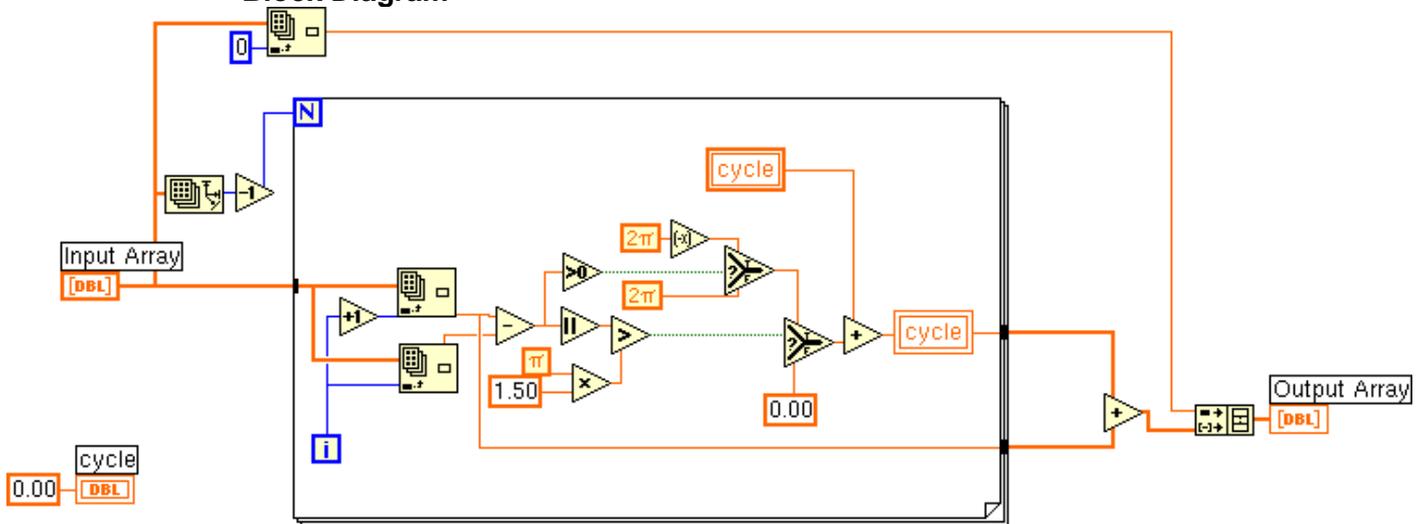
List of SubVIs

History
"Unlimited phase.vi History"
Current Revision: 5

		Open	
ERICSSON ≩		REPORT	69 (78)

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EMW/TB/RB Osvaldo Mendoza			EMW/TB/RB-02:003 Uen		
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Complex Eye Diagram.vi

Complex Eye Diagram : Prints eye diagrams for I and Q for incoming RX data. The data must be corrected for phase, amplitude and time shifts.

Connector Pane

Front Panel

Controls and Indicators

TF stop
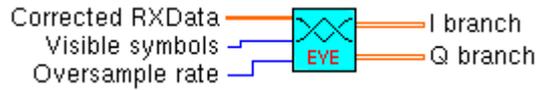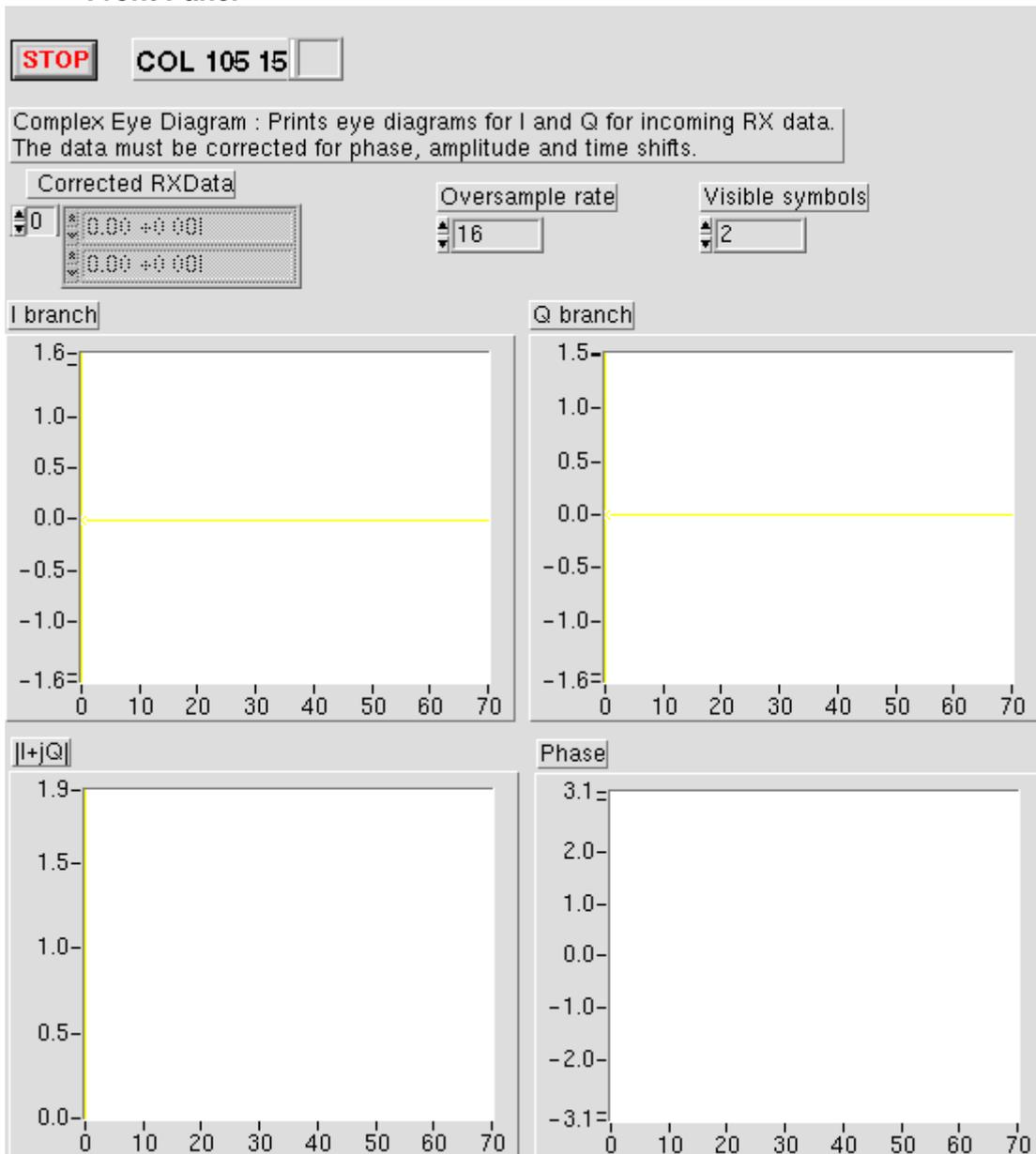
U16 Oversample rate

CSG Corrected RXData

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U16 Visible symbols
abc Rev
[SGL] I branch
[SGL] Q branch
[SGL] |I+jQ|
[SGL] Phase

Block Diagram
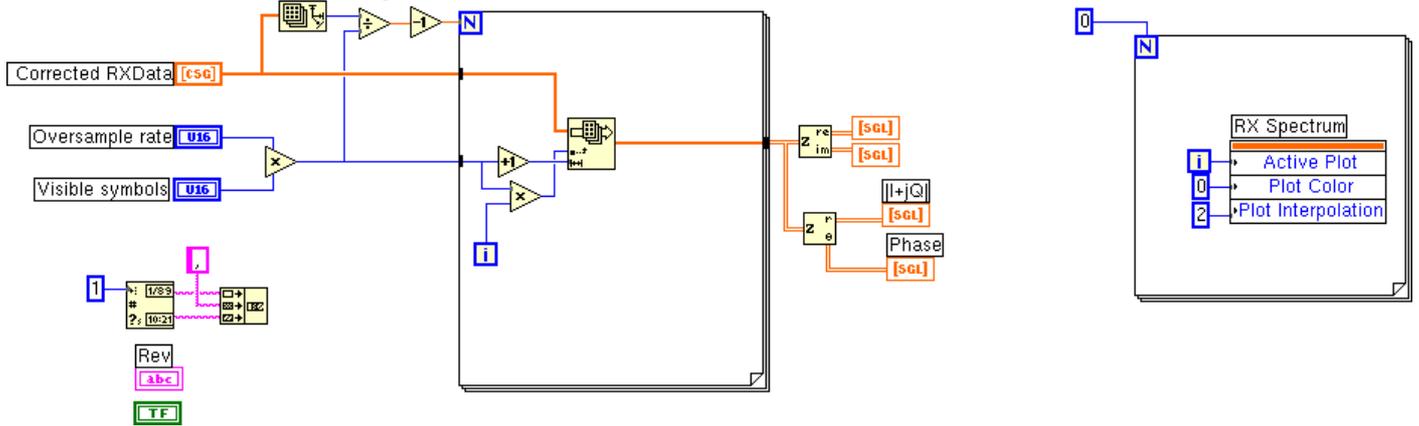
List of SubVIs

History
"Complex Eye Diagram.vi History"
Current Revision: 95

		Open		
ERICSSON ⧓		REPORT		71 (78)
Prepared (also subject responsible if other)		No.		
EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen		
Approved	Checked	Date	Rev	Reference
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Calculate EVM.vi

Calculates the RMS Error Vector Magnitude given the received signal Z(k) and a reference ideal signal S(k).

EVM = sqrt(sum(|Z(k)|^2)/sum(|S(k)|^2))x100

Connector Pane
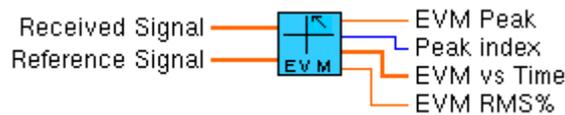
Front Panel
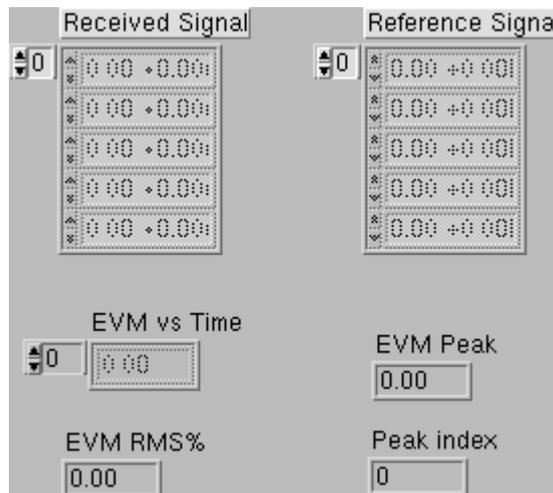
Controls and Indicators

[CDB] Received Signal
[CDB]

[CDB] Reference Signal
[CDB]

[DBL] EVM RMS%
[DBL] EVM Peak
[DBL] EVM vs Time
[DBL]
[I16] Peak index

Block Diagram

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List of SubVIs

Complex Dot Product.vi
/opt/LabView/5.1/sol2/fds/lv51/vi.lib/Analysis/7linalg.llb/Complex Dot Product.vi

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History

"Calculate EVM.vi History"
Current Revision: 22

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EMW/TB/RB Osvaldo Mendoza		EMW/TB/RB-02:003 Uen		
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Upsample phase correction.vi

Up-samples the phase correction.

Input: -Phase compensation to be applied (Burst containing N elements)

-Oversampling (T)

Output:

-Phase compensation to be applied (Burst containing NT elements)

Connector Pane

Front Panel

Controls and Indicators

[CDB] Data Input

[I32] Oversampling

[CDB] Phase

Block Diagram

List of SubVIs

History

"Upsample phase correction.vi History"

Current Revision: 4

Appendix B. I/O Summary

Table 0-1 and Table 0-2 summarise the inputs and outputs for the virtual instruments used in the implementation.

Inputs

SubVI	Input	Description	Type
Normalize IQ Magnitude.vi	Data Input	Data to be normalized	Double precision complex array
Interpolator Multiple.vi	Input Data	Data to be interpolated	Double precision complex array
Interpolator Multiple.vi	x	Adjusts the interpolation factor L in the following basis L=2^x. The variable x can have values from 1 to 4, (L=2,4,8,16). x=5 means interpolation off	Word
Interpolator Multiple.vi	Samples per Symbol (input)	Number of samples per symbol at the input	Word
Interpolator Factor 2.vi	Input Data	Data to be interpolated	Double precision complex array
Sample.vi	Samples per symbol	Number of samples per symbol at the input	Unsigned word
Sample.vi	Input Data	Data to be sampled (measured at the optimal sampling point)	Double precision complex array
Find Sampling Point.vi	Input Data	Data where the optimal sampling point is to be found	Double precision complex array
Find Sampling Point.vi	Samples per Symbol	Number of samples per symbol at the input	Unsigned word
Extract Samples.vi	Data Input	Data to be sampled (measured at the optimal sampling point)	Double precision complex array
Extract Samples.vi	Samples per Symbol	Number of samples per symbol at the input	Unsigned word
Extract Samples.vi	Sampling Point	Optimal Sampling Point (topt)	Unsigned word
LMS.vi	Received Signal	Data to be detected and compensated for frequency offsets (phase rotation)	Double precision complex array
LMS.vi	Beta	Adjusts the step size. 0<Beta<1. Its default value is 0.1	Double precision
Detection.vi	Received Symbol	Symbol to be detected	Double precision complex
LP Filter.vi	Input Data	Data to be filtered through the low pass filter.	Double precision complex array
Calculate EVM.vi	Received Signal	Measuered signal over which EVM is to be calculated	Double precision complex array
Calculate EVM.vi	Reference Signal	Ideal reference signal	Double precision complex array

Table 0-1 Virtual Instruments Inputs

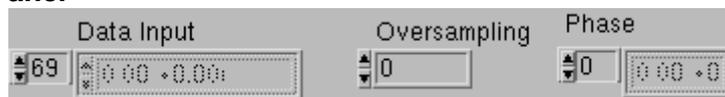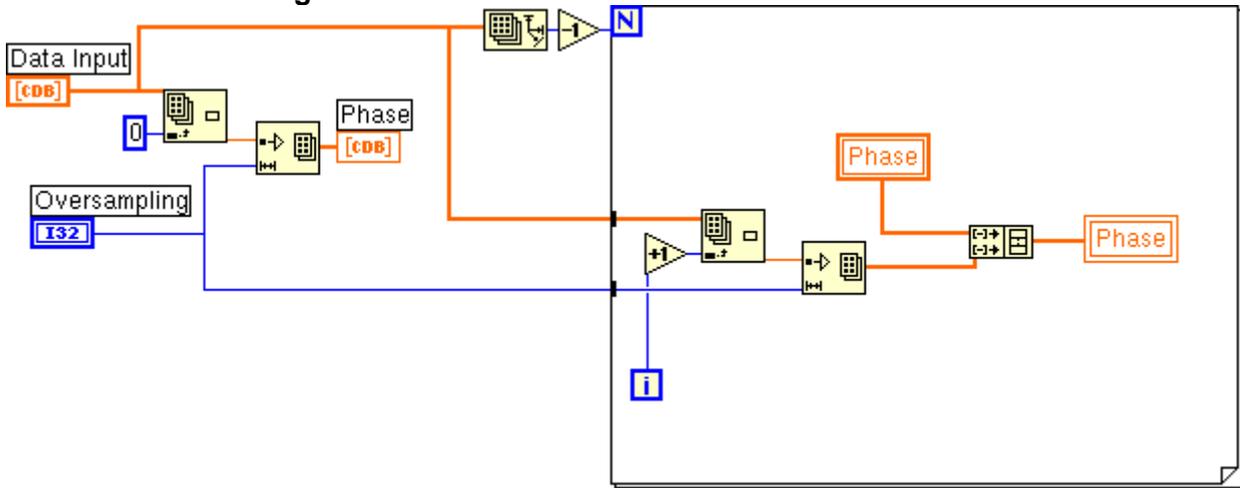| Prepared (also subject responsible if other)				
EMW/TB/RB Osvaldo Mendoza		No. EMW/TB/RB-02:003 Uen		
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Outputs

SubVI	Output	Description	Type
Normalize IQ Magnitude.vi	Normalized Data	Normalized data with magnitude mean equal to one over the burst	Double precision complex array
Interpolator Multiple.vi	Samples per Symbol (output)	Number of samples per symbol after interpolation.	Word
Interpolator Multiple.vi	Interpolated Data	Data after interpolation	Double precision complex array
Interpolator Factor 2.vi	Interpolated Data	Data after interpolation with *L*=2	Double precision complex array
Sample.vi	Optimal Sampling Position	Optimal measurement position (topt). Samples are extracted at nT+topt, where n=0...N-1, and T is the symbol period. N is the number of symbols per burst.	Unsigned Word
Sample.vi	Data Out	Extracted samples at *nT+topt*, where n=0...N-1, and T is the symbol period. N is the number of symbols per burst.	Double precision complex array
Find Sampling Point.vi	Optimal Sampling Point (topt)	Optimal sampling point for the data provided.	Unsigned word
Extract Samples.vi	Sampled Data	Extracted samples at nT+topt, where n=0...K-1, and T is the symbol period. N is the number of symbols per burst.	Double precision complex array
LMS.vi	*C(n)*	Contains an array of coefficients with magnitude equal to one and the phase corresponding to the frequency offset at the given instant.	Double precision complex array
LMS.vi	Estimated Signal $\hat{Z}(n)$	Contains the estimated signal $\hat{Z}(n)$ including the transient period	Double precision complex array
LMS.vi	Detected Signal *S(n)*	Contains the detected signal *S(n)* including the transient period	Double precision complex array
LMS.vi	Estimated Signal $\hat{Z}(n)$ without transient	Contains the estimated signal $\hat{Z}(n)$ without the transient period	Double precision complex array
LMS.vi	Detected Signal *S(n)* without transient	Contains the detected signal *S(n)* without the transient period	Double precision complex array
LMS.vi	Transient time	Time that takes to the algorithm to converge. (Settling time)	Unsigned word
LMS.vi	Step Size µ'	Shows the step size µ'. µ' equals Beta times µ	Double precision
Detection.vi	Detected Symbol	Detected ideal symbol, closest to the received one.	Double precision complex
LP Filter.vi	Filtered Data	Data passed through the low pass filter	Double precision complex array
Calculate EVM.vi	EVM Peak	Peak value of the EVM over the burst	Double precision
Calculate EVM.vi	Peak index	Position of the EVM peak over the burst	Unsigned word
Calculate EVM.vi	EVM vs Time	Contains the error vector magnitude variations over the time at the measuring points.	Double precision array
Calculate EVM.vi	EVM RMS %	Square root of the ratio of the mean error vector power to the mean reference power expressed as a %	Double precision

Table 0-2 Virtual Instruments Outputs

Prepared (also subject responsible if other)				
EMW/TB/RB Osvaldo Mendoza		No. EMW/TB/RB-02:003 Uen		
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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

Appendix C. Labview Main Panel Description

The main panel consists of a set of displays, indicators and controls. The parameters to be introduced by the user are the following:

- Interface: Can be switched between the gamma bus or the analogue to digital converters.

- Interpolation Factor: Some pre-defined values have been provided (2,4,8,16,interpolation off). The final number of points per symbol after interpolation is given by the original number of points per symbol times the interpolation factor. As the interpolation factor is increased the accuracy of the EVM measurement is improved.

- Beta: Adjusts the step size for the LMS algorithm. A default value of 0.1 is provided.

- Eye Diagram Correction: If turned on displays the eye diagram of the corrected signal over which the EVM calculation is done. This is just a visual tool and it is recommended to turn it off to speed up the time that takes to run the whole program.

- Eye Diagram Visible Symbols: Determines the number of symbols to be displayed at the eye diagram.

The algorithms generate as outputs the following parameters:

- EVM_{RMS}: This is the root mean square Error Vector Magnitude.

- EVM_{PEAK}: Highest EVM value over the burst. Represents the worst case.

- Numer of samples per symbol: This is the oversampling after interpolation.

- Step size μ': Step size used in the LMS algorithm.

- Best Sampling point: Indicates the optimal measurement position.

And the graphs displayed are:

- Received signal $Z(k)$

- Sampled signal $Z(n)$

- Estimated signal $\hat{Z}(n)$ without transient

- EVM vs Time

- Phase rotation (rads)

- Eye Diagram, I and Q branches.

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EMW/TB/RBC Ödgärd Andersson			February 2002	A	

Appendix D. Equipment

List of Devices

Device	Description	EMW SN	Factory SN
Signal Generator	Rohde & Shwarz SMIQ 03	YG 0711	847867/003
Logic Analyser	Hewllet Packard 16702A	A03117	NS40020218
Test Board	Test Board	ROA1174715/1	-
Device Under Test	TRX Board	ROJ61982/3..,R1A,|A533946963	-
Software	LabVIEW 5.1	-	-

Table 0-1

Signal Generator General Settings

3GPP Version	3.2.0 (Release 99)
Chip Rate	3.84 Mcps
Link Direction	Down/Forward
Sequence Length	1 Frame
Clipping Level	100%
Filter	WCDMA 0.22
RF Power	-45 dBm

Table 0-2

Prepared (also subject responsible if other)				No.			
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EMW/TB/RBC Ödgärd Andersson		February 2002	A	

Appendix E. Gamma Bus Format

30.72 MHz clock

Bit-0	Data	I1	Q1	I1	Q1	I1	Q1	I1	Q1	I1	Q1	I1	Q1	I1	Q1	I1	Q1
Bit-1	Data	I2	Q2	I2	Q2	I2	Q2	I2	Q2	I2	Q2	I2	Q2	I2	Q2	I2	Q2
Bit-2	Data	I3	Q3	I3	Q3	I3	Q3	I3	Q3	I3	Q3	I3	Q3	I3	Q3	I3	Q3
Bit-3	Data	I4	Q4	I4	Q4	I4	Q4	I4	Q4	I4	Q4	I4	Q4	I4	Q4	I4	Q4
Bit-4	Data	I5	Q5	I5	Q5	I5	Q5	I5	Q5	I5	Q5	I5	Q5	I5	Q5	I5	Q5
Bit-5	FFE	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
Bit-6	FFE	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
Bit-7	Parity	cp	cp	cp	cp	cp	cp	cp	cp	cp	cp	cp	cp	cp	cp	cp	cp
Bit-8	Factor/ID	0	F5	F4	F3	F2	F1	F0	Id5	Id4	Id3	Id2	Id1	Id0	X	X	X
Bit-9	Strobe	1	0/1	0	0	0	0	0	0	1	0	0	0	0	0	0	0