# Adaptive Secure Data Transmission Method for OSI Level 1

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Dissertation was accepted for the defense of the degree of Doctor of Philosophy in Engineering at Tallinn University of Technology in August 2005.

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Commencement: September 15, 2005

Declaration: I declare that this thesis is my original unaided work. It is being submitted for the degree of Doctor of Philosophy in Engineering at Tallinn University of Technology. It has not been submitted before for any degree at any other university.

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ISSN 1406-4731

ISBN 9985-59-560-2

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# Abstract

This thesis discusses the problems of physical level data transmission of biomedical information systems over different channels in telecommunication networks. Existing standards of band-limited data transmission (modem standards) are based on hardware technology and defined for fixed not varying channels characteristics (bandwidth, signal-to-noise ratio etc). The methods recommended are not adaptive and thus modems do not show high performance used over different channels. On the other hand software based modulation systems use Fourier theory and algorithms in generation and detection of complex multi-carrier waveforms, which gives the possibility to obtain a higher performance than traditionally. Adaptive data communication based on software algorithms is used increasingly in modern wired and wireless communication networks as a result of technology development. These adaptive data communication methods give security and near optimal performance (bit rate, bit error rate etc) in different cases. These systems are investigated in this thesis and applied to the biomedical data transmission needs of the future.

The research methods are:

- Literary research is used in problem formulation and review of studies.
- Measurement of data communication in digital network is an approach to investigate data transmission problems.
- A robust tool (worksheet modeling and simulation) was developed for evaluation of voice coding, Gaussian or granular noise channels, and legacy and adaptive modulation methods. Simulations with this worksheet method help in investigating new modulation methods (waveforms).
- Discrete Fourier transform is used as the main mathematical method throughout this thesis in the design of modulation methods and modeling of waveforms.

Results are verified as:

- Simulation results are verified with standard data modem measurements and literature.
- A prototype adaptive modem is designed based on the simulation research results.
- Research results are verified with field tests using adaptive modem prototypes.
- Error performance and sensitivity study of the modeled communication system is made with data transmission simulations using band-limited adaptive multi-carrier data transmission waveforms over different channels.

The results are: traffic analysis of a biomedical data system, design of a robust worksheet simulation system, formulation and analysis of the adaptive band-limited data transmission for Gaussian and granular channels, and an adaptive data modem prototype based on discrete Fourier transform (DFT) approach. The main contribution of the thesis is: formulation of the adaptive secure data communication method for OSI level one (physical), its potential application (a proposal) and new knowledge of the use of DFT in generation and detection of waveforms.

**Adaptiivne turvaline infovahetuse meetod OSI tasemele 1**

# Kokkuvõte

Käesolevas väitekirjas käsitletakse probleeme, mis on seotud biomeditsiinilise info vahetusega telekommunikatsioonivõrkudes, kasutades erinevaid infokanaleid. Olemasolevad, piiratud riba korral kehtivad, infovahetusstandardid (modemi standardid) baseeruvad riistvaralisel tehnoloogial ja kehtivad fikseeritud, mitte muutuvate infokanali parameetrite korral (ribalaius, signaal-müra suhe jne). Soovitatud meetodid ei ole adaptiivsed ja sellised modemid ei saavuta kõrgeid tehnilisi näitajaid muutuvate infokanalite korral. Samal ajal tarkvaral baseeruvad modulatsioonisüsteemid kasutavad Fourier teisendust ning algoritme nii komplekssete multikandesageduslike signaalide genereerimisel kui detekteerimisel, mis annab võimaluse saavutada paremaid tehnilisi näitajaid kui eespoolnimetatud traditsiooniliste süsteemide abil. Tehnoloogilise arengu tulemusena leiab
kaasaegsetes nii traat- kui ka traadita sidevõrkudes üha laiemat kasutamist adaptiivne andmevahetus, mis baseerub vastavatele tarkvaralistele algoritmidele. Sellised adaptiivsed infovahetusmeetodid kindlustavad parema turvalisuse ja optimaalsele lähedased tehnilised parameetrid (ülekandekiirus, vigade arv jne) väga erinevates olukordades. Käesolev väitekiri ongi pühendatud adaptiivsete süsteemide uurimisele, samuti nende praktilise kasutamise võimalustele tulevikus erinevates biomeditsiinilistes infosüsteemides.

Töös kasutatud uurimismeetodid:

- Kirjanduslikku uurimist on kasutatud probleemi formuleerimisel ja ülevaatelises osas.
- Infovahetuse parameetrite mõõtmist digitaalses võrgus on kasutatud vastavate andmevahetuse probleemide uurimisel.
- On välja töötatud lihtne töövahend (probleemi modelleerimiseks ja simuleerimiseks), mille abil on uuritud kõne kodeerimist, Gaussi ja digitaalseid mürakanaleid ning fikseeritud ja adaptiivseid modulatsioonimeetodeid. Selle väljatöötatud meetodiga simuleerimine võimaldas uurida erinevaid modulatsioonimeetodeid ja signaalikujusid.
- Põhilise matemaatilise meetodina on selles väitekirjas kasutatud diskreetset Fourier teisendust nii erinevate modulatsiooniviiside väljatöötamisel kui ka signaalikujude modelleerimisel.

Saadud tulemusi on kontrollitud, kasutades:

- Simulatsioonide tulemusi on võrreldud nii standardsete modemite mõõtetulemustega kui ka kirjanduse põhjal.
- Adaptiivse modemi prototüüp on välja töötatud kasutades simulatsioonidel saadud tulemusi.
- Saadud uuringutulemusi on võrreldud erinevate adaptiivsete modemite prototüüpidele korraldatud testidega.
- Modelleeritud infovahetussüsteemi vigade analüüsi ja tundlikkuse uurimisel on kasutatud piiratud sagedusribaga adaptiivse multikandesagedusliku signaaliga infovahetuse simuleerimist üle muutuvate kanalite.

Töö käigus on saadud järgmised tulemused: biomeditsiinilise andmevahetussüsteemi info liikumise analüüs, lihtsa simulatsioonisüsteemi väljatöötamine, adaptiivse piiratud sagedusribaga infovahetuse formuleering ja analüüs nii Gaussi kui digitaalsete kanalite jaoks ning diskreetsel Fourier teisendusel (DFT) baseeruva adaptiivse modemi prototüübi väljatöötamine.
Väitekirja põhiliseks panuseks on: adaptiivse turvalise infovahetuse meetodi OSI tasemele 1 formuleerimine, selle potentsiaalne kasutamine (ettepanekuna) ja uued teadmised DFT kasutamisest signaalide genereerimisel ja detekteerimisel.

# Acknowledgements

First of all, I thank my supervisor Professor Kalju Meigas accepting me as his student in Tallinn and helping me in starting and finding practical solutions during the research process.

I express my gratitude to many people in Helsinki University of Technology (HUT) for their excellent contacts to Tallinn University of Technology. Without their knowledge I would never have started my research in Tallinn. I express my gratitude to Professor Seppo J. Halme, the former chief of the Communications Laboratory at HUT, who was my instructor in simulation research and modem prototype development.

Special thanks to Mr. Lloyd Bethell.

I acknowledge the funding of the Foundation of Technology, the Finnish Society of Electronics Engineers, the Scientific Advisory Board for Defence, the National Technology Fund, and the Foundation for Finnish Inventions.

My family, especially my father, my wife Raili and our children have supported me - I would never have finished my PhD research without their encouragement.

# Abbreviations

| | |
|---|---|
| xDSL | All types of DSL |
| A/D | Analogue/digital, analog-to-digital |
| ACR | The American College of Radiology |
| A/D | Analog-to-Digital |
| ADC | Analog-to-Digital Converter |
| ADM | Adaptive Delta Modulation |
| ADSL | Asymmetric Digital Subscriber Line |
| ANSI | The American National Standard Institute |
| ARP | Address Resolution Protocol |
| ASCII | American Standard Code for Information Interchange |
| ASIC | Application-Specific Integrated Circuit |
| AWGN | Additive White Gaussian Noise |
| BER | Bit Error Rate |
| BICM | Bit Interleaved Code Modulation |
| B-ISDN | Broadband Integrated Services Digital Network |
| BLER | Block Error Rate |
| BW | Bandwidth |
| $C^2$ | Command and Control |
| CCD | Charge Coupled Device |
| CGI | Common Gateway Interface |
| CP | Cyclic Prefix |
| CR | Computed Radiography |
| CT | Computed Tomography |
| D/A | Digital-to-Analog |
| DAB | Digital Audio Broadcasting |
| DAC | Digital-to-Analog Converter |
| DF | Digital Fluoroscopic imaging, Dark Field microscopy |
| DFT | Discrete Fourier Transform |
| DICOM | The Digital Imaging and Communications in Medicine |
| DM | Delta Modulation |
| DMT | Discrete Multi-Tone |
| DPSK | Differential Phase Shift Keying |
| DSL | Digital Subscriber Line |
| DSP | Digital Signal Processor |
| DVB | Digital Video Broadcasting |
| ED | Euclidean Distance |
| ESM | European Simulation Multiconference |
| ETSI | The European Telecommunication Standard Institute |
| EUROCOM | Communications Committee of Eurogroup |
| FAX | Facsimile |
| FDF | Finnish Defence Forces |
| FDM | Frequency Division Multiplex |

| | |
|---|---|
| FEQ | Frequency Domain Equalizer |
| FFT | Fast Fourier Transform |
| FH | Frequency Hopping |
| FPGA | Field-Programmable Gate Array |
| FSK | Frequency Shift Keying |
| HDTV | High Definition TeleVision |
| HIS | Hospital Information System |
| IDFT | Inverse Discrete Fourier Transform |
| IEEE | The Institute of Electrical and Electronics Engineers, Inc |
| IETF | The Internet Engineering Task Force |
| IFFT | Inverse Fast Fourier Transform |
| IMAC | Image MAnagement and Communication |
| IN | Input |
| IP | Internet Protocol |
| ISAC | Image Save and Carry |
| ISDN | Integrated Services Digital Network |
| ISI | Inter Symbol Interference |
| ISO | International Standards Organization |
| ITU | International Telecommunications Union |
| ITU-T | The ITU Telecommunication Standardization Sector |
| JIRA | Code name of an issue tracking and project management application |
| JPEG | Joint Photographic Experts Group |
| JTRS | Joint Tactical Radio System |
| LAN | Local Area Network |
| MB | Megabyte |
| MCM | Multi-Carrier Modulation |
| MDIS | The Medical Diagnostic Imaging Support system |
| MFC | Multi frequency code |
| MFSK | M-ary noncoherent Frequency Shift Keying |
| MIB | Medical Instrumentation Bus |
| MILCOM | IEEE Military Communications Conference |
| MIPS | Million Instructions Per Second |
| MLA | Modulation Level Analyzer |
| MR | Magnetic Resonance |
| MRI | Magnetic Resonance Imaging |
| NC | Numerical Control |
| NEMA | National Electrical Manufacturers Association |
| OFDM | Orthogonal Frequency Division Multiplexing |
| OOA | Object-Oriented Analysis |
| OOK | On/Off Key modulation |
| OSI | Open Systems Interconnect model of ISO |
| OSE | Operational Spectrum Effectiveness |
| OUT | Output |
| PACS | Picture Archiving and Communications System |
| PAPR | Peak-to-Average Power Ratio |
| PC | Peersonal Computer |
| PCASSO | Patient-Centered Access to Secure Systems Online |
| PCM | Pulse Code Modulation |
| PCT | International PCT Patent Application (PCT/FI99/00952) |
| PHD-RS | Personal Health Data Recording System |
| PSK | Phase Shift Keying |
| PSP | Polynomial Signal Processing |
| PSTN | Public Switched Ttelephone Network |
| PTWC | The Pacific Tsunami Warning Center |
| QAM | Quadrature Amplitude Modulation |

| | |
|---|---|
| R2 | A regional signaling system of ITU-T |
| RF | Radio Frequency |
| RFC | Request for Comments |
| RI | Radiological Information |
| RIS | Radiological Information System |
| RND | Random |
| ROC | Receiver Operating Characteristic curve analysis (Metz, 1978; Zweig & Campbell, 1993) |
| RSTP | Rapid Spanning Tree Protocol |
| RX | Receiver |
| SDR | Software Defined Radio |
| SL | Loss of Synchronization |
| S/N | Signal power over average white noise |
| SNR | Signal-to-Noise Ratio |
| Sync | Synchronization |
| TCM | Trellis-Coded Modulation |
| TCP/IP | Transmission Control Protocol / Internet Protocol |
| TDM | Time Division Multiplexing |
| TEQ | Time Domain Equalizer |
| THD | Total Harmonic Distortion |
| TX | Transmitter |
| UHF | Ultra High Frequency 300 MHz – 3 GHz |
| UMTS™ | Universal Mobile Telephone Service |
| UWB | Ultra Wide Band |
| VDSL | Very high rate Digital Subscriber Line |
| VF | Voice Frequency |
| VHF | Very High Frequency 30 – 300 MHz |
| WLAN | Wireless Local Area Network |
| WNW | Wideband Networking Waveform |
| X-CT | X-ray Computed Tomography |
| XFT | eXtra Fast Transport |

# Chapter I

# 1. Introduction

This chapter is organized into background, motivation of the work, evolution of telecommunications, objectives and, outline of the thesis. The motivation includes earlier activities and work experiences and contribution papers of this thesis. The evolution section gives a review of other studies of telecommunications, standard modems and their performance, waveforms and the idea of multi-carrier systems. Research methods and research problems are discussed in the last section: objectives of the thesis and outline of the work. The objectives describe the aim of the research. Formulation of the adaptive secure data communication method for OSI level one, its potential applications and new knowledge of the use of discrete Fourier transform (DFT) in generation and detection of waveforms is the expected benefits of this study. The outline of the work defines the structure of the thesis.

## *1.1. Background*

The most unfortunate things in the modern world awoke the world, the World Trade Center on September eleven in 2001 and an earthquake followed by a tsunami in Asia on Boxing Day 26.12.2004, when we think of all the precautions against manmade catastrophes and convulsions of nature. The former perfected a military transformation in USA. The latter made it clear that an information system based on telecommunications could have saved most of the nearly 180 000 deaths. It is sad that a working alert system was not made there, in spite of the technology known and used in most countries in the Pacific Basin. The Pacific Tsunami Warning Center (PTWC), established in 1949, provides warnings for teletsunamis to the areas and US interests.

The communication system from the information source via radio waves (public broadcast, mobile cellular, satellite service etc) to the people exists. The completing alternatives to public broadcasting, for example messages with cellular phones, can serve as redundancy in securing the authority made alerts. The motto of the Signal Regiment of the Finnish Defence Forces (FDF) says "Denuntiatio solum translata valet". The tsunami also reminds the author of the development efforts within telemedicine in general as one of the most important issues. It is worth all the efforts already made and those that will be made in the future. This encourages the author in publishing some results of the secure adaptive data communication methods studied and developed during several years and especially during 2000-2005 at the Tallinn University of Technology. The results of this thesis will be presented on the following pages. An adaptive data communication method (soft data generation and DFT detection method) will be presented that can be included in public broadcasting systems all over the world. The idea is that broadcasting stations are coupled in the alerting data sources and local authorities have standard receivers equipped with a small additional audio circuit for detection of an alerting coded waveform. For telemedicine communications the waveform will be proposed as a secure OSI level 1 wireless data transmission.

## *1.2. Motivation of the Work*

The public switched telephone network (PSTN) in Finland was in an early phase of the automatic traffic during the time 1968-1970, when the author was an office engineer at the Post and Telegraph Administration in Helsinki. The value of automatic telephone networks was well understood during the automation projects in various places in Finland (Porvoo, Imatra, Keuruu, Lieksa, Kristiinankaupunki, Saarijärvi, Kemijärvi etc). The author completed his M.Sc thesis in 1969 about the R2, which is a MFC signaling in the automatic PSTN. The result of that work is described in reference [Par00].

During active military service in 1970-1977 the author was also involved in discussions concerning the automation of the, at that time, manual tactical military telephone services [Lei72]. The author's first post-graduate thesis (Lic.Tech 1975) and a publication in HUT (1987) was a result of the studies for the evaluation of the availability of a limited network, papers [Lal75] and [Lal87]. For eleven years, 1977-1988, the author worked at the vocational training centers at Jyväskylä leading a department and at Järvenpää as a leader. Among several activities during that time he was also involved in the development of training programs for numerically controlled (NC) machinery and microcomputers. At that time the evolution of microprocessors and personal computers (PC) was intensive. The use of PCs in education and small businesses with the idea of electronic sheet, Visicalc in 1978-1979 [Rad80], was started. The author also had the TRS-80 Model I at home.

The post of Assistant Professor of Applied Electronics at Lappeenranta University of Technology was founded in 1989 and the author was its first holder conducting courses on "Microprocessors" and "Machine language programming. The author published a textbook about microcomputers used in vocational training centers and Lappeenranta University of Technology. At Helsinki University of Technology the author has conducted courses on "Teleautomation" in 1984-1985 and 1988-1989, and "Digital systems" in 1990-1991 working at the Department of Electrical and Communications Engineering.

The automation and digitization of military networks, both tactical (moving) and strategic (fixed), and the start-up of the military information age with the Intranet and first personal computers (PC) was established in about 1988. This time was his second military service in 1988-2004 (retired on 31.12.2004). The author designed training guides and planning methods for new systems including modeling and simulation methods described in references [Cac93, Mis98] and a robust simulation tool of his own [Lal04a-b]. Design methods for the tactical radio links and wireless networks were one of the main efforts during 1998-2004. The network planning system is based on modeling and computer simulations using cartographic data from Finland (digital maps, elevation models, clutter and forest height information).

Oracular forecast is found in reference [Fdf03] about the roles and tasks of defense forces and their mutual relations. International peace keeping forces are gaining a more significant role as the threat of armored attack diminishes. Large-scale national disasters and defense against terrorism necessitate the use of armed forces in support of Civil Society. The reference presents a software radio program and its new adaptive wideband networking waveform for FDF [Fdf03]. Also the need of command and control (C2) systems with wireless communication networks using (VHF, UHF) digital radios motivated into several studies of adaptive data communication systems One result of the author's studies in these areas was the concept of an adaptive modem and later an adaptive security method for data communication [Lal99, Lal00, Lal02, Lal04b]. The adaptive modem was found as a novel solution in the PCT review in 1999. Both the adaptive securing method for OSI level one (physical) data transmission and the adaptive modem itself have been awarded with an invention prize of the FDF in 1999 and in 2002. This system is the main result in this thesis, discussed in several references [Lal97a-b, Lal99, Lal00, Lal01, Lal02, Lal04a-b].

The background described above gave the motivating force for the academic work at the Tallinn University of Technology during 2000-2005 with the results published in the following proceeding of

- Eurocom conference 2000 in München "Adaptive modem" [Lal00].
- MILCOM conference 2001 in MacLean, USA with title "Adaptive modem technology" [Lal01]
- MILCOM conference 2002 in Anaheim, CA, USA with title "Basic theory of adaptive data transmission" [Lal02].
- ESM conference 2004 in Magdeburg, Germany with title "Modeling and simulation of biomedical data networks" [Lal04a].
- MILCOM conference 2004 in Monterey, CA, USA with title "Robust simulation of waveforms" [Lal04b].

The main contributions of this thesis are in the theory of adaptive data communications, a simulation of telemedicine traffic, and the soft detection and generation method of adaptive band-limited waveforms based on the discrete Fourier transform (DFT), Appendix 1. The result of the studies is defined in the proposal for secure telemedicine and biomedical information data transmission method for OSI level 1.

# 1.3. Evolution of Telecommunications

A Shannon's result is the concept that every communication channel had a speed limit, measured in binary digits per second. The famous Shannon Limit is the familiar formula for the capacity of a white Gaussian noise channel [Sha48]. In a paper [Mos02] a new paradigm called "capability" is proposed, which gauges the effectiveness of a steganographic method. It includes payload carrying ability, detectability, and robustness components. A demonstration is made that a compressed image (JPEG) always has the potential to carry hidden information [Mos02]. In the adaptive data transmission method studied in this thesis hidden information will be carried in the signal space. The use of zero-error capacity for channel analysis is discussed. The error free channel capacity is one substance used in this thesis.

## 1.3.1. Studies of Telecommunication

Alec Reeves invented PCM in 1937 as a result of a research group in Paris [Alc92, Rob04]. Severe problems with noise and distortion were reported while bandwidth was large in radio links. Technologically it was too early to use PCM in practice. The encoding of signals with 1-bit code has been studied quite early 1946 [Del46]. The introduction of delta modulation (DM) by de Jager was made in 1952 [deJ52]. Voice coding methods (PCM and DM) produce granular noise. Bandwidth and signal-to-noise ratio (SNR) are important parameters used in evaluation of a system's performance in this thesis.

### Voice Grade Data Communication

At present, "capacity" is the prevailing paradigm for covert channels. Voice grade data transmission on wired channels has been studied by ITU-T as early as in March 1960 [Itt61, Itu60]. Modulation method of the first data transmissions was FSK and later MFSK. Modem development is based on modulation method, bit rate and symbol rate, Table 1.1.

Table 1.1. Standard modems [Lal97a]

| Year | Recommen-dation | Bit rate bit/s | Spectrum Hz Measured | Carrier frequency Hz | Symbol rate baud | Modulation |
|---|---|---|---|---|---|---|
| 1964 | V.21 | 300 | | | 300 | FSK |
| 1964 | V.23 | 1200 | **900 - 2500** | 1300, 2100 | 1200 | FSK |
| 1968 | V.26 | 2400 | **900 - 2700** | 1800 | 1200 | 4-DPSK |
| 1972 | V.26 bis | 2400 | **900 - 2700** | 1800 | 1200 | 4-DPSK |
| 1976 | V.27 ter | 4800 | 800 - 2900 | 1800 | 1600 | 8-DPSK |
| 1976 | V.29 | 9600 | | 1700 | 2400 | 16-QAM |
| 1980 | V.22 | 1200 | 600 - 2900 | 1200, 2400 | 600 | 4-DPSK |
| 1984 | V.22 bis | 2400 | 600 - 2950 | 1200, 2400 | 600 | 16-QAM |
| 1984 | V.32 | 9600 | 300 - 2950 | 1800 | 2400 | 16-QAM |
| 1984 | V.33 | 14400 | 300 - 3200 | 1800 | 2400 | 32-QAM |
| 1994 | V.34<br><br><br>version 96 | 28800<br><br><br>31200<br>33600 | | 1800 | 2400<br>2800<br>3000<br>3200<br>3429 | 16-QAM<br>32-QAM<br>64-QAM |

Modem development is seen in the modulation method: phase difference, phase and amplitude difference (QAM), and finally coding methods were used. The key idea was that the operations of modulation and coding are combined. Limitations on conventional block and convolutional codes on band-limited channels have motivated to introduce other coding methods TCM (Trellis-Coded Modulation) scheme, discovered by G. Ungerboeck in 1976 [Ung82] and its several variants for example iterative turbo BICM (Bit Interleaved Code Modulation) scheme [Big91 SNg03] for decreasing error probabilities.

## *Performance Analysis*

The performance analysis for a TCM system with a Viterbi decoder [Vit67] is derived in reference [Ung82] as

The upper bound on the node error rate is

$$P_e \leq Q\left(\sqrt{\frac{d_{free}^2 E_s}{2N_0}}\right) e^{\left(\frac{d_{free}^2 E_s}{4N_0}\right)} T(D) \tag{1.1}$$

with

$$D = e^{\left(\frac{E_s}{4N_0}\right)} \tag{1.2}$$

and the upper bound on BER is:

$$P_e \le \frac{1}{m} Q\left(\sqrt{\frac{d_{free}^2 E_s}{2N_0}}\right) e^{\left(\frac{d_{free}^2 E_s}{4N_0}\right)} \frac{\partial(D,I)}{\partial I} \tag{1.3}$$

with

$$D = e^{\left(\frac{E_s}{4N_0}\right)}, I=1 \tag{1.4}$$

Where *T(D)* is the generating function of the direct graph (state diagram of the trellis code). *T(D,I)* is the augmented version of *T(D)*with components of the I terms denote the number of information bits errors associated with each error event. $N_0$ is the one sided noise spectral density of AWGN, $E_s$ is the average energy of the signal constellation and m is the number of information bits carried by each symbol.

The derivation of generating function is complicated as the number of states in the TCM increases. At high S/N values an approximation is quite accurate without *T(D,I)* [Ung82]:

$$P_e \approx N(d_{free}) Q\left(\sqrt{\frac{d_{free}^2 E_s}{2N_0}}\right) \tag{1.5}$$

$$P_b \approx \frac{d_{free}}{m} Q\left(\sqrt{\frac{d_{free}^2 E_s}{2N_0}}\right) \tag{1.6}$$

Here the $N(d_{free})$ is the average number of sequences that are distance $d_{free}$ from the transmitted sequence.

For example, using TCM the error rate can be reduced by three orders of magnitude (from 1 in 10 to 1 in 10000). In a classical view the spectral efficiency as the number of bits per second transmitted per one hertz of bandwidth is decreased due to the increased code rate. However, the modulation symbol set could be enlarged when coding is used relative to that needed for uncoded case [Ung82] and:
- If the signal set dimensionality per info bit is unchanged the power spectrum remains unchanged (no BW expansion or change in spectral efficiency).
- The signaling rate does not change if coding and modulation is performed with respect to ED (Euclidean distance) between coded modulation sequences.

TCM code is recommended by ITU-T with
V.32 for 9.6 kbps over two-wire telephone lines and 14.4 kbps one-wire lines.
V.17 for use with 64-QAM and 128-CROSS 14.4 kbps FAX traffic over standard phone lines.
V.33 14.4 kbps and V.34 28.8 kbps.

For evaluation of a modulation method a formula for BER versus S/N (bit error rate versus signal-to-noise ratio) are generated.

In general the Gaussian error integral, function $Q(x)$, gives an approximation for error performance $P_e$ of sequences of transmitted symbols in signal space, where $d_{min}$ minimum Euclidean distance described in references [Vit67, Ung82]. At high signal-to-noise ratios [Ung87]:

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^{\infty} e^{\frac{-y^2}{2}} dy \tag{1.7}$$

$$P_e = Q\left(\frac{d_{\min}}{2\sigma}\right) \tag{1.8}$$

$$P_e = N_{free} Q\left(\frac{d_{free}}{2\sigma}\right) \tag{1.9}$$

Where $N_{free}$ is the average number of sequences at $d_{free}$. In TCM the Euclidean distance $d_{free}$ between signal sequences is increased to get a lower error rate. The Euclidean distance between two points $p$ and $q$ in N dimensions $i=1\ldots N$ as

$$d = \sqrt{\sum_{i=1}^{N} (p_1 - q_i)^2} \tag{1.10}$$

Formulae (1.11)-(1.12) for FSK and DPSK are proper references for other modems developed later, reference [Car86].

$$P_b = \frac{1}{2} e^{\frac{-\gamma}{2}} \tag{1.11}$$

For FSK, envelope detection, and modulation speed $r_b / B_T = 1$.

$$P_b = \frac{1}{2} e^{-\gamma}, \tag{1.12}$$

For DPSK, phase-comparison detection and modulation speed $r_b / B_T = 1$.

A comparison of modulation methods is presented in reference [Car86] pp. 553-554. The selection of a modulation method is made at a common standard for comparison purposes $P_b = 10^{-4}$. For example, M-ary DPSK is a reference for modulation methods, formula (1.13). The performance of 2-PSK…32-PSK is at $P_b = 10^{-4}$ about $r_b / B_T = 1 \ldots 5$ bps/Hz and $\gamma = 8 \ldots 21$ dB.

$$P_b = \frac{1}{K} Q\left(\sqrt{4K\gamma \sin^2 \frac{\pi}{2M}}\right) \tag{1.13}$$

Where bit error rate is $P_b$ and $\chi$ is energy to signal ratio, data conversion factor is $K = \log_2 M$ and number of symbols in a M-ary system, for example $M = 2^K$.

In this thesis FSK and DPSK are used as references in performance analysis.

### *Waveforms*

The basic waveform is a simple sinusoidal signal as presented in formula (1.14)

$$s(t) = A\sin(2\pi mft + P) \tag{1.14}$$

It has four information carrying parameters in it (A, f, t and P). These parameters are used in data modulation methods. The multi-carrier or multi-channel signals are described Waveforms as a Fourier presentation in formula (1.15)

$$S(t) = \frac{1}{2}c + \sum_{n=1}^{\infty} a_n \sin(2\pi nft) + \sum_{n=1}^{\infty} b_n \cos(2\pi nft) \tag{1.15}$$

Any signal can be presented as a Fourier series [Mar62]. An algorithm for the machine calculation of complex Fourier series was first presented in 1965 [Coo65].

### *Digital Generation of Waveforms - A symbol-based approach*

In present digital networks data transmission is virtually transmitted in digital form. The nature itself is analog and also analog are the waveforms of Internet access circuits and radio waves (presently called air interface, former name was ether). Thus the need for analog waveforms and modeling methods still exists. However, the waveforms themselves can be made with digital modulation methods. The present studies of waveforms are focused on wideband applications (B-ISDN, UMTS, WNW etc) both in fixed wired and moving wireless systems. Several applications are based on the OFDM waveform (Orthogonal Frequency Division Multiplexing) scheme. The basic idea of multi-carrier modulation was introduced and patented in the mid 60s by R.W.Chang [Cha66, Wei71]. The sampled waveform may be presented as x(t) in [Guo02] formula (1.17). The analog signal at DAC output after quadrature modulation in a software radio type modulator is

$$\tilde{s}(t) = R\{x(t)e^{j(\omega_C t + \phi(t))}\} \tag{1.16}$$

With amplitude of the input signal to the RF transmitter

$$x(t) = \sum_{i=0}^{+\infty} \sum_{k=0}^{N_S - 1} \sqrt{\{[s_I^2(k) + s_Q^2(k)]p(t - k\frac{T_S}{N} - iT_S)\}} \tag{1.17}$$

and the phase

$$\phi(t) = \tan^{-1}\left(\frac{\tilde{s}_Q(t)}{\tilde{s}_I(t)}\right) \tag{1.18}$$

where TS is the period of one time-domain OFDM symbol, NS is the number of samples in one symbol after CP and p(t) is the pulse function of the symbols. The high peak-to-average-power ratio (PAPR) of OFDM systems introduces non-linear distortion in the transmitter and causes both in-band and out-of-band spectrum re-growth. Undesired effects are [Guo02]:
- Nonlinearities.
- Intermodulation among subcarriers.
- Undesired out-of-band radiation.

In general, the complexity of the equalizer grows with higher symbol rates because as the transmission rate increases (symbol periods become shorter), the effects of ISI degradation are higher. In the case of both DMT and OFDM, the channel is divided into many sub-channels situated in different frequency bands, with each sub-channel utilizing a carrier with Quadrature Amplitude Modulation (QAM). This allows a bit stream with a very high transmission rate to be sub-divided into many bit streams with lower transmission rates, reducing the relative effect of ISI in each symbol period (which is now longer) and allowing simplification of the equalizer. Therefore, both OFDM and DMT systems can implement equalization in a simple fashion, due to their slower symbol rates. The OFDM waveform is resistive against multi-path errors.

In this thesis band-limited multi-carrier waveforms are the main subject.

### *Channel Modeling*

References of channel modeling [Sha48, Rum86, Agu03] and recommendations [Eur86, Itu88] give a classification of channels:
AWGN used for theoretical modeling and reference in evaluations of modulation methods.
Granular noise channel is generated in the digitization process of analog signals.
Multi-path channel is a wireless (mobile cellular, radio broadcast) channel.
This classification is used in this thesis.

## *1.3.2. Evolution to Adaptive Communications*

Adaptation is known as an adjustment by which a species improves its condition in relation to its environment. Culture is the human adaptive system.

### *Idea of Multi-Carrier Modulation*

The concept of Multi-Carrier Modulation (MCM) or Discrete Multi-Tone (DMT) has been known for many decades. In the 1960s, a multi-carrier modulation technique known as Orthogonal Frequency Division Multiplexing (OFDM) was invented, which utilized multiple sub-channels in the frequency domain [Cha66, Wei71]. The basic idea of multi-carrier modulation was introduced and patented in the mid 60's by R.W.Chang. The channel is sliced up into narrow little bands and multi-carrier modulation is used with individual modulation in each of the bands [Gal68, Gal01]. MCM techniques (OFDM and DMT) have evolved and have been adopted in various standard bodies such as IEEE 802, American National Standards Institute (ANSI), European Telecommunications Standards Institute (ETSI), and International Telecommunications Union (ITU). Multi-Carrier Modulation (MCM) [Sta99], divides the data into a number of low rate channels that are stacked in frequency and separated by 1/symbol rate. MCM, also called OFDM, is being proposed for numerous systems including audio broadcasting (DAB), video broadcasting (DVB), mobile wireless access (WLAN) and digital subscriber link systems (xDSL).

## *Definition of OFDM*

The recent OFDM method (sometimes called multi-carrier or discrete multi-tone modulation) is the basis of several standardized systems for data networks and cellular radio communications. A definition of OFDM systems [Eng03] describes it as: The available bandwidth W is divided into a number Nc of subbands (subcarriers or subchannels) each of width f=W/Nc. Data symbols are transmitted in parallel by modulating the Nc carriers. To assure a high spectral efficiency, the subchannel waveforms must have overlapping transmit spectra. They need to be orthogonal for enabling simple separation of these overlapping subchannels at the receiver. Multi-carrier modulations that fulfill these conditions are called Orthogonal Frequency Division Multiplex (OFDM) system.

## *OFDM System Model*

A high-level block diagram [Ram02] is shown in Appendix 2. The system model of an OFDM transmitter with RF is presented in reference [Guo02]. The information bits

$$[b_1 b_2 b_3 \ldots b_M] \tag{1.19}$$

are mapped into the I/Q channel baseband symbols using a modulation scheme such as phase-shift-keying (PSK) or quadrature-amplitude-modulation (QAM). Then each group of N symbols are packed into a parallel block

$$[S_1 S_2 S_{3.} \cdots S_N]^T \tag{1.20}$$

at the input to the IFFT.

OFDM symbols in the time domain over time interval

$$t \in [0, T_S] \tag{1.21}$$

are generated by the IFFT operation as,

$$s(k) = \frac{1}{\sqrt{N}} \sum_{n=1}^{N} S_n e^{j2\pi(k-1)(n-1)/N} \tag{1.22}$$

for $k=[1,2,\ldots,N]$

Then in Cyclic Prefix (CP) insertion, the first G coefficients are repeated after the original N coefficients and made serial for quadrature modulation. The analog signal at DAC output after quadrature modulation in a software radio type modulator is,

$$\tilde{s}(t) = R\{x(t)e^{j(\omega_C t + \phi(t))}\} \tag{1.16}$$

with amplitude of the input signal to the RF transmitter

$$x(t) = \sum_{i=0}^{+\infty} \sum_{k=0}^{N-1} \sqrt{\{[s_I^2(k) + s_Q^2(k)]} p(t - k\frac{T_S}{N} - iT_S)\} \tag{1.17}$$

and the phase

$$\phi(t) = \tan^{-1}(\frac{\tilde{s}_Q(t)}{\tilde{s}_I(t)}) \tag{1.18}$$

Where TS is the period of one time-domain OFDM symbol, N is the number of samples in one symbol after CP and p(t) is the pulse function of the symbols.

## *1.3.3. Discussion of Broadband Evolution*

The evolution to adaptive communications is due to many different factors including the development of MCM in the 1960s. Advances in the design of new communication systems are based, later in the 1970s, on the microprocessor, software memory, digital signal processing and chip manufacturing technology. Methods for doing coding, waveform shaping and equalization are still a developing area in communications. The terms "multilevel QAM digital radio system" or "variable rate QAM" or "symbol rate controlled modulation" were used in the late 1980s. Self-adjusting adaptive systems and OFDM (Orthogonal Frequency Division Multiplex) systems have been designed and used for many civil applications in the 1990s. The recent ubiquitous OFDM method is the basis of several standardized systems for data networks and cellular radio communications. In the 1990s also a new concept "adaptive modulation" is used.

Modeling and simulations have been used for telecommunication network planning since the development of simulation programming and the first simulators in 1964. Simulations are important in the evaluation and design of network structure, communication routes with traffic calculations, link budgets, radio wave propagation and circuit level analysis. At about the same time an algorithm for the machine calculation of complex Fourier series was presented in 1965. In the 1970s Discrete Fourier Transform (DFT) was studied. Soft generation and detection of waveforms with IFFT and FFT, Object-Oriented Analysis (OOA) were used in the 1990s.

Military applications will use new downloadable waveforms (a software algorithm) with Software Defined Radios (SDR). WNW (Wideband Networking Waveform) waveforms will be designed for military mobility and network access purposes according to some military development plans during 2004-2007.

## *1.3.4. Discussion of Some Communication Problems*

### *Biomedical Data Transmission Problems on Physical Level*

Biomedical data transmission problems on the physical level of the OSI model are presented according to reference [Var03]:

1. For the purposes of various databases and information systems off-the-shelf solutions are available. There are already many standards for platform independent representation and interchange of static or quasi-static medical records (patient data, insurance data, medical images etc.). These communication standards are based on the upper layers of the OSI model, and the lower layers, like the physical transmission, the media access and the routing methods are not defined.
2. As it can be established, to fulfill all of these requirements we must define the lower layers of the OSI model. Although many vendors are offering their monitoring systems, they lack an open communication technology and so lack the ability of interchangeability between devices of various vendors.

3. Clinical monitoring and intensive patient care mean a much different problem. In these applications the various bedside medical equipment are to be networked at a lower level of the communication hierarchy. This requires a dynamic, real-time, deterministic, fault-tolerant, and secure data exchange.

The 802.x standards lack some or more of these features. According to a survey in 1989 the main requirements of bedside medical monitoring systems are:
1. Interconnecting bedside devices of one or more patients.
2. Real-time, deterministic, secure data exchange
3. Frequent reconfiguration of the network, plug & play features (ease-of-use)
4. Support a wide range of existing hospital information systems and databases

### *Address Resolution Protocol (ARP) Problem*
Address Resolution Protocol (ARP) Problem is presented in reference [Noo99]. The Address Resolution Protocol (ARP) is the protocol used to map 32-bit IP addresses to the address scheme used by the data-link layer. The data-link layer (sometimes called the network link layer), which consists of the operating system device driver and corresponding network interface card, is responsible for dealing with the physical transport media. Each network interface has a unique hardware address, typically assigned by the manufacturer. ARP is often referred to as a "dynamic" protocol. This is due to the fact that its operation occurs automatically. The protocol works in the background, without concern to the application user or even the network administrator. It is the dynamic nature of ARP, which causes security issues.

## *1.4. Objectives of the Thesis*
The investigation of analog (voice, image) data communication waveforms over band-limited on voice grade circuits, soft modulation and detection methods in the mobile wireless or fixed wired networks is a general objective in this thesis. The focus is not in the wide bandwidth applications as in the general trend and in several other studies but in the limited base-band waveforms, modulation methods, and the security on the physical level (OSI model level 1), which has not been studied in the references. In summary the main objective of this study is the development or evaluation of a new concept of soft adaptive multi-carrier data transmission over band-limited wireless and fixed telephone channels especially for alert, telemedicine and authority needs.

According to reference [Var03] medical communication standards only exist at the moment for the higher level of medical care, like various databases and hospital information systems, and not for low-level communication between various diagnostic devices. The main goals are to ensure a real-time, deterministic and secure data exchange between the linked devices, and to present a user-friendly visualization of the patient' **s**tate.

Our main goal is to present a secure adaptive communication method for a medical data communication of OSI level 1. The method is also proposed for secure alarm signaling systems.

## *1.5. Outline of the Work*
This study is an analysis and synthesis of an investigation of an adaptive data communication method proposed for telemedicine and alert systems. The method is based on the present networks and different channel types (AWGN, granular noise and multi-path). Biomedical data processing and transmission are analyzed in Chapter 2. Transmission investigation methods are measurements in Chapter 3, modeling and simulations in Chapter 4, and field tests with adaptive data modem prototypes and new waveforms in Chapter 5.

The study is organized as follows: This Chapter is an introductory chapter including a review of adaptive communications. Chapter 2 provides an introduction to biomedical data processes and data transmission based on the literature. Biomedical data processes and transmission usually involves high quality biomedical images. Chapter 3 is an analysis and measurement review. The basic investigation results in developing new waveforms and evaluating their functionality in available transmission channels are presented in Chapter 4. Chapter 5 presents the theory of generation and detection of waveforms using a DFT based approach. In summary Shannon's, Fourier and Chang's theories are used in the formulation of the final results. A proposal and discussion of the adaptive secure data communication application is included for telemedicine and alert system. Chapter 6 is a summary of the thesis.

# Chapter II

# 2. Biomedical Data Processing and Transmission

Biomedical data processing uses image information. The importance and value of IMAC systems is found in medicine. Image management and communication (IMAC) has been developing since the First International Conference on IMAC held in June 1989 in Washington D.C. New technological innovations, PACS, HDTV and ISAC, were discussed in the second conference in 1991 in Kyoto. PACS is a picture archieving and communication system. HDTV is a high definition TV. ISAC is an abbreviation of "Image Save and Carry". A medical image analysis and diagnosis system has been developed in Australia [Ead01] and a system for patients as Patient-Centered Access to Secure Systems Online (PCASSO) in USA. Biomedical data processes and standards are described in several references [Ima91], [Bus02] e.g. European standardization in medical informatics is identified in a reference [Ima91] pp. 230-234. Medical computing and data standards are introduced in [Bus02] pp. 85-98. The future and advances in telemedicine are surveyed in reference [Bus02] pp. 129-136.

Security of biomedical or telemedicine data transmission on the OSI level, i.e. the physical level, is not available as a standard [Har05]. Wireless mobile information transmission in telemedicine is a new area of data transmission, legislation and studies. These areas and recent needs brought about by tsunamis, e.g. warning systems, are the focus of this thesis.

## 2.1. Biomedical Information Systems

### Development of E-Health and Telemedicine

Early days of and development of E-health, telemedicine applications and technologies are described in references [Eads01, Mah01, Rat05] as:

- The 1990s brought advances in image digitization and data compression technology, which enabled videoconferencing over lower bandwidth lines i.e. voice grade telephone lines and present wireless mobile phone connections.
- The typical telehealth model involves a hub hospital with satellite hospitals and clinics.
- Benefits of health care telecommunication technologies are: a. Distribution of resources. b. Access to resources. c. Cost of health care.
- Challenges: a. Professional practice. b. Guidelines. c. Malpractice issues. d. Reimbursement and legislation. e. Staff training.
- The boundary between medical and communication technologies will increasingly blur.

An important issue is the image resolution as:

- High resolution is crucial to teleradiology.
- Images have to be ideally transmitted by using lossless compression methods.

Three important attributes of image transmission are:

- Fidelity: Resolution, linearity and noise.
- Informativeness: The image conveys clinically important information.
- Attractiveness: The aesthetic properties.

## *Local Area Network*

Several different systems are in relation and connected together with a Hospital Information System (HIS) using a local area network (LAN). Systems are according to reference [Ima91]:

- Picture Archiving and Communications System (PACS).
- Radiological Information System (RIS).
- Hospital Information System (HIS).

HIS deals with patient identification, blood chemistry, diagnosis, medical history, infection and accounting. RIS consists of exposure record, file management, label print, reservation of examination and accounting. The main problems of large picture archieving and communication systems (PACS) are the need for high speed local network (LAN) and a mass storage device. Two major components of PACS are: the network and the data base.

Some conclusions about the hi-speed network for PACS were made in reference [Ima91] p.32-35:

- At least 100 Mb/s transfer rate in a network is equal to the present performance of the film-based system.
- The rate of data generation with the paths that the information flows in hospital were evaluated. ETHERNET had a signaling 10 MB/s but a transfer rate of about 200 kb/s. This transfer rate is a limiting factor.
- The very high required transfer rate could be achieved by the signaling rate of PACS network or the large efficiency of operation.

A prototype ETHERNET network XFT (eXtra Fast Transport) was designed in 1991 and is described in reference [Ima91] p.32-35. The performance, 500 Mbit/s signaling and 450 Mb/s fiber-optic serial point-to-point communication links, agrees with the specifications. In 1991 operating networks at gigabit per second speeds were demonstrated [Ste91].

## *Image Management and Communication*

The image management and communication (IMAC) system has not been utilized as quicly as expected since 1989 conference. Military operations (for example the Gulf War) is one area, where IMAC systems were needed. The image management and communication system must consist of [Ima91]:

- PACS (on-line).
- ISAC (off-line).

An Image Save and Carry (ISAC) committee was organised in 1989 supported by a foundation established by the Japanese government.

An estimation of the annual data of an entire hospital in Japan during one year (1988) is given in reference [Ima91] p. 67 as presented in Table 2.1.

Table 2.1. Annual data of a hospital [Ima91]

| | patients | images | data/image | data volume |
|---|---|---|---|---|
| Plain study, CR | 116,921+19,762 | 228,939+36,161 | 4.0 MB | 916 MB+145 MB |
| Enhanced study | 23,946 | 233,893 | 2.25 MB | 526 MB |
| CT, MR, nuclear | 29,556 | 361,289 | 0.5 MB | 180 MB |
| Total | 190,194 | 860,282 | 2.8 GB/disk | 1.7 TB<br>680 disks |

Another estimation of daily image volume in a hospital in Japan is also given in reference [Ima91] p. 330 as presented in Table 2.2.

Table 2.2. Daily image volume in a hospital [Ima91]

| | bits/image | no of exposures | bits |
|---|---|---|---|
| X-ray | 2048x2048x1 | 900 | 4.0E10 |
| CT | 512x512x1 | 800 | 2.5E9 |
| RI static | 256x256x8 | 10 | 0.3E8 |
| dynamic | 64x64x8 | 50 | 2.5E6 |
| DF | 1024x1024x8 | 600 | 0.6E10 |
| MRI | 512x512x8 | 500 | 1.6E9 |
| Total | | 2860 | 5.5E10 |

Another study evaluated the transfer speeds between neuroradiology network components. Transfer rates versus times of a 60 images study varied from 171 kb/s versus 24.5 min to 677 kb/s versus 1.55 min. The low speed problem is disturbing in scanning images. However, the lost images were reported to be even more of a disturbing problem, reference [Ima91] pp. 272-277.

These results are interesting as a reference for this thesis.

### *Medical Diagnostic Imaging Support*

The Medical Diagnostic Imaging Support (MDIS) system is a project of the U.S. Department of Defense. The goal was to achieve filmless medical imaging operations in the 1990s throughout the defense health care system.

A computerized analysis of lung textures for detection and characterization of interstitial diseases in chest images based on the power spectrum is presented in reference [Ima91] pp. 280-283. A comparison of ROC analysis curves obtained from radiologists and by means of the computerized method suggest that the computerized approach may provide perfornance similar to human observers in distinguishing lungs with mild interstitial diseases from normal lungs.

### *Picture Archieving and Communication Systems*

A simplified view of the Picture Archiving and Communications system (PACS) assumes that it can be decomposed into the following classes of subsystems [Ima91] p.12:

- Acquisition.
- Distribution.
- Storage.
- Processing.
- Display.

Since 1983, in Personal Health Data Recording System (PHD-RS) Japan and Picture Archiving and Communications system (PACS) in USA has started. The PHD-RS system is a personal filing system to carry all medical information of a patient: medical images, laboratory findings and past history under chronological editing. In the concept of PHD-RS in 1982, all personal data concering medical information must be gatherted onto in one magnetic tape cassette and carry with patient himself [Ima91] pp. 4-5.

A total information system using IBM 360 mainframe computer system was started in 1971 in Kitasato University Hospital, Japan. Microfilm system was considered but cost performance was not enough and they did not save manpower. Development in the 1980s based on computer technology, information transfer technology and information media progress as:

- In the field of radiology.
- X-ray computed tomography (X-CT).
- Magnetic resonance imaging (MRI).

- Single photon emission tomography (PET).
- Ultrasound tomography.
- Charge coupled device (CCD) camera for endoscopy.

All radiological images can be shown in digitalized patterns and recorded on optical disk. This tendency leads the medical information system to be image save and carry system.

### *A Web-Based Collaborative System for Medical Image Analysis and Diagnosis*

Reference [Ead01] explains the web-based collaborative system for medical image analysis and diagnosis. The system uses computer and network technologies and the Internet, to provide and support healthcare when distance separates the participants. There has been a lot of research carried out to develop the electronic Picture Archiving and Communications system (PACS) that is for a hospital wide network and for people to deal with medical image. The test results of the client-server system is presented in reference [Ead01]:

- The client system consists of a chat system, image system, and a CGI based system.
- The chat system provides information on all the participants and actual messages between them.
- The server was set up to test the consistency and feasibility of the system. The system's independence was checked using different platform machines such as Microsoft windows, Unix/x-windows, and Linux.
- It performed well in all kinds of operating systems but the performance depended on the network bandwidth. For future work, a video-capturing function could be added to future work to provide better presence of awareness.
- The image system consists of an image selector and drawing tools. The image selector downloads medical images from a server's database and displays a set of images. When a user selects one image it creates a new image object with drawing tools to handle the image and network connection to the server.

These results are interesting as a reference for this thesis because the bandwidth requirements will be simulated and new modulation methods will be evaluated. The goal of investigations is to develop adaptive methods for use in data transmission over physical band-limited channels.

### *Patient-Centered Access to Secure Systems Online*

At the time the project began, several prototypes existed, Web-based clinical data systems, which were explicitly designed to serve only health professionals, and, the most used security 'firewalls" to filter queries originating from outside the organization's private network times. The Patient-Centered Access to Secure Systems Online (PCASSO) project was designed to apply state-of-the-art security to the communication of clinical data over the Internet. The project has completed its initial field test and is not open at this time for new participant enrolments. Several references are made [Mas97, Mas98, Mas02].

The reported conclusion was: PCASSO applies state-of-the-art security technologies to the goal of extending the current World-Wide Web so that it may be used by healthcare providers and their patients to view person-identifiable health data. The project tests both the technology of security and the sociology of healthcare in an era where patients are given online access to their own medical data.

## 2.2. Standards

The main goals are to ensure a real-time, deterministic and secure data exchange between the linked devices, and to present a user-friendly visualization of the patient' s state. Storage and communication standards have been discussed since the 1980s (PACS, DICOM, JIRA, MIPS, ACR/NEMA etc). Technical standards relating to image format and communication protocols, as well as image processing including digital compression, have been the subject of discussions. Issues of safety and security within these digital networks are emerging to be critical aspects of medical multi-media data networks. Safety is usually defined in terms of hazards and risks as a systems engineering concept. In certain medical environmental conditions, a hazard is a set of conditions that can cause harm to a patient or other person. Safety is related to the reliable functioning of hardware and software. Security deals with issues of protecting the system against:

- unauthorized access to multi-media medical records and
- malicious or accidental corruption of data.

OSI Security Architecture defines a number of mechanisms for authentication, access control, data confidelity, data integrity, and non-repuditation. No standard mechanism exists or is likely to be available in the near future, which guarantees perfect safety of medical information systems and specially of IMAC systems [Ima91] p.183-185.

### *ACR-NEMA Standards*

The first U. S. PACS meeting in Kansas City in 1982 acquired image and image descriptive data. Different digitized video (often a loss of dynamic range) or magnetic tapes (different tape formats and pixel packing technique) were used in the manufacturer's systems. First a committee was formed in 1983 by two bodies, the American College of Radiology (ACR) and the National Electrical Manufacturers Association (NEMA). In 1985 the ACR-NEMA Standard was developed as an interface standard for the interconnection of two pieces of imaging equipment. The standard builds upon the OSI-ISO Reference Model but is not ISO compatible. Differences are in definitions of layer-to-layer connections (interfaces in ISO terminology), reference [Ima91] pp. 235-248.

### *Work items in ISO/TC 215 Work Group WG4*

The working group experts identified the following areas as possible candidates for new work items:

1. Definition of security terminology to be used in healthcare and in ISO/TC 215 standards in particular.
2. Guideline to existing standards and to point out where the documents can be found and when they may be applicable.
3. A framework document to show the scope of security standards for the healthcare field.
4. A standard for secure messaging (encapsulated objects with encryption for confidentiality and digital signatures to provide proof of origin and integrity).
5. A standard for secure channeling, particularly for web applications.
6. Discussions were held concerning the selection of encryption algorithms. "Consensus was almost reached that we should not make any restriction to the selection.... to draft a document describing the business model and a proposed policy on encryption... possibly suggest a work item on healthcare professional cards for authentication".
7. The need for standards on a supporting infrastructure for cryptographic techniques was further identified but not considered to be in the top priority for immediate action.

### *Medical Instrumentation Bus of IEEE*

Since 1992 IEEE had a draft proposal of the Medical Instrumentation Bus (MIB). The final standard has not yet been approved. Bridge MIB modules approved 802.1 technologies. IETF was in progress in 2004. IETF Bridge WG Transition is to IEEE 802.1 WG. The schedule is presented in reference [Har05]:

- November 2004: RFC1493 update, RFC2674 update, and RSTP-MIB.
- In 2005: Bridge WG documents to proposed standard and MIB module in some 802.1xx.

### *802.3 Ethernet*

Due to its economical price lots of hospitals and medical centers used and use the commercial 802.3 Ethernet to interconnect their departments, [Var03].

## *2.3. Conclusions*

There are many PACS, RIS and HIS in the market but they are seldom connected to each other and integrated. The reason has been lack of information exchange standards. Lack of standards for communication between health care applications is still one of the primary reasons holding up the wider use of information technology in health care. Movement of the standards towards alternate lower layer connections, TCP/IP protocol (de facto standard) and ISO-OSI stacks will contribute to the use of standards for network connection of devices [Ima91].

Today medical communication standards exist only for the higher level of medical care, like various databases and hospital information systems, and not for low-level communication between various diagnostic devices, see references [Sen95], [Var03] and [Rat05].

Transfer speed between network components and the problem of lost images are disturbing effects in PACS implementation.

# Chapter III

# 3. Analysis and Measurements

Security of biomedical or telemedicine data transmission on the OSI level, i.e. the physical level, is not available as a standard [Har05]. Thus the basic investigation problem is to find solutions and proposals to this problem formulation. Firstly the quality of data transmission over the present voice communication channels was investigated using measurements. Present networks are based on digital hierarchy and digital channels. Analysis and data transmission methods over these channels are discussed in the original papers written in 1997-2004. The papers are listed at the end of this thesis. Earlier networks were analog networks and used FDM. These networks and data transmission on their analog channels are discussed in the original papers [Lal75] and [Lal87].

## *3.1. Definition of Measurement Objects*

Based on the different standards in military and public telecommunication networks the attenuation distribution requirements are different, Figures 3.1-3.2. In Figure 3.1 one can find that delta modulation using 16 kbps bit rates have no requirements for frequencies higher than 2600 Hz.

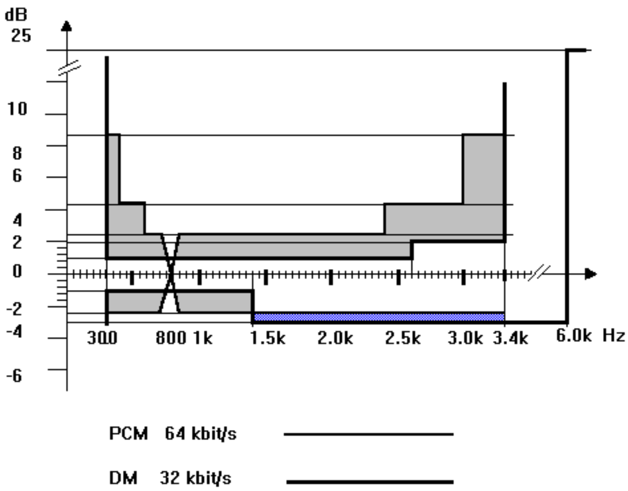


Fig. 3.1 Attenuation distortion requirements of 32 kbps ADM and PCM [Eur86], [Itu89]

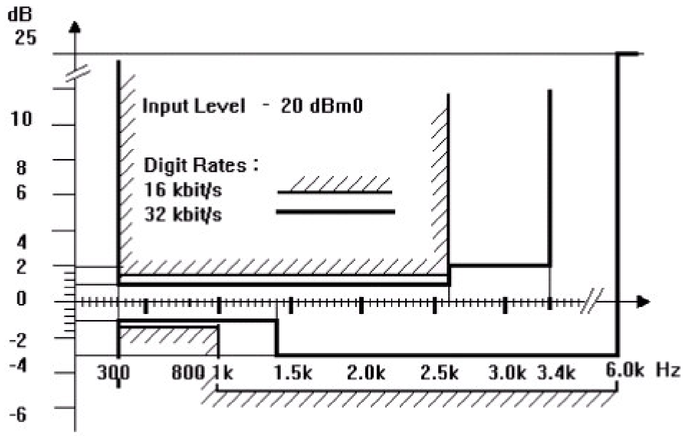


Fig. 3.2 Attenuation distortion requirements of 16 and 32 kbps ADM [Eur86]

On the other hand the 32 kbps delta modulation is quite near to the ITU-T speech channel, [Eur86, Itu89]. The following problems with analog data transmission using standard ITU-T modems over delta-modulated channels have been found in networks:

- Modern high-speed modems (telefax or 9600 bps) do not work.
- Low speed (1200-2400 bps) FSK V.23 and PSKV.26 modems have high BER.
- Packet switching does not work properly (long message delays).

There were problems in finding a proper standard modem or modems capable for use in 16 kbps networks defined in [Eur86]. The voice grade modems are optimized for use in the analogue speech channel of the public telecommunication network. The results of these measurements are discussed in the next sections.

## 3.2. Measurements

The measurements include ADM (adaptive delta-modulated) channel investigations and tests for analog data transmission over the ADM-channel. Data transmission performance in a delta modulated voice channel is limited by the channel performance. Thus the effect of several limiting factors and physical parameters are investigated in the measurements:

- Attenuation and attenuation distortion versus frequency.
- Phase characteristics and distortion.
- Total harmonic distortion (THD).
- Input level of the channel.
- Signal-to-Noise (S/N) ratio.
- Bit rate of delta modulation.
- Bandwidth of the channel.

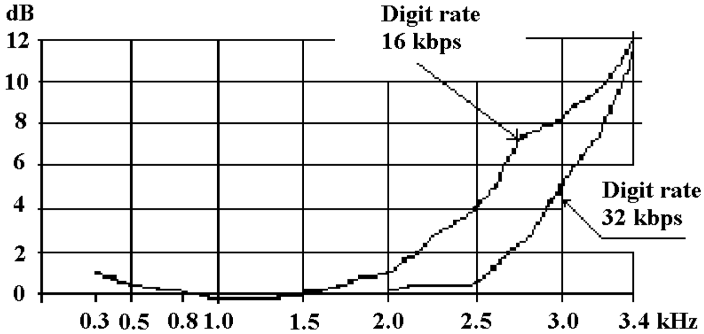


Fig. 3.3 Measured attenuation distortion of ADM-channel [Lal97a]

Measurements were made during 1990-1992 in The Signal School in Finland to find out some basic characteristics of the adaptive delta modulated speech channel of the 16 kbps network. The results are shown in Figure 3.3 [Mdd92]. The main channel and data transmission measurements were continued in the Signal Schools, Finland from 1991-1994 with a HP3567A measuring instrument made by the Hewlett-Packard Company, a data communication analyzer 2871 of Marconi and a Datatest 3 of Navtel Canada Inc. The HP3567A instrument has a built-in Fast Fourier Transform (FFT) capability. The analyzer 2871 and the Datatest 3 perform a bit error rate analysis.

### *Bandwidth for Voice and Data Transmission*

The 16 kbps ADM-channel voice bandwidth is about 2600 Hz and is thus not equivalent to the ITU-T requirements [Itu88]. The equivalent measured bandwidth of the 16 kbps and 32 kbps ADM systems is presented in Figure 3.3. The 32 kbps bit rate makes the ADM-channel more compatible with the ITU-T attenuation distortion requirements than 16 kbps.

The measurement results in Figure 3.4 show that the attenuation distortion of a 16 kbps ADM-channel is not in the limits of ITU-T requirements for analog data transmission. The result was expected based on the attenuation distortion requirements of the ADM-channel defined in the reference [Eur86].

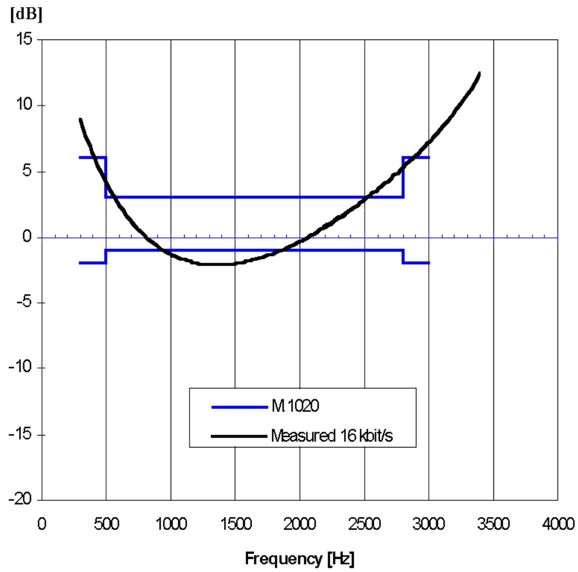


Fig. 3.4 Measured attenuation distortion versus ITU-T M 1020 requirements [Lal97a]

## *Message Intelligibility in DM and PCM*

For voice communications, the message intelligibility is the performance measure of greatest interest to the users [Sha79]. Then the performance of DM is better than the performance of PCM while human communication is possible with lower S/N-ratio and bit error rate. Experiments have shown that an error probability $P_e$ of the order $10^{-1}$ does not affect the intelligibility of voice signals in DM, whereas $P_e$ as low as $10^{-4}$ can cause serious errors leading to threshold in PCM [Sha79]. In PCM the weight of the detection error depends on the digit location $2^7$, $2^6$, …$2^0$. Errors in the detection of the first digits have a greater effect on the output signal waveform than errors in the later digits.

It was found in the analysis made at the Signals School in 1990-1992 that conversation in delta modulated channels was possible to the point where the channel synchronization was lost at bit error rate $>10^{-3}$ at S/N=9 dB.

In data communications the phase detection is difficult in delta modulation systems because of many reasons:

- Symbol timing in asynchronous systems.
- Slope overload in delta modulation.
- Granular distortion.

## *ADM versus PCM*

Comparing adaptive delta modulation (ADM) with pulse code modulation (PCM) one could find that the latter also follows the phase of a signal in a better way. This is based on the 8-bit code of the PCM, which adapts faster to the input signal than a one-bit code of ADM. In PCM one knows quite well the zero points of the signal which correspond to the phase = 0. In ADM the zero points are not as well defined because of the slope overload, continuous variable step size and the operative delay of the leaking integrator used in the delta coder.

## 3.3. Measurement Results and Analysis

The measurements of data transmission were made over granular noise channels and later in field tests multi-path channels as reported in papers [Lal97a], [Lal97b], and [Lal02b]. Several problems were found in analog data transmission in granular networks based on 16 kbps delta modulation (DM) voice coding:

- Modern ITU-T high-speed modems or telefax do not work there.
- Even low speed (1200-2400 bit/s) FSK and PSK ITU-T modems (V.23 and V.26) have high bit error rates.
- Delay times grow too high in packet switching.

The measurement results include

1) Magnitude and phase response of the granular channel.
2) Distortion characteristics of the ADM-channel.
3) Data transmission quality measurements using standard modem waveforms.
4) Analysis of block error probability in packet switching.

Figure 3.3 presents the measured attenuation distortion of a digital channel (granular ADM-channel). Attenuation depends heavily on digit rates. The attenuation at a digit rate of 16 kbps is not in the frames of the ITU-T recommendations for data transmission. Data transmission measurements over these channels are described in the paper [Lal97a]. Table 3.1 presents the considered standard modem types, which were tested. V.23 and V.26 modems were the only acceptable models. The bold spectrums in Table 1.1.correspond to the measured 16 kbps granular noise channel bandwidth shown in the Figure 3.3.

```
!RX:READY        TX:ON              23-NOV-93   14:41:13!
!                                                       !
!CONDITION   :       INTERFACE        BIT RATE   PATTERN !
!TRANSMITTER: RS-232-C     SYNC        1.200     FOX MSG !
!RECEIVER    : RS-232-C     SYNC        1.200     FOX MSG !
!                                                       !
! TEST PATTERN MEASUREMENTS            INDICATORS        !
!                                                       !
!TOTAL BITS   :     103351504   TEST DURATION: 23:59     !
!                               TEST MODE     : FOX      !
!TOTAL ERRORS:          124715                          !
!                                                       !
!ERROR RATIO :    1 IN 10^ 3   DATA SIGNAL   :           !
!                                                       !
!                                                       !
!                               SYNC GAIN: 23 14:31:57  !
!                               SYNC LOST: 23 14:31:55  !
!                                                       !
!                               STARTED   : 22 14:41:28 !
!                               STOPPED   : 23 14:40:29 !
```

Fig. 3.5 Measured bit error rate of V.26 modem data transmission
A 24-hour test

However, the data transmission test shows poor results, Figure 3.5. Modeling and simulation methods are used to find better analysis and solutions in the next chapter. Figure 3.6 presents the result using the ADM (adaptive delta modulation) as a granular channel. It is found the interesting capability of ADM to maintain the bit error rate level constant. The reason was expected and obvious due to the adaptation of the step size according to the signal level. The bit error (BER) tests of two modem types are presented in Figure 3.6. BER is in the range $10^{-4}$…$10^{-3}$. The recommendation of the digital 16 kbps networks [Eur86] describes only digital data transmission classes, which include error correction methods and faster bit rates (9.6 – 16 kbps).

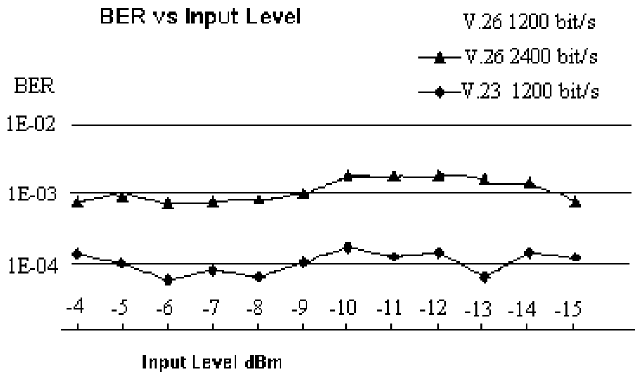


Fig. 3.6 Measured bit error rates of data transmission [Lal97a]

## *Packet Switching Problems in Internet*

Figure 3.7 illustrates BLER (block error rate) results with FSK waveform at low S/N ratios, paper [Lal97a]. The retransmissions show that packet switching does not work properly in a low S/N environment.

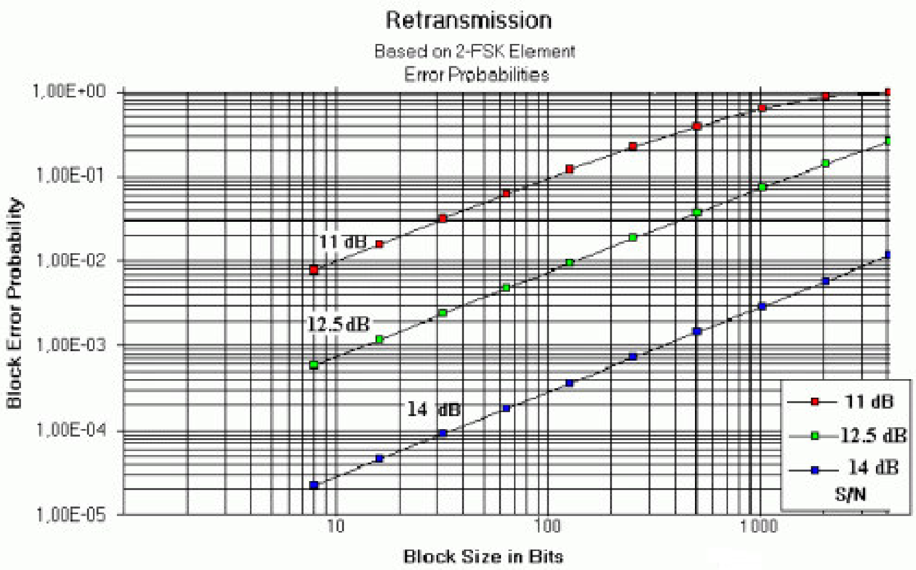


Fig. 3.7 BLER versus packet size

.

## *3.3.1. Discussion of Measurement Results*

The result based on the measurements is a recommendation to use FSK-modems in delta modulated 16 kbps networks for analog data communication. The standard 4-DPSK-modem is omitted from the recommendation because of its poor BER quality. Standard modems based on phase differences were not suitable in 16 kbps networks. In fact modems (>1200 bps) standardized by ITU-T using discontinuous phase are problematic in the present delta modulated voice channels. This result was found after many tests with carefully selected input levels according to EUROCOM standards and modem specifications.

In most references of delta modulation the slope overload is discussed. The slope overload is the error of amplitude and phase in a delta modulator output when the output signal does not follow the input signal. It will be simulated in the next chapter.

In data transmission a PSK modem has rapid phase shifts. A phase shift may have a rapid increase in amplitude, which a delta modulation system is not able to follow. An 8-bit code PCM-system has timing pulses and can follow amplitude and phase shifts (adapt to the signal) better than a one-bit code DM-system. The result is expected because of time varying slope overload in ADM. Figure 3.6 shows clearly that BER using standard modems in the analog data transmission over ADM voice channels is in the range 10-3…10-4. BER using the recommended digital data transmission method with error correction is about 100 times better according to reference [Eur86].

### *Loss of Synchronization*

The threshold for loss of synchronization (SL) was 1 in 16, using the data communication analyzer 2871 of Marconi Instruments Ltd. In the Figure 3.5 made by this analyzer the loss of synchronization is seen at 14:31:55 and sync gain at 14:31:57. Synchronization was lost in this case for two seconds. It is a serious problem in data transmission over the ADM-channel but not a big problem in voice communications. Later the loss of synchronization (SL) was again monitored now with Navtel Datatest 3. The mean period between the loss of synchronization was measured with 22 different ten-minute tests. The results were:

- FSK 1200 bps: 1.197 seconds.
- 4-DPSK 2400 bps: 0.985 seconds.

Measurements in another situation with another transmission line gave different results than in 1993 when the results were about 0.5 seconds for both modem types. Thus these measurements are unreliable and only indicative. In general to avoid using confusing results a simulation investigation will be made in the next chapter.

### *Real Data Rate*

Using ADM-channels with analog data modems the loss of synchronism happens very often. This result was seen as a decrease of the real data transmission speed of FSK to 587 bps and 4-DPSK to 2044 bps. This is an illustrative and suggestive measurement made in 1994. Practical situation are varying thus to get repeatable, reliable and more general results one will use simulation methods rather than measurements.

## *3.3.2 Analysis of Packet Switching*

An analysis of packet switching in a network is made using a FSK waveform based on the recommended measurement results. Figure 3.8 illustrates that packet switching does not work properly in a bad SNR environment as the probability of retransmission increases with the block size. The situation found and known in practice and is demonstrated here. In the example of Figure 3.8 the retransmission phenomena as block error rate (BLER) versus packet sizes is calculated for the FSK waveform at different low SNR ratios.

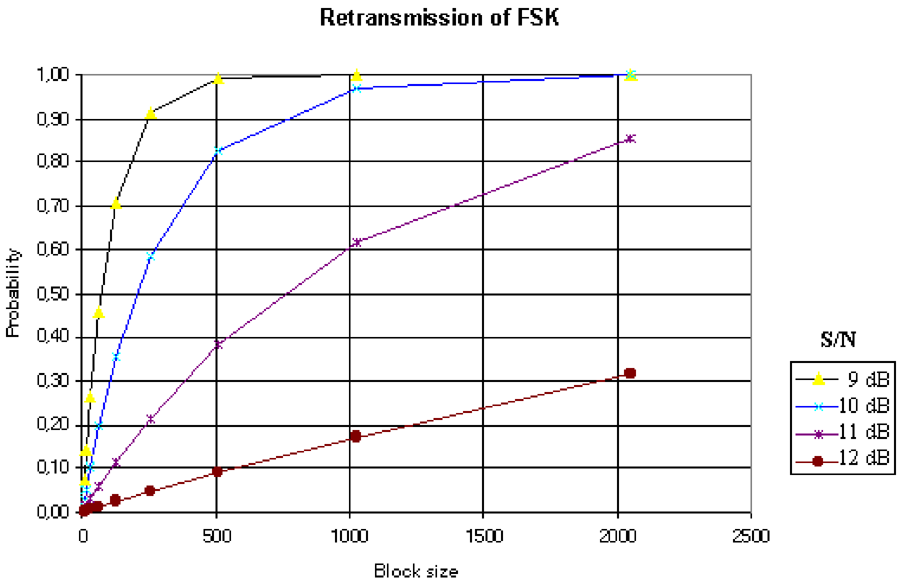


Fig. 3.8 Probability of a FSK retransmission versus packet size [Lal97a]

Formulae (3.1) – (3.3) were used in the evaluation calculations of Figure 3.8. The BER values correspond with the S/N = 9…12 dB of FSK-waveform. In local area networks (LAN) the problem is often message delays. A data network modeled with a network level simulator (Comnet 3) is presented in Figure 3.9.

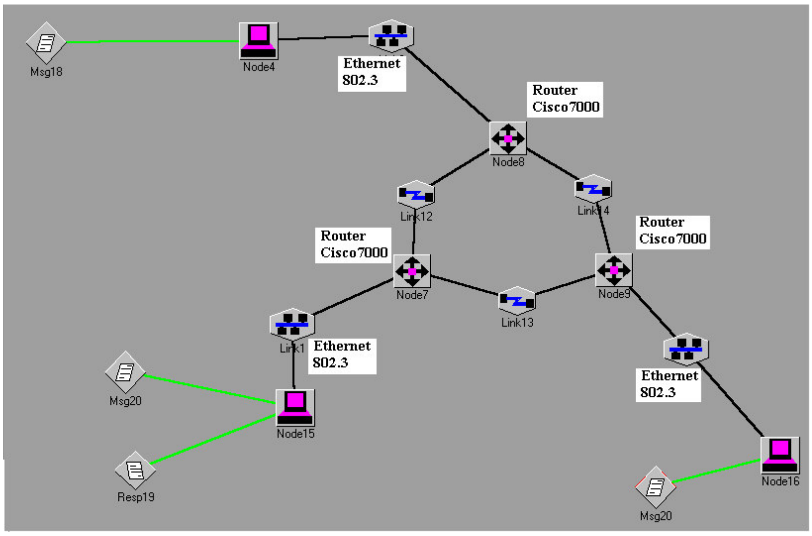


Fig. 3.9 Example: network model (CACI Comnet 3) [Lal04a]

### *Message Delay versus Bit Rate*

The simulation results in Figures 3.10- 3.12 show mean and deviation values of message delays as the effect of different capacity links used in the backbone network. The traffic bursts are seen in three different cases (600 bps, 9600 bps and 22.5 kbps), in Figure 3.11, smoothing with the proper channel capacities, in the example case at 22.5 kbps of Figure 3.12. The planning of data channels and network structures are essential in minimizing traffic delays and bursts. The simulated results in this study motivate to look for better modulation methods to avoid retransmissions and traffic delays. Comnet 3 is a traffic simulator and an example of simulator packages, which are discussed in the simulation study in the next chapter.

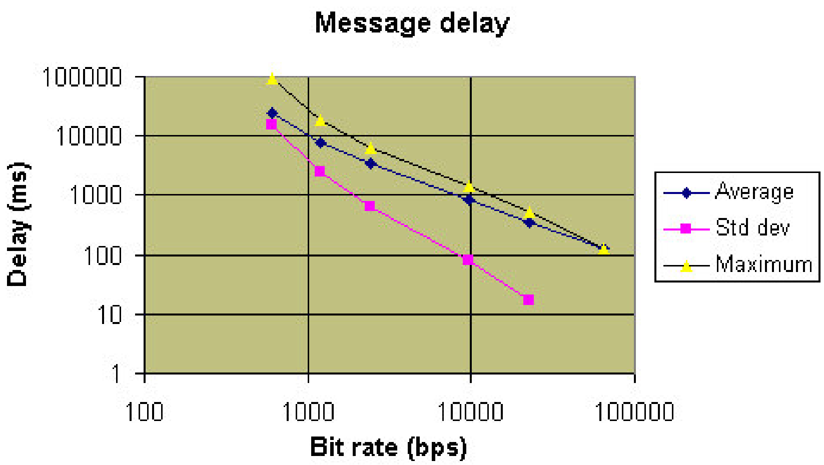


Fig. 3.10 Message delay versus channel bit rate [Lal04a]

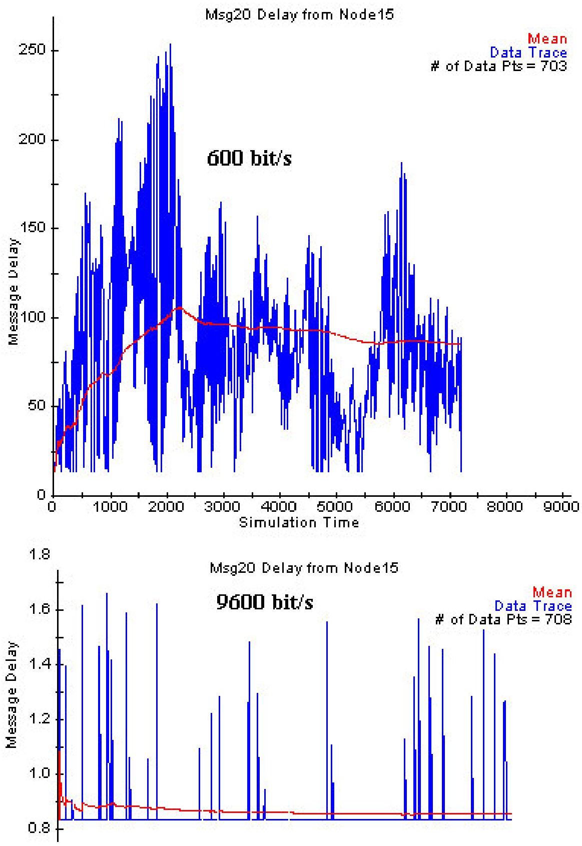


Fig. 3.11 Message delay bursts using different backbone networks [Lal04a]

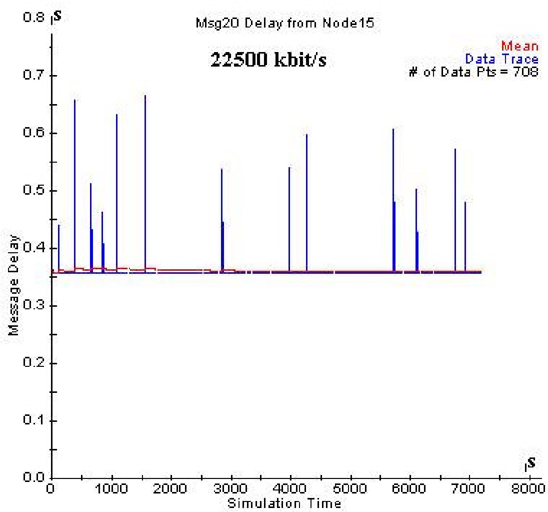


Fig. 3.12 Message delay bursts using a 22.5 kbps backbone network [Lal04a]

The block error probability may be used in the evaluation of packet transmission. The packet size especially has to be evaluated. Other parameters used in the evaluation of packet transmission are:
- Block size.
- Bit error rate.
- S/N.
- Block error probability (probability of retransmission).

In packet transmission some delay problems are eliminated in the proper selection of these parameters and network topology. Network topology is a large subject of other studies.

In reference [Com92] a formula (3.1) is used for the estimation of the probability $P$ of a correctly received message.

$$P = (1 - P_b)^N \qquad (3.1)$$

Where

$N$ = Number of bits in the received message
$P_b$ = Bit error rate. Bit errors are independent variables.
$P$ = Probability of a correctly received message.

The probability of retransmission $P_B$ is thus $1\text{-}P$, formula (3.2).

$$P_B = 1 - P \qquad (3.2)$$

Where $P_B$ = Block error rate or the probability of retransmission.

In reference [Car86] pp. 552-554 a comparison of digital modulation systems give a formula (3.3), which is used for the estimation of the probability Pe of an OOK or FSK modulation with envelope detection of a correctly received message.

$$P_e = \frac{1}{2} e^{\frac{-\gamma_b}{2}} \qquad (3.3)$$

Where $\gamma_b$ is the energy-to-noise ratio needed to get a specified error probability per bit $P_e$.

# Chapter IV

# 4. Data Transmission in Channels and Networks – A Simulation Study

## *4.1. Worksheet Simulator*

The use of a personal computer with programs already in use at the office was one goal in modeling and simulations in this study. Compilers of simulation languages are expensive and a simulation package is adapted to one type of problem only. The Excel worksheet program as a standard language was selected for rapid modeling and during this study the author programmed an Excel-simulator for simulations made in this study, presented in paper [Lal04b]. The reasons for this choice were mathematical, economical and practical. The simulator is based on the mathematics programmed in Excel cells forming a block model, Figure 4.1. The model blocks include mathematical entities. The simulations are executions of the programmed mathematical formulae in networked Excel cells [Lal97b] and [Lal04b]. The Excel worksheet program itself has all the mathematics and graphics needed. It is widely used and thus available for most PC-users. It is an effective way of programming and it has excellent graphics to present results. Most of the particular blocks and waveforms needed for this study were not available in the libraries of the reference [Com90]. Thus a new computer simulation method for evaluation of the characteristics of the ADM-channel and data transmission was needed and its first version MIL.xls was developed in November-December 1992, based on a standard worksheet program (Excel), Figure 4.1. The latest development of the robust worksheet simulation, 26 data channels (AWGN, granular, and multi-path) with an adaptive 1…160-point DFT calculation, is presented in reference [Lal04b].

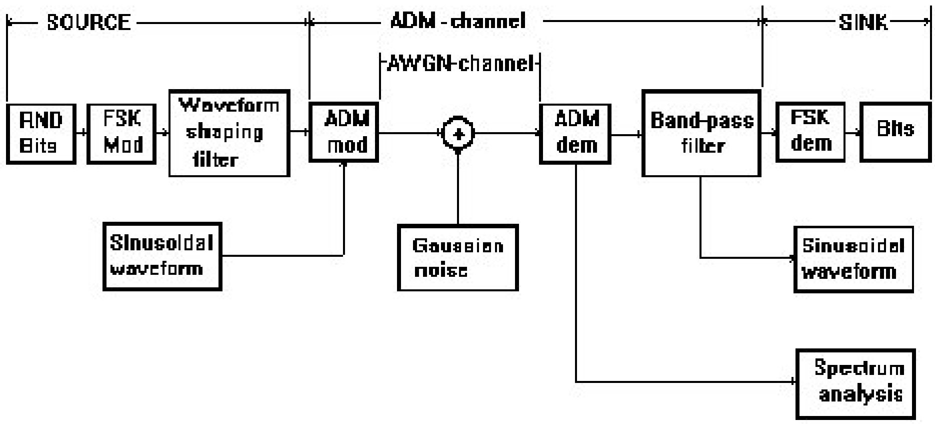


Fig. 4.1 Blocks of robust worksheet simulator

The ADM-channel model included an ADM-modulator, an AWGN-channel, and an ADM-demodulator. Adaptation simulations were made with this ADM-channel model using 2-bit, 3-bit or 4-bit memory in Modulation Level Analyzer (MLA). EUROCOM specifies the adaptation with 3-bit memory, which was used in simulations unless otherwise stated. The granular noise channel modeled is called here the ADM-channel model, Figure 4.1.

The data transmission simulation model used a random bit source, a waveform generator, the ADM-channel model, and a data modem receiver simulator using the Discrete Fourier Transform, which is called here the DFT-receiver. Most of the simulations were made with this simulator system, which is called here the Excel-simulator.

The present worksheet simulation package for modeling data transmission over different channels (incl. the adaptive delta modulated voice channel of the present 16 kbps network) includes generation of waveforms, a model of discrete Fourier transform receiver for waveform detection, random bit and symbol generation, calculation and estimation of simulated BER, setting of Gaussian noise level, setting of multi-paths, setting of interference signals, signal-to-noise ratio calculations, phase distortion calculations, and group delay calculations.

## *Limitation of Worksheet Program*

The limitations with a worksheet simulation are the memory size available in PC, PC throughput with Excel and Excel worksheet limitations. An Excel worksheet used in 1992 had 16384 rows and 256 columns. A minimum robust 1000-bit simulation used in this work needed about 6 MB memory to manipulate 13000 samples stored in Excel cells. This was the practical limit for the personal computers used earlier. These problems were minimized in ten years and 10000-bits simulation is not a practical limit. To get a high quality waveform in an Excel-simulator a sample rate about 10 times the highest signal frequency was used. The same limitation was also observed in other simulators [Tes92].

## *Use of Discrete Fourier Transform*

Two approaches were considered: Discrete Fourier Transform (DFT) and Fast Fourier Transform (FFT), definition in Appendx 1. DFT was selected instead of FFT for the calculation of the response of the ADM-channel and for the decision of the bits from the output waveforms of different data modems. The main reasons for this are:

- The use of DFT makes the approach adaptive. The number of samples (N) was freely selected.
- DFT is easy to program.
- DFT gives both amplitude and phase of a given signal.
- The number of multiplications of DFT is limited in calculations using N=13 or 26.
- The number of samples in FFT must be in powers of two. Thus the sample numbers used in this study were not optimal for FFT.
- DFT works in the simulator with any limited number of samples.

### *Computational Limitations of DFT and FFT*

To calculate one magnitude point of frequency response the first version of MIL.xls simulator (1993) made more than 20000 calculations. It took about 30 seconds while N=160 samples were used in a direct computation. In reference [Pro92] the computational complexity for the direct computation of the DFT is compared to the FFT algorithm. The number of multiplications needed in DFT ($N^2$) is much larger than in FFT $(N/2)\log_2(N)$, see Table 4.1. However the rapid development of processor power and RAM memories have made the time delay in simulations with DFT negligible. The FFT-values for N=13, N=26 and N=160 are "not powers of two" and thus not possible with FFT (N/A).

Table 4.1. Complexity of DFT versus FFT

| Number of points N | Multiplications DFT | Multiplications FFT |
|---|---|---|
| 13 | 169 | N/A |
| 26 | 676 | N/A |
| 128 | 16384 | 448 |
| 160 | 25600 | N/A |
| 256 | 65536 | 1024 |

## *4.2 Modeling of Data Transmission*

Simplified models are used in the simulation of data transmission over the ADM-channel (analog voice grade channel, ADM coding), Figures 4.1-4.4. A detailed presentation of the data transmission is found in reference [Ska01] and Appendix 2. In the simplified model random of Figure 4.2 digital data bits were generated (Bits IN) and analyzed (Bits out) in the PCs. The symbol waveforms or analog signals were generated and detected in the data modem (Data Mod, Data Dem). The data waveform was led to the multiplexer (Mux), which includes the equipment for the adaptive delta modulation coding of analog signals (A/D ADM, D/A ADM). Crypto equipment is needed in wireless communications but was not modeled for the simulations. The air interface is a radio link or a base station (Mod, Dem), which are modeled with an AWGN or a multi-path channel model. Noise was added to the signal in the receiver (Dem).

In Figure 4.3 the channel is a radio channel or a wired channel. DM and PCM are source encoding and decoding methods used for a base-band digital signal. Discrete channel encoders and decoders for base-band line signaling are not modeled in simulations. The analog radio or wired channel is robustly modeled with an AWGN, multi-path, and granular channel model.

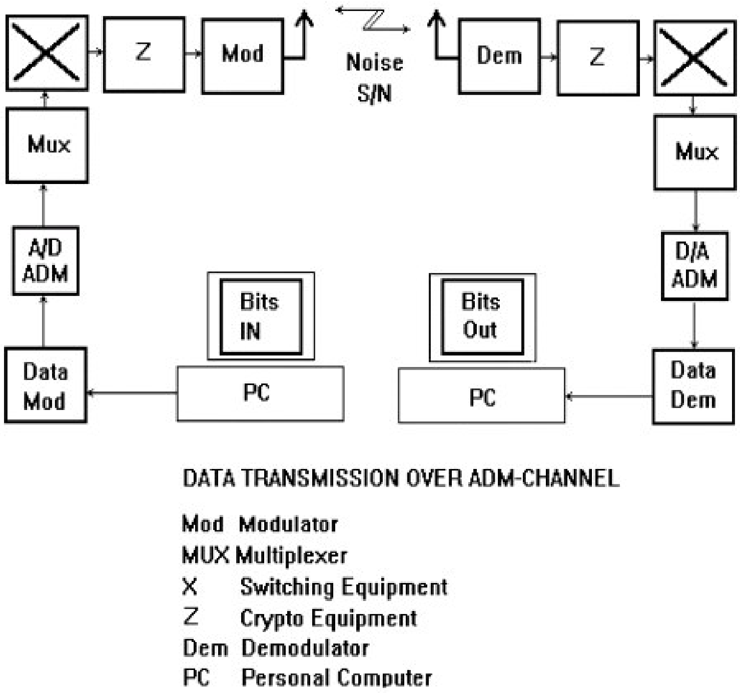


Fig. 4.2 Simplified model of analog data transmission

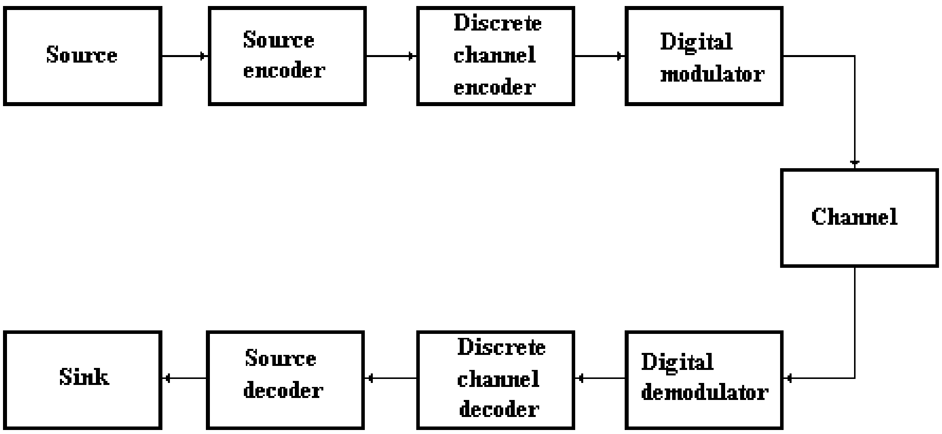


Fig. 4.3 Simplified simulation models of data transmission [Lal97a]

## *Problems and Research Methods*

The measurements demonstrated the quality of standard analog data transmission with 1200 bit/s or 2400 bit/s rate modems only using granular noise channels. The quality levels were acceptable or poor. This motivates to study other than standard data modulation methods for improvements of the data rate and transmission quality over granular (digital network), AWGN (theoretical) and multi-path (radio) channels. The investigation was made with a robust modeling and simulations method. The programming was made with a worksheet as discussed in papers [Lal97b], [Lal99], and [Lal04b]. The results were verified with measurements and reference simulators.

The information transmission chain is: digital data source or analog source waveform - granular noise in source coding – AWGN noise and multi-path channel - receiver was modeled. The simulation results show the probability of the correctly received message in different cases or BER (bit error rate). The information transmission blocks are analyzed and described in detail in papers [Lal97b], [Lal99], [Lal04a], and [Lal04b]. The three different channel models causing different problems in data transmission (quality impairment) are discussed in several references are available [Sha48, Cha66, Rum86]:

- AWGN.
- Granular noise.
- Multi-path interferences.

## *Discussion of the Results*

The simulation results, conclusions and proposals in the papers [Lal99], [Lal00], [Lal01], [Lal02], and [Lal04a-b] include:

- Simulation analysis of different granular noise channels (phase and amplitude distortion and a polynomial channel model)
- Comparison of standard data transmission methods with the developed adaptive multi-carrier data transmission methods.
- Qualitative results using AWGN, granular noise and multi-path channels for data transmission.
- Recommendation for selecting adaptive data transmission parameters and design principles for an adaptive modem.
- Results of simulations with a model for biomedical data network using adaptive versus standard data transmission at different bit rates.

A brief summary is presented next.

## 4.3. Adaptive Delta Modulation and Granular Channel

Figure 4.4.presents the adaptive delta modulator and demodulator of the Eurocom recommendation [Eur86] and the simulation model of the adaptive delta modulator, paper [Lal04b]. The ADM-channel (granular noise channel) discussed in this study is established between points C and C'. The demodulation process is simply the integrator and it includes the same modulation level analyzer as the modulator. The MLA (modulation level analyzer) and the first integrator in Figure 4.4 define the step size, which is propotional to the granular noise level, described in more detail in reference [Eur86].

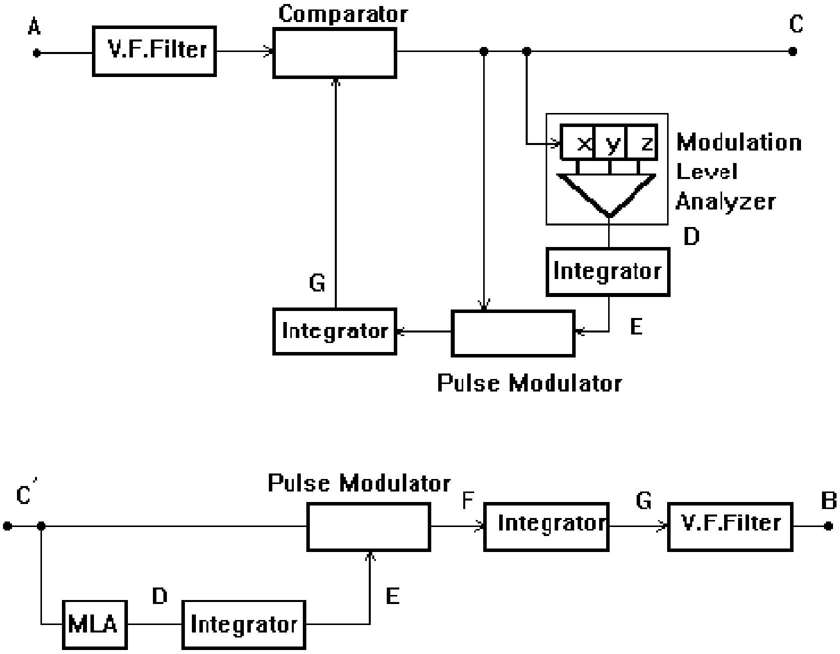


Fig. 4.4 ADM modulation demodulation process and blocks [Eur86]

Figure 4.4 shows the analogue/digital conversion of speech signals with a pulse modulator in the transmitter and digital/analog conversion in the receiver end of the digital transmission channel between (C-C´). The receiver has a leaky integrator (between F-G) and a VF (voice frequency) filter (between G-B).

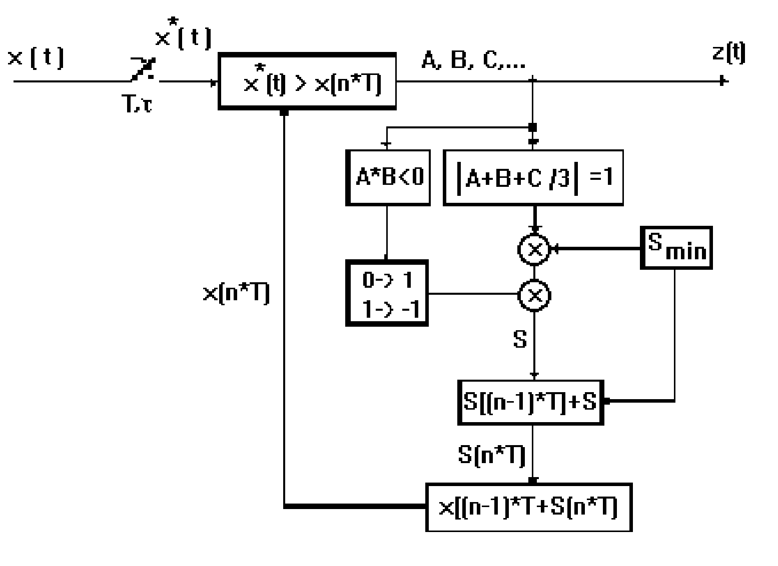


Fig. 4.5 Simulation model of ADM modulator [Lal04b]

In the simulation model of Figure 4.5 the modulation level analyzer is developed into different two, three or four bit versions and used for the ADM algorithm simulation process in paper [Lal97b]. In Figure 4.5 a three bit algorithm (A+B+C)/3 = 1 or –1 is used in the simulated result of the adaptive step size versus a continuous sinusoidal signal. The modulation simulation result, the adaptive discrete audio signal x(nT), is presented in Figure 4.6 with the original input signal 800 Hz.

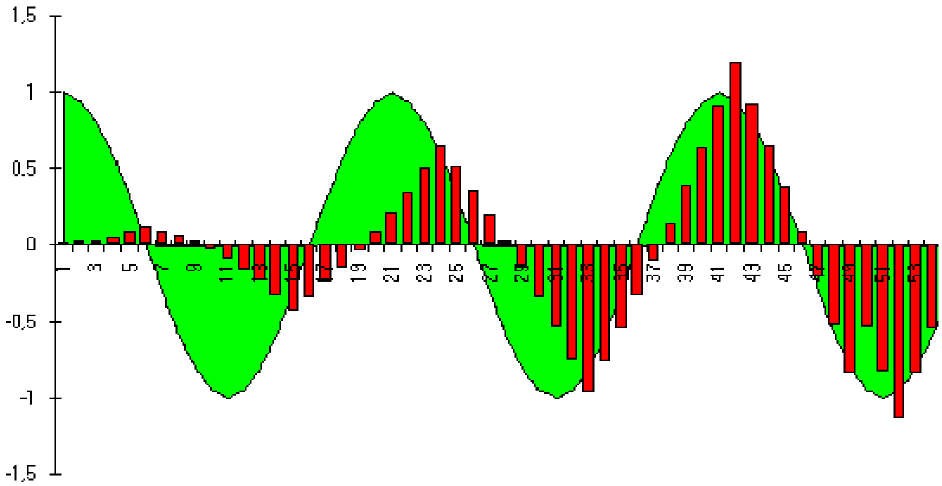


Fig. 4.6 ADM modulation of 800 Hz test signal [Lal97b]

The integration is controlled by the adaptive step size s(t), formula (4.1).

$$s(t) = \frac{1}{V_C}\sum_{i=0}^{n} S(t)\,y(iT) \tag{4.1}$$

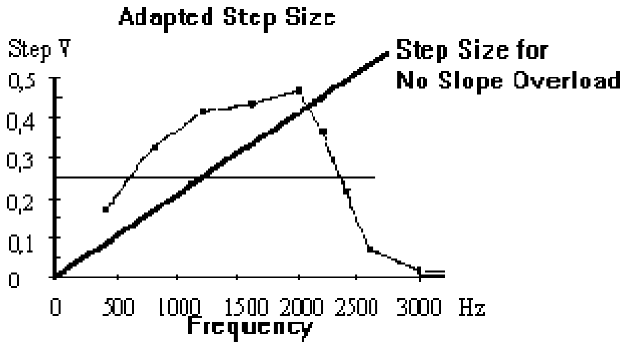


Fig. 4.7 Simulated adaptive step size [Lal97b]

Figure 4.7 presents the step adaptation at different frequencies in the adaptive delta modulation system. To avoid slope overload in a delta modulation system the step size S (line in the figure) must be greater than a minimum value.

## *Adaptive Digital Channel*

The performance of the delta-modulated voice channel is presented in the simulated results of Figures 4.8-4.10. The simulated magnitude and phase response functions for different amplitude to minimum step size ratios of the ADM-channel are seen.

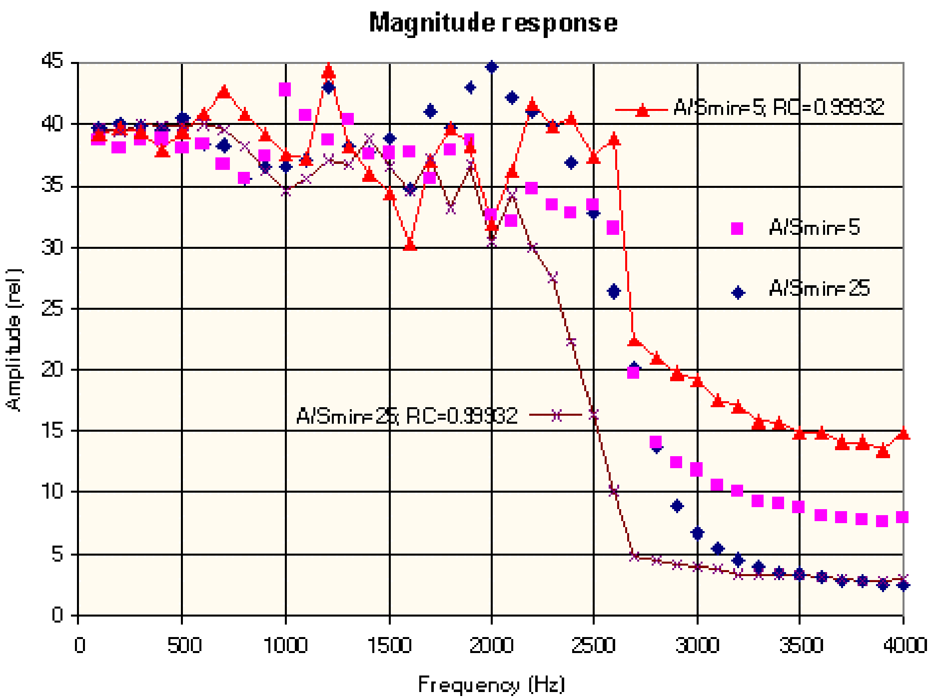


Fig. 4.8 Simulated magnitude response of ADM-channel [Lal97b]

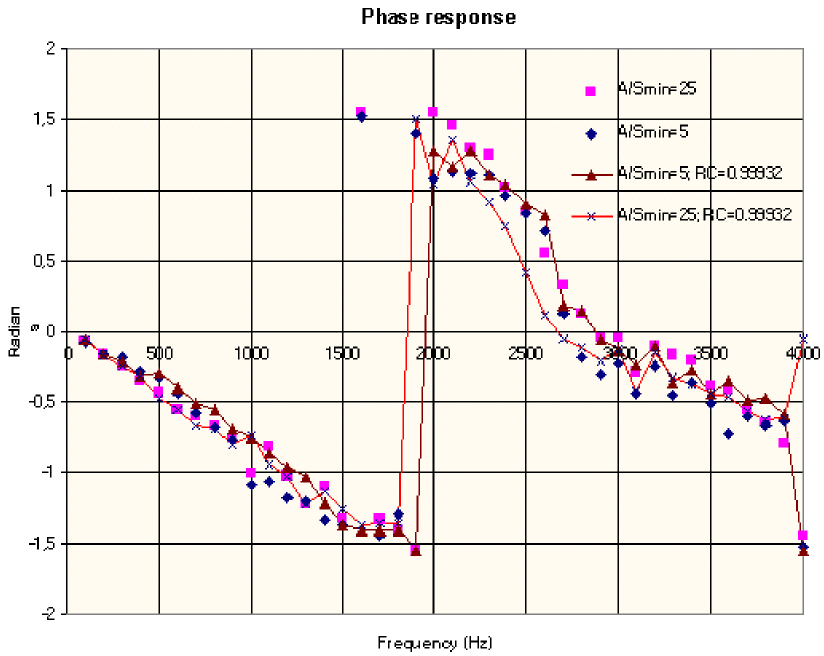


Fig. 4.9 Simulated phase response of ADM-channel [Lal97b]

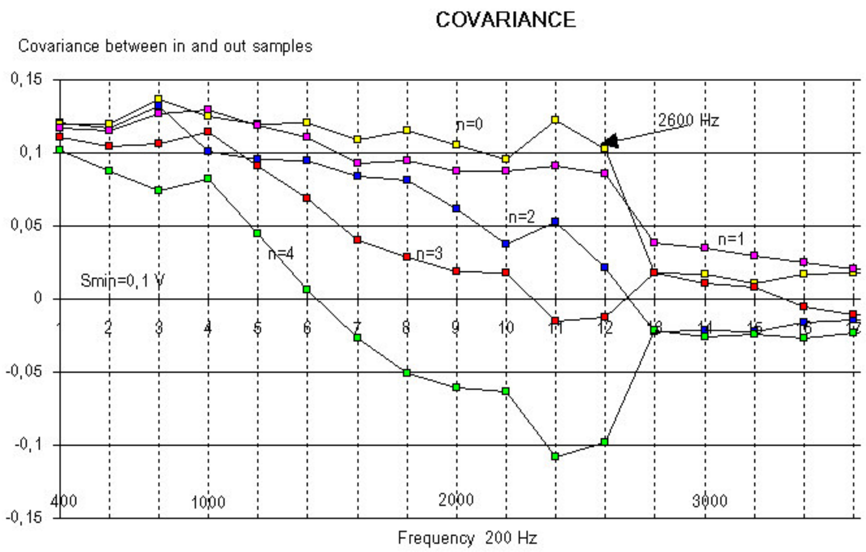


Fig. 4.10 Covariance between input and output samples

In Figure 4.10 the voice band signal has redundancy between delayed samples, which is seen in the calculated covariance results between the 1...4 - sample delayed 16 kbps signals.

## *Polynomial Channel Model*

The Polynomial Signal Processing (PSP) of the simulation result gives the channel model. The channel model polynomial in dB values is in formula (4.2) where the voltage y is normalized and the frequency x is given in kHz. The results are presented in Figure 4.11. The quality of the polynomial curve fitting was evaluated with the coefficient of determination r also called correlation coefficient. The best fitting r = 0.9474 for the amplitude response is achieved with the polynomial of degree 6 in Figure 4.11.

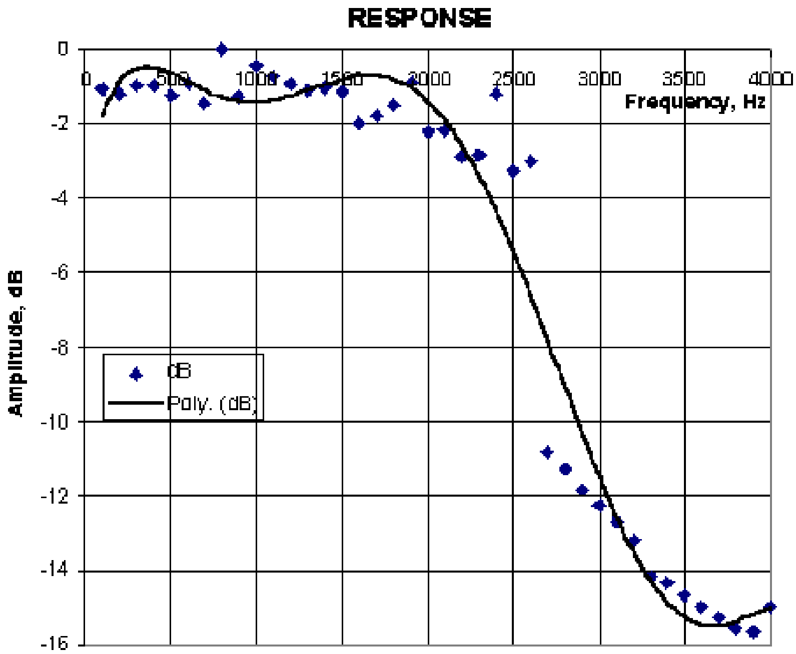


Fig. 4.11 ADM-channel amplitude response [Lal04b]

The polynomial amplitude response model [dB] of degree 6:

$$y = -0.0266x^6+0.2811x^5-1.2868x^4+2.6801x^3-2 .6318x^2+1.1013x+0.757; \quad (4.2)$$
$r = 0.9474$

The polynomial phase response model of degree 3:

$$y = -4E\text{-}9x^3+2E\text{-}5x^2-0.0682x+97.429; \quad (4.3)$$
$r = 0.9864$

A very good fitting for the phase response (r=0.9864) is achieved with the polynomial of degree 3 if the frequency range x is limited to 2600 Hz. The polynomial is presented in formula (4.3) where the phase y is in degrees and the frequency x is given in Hz.

### *Discussion*

The polynomial response is sensitive to additive Gaussian noise (AWGN), which was also studied. If noise is eliminated in simulations, the polynomial models of the ADM-channel are as presented in Table 4.2.

Table 4.2. Polynomial models for the ADM-channel amplitude response

| |
|---|
| y = -7E-18x6 + 4E-14x5 - 1E-10x4 + 1E-07x3 - 0,0001x2 + 0,0331x + 76,175<br>R2 = 0,7687 |
| y = -1E-14x5 + 6E-11x4 - 1E-07x3 + 7E-05x2 - 0,0206x + 80,933<br>R2 = 0,7654 |
| y = -2E-11x4 + 7E-08x3 - 0,0001x2 + 0,0546x + 72,384<br>R2 = 0,7454 |
| y = -1E-08x3 + 4E-05x2 - 0,0386x + 87,086<br>R2 = 0,6385 |
| y = -7E-06x2 + 0,0096x + 75,221<br>R2 = 0,5165 |
| y = -0,0086x + 83,745<br>R2 = 0,4054 |

Amplitude (relative amplitude vs frequency) and phase (phase in radians vs frequency) models are presented with polynomials in Figures 4.12 and 4.13.

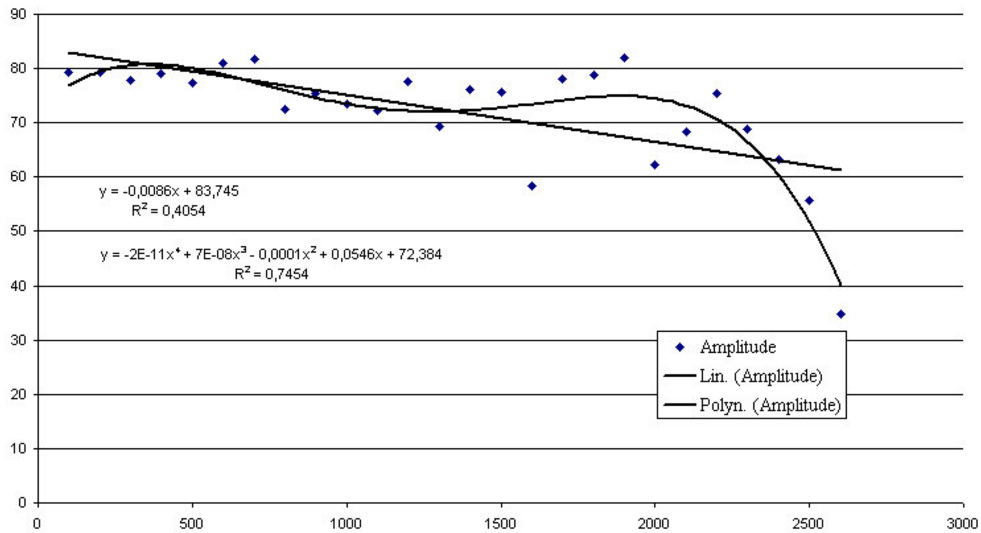


Fig. 4.12 ADM-channel amplitude response [V]

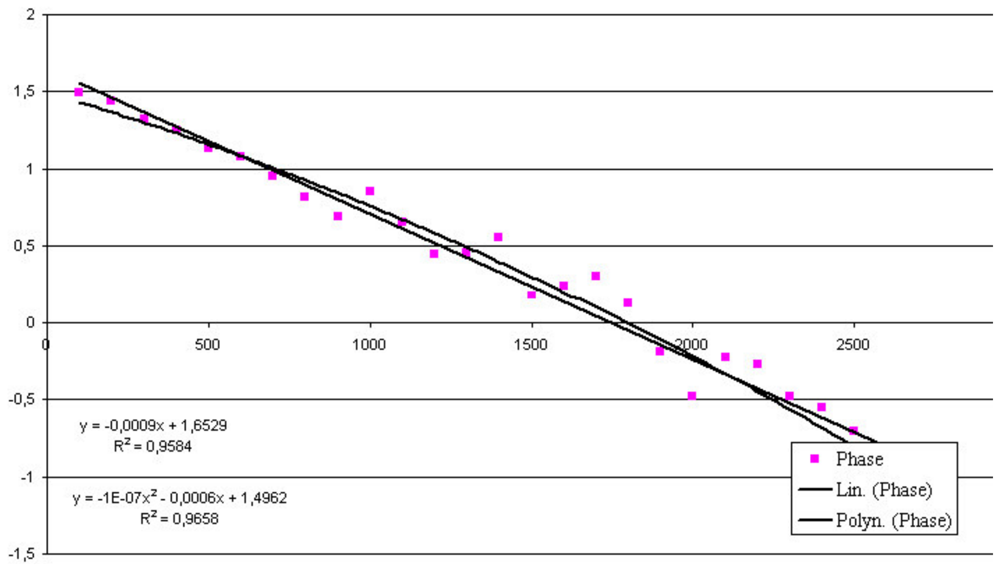


Fig. 4.13 ADM-channel phase response [rad]

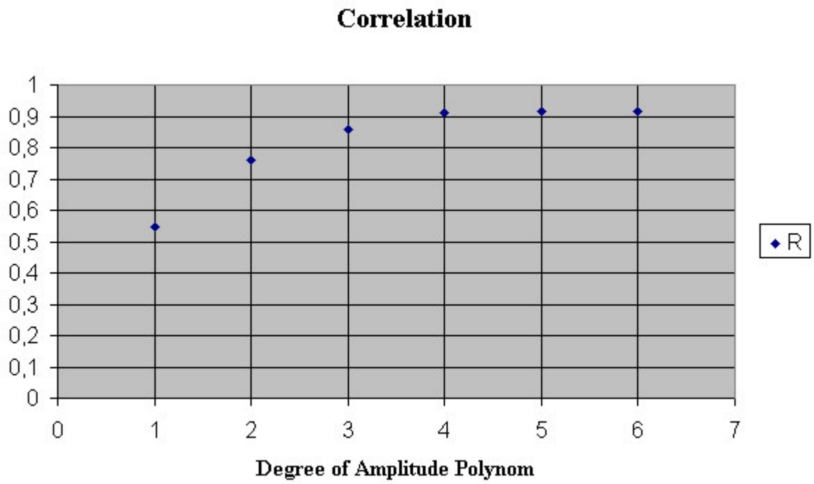


Fig. 4.14 Correlation of the ADM-channel amplitude model

Figure 4.14 shows the correlation (R) of linear and different polynomial (degree 2-6) models for the ADM-channel amplitude response. Third or fourth order models have better than 85% fitting. The 5-6 order models have the same level fitting as the fourth order model. Linear model is not a proper attenuation model for this ADM-channel (100-2600 Hz, 16 kbps sampling rate, 3-bit MLA). The amplitude response of the ADM-Channel is not linear. It has an attenuation distortion found in measurements of chapter 3 and regulated in recommendations [Eur86]. The attenuation distortion is qualitatively illustrated in simulations, Figures 4.12-4.13. Polynomial channel models are included in the figures. The phase is not linear as illustrated by the polynomial model in Figure 4.13 and Table 4.3.

Table 4.3. Polynomial models for the ADM-channel response of phase

y = -0,0009x + 1,6529
R2 = 0,9584

y = -1E-07x2 - 0,0006x + 1,4962
R2 = 0,9658

y = -2E-10x3 + 6E-07x2 - 0,0014x + 1,6802
R2 = 0,9716

y = -1E-13x4 + 5E-10x3 - 6E-07x2 - 0,0006x + 1,5623
R2 = 0,973

y = -2E-16x5 + 1E-12x4 - 3E-09x3 + 3E-06x2 - 0,0021x + 1,7356
R2 = 0,9746

y = -6E-19x6 + 5E-15x5 - 1E-11x4 + 2E-08x3 - 1E-05x2 + 0,0029x + 1,29
R2 = 0,9803

Table 4.3 shows that a linear model best describes the phase response of the ADM-Channel. The correlation coefficient of it is very high >95%. Thus the ADM-channel is almost phase linear.

### *4.3.1. Distortion of Signals in Granular Channel*

As stated earlier the ADM system causes granular distortion. Simulating with the ADM-channel voice-coding model, presented in Figure 4.5 and paper [Lal04b], the effect of using different sinusoidal frequencies (carrier frequency) is evaluated next.

#### *Distortion Components*

In Figure 4.15 one can find the simulated distortion components of the ADM-channel caused by a 2000 Hz sinusoidal input signal. The components are 20 dB below the input signal. These simulated results fulfill the requirements of the adaptive delta modulation method described in reference [Eur87].

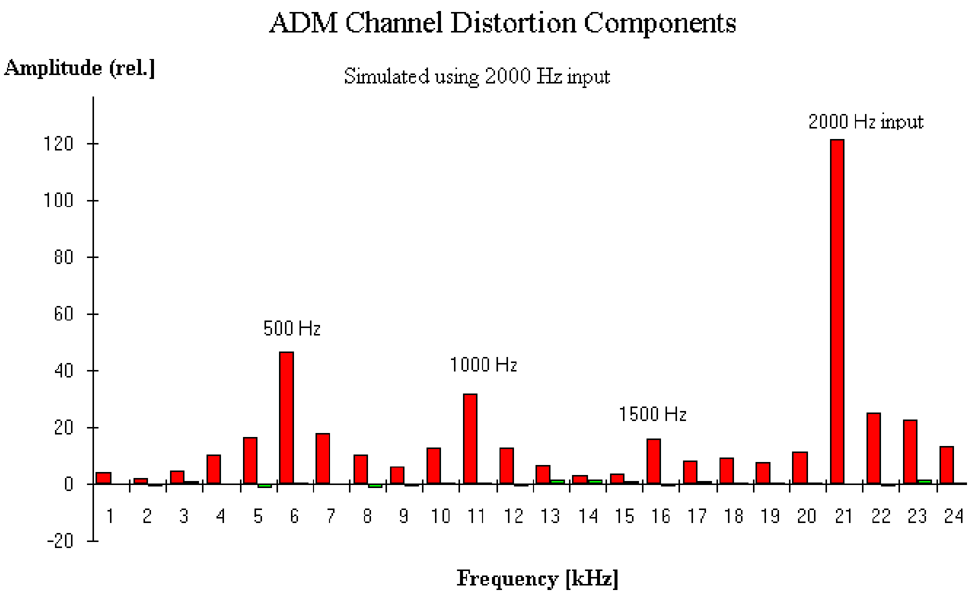


Fig. 4.15 Simulated distortion components of a 2000 Hz input sinusoidal signal

### *Distortion of a Granular Noise Channel*

A digital telecommunication network, in this case an ADM (adaptive delta modulation) system, causes granular distortion. The effects increase with the frequency as simulated results in Figure 4.16 present. The result is in accordance with the measured values. The measured value fits with the simulation results in the $A/S_{min}$ range 12.5…50.

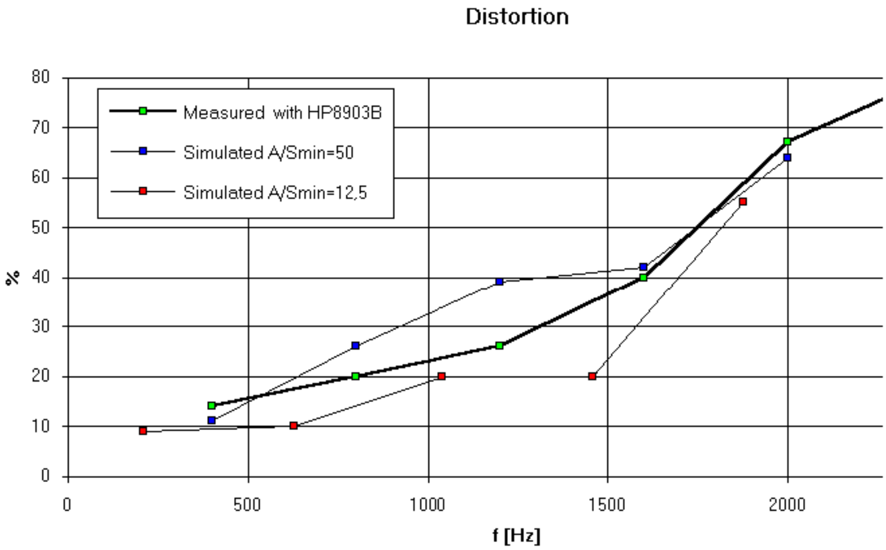


Fig. 4.16 Distortion of the ADM-channel [Lal97b] and [Lal04b]

## 4.3.2. Simulation Results of Analog Data Transmission

In this section data transmission with some traditional and new waveforms are studied with n-point DFT–algorithm (n=13, 26…N) software detection using simulations. The results are discussed and compared with measurement results of standard modems.

### Analog data transmission over a granular channel - simulated results

The effect of frequency is seen in the results of the simulation of the 8-PSK data transmission over the granular ADM-channel, Table 4.4 [Lal97b].

Table 4.4. Simulated phase jitter of ADM-channel

| Frequency Hz | Deviation of phase jitter | Maximal phase jitter | BER of 8-PSK |
|---|---|---|---|
| 615 | 4.29 | 18.4 | $<10^{-4}$ |
| 1231 | 7.41 | 31.8 | $<10^{-4}$ |
| 1846 | 12.39 | 55.5 | $<10^{-2}$ |
| 2462 | 15.10 | 73.8 | $>10^{-2}$ |

The adaptive delta modulation effects are seen here in phase jitter (in degrees) and in resulting bit error rate (BER). This result suggests an improvement of data throughput in the ADM-channel by the selection of a lower carrier frequency and a lower symbol rate.

Table 4.5 presents simulation results and a comparison between the developed waveforms (detection with a 26 sample DFT software algorithm) and the measured standard V.23 and V.26 modem waveforms. In this case the corresponding bit rates are 2400 bit/s and modems FSK&8-PSK with DFT26 detection and V.26 standard. The bit error is three times larger with the V.26 standard waveform compared to the developed soft detection with a DFT26-algorithm. Details are presented in the papers [Lal97a] and [Lal97b]. The papers present the adaptive waveform generation and selection principles.

Table 4.5. Comparison of standard modems and DFT26-algorithm

| Modem BER | Modulation bauds, bit/s | Carrier frequency, Hz |
|---|---|---|
| V.23<br>1E-4 | FSK<br>1200, 1200 | 1300,<br>2100 |
| V.26<br>5.7E-3 | 4DPSK<br>1200, 2400 | 1800 |
| DFT26<br><br>1.6...1.8E-3 | FSK & 8PSK<br>4FSK & 4PSK<br>600, 2400 | 615, 1231<br><br>1846, 2462 |

### *New Data Transmission Waveforms and Different Channels*

A combination of 2FSK and 8PSK or 4FSK and 4PSK is more reliable and acceptable than standard 4DPSK for the simulated granular channel based on lower and more suitable carrier frequencies. The use of low carrier frequencies and symbol rates has advantages compared to standard modems with these channels. Results and analysis of the detection method, a DFT (discrete Fourier transform) approach, are discussed in several original papers [Lal97b], [Lal99], [Lal00], [Lal01], [Lal02], and [Lal04b].

### *Signal Impairments in a Granular Channel*

A granular noise channel used in these simulations causes two kinds of impairments a. granular noise, b. slope overload seen earlier in Figure 4.6 and now in the Figure 4.17. Bit rates of 3000 bit/s may use with new complex waveforms (a combination of 4FSK and 8PSK). Figure 4.17 presents a simulated complex 2FSK-8PSK waveform before and after the granular noise channel. Table 4.4 suggests the use of lower frequencies for granular channel.

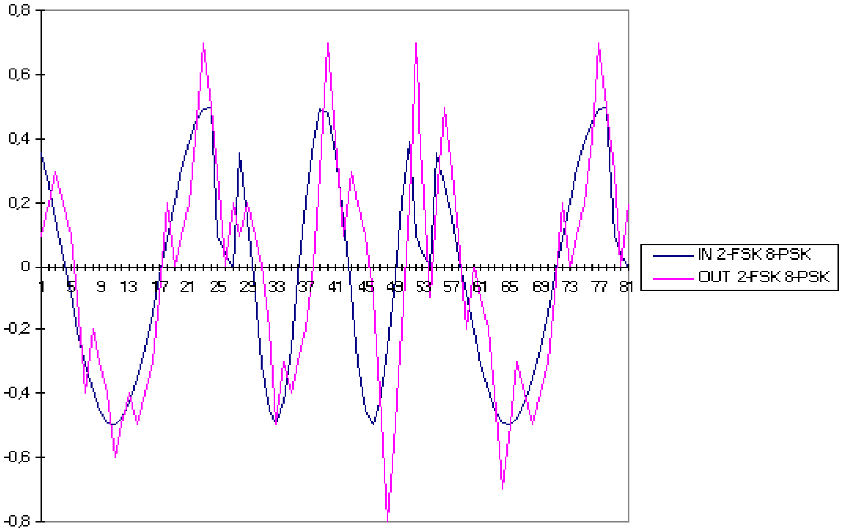


Fig. 4.17 2FSK-8PSK waveform in a granular channel [Lal97b]

## *Effects of a Multi-path Channels*

In the simulated results, presented in papers [Lal02] and [Lal04b], the data waveform has impairments starting at S/N<15 dB…20dB. The qualitative result shows the effect in amplitude and phase error, which can make the detection impossible in Figure 4.18. The normalized values of A and P are one. The subject is quite broad and needs a lot more investigations. The results in the paper show, that a robust worksheet modeling and simulation method can give rapid answers to most practical questions.

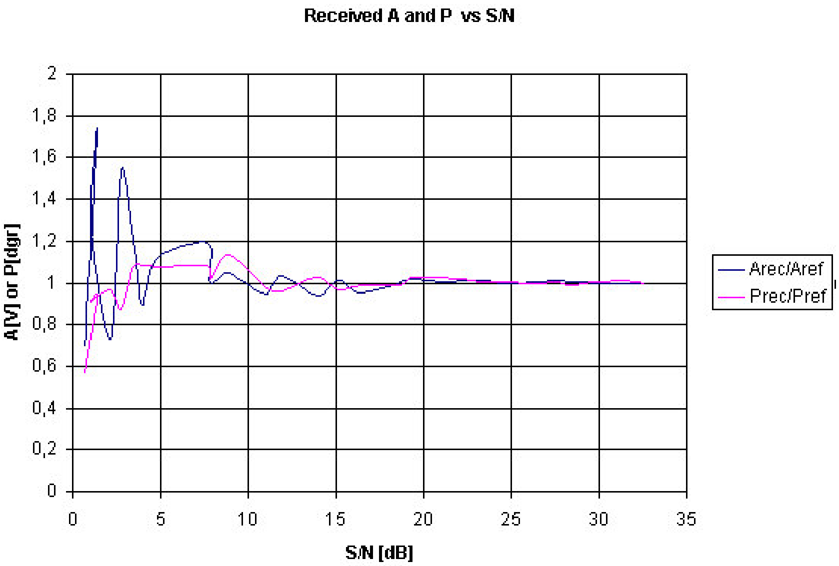


Fig. 4.18 Channel impairments effects in received A and P [Lal02]

## *Soft Detection of Waveforms*

Soft detection is the detection method used in simulations. It was performed with a N-point DFT based calculation process. First the method is used in the simple cases of FSK and PSK waveforms. Soon it was found that with N=26 and sampling frequency $f_S$ = 16000 detection of some multi-carrier modulation methods (MFSK) can be carried out. Later it was obvious that by using different sample numbers N and sampling frequencies $f_S$ more complex multi-carrier waveforms can be detected and with an inverse DFT multi-carrier waveforms were generated in the simulations. The band-limited multi-carrier approach is used in a prototype modem design described in chapter 5.

### *Soft Detection of Noisy FSK-Signals*

The first results with the FSK and PSK data transmission simulation over the 16 kbps ADM-channel were the evaluation of the number of samples needed in the detection of one symbol. Figure 4.19 presents qualitative results of soft detection of noise FSK-waveforms in a simulation using a 13-point DFT (N=13 samples per symbol) in detection. N=13 was enough for FSK but not for PSK where N=26 was needed.

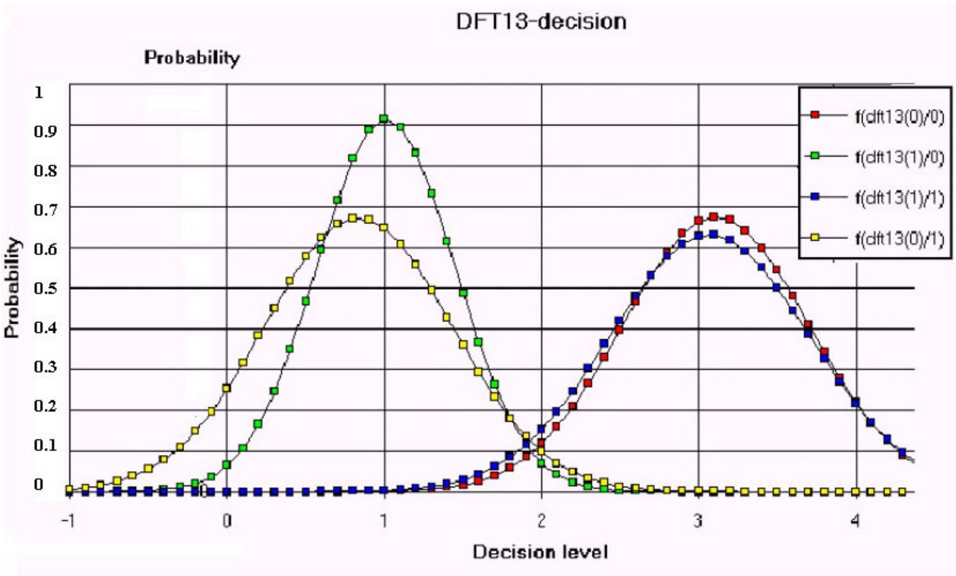


Fig. 4.19 Soft detection of noisy FSK-signals [Lal97b]

Robust examples of soft FSK detection simulation results with FSK waveform over granular AWGN noise channel are presented in Tables 4.6 and 4.7. The S/N is 65 and 7 dB in these cases giving an error in bit detection in the latter. Soft detection is based on the simple rule "select the largest FSK signal". Tables present the first 15 bits of the 205 bits in this example.

Table 4.6. Soft detection simulation results with FSK S/N = 65 dB

| | RESULT | | | |
|---|---|---|---|---|
| IN | T | FSK | SIGNAL | 1300 |
| bits | bits | DFT13(0) | DFT13(1) | BIT nro |
| 1 | 1 | 0,413343 | 0,879732 | 1 |
| 1 | 1 | 0,529696 | 1,380792 | 2 |
| 0 | 0 | 1,075283 | 0,679321 | 3 |
| 0 | 0 | 1,088647 | 0,569278 | 4 |
| 0 | 0 | 0,729953 | 0,668613 | 5 |
| 1 | 1 | 0,64243 | 1,301552 | 6 |
| 0 | 0 | 0,379822 | 0,285694 | 7 |
| 1 | 1 | 0,430707 | 0,86252 | 8 |
| 1 | 1 | 0,587002 | 1,354614 | 9 |
| 1 | 1 | 0,095181 | 1,487733 | 10 |
| 0 | 0 | 0,751494 | 0,444714 | 11 |
| 0 | 0 | 0,596189 | 0,347739 | 12 |
| 0 | 0 | 0,442083 | 0,251259 | 13 |
| 0 | 0 | 0,29109 | 0,156189 | 14 |
| 1 | 1 | 0,361007 | 0,651705 | 15 |
| 1 | S/N | 65,60253 | dB | |
| 7 | BER | 0 | 205 | 205 |

Table 4.7. Soft detection simulation results with FSK S/N = 7.5 dB

| IN | RESULT | FSK | SIGNAL |
|---|---|---|---|
| bits | bits | DFT13(0) | DFT13(1) |
| 1 | 1 | 0,413343 | 0,879732 |
| 1 | 1 | 0,529696 | 1,380792 |
| 1 | 1 | 0,430431 | 0,884521 |
| 1 | 1 | 0,361007 | 0,651705 |
| 1 | 1 | 0,353727 | 0,925599 |
| 0 | 1 | 0,203228 | 0,252977 |
| 0 | 0 | 0,294538 | 0,147042 |
| 0 | 1 | 0,215511 | 0,272362 |
| 0 | 1 | 0,203228 | 0,252977 |
| 0 | 0 | 0,29109 | 0,156189 |
| 1 | 1 | 0,361007 | 0,651705 |
| 1 | 1 | 0,28761 | 0,701455 |
| 1 | 1 | 0,45814 | 1,137804 |
| 0 | 0 | 0,791592 | 0,421765 |
| 0 | 1 | 0,203228 | 0,252977 |
| S/N | 7,455425 | dB | |
| BER | 0,019512 | 201 | 205 |

## Soft Detection of Noisy PSK-Signals

Phase detection is degraded due to phase jitter generated in the delta modulation process. Phase jitter of the ADM-channel is calculated with a 26-point DFT algorithm as a phase receiver. The software algorithm is programmed in Excel format in a worksheet simulator for the modeling of software detection of PSK and other waveforms. The 26-point discrete Fourier transform (N=26) is presented in formula (4.4). The simple calculation process is made assuming:

- The symbol waveform is sampled with sampling frequency $f_s$ = 16000.
- The number of samples used in a symbol waveform is N=26.
- The carrier frequencies $f_c$ are calculated and generated as $f_C = m \cdot \Delta(f)$.
- The selectivity in detection of waveforms is calculated as $\Delta(f) = f_S / N$.
- The reference signal is an 8-PSK-waveform in the simulated evaluation of soft detection of PSK-signals.
- The resulting lowest four carrier (m=1…4) frequency candidates, matching with the ADM-channel, are $f_c$ = 615, 1231, 1846 and 2462 Hz.
- The resulting frequency selectivity in detection of waveforms is *Δ(f)* = 615 Hz.

## 26-Point DFT Algorithm

$$S_X[m\Delta(f)] = \sum_{n=1}^{26} x[n\Delta(t)]e^{-2j\pi m\Delta(f)n\Delta(t)} \qquad (4.4)$$

The complex signal spectrum $S_x$ is the result of the DFT calculation in formula (4.4). The individual carriers $f = m \cdot \Delta(f)$ are separately evaluated and calculated for phase detection of the signal (for example an 8-PSK-signal as in Table 4.4-4.5. Every carrier *f* has its own $S_x$, which has a real part $x=Re(S_x)$ and an imaginary part $y=Im(S_x)$. The phase detection can be modified (some training is needed) from the phase estimate $P = \tan^{-1}(\frac{y}{x})$ depending on the phase constellation used. In general a phase estimate P for a frequency $f = m \cdot \Delta(f)$ is

$$P = \tan^{-1}\frac{\mathrm{Im}\{S_X[m\Delta(f)]\}}{\mathrm{Re}\{S_X[m\Delta(f)]\}} \qquad (4.5)$$

## *AWGN Channel and Noise Model*

Setting the noise level as demonstrated in reference [Bro93] the S/N-ratio was calculated for an average signal *s(t)* power and noise *n(t),* formula (4.6). In the case of Gaussian white noise, relationship between ó and the noise power spectrum $N_0$ is presented formula (4.7) [Bro93].

$$S/N = 10\log\frac{Var[s(t)]}{\sigma^2 Var[n(t)]} \tag{4.6}$$

$$\sigma^2 = \frac{N_o}{2} \tag{4.7}$$

Additive random noise source was designed in order to evaluate the error performance of the AWGN noise channels. Noise signal $N$ (positive or negative impulses) was added to every sample of AWGN channel waveform, see waveform examples earlier in Figures 4.6 or 4.17. The complex transmission chain was simulated with AWGN noise sources at the input and/or at the output of the granular channel, presented earlier in Figure 4.1. Figure 4.20 presents absolute values of $N$ for reference purposes with the normalized signal power of bits. The resulting simulated S/N ratio was also calculated and referred to the set parameter values of S/N. Generation of noise was based on the known simulated normal distribution by formula (4.8).

$$N = (\sum_{i=1}^{12} RND(i) - 6)\sigma + \mu \tag{4.8}$$

Where

$N$ = Noise amplitude
*RND(i)* = Random number 0...1, i=1, 2, 3,...,12
ó = Deviation of the noise distribution
ì = Mean of the noise distribution

Positive and negative noise impulses were added to the signal (absolute values in fig. 4.4) so that its mean ì was zero and its deviation óset the noise power for S/N ratio. In the simulations S/N was also calculated using the sample values of signal and noise peaks. The setting of S/N was compared to the actually simulated S/N values in the simulations presented in this thesis.

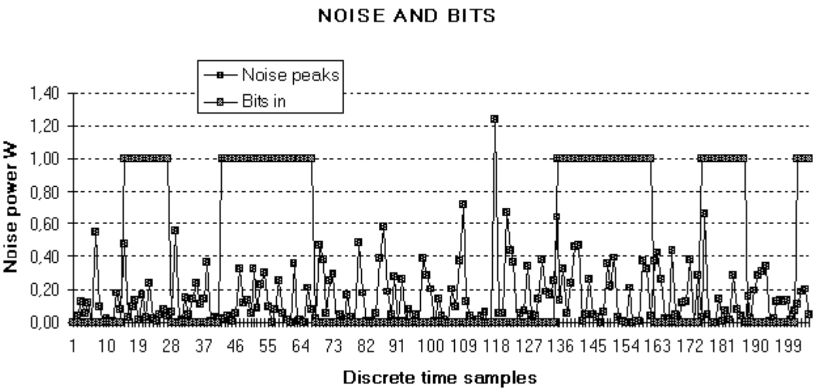


Fig. 4.20 Sampled noise modeling during symbol (bit) time

# 4.4. Simulations of Biomedical Data Networks

## Image Transmission

An evaluation of the digital image management is in reference [Rat05] as

1. The storage of radiological images in digital format is a non-trivial problem due to the very large volume of data that these images contain.
2. Projectional X-ray Images require very high resolution to be clinically acceptable. Such images must be acquired and stored in image matrices of more than 2000 by 2000 pixels, with a dynamic range of 8 to 12 bits per pixel. This represents between 4 to 8 Mbytes per image.
3. Digital imaging modalities such as computed tomography or magnetic resonance imaging generate images with smaller matrices (typically 256x256 or 512x512 with a dynamic range of 12 to 16 Bits per pixel).
4. The difficulty comes from the very large number of images generated for each patient examination. One examination can generate between twenty and more than one hundred images. This corresponds to storage requirements between 10 and 50 MBytes per study.

The conclusions of telemedicine simulations and the Comnet 3 LAN network traffic modeling are presented in paper [Lal04a]. The main results are briefly given here.

## Nature of band-limited data traffic– Simulated Results

The nature of data traffic was seen in Figure 4.21. The main difference between data and voice communication is traffic bursts in data transmission. Delays depend heavily on the data transmission lines used in simulations as the results of mean delays indicate:

<0.5 s delay with 9600 bps and

<100 s delay with 600 bps.

Many heavy unpredictable variations in traffic and delay times are seen in the low bit rate cases. In the Figure 4.21 we had a fixed 1000 byte mean message size. Delay times with the 600 bps lines are not acceptable but 9600 bps might be acceptable in some real world applications.

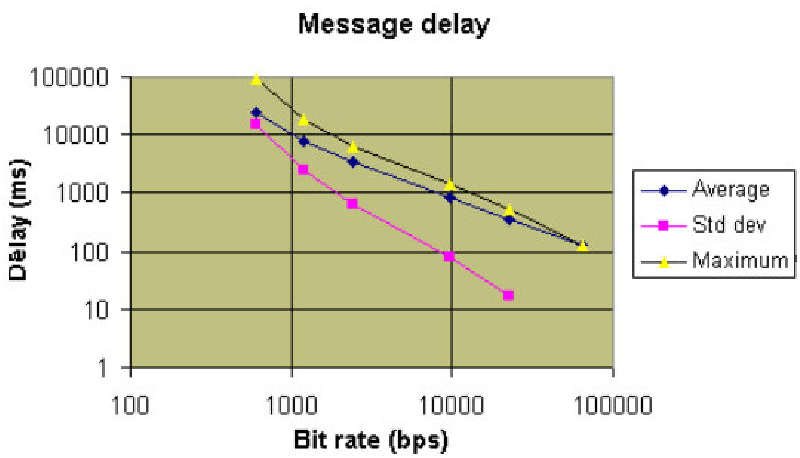


Fig. 4.21 Message delays [Lal04a]

### *Transaction Rate*

We investigated increasing transaction rates in a network using the 22.5 kbps adaptive waveform, as discussed earlier, in the backbone channels, results in Figure 4.22. We gradually increased the network traffic from message sources having a fixed normal distributed message size (100 kilobyte mean, 10 kilobyte standard deviation) and arrival time as variable 10 sec…160 sec. These arrival times gave us a practical transaction range of a biomedical institution with 65 000 to 260 000 transactions per year. Delays have an average (ave) and a standard deviation (std) value. We found that there is a transaction value limit below 250 000 per year at which this backbone network causes increasing delay time. In the evaluation of the simulated result the message size (100 kilobyte used) is a critical measure. If the messages include for example large images (1 Megabyte) then the number of transaction limit may drop to 25 000 per year. However, it is possible to work with a low data rate band-limited backbone but with larger delays. One conclusion is that the X-Ray images need wide band channels in order to minimize the delay times.

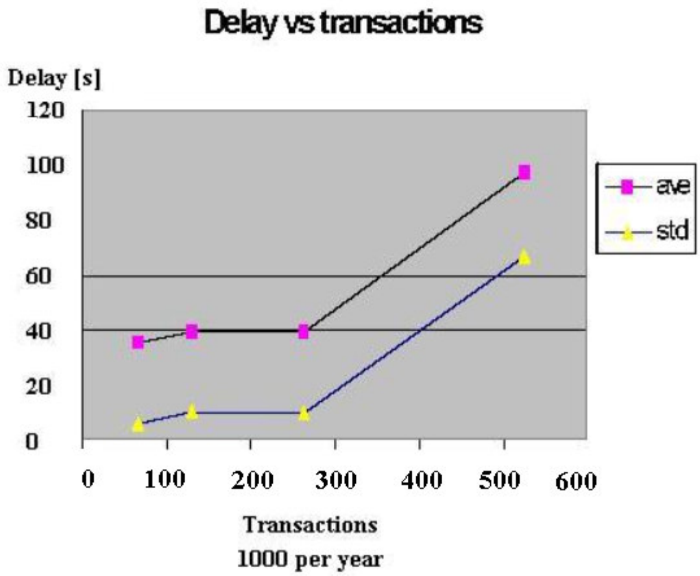


Fig. 4.22 Message delays [Lal04a]

## 4.5. Summary

The subject is quite extensive. We have investigated and discussed one model of data networks with limited simulated examples, network design principles using different channels (AWGN, granular and multi-path) and the selection of some modulation methods (waveform). This chapter serves as an introduction to the adaptive multi-carrier secure data communication system of the next chapter. The DFT based approach is based on the presented simulated results and the selection of waveforms using band-limited frequency hopping on the lowest OSI level in order to get security and optimal throughput for different channels.

The adaptive delta modulation (ADM) is used in several systems (mobile military and commercial digital recording systems) as a voice coding method. It was used in modeling digital granular channels. Both PCM and ADM are waveform-coding methods and their performance is similar in quality (S/N). Analog waveforms should stay at a high quality level (S/N) during the transmission over digital networks if the analog data transmission (waveform) is used end-to-end. The high S/N-ratio and low error rate are also generally important in biomedical information transmission.

# Chapter V

# 5. A DFT-Based Approach to Adaptive Data Communications

The expression "adaptive" means showing or having the capacity of or a tendency towards ada p-tation, while adaptation is the act or process of adapting or adjustment to environmental conditions [Web94]. Adaptive communications consist of a wide area of adaptive methods used in present communication technology and are described earlier. The following sections include a proposal and description of an adaptive multi-carrier data transmission system for telemedicine, alert systems and other authority use. It is a novel solution particularly with the security properties offered for OSI Model level 1 (physical), which have not been presented in the band-limited data transmission standardization. The modeling of the data transmission system and the simulation process is presented in papers [Lal04a] and [Lal04b]. It is capable of simulating all kinds of waveforms with given signal to noise (S/N) settings.

## 5.1. Basic theory

There are many elements, which have effects on the signal power, bandwidth, and time spent by a voice sample, an information byte or data block during the transmission over different channels, Figure 5.1. The data block or the information package of the figure is always a combination of signal power (S) needed, time (T) spent and bandwidth (B) used. The selectivity of time, frequency, amplitude and phase are the limiting factors and can be described as in chapter one with the minimum Euclidean distance $d$, formula (5.1).

$$d = \sqrt{\sum_{i=1}^{N} (p_1 - q_i)^2} \tag{5.1}$$

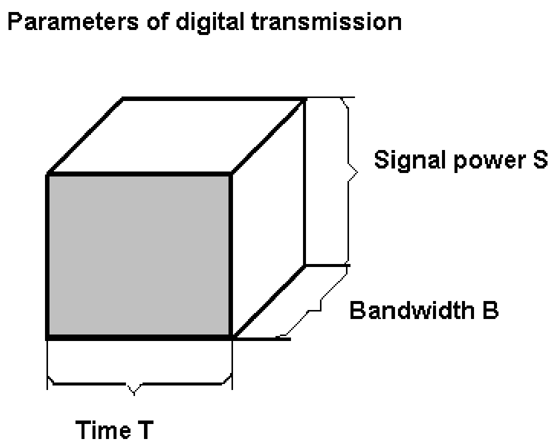


Fig. 5.1 Package of data in the channel [Lal97b]

The average signal power $S$ and additive white noise power $N_0$ are reference factors in the communication theory as presented in reference [Sha48]. Time element is effectively used in the present communication technologies like time division multiplexing (TDM). In the present systems the symbol time is fixed. The wide frequency band is used in the most modern telecommunication systems such as ultra wide band (UWB) systems.

In military telecommunication the signal power and time are often minimized because a data package presented in Figure 5.1 is a target of electronic warfare. The whole package is usually minimized. Many design parameters of the system have some effects on this package.

## *Adaptive Data Transmission*

Adaptive data transmission is introduced in the original papers [Lal97b], [Lal99], [Lal01], and [Lal02]. The three last papers can be found in the IEEE Communication Society Digital Library.

- Signal classification by discrete Fourier transform [Lal99] presents a signal classification method using Discrete Fourier Transform (DFT).
- Adaptive software modem technology [Lal01] presents a description of a new software modem technology.
- Basic theory of adaptive data transmission [Lal02]. The paper presents the basic theory of adaptive data transmission.

A brief description of adaptive data transmission, implemented as an adaptive modem, is here:

- The adaptive modem uses symbol time as a parameter. Thus the symbol rate is also a variable.
- The single adaptive modem operates on a very limited band (a channel). The full available bandwidth contains several channels. Thus the linearity of the total multi-carrier band is not necessary.
- The modulation scheme of each carrier is based on the channel properties, which can be automatically measured in a training process
- The best available complex digital modulation for the carriers is selected and thus the bit rate of the system is made optimal.
- The classical bit constellations of standard analog data modems of ITU-T have generally 1-5 bits in the digital modulation schemes. This is not limited in the adaptive modem theory because the transmission channel may give better performance in different cases.

The adaptive modem is described in more detail in the original paper [Lal00].

## *Selectivity in Soft Detection with Discrete Fourier Transform*

The main mathematical background in the generation and detection of waveforms is the discrete Fourier Transform (DFT), formula (5.2). All the parameters in the formula are adaptively selectable. Thus the full advantage of the adaptive modem is in the capability to change the carrier frequencies, amplitudes, phases, and symbol lengths. The complex form of the discrete Fourier transform includes the amplitude and phase information of the symbol.

The DFT is defined in several references, for example in [Mar62] and [Gol70], as an operation on an $N$-point vector $[x(0),x(1),\ldots,x(N-1)]$ as

$$X(k) = \sum_{n=0}^{N-1} x(n) W_N^{nk} \quad \text{, for k = 0, 1, 2, …, N-1} \qquad (.5.2)$$

where $W_N = e^{-j2\pi/N}$.

The complex form of the discrete Fourier transform includes the amplitude and phase information of the symbol.

$$S_X[m\Delta(f)] = \sum_{n=1}^{26} x[n\Delta(t)]e^{-2j\pi m\Delta(f)n\Delta(t)} \tag{5.3}$$

$$S_X[m\Delta(f)] = \sum_{n=1}^{13} x[n\Delta(t)]e^{-2j\pi m\Delta(f)n\Delta(t)} \tag{5.4}$$

The formula (5.3) calculates the discrete Fourier transform of a signal *x(t)* with N=26 samples. The formula (5.4) calculates with N=13 and could be used only for FSK-detection. Time is sampled $f_S$ times per second, which gives the sample time in the formula. The frequency selectivity is the ratio $f_S/N$. The individual mean filter frequency is *m* times the basic frequency selectivity m$f_S/N$, while $m = 1 \ldots M$ and $M$ = number of carriers. Thus the frequency selectivity depends on this relation. This is illustrated in Figure 5.2. In general time is sampled $f_S$ times per second, which gives the sample time in the discrete Fourier transform formula of a symbol of the signal *x(t)* with *N* samples. The total sampled signal *x(t)* consists of a piecewise continuous set of symbols that will be discussed later. The frequency selectivity comes from the ratio $f_S/N$. The individual mean frequency of filter is *m* times the frequency selectivity. Thus the frequency selectivity depends on this relation, which is illustrated in Figure 5.2.

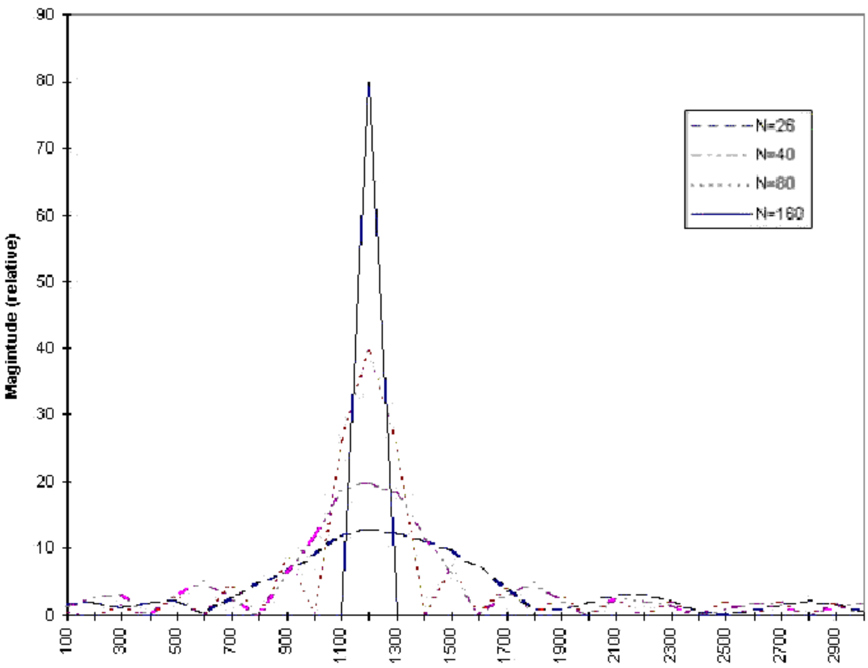


Fig. 5.2 DFT-filter [Lal99]

## *Limits for Bit Rates - Shannon's Channel Capacity Formula*

For the development of the adaptive modem the Shannon's capacity limit $C$ is the figure of merit. Formula (5.5), presented by Shannon, shows for *S/N* (average signal power over average white noise) the channel capacity is approximately $C$ for limited band $B$. The signal-to-noise ratio and bandwidth define Shannon's channel capacity limit for the AWGN-channel in 1940 [Sha48] as

$$C = B \log_2 (1 + \frac{S}{N}) \qquad (5.5)$$

The formula (5.5) includes important design parameters:
$C$ = Channel capacity bps
$B$ = Bandwidth Hz
*S/N* = Signal-to-noise ratio.

Shannon states that to approximate this limiting rate $C$ of transmission the transmitted signals must approximate, in statistical properties, a white noise. This approximation in a band-limited channel is done using multi-carrier systems (OFDM) with an adaptive modem designed and described later in this thesis. From the Shannon's limit one can generate the design principle used in the modem design and present wideband network development: To improve information transmission, in bits per second per Hz, it becomes necessary to increase the *S/N* or the bandwidth of the channel. In search for error free transmission, this theory yields to the use of complex modulation methods. OFDM has been found one of the most promising. Shannon's voice band capacity of different channels versus *S/N* is presented in Figure 5.3.

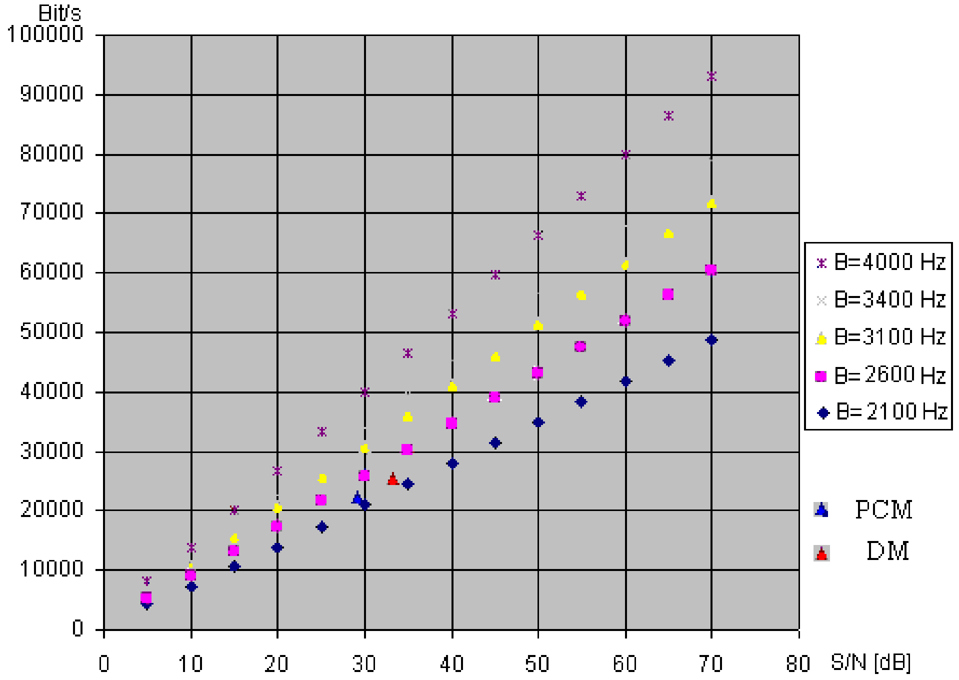


Fig. 5.3 Shannon's capacity of different channels versus S/N [Lal99]

## *5.2. Adaptive Modem*

The basic principle of the adaptive modem is the free selection of the data transmission parameters optimized to the channel conditions. This was implemented in the adaptive modem prototype, presented in papers [Lal00] and [Lal01].

In simulations a wide range of modulation method and bit rates were studied with the following results:

1. Bit rates 4000 –240 000 bps.
2. Bit constellation with 16-256 states.
3. Symbol rates 1000-3000 symbol/s.

In the modulation process of an adaptive modem several frequencies; symbol phases and amplitudes (QAM-states) are used. Its performance is calculated for example in a four-carrier $k$=4 as:

1. Using $f_S$ = 45000, $N$=26 the symbol rate is $R_S$ = 1730.7.
2. Using $M$ =5…8 bits per symbol the maximum bit rate is $R_b$ = 8653...13846 bps with one channel.
3. $R_b$ = 34.6...55.4 kbps with four channels (carriers), where $R_b = kMR_s$.

Figure 5.4 presents a simulated complex piecewise continuous waveform. There is a block of six symbols made with a multi-carrier soft modulation. Generation of these piecewise continuous symbol waveforms will be discussed in more detail later.

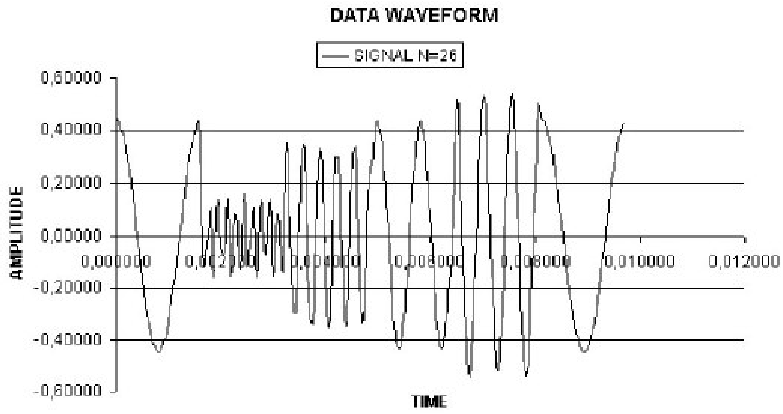


Fig. 5.4 Simulated complex waveform [Lal02]

In Figure 5.5 a step function is used as the input signal (IN) and the result is the simulated output of the 16 kbps granular channel (OUT). This result shows the performance of the robust worksheet method and the usefulness with different waveforms. This case is with for example detection of PPM (pulse phase modulation) used in UWB. This is a subject for other studies.

Sampling rate limits the performance in the received 'OUT" signal, which is the input signal for the adaptive modem. Figure 5.5 illustrates the granular channel properties. The performance and sensitivity of adaptive modem with different channel models are evaluated in a later section.

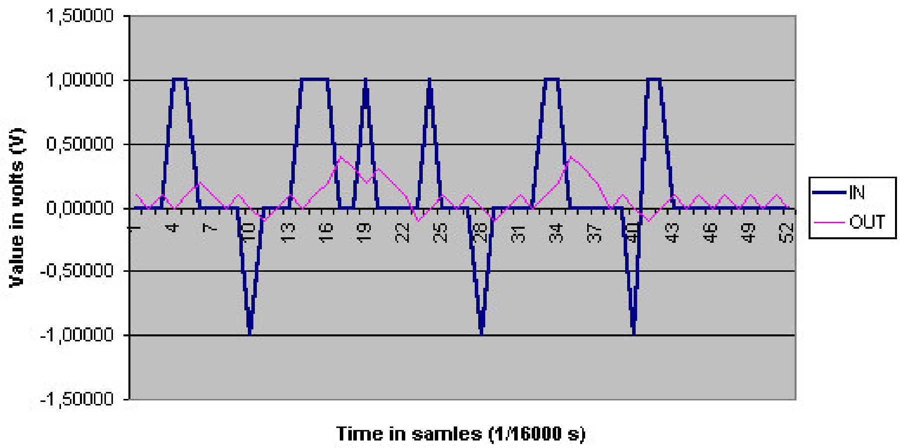


Fig. 5.5 Simulated narrow step functions

# 5.3. Generation of Symbol Waveforms

The best available complex digital modulation is selected and thus the bit rate of the system is made optimal. The present bit constellations of standard analog data of ITU-T have 1-5 bits in the digital modulation schemes. This is not limited in the adaptive modem basic theory because the channel may give better performance. In military telecommunication networks signal power and time are often minimized. A data package presented in Figure 5.1 is a target of electronic warfare. This target is usually minimized. Many design parameters of the system have some effects on this package. The simulation process, presented in papers [Lal97b], [ Lal04a] and [Lal04b], is capable of simulating all kinds of waveforms with given signal to noise (SNR) settings. Instead bits symbols (a specific number of bits) are converted to waveforms as described earlier in the simulation process.

A symbol is earlier called the data block and presented with its parameters in Figure 5.1. Several random digital bits (bit stream or block of bits) are collected together in a symbol using bit constellations as discussed earlier. A symbol waveform of one symbol (several bits) is a continuous waveform during the symbol time T. However, a waveform stream of several symbols is a piece-vice continuous waveform. Then a band-limited OFDM system is a combination of several piece-vice continuous symbol waveforms, which can be described as a three dimensional system (time, frequency and amplitude phase constellation) refer to Figure 5.1.

### Signal Source and Bit Stream

A digital information source generates bit streams. The signal source can be a computer, a human voice or any voice grade source or even a still picture or video source. However, if the source generates analogy information (voice or video), it must be coded into the digital form using different coding algorithms. The binary sequence $\boldsymbol{B_N} = \{B_N\}$ is usually made by a random process. One has the N-bit random digital stream $\{B_k\} = b_0, b_1, b_2, \ldots b_k, \ldots b_{N-1}$ available for the transmission over wired telephone lines, optical fibers or wireless radio channels, formula (5.6).

$$\{B_N\} = [b_0, b_1, b_2, \ldots, b_k, \ldots, b_{N-1}] \text{ ,k = 0,1,2,..,N-1} \qquad (5.6)$$

Where $b_k$ is the $k^{th}$ bit and $N$ is number of bits in this representation.

### Symbol Stream and Waveform Generation

This stream of bits $\boldsymbol{B_N}$ is transformed into a signal $\{S_L\}$ by using digital modulation methods discussed in Chapter 2. Thus the original binary signal is developed to a symbol stream $\{S_L\}$ presented in Formula (5.7)

$$\{S_L\} = [s_0, s_1, s_2, \ldots, s_k, \ldots, s_{L-1}] \qquad (5.7)$$

In a digital modulation process these bits or symbols (group of bits), performing with the digital modulation a digital to analog conversion (DAC), are converted and generated to analog voltages for transmission over analog telephone lines. Traditional telephone lines are band-limited thus the analog voltage (waveform) must be adapted to the line conditions (amplitude scale, frequency band, etc). A voltage representing these binary digits or symbols is transmitted over a communication channel as a waveform (a binary or symbol waveform). The present analog voice band modems, discussed in Chapter 2, can combine at least five bits into a symbol waveform.

In digital radio links the symbol waveforms can have more than ten bits in a symbol waveform while the useful bandwidth is not limited to the voice grade channel, see Table 5.1. The useful bandwidth is at present in the range of a few GHz. There are Giga-sample analog-to-digital converters (ADC) commercially available. All-digital UWB devices for indoor use have been proposed (Standard 802.15.3a for UWB-OFDM indoor system [Mil03]. Commercially available Giga-sample analog digital converters are presented in Table 5.1.

Table 5.1. Commercially available Giga-sample ADCs [Mil03]

| Vendor | Bits | Giga-samples per second | Bandwidth GHz | Power W |
|---|---|---|---|---|
| Maxim (Evaluation kit) | 8 | 1.5 | 2.2 | 5 |
| Atmel | 10 | 2 | 3 | 4.6 |
| Alma project (France) | 2/3 | 4 | 2-4 | 2 |
| Rockwell | 8<br>6 | 3<br>6 | -<br>- | 5.5<br>3.8 |

Let's say one has $M$ bits in one symbol $\{S_{L,k}\} = [b_0, b_1, b_2, ..., b_k, ..., b_{M-1}]$. By a teaching mechanism one can test the maximum bit constellation $M = M_{max}$ of symbols, which the digital modulation method can use in a particular channel. The standard data modems of ITU-T use 1…6 bit per symbol. The digital modulation methods make a D/A transformation bit-by-bit or symbol-by-symbol to waveforms, Appendces 1-3. Using M bits per symbol one gets the L-symbol stream $\{S_L\}$, formula (5.7).

## *Use of Design Parameters in Waveform Generation (Proposal)*

The waveform describes the symbol stream - a message. All symbols are formed of several bits. The following parameters were used in the software modem algorithm development in order to get the most suitable functionality. The performance of the modulation was then tested in the field (description of the test results in the coming sections).

| Parameter | Effects |
|---|---|
| Symbol time | Symbol rate - Sampling rate - Number of samples. |
| Amplitudes and phases | Bit constellation - A-P range and selectivity. |
| Modulation method | Bit rate - BER performance versus S/N. |
| Number of carriers | Bandwidth - Frequency selectivity. |

The adaptive parameter selection principle was proposed and used in field test: Parameter values were selected during the modulation training process for use in the symbol signal waveform as:

1. Amplitude.
2. Phase.
3. Symbol time.
4. Carrier frequency.

In the Figure 5.6 one finds a design example of the simulated adaptive and complex four-carrier waveform. One finds the design parameters: symbol length N=26 samples, 4 carrier frequencies, and bit constellation (several amplitudes and phases). The symbol waveform is adaptive when it is adapted optimally to the transmission channel.

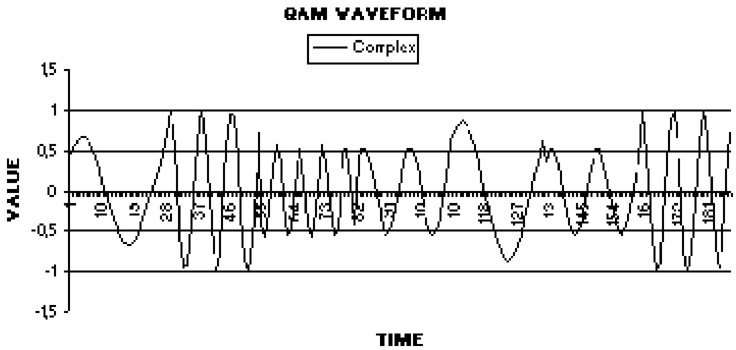


Fig. 5.6 Simulated complex four-carrier waveform [Lal01]

Several different waveforms and channel models were studied with simulations of the robust method [Lal04b]. Waveforms or their use are discussed in all original papers especially in [Lal97b], [Lal99], [Lal00] and [Lal01]. Table 5.2 summarizes the simulations and the harmful effects associated with different channels. The simple waveforms (FSK, PSK and DPSK) best resisted the different effects listed in the table. The sensitivity of the adaptive data transmission method against the harmful effects will be presented later in a sensitivity analysis section.

Table 5.2. Data waveform and channel models

| Modulation method | Granular DM, ADM | AWGN | Multi-path |
|---|---|---|---|
| FSK | Simulated, tested | Simulated, tested | Simulated, tested |
| PSK DPSK | Simulated, tested | Simulated, tested | Simulated, tested |
| MFSK | Granular noise | Noise level | Interference level |
| MPSK | Slope overload | Noise level | Interference level |
| Multi-carrier QAM | Granular noise Slope overload | Noise level | Interference level |

## *Adaptive Selection of Modulation Method*

It is an adaptive data modulation concept, where one defines adaptive data modulation as the modulation, where modulation parameters are optimized to the transmission parameters of the channel. Channels are:

- Analog channel (voice channel).
- Digital channel (data or digital coded voice channel).
- Multi-path fading channel (air interface).

The following channel parameters were in the adaptive modem algorithm selection process considered for different channel types:

1. Bandwidth.
2. S/N and S/(N+I+J) ratio.
3. Frequency range.
4. Multi-path signals in radio channel.

Adaptation of the bit rate to the channel characteristics is defined as selection of the number of carriers and with the change of modulation signal constellation (for example 4QAM to 16QAM). Examples of tested waveforms are presented in the session 5.3.2 field tests, where the selection method was:

- Number of carriers $k$ = 1, 2, 4, 5, 6 and 8.
- Bit constellation $M$ = 2-5 bits.
- Symbol rate $R_S$ = 703-2250 Bd.
- Number of samples $N$ > Symbol length.
- Sampling frequency $f_S$ > Symbol time > Symbol rate $R_S = f_S / N$.
- Number of amplitude and phase bits M > Modulation > Bit constellation.
- Number of carriers $k$ > Channels.

## *Base-Band Signal*

The voice grade channel of a telecommunication network has a limited bandwidth about 3.1 kHz (300 to 3400 Hz) for information transmission in a signal *m(t),* formula (5.8). The standard voice grade data modems have been limited to use this limited band.

$$m(t) = A_m(t)\cos[(\omega_C(t) + \phi_C(t)] \tag{5.8}$$

The data bits are presented by a base-band signal *m(t).* Information can be modulated at least into four basic development parameters: Time, frequency, amplitude and phase. The last two parameters are used in the well-known two-dimensional constellation mapping of bits into symbols. The basic formula (5.8) presents three generally used modulation parameters: carrier frequency $f_c$ in $\omega_C(t) = 2\pi f_C(t)$, amplitude $A_m(t)$, and phase $\phi_C(t)$. The resulting data waveform is a random (stochastic) piecewise continuous waveform, which represents the symbol stream of random bits.

If $A(t) = Am$, $f_c(t) = f_c$, and $\phi_C(t) = \phi_C$ i.e. all three are known constants, one has a deterministic waveform. Deterministic sinusoidal signals were used in the field tests for measurement of the telephone channel characteristic and in system synchronization processes (amplitude level adjustment and first symbol identification and detection) and random waveforms for data transmission by modems.

In this study random waveforms were used for the simulations of data transmission over ADM-channel. Constant amplitude $A_m$ and carrier frequency $f_c$ as a parameter selected from a band between 0…4000 Hz were used for simulation of the ADM-channel characteristics.

## *Adaptive Multi-Carrier Signal*

The band-limited adaptive signal is made using the sum (multiplex) signal *S(t)* of $M$ carriers each modulated with a selected modulation method as adaptive selection of the modulation method, Formula (5.9).

$$S(t) = \sum_{m=1}^{M} A_m(t)\cos[(\omega_{C,m}(t) + \phi_{C,m}(t)] \tag{5.9}$$

Where $M$ is number of carriers (sub-channels). Each carrier $f_{C,m}(t)$, amplitude $A_m(t)$, and phase $\phi_{C,m}(t)$ depend both on the selected adaptive modulation method, the symbol set (ASCII etc) used and the present symbol in transmission at the moment *t*. In some references the adaptive modulation method was selected in the wireless case according to the distance, attenuation and S/N ratio requirement of the particular modulation method (for example16QAM, 8PSK, 2FSK). The complex sampled multiplex signal *x(t)* is discussed in Chapter 2 in Section 2.3 'OFDM System Model", Formula (5.10) [Guo02].

$$x(t) = \sum_{i=0}^{+\infty} \sum_{k=0}^{N_S - 1} \sqrt{\{[s_I^2(k) + s_Q^2(k)]p(t - k\frac{T_S}{N} - iT_S)\}} \tag{5.10}$$

Where $N$ is the number of samples in a symbol waveform, $T_S$ = symbol time, discrete time is $iT_S$ , $i = 1 \ldots \infty$, number of carriers is $N_S$ , $k = 0 \ldots N_S - 1$, $x(t)$ amplitude is the resultant sum of I and Q-channels ($S_I$ and $S_Q$), $t$ is time and $p(i,k,t,N,T_S)$ is the pulse function of symbols.

# 5.4. Soft Detection of Symbol Waveforms

The formula (5.3) calculated the discrete Fourier transform of a signal *x(t)* with $N$=26 samples. The formula (5.4) calculated with $N$=13 and could be used only for FSK-detection. Time is sampled $f_S$ times per second, which gives the sample time in the formula. The frequency selectivity is the ratio $f_S/N$. The individual mean filter frequency is $m$ times the frequency selectivity, $m = 1 \ldots M$ and $M$ = number of carriers. Thus the frequency selectivity depends on this relation. This was illustrated in Figure 5.2. The detection of the constellation of a particular carrier $f_{C,n} = m\Delta(f)$ is made setting $m$ and $f_S/N$. Thus one gets the individual signal amplitudes as a complex value from formula (5.3) as

$$\operatorname{Re} S_X[m\Delta(f)] = \sum_{n=1}^{26} x[n\Delta(t)] e^{-2j\pi m\Delta(f)n\Delta(t)} \tag{5.11}$$

$$\operatorname{Im} S_X[m\Delta(f)] = \sum_{n=1}^{26} x[n\Delta(t)] e^{-2j\pi m\Delta(f)n\Delta(t)} \tag{5.12}$$

From (5.8) one gets I- and Q-signals for N = 26 as

$$\operatorname{Re} S_X[m\Delta(f)] = \sum_{n=1}^{26} x[n\Delta(t)] \cos(2\pi m\Delta(f)n\Delta(t)] \tag{5.13}$$

$$\operatorname{Im} S_X[m\Delta(f)] = \sum_{n=1}^{26} x[n\Delta(t)] \sin[(2\pi m\Delta(f)n\Delta(t)] \tag{5.14}$$

Simplifying real and imaginary parts one get I and Q signals as

$$S_I = \operatorname{Re} S_X[m\Delta(f)] \tag{5.15}$$

$$S_Q = \operatorname{Im} S_X[m\Delta(f)] \tag{5.16}$$

Amplitude $A$ and phase $P$ of each carrier $f_{C,n} = m\Delta(f)$ will be as

$$A = \sqrt{S_I^2 + S_Q^2} \tag{5.17}$$

$$P = \tan^{-1} \frac{S_Q}{S_I} \tag{5.18}$$

In simulations and prototype development of an adaptive modem the amplitudes $A_m$ of multi-carrier signal are normalized for all carriers $f_{c,m}$ , $m = 1 \ldots . M$ as

$$A_m = \frac{A}{M} \tag{5.19}$$

Phase in formula (5.18) is more difficult to adjust, because it is periodic. It can be trained to a proper value range for each particular symbol as will be shown a little later in an example for 8-PSK detection.

### *Modeling Software Detection*

Data waveforms have a constant symbol rate $R_S$. In software detection with the DFT algorithm the sampling frequency $f_S$ and the number of samples N give the symbol rate $f_S/N$. The detection of the waveforms is based on the set of parameters. The resulting bit rate is generated from the selected symbol rate, sample frequency, number of samples, and number of carriers used as:

- Symbol rate $R_S = f_S/N$.
- Sample frequency $f_S$.
- Number of samples in a symbol $N$.
- The bit rate $R_b = kMR_s$.
- The number of carriers used $k$.
- The number of bits in the symbol $M$.

Software detection is made with the DFT algorithm of the simulation system presented in Figures 5.7-5.8 based on paper [Lal04b]. This simulation system was used for the development of the adaptive detection of different waveforms and their performance. The result of the development work is the full adaptive modem with the ability of the selection of frequency $f$, amplitude $A$, phase $P$, and symbol time $T$.

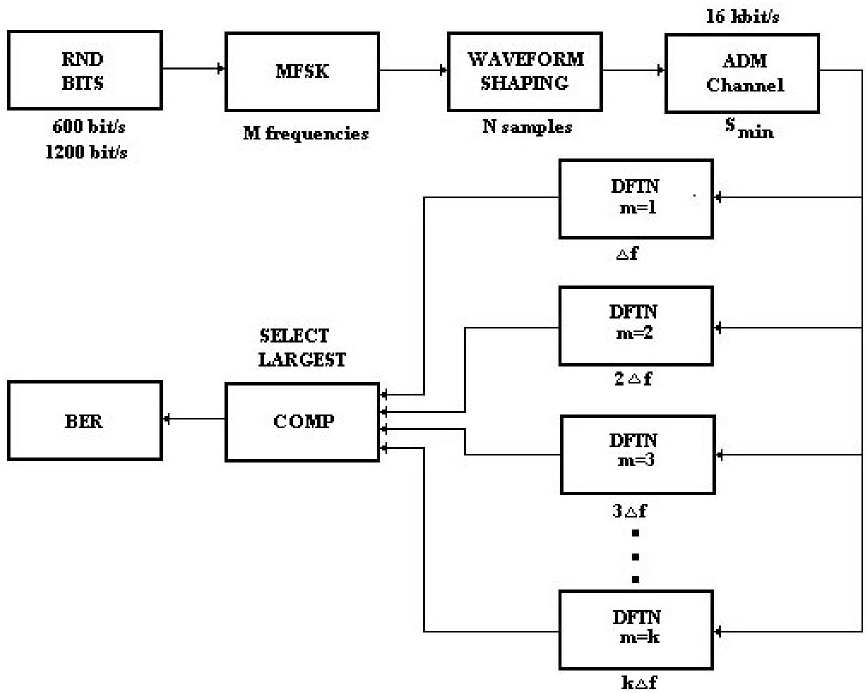


Fig. 5.7 Data waveform simulations with worksheet

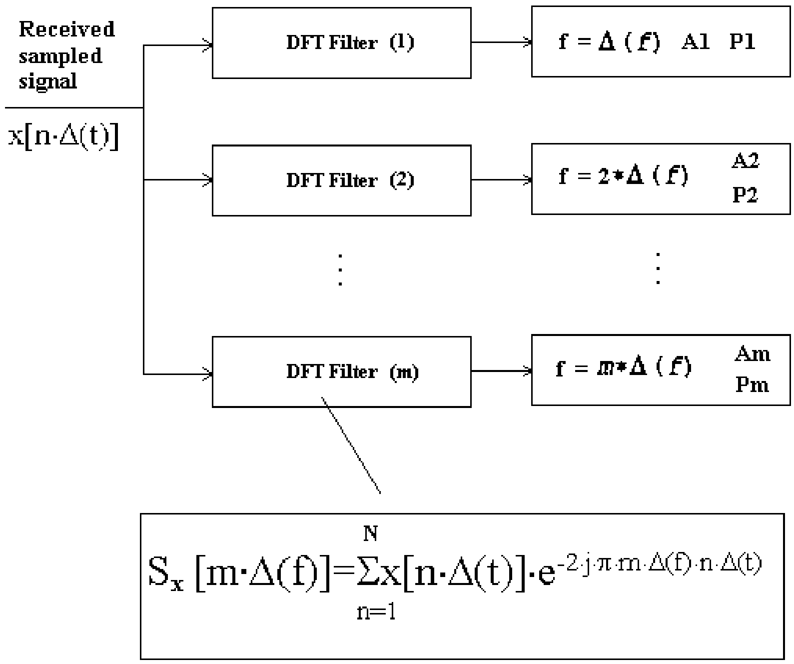


Fig. 5.8 DFT-Detection of sampled waveforms [Lal04b]

Figure 5.7 presents a MFSK detection case and Figure 5.8 a more generalized case, where QAM states of carriers are decoded (in literature called a finger or RAKE receiver).

## *Example*

In the simulations the different digital modulation schemes have to be modeled. In this example the detection within a 45 degrees range of 8-PSK is possible using different carrier frequencies. BER values presented earlier in the chapter were simulated with 8-PSK. The algorithm used for a bit decision of 8-PSK with DFT26-algorithm in a worksheet simulator was trained as:

```
=IF(AND(FY56>93;FY56<138);1;
IF(AND(FY56>139;FY56<188);2;
IF(AND(FY56>188;FY56<227);3;
IF(AND(FY56>229;FY56<259);4;
IF(AND(FY56>273;FY56<317);5;
IF(OR(AND(FY56>319;FY56<360);
AND(FY56>0;FY56<8));6;
IF(AND(FY56>8;FY56<49);7;8)))))))
```

The principle of using 2FSK-8PSK-signal in a granular voice channel was first presented in [Lal97b].

# 5.5. Implementation and Test Results of Adaptive Modem

The basic principle of the adaptive modem is the free selection of the data transmission parameters optimized to the channel conditions. This was implemented in the adaptive modem prototype, papers [Lal00] and [Lal01]. The use of adaptive waveforms in data transmission was also discussed in a Milcom 2002 conference tutorial and paper [Lal02]. Modeling software detection is a large subject for another study.

## 5.5.1. Simulated Waveforms

In simulations a wide range of modulation methods and bit rates were studied as an approach to the design of an adaptive modem prototype. For example high band-limited data rates as:

1. Bit rates 4000 –240 000 bps.
2. Bit constellation with 16-256 states.
3. Symbol rates 1000-3000 symbol/s.

In Figures 5.4 and 5.6 a simulated complex waveform of the adaptive modem were presented. In the Figure 5.4 a six-symbol block with four carrier frequencies and several phases and amplitudes is seen. Its performance is evaluated with formula (5.20) for example as:

1. Using $f_S = 45000$, $N$=26 the symbol rate is $R_S = 1730.7$.
2. Using $M$ =5…8 bits the maximum bit rate is $R_b$ = 8653...13846 bps with one channel and $R_b$ = 34.6...55.4 kbps with four channels (carriers).

$$R_b = kMR_s \tag{5.20}$$

Where

- Number of carriers is $k$.
- Bit constellation has $M$ bits.
- Symbol rate is $R_S$ [Bd].

## 5.5.2 Field Tests

### Field Test Arrangements

During a data transmission field-test the team made wave generation and detection experiments using the adaptive software modem, Figures 5.9-5.11. The team examined data transmission waveforms over an analog voice channel with AN/PRC-77 type VHF radio set upgraded with the adaptive modem. The upgrade device connected to the radio 'POWER" connector (a wide band audio modification) gave a 22.5 kbps (6.8 bit/Hz) band-limited wireless bit rate performance. The spectrum of the tested base-band waveform is presented in Figure 5.11. This was a good voice band result compared to cellular radios in 2000, which had 9.6 kbps. The present higher data rates were due to the wide band use in frequency domain (multi-slot in time domain), B in formula (5.5). The adaptive waveform generation was based on the software algorithm, which can be downloaded from Internet or built-in into the digital communication system control files.

In field tests a wide range of modulation method and bit rates were studied in wireless and wired environments, Figures 5.9-5.10. The following settings (parameters) over a VHF radio channel were used in the field tests of the adaptive modem prototype:

1. Symbol length 16, 20, 24, 48 and 64 samples.
2. Symbol time 444 -1422 microseconds.
3. Symbol rate 703-2250 Bd.
4. Modulation 4QAM, 8QAM, 16QAM and 32QAM.
5. Number of carriers 1, 2, 4, 5, 6 and 8.

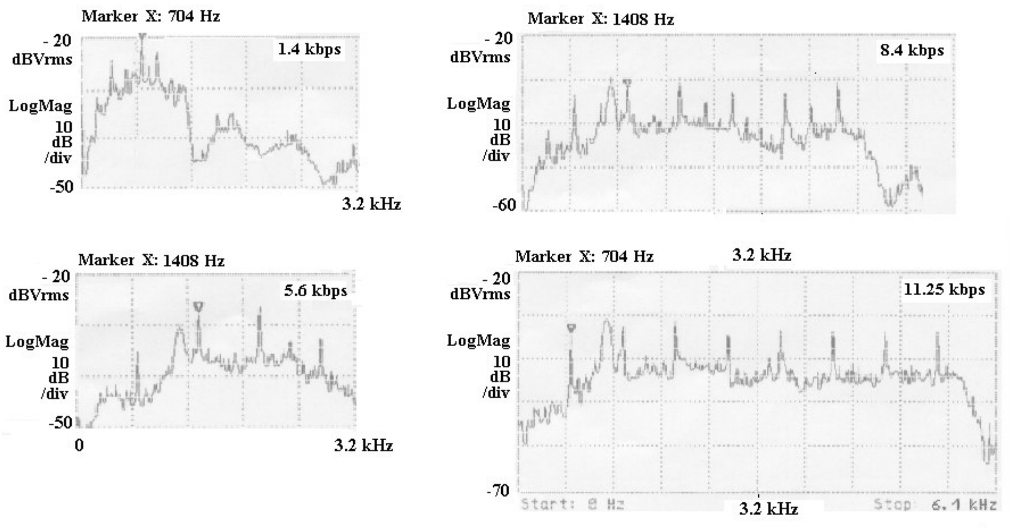


Fig. 5.9 Spectra of adaptive waveform measured in the VHF field test, [Lal02] and [Lal04b]

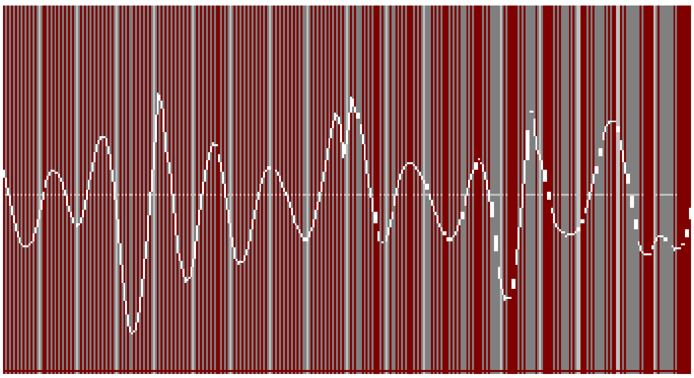

Fig. 5.10 Waveform captured in the VHF field test

### *Case 1 Wireless Channel*

Adaptive waveforms were generated with the earlier given settings. The results of the data transmission tests with these settings in the band-limited wireless case with VHF military radios were:

1. Bandwidth 600 - 4800 Hz.
2. Bit rates 1.4 … 22.5 kbps.

Figure 5.11 shows some results of the VHF-range field tests. The bit rate was developed with multiple carriers and thus using bandwidth. The bandwidth was limited to the voice grade. The received waveform in Figure 5.10 is 4-QAM with no impairments observed. By changing the parameters described earlier one could find an optimum throughput for the radio channel in question.

The best result of the VHF radio channel test (no error found in test messages) was about 6.8 bit/Hz sing a two carrier 16-QAM-modulation. The spectrum of the 22500 bps 16-QAM waveform is presented in Figure 5.11. Other examples of adaptive data communication development and DFT-detection are described in a paper presented in 2002 a Milcom conference [Lal02].

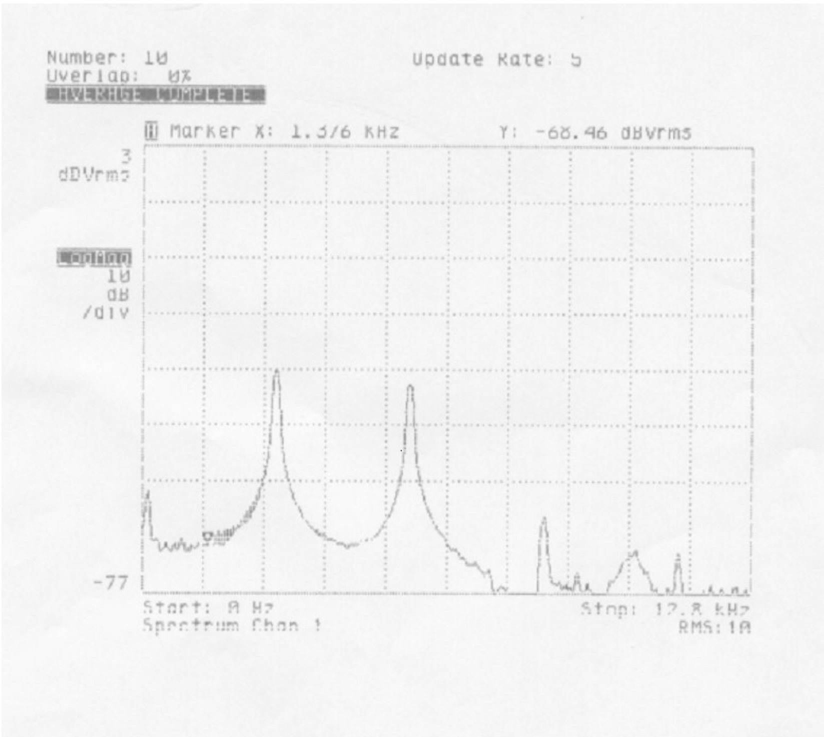


Fig. 5.11 Spectrum of 22.5 kbps signal – a result of the VHF field [Lal01]

Test results:
The best bit rate for the channel used in the test was $R_b = 2 \cdot 5 \cdot 2250 = 22500 bps$ using 32-QAM
The spectral efficiency was then about 6.8 bit/Hz.

In general the evaluation of bit rate $R_b$ of the adaptive modulation method is made in a multi-carrier ($k$ carriers) case with formula (5.20) as

$$R_b = kMR_s \tag{5.20}$$

Where
- Number of carriers $k$ = 1, 2, 4, 5, 6 and 8.
- Bit constellation $M$ = 2-5 bits.
- Symbol rate $R_S$ = 703-2250 Bd.

Table 5.3. Adaptive communication test results

| Band<br>Carriers | Symbol rate<br>Bit rate | Modulation A-P<br>QAM level |
|---|---|---|
| 550-1100 Hz<br>2 | 600<br>600 | 2FSK |
| 550-2200 Hz<br>4 | 600<br>1200 | 4FSK |
| 550-4400 Hz<br>8 | 600<br>1800 | 8FSK |
| 550-4400 Hz<br>8 | 1200<br>3600 | 8FSK |
| 550-4400 Hz<br>8 | 2400<br>7200 | 8FSK |
| 550-4400 Hz<br>8 | 2400<br>12000 | 8FSK, A=1 P=1 |
| 550-4400 Hz<br>8 | 2400<br>19200 | A=1 |
| 550-4400 Hz<br>8 | 2400<br>19200 | P=1 |
| 550-4400 Hz<br>8 | 2400<br>38400 | A=1 P=1<br>4QAM |
| 550-4400 Hz<br>8 | 2400<br>76800 | A=2 P=2<br>16QAM |
| 550-4400 Hz<br>8 | 2400<br>153600 | A=4 P=4<br>256QAM |

### *Case 2 Wired Channel*

The bit rates of the adaptive modem were 600-153,600 bps in the wired telecommunication network test, Table 5.3. During these tests the method of adaptive selection of waveforms was demonstrated as presented in the Table 5.3. The modulation method (algorithm) was selected during the test by changing the modulation parameters (soft detection). The table presents test results of data transmission over band-limited channels (A = amplitude bits, P = phase bits). Multi-carrier QAM-modulation methods are advantageous. MFSK offers much slower bit rates.

The wired channel bit rates of the adaptive modem were much higher than 5625-22500 bps the result of the wireless telecommunication network test, paper [Lal00]. Several other examples with DFT-detection algorithms are described in paper [Lal01] and [Lal02].

Conclusions of adaptive data transmission theory are summarized here:

- The adaptive data modulation method uses DFT in detection of waveforms.
- There are a lot of signal *f, A, P, T* and software detection parameters *N, m, fs* available in the optimization process of the data communications in radio or wired networks.
- The optimal use of the bandwidth is designed in this study by the proper selection of carriers *(f)*.
- Waveform is made to resist noise and interference with optimal selection of symbol time *T,* carrier frequency *f*, and bit constellations *A, P*.
- DFT-parameters, sampling frequency *fs*, number of samples *N* and number of frequencies *m*, are selected in reception for the best performance needed for BER or data rate or other metric.
- The throughput can be optimized in regard to BER versus S/N by selecting the most suitable bit constellation used in the digital modulation method.

The proposal for using adaptive waveforms in different channels in telemedicine (alert systems etc) or in the software-defined radios (SDR) is presented. In the future more studies on optimal waveforms for multi-path propagation should be made.

# 5.6. Discussion of Fourier Theory, Limitations and Applications

In this section some basic theories and applications of adaptive data communications are discussed. The basic theories are first of all Shannon's channel capacity formula and the Discrete Fourier Transform (DFT), which is used in the selection of adaptive data communication waveforms and soft detection. Waveforms are generated in an IDFT process and detected with a Discrete Fourier Transform (DFT) algorithm. Sampling is made with A/D or D/A devices. The discrete event simulation theory is used as a research tool, study and development of secure adaptive data communications and a prototype modem [Mit82]. Ciphering in the modulation is a proposed process for planning secure adaptive data communications. The focus in this chapter is on the realization of secure data transmissions with complex waveforms in accordance with the Shannon's channel capacity theory.

## Fourier Theory

Jean Baptiste Joseph Fourier did his important mathematical work on the theory of heat in 1807 publishing "On the Propagation of Heat in Solid Bodies". The Fourier series are based on Fourier's expansions of functions (trigonometrical series) of this old work [Bos17].

A waveform is a continuous signal in time domain. The Fourier Transform provides the means of transforming a signal defined in the time domain into one in the frequency domain. A digital symbol can be transformed into a finite waveform (signal) in a digital modulation process for transmission in different analog communication channels. The Discrete Fourier Transform (DFT) is an approximation of the continuous Fourier transformation. The Fast Fourier Transform (FFT) is a DFT algorithm developed by Tukey and Cooley in 1965 [Coo65]. It reduces the number of computations of a N-point transform (N samples) on the order of N2 to N log N in digital operations.

## Limitations of DFT

In data communications Discrete Fourier Transform (DFT), microcomputers and Digital Signal Processing (DSP) with high processing speed are used quite early [Har82]. DFT is commonly used for calculation of a power spectrum. DFT thus includes an algorithm for detection of MFSK signals.

The DFT algorithm can be used to approximate the transformation of a continuous time function with the following limitations:
- The signal must be band-limited.
- Aliasing. The sampling rate must be sufficiently high to avoid to any spectral overlap.
- Leakage. The observation of the signal is limited to a finite interval. The effect is a spreading or leakage of the spectral components and an undesirable modification of the total spectrum (distortion).
- Picket-Fence Effect. The inability of the DFT to observe the spectrum as a continuous function but only at discrete points. The spectrum is limited to integer multiples of the fundamental frequency $F$ (reciprocal of the sample length $N$). The major peak of a signal component might not be detected.

In Digital Signal Processing (DSP) different means are employed to avoid the problems:

- Use sampling rate high enough to avoid any spectral overlap or an anti-aliasing filter.
- Multiply the signal by a suitable window function that minimizes the spreading.
- A procedure for reducing the picket-fence effect is to vary the number of points in a time period by adding zeros at the end of the original record. The original record is intact.

## *Waveform Generation and Detection with DFT*

A DFT calculation gives complex values $z=x+jy$ of a finite signal for a finite time period and a given frequency. Amplitude and phase response (spectrum) is a Fourier series calculated by DFT. DFT performs symbol (bit constellation) detection of a digital modulation method. The inverse DFT is used as a signal generator, Figure 5.12. It digitally modulates the symbols stream

$$\{S_N\} = [s_0, s_1, s_2, \ldots, s_k, \ldots, s_{N-1}] \tag{5.21}$$

into a piece-vice continuous waveform *x(t)* in a symbol by symbol *(k)* waveform generation as

$$x(t) = \sum_{i=0}^{+\infty} \sum_{k=0}^{N_S-1} \sqrt{\{[s_I^2(k) + s_Q^2(k)] p(t - k\frac{T_S}{N} - iT_S)\}} \tag{5.22}$$

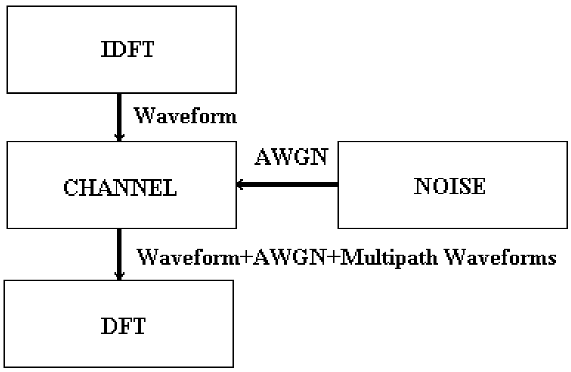


Fig. 5.12 Digital wave generation and detection

DFT is used in the waveform generation and detection of very high data rate OFDM systems [Kif01]. In a band-limited software modem design and in a data transmission simulator DFT is the starting point of the algorithm design [Lal04a, Lal04b], Figure 5.12.

A symbol waveform is a finite interval signal and thus with a proper selection of parameters one can use the Discrete Fourier Transform (DFT) for real time detection of adaptive multi-carrier waveforms. Fast Fourier Transform (FFT) was not quite suitable for the adaptive software modem, because the use of the number of samples N is limited to powers of two or in some cases to four. FFT is not fully adaptive as a DFT solution, when an adaptive selection of carrier frequencies, frequency selectivity, symbol rate or bit constellation states is wanted. All these are selected with a few parameters *N* and *Δ(t)* in DFT, formula (5.23). The sample interval *Δ(t)* is defined by the sampling rate $f_s$.

$$S_X[m\Delta(f)] = \sum_{n=1}^{N} x[n\Delta(t)]e^{-2j\pi m\Delta(f)n\Delta(t)} \tag{5.23}$$

The balance between the number of samples $N$ and the sample frequency $f_S$ is set in the detection. The other parameters used in the adaptive data communications theory are the bit rate $R_B$, symbol rate $R_S$, symbol time $T_S$, the number of bits in one symbol, bit constellation, digital modulation scheme, and the number of frequencies used in the channel. All these parameters are selectable variables. The channel characteristics are used in optimizing the waveform and for adjusting the modulation parameters. Then the software algorithm is optimized according to the selected test measure (BER versus S/N, bit rate versus bandwidth).

## *Adaptive Selection of Modulation Method*

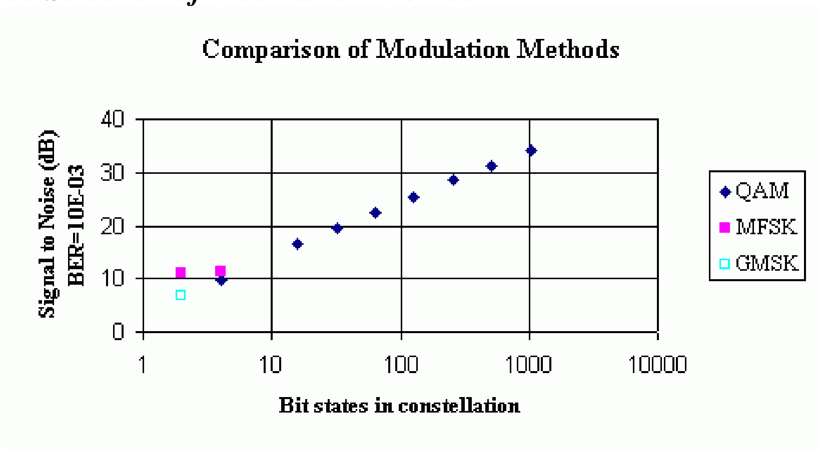


Fig. 5.13 Adaptive selection of modulation method,
S/N versus constellation

A comparison of modulation methods is made in reference [Car86] pp. 552-554. Adaptive selection of the modulation method is illustrated in Figures 5.13 - 5.15. Figure 5.13 presents a comparison of modulation methods based on the minimum signal-to-noise ratio needed for working on the BER-level = 1.0E-03. In cellular networks a modulation working on a low S/N ratio level can be economically a sound solution, because a large cell size is available. However, in the future low or high bit rates needed randomly in various services argue for an adaptive approach in the use of digital modulation methods and thus the selection of an optimized waveform. At the same time the adaptation of bit rates to distance can increase the tariff possibilities offered by the operators. Adaptation can follow technology in a way that high bit rates are in smaller cells and the lowest bit rates are offered in larger cells and for distant customers. Appendices 1-3 explain some proposals made for adaptive communications.

# 5.7. Adaptive Multi-Carrier Data Communications

The adaptive multi-carrier data communications like OFDM have been intensively studied [Ist99], Appendces 2-3. A band-limited prototype modem (an upgrade device for wireless radios) is one example of field-tested adaptive systems. Its main property is the adaptive selection of the following data transmission parameters [Lal01]:

- Channel bandwidth.
- Carrier frequencies.
- Bit constellation (amplitude/phase) states.
- Symbol and bit rate.

Table 5.4 presents examples of simulated band-limited adaptive multi-carrier waveforms, symbol rates, QAM modulation states, number of channels, equivalent MFC-code and the corresponding bit rates achieved. Symbol generation is made with an inverse DFT algorithm. Symbol detection is based on the complex DFT, formula (5.23). The complex DFT of the symbol calculated over the symbol time fully describes the particular bit pattern in the constellation diagram i.e. amplitude and phase at the DFT frequency used. Thus it gives us the symbol identification parameter estimates (amplitude, phase and frequency) with known frequency selectivity described later.

Table 5.4. Adaptive multi-carrier modulation methods and bit rates [Lal01]

| Symbol Rate | QAM | Channels | MFC-code frequencies | Bit/s |
|---|---|---|---|---|
| 1000 | 1*16 | 1 | (1 1) | 4000 |
| 1000 | 1*16 | 2 | (5 2) | 8000 |
| 2000 | 1*8 | 2 | (4 2) | 12000 |
| 2000 | 1*16 | 2 | (5 2) | 16000 |
| 2000 | 1*64 | 2 | (24 2) | 24000 |
| 3000 | 1*64 | 2 | (17 2) | 36000 |
| 3000 | 1*64 | 3 | (65 2) | 54000 |
| 3000 | 1*64 | 4 | (513 2) | 72000 |
| 3000 | 1*64 | 5 | (514 3) | 90000 |
| 3000 | 1*64 | 6 | (285 4) | 108000 |
| 3000 | 1*64 | 7 | (803 4) | 126000 |
| 3000 | 1*64 | 8 | (640 5) | 144000 |
| 3000 | 1*64 | 9 | (572 6) | 162000 |
| 3000 | 1*64 | 10 | (541 7) | 180000 |
| 3000 | 1*128 | 10 | (685 8) | 210000 |
| 3000 | 1*256 | 10 | (836 9) | 240000 |

### *Adaptive Data Communication Applications*

Adaptive data communication applications are discussed in this section. They can be designed according to the principles studied in earlier sections and chapters using adaptive selection of modulation method and waveforms. This section presents a method for adaptive selection of waveforms, adaptive filters and filter banks and a secure communication system. In the detection of waveforms adaptive filters or filter banks are needed. In securing data transmission an adaptive multi-carrier modulation system can be used.

### *Bit Stream*

One describes a bit stream, which has to be transmitted on-line or via the air. In the transmission of the bit stream adaptive digital modulation methods symbol by symbol are used. One describes a symbol with a waveform that contains several bits. The adaptive modulation method means that one can adapt the generated symbol waveform to the analogue channel used. The modulation of the particular carrier is made by changing the software algorithm, which converts each symbol to a specific amplitude-phase constellation point and uses a proper symbol time. Depending on the channel bandwidth $B$ (wired or radio) there is one or several transmission carriers (channels) in use for the optimal Shannon's capacity. Depending on the channel characteristics or signal to noise ratio one can select the best amplitude phase constellation. One uses the discrete Fourier transform (DFT) in the detection and the demodulation of the symbol waveform.

## *5.8 Adaptive Selection of Modulation Method*

A long-range transmission of bits is not possible in the form of a two DC state signal i.e. a binary signal, which is the way a PC operates. The sinusoidal waveforms of a symbol sequence can travel long ranges in the air or on-line. There are effective digital modulation methods, which combine several bits into one symbol and make a corresponding symbol waveform, Figure 5.14. One describes the adaptive software modulation method instead of standard digital methods. The reason for the use of adaptive waveforms is due to practical telecommunication networks, where one has a variety of different channels. They offer different bandwidths, SNR and continuously varying characteristics in mobile cases. Design of an adaptive modulation method begins in the selection of dynamic range and selectivity of the system using carrier frequency $f$, amplitude $A$, phase $P$ and time period $T$. The limits for the selectivity and dynamic range of each parameter are set according to the channel characteristics.

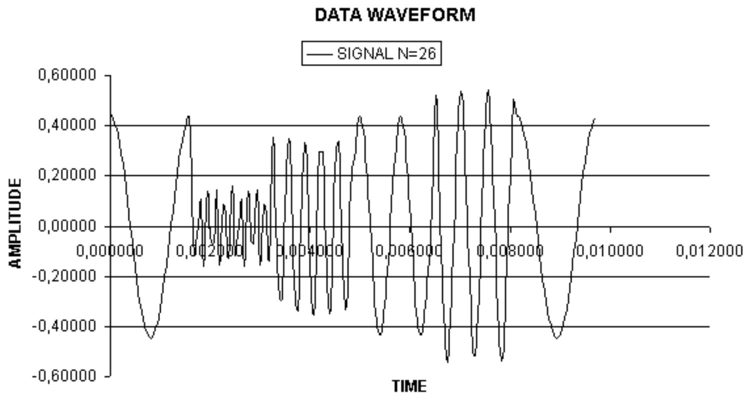


Fig. 5.14 Symbol waveform stream [Lal04b]

The adaptive modulation method and waveforms are a generalization of known digital modulation methods and waveforms. A development team has carried out experiments with a prototype adaptive modem during the field tests in order to demonstrate the effectiveness of different modulation methods, Figure 5.15.

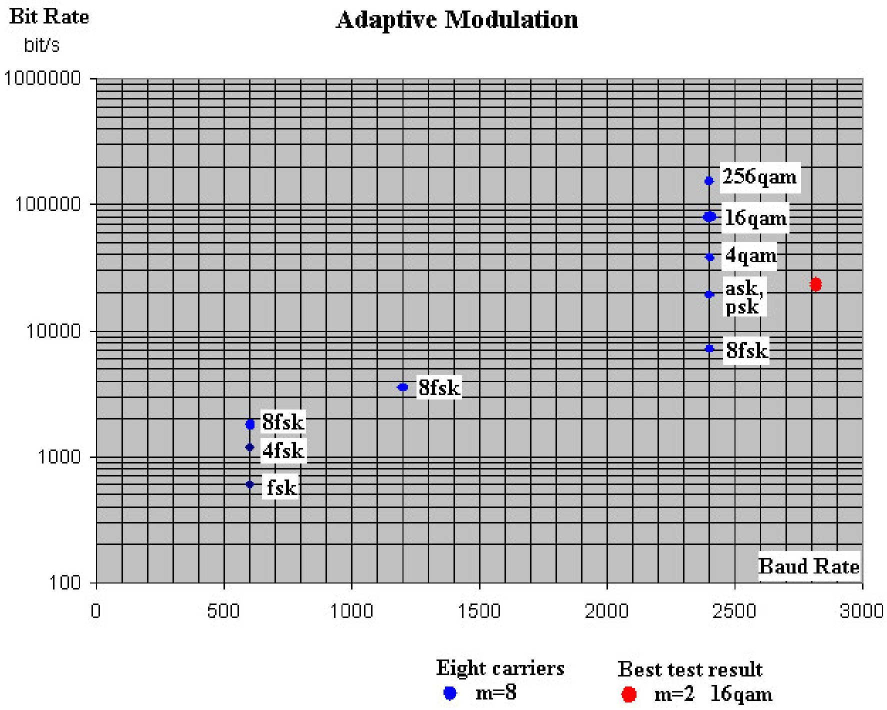


Fig. 5.15 Adaptive selection of modulation method,
bit rate versus baud rate

The MFSK method is not very effective as one can see in Figure 5.15. Increasing the symbol rate (baud rate) does not help the situation. To get high bit rates one has to use more complex modulation methods and multi-carrier systems. OFDM is a data transmission solution, which gives high bit rates and is the basis for the most recent communication system development projects [Kif01]. The adaptive selection of the modulation method gives advantageous bit rates and also the proper bit error performance for the individual cases. This is an optimization problem discussed in paper [Lal04b].

## *Adaptive Filter*

A human ear can detect fine frequency differences. The same with software detection using DFT was presented earlier in Figure 5.2. It explains the situation with DFT i.e. by increasing the number of samples one gets more narrow filters and the possibility to use narrow frequency bands and channeling. One can adapt multi frequency signal waveforms to the bandwidth in use. The number of samples $N$ and the sampling frequency $f_s$ define the frequency selectivity and the useful channel bandwidth. Figure 5.16 presents simulation results using two, the 13-point adaptive filters in the detection of FSK-signal transmitted over ADM-channel.

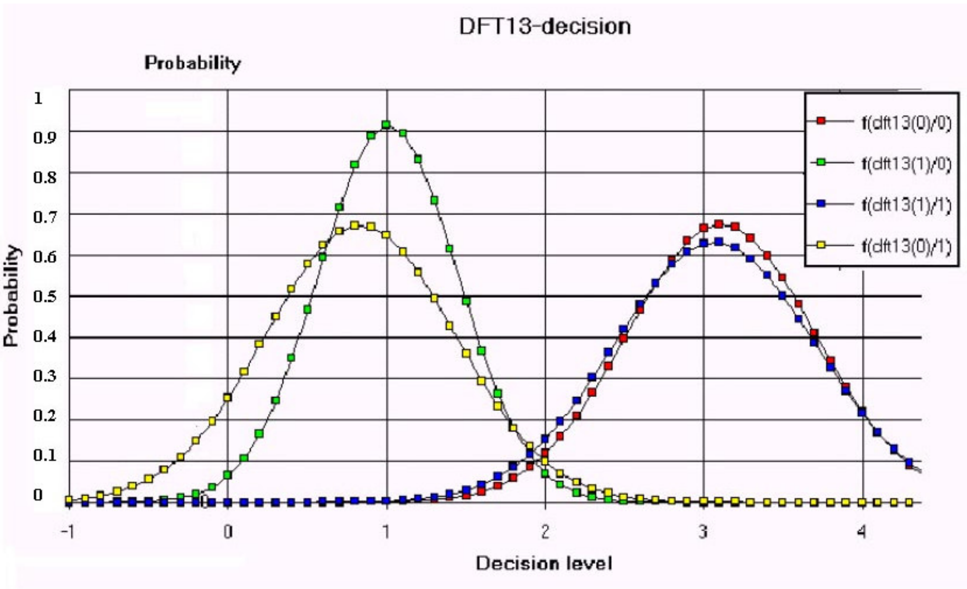


Fig. 5.16 DFT as filter [Lal97b]

## *Adaptive Filter Bank*

The detection of multi-carrier waveforms is made with adaptive DFT filter banks. A simulation model and example of a 26-point DFT filter bank is in Figure 5.17. The first six filters and their output amplitudes at different frequencies are shown in the figure. The center frequency of the first filter is 615 Hz and the next is 1230 Hz etc. The bandwidth of the filters is also 615 Hz. The noise floor (S/N) at 615 Hz is 15 dB. The noise floor with DFT detection depends on the number of samples used in detection and thus the relation between sample frequency and the signal frequency. Thus the realization of the DFT-based software filters is not easy at higher frequencies. The present technology level might be at the 2 GHz [Mil03].

The important components are A/D or D/A devices and the processors, which are used for the Joint Tactical Radio System (JTRS). One can easily change all the parameters of the waveform and thus also the bandwidth of a band-pass filter and carrier frequencies etc. The change of a parameter may also have an effect on the resulting throughput bit rate. The noise level and the bit rate must be balanced with the selected digital modulation method for an error free result.

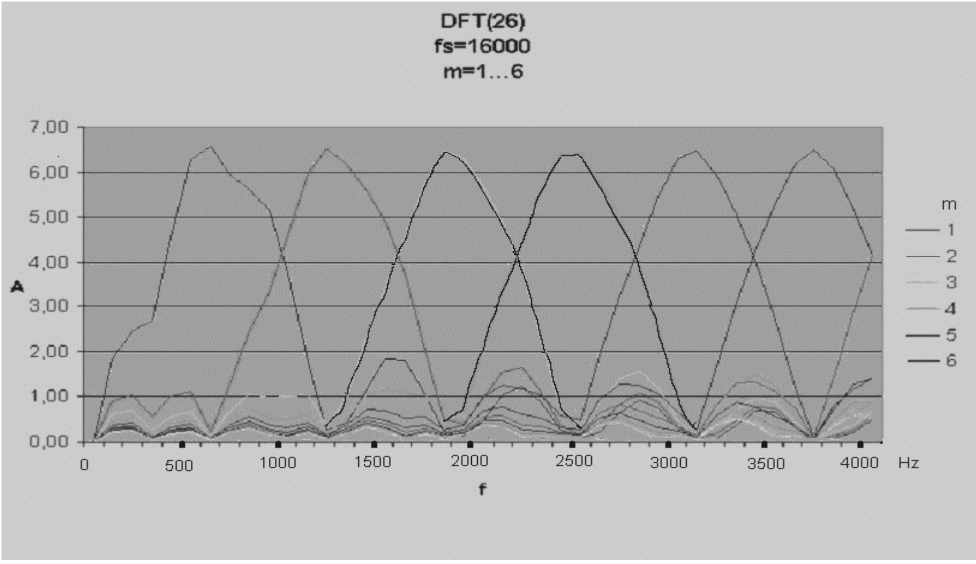


Fig. 5.17 Adaptive DFT filter bank [Lal02]

## *Secure Communication*

An example of secure communications is a multi-carrier system with $m$=8 carriers tested with the adaptive modem prototype, later in Figure 5.18. The figure includes a synchronism signal sent first in the waveform stream. A proposal for securing data communication is proposed by applying a band-limited frequency hopping (FH) waveform. As an example a FH waveform using $m$=8 carriers is analyzed. One designs a secure waveform with FH carriers representing a symbol sequence $\boldsymbol{S} = \{Sk\}$, formula (5.24).

$$\{S_N\} = \left[s_0, s_1, s_2, ..., s_k, ..., s_{N-1}\right] \tag{5.24}$$

Where

$s_k$ = the k$^{th}$ symbol
$k = 0,1,2,...,N-1$
$N$ = the number of symbols.

A secured band-limited signal is the sum waveform of eight carriers (a multiplex signal). The physical securing of $\boldsymbol{S}$ is made using a random FH signal as one in the multiplex signal of eight carriers. The ciphered data is digitally modulated in one FH signal among the multiplex signal of the eight carriers. The other seven carriers are digitally modulated with random data. Some problems might be in the selection of carriers. The problem with large m is the Picket-Fence Effect as described earlier. Thus the carriers for the sum waveform should be selected carefully in order avoid major peaks in the signal. Also the selection of a digital modulation method depends on the particular channel quality.

# 5.8. Adaptive Secure Data Transmission Method for OSI Level 1

## Security

By passing adaptive waveforms over the Internet or over a telephone network to another LAN server we can build our own VPN tunnel channel with OSI level one securing. The securing is made in the adaptive modem with band-limited frequency hopping in the voice frequency band. This can be done with the adaptive modem software algorithm in the data modulation process with a multi-carrier system. The basic theory is presented in paper [Lal02b]. It is a question of a signal microscope in the signal space, when a N-point DFT with a large enough N is used for symbol detection as described in chapter 3. In a hardware world such a filter is impossible to make. A DFT-based soft detection method uses such algorithms adaptively.

## A Proposal Application for Secure DFT-based Alert System Using Broadcasting

Remembering the worst earthquake catastrophe in Asia on 26.12.2004, with a missing swell alert system, it is obvious that a reliably working global alert system is needed. A general warning system for any kind of catastrophes based on public radio broadcasting with added multi-tones is proposed here. Short multi-tone voice messages describing the alert message in question can be detected with a DFT-based software algorithm application described earlier in this chapter. Normal radio receivers should have a small additional device for receiving these messages. The government offices (police stations, rescue authorities etc) should have the responsibility to listen to the broadcasting 24 hours a day and at the same time possible alerts and carry out necessary actions. For example, a three-tone detection and DFT filter needed with the receiver is easy to realize with an adaptive DFT detection algorithm (ref. fig.5.14) formula *S(m1,m2,m3)* for a three-tone signal as

$$S(m_1, m_2, m_3) = \sum_{n=1}^{N} x[n\Delta(t)]e^{-2j\pi m_1 \Delta(f) n\Delta(t)} + \sum_{n=1}^{N} x[n\Delta(t)]e^{-2j\pi m_2 \Delta(f) n\Delta(t)}$$
$$+ \sum_{n=1}^{N} x[n\Delta(t)]e^{-2j\pi m_3 \Delta(f) n\Delta(t)} \tag{5.25}$$

Where $m_1$, $m_2$, and $m_3$ define the three multi-tone carriers. In a reliable detection method the thresholds are set for all detected individual tones $S_1$, $S_2$, and $S_3$. The alert is accepted with the simultaneous detection of all tones.

## *Secured Waveform*

It is well known that wireless Internet access or LANs are open to any recording, taping, interfering etc. There is no standardization for security in the physical OSI reference level. Thus there is a need for data encryption as a minimum requirement in Internet traffic with TCP/IP-protocol systems. For example in biomedical and telemedicine data traffic one needs a standard for the physical level data security [Var03].

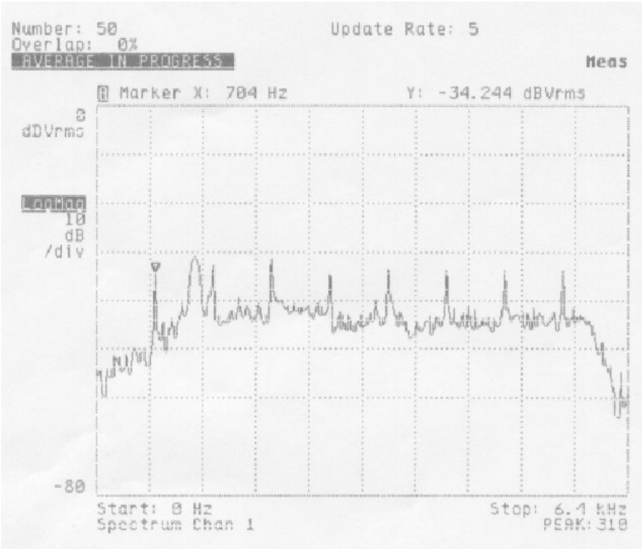


Fig. 5.18 Secured multi-carrier (m=8) waveform [Lal04b]

A simple version of the proposal for securing data transmissions on the physical or modulation level is illustrated in Figures 5.14, 5.17 and 5.18. Figure 5.14 gives an example of a secured waveform. Figure 5.18 shows the signal on the frequency band seen by the receiver and Figure 5.17 presents the filter bank function for its detection.

Suppose one has a symbol sequence $S_k$ for data transmission, formula (5.24). One defined symbol stream vector $\boldsymbol{S} = \{S_k\}$, which is the transmitted message. In a digital modulation process this stream is converted into a piecewise continuous waveform stream vector $\boldsymbol{W}=\{Wk\}$, formula (5.26).

$$\{W_N\} = [w_0, w_1, w_2, \ldots, w_k, \ldots, w_{N-1}] \tag{5.26}$$

Where $w_k$ = the k$^{th}$ waveform.

The whole multiplex signal matrix $\boldsymbol{M}=\{M_{m,k}\}$ is made in a random process. For a message of $N$ symbols and $M$ carriers one gets a signal as

$$\{M_{k,N}\} = \begin{bmatrix} m_{1,1} & m_{1,2} & \cdot & m_{1,N} \\ m_{2,1} & m_{2,2} & \cdot & m_{2,N} \\ \cdot & \cdot & \cdot & \cdot \\ m_{M,1} & m_{M,2} & \cdot & m_{M,N} \end{bmatrix} \tag{5.27}$$

Using an orthogonal frequency base of $M$ frequencies one can modulate each carrier using an adaptive digital modulation method i.e. by selecting adaptively the proper QAM-level for each carrier according to the channel quality.

A secured waveform for a particular message transmission is generated by hopping the frequency in a random sequence, which is at least as long as the message $N$. Taking the advantage of frequency hopping (FH) in a base-band modulation process one has now the secured waveform on a physical level.

Cryptographic methods are not studied here, however, for additional securing of the message one can use secret bit constellations *(A, P)* presented by formulae (5.17-5.18), which are other than those defined in standards.

In decoding the signal from the multiplex of $M$ signals one needs to know the hopping code *{C}* for the particular message signal as

$$\{C_{k,N}\} = \begin{bmatrix} c_{1,1} & c_{1,2} & \cdot & c_{1,N} \\ c_{2,1} & c_{2,2} & \cdot & c_{2,N} \\ \cdot & \cdot & \cdot & \cdot \\ c_{M,1} & c_{M,2} & \cdot & c_{M,N} \end{bmatrix} \qquad (5.28)$$

$N$ message elements $c_{k,N}$ one in each column are 1, $k=1 \ldots N,$ while all other elements are 0. In the similar way hopping codes for other messages can be constructed in a hopping system.

One gets the decoded signal $\boldsymbol{S}$ in general as resulting signal carriers in a matrix operation as

$$S = CF^T \qquad (5.29)$$

Hopping sequence is secret i.e. the carriers are selected with a secure random process known only by the two end users of the secure adaptive end-to-end communication. Vector $\boldsymbol{F}=\{F_M\}$ defines in general the M frequencies used.

$$\{F_M\} = [f_0, f_1, f_2, \ldots, f_k, \ldots, f_{M-1}] \qquad (5.30)$$

### *Robust Complexity Evaluation*

In Figure 5.18 the number of carriers is $M = 8$ (the second broader peak is caused by a synchronizing deterministic sequence of symbols). FH is now a random sequence of eight carriers and the hopping rate is the symbol rate. The multiplex waveform includes eight random waveforms. If the message size is $N = 10$ symbols, there are 10 columns in the code $\{C_{10,8}\}$ and matrix element 1 means a symbol in each column of $\{C_{10,8}\}$ as

$$\{C_{10,8}\} = \begin{bmatrix} 1 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 0 & 0 & 1 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 1 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 1 & 0 & 1 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 1 & 0 & 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \end{bmatrix} \tag{5.31}$$

The code matrix is thus a 10x8 matrix and may be called a block code for eight messages. Eight carriers can send eight messages each containing ten symbols in this example case. Each message has an individual code matrix, where in every column only one element on a random row one. All other elements of that code are zero.

Supposing the carrier set (M=8) is the same for all symbols k=1…10 and it is a priori known and the eight carriers are presented in the carrier frequency vector as

$$\{F_{k,8}\} = [705,1410,2115,2820,3525,4230,4935,5640] \tag{5.32}$$

Then in the special case of fig. 5.15 the carriers in the message of ten symbols are calculated as

$$S = CF^T \tag{5.29}$$

The result is

$$\{S_{10,8}\} = [615,4920,1845,2460,4305,1845,3075,1230,3075,1230] \tag{5.33}$$

It is supposed that the digital modulation method is a priori known. Thus final decoding of the information from the waveforms represented in *$\{S_{10,8}\}$* is made as described earlier in formulae (5.17-5.18).

The complexity of this cipher system (one of eight carriers) is 5.85E+48. A reference value of the DES code with 54-bit sequence is 1.8E+16. Using a 512-carrier system in the same way as the proposed cipher system one gets the reference level of complexity 2E+146.

# 5.9. Sensitivity Analysis of Adaptive Data Communications

A simulation method can only give qualitative results of the error performance of the investigated systems. However, due to mathematical complexity several systems can only be investigated with modeling and simulation. The sensitivity analysis thus gives robust information of error performance of the presented adaptive data communication method, which uses DFT-based soft detection and IDFT-waveform generation. The channel models are granular, AWGN and multi-path.

## 5.9.1. Sensitivity of the Soft DFT Detection in AWGN-Channel

Simulation settings in Figure 5.19 are:

- AWGN noise is a parameter. It is generated in the transmission channel and a received signal is disturbed by this noise.
- Signal amplitude A is a variable.
- f1=615 Hz, f2=1230 Hz, and f3=1846 Hz.
- N=26.

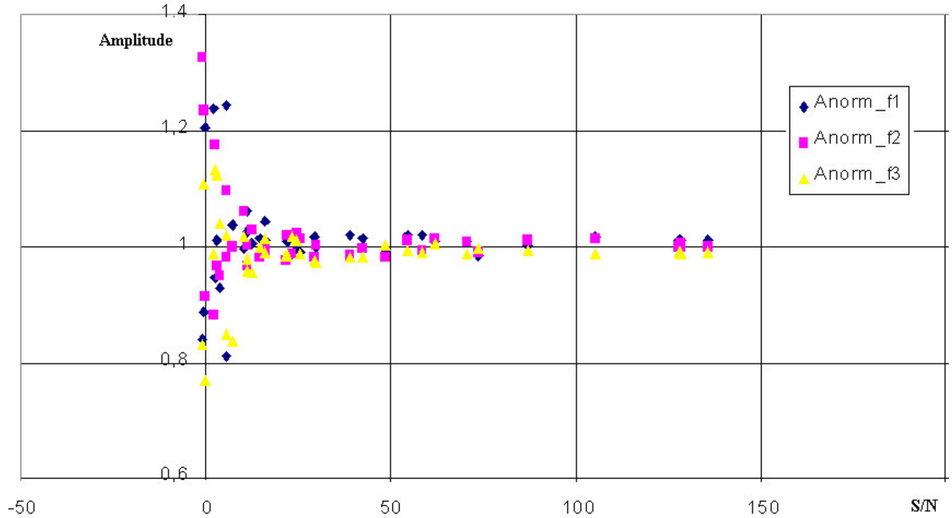


Fig. 5.19 Error performance of DFT soft detection in presence of AWGN

Figure 5.19 presents amplitude sensitivy of a multicarrier signal transitted over an AWGN channel and detected using using soft DFT detection. Error performance of a 26-point DFT detection is a simulated result. The amplitude sensitivity against AWGN noise sets the limits for usable S/N and amplitude selectivity in the selection of the modulation method. The qualitative result gives at S/N=10 dB and amplitude variation of 10%.

### *5.9.2. Sensitivity of the Signal Detection in Granular Channel*

Simulation settings in Figure 5.20 are:

- Granular noise is generated in the adaptive delta modulation process and is in the received signal.
- AWGN noise is generated in the transmission channel in the same way as presented in Figure 5.19.
- Signal amplitude is a variable.
- F1=615 Hz, f2=1230 Hz, and f3=1846 Hz.
- N=26.

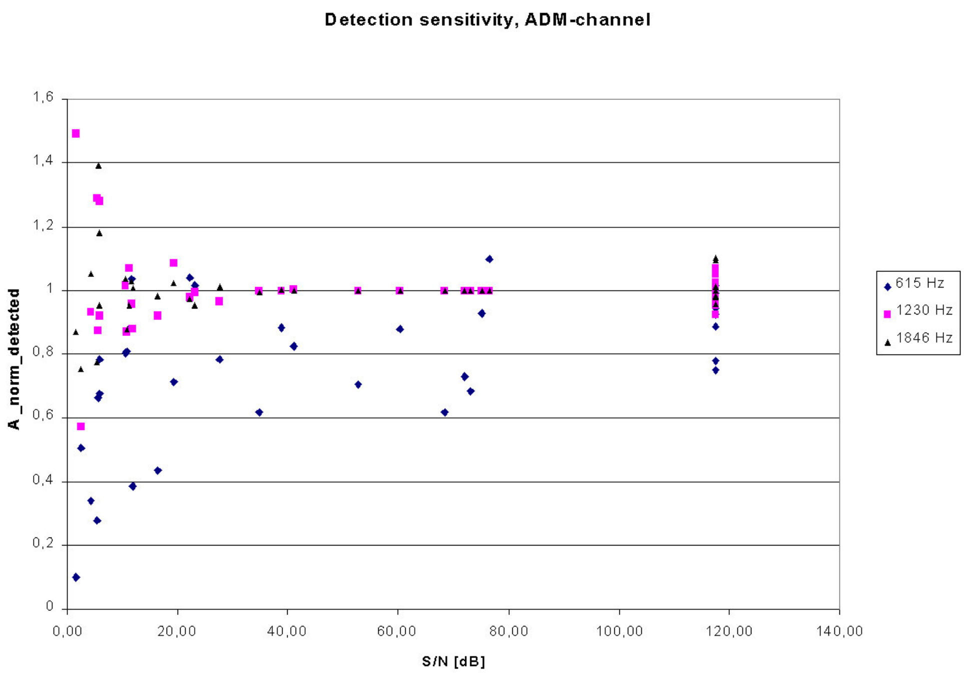


Figure 5.20 Error performance of DFT soft detection in presence of granular and Gaussian noise

Error performance of a 26-point DFT detection is a simulated result. The normalized amplitude sensitivity is disturbed by the granular noise all the time. The adaptive delta modulation proces sets the limits for the amplitude selctivity in the selection of modulation method. The qualitative result in Figure 5.20 gives basic variations in the received amplitude, which is at least 10% at S/N=120 dB. AWGN noise increases the variation of the recived signal at S/N= 10 dB to about 40%.

### *5.9.3. Frequency Deviation Sensitivity of the Soft Signal Detection*

Simulation settings in Figure 5.21are:

- Granular is not present.
- A three tone signal is used f=615, 1230 and 1846 Hz, A=0,5 V.
- N=26 samples and sampling rate is 16000 giving 615 Hz selectivity of DFT.

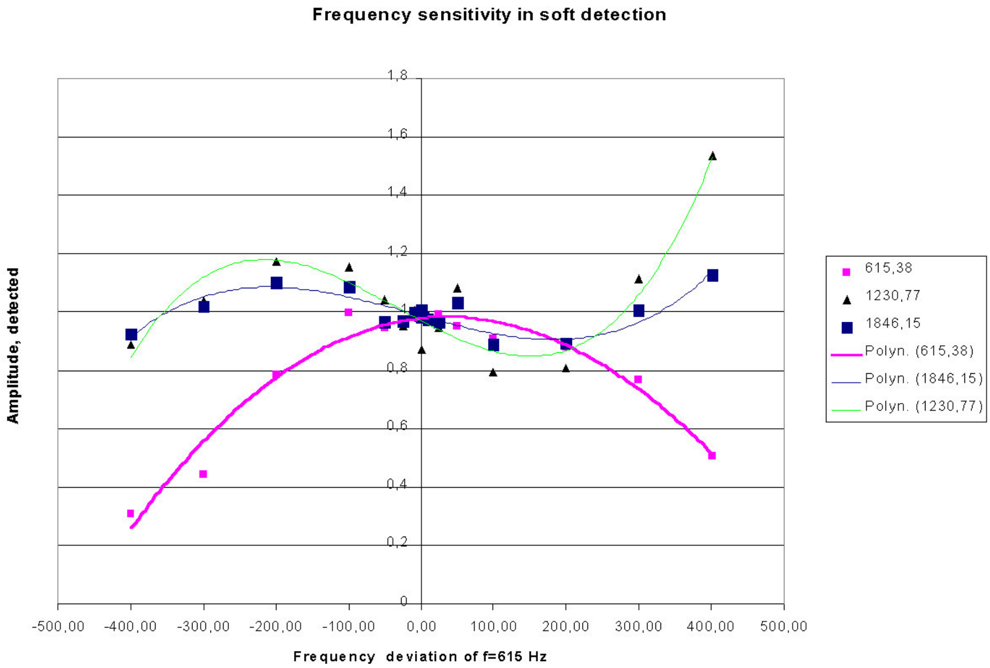


Fig. 5.21 Performance of DFT soft detection in function of frequency deviation

A three-tone signal of 615, 1230 and 1846 was used in the evaluation of frequency sensitivity of DFT detection. Frequency deviation of 300 Hz in the basic 615 Hz signal gives about 40 degrees error in the received signal value in detection. However, the soft detection is working well at the whole measured deviation area as seen in the values of other signal elemets (1230 Hz and 1846 Hz). The selectivity of the DFT filter is given by a sampling rate per number of samples, and is 16000/26=615 Hz.

The figure shows that the whole pass-band 615 Hz is usable and the soft detection is not very sensitive to frequency deviation of the carrier.

## *5.9.4. Frequency Deviation Sensitivity of the Soft DFT Detection in ADM-Channel*

Simulation settings in Figure 5.22 are:

- Granular noise is generated in the ADM-channel and it is present in the detected signal.
- The same three tone signal parameters as before are used f=615, 1230 and 1846 Hz, A=0,5 V and random phase.
- N=26 samples at sampling rate 16000 Hz gives 615 Hz selectivity of DFT.
- Detection of three tones is made with a 3-finger receiver using DFT.

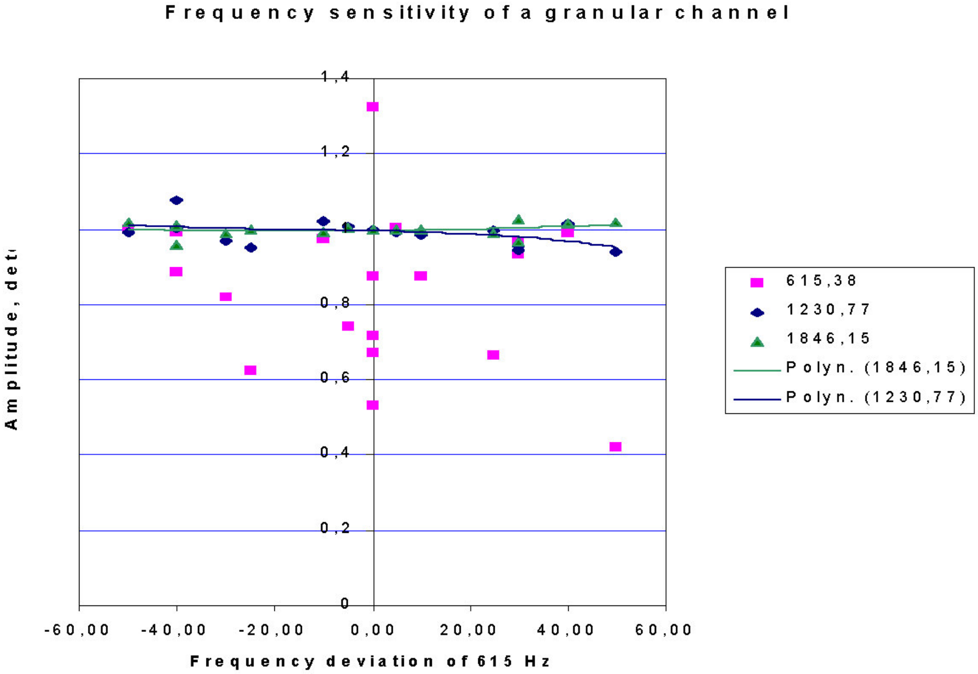


Fig. 5.22 Performance of DFT soft detection in ADM-channel in case of frequency deviation

The evaluated granular channel is not very suitable for analog data transmission. The granular channel introduces amplitude variations in the detected analog signal values (615.38 in figure). Reference analog signal components values (1230.77 and 1846.15 Hz) are not granular signals (thus only small variations in the amplitude). Reliable amplitude and phase detection of a multi-carrier data transmission system working in the ADM-channel is not possible. A MFSK system is one solution.

## *5.9.5. Sensitivity of the Soft DFT Detection in Granular Noisy Channel (ADM-channel)*

Simulation settings in Figure 5.23 are:

- Granular noise is generated in the ADM-channel and it is present in the detected signal.
- AWGN noise is added to the signal in the granular channel before D/A conversion in the ADM decoder.
- Analog waveform and the symbol value detection method is soft detection using a 26-point DFT.

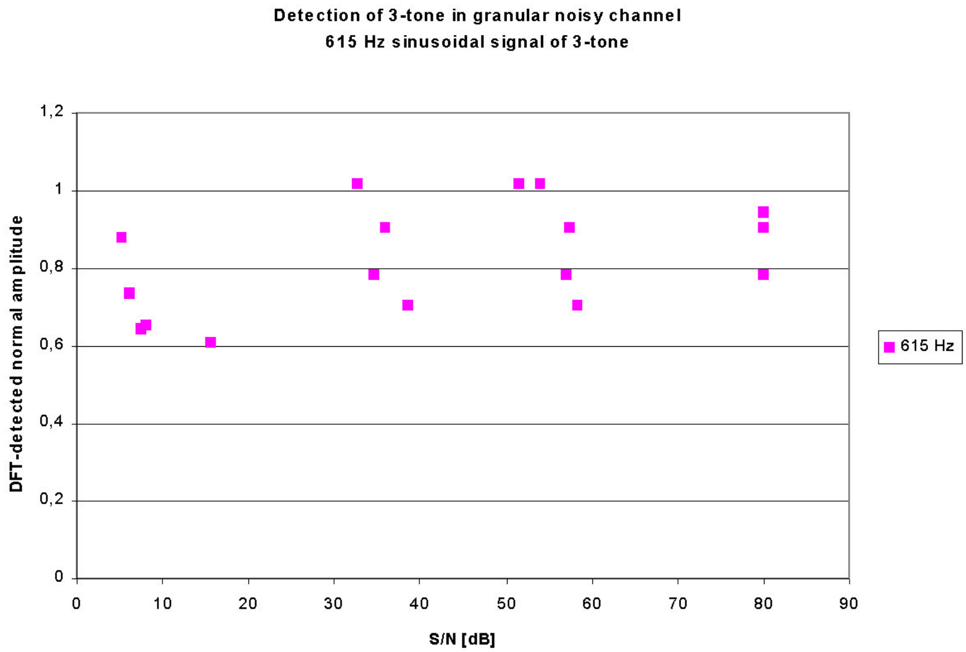


Fig. 5.23 Performance of DFT soft detection in AWGN noise ADM-channel

The evaluated granular channel was found to be not very suitable for analog data transmission. Figure 5.23 presents the amplitude variations of a three-tone component in respect to granularity and AWGN noise level. The granularity is the main source of variations. The signal amplitude values vary form normal to about -20 %. The increasing AWGN noise increases the amplitude variations slowly to about -40%.

### *5.9.6. Sensitivity of the Soft DFT Detection in Multi-Path Noisy Channel*

General simulation settings in Figure 5.24 are as before and:

- Granular noise is not present.
- AWGN noise is set on the noise levels S/N= 28…77 dB dB.
- Single-tone transmission (ASCII, QAM) is used as the signal.
- The reference received normal value used in the simulation is without granular or additive noise (at noise floor S/N>>80 dB) and without multi-path propagation.
- Multipath signals (I) are generated with additive signal components.

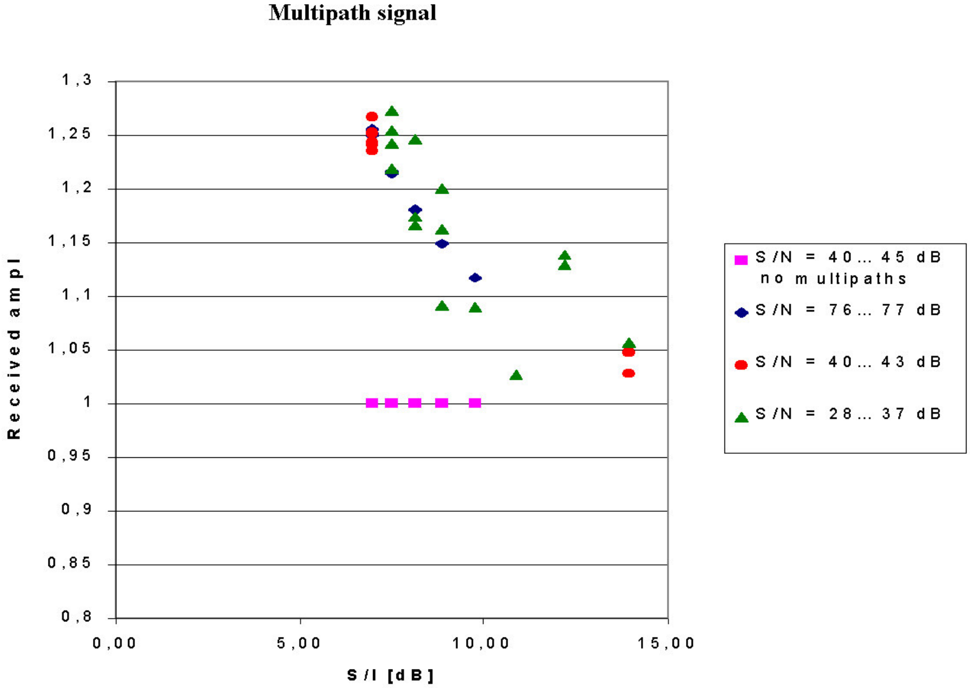


Fig. 5.24 Performance of DFT soft detection in multi-path noise channel

Figure 5.24 presents the received signal amplitude versus the S/I in the multi-path signal case. The multi-path components are made according to random phase and an exponential amplitude distribution. In the figure the mean power of all interfering multi-path components are calculated and the resulting S/I is used as a variable. In general, the first multi-path component is the main component of interfering signal power I of the S/I ratio. The reference signal is on the normal amplitude level, amplitude=1.

The multi-path error performance of the soft detection system is seen as the deviation of the amplitude compared to the normal amplitude value (ampl = 1). In the simulated results the multi-path signal varies with the received interference signal level. Error increases to 25% with the decreasing S/I levels 15…7 dB. The deviation in the variation is higher at lower S/N levels as expected. High S/N-values (76…77 dB) represent pure multi-path effects.

### *5.9.7. Orthogonal Signal Space*

Simulation settings in Figure 5.25 are:

- Granular noise is not present.
- AWGN noise is on the noise floor level S/N>>80 dB.
- The single-tone, two-tone and three-tone signals are generated as before in an IDFT-process, A=0.5 and random phase.
- Detection is made using 26-point DFT optimized to each frequency used in the multi-carrier signal.

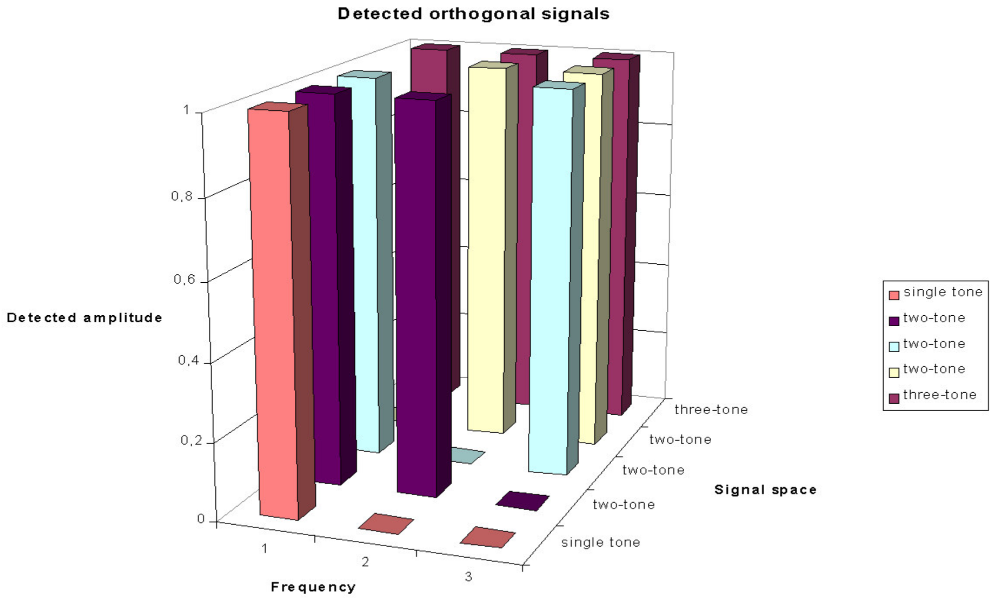


Fig. 5.25 Performance of DFT soft detection method

Figure 5.25 presents the performance of DFT in multi-carrier signal detection. Five different signal spaces are evaluated one-tone, two-tone and three-tone spaces. Each signal is detected with its normal amplitude value. No distortion components are found. The system is orthogonal and the detection method is matched to the signal frequency.

Errors in the soft detection are due to noise channel, granular noise generated in voice coding and parameter errors (frequency) in transmission. These situations have been discussed earlier and presented in Figures 5.19-5.24.

# Chapter VI

## 6. Summary

A new adaptive method for data transmission over different radio or telecommunication channels has been developed. The results are based on the modeling and simulation system, described in chapters 4-5, and the use of DFT (discrete Fourier transmission) in the soft generation and detection of waveforms. It is a band-limited variant of a generally known OFDM technology. The basic element is the adaptive modem, which has a standard electronic interface to supported communication systems and software algorithms for selecting its own functionality (synchronization, waveform, modulation etc). The basic theory used was made by Shannon, Fourier and Chang.

The complex waveforms generated in the simulations and in a prototype modem are the practical solutions for the capacity limit of Shannon's theory. The selection of adaptive waveforms have been simulated and tested in the field in 2000, which ended the investigation of the method by proving its functionality in practice.

The main result and benefit of this work is a theory and an early version (prototype modems) of the adaptive band-limited multi-carrier data communication system for alert communications, telemedicine purposes and other security communication needs on the physical level. The including of a warning multi-tone into a broadcasted waveform is proof of the effectiveness of a steganographic method used in the signal space.

Data communication is improved with the presented adaptive waveforms and the secure adaptive communication (modulation) method compared to known band-limited methods as

- We have the physical OSI level security in local networks.
- We have secure band-limited end-to-end voice channels.
- We can optimize the bandwidth versus bit rate at the selected quality level (S/N).
- We can manage bit error rates in communication with adaptive selection of waveforms (modulation method).

These results are simulated and tested in the field proving that the goals of the thesis have been achieved in general with the presented adaptive data communication method.

# List of Original Papers

This thesis is based on the following publications of the author.


1. [Lal97a] Lallo, P., Analysis and Measurements of Information Transmission over an Adaptive Delta Modulated Voice Channel, IEEE Military Communications Conference November 2-5, 1997, in *Proceedings of MILCOM 97*, Monterey, CA 1997, pp. 1057-1061.

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# Appendix 1: Mathematical background of some transforms

**A1 Definition of the Fourier Transform**

Definition of the class $L^p$ is as follows:
Suppose $1 \le p < \infty$. The function f on $(-\infty, \infty)$ is said to be of class $L^p$ (written $f \in L^p$) if

$$\int_{-\infty}^{\infty} |f(x)|^p dx < \infty$$

For each $f \in L^1$, the integral

$$\int_{-\infty}^{\infty} e^{ixt} f(t) dt$$

exists for all real x.

The Fourier transform $F$ of $f \in L^1$ formula is defined by

$$F(x) = \int_{-\infty}^{\infty} e^{ixt} f(t) dt \quad , \; -\infty < x < \infty$$

F(x) is continuous at x, which is shown using the Lebesque convergence theorem [Gol70].

Definition of the Fourier transform on $L^1 \cap L^2$ is also shown in [Gol70]:
It turns out that if $f \in L^2$ then the Fourier transform $F$ of $f$ is also in $L^2$ and $\|F\|_2 = (2\pi)^{½} \|f\|_2$, where $\|f\|_p$ is defined to be

$$\left(\int_{-\infty}^{\infty} |f(x)|^p dx\right)^{1/p}$$

The symbol $\|f\|_p$ is read as the $L^p$ norm of f.

**A2 Inverse Fourier Transform**

If we know that a function $F$ is the Fourier transform of some $f \in L^1$ we can determine the function $f$ from the values $F(x)$ of $F$. The inversion $f(t)$ is as follows [Gol70]:

If $F(x) = \int_{-\infty}^{\infty} e^{ixt} f(t) dt$

Then $f(t) = \dfrac{1}{2\pi} \int_{-\infty}^{\infty} e^{-itx} F(x) dx$

These equations may be written symmetrically by replacing *f(t)* with $\frac{1}{\sqrt{2\pi}} f(t)$ as

$$F(x) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\infty} e^{ixt} f(t) dt$$

$$f(t) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\infty} e^{-itx} F(x) dx$$

The functions F and f are called a pair of Fourier transforms i.e., F is the Fourier transform of f and vice versa. Such pairs are of great importance in the analysis of electrical impulses etc. The Fourier transform is valid for both periodic and non-periodic f(t). All signals encountered in the real world easily meet the requirements. [Mar62, Gol70]

**A3 Discrete Fourier Transform (DFT)**

The DFT is defined in references an operation on an *N*-point vector [x(0),x(1),..,x(N - 1)] as

$$X(k) = \sum_{n=0}^{N-1} x(n) W_N^{nk} \quad \text{, for k = 0, 1, 2, .., N -1}$$

where $W_N = e^{-j2\pi/N}$.

The operation is a transformation from the *N*-point vector in time domain to another N-point vector *X(k)* in frequency domain. The definition is interpreted as a frequency sampling of the discrete-time Fourier transform.

**A4 Inverse DFT (IDFT)**

The inverse DFT (IDFT) can be computed using a forward DFT algorithm. The formula for the IDFT is nearly identical to that for the forward DFt, except for a minus sign in the exponent and a factor 1/N as

$$x(n) = \frac{1}{N} \sum_{n=0}^{N-1} X(k) W_N^{-nk} \quad \text{, for k = 0, 1, 2, .., N -1}$$

A simplification for the part $e^{-j2\pi nk/N}$ using Euler's rule is

$$\cos(2\pi nk/N) - j\sin(2\pi nk/N)$$

The Euler's rule lets us state a more familiar form of DFT as

$$X(k) = \sum_{n=0}^{N-1} x(n)[\cos(2\pi nk/N) - j\sin(2\pi nk/N)]$$

and the inverse DFT (IDFT) as

$$x(n) = \frac{1}{N}\sum_{n=0}^{N-1} X(k)[\cos(2\pi nk/N) + j\sin(2\pi nk/N)]$$

Now we can calculate amplitude and phase from the complex value of *X(k)* as

$$X(k) = a + jb$$
$$|X(k)| = \sqrt{a^2 + b^2}$$
$$\phi(t) = \tan^{-1}\frac{b}{a}$$

To compute the Fourier Transform digitally we do perform a numerical integration. The result (DFT) is an approximation to a true Fourier Transform. A limitation is the finite time record of input signal (finite-length vector). The calculations are made at discrete points on the frequency and time domain. The frequency spacing of the result is the reciprocal of the time record length.

**A5 Fast Fourier Transform (FFT)**

The Fast Fourier Transform (FFT) is an algorithm for computing the Discrete Fourier Transform first described in [Coo65]. FFT is a fast algorithm for computing the DFT. FFT is used and described in references MATLAB [Bur94] and SPW [Com90].

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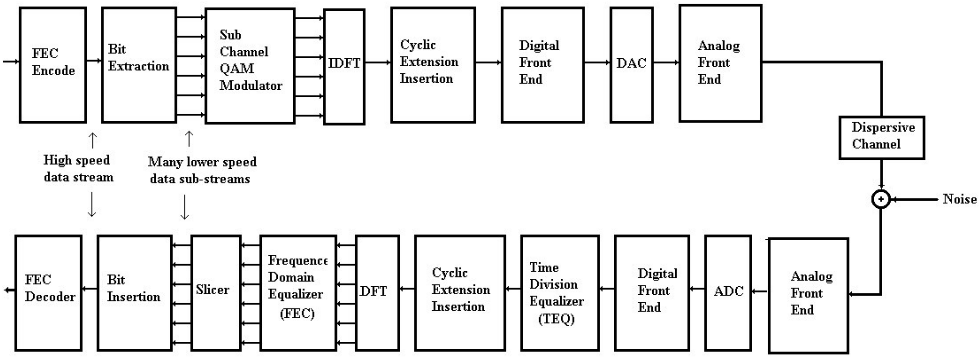
# Appendix 2: A high-level block diagram of a dmt/ofdm system




**Reference**

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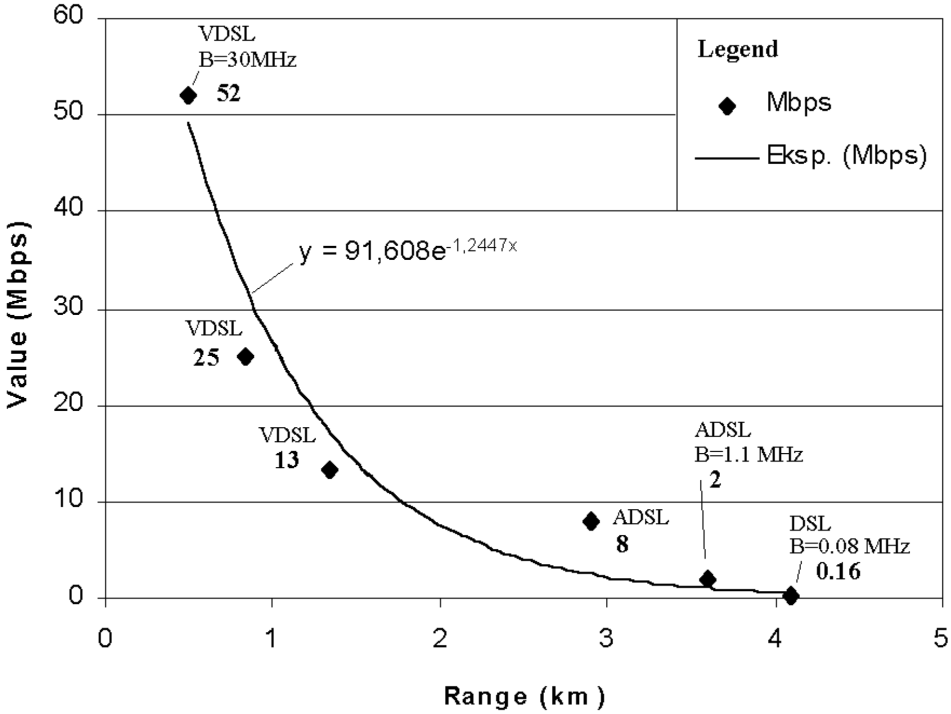
# Appendix 3: xDSL Capacity versus Distance

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