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Adaptive Coding and Modulation Techniques for HF Communication

Performance of different adaption techniques implemented with the HDL+ protocol

Martin Carlsen

Master of Science in Electronics
Submission date: June 2009
Supervisor: Lars Magne Lundheim, IET
Co-supervisor: Vivianne Jodalen, FFI

Norwegian University of Science and Technology
Department of Electronics and Telecommunications

Problem Description

In digital HF point-to-point defense communication systems, data rate adaptation schemes have been suggested based on either packet errors, bit errors or channel information estimated at the receiver. One such scheme, which is patented, is the HDL+ protocol for STANAG 4538.

The project shall investigate alternative solutions for the adaptive parts of the HDL+ protocol avoiding patented technology. A performance analysis of these solutions shall be performed using simulation, and the results should be compared with the existing scheme.

Assignment given: 29. January 2009
Supervisor: Lars Magne Lundheim, IET

Preface

The following report is a result of my Master's thesis work at the Norwegian University of Science and Technology (NTNU). The assignment was given by the Norwegian Defense Research Establishment (FFI) as part of an investigation around a new protocol proposed for ratification in NATO. The area of research, in particular, has been the implementation of different adaption techniques for the wireless channel. The work done has been a motivating, but has also imposed some challenges for me to overcome with the help of my advisors.

Hereby, I would like to express my special thanks so my advisors, Lars Lundheim at NTNU and Vivianne Jodalen at FFI, for their guidance and support. I would also like to thank Roald Otnes for technical help, and fellow student, Jon Even Corneliussen for his esthetical reflections along the way.

Martin Carlsen
June 2009

Abstract

The main goal of this thesis is to present two good alternatives for the HDL+ protocol proposed for ratification in STANAG 4538, as this partially is restricted by a patent claims.

The HDL+ protocol is used as a starting point, and in order to accommodate for the patented parts, the adaptive process is altered, and the code combining process is removed for the highest rate.

For simplifying the comparison between the performance of the proposed protocols, and the HDL+, both proposed protocols is simulated in a MATLAB environment, over the same channels as Harris has presented the throughput capabilities of the HDL+. These channels include the AWGN, single tap channel with flat fading, the ITU-MLD channel, and the ITU-MLD channel with Long- and Intermediate- Time SNR variations.

By analyzing the results, it is clear that the current implementation of the proposed protocols does not achieve as high throughput as the HDL+, but there are indications that there is potential for better results if further development is performed.

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Abbreviations

2G	Second Generation
3G	Third Genereation
ACK	Acknowledge
ALE	Automatic Link Establishment
ALM	Automatic Link Maintenance
AR	Auto Regressive
ARQ	Automatic Repeat Request
BER	Bit Error Rate
bps	bit per second
BPSK	Binary Phase Shift Keying
BW	Burst Waveform
CRC	Cyclic Redundancy Check
dB	deciBel
DRC	Data Rate Change
EOT	End Of Transmission
FEC	Forward Error Correction
FER	Frame Error Rate
FER-DRC	Frame Error Rate - Data Rate Change
FFI	Norwegian Defense Research Establishment
FSK	Frequency Shift Keying
GPS	Global Positioning System
GSM	Global System for Mobile communications
HDL	High Throughput Data Link
HDL+	High Throughput Data Link+
HF	High Frequency
IPR	Intellectual Property Rights
ITU	International Telecommunication Union
ITU-MLD	ITU- Mid-latitude Disturbed Conditions

ITV	Intermediate-Term Variations
LDL	Low Latency Data Link
LSSE	Least Sum of Squared Errors
LTV	Long-Term Variations
LUT	Look Up Table
NATO	North Atlantic Treaty Organization
NTNU	Norwegian University of Science and Technology
PSK	Phase Shift Keying
PU	Participating Unit
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
R-DRC	Receiver Data Rate Change
RSVD	Reserved
SNR	Signal to Noise Ratio
STANAG	Standard Agreement
TX	Transmission
UMTS	Universal Mobile Telecommunications System
WF	Waveform
WLAN	Wireless Local Area Network
xDL	Data Link Protocol

Chapter 1

Introduction

Wireless data communication has become a necessity in all layers of the community. Cellular phones and wireless routers can be found whereever you go, and the most common standards in Norway is GSM, UMTS and WLAN. These standards achieve high data rates up to a few kilometers in length, but is dependent on base stations, routers and infrastructure in order to communicate over longer distances.

The necessity for long range wireless data transmission is also found in the armed forces, but the availability of base stations and infrastructure are often restricted in a military environment. The source of information might be far at sea, or in hostile territory, where the use of landlines might impose a security risk. In order to communicate over longer distances, the armed forces within NATO communicate over the *High Frequency* (HF) band which is suited for transmissions up to several 1000 km without the need for relay points and infrastructure. The reason why HF communication is highly effective over long distances is the advantages the short waves (10-100 meters) has in combination with the properties of the ionosphere. Under the right circumstances the ionosphere can be used as a giant mirror, reflecting the waves back to earth, effectively increasing the range far beyond line of sight. As a consequence of long distances and the fluctuating nature of the ionosphere, the HF radio medium has a number of key challenges to overcome. These include low *Signal/Noise Ratio* (SNR), long multipath delay and fading channels. The data link protocols currently in use (see section 2.2) are quite robust in order to cope with these channel conditions, but does not achieve very high throughput.

In order to reach a desired goal of high performance in this varying channel, a system which is able to adapt the robustness is needed. This is often done by *Data Rate Change* (DRC). Harris, which is one of the leading vendors in developing radio systems for military HF communication, has proposed and patented a new data link protocol, the *HDL+*, which utilize DRC for data transfer over the HF channel [1]. The HDL+ is based on the earlier data link protocols, *xDL*, described in section 2.2 and in [2].

1.1 Problem Statement

One of the improvements in the proposed HDL+ protocol, described in section 2.2, is to take advantage of the already existing return channel used for *Acknowledge* (ACK) and *End Of Transmission* (EOT) messages. ACK-messages are sent from the receiving *Participating Unit* (PU) to the transmitting PU after each received frame, and by adding a few extra bits in the ACK-frame, the system is able to transfer information about estimated SNR and fade rate. By using this information effectively, combined with a smart way of code combining retransmissions, Harris is able to achieve promising results with higher throughput than the existing *High throughput Data Link protocol* (HDL),described in section 2.2 at high SNRs. [3][4]

Despite the promising results, the suggested solution might still not be the very best option for NATO. The reason is the drawbacks of using patented technology in military standard protocols, where several vendors compete in performance and cost, to develop the best hardware solution. Clearly, Harris will have a great advantage in development, cost and maybe performance if their patented technology [5] is implemented in a *Standard Agreement* (STANAG). So in order to encourage development of efficient and cheap radio solutions, there is a need for fair competition between different vendors. One of best ways to achieve this is to exclusively use non-patented solutions in the STANAG. For this reason, alternative solutions to the patented adaption techniques of the HDL+ protocol, both with and without utilizing channel information will be proposed.

1.2 Structure and Goal of this Thesis

In this report, the performance of two alternative solutions for rate adaption in HDL+ is proposed. The first solution, the *Frame Error Rate-DRC* (FER-DRC) described in section 2.4, is based on information about errors in the received frame. The frame error rate in this setting refers to the rate of packets received in error, in each frame.

The other proposed solution, *Receiver-DRC* (R-DRC), utilize information from a channel estimator and process this in the receiving PU, described in section 2.4. The two adaption systems has been simulated in an earlier project performed at NTNU [6], but more like a proof of concept of the adaption techniques, simulated on the S4539 protocol, rather than on the HDL+ protocol. The project gave practical insights on the nature of adaptive systems, but the results were not comparable to other HDL+ results, as the S4539 platform does not resemble the HDL+ protocol very well in packet sizes and interleaver lengths, and because the long term channel variations were generated numerically by human made tables which were not statistically correct.

The goal of the project is to achieve a high throughput on the HDL+ protocol while avoiding patent violations. In order to find the best alternatives and to compare these to the existing protocols, the system is simulated in a MATLAB environment described in chapter 3. The simulation results for both alternatives are presented in chapter 4, and the results are discussed in terms of adaptivity and throughput, and compared with the HDL+ protocol. Discussions and conclusions about how relevant it is to implement the alternatives in the STANAG, is presented in chapter 5 along with some proposed further work, based on results and experiences gained during the project.

Chapter 2

Background and Related Works

This chapter contains basic information about adaptive radio systems with references to additional information about the subject. It will also provide the reader with some background information, the motivation for the development of the HDL+ and the protocols presented in this thesis.

At the end of the chapter some other works on the subject are considered and discussed, followed by a presentation of the two proposed alternatives to the HDL+ protocol.

2.1 System Design

The crave for higher data rates over long distances encourage continuous development of the STANAGS. The wireless standards are evolved as more experience is gained, and more technically advanced solutions becomes available. The goal is to achieve the highest possible *throughput*, to transfer as much error free data as possible, in the shortest possible time. As the channel continuously changes, both in short term and long term variation, there will be times where the channel conditions are poor, and in some cases it will be quite good. The variations caused by fast fluctuations and short term fading varies too fast for the adaption process, but the slower variations can be tracked and adapted to.

It is a challenge to decode the received symbols correctly when the channel conditions are bad, and to achieve any throughput at all, robust modulation schemes (few bits per symbol) and redundancy is required. As a result of this, the transmission requires a low data rate.

When the channel conditions are good, higher order modulation schemes (several bits per symbol) can be used. It might also be possible to send fewer redundant bits for the error correction, which also result in a higher data rate. A typical radio system is illustrated in Figure 2.1 and the most common adaptive parts, *Forward Error Correction* (FEC), interleaver and modulation, is explained. For more information about the design and function in a radio transceiver, the reader is encouraged to read more about this in [7] and [8].

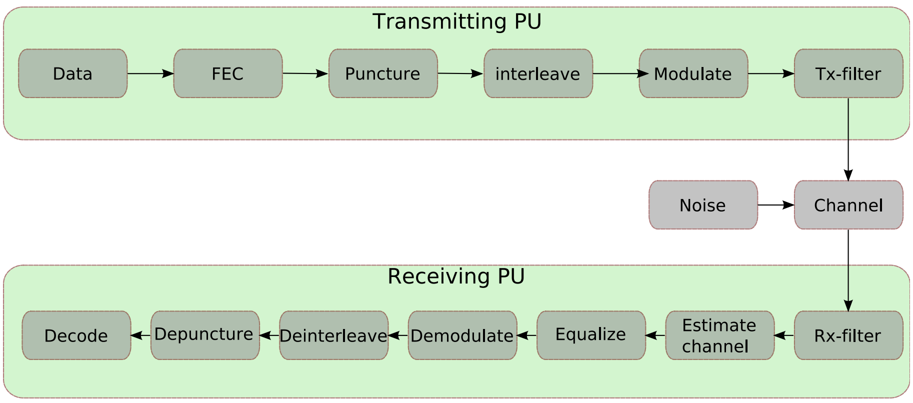

Figure 2.1: A typical radio system. Both the transmitting and receiving PU are illustrated as several modules with different purposes.

Forward Error Correction

In order for a system to be able to correct errors produced by a noisy channel, the information is coded with an error correcting code, adding redundancy to the data before transmission. As redundancy is added to the system, the bandwidth efficiency drops, but the overall throughput is often increased, as it results in a lower *Bit Error Rate* (BER) and fewer retransmissions. How much redundancy needed, is channel dependent, so the code rate in a system is preferably adjustable. There are several ways to encode and decode information, both for compression and redundancy, but the encoder used in the radio systems in this report is a convolutional encoder described in detail in [2] and in order to change the code rate, some bits are removed prior to transmission by a *puncturing frame* and reapplied prior to decoding. [7] [9].

Interleaver

Error correcting coding usually performs bad when many consecutive bits are received in error. This is quite unfortunate as the nature of the wireless channel behaves in exactly this way. Errors often occur in bursts, short or long, depending on the channel conditions. In order to prevent several successive errors caused by a channel fade, the coded bits are shuffled prior to transmission, separating successive bits as far as possible, hopefully longer in time than the channel fade [10]. As the coded bits are buffered in a register to enable shuffling, a latency occurs which slows down the transmission process. A longer interleaver can cope with longer channel fades, but also induces more latency, which is why the interleaver length preferably should be adaptable. In the HDL+ protocol reviewed in this thesis, the interleaver length for each combination of data rate and packet size is described in the protocol. This means that the interleaver length follows the choice of modulation and packet size and is thereby not adaptable in its own, but as a consequence of other adaptive parameters.

Modulation

In order to transfer the coded bits, it is necessary to map them as symbols which can be sent over a physical medium. There are several different ways to do this, but they all map one or more bits to a symbol prior to transmission. As more bits are resembled in each symbol, the data rate increases, but as the symbols now appear more similar to each other, they are also harder to demodulate without error. For more information about different modulation techniques, [7] and [8] are recommended reading

There are several ways to adapt the rate to the current channel conditions, and the easiest way might be to measure the BER or *Frame Error Rate* (FER) and adjust the rate accordingly. The FER is the rate of packets containing errors in a frame. If there are no errors, the system will send at a higher rate and vice versa. This solution is quite simple and easy to implement but does not utilize any direct information about the channel. It should be possible to achieve better results by adapting the rate to information about the channel estimates, but as this is a more complex solution, it also introduces additional possible sources of error.

In an ideal system, the receiving PU will have complete knowledge of the current channel conditions and continuously adapt to the best suited data rate. This is not realizable, as the only available channel information is non-instantaneous estimates of what the channel has been.[11][10]

2.2 Military HF Radio Systems

In this section, some background information about the development of the automatic radio system for HF is provided. As technical descriptions of the different protocols are not the goal of this project, this is left to be covered in the STANAGS [2][12]. The major differences between the data link protocols, and the motivation for the development of HDL+, is considered important for the reader and is thereby described in the following sections. Some figures from the HDL+ protocol are implemented to ease the understanding of what is omitted in the simulation model.

xDL

xDL is the common notion for the two data link protocols defined in S4538, the *High Throughput Data Link* (HDL) and the *Low Latency Data Link* (LDL). Both utilize synchronous *Automatic Link Establishment* (ALE) and *Automatic Link Maintenance* (ALM) which enables a simple but effective variation of *code combining* in the ARQ process in order to achieve higher throughput in difficult channel conditions. If a packet is received in error, the retransmitted packet will be coded with an alternate phase of the encoder. This enables the receiving PU to decode the packet by using information from both the sent phases, effectively reducing the probability of error in the retransmitted packet. The utilization of code combining in the retransmitted packets is called *ARQ type II*. [2][13]

HDL

The HDL is the preferred protocol for transmitting large amounts of data over good channels. Prior to transmission, the data to be sent is divided into packets of a given size. The number of packets contained in one transmitted frame in the HDL protocol is designated by a number attached to the protocol name e.g. HDL_24 will transmit 24 HDL packets (233 bytes in one packet). Available frame sizes for HDL are 3, 6, 12 and 24 packets. The receiving PU decodes eachpackets separately, and is able to send an ACK message with information about which packets contained errors (selective ACK). This enables retransmission of failed packets only. The maximum data rate, not considering protocol overhead, is 4800 bps (bit/s) which gives an approximate throughput of 3200 bps when the highest amount of packets (24) are sent in each frame. [2][13][14]

LDL

The LDL protocol is more robust than the HDL and is better suited for tougher channels and/or smaller amounts of data. The amount of bits to be sent is also designated by a finite set, but the designator in LDL, defines how many bytes are contained in one packet, and only one packet is sent in each frame. The size of the LDL transmission frame can vary from 32 bytes (LDL_32) to 512 bytes (LDL_512). As only one packet is sent, no selective ACK is available in this system, and if an error occurs, the whole frame is retransmitted. The maximum throughput for LDL is approximately 500 bps. [2][13]

Development, 2G vs. 3G

As new protocols and standards are developed, they are often sorted in groups of generations, classified by the technology used. STANAG 5066 (S5066) and S4539 are commonly defined as second generation (2G), while the newer standard, S4538, containing the xDL protocols, are defined as third generation (3G) automatic radio system.

Table 2.1: Comparison of 2G vs 3G automatic radio systems at HF.[13]

Automatic HF 2G	**Automatic HF 3G**
(S5066 & S4539)	(S4538)
Modular, different functionalities may be located at different pieces of hardware.	Integrated, all functionalities located in the radio
Asynchronous calling, no GPS time reference, gives longer call times	Synchronous calling, uses GPS time reference, gives short call times for members of the net
Linking using 8-FSK, not particularly robust at low SNRs	Linking using 8-PSK and Walsh-functions, very robust at low SNRs
Data rate adaptation based on an explicit change of waveform	Data rate adaptation based on adapting the code rate (code-combining)
Can utilize high data rate waveforms (up to 12800 bps) defined in STANAG 4539	Is limited to a maximum data rate of 4800 bps defined in the STANAG
Offers a point-to-point service and a broadcast service for both packet and circuit switched data	Offers a point-to-point service for packet and circuit switched data and a point-to-multipoint service for circuit switched data only
Allows a more flexible frame size of forward transmissions, throughput efficient	Finite number of forward transmissions sizes, less throughput efficient

In the the 3G system, robust transmission and fast channel setup is prioritized, and as seen in Table 2.1, this has resulted in a lower maximum, data rate (4800 bps) than in the older S4539 system (12800 bps). As a consequence of the lower maximum data rate, Harris corporation has proposed the new data link protocol, HDL+, for ratification.

HDL+

One of the key improvements in the HDL+ protocol is the combination of the high data rate *waveforms* from S4539 or MIL-STD-188-110B listed in Figure 2.2 ,and the code combining techniques in the xDL data link protocols, achieving a maximum throughput of up to 10000 bps in a 3kHz channel.

Each HDL+ forward transmission begins with an informational header transmitted in the more robust *Burst Waveform* 6 (BW6),described further in section 3.1, enhancing the probability of correct reception. A burst waveform defines a way to transmit a set of data and is described in the STANAGS. S4538 describes a total of seven Burst waveforms for different usage. A burst waveform can consist of several combinations of modulation and code rate, which are referred to only as waveforms. The actual data transmission of the HDL+ is performed with BW7, described in Data Rates on page 13 and illustrated in Figure 3.1. BW7 consists of seven different waveforms, and two packet sizes (280 or 568 bytes). In order for the receiving PU to demodulate and decode correctly, the waveform and packet size is described in the header.

Header

The header is 51-bit long and is transmitted in 386.67 ms. As seen in Figure 2.2, the header contains 3 bits to distinguish the header from the ACK messages transmitted with the same Burst Waveform (BW6). The source address is a 10 bit individual code, identifying the transmitting PU. One of the most appreciated functions of the header is the information contained about modulation, code rate and packet size in the following data transmission, enabling the receiving PU to decode instantly, without any negotiation of data rates, which has been common in earlier systems [2]. This solution enables the transmitting PU to instantly adapt any of these parameters (modulation, code rate and packet size) between successive frames, making the system highly adaptive to varying channel conditions. The header also contains an estimate of the SNR in the return channel, information about which packets in the frame are to be sent and a 12-bit *Cyclic Redundancy Check* (CRC) to validate the information.

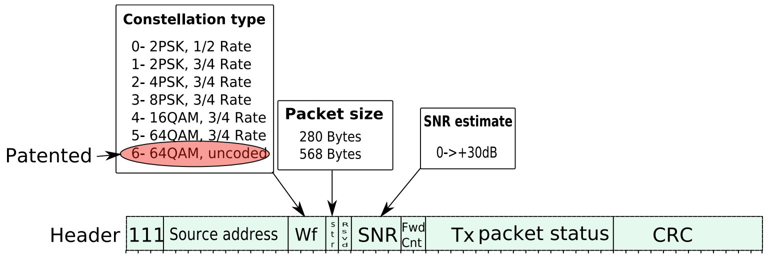

Figure 2.2: HDL+ Header with patented parts highlighted [1][2]

ACK

The return message, for acknowledging the received frame, is also an important part of the adaptive system patented by Harris. There are some similarities to the header, but there are also some fundamental differences. The ACK contains status about which packets were received correctly and estimates of the fade rate and SNR in the forward channel. This means that the transmitting PU knows which packets to retransmit and how the channel conditions are, so it can adapt the modulation and code rate accordingly. This is one of the patented parts of the HDL+ protocol, and is highlighted in Figure 2.3.

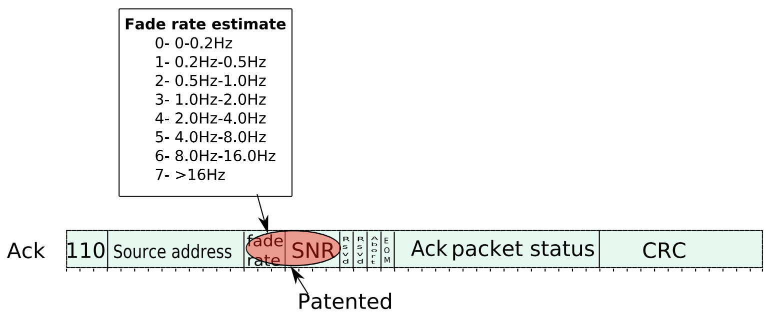

Figure 2.3: Acknowledge message for HDL+. Patented parts are highlighted.[1][2]

Data Rates

The data rates are presented in Table 2.2 and should give an efficient throughput from 1-10 kbps. For information about the modulation, the reader is encouraged to read [12] as these are implemented according to the standard.

Table 2.2: Data rates utilized in BW7 data transmission

Data Rate [bps]	Signal Constellation	Convolutional FEC	Scrambling
1000	BPSK	R1/2	Symbol
1600	BPSK	R1/4 (Punctured)	Symbol
3200	QPSK	R1/4 (Punctured)	Symbol
4800	8-PSK	R1/4 (Punctured)	Symbol
6400	16-QAM	R1/4 (Punctured)	Bit
9600	64-QAM	R1/4 (Punctured)	Bit
12800	64-QAM	R0 (Uncoded)	Bit

As illustrated in Figure 2.4 the Burst Waveform 7 (BW7) consists of an initial probe sequence of 64 symbols followed by a data sequence of 256 data symbols alternated with 32 symbols long mini-probes for channel estimation (section 3.3)[2].

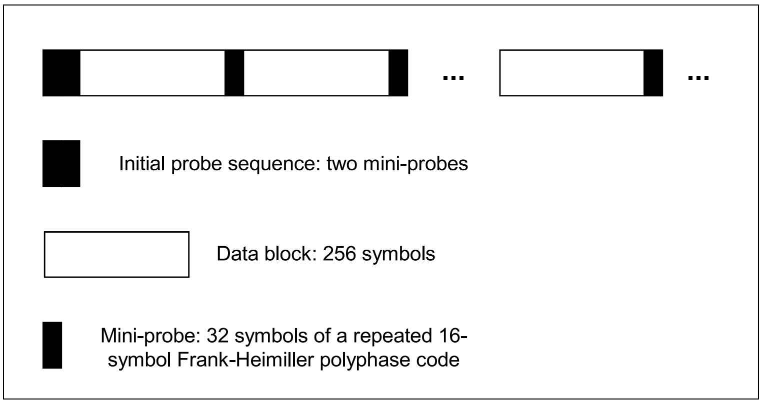

Figure 2.4: The total TX-frame in BW7 consists of an initial probe sequence and several data blocks of 256 data symbols, divided by 32 symbols long mini-probes for estimation.[2]

IPR-FREE

As it is quite obvious that a patented system is not preferred in a NATO-standard, the developers at Harris has also proposed an alternative version of the protocol. The solution is known as IPR-free HDL+ and is supposed to be interoperative with the patented HDL+. This basically means that Harris is able to produce radios with the patent and the advantages that comes with it, while other vendors can produce radios able to communicate with the HDL+ radios, but at lower rates. The IPR-free solution is not allowed to communicate channel information in the ACK message or utilize the highest data rate. By following these rules, the maximum data rate of the IPR-free solution is limited to 7900 bps while the patented version radios will be able to communicate at up to 10.000 bps. The limitations of this alternative solution is so comprehensive that it will give Harris a great advantage as a vendor, making it a non-ideal solution for ratification in NATO [1][4].

2.3 Related Works

During the initial stages of the project, several other works on adaptive algorithms are investigated. There is not a lot of published material on the issues of adaption techniques on the HDL+, but there are some interesting works on data rates from S4539 in combination with S5066. Much of the work is focusing on the selection of packet sizes, frame sizes and interleaver lengths, which not is an issue for the HDL+, but there are also some recommendations on how to adapt the data rate, and suggestions for further work on adaptive ARQ protocols.

The typical issue when investigating these earlier works, is that they focus a lot on details which are not valid for the HDL+ protocol. A good example is Trinder and Browns three main requirements for an effective data rate change algorithm, as described in [15]:

- Optimization of data throughput to the prevailing channel conditions.
- Avoidance of unnecessary data rate changes (e.g. "oscillations" or upward changes in data rate followed immediately by downward changes in data rate, with consequent losses in overall throughput).
- Robustness (e.g. the algorithm should not attempt to change to a data rate higher than the channel conditions can support, as this could cause failure of the data rate change mechanism and a loss of throughout whilst the link is re-established.)

When developing an alternative DRC algorithm for the HDL+ protocol, these requirements are quite obsolete. The avoidance of unnecessary data rate changes, is not as important when code combining is added to the system, and some of the information can be utilized by later packets. The oscillations were also unwanted because of the extra redundancy added to the system when changing the data rate, but this is not an issue in the HDL+ protocol. As the header of the frame is sent with BW6, regardless of the data rate chosen, re-establishment of the channel will not be an issue as a consequence of the data rate chosen, making the requirement for robustness less important.

Some of the most interesting publications on DRC algorithms are written by Trinder and Gillespie [16], Nieto [17], and Schulze [18] who worked on combining the data rates from S4539 and STANAG 5066 with optimum interleaver and data rate selection, as this is the same data rates utilized by the HDL+. All seem to conclude with the best DRC algorithm based on packet errors is to decrease of data rate when the frame error rate exceeds 50% and an increase when the frame error rate is below 20%.

Trinder and Brown also concludes by stating that a DRC algorithm based purely on frame error statistics will occasionally fail (regardless of the choice of error rate threshold) because of the highly variable nature of the HF propagation characteristics. From these conclusions, they recommend further analysis of data rate changes based on estimation of additional channel quality measures, such as SNR estimates, in combination with the frame error statistics.

Some of these recommendations are followed in this project, but the threshold for increasing the data rate is changed to 10%, as this resulted in better results on simulations performed. The recommendation of utilizing more channel data is also investigated, and the two algorithms are compared in chapter 4.

Work on analyzing performance on the HDL+ is well covered by Chamberlain and Furman in [4] and [14]. These publications provides performance measures for the utilization of S4549 waveforms in combination with the data link protocols on S4548, while the work done in [19] illustrates the adaptive properties over varying channel conditions. The later has been used as a reference throughout this thesis, both for channels to use, presentation, and performance results for comparison.

2.4 Proposed Solutions

The focus in designing the alternative DRC systems, is to avoid infringing the patented parts of the HDL+ protocol [5] and to achieve as high throughput as possible. I am not educated in patent law, but to my understanding the two system designs presented, fulfills the required design goals. The proposed solutions are simulated in order to tweak for best performance and for comparison with existing systems. The results of these simulations are presented in chapter 4. For more information about the basic idea behind the two proposed solutions, with figures, the reader is encouraged to read [6].

R-DRC

In order to adapt the system to varying channel conditions without transferring information about the channel in the ACK message, it is possible to process the channel information at the receiving PU, only returning information about which rate to use in the next forward transmission, hence the name *Receiver-DRC*

The processing might be done by a simple *Look Up Table* (LUT), described in section 3.5, or by advanced formulas, but in order to encourage continuous development it should be up to different vendors to develop the best system for channel information processing. In this project, R-DRC is investigated by using a LUT in the receiving PU for adaptivity. Some other adjustments to the HDL+ system are incorporated in order to simulate the system without patent violations and to compare the results with other possible solutions and existing HDL+ results.

FER-DRC The other proposed solution is to adapt the system purely on basis of the amount of packets received in error (FER). As this information is already contained in the ACK message, it requires no changes to the existing HDL+ protocol, and because it is not utilizing any channel information, it will neither be infringing the patent [5] in any way. This solution is referred to as *Frame Error Rate DRC*, FER-DRC. Earlier implementations of this principle of adaption, has often experienced unwanted oscillations in the data rate selection. This has been quite a problem since the throughput goes drastically down when retransmitting and because change of data rate often has caused a delay in the system. Data rate change in the HDL+ system does not apply any delay to the system, and the code combining techniques of the ARQ type II, makes the consequence of a wrong choice less significant. Still, an oscillating effect in the packet size would spoil the benefits of the code combining, as it is difficult to combine packets of different sizes. For this reason, packet sizes are not adaptive in the FER-DRC. It should be quite easy to implement this relatively simple adaption protocol, but it might require some effort in designing the adaption strategies and rules in order to achieve the most stable system with the highest throughput over different channels. If this protocol is to be implemented in a STANAG, the principle should be specified in the protocol and the different ways to solve it should be vendor specific. The principle is simulated and tested in the same environment as the R-DRC system, and is developed in order to achieve as high throughput as possible over different varying channels. The choices made for the adaptive algorithms, and suggestions for further work is presented in detail later in the report.

Chapter 3

System Model and Specifications

For the analysis of the different adaptive systems, our simulation model is based on a MATLAB code developed by Roald Otnes at the *Norwegian Defense Research Establishment* (FFI). The code is originally developed in order to simulate the robustness of S4539 waveforms in different channels when implementing a Turbo Equalizer at the receiver. As the waveforms in the original simulation environment are pretty much the same as the waveforms of HDL+, this is a good starting point. The changes implemented, is the different DRC algorithms, a time varying channel, packet sizes, and ARQ techniques as described in the HDL+ protocol. As some assumptions and simplifications are done throughout the simulation process, the HDL+ protocol is not exactly replicated. So in order for others to replicate the results, the simulation setup which is not implemented according to the STANAG is described in detail.

3.1 Model Specifications and Assumptions

The simulated radio system corresponds to Figure 2.1. The data to be sent are randomly generated and then encoded by a 1/2 rate convolutional encoder with six states [7][12]. In the 3/4 rate waveforms, the coded bits are punctured after coding, which means that three of the six bits from the encoder are removed prior to transmission, and reinserted as zeros before *soft decoding* is done. Soft decoding is a smart way of using information from the turbo equalizer used in this simulation. Each coded bit is represented by a floating positive or negative value, as opposed to regular decoding where each bit is simply declared positive or negative. When the coded bits are assigned in float, they also contains information about the certainty of each bit, enabling the decoder to weight certain bits more than uncertain bits in the decoding process. As the reinserted bits are valued zero, they are the most uncertain, and does not add any information to the decoder. If a packet is received in error, the packet is retransmitted with another puncturing frame. The received soft decisions from the equalizer are added to the previously received soft bits, effectively increasing the amount of information available in the receiving PU, possibly increasing the confidence level of the coded bits which are received correctly, and decreasing it for those received in error. As it is known that the first version of the packet contains errors, it should be assigned less certainty. In an attempt to increase the efficiency of the system, I have weighted the first and the second copy of the received packet differently. Saved information is weighted 0.3 and new information is weighted 1.

The code combining is not implemented on the highest rate, as the use of uncoded rate (12800 bps) in combination with code combining, is restricted by the patent. As an alternative, the highest rate is sent with no coding at all (the highest rate in HDL+ is coded and punctured to 0 rate, also called uncoded), only adding the information from retransmitted copies to the soft decision process, leading to a severe decrease in robustness for the highest rate.

Transmission Frame

The forward transmission in HDL+ is divided into frames, also called datagrams, as illustrated in Figure 3.1. Each frame contains a header, which is transmitted at BW6 for robustness, and a data section sent at BW7 for higher speed. These waveforms are described in detail in [2].

The BW6, which is used for the Header and ACK messages, is much more robust than the BW7, and the probability of error here is quite small compared to in the actual data transfer. Errors occurring in a BW6 message, will most probably occur in large numbers in the BW7 data packets and the packets will have to be retransmitted regardless of the BW6 error. By this argument, the header and ACK messages can be omitted from the simulation without affecting the end results, as long as the transmission time of the two 51 bit BW6 packets (2×386.667 ms) are added in the throughput calculation. [2]

The data section contains several smaller packets (up to 15) complemented by a CRC check and seven encoder flush bits for each packet. In the simulation environment, the error check is easily performed by comparing sent and received bits, so the CRC check is skipped for convenience. The CRC bits are sent as coded data, but are not accounted for in the throughput calculation. If there are fewer data bits to be sent than 15 packets, the minimum number of packets needed is sent, and the last packet is filled with random bits. The filled bits is not accounted for in the throughput calculation and the same procedure is used for each packet's count/sequence bits. (See Figure 3.1 for clarification.)

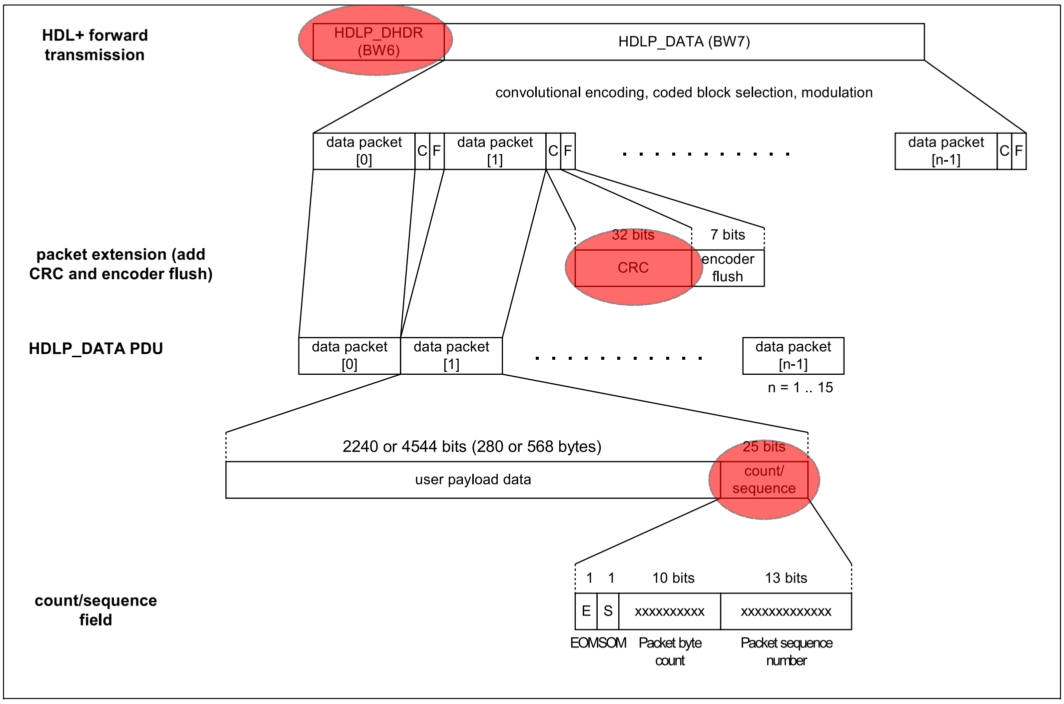

Figure 3.1: HDL+ Transmission frame divided in packets and combined with control bits of different functions. Highlighted bits are omitted in the simulations. The figure is from [20].

3.2 Channel

The original channel model implemented in the simulation environment is a standard Watterson Model described in [21]. This is a good model for simulating the Doppler and multipath spread which perturbs the signal over periods of a few seconds, but fails to implement the channel variations that occur over longer time periods. On the ITU recommended channels, as the ITU-MLD [22] are easily implemented by setting parameters parameters for delay spread and Doppler spread accordingly.

In order to analyze the adaptive properties of the proposed solutions, a channel with long term variations in the SNR is needed. Hence a model for *Long- and Intermediate Term Variations* (LTV & ITV) in the SNR is proposed by Furman et al. in [19] and [23]. LTV is the SNR variations occurring mainly below 0.01 Hz, and ITV resembles the SNR variations at higher frequencies (0.01 Hz-0.185 Hz, where 0.185 Hz is a result of the sampling frequency of the estimator). The SNR mentioned here, corresponds to mean Signal to Noise ratio in a 3 kHz bandwidth over a longer time period, during this time period there will occur short time fading, LTV and ITV, which varies the instant SNR.

The channel model, illustrated in Figure 3.2, implements both LTV and ITV to the Watterson model by generating two sources of Gaussian noise, and filtering these with two separate AR(1) filters (Auto Regressive), also referred to as alpha filters, with different time constants. The LTV and ITV gain control is implemented in order to adjust the spectrum of the channel SNR variation, and the high pass filter (5. order Chebychev) removes any unwanted "DC energy" [19] [23].

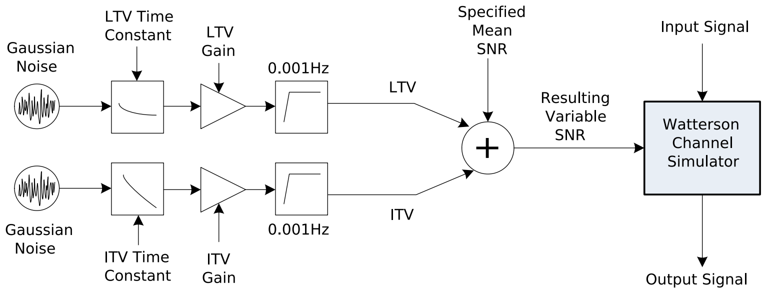

Figure 3.2: Implementation of LTV and ITV to the standard Watterson Model. Figure from [19].

The ITV and LTV simulation parameters implemented corresponds to Table 3.1. In order to compare the generated SNR variation with other channels, without the noisy variation, the channels' cumulative spectrum profile is generated from a 2048-point FFT ($s[i]$), and illustrated in Figure 3.3. The *Cumulative Spectrum Profile* $p[n]$ is defined:

$$p[n] = \frac{\sum_{i=1}^{n} s[i]}{\sum_{i=1}^{1024} s[i]} \tag{3.1}$$

Table 3.1: The Simulation Parameters used for generating the LTV & ITV.

Parameter	**LTV**	**ITV**
Std. Dev. (dB)	3.85	3.95
Time Constant (s)	180	5.2

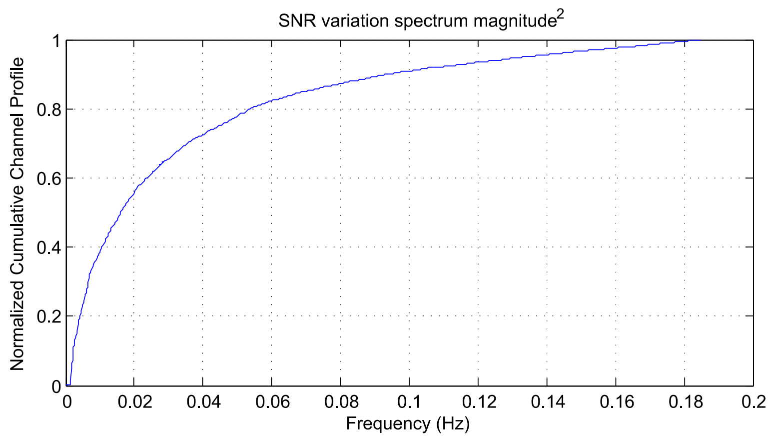

Figure 3.3: The channel profile (cumulative spectrum) for the LTV and ITV implemented in the simulation.

The resulting channel has a SNR variation around the mean SNR defined in the Watterson model both in short term, as illustrated in Figure 3.6, and longer term, as illustrated in Figure 3.4.

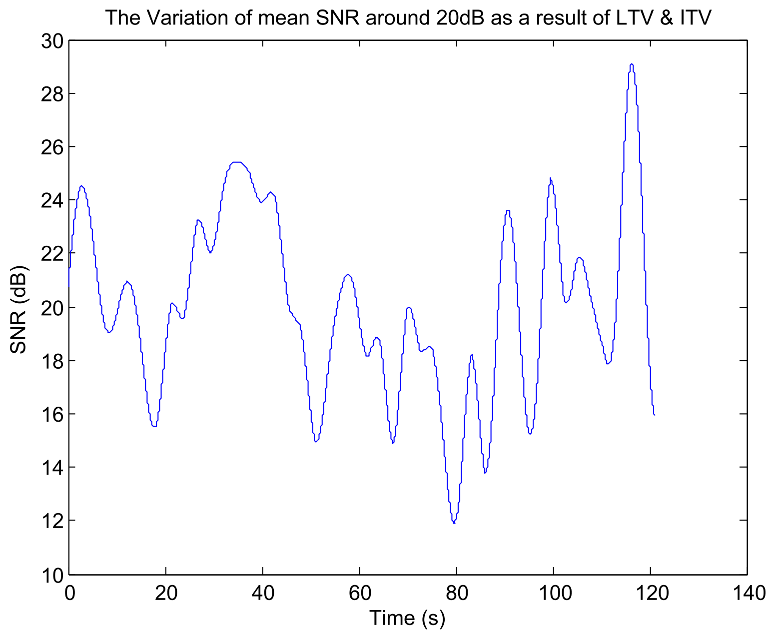

Figure 3.4: The SNR variation around a 20 dB offset as a result of LTV and ITV.

3.3 Estimation

In order for the equalizer to get the best results, the channel conditions are estimated prior to equalizing. The channel is estimated by analyzing the known probes in the transmitted frame (see section 2.2). By observing how these known symbols are distorted by the channel, the channel estimator is able to make a qualified guess of how the channel behaves between the probes by using linear interpolation. The algorithm used for channel estimation is called *Least Sum of Squared Errors* (LSSE), and is described in detail in [24].

SNR Estimate

The channel estimates is also used in the R-DRC process in order to generate and use the LUT introduced in section 2.4 and further described in section 3.5. The HDL+ protocol estimates SNR and fade rate in the receiving PU and transfers this in the ACK message, but instead of using estimates of the SNR in dB, the error variation (noise) σ^2 on the received mini-probes is calculated after removing the effects of the estimated channel variation. The σ^2 is used in both generating the LUT and for the adaption process, as this resembles the actual SNR quite good. The error variance estimates, for different SNRs, used for the LUT is illustrated in Figure 3.5.

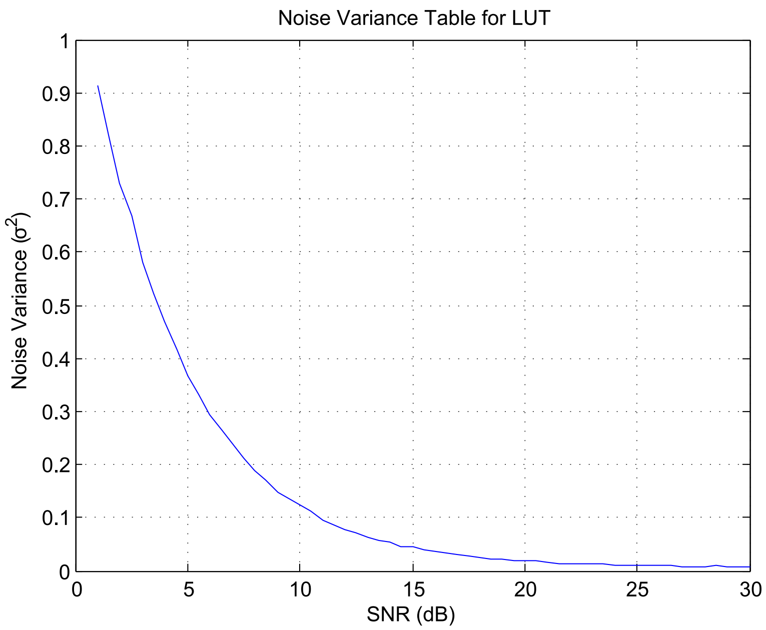

Figure 3.5: The Noise Variance used as Parameter for the LUT.

Performance in Terms of Adaptivity

The fade rate is calculated by counting the number of tops and bottoms of the absolute value of the estimated channel over a time interval corresponding to a frame length (illustrated in Figure 3.6). By dividing this number by the duration of the frame, and account for the estimated SNR, which affects the Doppler spread estimate, a good estimate is acquired.

The estimate does not correspond to the actual fading rate, but can be used as a parameter in the adaption process, as long as the LUT is generated with the same parameter. The accuracy of this method is limited, but it seems to work quite well in the simulations. Because of the low accuracy, there is no point in having a large LUT which requires long computation time in order to be generated statistically correct. Computation time has been a scarce resource during this work, but in a real life implementation, the size of the LUT would only be limited by the accuracy and confidence of the estimator.

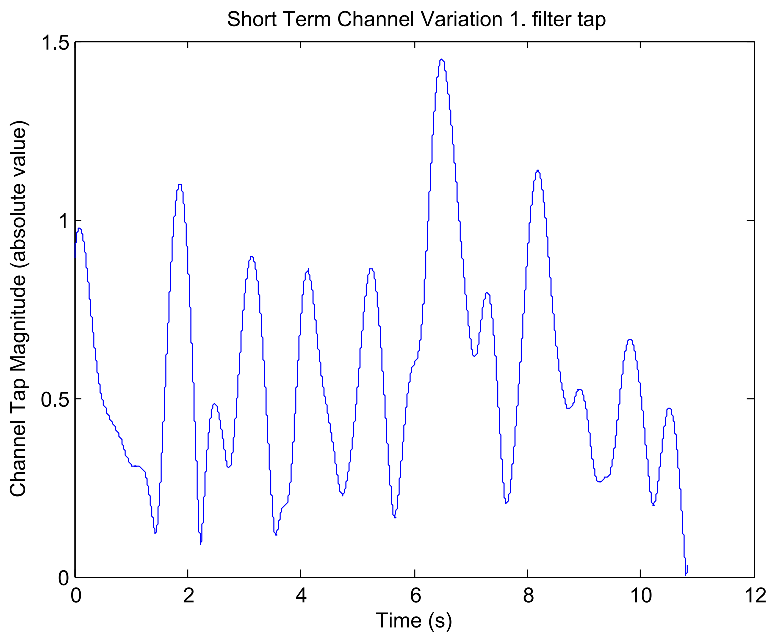

Figure 3.6: The absolute value of the first channel tap over time, in order to illustrate short term fading.

3.4 Rules of Adaption, FER-DRC

The adaptive rules of the FER-DRC is based on the recommendations made in [16] and [18] and is adjusted in order to match the properties in the HDL+ protocol. The basic rules implemented is to increase the data rate if the FER is less than 0.1 and to decrease it if the FER is above 0.5. In order to accommodate for the properties of the code combining, errors in retransmitted packets is weighted double. If one packet is retransmitted and still contains an error, this error will be counted as two errors, effectively meaning that it is possible to achieve a FER above 1. The equation for the modified FER (FER2) corresponds to Equation 3.2.

$$\mathrm{FER2} = \frac{2 \cdot (\mathrm{Packet\_errs\_retransmitted}) + (\mathrm{Packet\_errs\_new})}{\mathrm{Received\_packets}} \tag{3.2}$$

A packet error is defined as a packet containing one or more bits decoded incorrectly. The simulated system is unable to recombine packets sent in different sizes, so adaptivity of packet size is omitted in the FER-DRC, since, by removing the benefits of the ARQ type II, unwanted oscillations in packet sizes can lead to a severe decrease in the system efficiency.

3.5 Rules of Adaption, R-DRC

The adaption choices made in the R-DRC algorithm is based on a pregenerated LUT (Table 5.1 in the Appendix). The LUT is generated by a brute force method, all data rate and packet size combinations available are simulated on each entry in the two dimensional LUT (SNR and fade rate), and the combination which returns the highest throughput for that particular channel, is saved. This method in particular, is described in more detail in [6], but the LUT was then in three dimensions (SNR, fade rate and delay spread). A channel is here referred to as a combination of a specified mean SNR and a fade rate. The different channel conditions are specified by a SNR and a fade rate value. The SNR spreads from 1 to 30 dB divided in steps of 0.5 dB, and the fade rate is divided as follows: 0.5, 1, 2, 3 Hz. The rate and packet size choices for varying SNR is illustrated in Figure 3.7, the fade rate is set to minimum for the example.

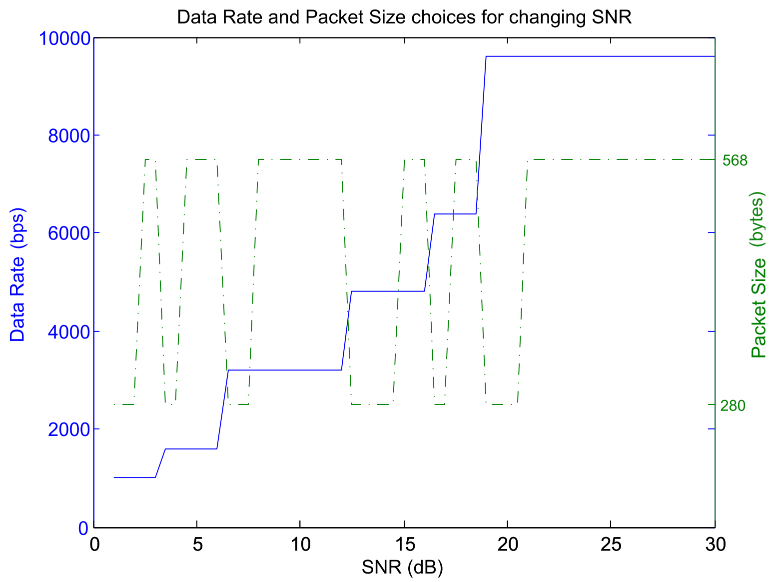

Figure 3.7: The generated Look Up Table, plotted for the varying SNR and the lowest fade rate.

3.6 Simulation Setup

Performance in Terms of Adaptivity

In order to investigate the special abilities and characteristics of the two adaptive solutions, one TX-frame was simulated sent over a ITU-MLD channel (2 ms delay spread, 1 Hz fading bandwidth [22]) with LTV & ITV for both systems, and illustrated in Figure 4.1 and in Figure 4.3. The data rate choices made is plotted in the same figure as the varying SNR in order to illustrate the pros and cons of the different solutions. In Figure 4.2 and Figure 4.4, the data rate choice and corresponding throughput is provided in order to see the results of the choices made and to compare the protocols with each other. The chosen data rate for the first TX-frame in each protocol is set to 4800, as this is the middle rate. The effects of a higher or lower start rate will be investigated.

Performance in Terms of Throughput
The most important performance measure of the adaptive systems, is the throughput at different mean SNR. The simulations are performed with messages of 50,000 bytes for SNRs below 12 dB, while messages of 100,000 bytes were used for the others. The amount is fairly large, but the chosen start rate for the system will probably still influence the overall rate of the system. By implementing memory and testing different start up rates, the effects of the selected start rate is illustrated in Figure 4.10 and in Figure 4.11.

When memory is implemented, data rate from the last TX-frame is used as a startup rate in the next iteration or SNR value simulated.

The protocols are simulated at SNRs from 4 to 28 dB with an increment of 4 dB. Each plotted value is a result of four separate simulations (iterations) with the same setup, but with different random generated data, channel variations and noise. The system memory (knowledge about data and SNR) is cleared between each iteration and each increment in SNR.

Both systems are simulated over the the following channels:

- Additive White Gaussian Noise channel with one tap and flat fading.
- ITU Recommendation Mid-latitude, Moderate conditions [22].
- ITU-MLD with LTV & ITV added to the mean SNR [19].

Chapter 4

Performance Analysis

In this chapter, the results of the simulations are presented as plots, and performance is discussed in terms of adaptivity (speed and correct data rate choices) and overall throughput for different channels. The results plotted for the HDL+ is replicated from [23] to the best of my ability.

At the end of the chapter, the effects of some alternative adaption rules are plotted and discussed.

4.1 Performance in Terms of Adaptivity

By plotting the varying SNR as a result of LTV & ITV in an ITU-MLD channel in the same plot as the data rate choices made, (as done in Figure 4.1 and in Figure 4.3), we are able to analyze how fast the different systems adapt, and the ability to track the varying channel. When the data rate and corresponding throughput for each TX-frame are plotted in the same plot, we we also able to analyze how good the choices are, and observe if the DRC is too optimistic or pessimistic. As it is rare to observe two successive frames with equal throughput, the frame duration is easiest observed as the time between a change in throughput in Figure 4.2.

Adaptivity for the FER-DRC Algorithm

The data rate choices for the FER-DRC algorithm is presented in Figure 4.1, and it seems to follow the channel quite good, although the channel might be a bit too fast changing at some points, making the algorithm lag one rate step behind. The corresponding throughput observed in Figure 4.4, often has a tendency to drop after an increase in data rate. This is obviously not wanted, but is hard to avoid without predicting the channel variation in advance. Shorter packet sizes could possible make the system able to track the channel better as it would increase the adaption frequency, but would also lead to more redundancy and probably lower data rate, at least in the lowest SNRs. As the FER-DRC currently is unable to adapt more than one step at a time, it was expected that it would a bit slow. It was also expected to lack a bit in throughput as a result of static packet size. As the start rate of 4800 bps is a good choice for the 20 dB mean SNR channel, the expected slow start is not observed.

When comparing the rate choices of the FER-DRC with the choices made by R-DRC in Figure 4.3, it seems that, despite the fact that it only adapts one step at a time, the FER-DRC has a lower threshold for more extreme data rate choices. The Fer-DRC chooses data rates from 3600 bps to 9600 bps in the same channel as the R-DRC only alternates between 4800 bps and 6400 bps.

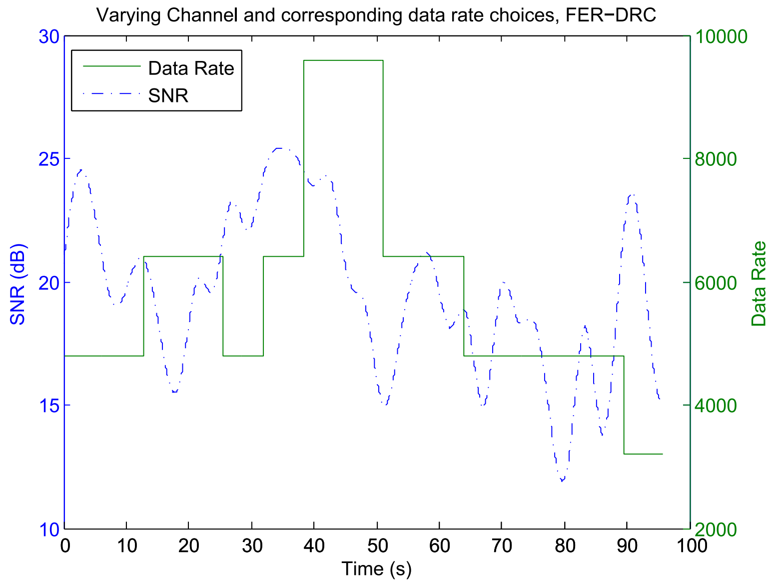

Figure 4.1: The varying mean SNR, as a result of LTV & ITV (Figure 3.4) and the corresponding data rate choices for FER-DRC. Total throughput for the example is 3953 bps.

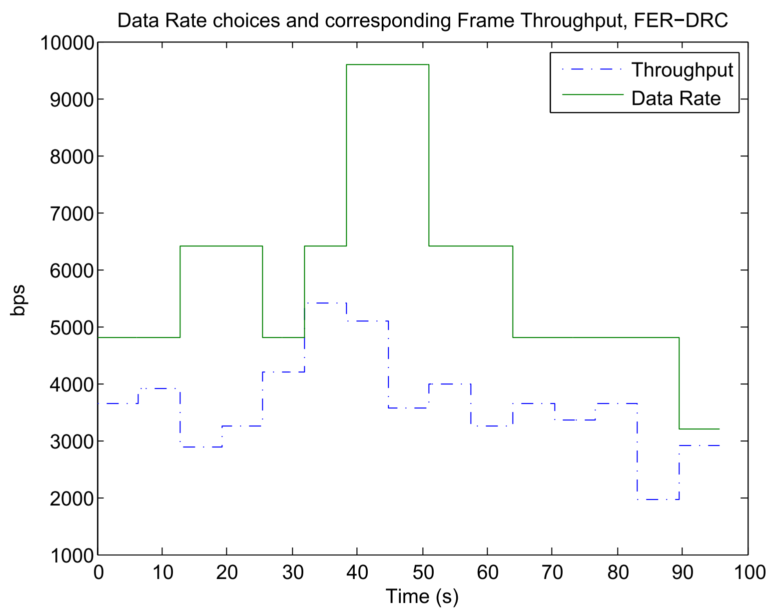

Figure 4.2: The frame throughput as a result of the data rate choices made by the FER-DRC algorithm in the channel illustrated in Figure 3.4.

Adaptivity for the R-DRC Algorithm

The data rate choices made by the R-DRC algorithm in Figure 4.3, seems a bit more modest than the choices made by the FER-DRC, and is quite typical when comparing the two algorithms. The FER-DRC will always keep trying higher or lower in order to find the optimal rate, while the R-DRC seems to already "know" the best alternative, and choose accordingly. It is evident that the R-DRC sustain the same problem as the FER-DRC in trying to track the channel variation, and seems to lag one step behind. The fact that the R-DRC algorithm also utilize the large packet size, makes the adaptive abilities even slower at some times.

By observing the throughput for the different data rate choices (Figure 4.2 and Figure 4.4), the effects of the ARQ type II becomes evident. After a frame is received with many errors and low throughput, the following frame often has a higher throughput. In return of this almost oscillating effect in throughput, the system gets a total throughput which corresponds better to the actual channel than earlier systems, as they got punished hard for choosing too optimistic data rates. Wrong data rate changes seems to be smoothed out in the overall throughput.

I have not found any similar results for HDL+, and is thereby not able to compare the adaption process of the proposed solutions with the original protocol.

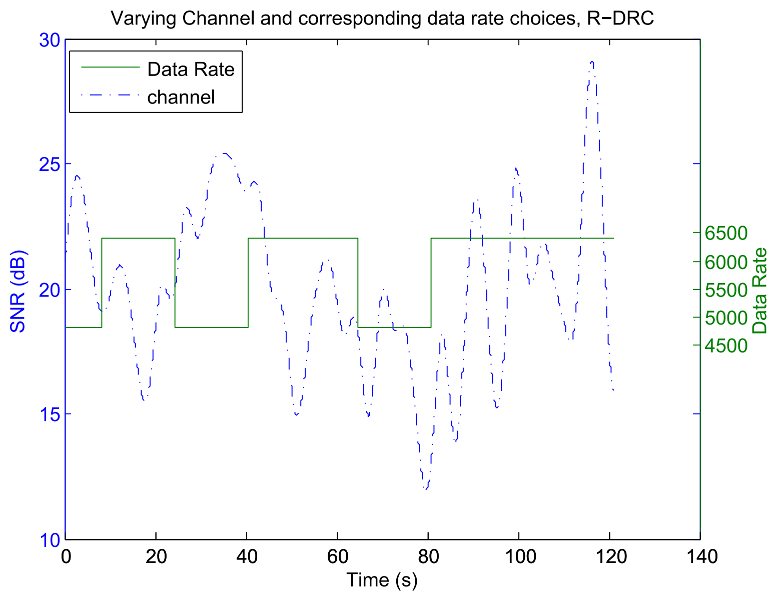

Figure 4.3: The varying mean SNR, as a result of LTV & ITV (Figure 3.4) and the corresponding data rate choices for R-DRC. Total throughput for the example is 4066 bps.

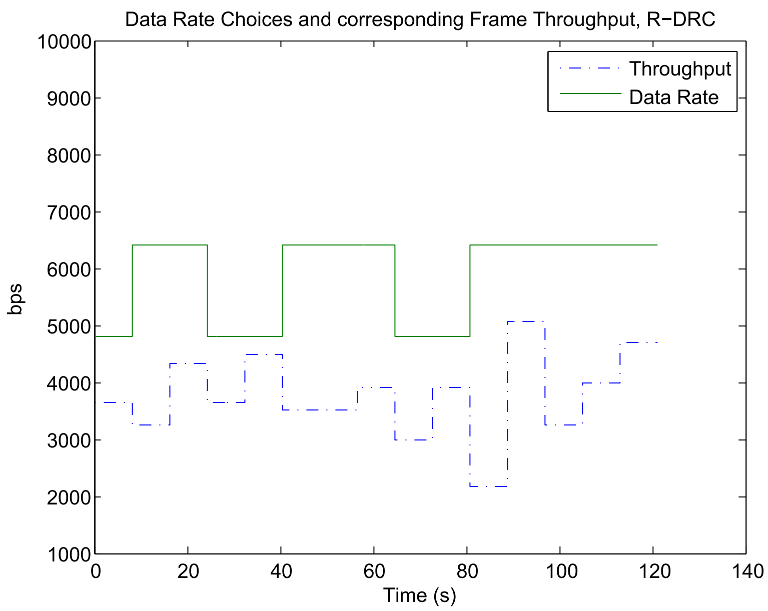

Figure 4.4: The frame throughput as a result of the data rate choices made by the R-DRC algorithm on the channel illustrated in Figure 3.4.

4.2 Performance in Terms of Throughput

In order to analyze the throughput for the different algorithms at different SNRs, the throughput for both FER-DRC and R-DRC are plotted in the same figure for different channels.

In an attempt to achieve good performance over several channels, the data rate choices are not ideal for each channel. This is explained for each channel, and examples of other DRC rules are presented at the end of the chapter.

AWGN, Single Tap, Flat Fading

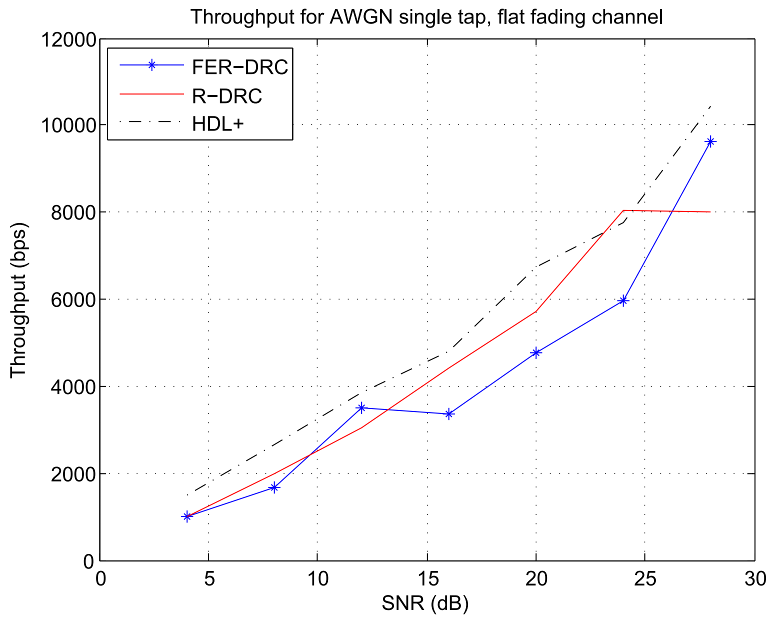

Figure 4.5: Simulated throughput for both algorithms on an AWGN single tap non fading channel with varying mean SNR on the x-axis.

The two algorithms is simulated for the best possible channel conditions, the Additive White Gaussian Noise, Single Tap channel with flat fading.

In Figure 4.5 The FER-DRC chooses the highest available data rate at the highest SNR, but is unable to reach the highest rate before three frames are sent, with respectively 4800, 6400 and 9600 bps, and this effectively prevents a higher throughput at these amounts of data. The R-DRC should not have this problem, but does actually not choose a higher rate than the 9600. The reason for this is that the system is too pessimistic in order to not choose too high rates in the other channels, which have a significantly higher probability of error for the same SNR. Both of these possible limitations are investigated in Figure 4.10.

ITU-MLD

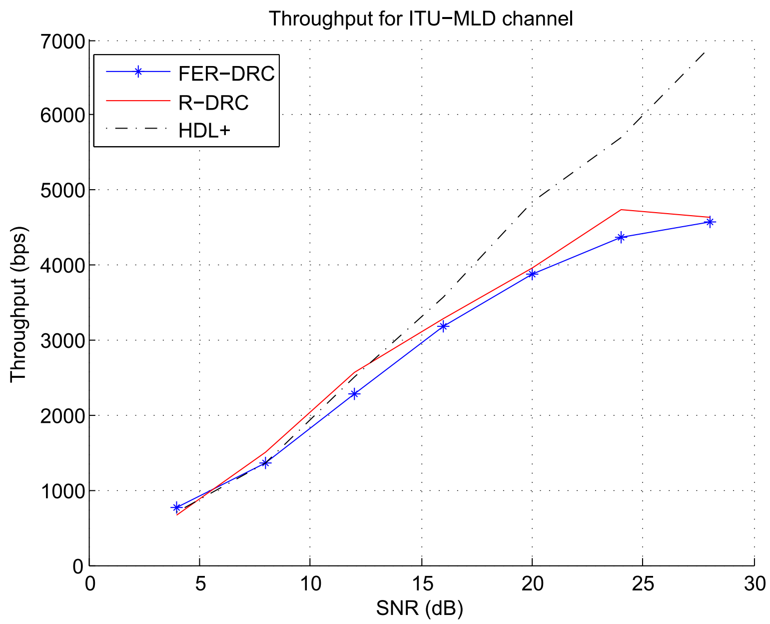

Figure 4.6: Simulated throughput for both algorithms on a ITU-MLD channel with varying mean SNR on the x-axis.

On the ITU-MLD channel (Figure 4.6), the R-DRC performs better than the FER-DRC for almost all SNRs except from the lowest, where FER-DRC performs slightly better. The reason for the better result at the low data rate is that the R-DRC is a bit too pessimistic and chooses only the lowest data rate, while the FER-DRC alternates between the data rates 1000 bps and 1600 bps.

Worth noting, is the decrease in throughput for R-DRC at 28 dB. By investigating the simulation data, it becomes clear that the reason for this is a combination of large packet sizes and high data rate selections, which leads to an excessive amount of retransmissions.

When comparing the results of the ITU-MLD channel with the HDL+ results presented by Harris [23], it becomes evident that the performance of the proposed algorithms is not good enough. The reasons for this, is hard to tell for sure, but I suspect that it is not the concept of the algorithm, but more likely the rules of adaption, the accuracy of the LUT, and the estimators used for adaption. Harris' adaption algorithms are probably a trade secret, so it is difficult to compare with the adaption strategies proposed.

The FER-DRC is performing even worse. This might be a consequence of the static packet size, and possibly not optimal adaption rules. It seems that the FER-DRC is a bit too pessimistic and does not take in to account that the following retransmission will gain from the previously received errors.

ITU-MLD with LTV & ITV

The results from the ITU-MLD channel with LTV & ITV (Figure 4.7) is a bit lower than the results from the ITU-MLD channel (Figure 4.6). This is not surprising, since it was expected that the channel variation would lead to lower throughput as a consequence of more unpredictable channel conditions. I suspect that the R-DRC advantage over the FER-DRC in 12 dB is a result of the packet sizes, whilst the higher maximum is achieved because the R-DRC chooses the highest data rate, while the FER-DRC oscillates between 9600 and 6400 bps. The fact that R-DRC is choosing the highest data rate in this channel, and not in the AWGN channel, is quite disturbing and tells me that it most probably is an error in the Doppler spread estimation. The LUT is generated on a two-path channel

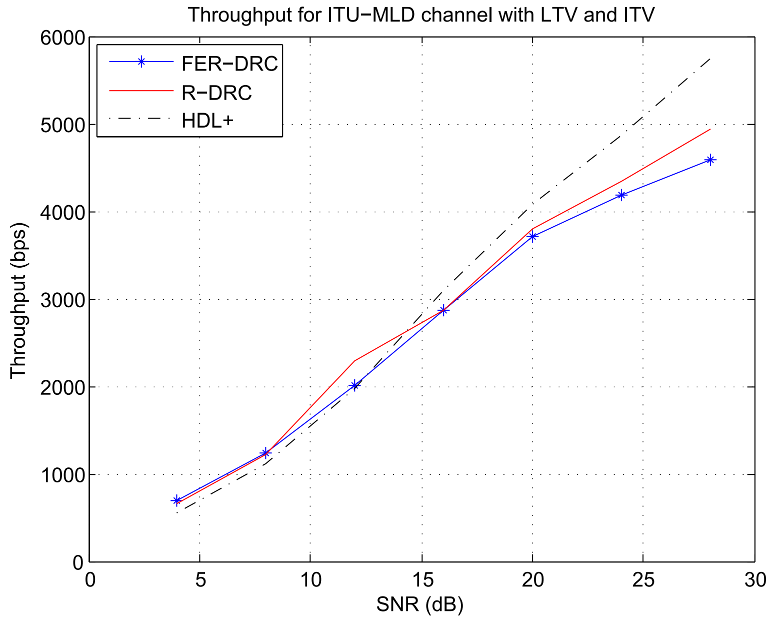

Figure 4.7: Simulated throughput for both algorithms on a ITU-MLD channel with additional LTV and ITV around the mean SNR on the x-axis.

environment, while the AWGN channel does not provide more than one tap. This might lead to errors in the estimation process.

By simulating both algorithms over several ITU-MLD channels with different LTV and ITV standard variation, (Figure 4.8 and Figure 4.9) the adaptive properties on faster and slower varying channels becomes apparent. As expected, a high variation leads to lower throughput, as the channel variations are more significant. It seems that The R-DRC has the highest throughput when the standard deviation is around 3 dB for both LTV & ITV. An explanation to this phenomena is that the LUT is generated at these channel conditions. As there currently is no estimation of the longer term variations, this is hard to correct for.

The FER-DRC has a lower maximum throughput, but is more stable over the different channels, and the variation of the standard deviation in LTV & ITV does not affect the throughput as much as it does on the R-DRC protocol.

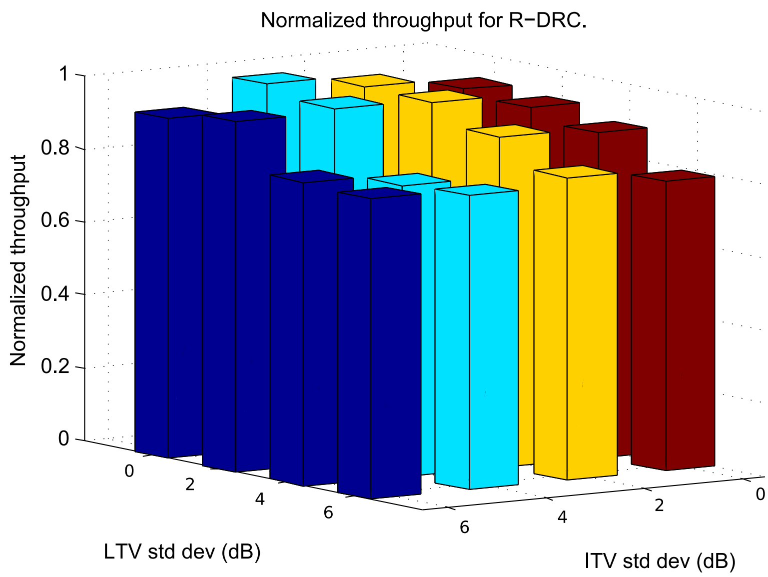

Figure 4.8: Simulated normalized throughput for R-DRC for various std. dev. in the LTV & ITV. ITU-MLD channel with mean SNR = 20dB

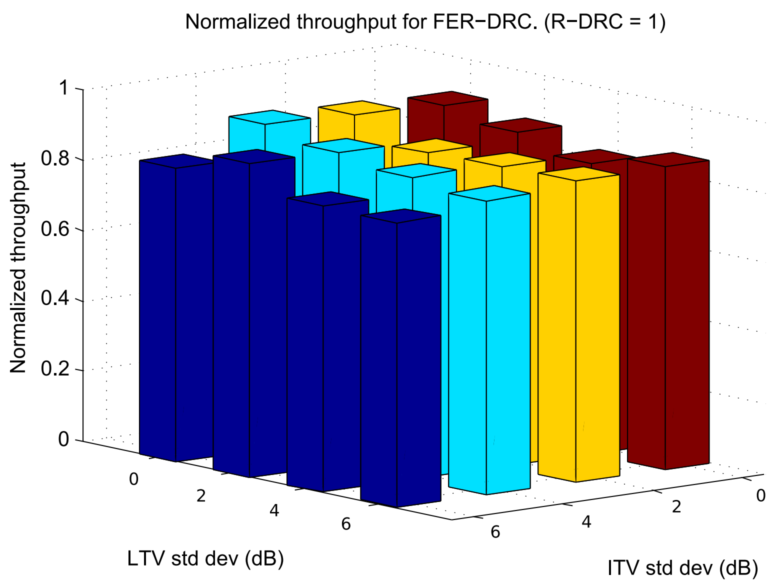

Figure 4.9: Simulated normalized throughput for FER-DRC for various std. dev. in the LTV & ITV. (R-DRC=1) ITU-MLD channel with mean SNR = 20dB

4.3 Performance of Alternative Adaption Rules

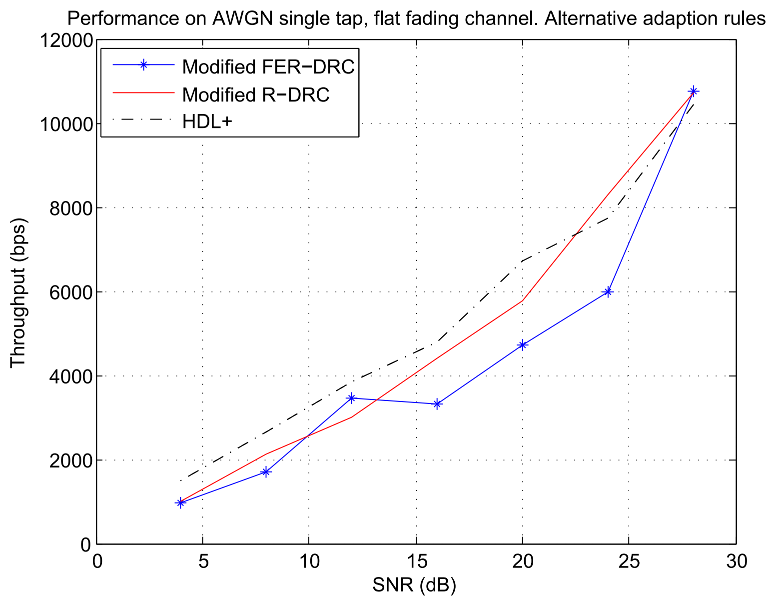

Figure 4.10: The effects of different adaption rules

The effects of modifying some of the adaption rules are presented in the following figures. In Figure 4.10, the R-RDC is modified to choose the highest rate in the AWGN channel, while the FER-DRC is implemented with utilization of the large packet size and memory between successive data transmissions.

It is observable that the throughput of the R-DRC is higher than in the original setup (see Figure 4.5), while the results of the modification of the FER-DRC does not seem to affect the throughput as much as expected.

In Figure 4.11, the effects of a different startup rate is illustrated by plotting several implementations of the FER-DRC with different startup conditions. Both a lower and a higher startup rate is presented together with a FER-DRC system where memory is implemented for the startup rate.

It was expected that the lower startup rate would perform best at the lowest SNRs, but all implementations seems to perform equally. This is probably

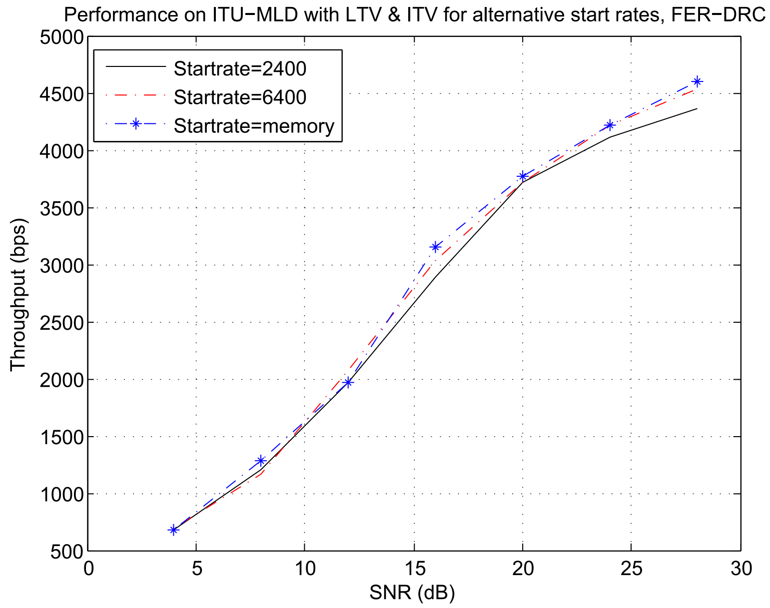

Figure 4.11: The effects of different start rates on the FER-DRC over a ITU-MLD channel with LTV & ITV.

because the choice of a too high data rate, is corrected for quite fast since the transmission time is shorter. Some errors might also be corrected by the code combining techniques in the following frame. At the higher SNRs it was suspected that the highest startup rate would perform best, and this is also the case. When choosing a too low data rate, the adaption process will be slower, and there is no way of correcting for the wrong choice. The best results is achieved by implementing memory in the system, but the achieved gain was not as high as expected.

Chapter 5

Conclusion

In this project, two separate alternative solutions for the adaption process of the HDL+ protocol is presented. The main focus is to design protocols which does not infringe the Intellectual Property Rights of Harris, which has developed the HDL+ protocol. Both of the protocols presented in this thesis, the FER-DRC and the R-DRC, utilize known estimation techniques in combination with the innovative ideas in the HDL+ protocol in order to get as high performance as the HDL+ without the use of patented technology. In order for easy comparison, both proposed protocols are simulated in a MATLAB environment over similar channels as the HDL+ results are presented. Performance is analyzed in terms of throughput, and the results are illustrated in plots and discussed throughout chapter 4.

More work may still be done in order to fully understand the strengths and weaknesses of the different adaptive data link protocols, some of which are mentioned in section 5.2. Still, some conclusions could be drawn. These are presented in the following section.

5.1 Main Findings and Results

By investigating the results of this thesis, we can conclude with the fact that it is fully possible to design DRC systems with the same packet system and structure as the HDL+ without violating any patents.

The FER-DRC is easy to implement and the fact that it performed satisfactory in comparison with the more complex R-DRC is a bit surprising, but is most probably because of a lower throughput in the R-DRC algorithm than expected.

The R-DRC is a bit more complicated than the FER-DRC, but absolutely implementable. During my work, it became clear the generation of the LUT and the channel estimation are the most crucial elements of the algorithm. Better implementation of these factors, might result in better performance.

The fact that the performance is not as good as the HDL+, is of course quite disappointing. But when the effectiveness of the highest data rate is reduced by removing the code combining, the maximum data rate is naturally decreased. There were also lower throughput at the lower data rates, which means that the adaptive rules of the proposed protocols probably has potential for further development.

5.2 Future Work

FER-DRC
The simplicity in the FER-DRC algorithm, makes it quite easy to tweak for performance, and works satisfactory for all the channels presented, but there are still potential for further development.

- Implementation of adaptable packet sizes would probably increase the overall throughput.
- The start data rate is crucial for the performance, and channel memory or faster adaption for the first frame should be considered.
- The possibility of adapting more than one step at a time might increase the total throughput.

R-DRC
As the R-DRC is a more complicated DRC algorithm, there are also more possibilites for further research. I recommend that the following are closer examined:

- A crucial part of the R-DRC is the channel estimation and the generation of the LUT. These are potential sources of error, and can influence the end result quite a lot. In order to simulate the best possible performance of the R-DRC, these possible error sources should probably be replaced with perfect estimation.
- Further development of the R-DRC could also be to improve the channel estimation and the LUT generation. Estimation of delay spread should also be considered, as it has become evident that this parameter influences the throughput quite a lot. There are no limitations in how many estimators can be used, since these are not transmitted as in the HDL+, so there should be no problem adding this additional information to the adaption process.

- Where the FER-DRC will always adapt when a wrong choice is made, R-DRC runs the risk of keeping make the wrong choice in successive transmissions, halting the whole transmission process. Good ways to account for this disadvantage is to have very good estimation and LUT, or monitor throughput vs. data rate in addition to the channel estimation.
- As with the FER-DRC, a better choice of the first data rate, should be considered, but is not so crucial as in the FER-DRC.
- The highest observed throughput is when memory is implemented in the adaptive process, leading to a better data rate for the first frame in a transmission.

It would also be interesting to compare the throughput of the protocols proposed in this thesis, with the original HDL protocol, as it is this protocol the HDL+ is developed from and compared with.

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Appendix

The Look Up Table used for adaption in R-DRC is provided in Table 5.1. D is the Doppler Spread. The data rates available is 1000, 1600, 3200, 4800, 6400, 9600 and 12800 bps. The last zero for each data rate is replaced with either 1 or 2, corresponding to small or large packet size (280 or 568 bytes).

Table 5.1: The Generated Look Up Table used by R-DRC.

		D = 0.5 Hz	**D = 1Hz**	**D = 2Hz**	**D = 3Hz**
SNR (dB) =	1	1001	1001	1001	1001
SNR (dB) =	1,5	1001	1001	1002	1002
SNR (dB) =	2	1001	1001	1001	1002
SNR (dB) =	2,5	1002	1002	1001	1002
SNR (dB) =	3	1002	1002	1001	1002
SNR (dB) =	3,5	1601	1601	1002	1002
SNR (dB) =	4	1601	1601	1602	1002
SNR (dB) =	4,5	1602	1601	1602	1002
SNR (dB) =	5	1602	1602	1602	1002
SNR (dB) =	5,5	1602	1602	1602	1002
SNR (dB) =	6	1602	1602	1602	1601
SNR (dB) =	6,5	3201	3201	3202	1602
SNR (dB) =	7	3201	3201	3202	1602
SNR (dB) =	7,5	3201	3201	3202	3202
SNR (dB) =	8	3202	3202	3202	3202
SNR (dB) =	8,5	3202	3202	3202	3202
SNR (dB) =	9	3202	3202	3202	3202
SNR (dB) =	9,5	3202	3202	3202	3202
SNR (dB) =	10	3202	3202	3202	3202
SNR (dB) =	10,5	3202	3202	3202	3202
SNR (dB) =	11	3202	3202	3202	3202
SNR (dB) =	11,5	3202	3202	3201	3202
SNR (dB) =	12	3202	3202	3202	3202
SNR (dB) =	12,5	4801	4801	4802	4802
SNR (dB) =	13	4801	4801	4802	4802
SNR (dB) =	13,5	4801	4801	4802	4802
SNR (dB) =	14	4801	4802	4802	4802
SNR (dB) =	14,5	4801	4802	4802	4802
SNR (dB) =	15	4802	4802	4802	4802

		D = 0.5 Hz	D = 1Hz	D = 2Hz	D = 3Hz
SNR (dB) =	15,5	4802	4802	6402	4802
SNR (dB) =	16	4802	6401	6402	4802
SNR (dB) =	16,5	6401	6401	6401	6402
SNR (dB) =	17	6401	6401	6402	6402
SNR (dB) =	17,5	6402	6401	6402	6402
SNR (dB) =	18	6402	6401	6402	6402
SNR (dB) =	18,5	6402	6402	6402	6402
SNR (dB) =	19	9601	6402	6402	6402
SNR (dB) =	19,5	9601	6402	6402	6402
SNR (dB) =	20	9601	6402	6402	6402
SNR (dB) =	20,5	9601	6402	9602	6402
SNR (dB) =	21	9602	6402	9602	9602
SNR (dB) =	21,5	9602	6402	9602	9602
SNR (dB) =	22	9602	9601	9602	9602
SNR (dB) =	22,5	9602	9601	9602	9602
SNR (dB) =	23	9602	9601	9602	9602
SNR (dB) =	23,5	9602	9601	9602	9602
SNR (dB) =	24	9602	9601	9602	9602
SNR (dB) =	24,5	9602	9602	9602	9602
SNR (dB) =	25	9601	9601	9601	9601
SNR (dB) =	25,5	9601	9601	9601	9601
SNR (dB) =	26	9601	9601	9601	9601
SNR (dB) =	26,5	9601	9601	9601	9601
SNR (dB) =	27	9601	9601	9601	9601
SNR (dB) =	27,5	9602	9602	9602	9602
SNR (dB) =	28	9602	9602	9602	9602
SNR (dB) =	28,5	12802	9602	9602	9602
SNR (dB) =	29	12802	12802	9602	9602
SNR (dB) =	29,5	12802	12802	12802	9602
SNR (dB) =	30	12802	12802	12802	9602