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A multicarrier modem architecture for VDSL

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A multicarrier modem architecture for VDSL

Michiel Esvelt

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Abstract

To provide high bit-rates to the customer a number of new access technologies is being developed: cable modems, optical fiber transmission systems, and Digital Subscriber Line (DSL) technologies.

This report concentrates on a Discrete Multitone (DMT) multicarrier implementation of the DSL technologies. DSL uses the twisted pair of the plain old telephone system.

A simulation model is presented together with an implementation on a TMS320C40 Digital Signal Processor. Performance studies of this model show that 52 Mbit/s VDSL is possible using DMT.

Acknowledgment

I would like to express my gratitude to prof. dr. ir. W.G.M. van Bokhoven and dr. ir. L.K.J. Vandamme, from the Eindhoven University of Technologie and dr. ir. M. Engels from the Interuniversity Microelectronics Center (IMEC) for giving me the opportunity to go to Belgium for my graduation project.

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Last but not least I would like to thank my parents and brother for their never ending support during my studies.

Michiel Esvelt

Leuven, February 1998.

Contents

List of Figures

List of abbreviations

ADC	Analog to Digital Convertor
ADSL	Asymmetrical Digital Subscriber Line
ANSI	American National Standards Institute
BER	Bit Error Rate
CAP	Carrierless Amplitude Phase modulation
DAC	Digital to Analog Convertor
DFT	Discrete Fourier Transform
DMT	Discrete MultiTone
FEQ	Frequency Domain Equalizer
FFT	Fast Fourier Transform
HDSL	High speed Digital Subscriber Line
ICI	Inter Carrier Interference
ISDN	Integrated Services Digital Network
ISI	Inter Symbol Interference
LSI	Loughborough Sound Images
OFDM	Orthogonal Frequency Division Multiplexing
PISO	Parallel in Serial Out
QAM	Quadrature Amplitude Modulation
RADSL	Rate adaptive Asymmetrical Digital Subscriber Line
RFI	Radio Frequency Interference
SDSL	Symmetrical Digital Subscriber Line
SIPO	Serial In Parallel Out
TEQ	Time Domain Equalizer
VDSL	Very High speed Digital Subscriber Line

1 Introduction

Several new high-bandwidth services like video-on-demand, high-bitrate Internet access and digital audio and video broadcast systems are being developed to give the consumer various multimedia applications e.g. interactive games, High Definition and wide-screen television and cd-quality audio.

To provide these high bit-rates to the customer a number of new access technologies is being developed: cable modems, optical fiber transmission systems, and Digital Subscriber Line (DSL) technologies.

These systems bring high bitrate services (2 megabit/s up to 50 megabit/s) to the end-user. The cable modem and DSL technologies reuse the existing infrastructure and therefore reduce the investments in the network. Cable modems use the television broadcast coaxial cable and DSL uses the twisted pair of the plain old telephone system.

This report concentrates on the DSL technologies and more specific, on a multicarrier implementation of these technologies. Simulation of a multicarrier modem using Matlab and C, and implementation of this model on a Texas Instruments TMS320C40 are described.

It is a collection of algorithms and architectures found in different publications combined into one architecture with a new frame level synchronization algorithm and a new equalizer training algorithm.

This paper is a report for the Eindhoven University of Technology and IMEC to show the results of my thesis project. It is also intended as a starting point for a colleague or successor to carry on the work on multicarrier modem architectures.

First, the DSL access networks will be explained. Second, the DMT modulation scheme will be discussed. Third, the Matlab simulation model will be presented. Finally, the DSP implementation on the TMS320C40 will be elaborated.

2 Digital Subscriber Lines

2.1 An introduction to DSL

The ever growing need for more bandwidth makes people look for new ways to get a high bandwidth connection in their home. There are several ways to accomplish this (Figure 1), one possibility is to put a new optical-fiber line to every home. Another is to use the cable-TV coax for two-way communication. A third possibility is to reuse the telephone wires for a high bandwidth connection. The DSL technologies use the twisted pair used by the Plain Old Telephone System (POTS) to provide high bitrate services [7, 12].

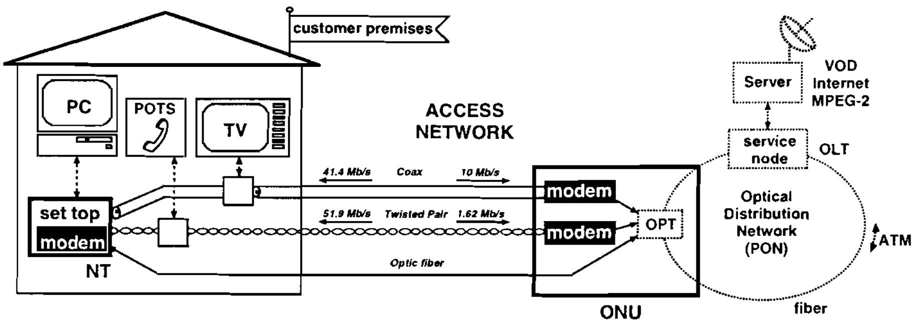


Figure 1: High bit-rate access networks

The first member of the DSL family is the known Integrated Services Digital Network (ISDN) system. DSL uses a bit-rate roughly between 160 kilobit/s and 52 Megabit/s and uses a point-to-point type of connection, as in Figure 2.

Several types of DSL connections are available:

ISDN ISDN is the first system used for bringing high bitrate services into the home. It provides two basic rate channels of 64 kbit/s and one 16 kbit/s signaling channel.

ADSL Asymmetrical Digital Subscriber Line (ADSL) uses an asymmetrical bandwidth type of service. ADSL delivers rates of up to 6.144 Mbit/s downstream and up to 640 kbit/s upstream.

RADSL Rate Adaptive Digital Subscriber Line is the rate adaptive variation of ADSL. The major benefit of this technology is its flexibility. RADSL allows to adjust the bandwidth of the DSL link to fit the need of the application and to account for the line length and quality on the fly. When ADSL is mentioned, RADSL is also included except when explicitly stated otherwise.

HDSL High-bit-rate Digital Subscriber Line (HDSL) uses a dedicated twisted copper pair to transport a full duplex E1 (2.048 Mbit/s) channel.

SDSL Symmetric Digital Subscriber Line provides symmetric (bi-directional) high-speed, variable rate communications. SDSL data rates range from 160 kbit/s to 2.048 kbit/s.

VDSL Very high-bit-rate Digital Subscriber Line technology provides both symmetric and asymmetric services with bitrates up to 52 Mbit/s.

These systems are already available for a number of years. The ISDN system is available for about 15 years. HDSL and SDSL are commercially deployed for several years. The use of ADSL is growing rapidly throughout the world. The first VDSL devices are starting to be commercially available. This report mainly focuses on the VDSL type of connections.

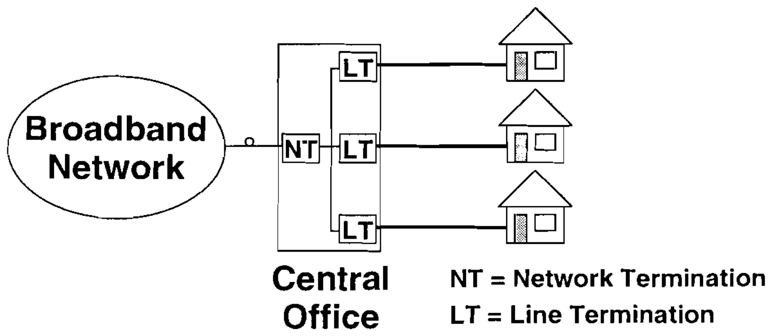


Figure 2: DSL access network configuration

ADSL, VDSL and SDSL use the spectrum of the twisted pair cable above the ISDN and POTS system. This means they are spectrally compatible with ISDN and POTS [8]. Both the POTS/ISDN and the DSL modem can be used at the same time, see Figure 3.

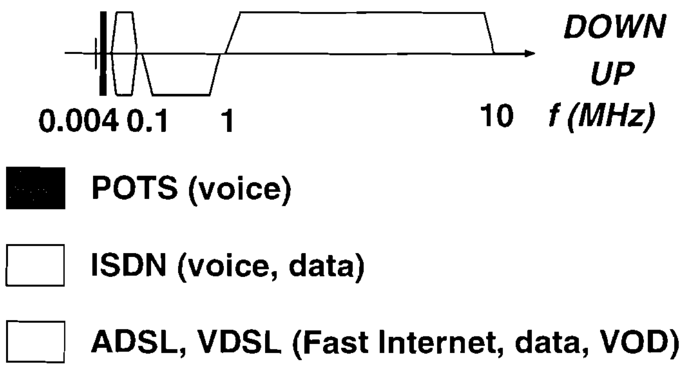


Figure 3: ADSL/VDSL spectrum with ISDN and POTS spectrum compatibility

2.2 VDSL standard

The information in this section is based on the draft ANSI VDSL standard [15]. VDSL is defined for a number of bit-rates with corresponding distances (Table 1). These values represent the worst-case situation. Higher bit-rates are possible if the cable has less distortions as the average cable mentioned in the standard. Typical DSL configurations are described in the standard, together with a test procedure to provide a reliable method to check if a device complies with the standard.

Name of Service Type	Downstream [Mb/s]	Upstream [Mb/s]	Range [m]
Asymmetric			
Short Node Asymmetric	52	6.4	300
	34 or 38.2	4.3	
Medium Node Asymmetric	26	3.2	1000
	19	2.3	
Long Node Asymmetric	13	1.6	1500
	6.5	1.6 or .8	2000
Symmetric			
Short Node Symmetric	34	34	300
	26	26	300
	19	19	300
Medium Node Symmetric	13	13	1000
Long Node Symmetric	6.5	6.5	1500
	4.3	4.3	
	2.3	2.3	

Table 1: VDSL bitrates and distances

Two types of modulation are proposed for VDSL,

1. Discrete Multitone Modulation(DMT), a multicarrier system,
2. Quadrature Amplitude Modulation (QAM)/Carrierless Amplitude Phase modulation (CAP), a single carrier system.

A standard has not been decided upon yet. This report describes the DMT type of modulation.

2.3 Available systems

To make a valid comparison this section will be limited to DSL technologies which also use a multi-carrier type modulation. The technologies described are ADSL and VDSL. For ADSL, DMT is the approved standard [14]. For VDSL standardization is ongoing for the time of writing.

ADSL modems are available for up to 10 Mbit/s downstream and 1 Mbit/s upstream supplied by a number of companies. Both DMT and CAP/QAM implementations are available.

Four companies are selling VDSL related equipment at the time of writing. Broadband technologies and Orckit are selling products. Lucent technologies is selling chips and AMATI is both selling products and licensing technology [31].

3 DMT overview

3.1 An introduction to DMT

The basic idea of Discrete Multitone modulation (DMT) is to split the available bandwidth into a large number of sub-carriers. The assumption is made that the spectrum inside a sub-carrier has flat amplitude, and linear phase distortion. DMT probes all sub-carriers separately to determine how much information can be sent per second on a sub-carrier. By doing this DMT is able to maximize the throughput of every single sub-carrier. Every sub-carrier can be turned on and off separately if for some reason it is not possible to send data on this channel. The examples in Figure 4 illustrate the basic functioning of DMT.

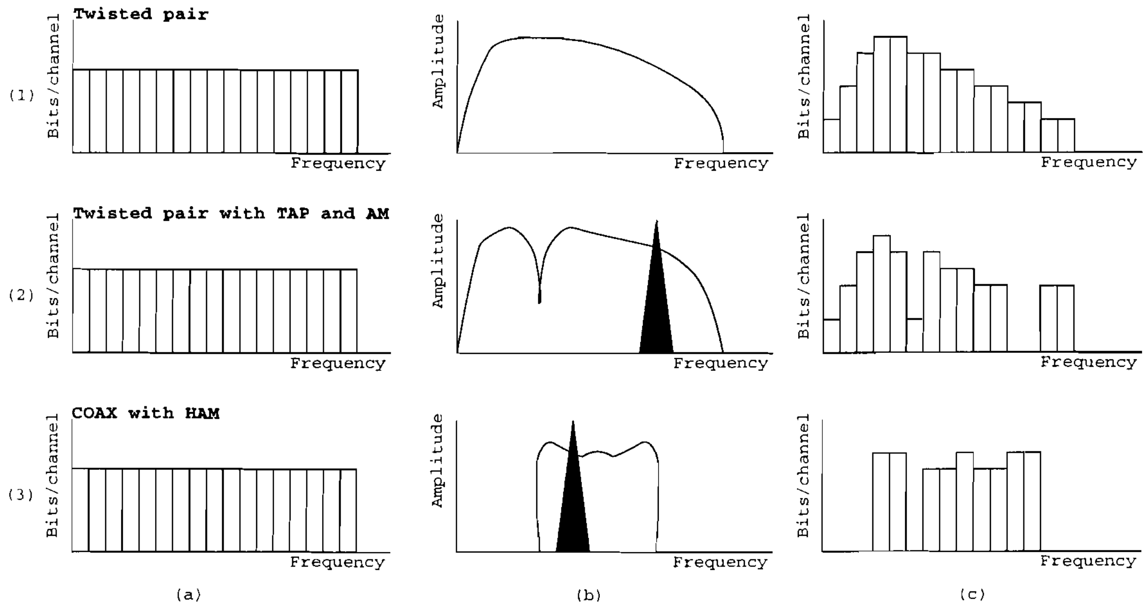


Figure 4: DMT examples

The channel characteristic of the second channel in Figure 4, part 2b, contains a notch and AM interference. During the estimation, an equal number of bits is transmitted to estimate the distortions per sub-carrier. 3b shows the result of this estimation. The sub-carriers that are affected by the noise are set to transmit a smaller number of bits than the adjacent sub-carriers. The AM interference has a similar effect, the sub-carriers that are affected are disengaged.

Several steps have to be taken to get a valid DMT frame encoded or decoded, these are the basic steps:

1. The bitstream is divided over the different carriers. Every carrier gets a number of bits (b_i) according to the characteristics of the carrier (Figure 5).

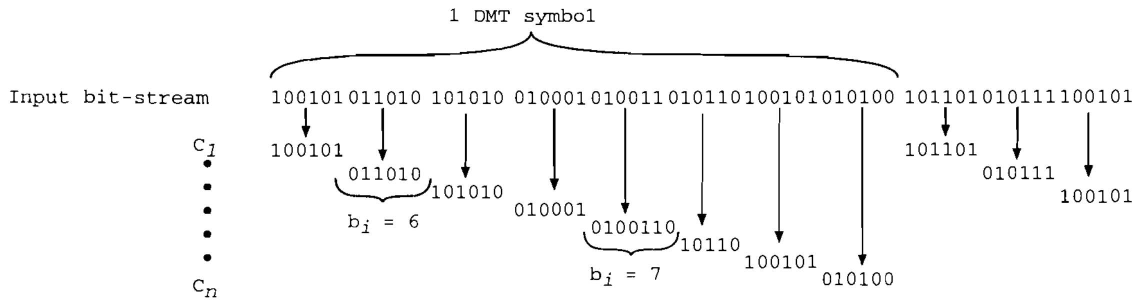


Figure 5: Tone ordering

2. For every carrier, the bits per carrier are converted into a QAM symbol in-phase and quadrature component pair according to a QAM like constellation (Figure 6).

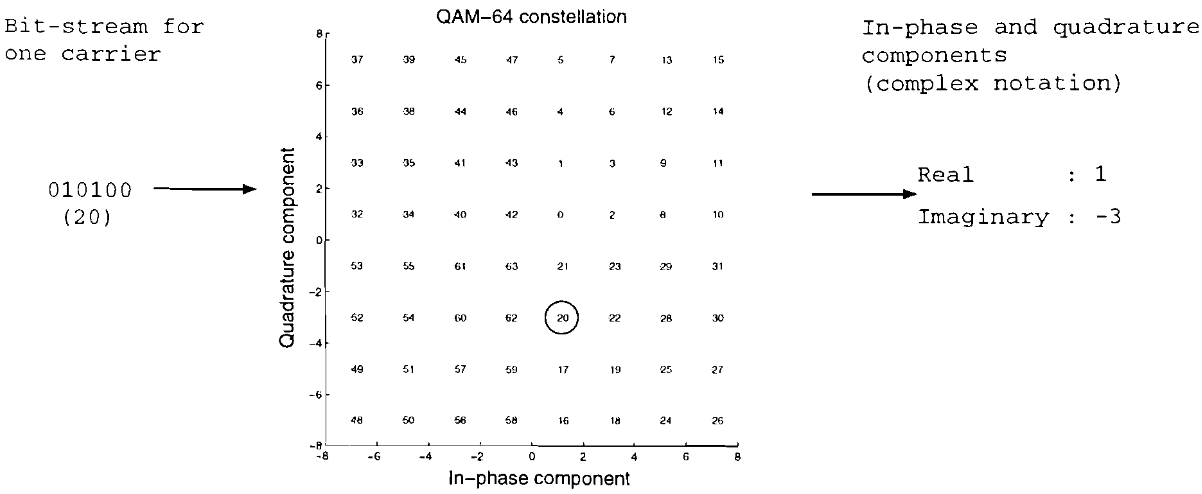


Figure 6: For every carrier, the symbol is converted to a QAM symbol in-phase and quadrature component

3. The different QAM symbol component pairs are combined to form one spectrum (Figure 7). Every QAM symbol has two pairs of values in the spectrum, one pair of real and imaginary values at F_c and one at $F_s - F_c$.

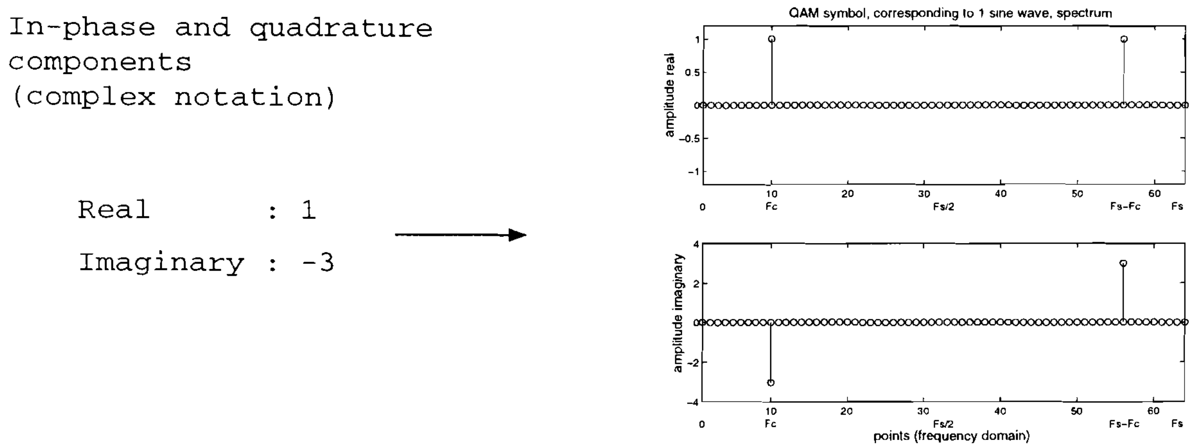


Figure 7: QAM symbol pair to spectrum mapping

4. This spectrum is transformed into a time-domain DMT frame using the inverse Discrete Fourier Transform (DFT) (Figure 8).

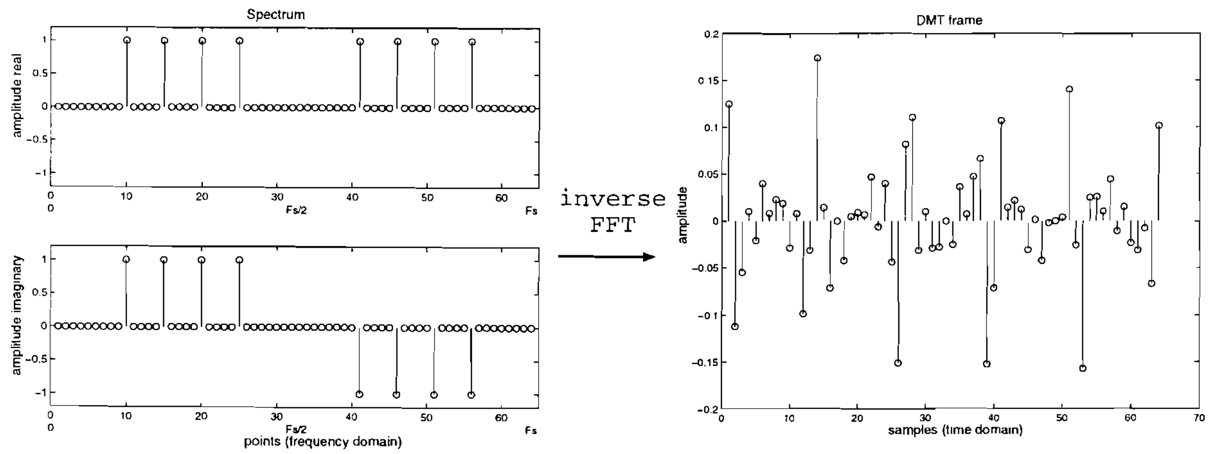
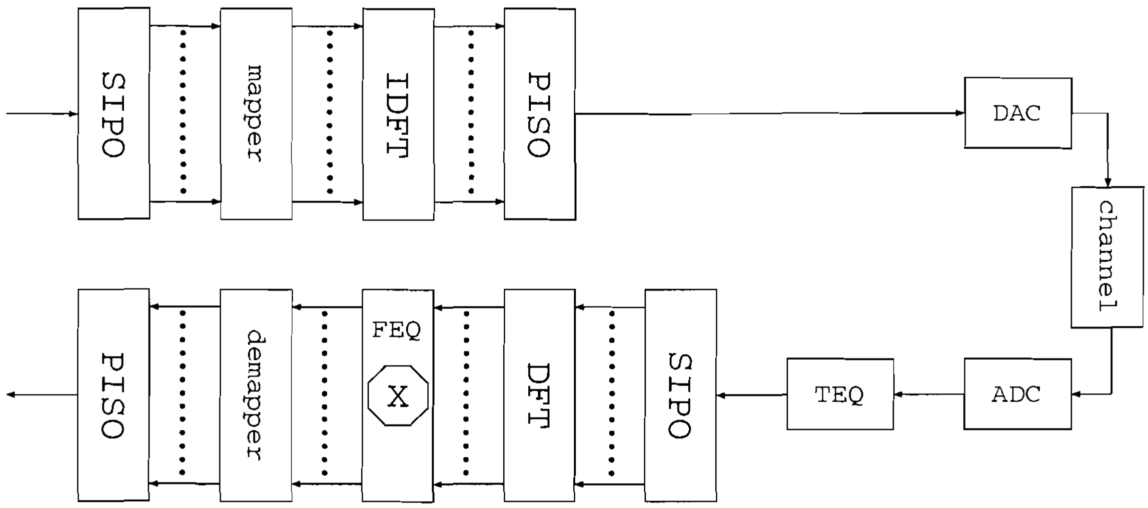


Figure 8: DMT frame in frequency and time domain

These transformations generate a valid DMT frame, ready to be transmitted. The receiver makes the inverse transformations to retrieve the send information from the time-domain frames.

The block diagram in Figure 9 contains all the blocks necessary to do the different steps. It consists of an encoder and a decoder path, together with a channel. The Serial In to Parallel Out convertor

(SIPO) divides the incoming bitstream over the different carriers. The mapper converts the different bitstreams into a frequency-domain spectrum, which is transformed into a time-domain DMT frame by the inverse DFT. The signal is then serialized by a PISO. The DMT frame is send through a Digital to Analog Convertor (DAC), channel, Analog to Digital Convertor (ADC) and Time domain Equalizer (TEQ) to the SIPO of the receiver.

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SIPO = Serial In Parallel Out
PISO = Parallel In Serial Out
IDFT = Inverse Discrete Fourier Transform
DFT  = Discrete Fourier Transform
FEQ  = Frequency domain Equalizer
TEQ  = Time domain Equalizer
ADC  = Analog to Digital Converter
DAC  = Digital to Analog Converter
X    = Complex multiplication
```

Figure 9: DMT system block diagram

Both the TEQ and FEQ are mentioned to give a complete picture of a DMT type transceiver system, they will be explained in section 3.4.

3.2 DFT properties

The operation of a DMT modem depends on the properties of the Discrete Fourier Transform (DFT). The forward and backward DFT formulas are given:

Forward:

$$X_n = \frac{1}{N} \sum_{k=0}^{N-1} x(k\Delta t) e^{-j2\pi nk/N}$$

$$(n = 0, 1, ..., N-1)$$

Backward:

$$x_{(n\Delta t)} = \sum_{k=0}^{N-1} X_k e^{j2\pi nk/N}$$

$$(n = 0, 1, ..., N-1)$$

To use the DFT to transform the created spectra into valid time domain DMT frames and vice versa several conditions have to be met:

Spectrum symmetry The created spectra must have symmetrical real parts and point-symmetrical imaginary parts to get a real time domain signal from the DFT.

Interval length The interval transformed by the DFT has to be the same length as the interval transformed by the inverse DFT otherwise the spectra will not be the same and leakage will occur.

Periodicity The DFT works only on periodical signals. Signals created by the inverse DFT are one period out of an endless periodical signal. Therefore the signal transformed by the DFT must have the same property, it has to be possible to extend the signal endless without anomalies at the transitions.

3.2.1 Spectrum symmetry

The inverse and forward DFT algorithms are used for transforming the information to be modulated from a spectrum to a DMT frame. A spectrum with the characteristics of a DMT frame is composed and this is inverse-transformed into a DMT frame (Figure 10). The spectrum is composed by putting both imaginary and real data corresponding to the to be transmitted QAM symbols at the places corresponding to the sub-carriers being used,

The symmetry in the spectrum determines the type of signal that is calculated by the inverse DFT. To get a real, time-domain signal, the real part of the spectrum has to be symmetrical around $\frac{F_s}{2}$, and the imaginary part of the spectrum has to be point symmetrical around $\frac{F_s}{2}$.

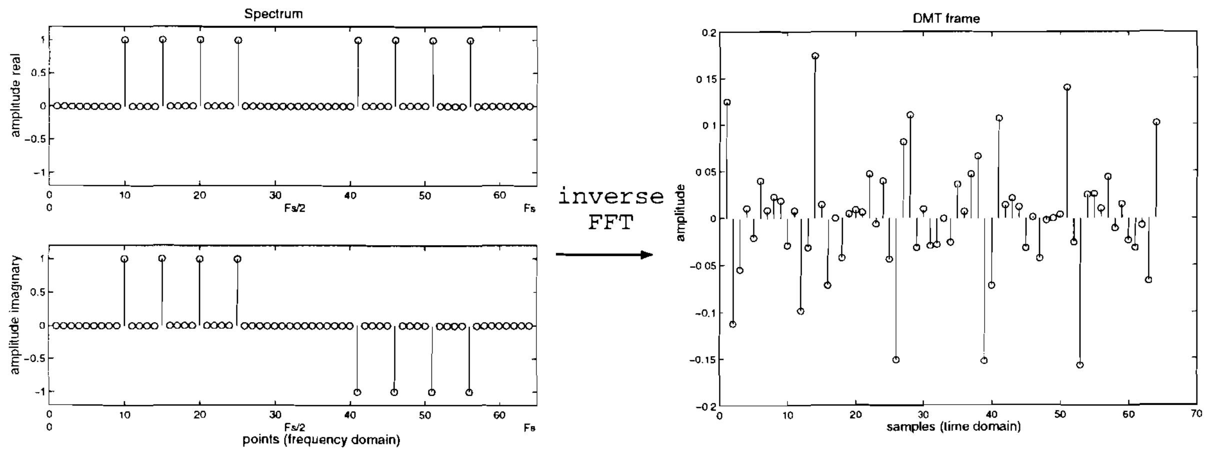

Figure 10: DMT frame, symmetry in the frequency domain produces real time domain signals

3.2.2 Interval length

The same interval as encoded by the transmitter has to be decoded by the receiver, or the spectra will not be the same. The number of points at the receiver has to be the same as the number of points at the transmitter otherwise leakage will occur and result in incorrect detection of the symbols [18]. Figures 11 a thru d show this:

a A spectrum is created with correct symmetry to generate a real time domain signal. Length is 64 points.

b The inverse DFT transforms the complex data of the spectrum into a real time domain signal, with a length of 64 samples.

c The time domain signal is truncated to 60 samples. This signal will be transformed back to a spectrum by the DFT.

d The spectrum gained from the DFT of part (c), leakage occurs because the interval which is transformed is not of the same length as the signal created.

Leakage occurs in this case because the interval decoded can not be repeated to get an endless, continuous signal [17, 18].

The same happens with DMT frames. In Figure 12 the time domain signal is the same as Figure 10, but the DFT has a length of 60 samples and 64 as in the original. Leakage occurs because the DFT borders of the interval do not connect properly if the signal is repeated.

3.2.3 Periodicity

Leakage does not only occur when periodicity of a signal is lost due to different DFT lengths, but also when the signal is send through a channel with an impulse response which is longer than T_s due to group delay[1] distortions.

[1]With group delay distortion, some frequencies propagate faster than other frequencies, therefore ISI will occur if adjacent symbols consist of different frequency and phase combinations.

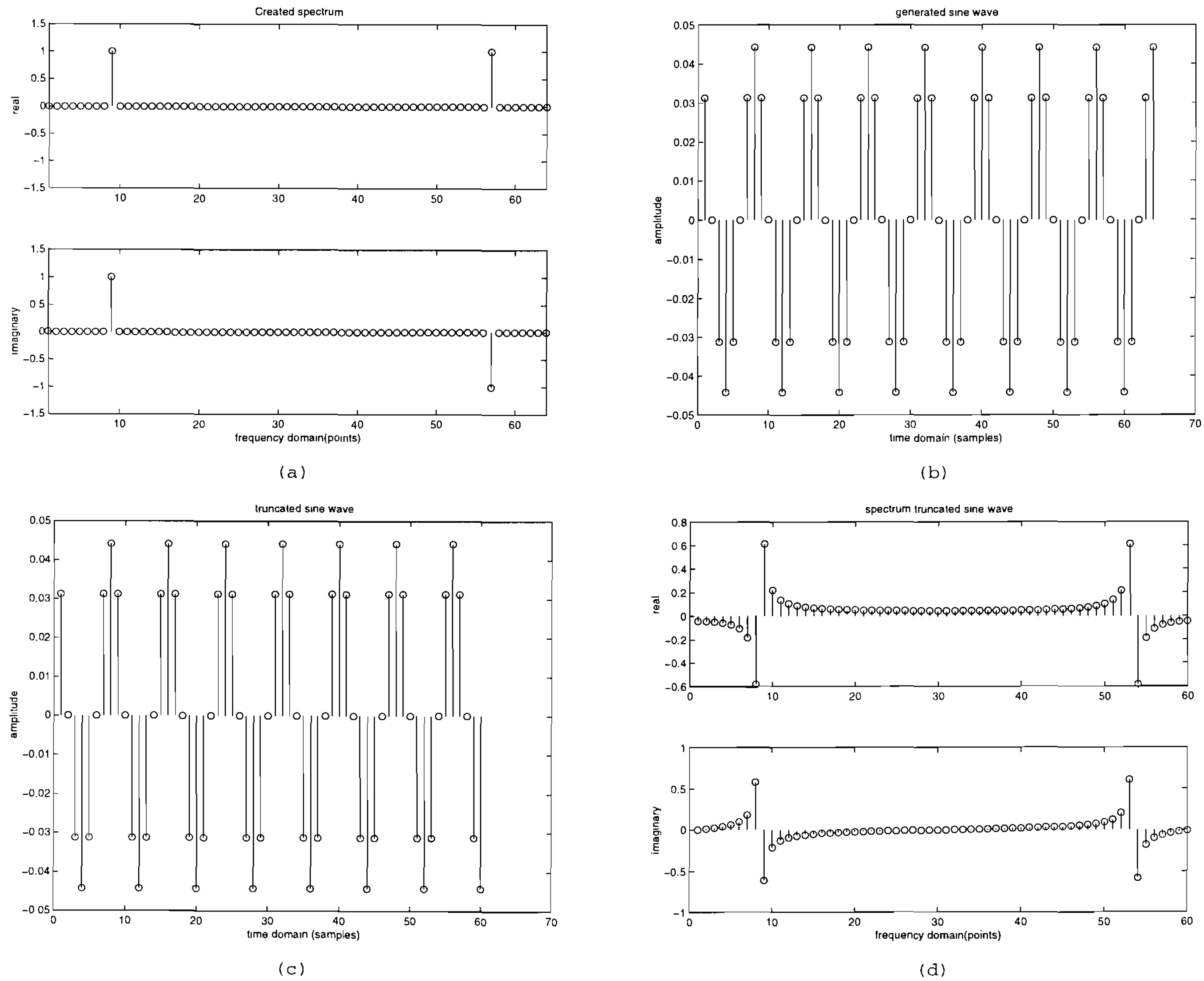

Figure 11: Leakage (d) due to DFT length differences

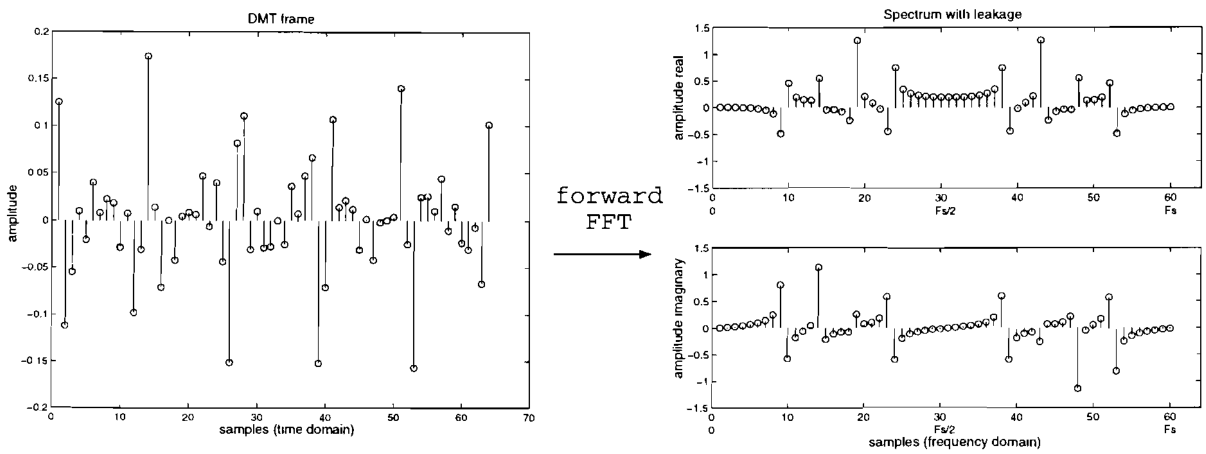

Figure 12: DMT frame and spectrum with leakage because of wrong number of points

The channel distortions work like a filter on the signals transmitted. Figure 13 shows an example of this. The dotted sine wave is the signal transmitted, the continues sine wave is the signal received. The first half period is heavily distorted, the channel works like a filter.

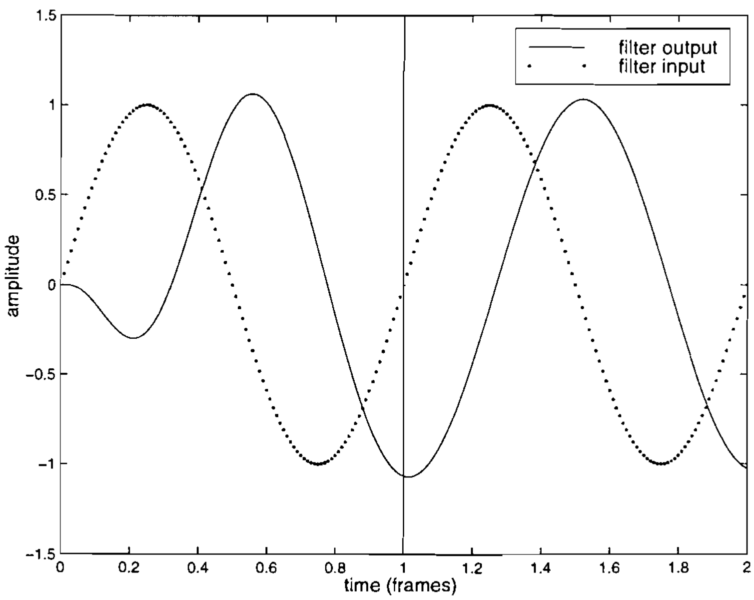

Figure 13: Frame distortion due to group delay

An other effect off the channel is shown in Figure 13. This is of minor importance to the DFT algorithm, the phase of the signal has changed. Figure 14 shows the effect of this distortion on the DFT.

Again the leakage occurs because the continues signal in part (a) can not be continued periodically without having discontinuities in the time domain (Figure 15).

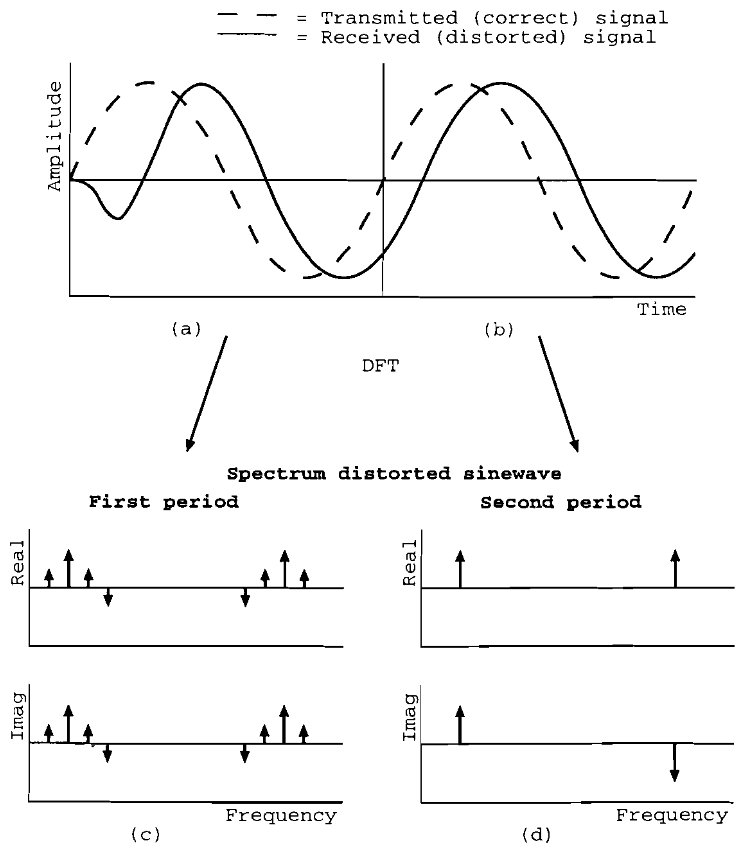

Figure 14: Leakage (c) due to group delay.

To avoid this problem the guard interval is introduced. This means the transmitted symbol is repeated $\frac{N_i+N_{DFT}}{N_{DFT}}$ times to compensate for the total length of the impulse response, N_i, and to assure the signal in a frame behaves like a periodic signal. The symbols no longer interfere and the spectra can be decoded without ISI and ICI. In Figure 16 the marked blocks in the time domain are the blocks that will be processed by the DFT. Figure 16a shows there will be a substantial amount of information from the previous frame in the current frame. In Figure 16b this is not the case, the frames have no interference from the previous frame.

A disadvantage is the reduced efficiency, because no information is transmitted during the guard interval. A DMT data frame has the length of the used DFT plus the guard interval length. This makes the efficiency equal to this formula.

$$\frac{N_{DFT}}{N_{guard}+N_{DFT}} \leq 1$$

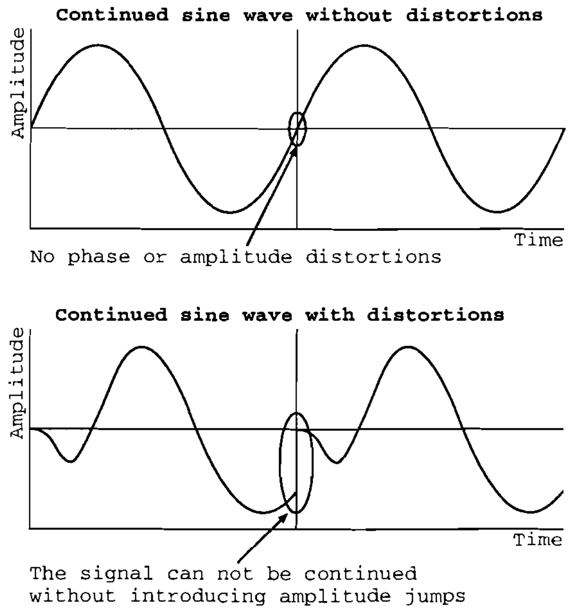

Figure 15: A distorted sine wave can not be continued without phase or amplitude jumps

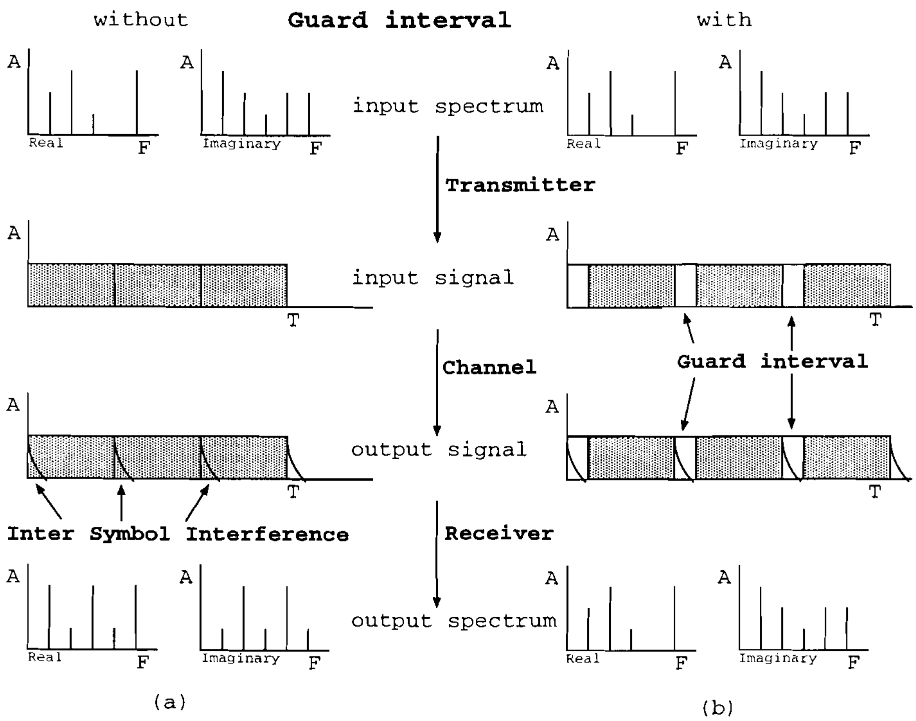

Figure 16: Guard interval to avoid channel distortions

3.2.4 Impulse noise

The influences of impulse noise are difficult to avoid because the DFT will transform the impulse into noise in every sub-carrier. Noise which will distort all information in one frame. Impulse noise will therefore generate large bursts of errors during transmission.

3.3 Synchronization

Synchronization is done at 3 levels, i.e. the sample level, frame level and super-frame level.

3.3.1 Sample level

The sample level is necessary to avoid losing information in the sub-carriers near the Nyquist frequency ($F_N = \frac{F_s}{2}$). The information in the sub-carriers near half the sample frequency is transformed by the inverse DFT into signals with ≈ 2 samples per period. Figure 17 shows the result of this, because there are only 2 samples per period the possibility exists that these two samples are situated at the zero-crossings of the sine wave, therefore the information can not be retrieved.

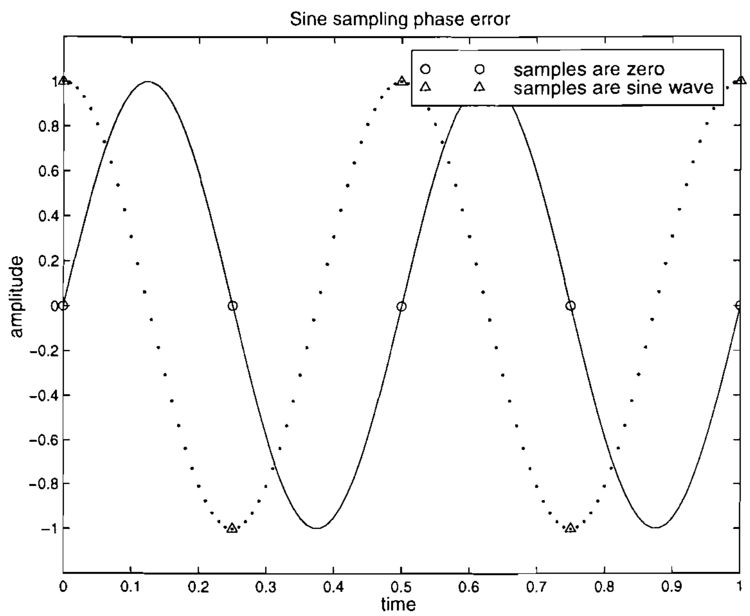

Figure 17: Erased data due to sample timing error

What does this mean for the DMT receiver? The next 4 situations illustrate the problem in relation to the sub-carriers with close to 2 samples per period and the timing error in sample time ($T_s = \frac{1}{F_s}$).

No oversampling, $T_e = 0$ First of all the received signal has no timing error. It is equal to the transmitted signal. The constellation of the received carriers is displayed in Figure 18. Every X corresponds to a combination of 2 bits (b_I, B_Q). A vector v_i with length (A) and angle (PHI) corresponds to the sine wave transmitted on carrier c_i.

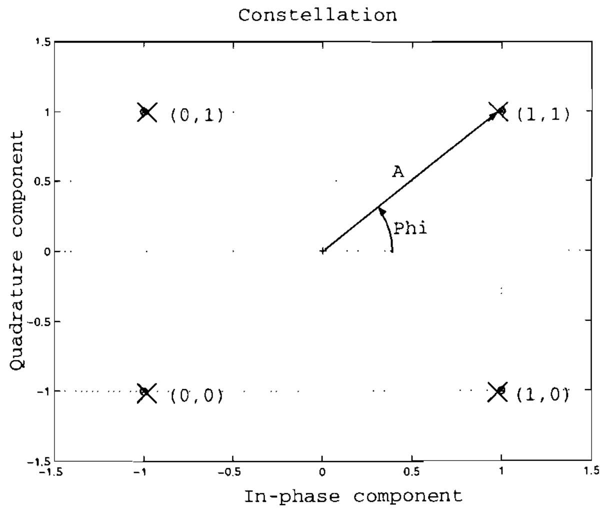

Figure 18: Constellation of 255 carriers (1 DMT frame), $T_e = 0$

No oversampling, $T_e = 0.3 * T_s$ In this situation the timing error $T_e = 0.3 * T_s$, this is the same as a phase shift of 60 degrees in the carrier c_{n-1} near F_N. The DFT decomposes the received signal into the most basic set of sine waves which corresponds to the points in the DMT frame. Figure 19 shows the phase and amplitude information of 255 decoded sub-carriers.

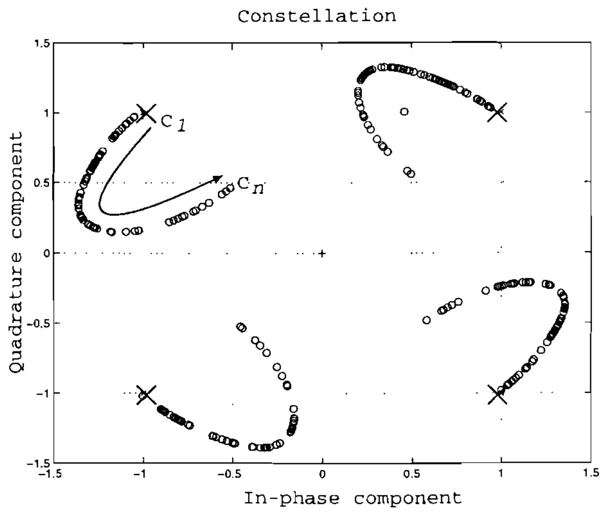

Figure 19: Frame distortion due to group delay, $T_e = 0.3 * T_s$

The sub-carriers close to C_0 are the lowest frequencies. No information is lost in these carriers because T_e causes a phase shift of $\frac{360}{512} * 0.3 = 0.21\,\text{deg}$ in carrier c_1.

The sub-carriers close to C_n are the highest frequencies. Information loss occurs in these carriers because T_e causes a phase shift of $\frac{360}{2} * 0.3 = 54\,\text{deg}$ in carrier c_{n-1}.

In this case all sub-carriers can be decoded because the symbols can be uniformly decoded ($A \gg 0$), even if noise would be present.

This particular pattern is the result of the DFT interpretation of the samples from the DAC of the receiver (Figure 20).

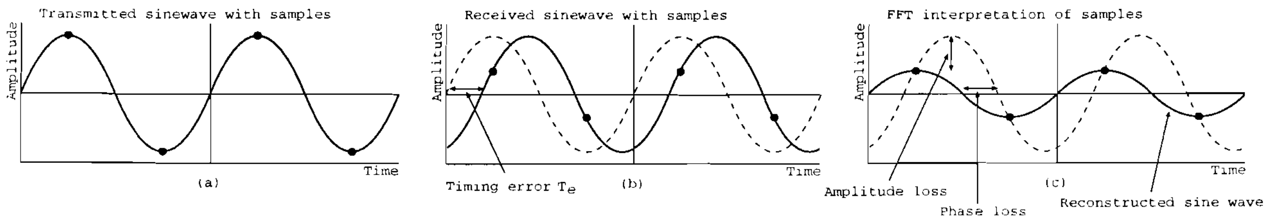

Figure 20: DFT sample mis-interpretation causes loss of amplitude

The DFT reconstructs the most simple sine wave which fits the samples received. Because of the limited number of samples per period (2 in this case) both amplitude and phase information is lost in part (c). This effect mainly occurs with signals with a limited number of samples per period e.g. the sub-carriers close to F_N.

No oversampling, $T_e = 0.5 * T_s$ In this situation, the timing error is critical, $T_e = 0.5 * T_s$. The sub-carriers near F_N can not be decoded if noise is present because the amplitude $A \approx 0$, Figure 21.

The sub-carriers close to C_0 are the lowest frequencies. No information is lost in these carriers because T_e causes a phase shift of $\frac{360}{512} * 0.5 = 0.35\,\text{deg}$ in carrier c_1.

The sub-carriers close to C_n are the highest frequencies. Information loss occurs in these carriers because T_e causes a phase shift of $\frac{360}{2} * 0.5 = 90\,\text{deg}$ in carrier c_{n-1}.

In this case the sub-carriers close to C_n can not be decoded uniformly because $A \approx 0$. If noise would be present carriers can easily shift to another quadrant.

2x oversampling, $T_e = 0.5 * T_s$ This situation shows a solution for the timing error problem. Figure 22 shows a two times oversampling ($F_{oversampling} = 2 * F_s$) scheme. The number of carriers and F_N are twice as high. The highest frequency used is half of the new Nyquist frequency. It will have 4 samples per period and not have the amplitude loss problem.

3.3.2 Frame level

The frame level synchronization determines the start of each DMT frame (with guard interval). This can be done in a number of different ways, both decision directed and not decision directed. The decision directed method uses information from the DFT while the non-decision directed method

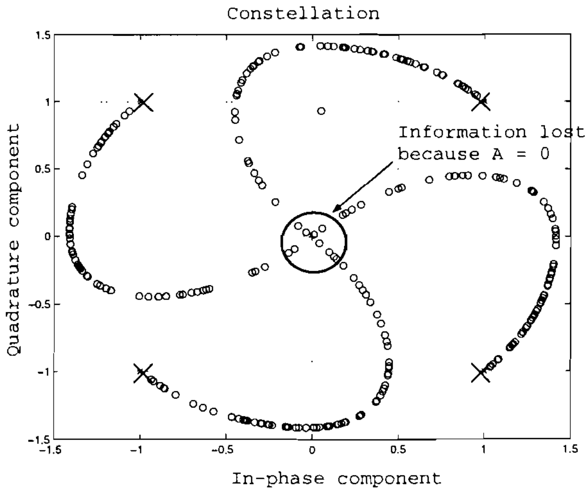

Figure 21: Frame distortion due to group delay, $T_e = 0.5 * T_s$

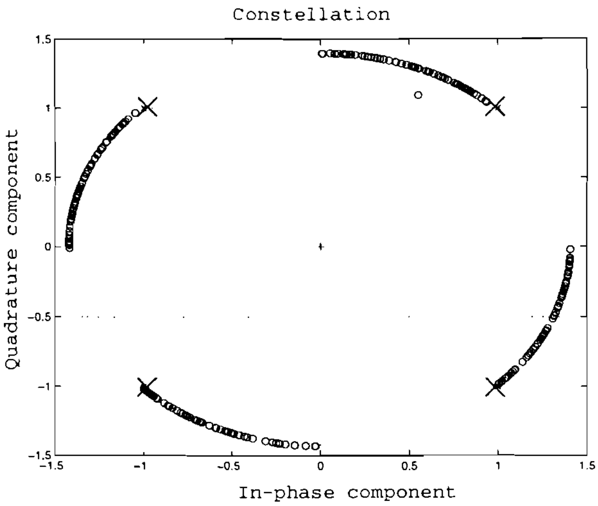

Figure 22: Frame distortion due to group delay, $T_e = 0.5 * T_s$ with oversampling

relies on information extracted from the input data stream. The pilot tone (section 3.3.4) can be used for both types of synchronization.

Figure 23 shows two architectures. The start of a DMT frame can be found with frequency domain information retrieved from the DFT, or from information retrieved from the data stream itself. The second algorithm has the advantage that a pipelined architecture can be used, because no information retrieved from the DFT has to be used in front of the DFT.

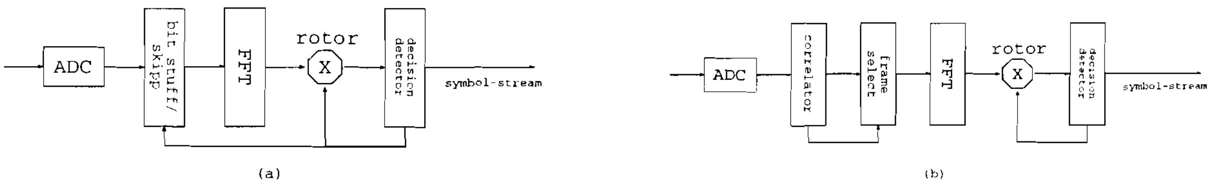

Figure 23: DMT receiver with (a) or without (b) feedback across the DFT

3.3.3 Super-frame level

The super-frame synchronization uses special frames to synchronize on, Figure 24. These are known frames with a pseudo-random data sequence as information [14]. The super-frame synchronization is used for retraining after a major distortion, or to make adjustments to the equalizers.

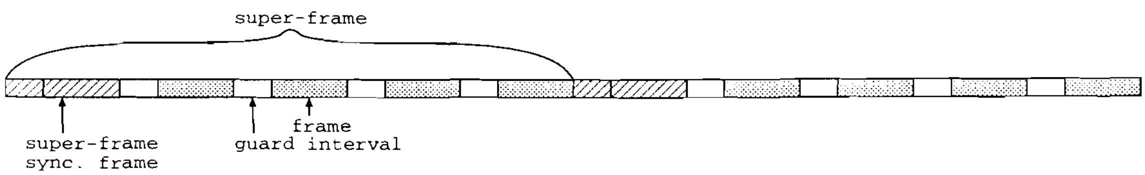

Figure 24: Super-frames

3.3.4 Pilot carrier

For synchronization purposes a pilot tone can be added like in an ADSL system [14]. The pilot tone is one tone with a constant modulation. This tone generates a constant sine wave throughout the signal. This is accomplished by choosing the carrier with the same period as the length of the guard interval, $N_{guard} = N_{pilot}$, see Figure 25.

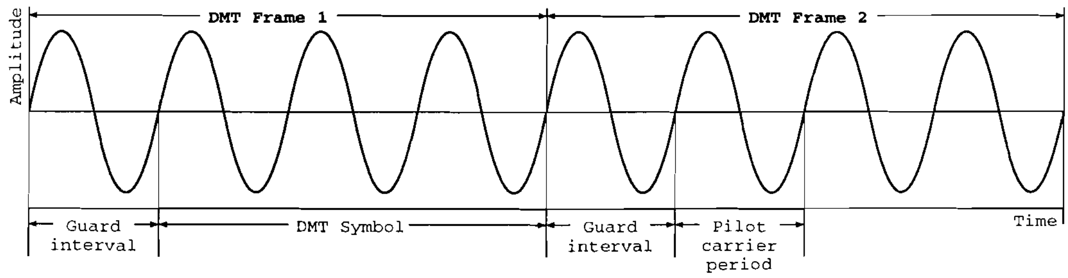

Figure 25: Pilot tone length is same as guard length

The result of this is a continues sine wave throughout the DMT signal. It can be used for synchronization purposes on both sample level and frame level. A PLL can be locked on this frequency and than the sample clock can be synchronized on this clock [23, 24, 25, 26].

3.4 Equalization

Several channel distortions have to be corrected in a DMT receiver to be able to detect the transmitted symbols:

1. Amplitude and phase distortions per sub-carrier.
2. Impulse response of the channel [4].
3. Sample timing error compensation.
4. Downsampling.

Equalization of these three types of distortions can be done both in the time domain and in the frequency domain. Although it is possible to compensate for the amplitude and phase distortions in a time domain equalizer the logical choice is a Frequency domain equalizer (FEQ). The opposite holds for the sample timing error compensation and downsampling. These can be equalized in the frequency domain, but the logical choice is to make a Time domain equalizer (TEQ) to compensate these distortions [4].

A time domain equalizer is not strictly necessary, because both the sample timing error and the impulse response problem can be solved by choosing the parameters of the modem different, although the performance will drop if the TEQ is left out. Sample timing errors can be solved by oversampling the FFT. Impulse response shortening can be avoided by making $N_{guard} \geq N_{impulse\ response}$.

A typical FEQ consists of a complex multiplication per carrier, which will be trained during the super-frame synchronization frames. This scheme is used in the ADSL standard. Phase and amplitude errors are estimated per sub-carrier to assure that the symbols will be decoded correct. During data transfer the estimated complex values $k_1 ... k_i$ are used for compensating the phase and amplitude error for every sub-carrier, see Figure 26.

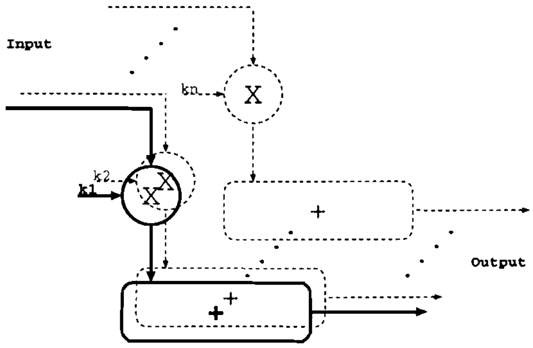

Figure 26: Frequency domain equalizer for n sub-carriers.

3.5 Rate Adaptive DSL

As stated before, the bandwidth of a DMT modem can be adjusted by changing the number of bits b_i assigned to a sub-carrier. Initially an estimation is made for the channel, but there is also the possibility to change b_i during operation. This requires a synchronization protocol which can estimate N_b and send the estimated $b_0 ... b_i$ back to the transmitter during operation. By estimating the quality of a channel by measuring the phase and amplitude error per sub-carrier during operation and sending $b_0 ... b_i$ back during the super-frame synchronization frames.

3.6 Radio Frequency Interference

There are two kinds of Radio Frequency Interference (RFI), the continuous broadcast systems in the frequency band also used by the DMT modem, and the Radio amateur type which can be both in and outside the band, but which is switched on and of during transmission of the modem.

1. The interference of the broadcast systems is not a problem. It will influence a number of bands. This can reduce overall performance because these bands can not be used.
2. The interference of Radio amateurs has two aspects.
   (a) The first is the continuous in-band interference which has the same effects as the broadcast systems.
   (b) The second is the fact that Radio amateur interference can switch on and off during a transmission is a problem. To detect this kind of interference the transceiver has to use a long training sequence to be sure it has got all different kinds of interference which might occur during operation. This is not possible, therefore it has to keep track of the distortions which occured during the last period of operation and use this information to adapt itself for the coming period of operation.

By doing this influences or Radio amateur RFI are avoided.

4 Simulation model

4.1 Overview

Matlab[2] is used for modeling and simulating the VDSL modem and channel. The model is a basic interpretation of the DMT modulation explained before. It includes all features described in the DMT section except the TEQ[3].

For the DFT implementation the matlab built-in Fast Fourier Transform (FFT) function is used.

Figure 27 shows the main parts of the Matlab simulation model block diagram. For most blocks there is both Matlab code and embedded C code. Embedded C code is faster than plain Matlab code. The C version was made to get a speedup in simulation time [30].

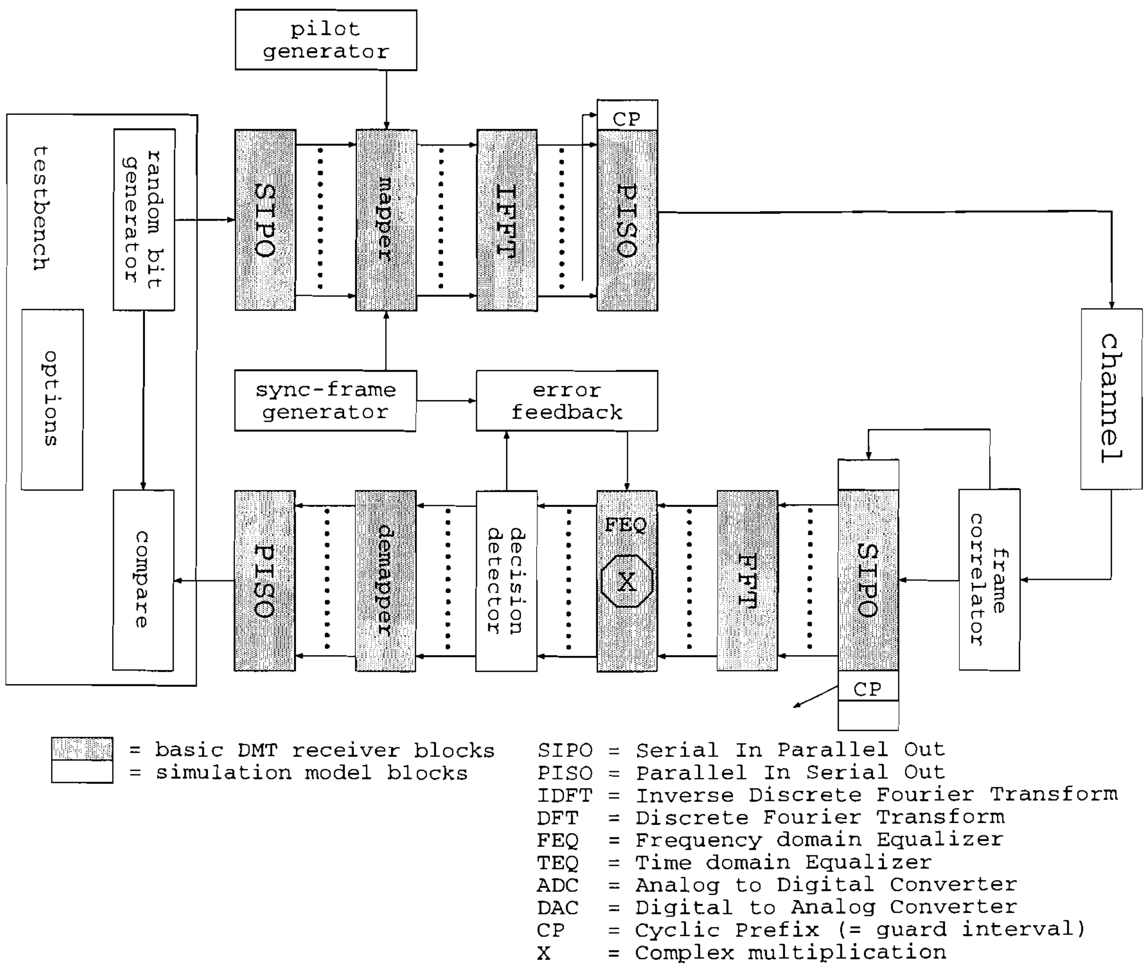

Figure 27: Simulation model block diagram

The grey blocks are the same as the DMT model shown in Figure 9. The white parts are mainly for synchronization and simulation purposes:

[2]The Matlab version used is 5.1 for HP-UX.

[3]because of a slowly varying channel

Testbench The testbench has 3 main functions:

- Stimuli generation for the transmitter (random bit generator)
- Checking for errors (compare between generated and received bit-streams and count erroneous bits)
- Control the functioning of the modem (all parameters for the modem are set in the options variable)

Pilot generator The pilot generator generates the sub-carrier which always has the same information, for synchronization purposes.

Sync-frame generator The sync-frame generator generates the super-frame synchronization frames.

Error feedback The error-feedback adjusts the FEQ parameters according to the information extracted from the super-frame synchronization phase. It compares the send and received data and calculates both phase and amplitude differences to compensate.

Frame correlator The frame correlator searches for the start of data frames by correlating between a super-frame synchronization frame and the data-stream. Whenever a threshold is reached a frame is detected. This receiver is a type without feedback across the FFT.

4.2 Assumptions

The model was developed incrementally, which means that for every problem encountered there are a few blocks added. During this process some assumptions were made to be able to take small steps at a time. Mainly because it was not possible to create solutions which can handle every possible situation. Eventually the impact of these choices has to be investigated. The choices are listed below, with a short explanation why these choices where made.

Slowly varying channel A slowly varying channel means that during the transmission the characteristics of the channel do not chance rapidly, the equalizers do not need to be adjusted. The parameters of the frequency domain equalizer are adjusted only during the super-frame synchronization phase. This assumption was made because an equalizer which adjust only during synchronization is easier to implement, a one-shot adaptive algorithm is used.

Guard interval can be as long as channel impulse response In order to be able to skip the TEQ the guard interval has to be stretched to the same length as the channel impulse response (there is no TEQ to shorten the impulse response). Only phase, amplitude and noise are the distortions to compensate when the guard interval is long enough [23].

D/A and A/D have unlimited dynamic range (no fixed-point) The model is completely floating point, quantization noise is not taken into account. This noise has to be investigated when the modem is converted to fixed-point.

Radio Frequency Interference (RFI) is not taken into account RFI was left out of consideration because of two reasons.

1. If RFI was taken into consideration it was not possible to use a slowly varying channel.
2. RFI can be very bursty, for instance when there is a radio amateur nearby.

Out of band interference is not considered (no spectral shaping) No action was taken to prevent or suppress out of band interference. Spectral shaping can be done by a filter after the IFFT or a smart way of modulating unused carriers before the IFFT. The filter would produce extra noise and distortions, the modulation of unused carriers would require an algorithm to determine the symbols to be modulated. Such an algorithm was not available.

Per carrier, the amplitude is constant and the phase is linear The frequency band taken by one carrier is assumed to be flat. This means the phase and amplitude distortion are assumed to be constant in a carrier. In reality this is not the case, but it can be assumed if a carrier is small enough.

4.3 Features

During the development a lot of parameters where added to the model. In this section the most important parameters are listed. To get a full list see the testbench.m file. The parameters listed below are a good view on the features of the model, also with an explanation.

FFT length (N) The length of the FFT determines the number of usable carriers. If the length of the FFT is for instance 512, the maximum number of carriers is 256. This is a theoretical value, in practise the limit is about 254, because it's difficult to use the DC and Nyquist frequency. The length of the FFT can be set to every desired value, although mainly the 2^n values are used, because these can be calculated very efficiently.

Carriers used (n) The number of carriers ($c_1...c_n$) can be set to any number as long as its less than half the FFT. The position of the carriers can also be set by hand to every number required.

Bits per carrier (b_i) The number of bits per sub-carrier can be set by hand, or can be estimated with the channel characteristics estimation feature. Every carrier can have its own constellation, max. $b_i = 11$.

Pilot carrier The pilot carrier is a feature borrowed from ADSL to compensate sample timing errors. By modulating a predefined symbol on one sub-carrier (c_p) it is possible to estimate and compensate for sample timing errors. The pilot carrier can be enabled and disabled and can be assigned to every available carrier. In this way it is possible to try different combinations to see which combination gets the best results.

Guard interval The length of the guard interval (or cyclic prefix) can be set to every desired value, to check the influence of this on the performance of the modem..

Synchronization frames The number of synchronization frames can be set, because more sync. frames gives the opportunity to get a better characterization of the channel, but this will reduce performance considerably.

Frequency domain equalizer The frequency domain equalizer can be enabled and disabled to be able to investigate the performance gained by this component, and to show the distortions which are compensated.

Oversampling Oversampling is incorporated to see if an oversampling receiver will benefit in the quality of reception. The oversampling factor can be set separately for the transmitter and receiver, as long as the channel can compensate for the differences. The channel also has to be configured for oversampling.

Channel characteristics estimation To adjust to every possible channel, a channel characteristics estimation feature is developed. This feature is able to estimate b_i to send per sub-carrier across the current channel and to estimate the phase and amplitude distortion per carrier. It uses a known training sequence of an adjustable length. The accuracy of the estimation depends on the length of the training sequence.

4.4 Performance

To be able to make a recommendation whether investigating DMT and in particular this model is useful, a performance measurement was done according to the Monte Carlo analysis. This part describes the Monte Carlo analysis of the modem, performed on a typical VDSL type channel.

4.4.1 Channel model

The channel model used for the Bit Error Rate (BER) simulations has amplitude and phase distortions as shown in Figures 28 and 29. It is standard testloop nr. 6 from the VDSL draft technical report [15].

This channel represents a 1000 meter unshielded twisted pair copper line with one bridged tap causing a notch and a 90^o phase jump. It is modeled with a sampling rate of 19.440 MHz. The channel amplitude and phase responses are converted into a IIR filter with 7 forward and 7 backward taps. To simulate the BER rates in the next paragraph the IIR filter was used.

The power spectral density (PSD) of a data stream generated by the modem is shown in Figure 30. The data streams contains 6 DMT frames with 210 carriers (from $c_{20} = 759380$ until $c_{230} = 8732870 Hz$). The PSD of the same stream after it has passed the channel is shown in Figure 31. The PSD figures are generated by the PSD function of matlab, using a 512 points FFT and no windowing.

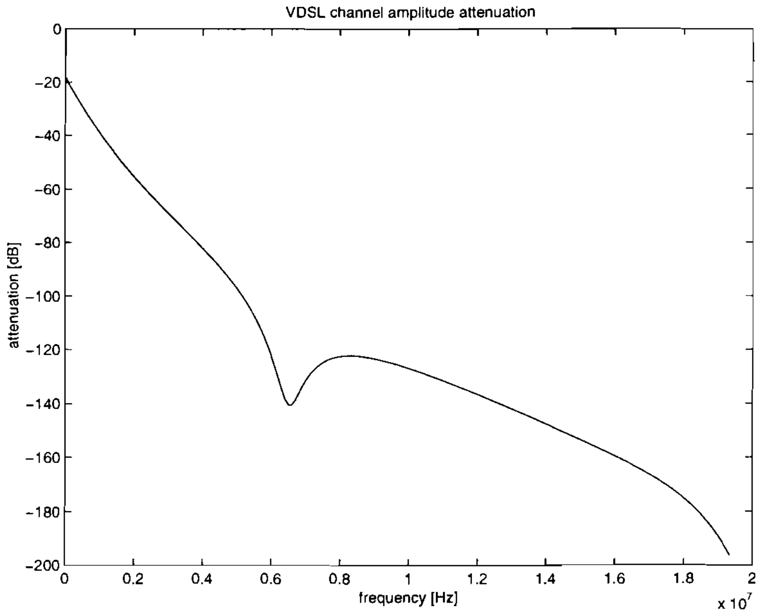

Figure 28: Channel amplitude characteristic

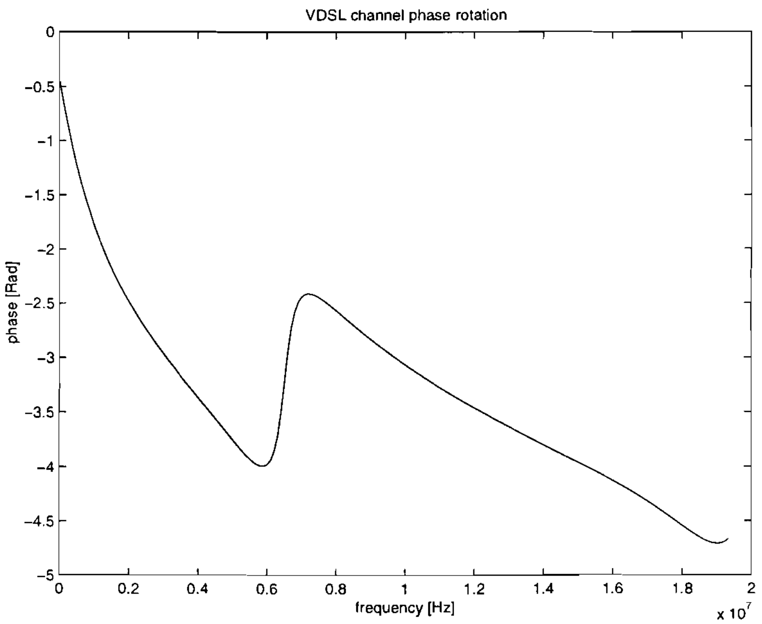

Figure 29: Channel phase characteristic

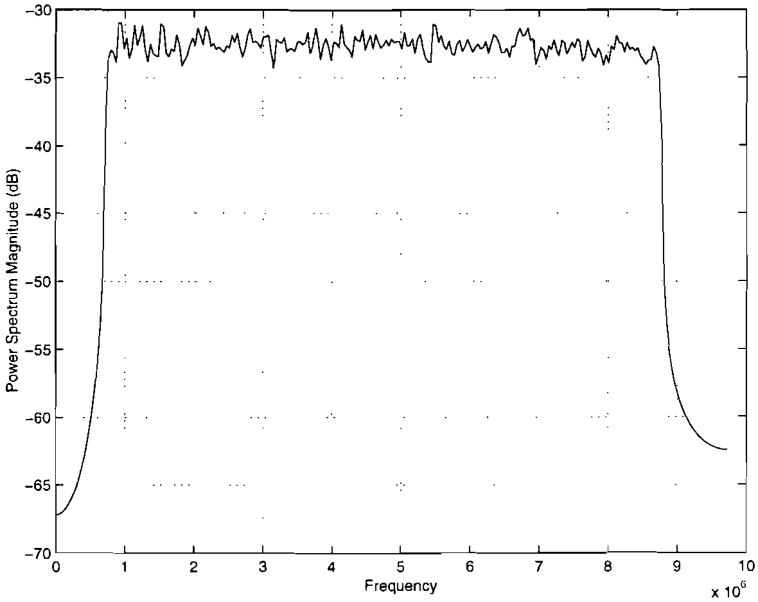

Figure 30: Power spectral density, F_s = 19.440 MHz, 210 carriers.

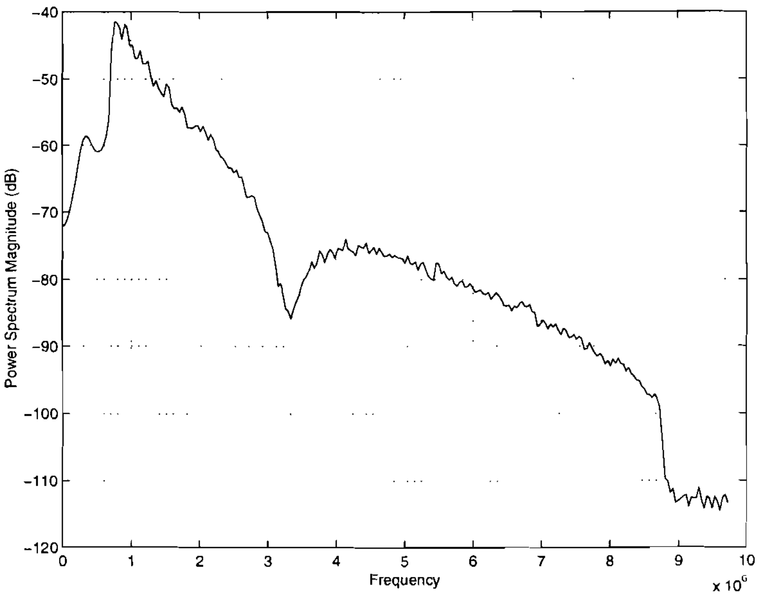

Figure 31: Power spectral density after channel, F_s = 19.440 MHz

4.4.2 Bit error rates

The bit error rate (BER) simulation where done with the embedded C (mex) Matlab model and the channel model described before. The Monte Carlo simulation consists of sending a random bit-stream through the transmitter, channel and receiver until 10 errors have occured. Probability calculations show that the BER is estimated within a factor of two by using this method [29]. The VDSL draft technical draft requires noise margin (E_b/N_0) estimations to be performed at an bit error probability of 10^{-7} [15].

Figures 32 and 33 show the BER simulation results. The results where obtained with the parameters set to the values in Table 2.

Parameter	Value	Units	Comment
N_{DFT}	512	points	FFT length
C_n	210		$C_i, 20 \leq i \leq 230$
F_s	19.440	MHz	Sample frequency
c_{bwidth}	37.969	kHz	Carrier bandwidth
c_{pilot}	c_{100}		Pilot carrier is carrier nr. 100
N_{guard}	700	T_s	Guard length
$N_{sync\ frames}$	3		Nr. of synchronization frames per super-frame
FEQ	enabled		
$N_{channel\ delay}$	0	T_s	
$N_{training\ sequence}$	100	frames	Length training sequence

Table 2: Simulation model parameters during performance analysis

Noise generation is done by adding a random signal to the data with a normal, Gaussian distribution with mean 0 and variance σ^2. σ is calculated according to the following formula [29].

$$\sigma = \sqrt{\frac{M_s}{2 * b * N * E_b/N_0} * P_T}$$

with:

- P_T = signal power
- M_s = symbol length = $N_{DFT} + N_{guard} = 1212$
- b = nr. of bits per symbol
- E_b/N_0 = noise margin
- N = total length data stream

During the simulation, the E_b/N_0 value is varied between 5 and 35 dB, in steps of 5 dB.

The model itself determined $b_1...b_n$. By adjusting the signal to noise ratio during the training sequence a distribution was estimated which fitted best in the channel response.

The figures show that a 22.2 Mbit/s implementation according to this simulation and channel model requires an E_b/N_0 value of > 28 dB if a BER of 10^{-7} is required (Figure 32). The performance parameters and results are listed in Table 3. The guard interval is 700 samples long in these simulations.

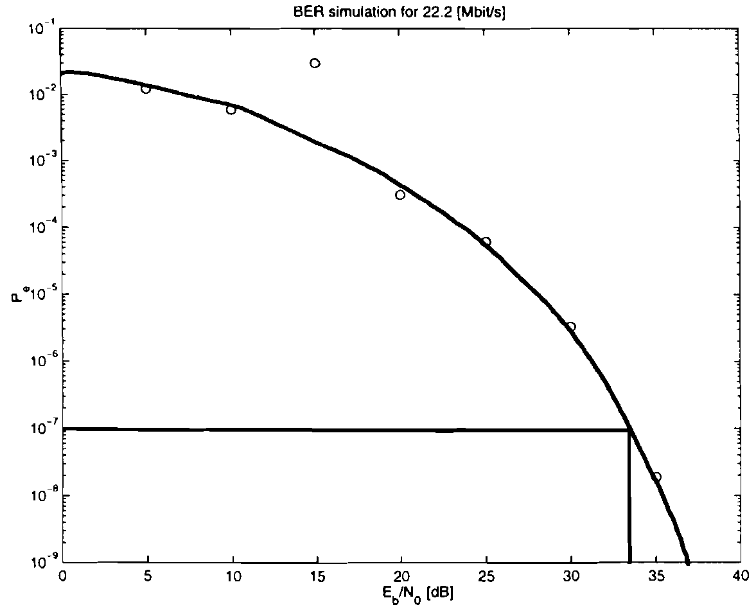

Figure 32: Bit Error Rate simulation for 22.2 Mb/s using a VDSL channel (testloop 6 from ANSI VDSL standard)

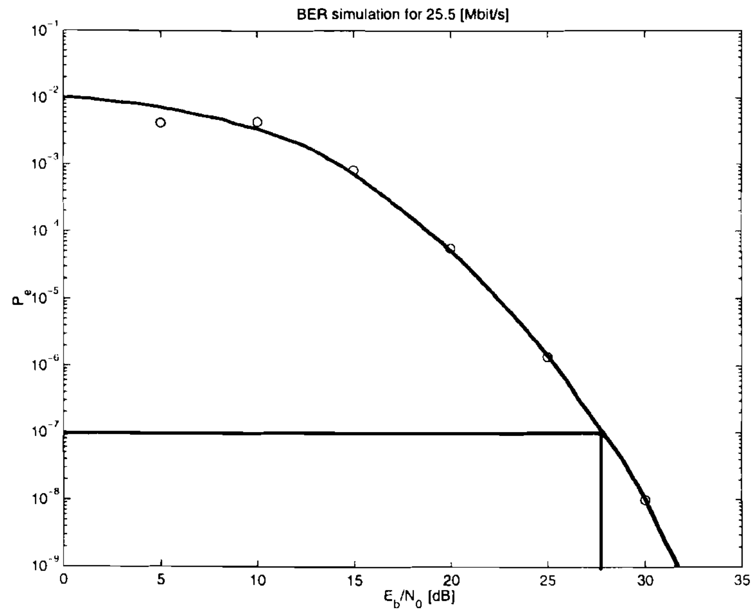

Figure 33: Bit Error Rate simulation for 25.5 Mb/s using a VDSL channel (testloop 6 from ANSI VDSL standard)

This value is obtained from analyzing the channel without adding noise. With a guard interval of 700 samples no ISI occurs larger than machine accuracy.

If the impulse response can be shortened to 50 T_s, 44 MBit/s is possible.

At a BER of 10^{-7} the 25.5 Mbit/s simulation requires a noise margin of $\approx$ 34 dB (Figure 33). The performance parameters and results are listed in Table 3. This case also uses the guard interval of 700 samples, if it can be shortened to 50 T_s, 52 Mbit/s will be possible.

Bit-rate [Mbits/s]	bit/frame	frames/s	N_{guard}	Noise margin [dB]	Spectral efficiency [bits/HzBw]
22.2	1390	16040	700	28	2.80
25.5	1590	16040	700	34	3.20

Table 3: Performance analysis parameters and results

The length of the guard interval can be shortened because noise has a greater impact in these simulations. To estimate an acceptable length at a certain noise margin the effect of different guard intervals at different noise levels is simulated. The effect of the length of the guard interval is shown in Figure 34. The guard interval length is varied between 100 T_s and 400 T_s. The effect of increasing the guard interval from 300 to 400 T_s is negligible.

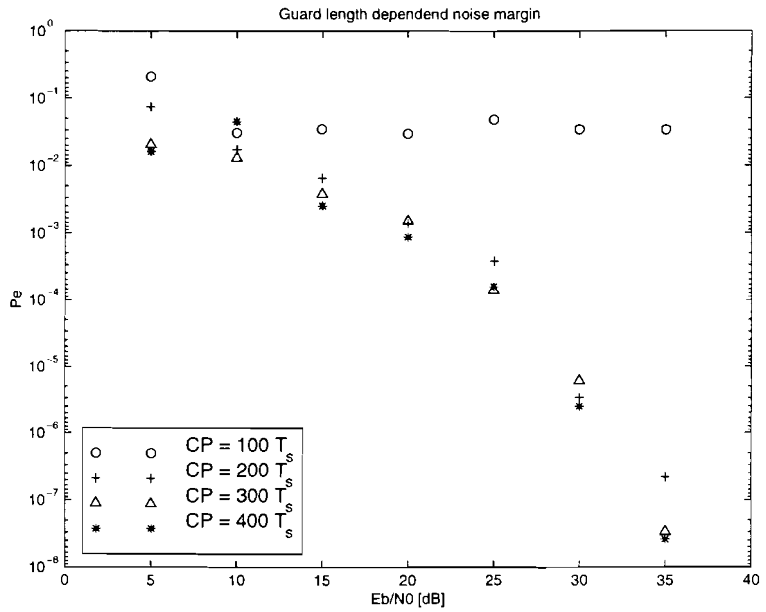

Figure 34: BER for guard intervals between 100 and 400 samples.

5 DSP implementation

5.1 Overview

The computational intensive parts of the DMT modem (like FFT and frequency domain equalizer) are taken over by the Digital Signal Processing (DSP) module. The input and output is taken care of by the PC host computer. Loughborough Sound Images (LSI) PC-card with a Texas Instruments TMS320C40 Digital Signal Processor (DSP).

5.2 Hardware/Software

The hardware and software listed below was needed to be able to program the DSP in C and to communicate with it.

- LSI PC-card with a Texas Instruments TMS320C40 at 40 MHz.
- Dell Pentium 90 host computer, running MS-DOS.
- Texas Instruments TMS320C40 C compiler development kit.
- Microsoft C version 6.00 for MS-DOS.
- LSI development kit with Microsoft libraries to support the DSP-card.

5.3 Differences between Matlab-C and DSP-C

A number of changes has to be made when porting from a HP-UX to a DSP environment. The list represents the changes made to the DMT simulation model:

FFT In the case of the DMT simulation model the optimized function for the DSP is (obviously) the FFT routine. This routine is obtained from the Texas Instruments bulletin board system available on the Internet.

Communication Due to the fact that the DSP card does not have its own display and/or file system, the input and output is taken care of by the host system. A communication protocol has to be included in the DSP software to supply the DSP with the necessary data and to get the results back to the PC. The LSI card has 4K word of dual ported RAM which makes it possible to send data to and from the PC. Unfortunately only the PC side of this communication mechanism was provided. To be able to access the common memory parts of the LSI card from de DSP a library with C-callable machine language routines were written.

Memory management The memory management for the model is changed from letting the operating system determine where certain variables are located to fixed data blocks with fixed number of bytes and variables.

Features The DSP implementation of the DMT type modem has less features as the simulation model. Some of the functionality and parameters are stripped to be able to get an acceptable simulation time. The synchronization and bit-per-carrier-allocation are not implemented.

5.4 Performance estimations

To estimate the performance of an DMT type receiver implementation which would be running on a TMS320C40 a calculation is made to estimate the number of cycles required to decode one DMT frame:

TMS320C40 at 40 MHz The cycle time of the processor is 25 ns.

FFT time The FFT is 512 long, in a RADIX-2 type algorithm this requires 8633 cycles if a TMS320C40 optimized architecture is used [27, 28]. This means each FFT takes $8633 * 1 * 25 = 215\mu s$ to calculate.

nr. of sub-carriers With an FFT length of 512 the maximum number of carriers which can be used is 254. For calculation purposes it is easy to set the number of carriers to 250.

Frequency domain equalizer The frequency domain equalizer requires one complex multiplication per sub-carrier. One complex multiplication requires 4 multiply instructions, this is 4 cycles, 100 ns per sub-carrier. The total required time is $250 * 4 * 25 = 25\mu s$.

Decoding QAM info Decoding the QAM information takes roughly 1 cycle for slicing, 3 cycles for converting the QAM in-phase and quadrature information into a symbol and 1 cycle for outputting the right bitstream. This has to be done for every carrier, therefore it takes $5 * 250 * 25 = 31.25\mu s$.

When all these figures are added the theoretical performance for decoding one DMT frame is $\approx 280\mu s$, this means $\frac{1}{280 \cdot 10^{-6}} = 3571$ frames per second can be decoded. If the number of bits send is equal to the performance measurements for the 26 Mbit/s situation (1640 bit/frame), the total bitrate will be $3703 * 1640 = 5.8$ Mbit/s. If a 25 Mbit/s receiver has to be build with these DSPs 5 will be needed for decoding the frames only.

6 Conclusion

Two conclusions can be drawn from this research project:

- The performance analyses of the simulation model shows DMT is an option for VDSL, both the 25 and 52 Mbit/s implementation are possible, where the 25 Mbit/s implementation does not require a TEQ which does impulse response shortening.
- Implementation of a 25 Mbit/s DMT modem on a single TMS320C40 DSP is not possible, because the number of frames/s transmitted is 5 times larger than the number of frames/s one TMS320C40 at 40 MHz can decode.

Future work

Simulation model time domain/frequency domain equalizer Both time domain and the frequency domain equalizer performance and implementation has to be investigated.

Echo cancelation Echo cancelation has not been investigated, it can increase performance without increasing the total bandwidth required, see [1].

Bandwidth taken by one sub-carrier The assumption that the amplitude characteristic is flat and the phase is linear in a sub-carrier is valid only when the sub-carrier frequency band is small. The frequency bands in the simulated VDSL modem are $1.944 \cdot 10^6/512 = 37.969 \cdot 10^3$ [Hz] wide. The effect of the width of these bands has to be investigated.

Memory management C model Some combinations of parameters of the C model result in a Matlab Bus error due to memory management errors.

Speed optimizations DSP C model Assembly code has to be written for the DSP to get the estimated bitrate.

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