José Filipe Morgado Vieira

Licenciado em Ciências da Engenharia Electrotécnica e de Computadores

A Random Access MAC Protocol for MPR Satellite Networks

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Orientadores : Luís Bernardo, Professor Auxiliar com Agregação, FCT-UNL

Rui Dinis, Professor Auxiliar com Agregação, FCT-UNL

Júri:

Presidente: Prof. Doutor Rodolfo Oliveira

Arguente: Prof. Doutor Francisco Cercas

Vogais: Prof. Doutor Luís Bernardo

Prof. Doutor Rui Dinis

FACULDADE DE CIÊNCIAS E TECNOLOGIA UNIVERSIDADE NOVA DE LISBOA

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"In my hands the means, in my heart the will. D.N"

To my father, mother, sister, Marisa and to the rest of my family.

"Having a place to go, is a home. Having someone to love, is a family. Having both, is a blessing"

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Resumo

Os cenários de acesso aleatório em redes de satélite de baixa altitude (LEO), são tipicamente incompatíveis com os requisitos de qualidade de serviço (QoS) para tráfegos multimédia, especialmente quando os terminais móveis necessitam de operar a baixa potência.

As arquitecturas com optimização entre camadas, combinadas com recepção multipacote (MPR), são uma boa escolha para melhorar o desempenho geral de um sistema sem fios. O protocolo Hybrid Network-assisted Multiple Access (H-NDMA), exibe elevada eficiência energética, com capacidade de MPR, mas a sua utilização em satélites está limitada pelo elevado tempo de propagação. Este protocolo foi adaptado para satélites, com Satellite-NDMA, que requer uma fase prévia de agendamento introduzindo um atraso significativo.

Nesta dissertação desenvolveu-se um protocolo de acesso aleatório que usa H-NDMA, para redes de satélite LEO, denominado Satellite Random-NDMA (SR-NDMA). O protocolo foca-se no problema inerente às redes de satélite (tempos de propagação e consumos energéticos elevados) definindo uma abordagem híbrida com um conjunto inicial de accessos aleatórios juntamente com um conjunto de possíveis retransmissiões agendadas. Um receptor MPR combina as multiplas cópias recebidas, reduzindo gradualmente a taxa de erro. São propostos modelos de desempenho para o débito, atraso, jitter e eficiência energética considerando filas finitas nos terminais. É também estudado o problema da optimização da eficiência energética do sistema, sendo calculados os paramêtros que permitem cumprir os requisitos de QoS.

O desempenho do sistema é avaliado para um receptor que usa Single-Carrier with Frequency Domain Equalization (SC-FDE). Os resultados demonstram que o sistema é energéticamente eficiente e consegue fornecer um QoS suficiente para suportar serviços como video conferência.

Palavras Chave: Recepção multi pacote, Satellite Random Network-assisted Diversity Multiple Access (SR-NDMA), Acesso aleatório, Redes de satélite LEO, SC-FDE.

Abstract

Random access approaches for Low Earth Orbit (LEO) satellite networks are usually incompatible with the Quality of Service (QoS) requirements of multimedia traffic, especially when hand-held devices must operate with very low power.

Cross-Layered optimization architectures, combined with Multipacket Reception (MPR) schemes are a good choice to enhance the overall performance of a wireless system. Hybrid Network-assisted Diversity Multiple Access (H-NDMA) protocol, exhibits high energy efficiency, with MPR capability, but its use with satellites is limited by the high round trip time. This protocol was adapted to satellites, in Satellite-NDMA, but it required a pre-reservation mechanism that introduces a significant delay.

This dissertation proposes a random access protocol that uses H-NDMA, for Low Earth Orbit (LEO) satellite networks, named Satellite Random-NDMA (SR-NDMA). The protocol addresses the problem inherent to satellite networks (large round trip time and significant energy consumption) defining a hybrid approach with an initial random access plus possible additional scheduled retransmissions. An MPR receiver combines the multiple copies received, gradually reducing the error rate. Analytical performance models are proposed for the throughput, delay, jitter and energy efficiency considering finite queues at the terminals. It is also addressed the energy efficiency optimization, where the system parameters are calculated to guarantee the QoS requirements.

The proposed system's performance is evaluated for a Single-Carrier with Frequency Domain Equalization (SC-FDE) receiver. Results show that the proposed system is energy efficient and can provide enough QoS to support services such as video telephony.

Keywords: Multipacket Reception, Satellite Random Network-assisted Diversity Multiple Access (SR-NDMA), Random Access, LEO Satellite Networks, SC-FDE.

Acronyms

ARQ *Automatic Repeat-reQuest*

BER *Bit Error Rate*

BS *Base Station*

CC *Code Combining*

CDMA *Code Division Multiple Access*

CDPD *Cellular digital packet data*

DAMA *Demand Assigned Multiple Access*

DC *Diversity Combining*

DTMC *Discrete Time Markov Chain*

EPUP *Energy Per Useful Packet*

FDE *Frequency Domain Equalization*

FDM *Frequency Division Multiplexing*

FF-NDMA *Feedback Free-NDMA*

FDMA *Frequency Division Multiple Access*

FEC *Forward Error Correction*

FFT *Fast Fourier Transform*

GPS *Global Positioning System*

GSO *GeoStationary Orbit*

H-ARQ *Hybrid Automatic Repeat-reQuest*

H-ND MA *Hybrid Network-assisted Diversity Multiple Access*

HIPERLAN *High Performance Radio LAN*

HSDPA *High-Speed Downlink Packet Access*

HSUPA *High-Speed Uplink Packet Access*

IC *Interference Cancelation*

ID *Identification*

IFFT *Inverse Fast Fourier Transform*

ISI *Intersymbol Interference*

ISL *Inter Satellite Link*

LAN *Local Area Network*

LEO *Low Earth Orbit*

LTE *Long Term Evolution*

MAC *Medium Access Control*

MEO *Medium Earth Orbit*

MIMO *Multiple Input Multiple Output*

MMSE *Minimum Mean Squared Error*

MLSE *Maximum Likelihood Sequence Estimation*

MPR *MultiPacket Reception*

MT *Mobile Terminal*

MUD *Multi-User Detection*

NDMA *Network-assisted Diversity Multiple Access*

NGSO *NonGeoStationary Orbit*

OBP *OnBoard Processing*

OFDM *Orthogonal Frequency-Division Multiplexing*

OSI *Open Systems Interconnection*

PAR *Peak-to-Average Ratio*

PAPR *Peak-to-Average Power Ratio*

PER *Packet Error Rate*

PIC *Parallel Interference Cancelation*

PHY *Physical*

PSK *Phase-Shift keying*

PSTN *Public Switched Telephone Network*

QAM *Quadrature Amplitude Modulation*

QoS *Quality of Service*

QPSK *Quadrature Phase-Shift Keying*

RA *Random Access*

RTT *Round-Trip Time*

S-NDMA *Satellite Network-assisted Diversity Multiple Access*

SA *Scheduled Access*

SC *Single Carrier*

SC-FDE *Single Carrier-Frequency Domain Equalization*

SIC *Successive Interference Cancelation*

SR-NDMA *Satellite Random Network-assisted Diversity Multiple Access*

TDMA *Time Division Multiple Access*

WAN *Wide Area Network*

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q_m	Number of packets in the MT's queue in the beginning of the m group of RA slots	47
Q_{max}	Maximum capacity of the queue	47
Q_x	Gaussian error function	41
R	Scheduled retransmissions	35
R_e	Earth's radius	58
r_s	Distance from a MT to the satellite in km	58
S	Throughput	49
ST	Service Time	54
S_f	CDMA spreading factor	59
$S_{k,p}$	Data block transmitted by a user p, in the frequency domain	39
$s_{n,p}$	Data block transmitted by a user p, in the time domain	39
T	Round Trip Time in slots	35
T_{on}	Slot transmission time	50
V^l	Number of packets successfully received during the lth SA retransmission slots	48
X^l	Number of packets being transmitted in lth SA retransmission slots	47
X^0	Number of packets being transmitted in the RA slots	47
$Y_{k,r}$	Received signal, from multiple MTs, at the receiver in the frequency domain for a given transmission r	39
α	Peak-to-Average-Ratio of the radio frequency power amplifier	50
η	Drain efficiency of the radio frequency power amplifier	50
λ	Average packet generation rate per MT	36

Chapter 1

Introduction

1.1 Current context

An important requirement in the evolution of telecommunications is its capability to provide permanent and ubiquitous connectivity regardless of the location. The integration of the satellite communications in the terrestrial network will bring the telecommunications network to a level where global connectivity, anywhere and any time, can be achieved. The ubiquitous coverage of the satellite network provides a complementary service to the very high data rate available in the areas covered by terrestrial cellular services. For example, it will be possible a reachability on inaccessible areas, or areas involved in unpredictable catastrophic events where the infrastructure was destroyed.

With the advent of 4G Long Term Evolution (LTE)-Advanced, mobile ultra-broadband Internet access for Mobile Terminals (MTs) enabled services that rely heavily on broadband and Quality of Service (QoS), such as high-definition mobile TV, gaming services, IP telephony, mobile web access, video conferencing and cloud computing. To truly appreciate the user's experience anywhere in the world, it makes sense to integrate the terrestrial network with a satellite network. However, compared to terrestrial cellular infrastructures, satellite networks have higher propagation delays and require much higher transmission power due to the large distance between the terminal and the satellite. Therefore, to seamlessly integrate the 4G LTE terrestrial network with satellite networks, and at the same time provide multimedia services with guaranteed QoS, MTs must have low cost and operate with low power. Future technology will have energy efficiency as a major requirement for such systems and it should be fully compliant with the users demands.

1.2 Objectives and Contributions

Most of the medium access control (MAC) protocols proposed for satellite networks that satisfy QoS requirements [Pey99], [DGDRH09] use Demand Assigned Multiple Access (DAMA) or a hybrid mode with random access and reservations mechanisms. However, these options lead to an overhead increase and higher delay, due to reservations mechanisms in DAMA, or result in the loss of some packets due to collisions during the random access phase.

Multipacket Reception (MPR) and cross-layered techniques are a good solution to enhance the communication in wireless mediums, but with the cost of higher receiver and/or MAC protocol complexity. Network Diversity Multiple Access (NDMA) [TZB00] is a cross-layer solution that uses MPR, which was proposed for terrestrial cellular networks, where the Base Station (BS) forces the transmission of a number of MAC packet copies equal to the number of MTs involved in a collision. An evolution of NDMA was proposed in [GPB+11], called Hybrid-NDMA (H-NDMA), which applied Hybrid Automatic Repeat reQuest (H-ARQ) to NDMA in a terrestrial cellular network, forcing additional transmissions to enhance packet reception when difficult reception conditions persist, e.g. low Signal-to-Noise Ration (SNR) or high interference. However, H-NDMA is unsuitable for satellite networks due to the multiple control packets required for additional retransmissions and acknowledgements, which may introduce delay and jitter incompatible with several kinds of QoS requirements [AMCV06]. In order to overcome these issues, Satellite NDMA (S-NDMA) [GBD+12] was proposed, which applied the H-NDMA approach in a satellite network, considering a Demand Assigned Multiple Access (DAMA) approach. The satellite schedules multiple concurrent MTs, packet transmissions, defining the transmission power that minimizes the MTs average energy consumed for each packet correctly received. However, the DAMA approach introduces an initial extra Round-Trip Time (RTT) delay for slot reservation. This dissertation proposes the Satellite Random-Access NDMA (SR-NDMA) protocol, which adds the random-access capability to S-NDMA. It combines random access and scheduled access slots, and lets MTs transmit without requiring any pre-reservation mechanism. The developed work considers a Low Earth Orbit

(LEO) satellite network, based on the Iridium satellite constellation, using Single Carrier-Frequency Domain Equalization (SC-FDE) scheme for the uplink transmission. Analytical performance models are proposed to compute the throughput, energy consumption, delay and jitter, considering a constrained bandwidth and limited queue size restrictions, and assuming homogeneous Poisson load. This model is used to calculate the optimal energy transmission parameters that guarantees the QoS requirements, given a know uniform average load.

In result of the work developed in this dissertation, a paper was accepted in the IC-CCN 2013 conference, [VGB+13], and another paper was submitted to the New2An 2013 conference. A journal article is also being prepared using the work developed.

1.3 Dissertation Structure

The dissertation structure is briefly organized as follows: Chapter 2 contains a related work overview. A brief overview about satellite networks is followed by the typical modulation schemes used, error control mechanisms, typical MAC protocols for satellite networks and finalizes with the presentation of collision resolution methods, which include an overview about MPR and NDMA. Chapter 3 presents the SR-NDMA protocol proposed, covering the system overview, the MAC protocol and the analytical performance models, which describe the system and the optimization approach followed to calculate its configuration. Chapter 4 presents the SR-NDMA system configuration parameters considered and a set of performance results, i.e. throughput, energy consumption and transmission delay. Finally, Chapter 5 contains the conclusions and incorporates future work that could be done by taking this dissertation as a reference.

Chapter 2

Literature Review

2.1 Satellite Communications

Satellite networks have properties which make them attractive for a variety of applications and a set of situations where they are the only resource to communicate. For instance, satellites have a coverage area that greatly exceeds that of a terrestrial system, they allow the use of higher bandwidths and support on a global basis all forms of communications, ranging from simple point-of-sale validation to bandwidth intensive multimedia applications [IRS$^+$04].

Satellites traditionally serve as bent pipes, i.e. a satellite is constituted by several transponders, each of which is listening to a portion of the radio-frequency spectrum. Each transponder amplifies the incoming signal and rebroadcasts it into another frequency to avoid interference with the incoming signal. As bent pipes, they act as repeaters between two communication points on the ground, so there is no OnBoard Processing (OBP)[HL01]. However, some of the satellite systems do allow OBP. This can be advantageous in a communication satellite by providing baseband processing and switching; i.e. its functions are similar to those performed in terrestrial local area networks, which include, demodulation, demultiplexing, error detection and correction, switching, congestion control, buffering, remultiplexing, modulation and network synchronization [Ins00, p. 39].

We are faced with the challenge of integrating the terrestrial and satellite systems. With the forthcoming appearance of the fourth generation of wireless communication systems, future satellite systems should support applications with stringent Quality of Service

(QoS) and demanding bit-rates [IRS$^+$04].

The current satellite environment is quite rigid [LK07]. In the first place, the cost associated with designing, developing and operating a space vehicle is extremely high. Secondly, the requirements are stringent because satellites have to survive in a harsh environment without the easy access for repairs, maintenance or upgrades. This leads to developers having the obligation to build inflexible specialized vehicles with long life expectancy. In [LK07] a study was made with the purpose to change the mindset of the future satellite system to a more flexible one, showing that a long term benefit can be obtained. For example, to improve flexibility, a software update can be sent on an uplink to easily adapt the function of the satellite. Another example is shared processing in a satellite network; as processing capability improves, all satellites utilizing the shared processor would be able to take advantage of the performance improvement [KB00].

2.1.1 Satellite Constellations

A satellite system can be classified into Geostationary Orbit(GSO) and Nongeostationary orbit(NGSO), which includes Medium Earth Orbit(MEO) and Low Earth Orbit (LEO) [PD11, p. 117]. The GSO orbit contains the majority of the satellites in operation. These satellites are synchronized with the Earth's rotation, appearing fixed to an observer on Earth. Along with its high altitude, 35,786 km above the equator, a GSO satellite can cover one third of the Earth's surface, meaning three GSO satellites are enough for a global coverage. In spite of this, the cost associated with the launching of GSO satellites is high, and there is a inherent signal degradation with distance, due to its high altitude [HL01]. These disadvantages force the use of large antennas and large transmission power for both the satellites and the ground terminals. Adding to this, there is a high propagation delay for these links (typical value of 250 ms), which does not help for real-time traffic [Uni02, p. 356].

The NGSO satellites systems have the unique feature of viewing the entire Earth's surface periodically from a single satellite. If simultaneous viewing of the Earth is required, a constellation of satellites can be employed [Uni02, p. 419].

The LEO satellites systems (with orbits ranging from 160 to 1500 km above the ground [BWZ00]) have been proposed to overcome the disadvantages of GSO systems in high speed data and voice communications. In contrast to GSO systems, due to their rapid motion, which causes more frequent handovers [CAI06], LEO orbits require more satellites to provide the same communications services. However, they are much smaller and require significantly less energy to insert into orbit [Ipp08, p. 28]. Due to the shorter path distances to the ground stations, the LEO orbits require smaller power and smaller antennas sizes of the earth terminals and have lower propagation delays, between 10 and 25 ms [JT01]. The LEO satellites can also, with proper inclinations, cover high latitude locations, including polar areas, which cannot be reached by GSO satellites [Ipp08, p. 27].

The MEO satellites orbit at an altitude between 2000 and 4000 km [BWZ00], hence the MEO satellites cover larger areas of the Earth surface, are visible for longer periods and a lower number of such satellites is necessary to cover the entire Earth [PP07], compared to LEO satellites. The MEO orbit is very similar to LEO's but has higher propagation delays, on the order of 80 ms [BWZ00]. MEO satellites have been found useful for meteorological, remote sensing, navigation and position determination applications. The Global Positioning System (GPS), for example, employs a constellation of up to 24 satellites operating in 12-hour circular orbits [Ipp08, p. 28].

2.1.2 Systems in Low Earth Orbit

LEO satellite constellations (Figure 2.1) are promising solutions for satellite networks, due to their low delay and bit error characteristics [NBSL11]. Although, since they do not have a high view angle to offer adequate global coverage, these systems require a large number of satellites (24 to 66 have been proposed). These satellites must be served by a large number of earth stations, perhaps 200 or more. Low altitude may also increase the risk of shadowing [1] of the signal by vegetation, terrain and buildings. As with the cellular service, this may cause interruptions in transmissions [Man95].

LEO networks can be separated according to the kind of applications they offer. They can be separated into Low Bit Data Rate (Little LEO), Mobile Telephony (Big LEO) and High Bit Data Rate (Broadband LEO). A LEO system which provides Low Bit Data Rate is ORBCOMM. Example of Big LEOs are Iridium and Globalstar, while for Broadband LEO systems there are Teledesic and Skybridge [Man95] [Uni02, p. 420].

[1] shadowing or shadow fading, affects the wave propagation resulting in fading. Fading is deviation of the attenuation affecting a signal over certain propagation media

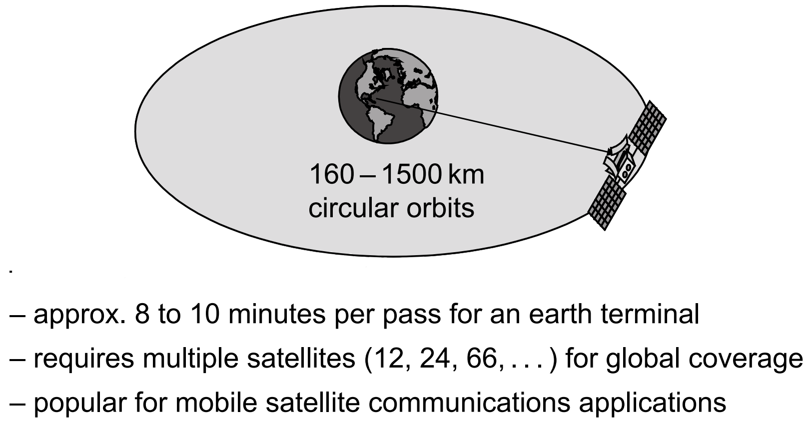

Figure 2.1: LEO-Low earth orbit. Adapted from [Ipp08]

Iridium and Globalstar provide narrow-band mobile voice services, whereas Teledesic and Skybridge provide primarily fixed broadband connections comparable to urban wireline service. ORBCOMM is used for commercial purposes primarily to provide electronic mail and paging to portable and mobile devices. It is not intended to carry voice calls [Man95].

2.1.2.1 Iridium Satellite System

Iridium system was completely deployed in May of 1998. It is based on a constellation proposed by [AR87], with 66 cross-linked operational satellites, plus seven in-orbit spares. These 66 satellites are arranged in six north-south necklaces, with one satellite every 32 degrees of latitude. All the satellites belonging to this constellation are located 780 km above the Earths surface [PRFT99].

Satellites that are part of Iridium system use Inter-Satellite Links (ISLs) to route traffic. Call setup procedures and the interface of Iridium with the existing Public Switching Telephone Network (PSTN) are handled by regional gateways. Iridium provides a network where the satellites communicate with other satellites that are near and in adjacent orbits. This kind of operation allows a simple call to roam over several satellites, coming back to the ground when downlinked at an Iridium gateway, and patched into an PSTN for subsequent transmission to destination. The existence of 48 cells (spot beams) per satellite with a capacity of 3840 channels, with 402Km of diameter apiece on the Earths surface for each satellite, is important to decrease the probability of existing dropped calls or missed connections. Of all channels available, some are used for paging and navigation while others are used for data and voice [PRFT99].

The ISLs and the links to the ground gateway stations work in the Ka band[2]. The frequency band used for mobile, pager and navigation services is the L band[3], which use the band 1.61 to 1.63 GHz [BWZ00]. Figure 2.2 shows the Iridum system architecture with its typical services.

Iridium satellites are also programmable, so it is possible to upload new instructions to them, in order to maintain good performances and high reliability levels [www12].

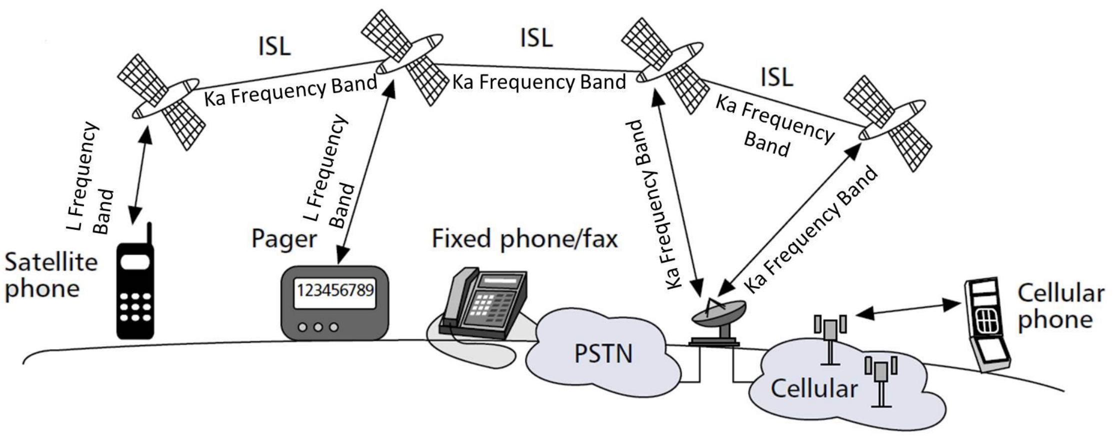

Figure 2.2: Generic architecture of the Iridium system and its typical services. Adapted from [JT01]

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2.2 Satellite Multiple Access Schemes

In satellite systems, there exist several ways to define separate communication channels, shared by several terminals [Ret80] [Uni02, p. 287]: Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA), Code Division Multiple Access (CDMA) and Space Division Multiple Access (SDMA), which are a set of common access techniques. More recently, in order to support higher data rates and to be able to cope with severe multipath propagation [4] two additional solutions were adopted in the WiMax/3GPP-LTE systems: Orthogonal Frequency Division Multiplexing (OFDM)

[2]frequency band ranging from 18 to 31 GHz

[3]L band ranges typically from 1Ghz to 2Ghz

[4]multipath propagation is the phenomenon that results in radio signals reaching the receiving antenna by two or more paths. Causes of multipath include atmospheric ducting, ionospheric reflection and refraction, and reflection from water bodies and terrestrial objects such as mountains and buildings. The effects of multipath include constructive and destructive interference, and phase shifting of the signal.

and Single-Carrier Frequency-Domain-Equalization (SC-FDE) [Wan11]. In this section, a short overview is given to CDMA, OFDM and SC-FDE since they are considered in the work developed in this dissertation.

2.2.1 Code Division Multiple Access

CDMA is a form of spread spectrum communication in which a narrowband signal is deliberately spread in the frequency domain, resulting in a signal with a wider bandwidth. This can make it more tolerant to interference, noise and jamming as well as allowing multiple signals from different users to share the same frequency band [PD11, p. 135]. The last characteristic is feasible since CDMA employs a special coding scheme that allows multiple users to be multiplexed over the same physical channel. The transmissions are distinguished by using different codes for each user, which are approximately orthogonal. That code allows CDMA on the receiver side to reject everything except the desired signal, using a frequency correlation operation. The receiver has the obligation to know the codeword that the transmitter used, in order to detect the message signal. There is no knowledge among users, so it means that each users operates in a independent way [Rap01, p. 405].

CDMA is practical for digital formatted data and offers the highest power and spectral efficiency operation of the three fundamental techniques (FDMA,TDMA,CDMA). A good spectral efficiency is achieved since several CDMA networks can share the same frequency band, because the undetected signal behaves as noise to all receivers without knowledge of the code sequence. This is particularity useful in applications such as NGSO Mobile Satellite Service systems, where bandwidth allocations are limited. Also, only a small portion of the signal energy is present in a given frequency band segment at any time, therefore, frequency selective fading or dispersion will have a limited effect on overall link performance. Again, because only a small portion of the signal energy is present in a given frequency band segment at any time, the signal is more resistant to intentional or unintentional signals present in the frequency band, thereby reducing the effects on link performance [Ipp08, p. 285].

However, CDMA can have performance issues due to the near-far problem. This occurs when there are huge signal strengths variations between the terminals in the same cell.

Stronger signals, raise the noise level for weaker signals at the receiver, making them less likely to being detected. The near-far problem is particularly difficult in CDMA systems, where transmitters share transmission frequencies and transmission time. To combat the near-far problem, power control is used in most CDMA implementations. Power control is provided by each base station in a cellular system and assures that each terminal in the base station coverage area provides the same signal level to the base station receiver. This solves the problem of a nearby subscriber overpowering the base station receiver and drowning out the signals of far away subscribers. Despite the use of power control within each cell, terminals out of the cell provide interference which is not under the control of the receiving base station [Rap01, p. 406].

2.2.2 Frequency Division Multiple Access

FDMA is a channel access method based on the Frequency Division Multiplex (FDM) scheme, which associates different frequency bands to different data streams. In the FDMA case, the data streams are allocated to different users. As a result, each user is individually allocated to one or several frequency bands (or channels) [PD11, p. 133].

For a large bandwidth, FDMA has some limitations due to equalization complexity. Equalization has the purpose of reducing Intersymbol Interference (ISI) to allow the recovery of the transmitted symbols. FDMA was improved by two additional approaches, OFDM and SC-FDE.

2.2.2.1 Orthogonal Frequency Division Multiplexing

OFDM is an evolution of FDM, and works by transmitting several modulated sub-carriers in parallel where each one occupies a narrow bandwidth [Cim85]. This way, the channel bandwidth is divided into many sub-carriers that independently send data. Each subcarrier is modulated with a conventional modulation scheme such as Quadrature Amplitude Modulation (QAM) or Phase-Shift Keying (PSK) at a lower symbol rate than the original data stream, maintaining total data rates similar to conventional single-carrier modulation schemes in the same bandwidth [NP00, p. 33].

OFDM, as opposed to single-carrier schemes, has the ability to handle severe channel conditions such as frequency-selective fading due to multipath or narrowband interference, without the use of complex equalization filters. Since the channel affects only the amplitude and phase of each subcarrier, equalizing each sub-carriers gain and phase does the compensation for frequency-selective fading [FABSE02]. The equalization is less complex because the many narrowband signals are modulated at a low symbol rate instead of one rapidly modulated wideband signal. Multipath propagation, which can cause ISI, can be eliminated by introducing a guard interval between each symbol. The feasibility of introducing guard intervals is bigger in OFDM since low symbol rates make longer symbol durations, as long as the channel dispersion is not longer than the guard interval, so the use of time domain equalization is not usually mandated. However, in case of higher data rates and channels with extensive time dispersion an equalizer is unavoidable [BS04, p. 103].

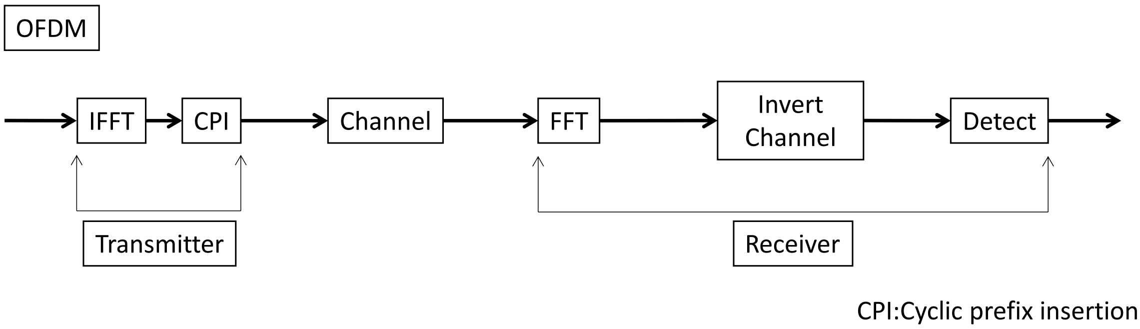

Figure 2.3: OFDM - signal processing. Adapted from [FABSE02]
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In OFDM (Figure 2.3), Inverse Fast Fourier Transform (IFFT) is applied on blocks of M data symbols at the transmitter side to generate the multiple sub-carriers. On the other hand, the receiver can extract the sub-carriers by applying a Fast Fourier Transform (FFT) on received blocks [FABSE02]. There is also a cyclic prefix whose goal, besides helping in avoiding ISI with the previous data block, is to make the received block look periodic, simulating a circular convolution, allowing an efficient FFT operation. Cyclic prefix carries the repetition of the last data symbol in a block, being consequently discarded at the receiver [BS04, p. 27].

It is widely recognized that a serious problem in OFDM is the possibility of extreme amplitude excursions of the signal. The signal is the sum of the several slowly modulated sub-carriers. In the most cases, the different carriers line up in phase at some instant in time, and therefore produce a very high Peak to Average Power Ratio (PAPR). The problem of these high peak amplitudes is most severe at the transmitter output. In order to transmit these peaks, not only must the D/A (Digital-to-Analog) converter have enough bits to accommodate the peaks, but more importantly, the power amplifier must remain linear over an amplitude range that includes the peak amplitudes. This leads to both high cost and high power consumption [BS04, p. 57]. The multiple access is achieved in OFDM by assigning subsets of sub-carriers to individual users. This allows simultaneous low data rate transmission from several users.

As a proof of its maturity, OFDM was selected as the High Performance Local Area Networks (HIPERLAN) transmission technique, became part of the IEEE 802.11 (Wi-Fi) standard as well as one of the the radio technologies for the LTE-Advanced.

2.2.2.2 Single Carrier with Frequency Division Equalizer

SC-FDE is an alternative to OFDM. While OFDM is a recognized multi-carrier solution to combat the effects of multipath conditions, mainly due to the favorable trade-off it offers between performance in severe multipath conditions and the signal processing complexity [Cim85], single-carrier modulation when combined with frequency domain equalization (SC-FDE) is capable of delivering a similar performance with essentially the same overall complexity [FABSE02], plus some advantages characteristic to single-carrier schemes.

SC-FDE has data sent at a high rate through one carrier and the data is modulated, for example with QAM or PSK, with equalization at the frequency domain. The frequency domain equalization is the frequency domain analogue of what is done by the conventional linear time domain equalizers. Frequency domain equalization is computationally simpler than the corresponding time domain equalization, since equalization is performed on a block of data at a time, and the operations on this block involve an efficient FFT operation and a simple channel inversion operation. When combined with FFT processing and the use of a cyclic prefix, a single-carrier system with FDE has essentially the same performance and low complexity as an OFDM system [FABSE02].

In OFDM and SC-FDE, one FFT and one IFFT block are employed in the system, even though in different places (Figure 2.4) and for different reasons. In the OFDM system, Fourier transforms are used for modulation and demodulation, whereas in the single-carrier system they are all incorporated in the receiver for converting time domain signals to the frequency domain and back, so that compensation for channel distortions can be accomplished in the frequency domain [PVK+08].

The use of SC modulation and FDE by processing the FFT of the received signal has a very attractive features: Single-Carrier modulation uses only one carrier, so the PAPR is lower. As SC-FDE systems have low PAPR, the power amplifier of an Single-Carrier transmitter does not need a big linear range to be able to support a given average power, so the power amplifier is less complex on these systems, resulting in a more efficient power consumption due to the reduced power backoff [FABSE02]. This main feature also gives good possibilities of both systems coexistence. For instance, in 3GPP-LTE, the Base Station (BS)[5] uses an OFDM transmitter and an SC-FDE receiver and the Mobile Terminal (MT) uses a SC-FDE transmitter and an OFDM receiver, avoiding IFFT operation complexity on the transmitter side and as result, improving the terminal battery resources [ZCM12].

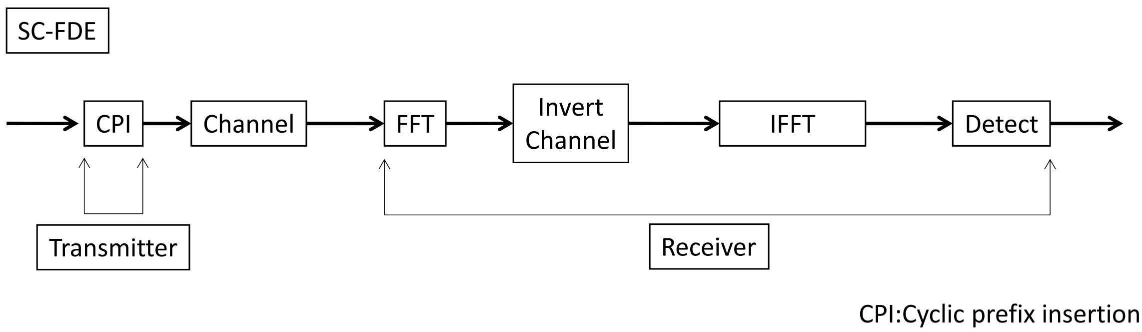

Figure 2.4: SC-FDE - signal processing. Adapted from [FABSE02]

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2.3 Error Correction Schemes

When data is sent on a channel, it is subject to errors for various reasons such as multipath fading, shadowing, among other reasons [Rap01, p. 139]. In order to ensure reliable communication, Forward Error Correction(FEC) (or channel coding), and/or Diversity Techniques may be employed.

[5]The base station is the section of a traditional cellular telephone network which is responsible for handling traffic and signaling between a mobile terminal and the network switching subsystem. The BS carries out transcoding of speech channels, allocation of radio channels to mobile terminals, paging, transmission and reception over the air interface and many other tasks related to the radio network.

2.3.1 Forward Error Correction

FEC relies on additional redundancy of the codewords [6] while diversity techniques rely on the redundancy from several copies of the same data symbol. FEC differs from error-detecting codes, since they include less redundancy than FEC, only enough for the receiver to know that an error exists, but without knowing what the error is. The strategy used in FEC is to include redundant information on sent data blocks, allowing the receiver to analyze it and see if data was correctly received, and if not, to know what was the error [PD11, p. 202].

Satellites and MTs rely on FEC to avoid data errors on the channel. Shannon [Sha48] studied the maximum rate at which data is reliably sent on a physical channel. He showed that, for a given transmission rate less or equal than the channel capacity, there is a forward correction scheme that allows data transmission with a small error probability, at the cost of spectral efficiency. If the rate is higher than the channel capacity, there is no reliable scheme. However, it is possible to increase the spectral efficiency at the cost of a higher bit energy noise ration, E_b/N_0, where E_b is the average bit energy and N_0 is the average channel noise [CHIW98].

2.3.2 Diversity Techniques

Diversity techniques refers to a kind of schemes where the method used for improving the reliability of a message signal is using two or more communications channels with different characteristics. Diversity techniques play an important role in order to lessen fading effects and channel interference as well as avoiding error bursts (i.e. in burst error channels, errors occur in clusters) by combining multiple versions of the same signal and make a single improved signal. Diversity techniques can be employed in frequency, spatial and time domains.

Frequency diversity arises when the signal is transmitted using several frequency channels or spread over a wide spectrum. The frequency response is no longer linear in these conditions, essentially due to the occurrence of fading and ISI. The main problem is then how to deal with the ISI while at the same time exploiting the inherent frequency diversity in the channel. There are three common approaches to deal with this, Single Carrier sys-

[6] a codeword is a n-bit frame containing data and check bits

tems with equalization (see section 2.2.2.2), spread spectrum methods and Multi-Carrier systems like OFDM (see section 2.2.2.1). For more information on frequency diversity consult [TV05].

In spatial diversity the signal is transmitted over several different propagation paths. In the case of wireless transmission, it can be achieved by using multiple transmitter antennas and/or multiple receiving antennas. A well known example of such scheme is the Multiple Input Multiple Output (MIMO) technique, which has become a standard for WiMax and 3GPP LTE, where multiple antennas are used at both the transmitter and receiver to improve communication performance [SBT11, p. 249].

Time diversity consists on obtaining multiple versions of the same signal, as any other diversity scheme, but at different time instants [TV05, p. 76]. It usually works by the BS demanding more data retransmissions from a MT when the MT data symbols are received with error. Automatic Repeat reQuest (ARQ) and Hybrid-ARQ schemes can be considered as a subset of the time diversity class and will be explored in the following section.

Notice that all these diversity techniques are employed in conjunction with multiple access schemes, some of which were explored in section 2.2.

2.3.3 ARQ Schemes

ARQ protocols are reliable data transfer protocols that are based in retransmissions. Normally, in this type of protocols, the receiver can inform the sender of what has been received with and without errors in order for the sender to retransmit what was not received correctly. This information exchange is performed by control messages.

An ARQ error-control system consists in the incorporation of an error detection code, with a certain retransmission protocol. Typically, if an error is detected, the receiver discards the erroneously received data and requests a retransmission of the same data. This process is repeated over and over, until the codeword is successfully received.

2.3.4 Hybrid ARQ Schemes

Hybrid-ARQ schemes are known as the combination of ARQ and FEC schemes, both already approached, and basically consists of a FEC subsystem contained in an ARQ system. With this combination done in a proper way, the disadvantages of both schemes can be overcomed [LCM84]. The purpose of embracing FEC as a function of an ARQ system is to reduce the number and the frequency of retransmissions by correcting some error patterns which may occur most frequently. Thus, the system throughput may be increased when some of the errors are corrected. Another advantage is that, unlike pure FEC systems, even when an uncommon error pattern is detected the receiver requests a retransmission rather than passing the erroneous decoded codeword to the end user. In general, a proper combination of ARQ and FEC schemes open the door to a more reliable system than an FEC only and also an higher throughput system than an ARQ only, thereby combining the assets of each system. The H-ARQ schemes can be divided in two categories: Type-I Hybrid ARQ and Type-II Hybrid ARQ [LCM84].

Recent interest in Hybrid ARQ schemes comes from the quest for reliable and efficient transmission under fluctuating conditions in future wireless networks. Hybrid ARQ techniques are currently used in 3GPPs High-Speed Downlink Packet Access (HSDPA), High-Speed Uplink Packet Access (HSUPA) and LTE [LL08]. Also, since the work done in this thesis relies on Hybrid-ARQ architectures, an overview will be given about this topic.

2.3.4.1 Type-I Hybrid ARQ Schemes

The type I hybrid ARQ protocol is the simplest of hybrid protocols. In this protocol, each frame is encoded for error correction and error detection [Wic95]. The message and the error detecting parity bit are encoded using an FEC code. The error correction parity bits are used in order to correct channel errors at the receiver side. The message estimation and the error detection parity bits are outputted to an FEC decoder, which tests it for error detection to determine if the message should be accepted or rejected due to errors. When the message is long or the channel signal strength is low, Type I H-ARQ can increase the efficiency, because this protocol decreases unsuccessful transmissions probability by adding extra FEC parity bits.

It is possible to have a coding gain if a compensation between the reduction of transmissions and the increase of message length is made. There is a crossover point in terms of strength between ARQ protocols and Type I H-ARQ protocols when the protocol's efficiency is the main subject. It happens in cases where signal strength is high. In these cases, this hybrid protocol type does not improve the efficiency, because the strong signal allows the delivery of free error messages. So the extra FEC parity bits are wasted. Hence, H-ARQ protocol type is not the best option in this case, as opposed to plain ARQ protocols [Wic95].

The Cellular Digital Packet Data (CDPD) wireless data protocols are an application that use these strategies [CHIW98].

2.3.4.2 Type-II Hybrid ARQ Schemes

Type-II hybrid ARQ protocols are more sophisticated than the type I, since the schemes used take into account when the channel becomes noisy and increasingly more errors start to occur. It is a sort of adaptive scheme by behaving like a pure ARQ when the channel is smooth and steady and while the channel becomes noisy, extra parity-check bits are added to the codeword. [Man74] and [Sin77] were the precursors of the fundamentals ideas behind the functioning of type-II H-ARQ. Here, a message in its first transmission is coded with parity-check bits for error detection only, as in a pure ARQ scheme, forming a codeword. If the receiver detects an error in the respective codeword, it saves the erroneous word and requests a retransmission. Now, the retransmission is not necessarily the original codeword, as is in type-I scheme, but can be a block of parity-check bits formed based on the original codewords and an error-correcting code. After this block of parity-check bits has been received, it is used to correct the errors presented in the previous stored codeword. If it does not succeed, a second retransmission is requested. The second retransmission may be either a repetition of the first and original codeword or another block of parity-check bits, depending on the retransmission strategy and the type of error-correcting code to be used. The main goal of Type II H-ARQ protocols is to work with the efficiency given by plain ARQ protocols in strong signals and to obtain the improvement of type I H-ARQ when the quality of the signal is low [Wic95, p. 417].

Code-Combining and Diversity Combining

Costelo et al [CHIW98] defined two main categories to classify the type-II Hybrid ARQ family, commonly known and referred as packet combining systems: Code-Combining (CC) (e.g. [Cha85]) and Diversity-Combining (DC) (e.g. [Sin77]) systems.

In code-combining systems, the packets are concatenated to form noise-corrupted codewords from increasingly longer and lower rate codes. In diversity-combining systems, the individual symbols from multiple, identical copies of a packet are combined to create a single packet with more reliable constituent symbols. These identical copies are obtained by straightforward retransmissions. DC systems are generally suboptimal with respect to CC systems, but are simpler to implement [CHIW98]. DC techniques can easily be extended to SC-FDE schemes and provide significant improvements in terms of delay and throughput performance [DCM08] [PBD$^+$10].

2.4 MAC Protocols in Satellite Communications

Medium Access Control (MAC) schemes are important in a wireless medium to coordinate MTs eligible for packet transmission. This means that MAC protocols have the job to control the access of communicating stations to the wireless medium, to share the network bandwidth [PD11, p. 258]. In the wireless medium multiple MTs might contend the wireless channel and interfere with each other [GL00]. Therefore, the design of the access scheme is of extreme importance, since it should enhance the throughput of the system and diminish the interference between the different MTs. However, not all MAC protocols are useful for satellite communications since some requirements are not achieved, primarily due to the long propagation delay. A large range of protocols that are applied in Local Area Networks (LANs) and Wide Area Networks (WANs) can not be used for this purpose. According to [Pey99] the main architectural objectives when designing MAC protocols for satellite communications are high channel throughput, low transmission delay, channel stability, protocol scalability, channel reconfigurability and low complexity of the algorithms used for the protocols.

MAC protocols can be generically classified as distributed or centralized [GL00]. Distributed MAC protocols rely on carrier sensing and collision avoidance algorithms, run locally by the MTs. The 802.11 DCF protocol is an example but they are also designed for distributed wireless networks (Ad-Hoc networks, Wireless Sensor Nets).

Centralized MAC protocols, as opposed to distributed ones, are coordinated by a BS. All the communications must go through the BS. In a satellite communication system, which normally covers a large distance between the satellites and the MTs, a centralized

MAC protocol allows higher throughputs and energy efficiency.

Centralized MAC protocols divide themselves into three categories: Random, Hybrid and Guaranteed Access [GL00]. Inside Hybrid Access exists two subcategories: Random Access and Demand Assignment. Let us focus on Random Access and Demand Assignment protocols, since these are the most important regarding this thesis. Demand Assigned are a class of protocols intended to increase efficiency by an advanced capacity reservation procedure while in random access protocols each station makes it own decision regarding when to access the channel, making them easy to implement and adaptive to varying demand.

2.4.1 Random Access Protocols

In the case of satellite mobile networks, a large number of user terminals may generate infrequent packets, generating "bursty" traffic with frequent inactivity periods in the return link[7]. A Demand Assignment protocol in these operating conditions will not perform optimally [DGDRH09]. Random Access techniques are by nature, very robust to this type of traffic and to large populations of terminals sharing the same capacity. A list of protocols based on random access used in satellite networks are available in [MB02], which includes the ALOHA protocol, slotted ALOHA (S-ALOHA) protocol, selective-reject (SREJ) ALOHA protocol, the Time-Of-Arrival Collision Resolution Algorithm protocol and the Announced Retransmission Random Access (ARRA) protocol.

ALOHA protocol

In ALOHA protocol, stations are not synchronized and only transmit packets when they are ready. When a collision occurs, each station knows that it happened and retransmits the packet after a random period. This random period provides stability to the protocol [Pey99]. S-ALOHA is an improvement to the original ALOHA protocol, which introduces discrete time slots and doubled the maximum throughput. A station can send only at the beginning of a time slot, and thus collisions are reduced. The time slot is defined by the network clock and equal to the common packet duration [Pey99].

[7] The transmission link from a user terminal to the base station.

SREJ-ALOHA protocol

With ALOHA, collisions between packets are most often partial [8], however the coherence of the packet is destroyed by even a partial collision. This leads to retransmission of the contents of the whole packet, although only a part has suffered a collision. The SREJ-ALOHA [Ray87] was designed to avoid a complete retransmission. The transmitted packet is divided into sub-packets, each having its own header and protocol bits. When a collision occurs, only the sub-packets involved are retransmitted. The efficiency of the protocol is greater than that of the ALOHA protocol. The SREJ-ALOHA protocol is well suited to applications in which the messages have variable lengths.

Time-Of-Arrival Collision Resolution Algorithm protocol

The Time-Of-Arrival Collision Resolution Algorithm protocol [Cap79] provides an improvement to the ALOHA protocol by avoiding the possibility that a packet which has already been subjected to a collision encounters another packet during its retransmission. To achieve this, stations avoid transmitting new packets in the time slots provided for retransmission of packets which have suffered a first collision. This protocol implies a procedure for identifying packets which have suffered a collision and the setting up of temporary coordination of transmissions. However such a protocol tends to be complex to implement.

ARRA protocol

The ARRA protocol increases the efficiency of S-ALOHA by introducing a frame structure which permits numbering time slots. Each packet incorporates additional information indicating the slot number reserved for retransmission in case of collision. This protocol avoids collisions between new messages and retransmissions [MB02, p. 289].

Random access protocols comparison

In terms of efficiency, ALOHA protocol does not exceed an average normalized throughput of 18% and the mean transmission time increases as the traffic increases due to the in-

[8] With asynchronous transmissions, as opposed to S-ALOHA which are synchronous, the collisions do not occur in a regular time slot

creasing number of collisions and packet retransmissions. S-ALOHA reaches a maximum average normalized throughput values around 36% [Pey99]. The SREJ-ALOHA has a practical limit, in the order of 30%, which is caused by the addition of headers to the sub-packets. The Time-Of-Arrival Collision Resolution Algorithm protocol has an efficiency of 40% to 50% while ARRA protocol has an efficiency of about 50% to 60% [Pey99].

2.4.2 Demand Assignment Protocols

Demand Assignment Multiple Access (DAMA) protocols are intended to increase efficiency even more by an advance capacity reservation procedure. According to [GL00], DAMA protocols allocate bandwidth to MTs according to their QoS constraints.

A mobile terminal reserves a particular time slot within a frame for its own use. The efficiency can be as high as 70% to 90% depending on the fraction of capacity used for the signalling information associated with the reservation procedure [Pey99]. Reservation can be implicit or explicit [Ret80].

Implicit reservation is reservation by occupation; that is, every slot occupied once by the packet from a given station remains assigned to this station in the frames which follow. This protocol is called R-ALOHA [Rob73]. The disadvantage is that a station is in a position to capture all the time slots of a frame for itself. The advantage is the absence of set-up time for a reservation.

Explicit reservation involves a station sending a request to occupy certain time slots to a base station. Two examples of this are R-TDMA and contention-base priority-oriented demand assignment (C-PODA) [JBH78]. The disadvantage of these protocols is the establishment time, which can be prohibitive for some interactive applications.

2.5 Collision Resolution Techniques

As was already said, Demand Assignment schemes are generally good for constant bit rate traffic with limited buffering or strict delay constraints, but they fail when arrivals are more bursty. In this case, random access schemes are generally preferable since they allow low mean delay, provided that the overall traffic load is limited. However, they also open the door for a collision environment.

When separating the MAC and Physical (PHY) layer, a perfect channel abstraction is usually considered for the PHY layer when designing the MAC protocols. Errors only exist when collisions occur due to simultaneous transmissions and when only one user transmits, the packet arrives at the receiving end error free. H-ARQ techniques were discussed in section 2.3.4, but they are not appropriate when packets are lost due to collisions. In those cases, the traditional approach is for the MAC layer to ask for the retransmission of the packets associated with different users with different probabilities to avoid more collisions. Random access protocols were based in this collision model and therefore they were designed to resolve collisions. This model does not consider however the possibility that packets may be successfully decoded in the occurrence of a collision. So, with the evolution of wireless communications, the design of both PHY and MAC layer went beyond the collision model.

Packet reception in the presence of collisions was improved and at the same time, the overall network efficiency was increased by upgrading or making the different protocol layers cooperate. This section serves as an overview on some collision resolution techniques as well as recent improvements that emerged when trying to resolve the collision problems.

2.5.1 PHY-Layer Solutions

Traditionally, the PHY layer leaves to the MAC layer the task of separating users via scheduling. However, signal processing techniques at the PHY layer have evolved in a way that the problem of packet collisions can be solved. For instance, in [ZR94] and [HKL97] was observed, that the signal capture mechanisms can decode a packet that has a higher power, in comparison with all the other packets involved in a certain collision. A conclusion can be taken; collision problems could not be exclusively solved by MAC layer. In the next sub-section, a PHY layer solution to solve collisions, called Multipacket Reception (MPR), is explained.

2.5.1.1 Multipacket Reception

MPR is the capability of simultaneous decoding of more than one packet from multiple concurrent transmissions, or in another words, packets involved in collisions. MPR is currently an active area of research (e.g. [SL11] [BCA12]) since it provides improvements to both throughput (i.e. multiple packets are received at a given time as opposed to one packet) and capacity of a wireless network. There are a number of techniques which allow simultaneous decoding of packets on a receiver, but in many papers, MPR is said to be realized with CDMA or MIMO. The appearance of these notions together with MPR brings some ambiguities to the understanding of MPR enabling techniques. To this end, [LSW12] made a survey about MPR in wireless random networks and according to it, MPR techniques can be classified based on the Transmitter perspective, Trans-Receiver perspective and Receiver perspective, as seen in Figure 2.5.

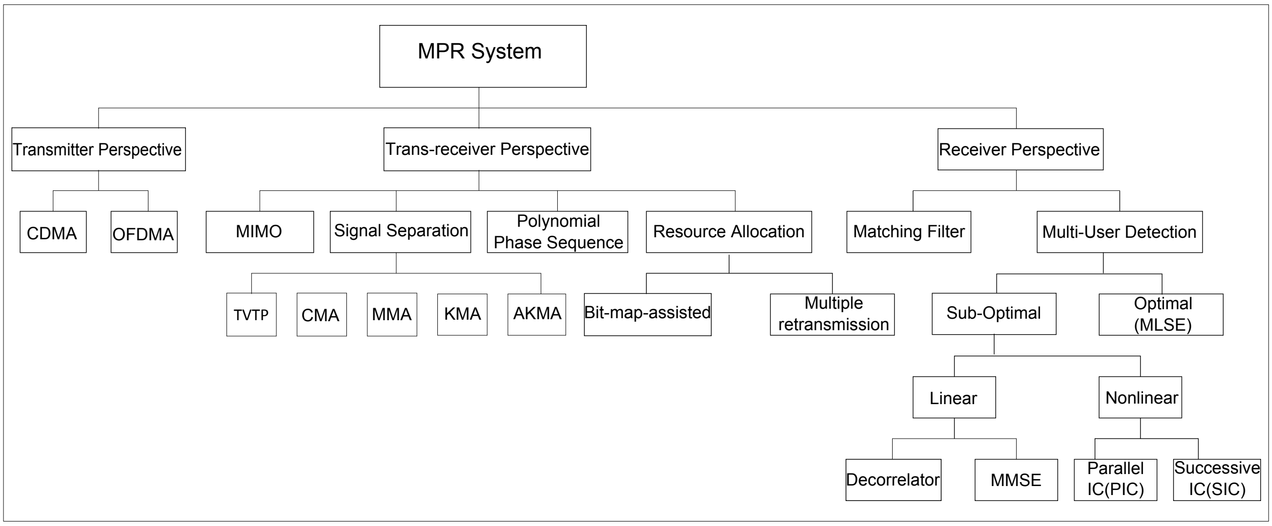


Figure 2.5: Classification of techniques applied for MPR. Adapted from [LSW12]
.

Let us focus on the Receiver perspective, more precisely in the Multi-User Detection (MUD), since in this work a MUD technique is used in the receiver implemented.

MUD is the ability of discerning data symbols from multiple interfering users, [Mos96]. The optimum multiuser receiver for CDMA was discovered and advanced in [Ver86]. Although the receiver is able to optimally decode multiple users in parallel with dramatic gains, it is extremely complex to implement and computationally exhaustive [And05]. In Figure 2.5 this receiver is known as the maximum likelihood sequence estimation (MLSE). But since the MLSE was too complex to implement, the research industry led the efforts in achieving sub-optimal multiuser receivers with lower complexity. Sub-optimal MUD techniques can be linear or non-linear.

Linear Techniques

Decorrelated detectors ([Sch79]) and Minimum Mean Square Error (MMSE) detectors (

[XSR90]) are the most known linear MUD techniques. These linear detectors apply a linear mapping to the soft output of a conventional detector to reduce interference from other users. The decorrelating detector will decorrelate the channel making the information of interest orthogonal to the interference. Nevertheless, its major drawback is the noise enhancement due to the inverse filtering. The MMSE detector does not completely remove the interference, which for low signal-to-noise ratios results in better performance since the noise is not enhanced in the same way as the decorrelator does. Linear detectors have however high complexity in comparison with non-linear detectors [Mos96] but yield an optimal value of the near-far resistance performance metric [9].

Non-linear Techniques

Non-linear MUDs use interference estimators and remove the interference from the received signal before detection. They are much simpler but have an inferior performance compared to linear MUD [LSW12]. The most known in this category is the Multistage Interference Cancelation (IC), where cancelation can be carried out either successively (SIC) or in parallel (PIC). For SIC ([KIHP90]), the signals from the various users are demodulated and canceled from the strongest to the weakest according to their received signal power [LSW12]. For PIC ([BCW96]), without the exact knowledge of the interfering bits, their estimates in the previous stage are used instead. To enhance the performance of PIC, a multistage approach is often adopted, in which the detector at nth stage use the bit decision from the (n-1)th stage. In theory, PIC could support more simultaneous packets from different users, but a perfect power control is necessary [LSW12]. These techniques can employ several stages, where performance is improved at each successive stage by using more retransmissions. Nonlinear detectors have the disadvantage of requiring reliable estimates of the channel (i.e. the interference estimators).

For more information on multiuser detection techniques refer to [Ver11].

[9] The near-far resistance is used to quantify the degree of robustness of multiuser detectors against the near-far problem

2.5.2 PHY-MAC Cross-Layered Designs

The Open Systems Interconnection (OSI) model specifies that layers do not share information between them. However some works emphasize the opposite, referring the importance of sharing information from lower layers to enhance the performance of higher layers on a wireless context [SRK03] [SM05]. This interaction brings some advantages (see Figure 2.6, where a PHY-MAC cross-layer designing is proposed and where the PHY layer possesses MPR capabilities[10]), because all levels of network protocol stack are affected by wireless link characteristics. Hence, all layers must respond to changing channel conditions, leading to a strong union among protocols at different layers. Some conditions have to be present at all layers in order to provide QoS delivery and adaptability to channel transmission. For instance, at the physical layer, dynamic adjustment of receiver filters can be made to respond to interference changes; at the link layer, the interference level can be affected by adapting power, rate and coding; finally at the MAC sub-layer, it is possible to adapt scheduling, based on the current level of interference and on the quality of the current link [BNNK08].

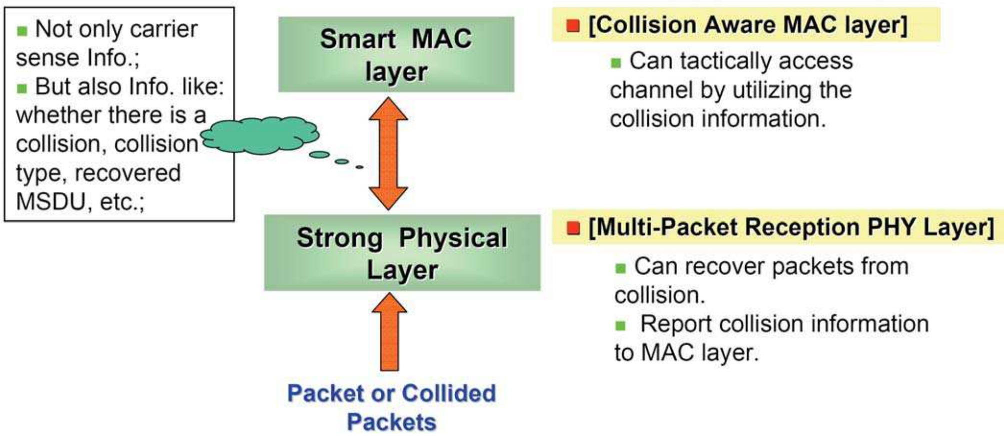


Figure 2.6: A cross-layer design on the basis of multiple packet reception. Adapted from [LWLK07]

Over the years some research work has been made to develop PHY-MAC cross-layered solutions. A union of MPR with MAC and the creation of an adaptive resource allocation algorithm for MIMO Wireless Local Area Network (WLAN) was made in [HLZ08]. In [GLASW07] a study of a cross-layer MAC algorithm for WLAN having single antennas

[10]MSDU stands for MAC Service Data Unit

terminals and multiple antenna access points was made taking into consideration an error free transmission channel. Another PHY-MAC protocol was developed in [CLZ08] for Ad Hoc networks, where the MAC layer uses the PHY layer knowledge to implement proper policies for distributed traffic control and robustness against interference. These are some examples of recent works done in PHY-MAC designs, but one has gained some relevance and it is the base for the work in this dissertation, which is approached in the next sub-section.

2.5.2.1 Network-assisted Diversity Multiple Access Protocols

Random access schemes rely on retransmissions to resolve current collisions, and most of the effort made in the random access literature has been essentially on retransmission schemes that minimize future collisions. In those schemes, the throughput penalty incurred by collisions cannot be eliminated unless some way is devised to extract useful information from that collided packets, since the collided packets are typically discarded when a collision occurs.

In [TZB00], a novel approach was proposed to the collision resolution problem to recover packets from multiple collisions and thus boost throughput, the so called Network-assisted Diversity Multiple Access (NDMA). The NDMA class of protocols essentially consists in not discarding collided packets and relying on proper retransmissions and signal separation principles to resolve collisions. In particular, if P users collide in a given time slot, they repeat their transmission for a total of P times so that P copies of the collided packets are received. Then, the receiver has to resolve a $P \times P$ source mixing problem and separate each individual user. All in all, it requires only P slots to transmit P colliding packets. The collided packets are stored for future decoding much like a H-ARQ type-II technique, in which packets are not discarded to enhance data reception, by using diversity combining ideas to separate the collided packets. The MAC layer is explored to create diversity at the physical layer, by asking for the retransmissions. So NDMA protocols should introduce throughput penalty in the presence of collisions (at least in theory), but for an enormous number of MTs accessing the wireless medium, NDMA can become problematic without an appropriate backoff scheme due to the high complexity of the receiver . In [PBD$^+$09] this problem was addressed and the original NDMA was extended, to allow a number of MTs contending the wireless medium higher than the maximum number of decodable colliding packets, with an appropriate backoff mechanism.

After [TZB00], some research work was done in order to evolve NDMA. The initial NDMA was designed for flat fading channels, which are not very appropriated for wireless communications. So in [ZT02] a new strategy was built for a frequency selective channel environment using multiuser receivers and CDMA systems. They implemented transceiver architectures and random access strategies to separate collided packets when unknown propagation channels are present. User identification (ID) signature sequences were used, making easier the collision detection and resolution process when multipath effects are present. [ZT02] also achieved a good throughput performance. The disadvantage brought by ID sequences is that they grow linearly (instead of logarithmically) with the number of users, introducing a considerable overhead. Another problem is that due to the linear nature of the receivers used (see section 2.5.1.1), the residual interference levels can be high and/or significant noise enhancement can be obtained. To address these issues, frequency-domain multiuser detection schemes that allow efficient packet separation in the presence of successive collisions were proposed in [DCB+07] [DMB+09]. These receivers are suitable to severely time dispersive channels and do not require different channels (i.e. uncorrelated channels) for different retransmissions. In some receivers, ideally, the retransmissions should be made with uncorrelated channels to allow an efficient packet separation. Since this is not practical in many cases, different packet phase rotations can be employed for different users (which is used in [ZT02]), interference cancellation procedures (as seen in section 2.5.1.1), or cyclic shifts (as in [DMB+09]).

Some evolutions on NDMA protocols

Several improvements have been proposed since the first appearance of the NDMA protocol. In [SRGM06] a constant p-persistent access mode for NDMA protocol was proposed to mitigate the unfavorable joint detection conditions at finite SNR. The p-persistent mechanism is optimized to reduce the average number of colliding users so as to optimize the throughput at the expense of a small delay degradation.

In [SRGM07] it is presented a new resource allocation mechanism, which is designed to improve the multiuser detection in NDMA protocols. The mechanism consists on adjusting the probability of false alarm of each user controlling the average number of active time-slots per user. By adjusting this PHY layer parameter, it is possible to regulate the typical delay degradation of a fixed resource allocation scheme, ensuring different levels of QoS.

In [SRGM09] it was presented a study on stability, throughput and delay properties of asymmetrical[11] non-blind[12] NDMA protocols under the assumption of imperfect collision-multiplicity[13] detection.

To overcome the difficulties mentioned above about the scalability of the ID sequences, [ZST02] explored the applicability of blind signal separation methods to the collision resolution problem. The signature of each colliding MTs packet has a Vandermonde form [14] and can be blindly estimated through rotational invariance techniques. For more examples in blind solutions refer to [OD06].

Other improvements and studies have been presented concerning NDMA aspects or variations. In Madueno and Vidal [MV05a] addressed the applicability of the NDMA protocol in an ad-hoc network. Guidelines for this design were presented as well as an investigation of the benefits of retransmission combining for broadcasting. As a result, it was proposed the Feedback Free - NDMA (FF-NDMA) protocol, which does not require feedback from the receiver. The solution went through forcing every node to transmit the same packet over a fixed number of R slots, and thus allowing resolving collisions of up to R packets without feedback. Different values of R in different setups were also studied.

A cross-layer analysis of the FF-NDMA for both Single Input Single Output (SISO) and MIMO configurations with orthogonal space time block codes is given in [MV05b]. In the SISO case, collision resolution is performed by repeating collided packets whereas in MIMO case both time and spatial diversity are exploited.

H-NDMA and S-NDMA

The combination of an H-ARQ technique with NDMA was proposed in [GPB+11], who

[11] configurations where the users have different channel and queuing statistics
[12] methods that require a known ID sequence to resolve different users collisions
[13] the number of contending users
[14] [SV06] for more information

named this mechanism by Hybrid-ARQ NDMA (H-NDMA). Basically, the access mechanism may force MTs to transmit a number of packet copies greater than the number of collided MTs to resolve collisions with low SNR. The BS defines the time slots, which are used by MTs to send data frames and controls the maximum number, J, of MTs that can use a given channel. The BS has also the duty of detecting collisions and to inform the MTs that it occurs through a broadcast downlink channel. After the involved MTs receive the collision information signal, they resend their packets. H-NDMA is considered a "slotted random access protocol with gated access", allocating the uplink slots in a organized way, which can be called a sequence of epochs, and using an SC-FDE scheme for uplink proposes. The BS transmits a synchronization signal (SYNC) to alert the MTs that a epoch is starting, so they are allowed to transmit at the next slot. MTs with new packets to transmit wait for the start of a new epoch. The epoch duration is defined by the number P of MTs that transmit data, and it was assumed that this number fits $1 \leq P \leq J$. When P MTs are linked to a collision, the base station requests P-1 initial retransmissions. Before each additional retransmission, an acknowledgment signal is sent by the BS to MTs, defining the ones that must retransmit at the next slot. This stipulated epoch ends when all packets are correctly received, or when the maximum number of additional retransmissions are sent. This research work concludes that H-NDMA has advantages in terms of network capacity and packet delay when compared with the classic NDMA and Hybrid-ARQ protocols. Scalability was another characteristic shown by this new protocol. The performance increases when more MTs transmit in a certain epoch.

NDMA was proposed for terrestrial cellular networks and never applied in a satellite environment. In a terrestrial network is a very flexible approach. It is more flexible than TDMA or CDMA because it is almost equivalent to a dynamic CDMA system where the spreading factor is equal to the number of colliding terminals. But for satellite networks, NDMA, has several drawbacks: MTs get to know the number of repetitions from a control packet received from the base station after a round trip time and has limited capability to handle errors due to channel effects. H-NDMA enhanced the error resilience capability by forcing additional packet transmissions after the initial packets. However, H-NDMA is unsuitable for satellite networks due to the multiple control packets required to control additional retransmissions and acknowledgments, which introduce incompatible delay and jitter for several kinds of QoS requirements.

In [GBD+12] a satellite demand assigned scheme using NDMA was developed, called Satellite NDMA (S-NDMA) protocol. S-NDMA is very similar to H-NDMA since it also uses hybrid protocols combining H-ARQ and NDMA. The first transmission in H-NDMA forces the MTs to transmit exactly P copies of each packet when P packets collide. Additional retransmissions of the packets are asked individually to improve the error resilience. What happens in S-NDMA can be seen in Figure 2.7. Besides the P copies of the packet in the first transmission, it is also added a group of slots, where in each slot there is a copy of the packet, to introduce redundancy and improve error resilience. The retransmissions may not send only one copy of the packet, but several, in groups of slots, which are organized in the uplink channel as super-frames.

S-NDMA can be regarded as an H-NDMA protocol especially designed to provide QoS guarantees for scenarios with a high RTT at a cost of some energy degradation due to the increased redundancy compared to H-NDMA.

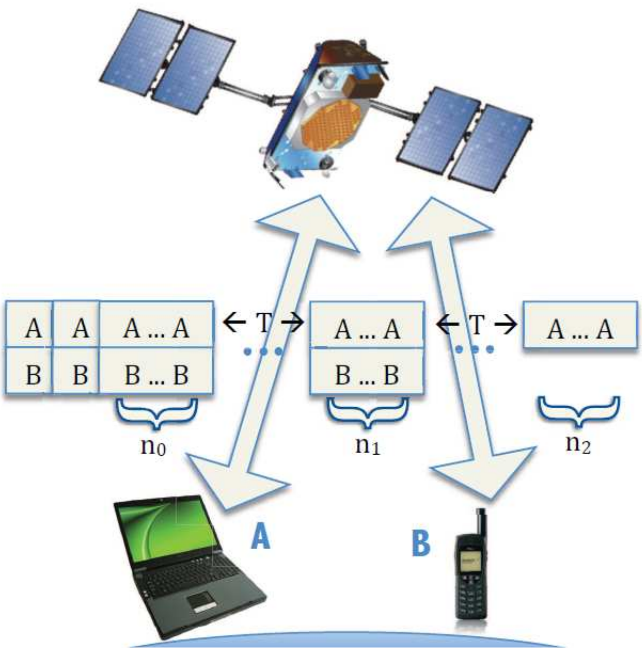


Figure 2.7: S-NDMA Demand Assigned scheme. Adapted from [GBD+12]

Chapter 3

Satellite-Random-NDMA Protocol

3.1 System Characterization

The system hereupon described consists of a network composed by a constellation of satellites and a set of Mobile Terminals (MTs). The scenario analyzed in this dissertation is the uplink communication between a set of MTs and a satellite which serves as Base Station (BS). MTs are low resource battery operated devices , whereas the satellite is a high resource device that runs a H-ARQ error control in real-time with a receiver that supports Multipacket Reception (MPR). The MTs employ SC-FDE in the uplink transmission, while the satellite employs OFDM for the downlink transmissions.

A scenario using a pure DAMA approach was already studied in [GBD+12]. However, demand assigned scenarios are only effective in environments comprising multiple MTs each having predictable types of traffic or high demanding loads, where a good spectral efficiency is indispensable. When, for instance, a large number of MTs generate infrequent packets for signaling transmission as well as for position reporting or other messaging applications, DAMA protocols fail to perform optimally, since they waste a lot of time to reserve and reassign transmission channels, increasing overall delay. This weakness, for bursty type of traffic, is the motivation to propose a random access protocol, which by nature, when compared to DAMA approaches, is able to provide lower transmission delay and limited complexity on the terminal side. So, in this chapter, a random access protocol is proposed named Satellite-Random NDMA (SR-NDMA).

In SR-NDMA, the uplink channel is composed by a sequence of time slots. Time slots can be Random Access (RA) or Scheduled access (SA) and are organized as a sequence of super-slots, defined by the satellite using a control downlink channel. A super-slot is a group of slots that carry a copy of one data packet; different super-slots carry different data packets.

This work considers an hybrid access approach: initially the MT transmits in one of the RA super-slots; when the data packet is not received, the satellite schedules further MTs retransmissions in the SA super-slots using a control downlink channel. A sequence of consecutive super-slots with the RA and SA slots that deal with the packet's recovery is denoted as an epoch. SR-NDMA differs from S-NDMA in the first access to the satellite, which is random. However, the remaining slots are scheduled following a similar approach to S-NDMA.

This protocol also considers a constrained bandwidth allocated to the set of MTs, B_{max} slots per RTT. The RA and SA slots are assigned in the allocated bandwidth, and their total number must not surpass B_{max}.

For the sake of simplicity, it is assumed in the following explanation that the MAC data packets associated to each MT have the same duration, the data packets on each slot arrive simultaneously and the MTs have a full-duplex radio. Besides defining time slots, the schedule specifies which MTs transmit in each slot (there can be more than one per slot) and the downlink channel still serves to select what transmission power is used by each MT. This control mechanism allows the optimization of the uplink transmissions using the algorithm specified in section 3.3.

3.1.1 Medium Access Control Protocol

The SR-NDMA protocol starts by organizing all the MTs associated to a satellite into groups who share a finite set of RA time slots defined by the satellite in the super-slots, using a downlink control channel. The maximum number of MTs in a group, J, is limited by the MPR capability of the receiver (the maximum number of MTs that can be received concurrently when using the multipacket receiver adopted). If the number of MTs in a group surpasses J, smaller groups of MTs have to be formed and the available bandwidth, is shared among the different sets of MTs. This way the complexity at the receiver is not increased but the total number of slots allocated to each group decreases[1].

The multiuser detection technique used by the satellite is capable of discerning all colliding data packets using user-specific orthogonal ID sequences for each MT of a group. Packets are transmitted within epochs, which include an initial transmission RA super-slot, followed by scheduled retransmissions in up to R SA super-slots, in result of reception failures. In the description below a MT set with J MTs is considered.

3.1.1.1 Random Access Mode

The RA mode is used for the initial transmission of each MAC data packet and the allocation of RA slots should satisfy bandwidth and QoS requirements. RA slots are allocated in groups of n_0 slots, **per packet**, which will name "RA super-slots". n_0 defines the number of redundant data packets transmissions used, in the RA phase, to improve error resilience and packet separation at the receiver. The number of RA super-slots allocated per RTT is denoted by N, meaning that up to N **different packets** can be sent, starting N concurrent independent epochs. In this work, it is assumed that the N RA super-slots are allocated periodically, with a period T equal to the highest RTT in a given MT set. The total number of RA slots allocated to the set of J MTs is $N \times n_0$ as can be seen in Figure 3.1. The number of packets actually transmitted depend on the number of packets in the queues of the MTs. In the example represented in Figure 3.1, three packets are transmitted (A, B, and C) by one MT and the remaining slots are idle.

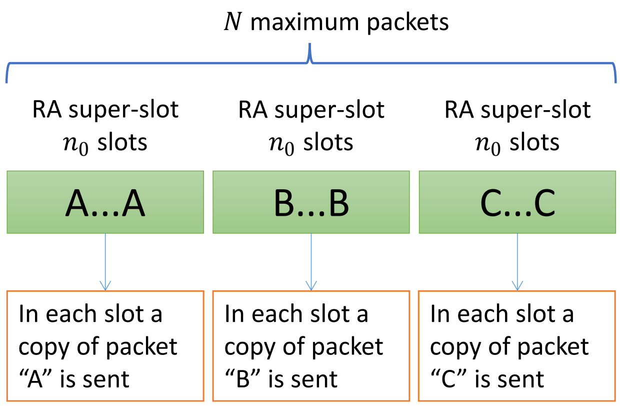


Figure 3.1: Maximum allocated RA slots and the structure of each RA super-slot

[1]The consequences of a smaller B_{max} value are seen in section 4.2.2

In order to transmit, the MT must wait for the next group of RA super-slots announced in the downlink channel and then randomly chooses up to N RA super-slots to transmit the packets in its queue. Therefore, the number of MTs that transmit in each RA super-slots is a random variable P , which satisfies $P \leq J$.

The maximum throughput of the system is given by $\frac{N \times J}{T}$, obtained when the maximum number of MTs supported by the MPR receiver, J, transmit. This limit should be above or equal to the total average load, λJ, where λ denotes the average packet generation rate per MT. When n_0 is set high enough, it is possible to transmit packets using only the initial RA transmissions. However, this also reduces the energy and bandwidth efficiency. They are maximized when the minimum number of packet copies are sent, which occurs when they are sent one by one, as in H-NDMA. Since this is not possible due to the high RTT, the value of n_0 should be balanced considering the QoS requirements.

3.1.1.2 Scheduled Access Mode

After the initial random transmission, the satellite becomes aware that P MTs transmitted. If the reception of all packets was not possible after the initial concurrent transmissions, additional SA slots are scheduled. From this point on, the protocol behaves like in S-NDMA [GBD$^+$12] and uses a pure DAMA approach for further retransmissions.

Individual packets not received at the satellite at retransmission super-slots l are scheduled for retransmission in super-slot $l + 1$. A packet transmission can be distributed over up to R retransmission super-slots, where the epoch ends. If all packets are correctly received before the Rth retransmission, the epoch ends as well. The packets that are not received after the $R + 1$ transmissions attempts (+1 taking into account the first random access transmission) fail its transmission. Depending on the QoS requirements, the packets can be transmitted again using a new epoch (remaining in the queue of the MT until then), or they can be dropped, for example, for real-time traffic applications.

The number of SA slots allocated for a packet in the lth retransmission super-slot of an epoch is denoted as $n_l^{(P)}$, where $l \leq R$ and P is the number of MTs that transmit in the epoch. The matrix with all $n_l^{(P)}$ values including the RA slots, is denoted as $\mathbf{n}$ and can be represented by equation (3.1). A vector of the matrix that is represented by $n^{(P)}$.

$$\mathbf{n} = \begin{vmatrix} \left[n_0, n_1^{(1)} \cdots, n_R^{(1)}\right] \\ \left[n_0, n_1^{(2)} \cdots, n_R^{(2)}\right] \\ \vdots \\ \left[n_0, n_1^{(J)} \cdots, n_R^{(J)}\right] \end{vmatrix} \tag{3.1}$$

The R value should be bounded by the delay requirements specified for the QoS traffic class that is being transmitted on a given epoch. When QoS specifications include a maximum delay, τ_{max} , R's value should be below $\lfloor \tau_{max}/T \rfloor$[2]; otherwise R can take higher values allowing an higher energy efficiency [GPB$^+$11]. Queueing delay is also influenced by the allocated slots per RTT, B_{max}. A low value may, limit the maximum number of packet copies that can be retransmitted per RTT. This affects the total cumulative queuing delay, increasing the total delay.

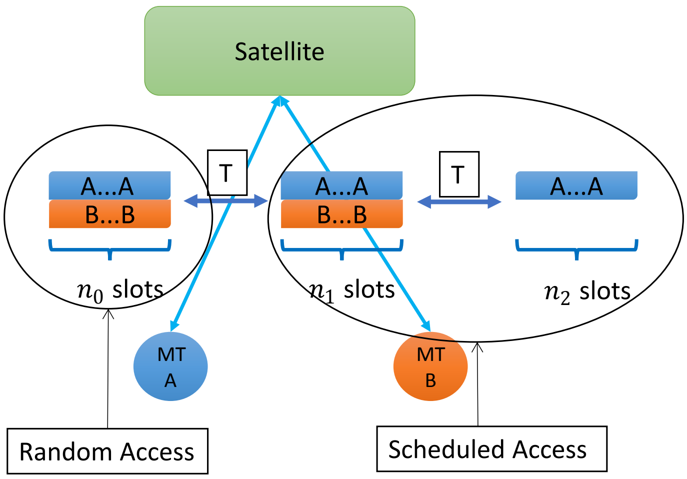


Figure 3.2: Two MTs sending a packet using SR-NDMA

In Figure 3.2, it can be seen an example of how packets are sent when $P = 2$ MTs try to transmit. In this example, the MTs transmit their packets in a super-slot with n_0 RA slots, defining the beginning of an epoch that may include two additional retransmissions ($R = 2$), corresponding to a vector of the matrix $\mathbf{n}$, defined by $\mathbf{n}^{(2)} = [n_0, n_1^{(2)}, n_2^{(2)}]$. MTs A and B want to transmit packet A and packet B respectively. The MTs start by transmitting the copies of their packets in n_0 RA slots of the first super-slot of the

[2] where $\lfloor x \rfloor$ defines the floor operation, that returns the maximum integer below or equal to x.

epoch. Since both MTs fail transmission, they use $n_1^{(2)}$ slots in the second super-slot to retransmit their respective packets. In the third super-slot only MT A transmits, because B managed to successfully send his packet after the second super-slot. The retransmissions are conditioned by the availability of enough free slots in the channel, which for heavy loads, the satellite may be forced to delay, until enough empty slots are available.

3.1.1.3 Operation of the SR-NDMA protocol

This section provides a full integrated description of how a packet is sent in SR-NDMA, using Figure 3.3 to illustrate it. This figure represents an example of how MTs send their packets. In this case $J = 3$ MTs, respectively A B and C, where MT A has four packets in his queue, B one and C two packets before the initial transmission. The maximum number of retransmissions is $R = 2$, so the vector with the number of slots per super-slot becomes, $\mathbf{n}^{(3)} = [n_0, n_1^{(3)}, n_2^{(3)}]$. The satellite allocates slots for the transmission of N packets ($N \times n_0$ slots).

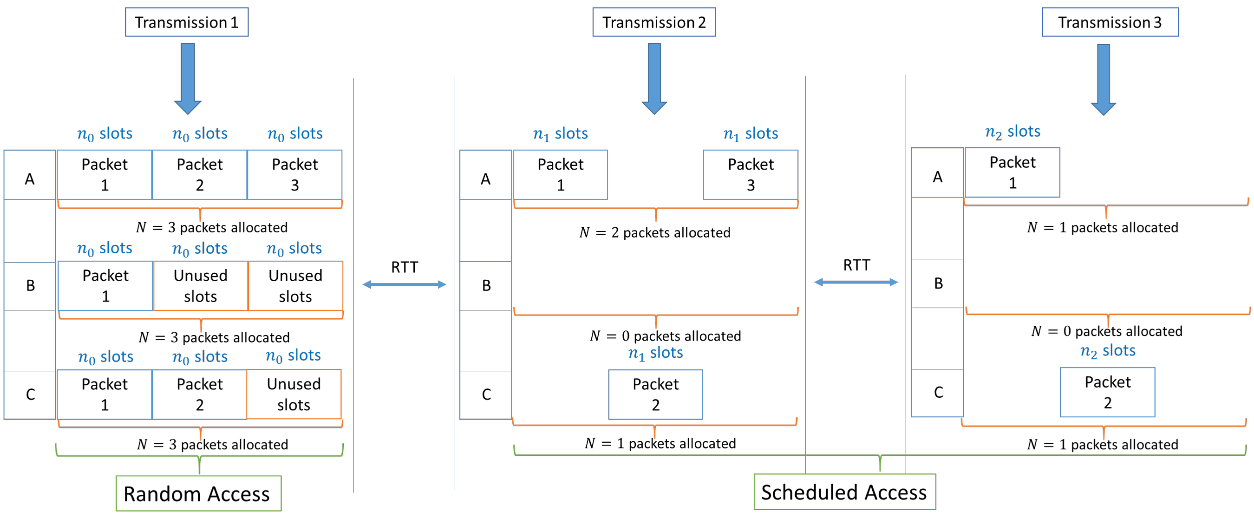


Figure 3.3: $P = 3$ users transmitting MAC data packets using SR-SNDMA

Due to the queue's occupancy, MT A sends the maximum packets possible, while B and C send the ones they have in queue using RA super-slots. After the first transmission, the protocol starts operating in a demand assigned scheme and only allocates the necessary slots, since it knows exactly the number of packets and the number of MTs that transmitted in the first access. It should be pointed out that MT detection errors might occur, invalidating the RA super-slot when this occurs. However, if the user ID contained in the packet header is transmitted using higher transmission power or using lower data rate than the data part, this probability can be made low compared to the packet error rate (PER). Therefore, the detection error problems may be reduced by properly configuring the system to address a PER requirement.

In the example, packet 2 of MT A is correctly received after the random access phase, but packets 1 and 3 are not received. Therefore, in the first packet retransmission (or transmission 2), MT A only resends packet 1 and packet 3, using a SA super-slot with $n_1^{(3)}$ slots. MT C only resends packet 2, using a super-slot with $n_1^{(3)}$ slots. MT B does not resend any packet, since its only packet was received without errors. In the last retransmission, A only resends packet 1 because packet 3 was successfully received, while C still is trying to send packet 2, both using super-slots with $n_2^{(3)}$ slots.

SR-NDMA differs from S-NDMA because its configuration parameter $\mathbf{n}$ defines a matrix, since all possible values of $0 < P \leq J$ must be specified, whereas for a pure DAMA approach P is scheduled by the satellite. SR-NDMA trades-off a lower delay for a lower bandwidth efficiency, since some initially allocated RA super-slots might not be used if transmitted individually. At the end of this chapter, the problem of defining the optimal values for the matrix $\mathbf{n}$, given a constrained bandwidth allocated to the group of MTs, B_{max} slots per RTT, is addressed.

3.1.2 Multipacket Reception - Receiver Structure

In the satellite system considered, the uplink transmission is done by employing SC-FDE. In this dissertation it is also considered the use of the linear multipacket receiver running the MMSE detector (see section 2.5.1.1) for SC-FDE systems proposed in [GDB$^+$11], and represented in Figure 3.4. An analytical expression for the Packet Error Rate (PER) was obtained in [GDB$^+$11] and briefly described in this section.

Nodes contend for the channel at each epoch and, as expected, collisions might occur. A data block, of N_s symbols, transmitted by a user p, can be expressed, on the time domain, as $\{s_{n,p}; n = 0, ..., N_s - 1\}$, and its correspondent on the frequency domain as $\{S_{k,p}; k = 0, ..., N_s - 1\}$. At the receiver, at the frequency domain, the received signal from multiple MTs for a given transmission r is given in equation (3.2). A block diagram of the receiver is given in Figure 3.4.

$$Y_k^{(r)} = \sum_{p=1}^{P} S_{k,p} H_{k,p}^{(r)} + N_k^{(r)}, \tag{3.2}$$

where $H_{k,p}^{(r)}$ is the channel response for the pth MT at the rth transmission, which is zero when the MT does not transmit. $N_k^{(r)}$ is the channel noise for the rth transmission. The total number of slots allocated until the $l+1$th super-slot for an epoch with P MTs is given by $\zeta_l^{(P)} = \sum_{i=0}^{l} n_i^{(P)}$. Considering that P MTs transmit $\zeta_l^{(P)}$ times and $0 \leq l \leq R$, then the received $\zeta_l^{(P)}$ transmissions are characterized as follows:

$$\begin{aligned} \mathbf{Y}_k &= \mathbf{H}_k^T \mathbf{S}_k + \mathbf{N}_k \qquad (3.3) \\ &= \begin{bmatrix} H_{k,1}^{(1)} & \dots & H_{k,P}^{(1)} \\ \vdots & \ddots & \vdots \\ H_{k,1}^{(\zeta_l^{(P)})} & \dots & H_{k,P}^{(\zeta_l^{(P)})} \end{bmatrix} \begin{bmatrix} S_{k,1} \\ \vdots \\ S_{k,P} \end{bmatrix} + \begin{bmatrix} N_k^{(1)} \\ \vdots \\ N_k^{(\zeta_l^{(P)})} \end{bmatrix} . \end{aligned}$$

For a given MT p, the estimated signal at the frequency domain is

$$\tilde{S}_{k,p} = \begin{bmatrix} F_{k,p}^{(1)} & ... & F_{k,p}^{(\zeta_l^{(P)})} \end{bmatrix} \mathbf{Y}_k = \mathbf{F}_{k,p}^T \mathbf{Y}_k \ . \tag{3.4}$$

$\mathbf{F}_{k,p}$ corresponds to the feedforward coefficients of the proposed system, and these are chosen to minimize the mean square error (MSE) $2\sigma_{E_{k,p}}^2$ for a MT p. Considering that $\boldsymbol{\Gamma}_p = [\Gamma_{p,1} = 0, \dots, \Gamma_{p,p} = 1, \dots, \Gamma_{p,P} = 0]^T$, $2\sigma_{E_{k,p}}^2$ is evaluated as follows

$$\begin{aligned} 2\sigma_{E_{k,p}}^2 &= \mathbb{E}\left[|\tilde{S}_{k,p} - S_{k,p}|^2\right] \\ &= \left(\mathbf{F}_{k,p}^T \mathbf{H}_k^T - \boldsymbol{\Gamma}_p\right) \mathbb{E}\left[\mathbf{S}_k \mathbf{S}_k^H\right] \left(\mathbf{F}_{k,p}^T \mathbf{H}_k^T - \boldsymbol{\Gamma}_p\right)^H \\ &\quad + \mathbf{F}_{k,p}^T \mathbb{E}\left[\mathbf{N}_k \mathbf{N}_k^H\right] \mathbf{F}_{k,p}^* \ . \qquad (3.5) \end{aligned}$$

Regarding $\mathbb{E}\left[|S_{k,p}|^2\right] = 2\sigma_S^2$ and $\mathbb{E}\left[\left|N_k^{(r)}\right|^2\right] = 2\sigma_N^2$, the optimal $\mathbf{F}_{k,p}$ is obtained by applying the method of Lagrange multipliers to (3.5), which results[3]

$$\mathbf{F}_{k,p} = \left(\mathbf{H}_k^H \mathbf{H}_k + \frac{2\sigma_N^2}{2\sigma_S^2} \mathbf{I}_{\varsigma_l^{(P)}} \right)^{-1} \mathbf{H}_k^H \mathbf{\Gamma}_p \left(1 - \frac{1}{2N\sigma_S^2} \right) . \tag{3.6}$$

From equations (3.5) and (3.6) results

$$\sigma_p^2 = \frac{1}{N_s^2} \sum_{k=0}^{N-1} \mathbb{E}\left[\left| \tilde{S}_{k,p} - S_{k,p} \right|^2 \right] . \tag{3.7}$$

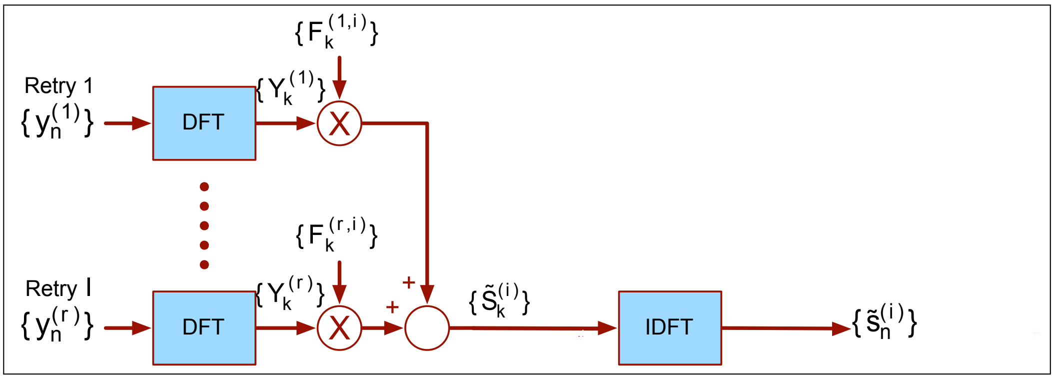


Figure 3.4: Structure of the receiver for the l packet reception

For a QPSK constellation and being $Q(x)$ the well known Gaussian error function, the Bit Error Rate (BER) of a given user p is

$$BER_p \simeq Q\left(\frac{1}{\sigma_p}\right) . \tag{3.8}$$

For an uncoded system with independent and isolated errors, the Packet Error Rate (PER) for a fixed packet size of M bits is

$$PER_p \simeq 1 - (1 - BER_p)^M . \tag{3.9}$$

Figure 3.5 depicts the BER over E_b/N_0 for increasing copies of the original data packet. It shows that for a given E_b/N_0 the BER diminishes considerably, when a higher number of retransmissions is involved. Therefore, the packet reception is improved, when there are more copies of the packet available at the receiver. This receiver uses the Equal

[3] It should be noted that σ_s^2 and σ_N^2 denote the variance of the real and imaginary parts of $S_{k,p}$ and $N_k^{(r)}$ respectively.

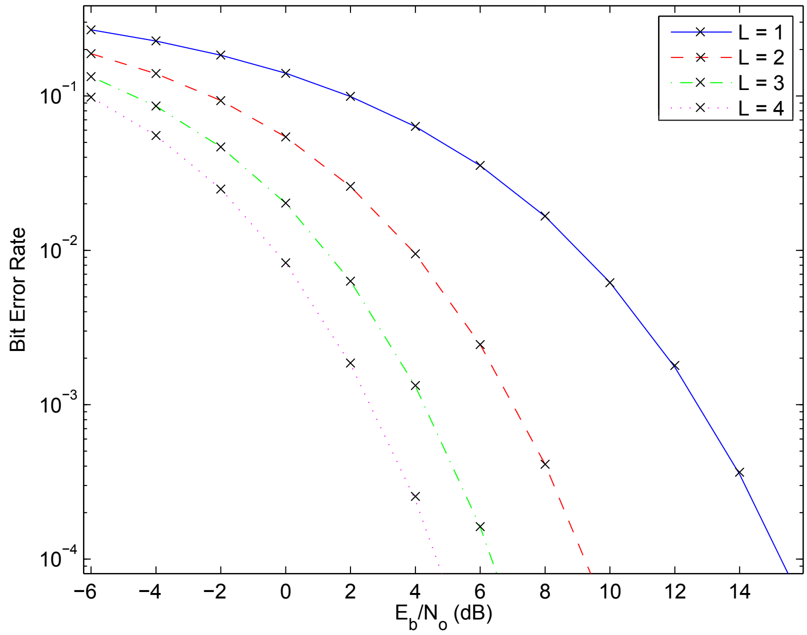


Figure 3.5: BER performance for the adopted receiver. Adapted from [GDB+11]

Channel with Shifted Packet (SP) technique [DMB+09] for the different retransmissions in order to reduce the correlation between packet copies, thus improving packet separation. This technique allows resolving equation (3.6) even when the channel does not change, as long as the channel is selective in the frequency. These shifts in frequency applied to the blocks translate into using different frequencies to carry different parts of the packet in each retransmission, making them independent.

3.2 Analytical Performance Model

This section proposes an analytical model for the delay, throughput, channel utilization and energy consumption of the uplink channel on SR-NDMA system, which quantifies the influence of the PER, transmission power and the distance to the satellite. The following modeling conditions were considered:

- The BS is able to discern up to J colliding packets. So, it is assumed that P MTs access simultaneously the channel during an epoch with $P \leq J$.

- Perfect average power control and time advance mechanisms exist in order to compensate different propagation times and attenuations. Perfect average control leads to a uniform average E_b/N_0 value for all MTs at the satellite.

- Feedback delayed by a delay $T_f \geq RTT$: After the end of the transmission of each packet, the MTs are informed of the outcome of the transmission with a delay of T_f. It is also assumed that the number of MTs colliding is precisely determined by the BS (i.e. detection errors are not considered) and that no error occurs in the control channels.

- Poisson arrivals: It is assumed that the buffer of each of J MTs receives packets that are generated according to an homogeneous Poisson source, with a generating rate λ.

- M/D/N/Q_{max} queue model in the access to the RA super-slots: The M stand for a Markovian Poisson arrival process, the D for being a deterministic distribution, N is the number packets that can be serviced at time and Q_{max} defines the size of the packet queue in the MTs. The queue model also considers that if a packet is not received after the end of the epoch is dropped. With a limited queue size, if the queue is full (saturated) new packets are dropped and do not receive service (i.e. are not transmitted).

- Constrained allocated bandwidth: It is assumed that the satellite only allocates a maximum number of slots per RTT, given by B_{max}. It is maximum possible value is equal to RTT (the number of slots contained in RTT). The RTT has a duration of T (in slots) given by the MT furthest of the satellite, in a set of MTs .

A Markov chain model is proposed in this section to model the queue's steady state distribution, which supports the performance models that follow.

3.2.1 Scheduled Access Phase

The transmission of a packet during the scheduled access phase applies an H-ARQ approach for scheduled NDMA transmissions, which is similar to S-NDMA. Given that the number of MTs that transmitted during the initial RA super-slot is already known during the SA phase, the analytical performance model of S-NDMA [GBD+12] applies, and can be used to build the SR-NDMA model. It was shown in [GBD+12] that the performance of the scheduled access phase only depends on the initial number of MTs that transmitted, P, on the slot allocation vector used, $n^{(P)}$, and on the E_b/N_0 value at the reception. The packet transmission behavior can be defined by a hidden Discrete Time Markov Chain (DTMC) with a random state vector denoted by $\Psi^{(P,R)}$.

$$\Psi^{(P,R)} = \{\psi_k^{(P,R)}, k = 0...R\} \tag{3.10}$$

Equation (3.10) defines the state space of the number of MTs whose packets were successfully received and stopped transmitting at the end of the retransmission super-slot $k = 0, ..., R$, for an epoch with P MTs and using up to R retransmissions. Considering only the retransmission super-slots up to $l \leq R$, the state space of $\Psi^{(P,l)}$ is denoted by the set $\Omega_P^{(l)}$. It contains all the vector elements $K^{(l)}$ (with dimension $l + 1$) that satisfy equation (3.11).

$$\sum_{k=0}^{l} K_k^{(l)} = P \tag{3.11}$$

Each state $\Psi^{(P,l)} = \{\psi_0^{(P,l)} = K_0^{(l)}, ..., \psi_l^{(P,l)} = K_l^{(l)}\}$ defines the set of transmission sequences where $K_0^{(l)}$ MTs stopped transmitting after the initial RA super-slot (super-slot 0), $K_1^{(l)}$ MTs stopped packet transmission after the SA super-slot 1, and so on until $K_l^{(l)}$ MT that transmitted in the SA super-slot l.

The average PER at the $(l+1)$th super-slot with P MTs is denoted by $PER_P\left(\Psi^{(P,l)}\right)$, and for the proposed receiver it is calculated using equations (3.6) and (3.9), where H_k is a $(\zeta_l^{(P)}) \times P$ matrix with the channel response. This matrix has zero coefficients for the epoch's slots where the MTs did not transmit. The probability mass function for $\Psi^{(P,l)}$ can be calculated using equation (3.12) [GBD+12].

$$Pr\left\{\Psi^{(P,l)} = K^{(l)}\right\} =$$
$$\frac{P!}{\prod_{k=0}^{l} K_k^{(l)}!}\left(\prod_{i=0}^{l-1} PER_P\left(\Psi^{(P,i)}\right)\right)^{K_l^{(l)}} \times$$
$$\prod_{j=0}^{l-1}\left(\left(1 - PER_P\left(\Psi^{(P,j)}\right)\right)\prod_{i=0}^{j-1} PER_P\left(\Psi^{(P,i)}\right)\right)^{K_j^{(l)}}, \tag{3.12}$$

where

$$\psi_m^{(P,i)} = \begin{cases} K_m^{(l)} & m < i \\ \sum_{k=i}^{l} K_k^{(l)} & m = i \end{cases}.$$

A packet is not correctly received after l retransmission super-slots if it is transmitted on all super-slots and its reception fails at the last retransmission. Consequently, the expected number of packets received with error during an epoch until the lth retransmission super-slot is given by equation (3.13).

$$\mathbb{E}\left[err\left(\Psi^{(P,l)} = K^{(l)}\right)\right] = K_l^{(l)} PER_P\left(\Psi^{(P,l)}\right). \tag{3.13}$$

Assuming that packet failures are independent, the packet error probability for an epoch $\Omega_P^{(l)}$ (with up to l retransmission super-slots) is given by equation (3.14).

$$P_{err}\left(\Omega_P^{(l)}\right) = \sum_{K^{(l)} \in \Omega_P^{(l)}} \frac{1}{P} \times Pr\left\{\Psi^{(P,l)} = K^{(l)}\right\} \mathbb{E}\left[err\left(\Psi^{(P,l)} = K^{(l)}\right)\right] \tag{3.14}$$

For a given E_b/N_0, the PER can be reduced by increasing the number of retransmissions of a packet. For the same number of retransmissions, the PER decreases when the number of interfering concurrent transmissions is decreased. So, when a MT transmits $\zeta_l^{(P)}$ copies of a packet, an upper bound for the packet error probability of an epoch can be calculated using equation (3.15) [GBD+12].

$$P_{err}\left(\Omega_P^{(R)}\right) \leq PER_P\left(\Psi^{(P,R)} = [0,0,...,P]\right) \tag{3.15}$$

The average number of slots used by a MT to transmit a packet during an epoch where $\Psi^{(P,R)} = K^{(R)}$ is given in equation (3.16).

$$tx\left(\Psi^{(P,R)} = K^{(R)}\right) = \frac{1}{P}\sum_{l=0}^{R}(\zeta_l^{(P)})K_l^{(R)} \tag{3.16}$$

The expected number of slots used, given in equation (3.17), is calculated using a Bayesian approach for all $K^{(R)}$ in $\Omega_P^{(R)}$.

$$\mathbb{E}\left[tx\left(\Omega_P^{(R)}\right)\right] = \sum_{K^{(R)}\in\Omega_P^{(R)}} Pr\left\{\Psi^{(P,R)} = K^{(R)}\right\} tx\left(\Psi^{(P,R)} = K^{(R)}\right) \tag{3.17}$$

3.2.2 System's Steady State

In order to properly configure the SR-NDMA system, it is necessary to know the average number of RA and SA slots that are being used by each set of MTs. Assuming an homogeneous system with uniform load, where the MTs packet transmissions are i.i.d. (independent and identically distributed), it is possible to analyse the behaviour of the entire system modelling a single MT behaviour. This section proposes a DTMC model for the average number of packets being transmitted at each retransmission during an epoch.

When a MAC data packet arrives at a MT's queue, it waits until its first transmission in a RA super-slot, defining the beginning of a new epoch. New epochs may be started separated by a RTT, time T. Therefore, there might be concurrent epochs running for each node, aligned with the generation of the RA super-slots. Assuming that packet arrivals are defined by a Poisson process with an average load of λ, the number of packets arriving between the groups of RA super-slots is defined by equation (3.18).

$$Pr\{A = k\} = \frac{(\lambda T)^k e^{-\lambda T}}{k!}, \qquad k \geq 0. \tag{3.18}$$

Let us denote q_m as the number of packets in the MTs' queue in the beginning of the mth occurrence of the RA super-slot in the uplink channel. The set of RA super-slots is repeated every T slots, and has a service capacity of N packets per RTT (as seen in section 3.1.1.1). Assuming that the queue has a maximum capacity of Q_{max} packets and the lost packets are dropped. The q_m distribution can be defined by the DTMC in equations (3.19) and (3.20).

$$\begin{aligned} Pr\{q_{m+1} = Q\} = Pr\{q_m < N\}\, Pr\{A = Q\} + \\ \sum_{i=0}^{min(Q_{max}-N,Q)} Pr\{q_m = N + i\}\, Pr\{A = Q - i\}\ , \end{aligned} \tag{3.19}$$

for $Q < Q_{max}$, and by

$$Pr\{q_{m+1} = Q_{max}\} = 1 - \sum_{i=0}^{Q_{max}-1} Pr\{q_{m+1} = i\}\ . \tag{3.20}$$

According to [BGdMT98], the individual MT queue's utilization, ρ_R, is defined by equation (3.21). Limited size queues are not inherently unstable, namely the queue always converges to an equilibrium state. However, they may introduce massive loss of packets due to queue drops, when the system is saturated (i.e. its utilization is above one). Therefore, to reduce the packet drops, ρ_R, should be below 1. This DTMC converges to a steady state distribution, denoted by ξ, which can be calculated using an iterative numerical method [BGdMT98].

$$\rho_R = \frac{\lambda T}{N}\ . \tag{3.21}$$

Let X^0 denote the number of packets being transmitted at the initial N sets of the RA super-slots, and X^l denote the number of packets being retransmitted at lth SA super-slots, with $1 < l \leq R$. The number of packets being sent in the RA super-slots is directly related with the number of packets in the queue, and is limited by N. Therefore, the probability mass function of X^0 is given by equation (3.22).

$$Pr\left\{X^0 = x\right\} = \begin{cases} Pr\left\{\xi = x\right\}, x < N \\ Pr\left\{\xi \geqslant N\right\}, x = N \end{cases} . \tag{3.22}$$

The probability of a MT transmitting a packet in one of the N RA super-slots, P_{tx}, is given in equation (3.23).

$$P_{tx} = \frac{\mathbb{E}\left[X^0\right]}{N} = \frac{1}{N}\sum_{x=1}^{N} x\, Pr\left\{X^0 = x\right\} \tag{3.23}$$

The sending of packets in subsequent transmissions depend on their success on previous transmissions. Therefore it is introduced the probability of v packets having success when x packets are transmitted during the lth retransmission, $Pr\left\{V^l = v \mid X^l = x\right\}$, for $0 \leq l \leq R$. This probability depends on the number of MTs contending in each epoch. Assuming that epochs are independent and that errors are uncorrelated with q_m, the number of MTs contending in each epoch and the number of packets successfully received can be approximated by binomial distributions, and we get equation (3.24).

$$Pr\left\{V^l = v \mid X^l = x\right\} = \sum_{p=0}^{J-1} f_b\left(J-1, p, P_{tx}\right) f_b\left(x, v, 1 - P_{err}\left(\Omega_{(p+1)}^{(l)}\right)\right), \tag{3.24}$$

where $f_b(J,k,p) = \binom{J}{k} p^k (1-p)^{J-k}$ denotes the Binomial probability mass function. The PER can be calculated using equation (3.14), and depends on $\mathbf{n}$ and E_b/N_0 values.

The conditional probability mass function of the number of packets retransmitted at the lth super-slot can be obtained using equation (3.24). The probability mass function of the number of packets transmitted for the lth time in an epoch is defined recursively using equation (3.25).

$$Pr\left\{X^{l+1} = x \mid X^l\right\} = \sum_{i=x}^{N} Pr\left\{X^l = i\right\} Pr\left\{V^l = i - x \mid X^l = i\right\} . \tag{3.25}$$

The transmitted number of packets has an average value of

$$E\left[X^l\right] = \sum_{x=0}^{N} x\,Pr\left\{X^l = x\right\}. \tag{3.26}$$

The expected number of packets successfully received at retransmission l, $l < R$, $E\left[V^l\right]$, is simply the number of packets that were successfully transmitted during super-slot l, i.e.,

$$\mathbb{E}\left[V^l\right] = \mathbb{E}\left[X^l\right] - \mathbb{E}\left[X^{l+1}\right]\ . \tag{3.27}$$

The expected number of packets successfully received in the last retransmission R is given by equation (3.28).

$$\mathbb{E}\left[V^R\right] = \sum_{v=1}^{N} v \sum_{x=0}^{N} Pr\left\{X^R = x\right\} Pr\left\{V^R = v \,|\, X^R\right\} \tag{3.28}$$

3.2.3 Throughput and Channel Utilization Analysis

The throughput is calculated taking into account the number of packets received with success during a RTT (T), for all retransmission stages. Considering all J MTs, it is given by

$$S = \frac{J\left(\left(\sum_{l=0}^{R-1} \mathbb{E}\left[V^l\right]\right) + \mathbb{E}\left[V^R\right]\right)}{T} \tag{3.29}$$

The measured throughput results from packets lost due to channels errors and queue overflow, related to the queue size limit of Q_{max}. From S, it is possible to calculate the total packet loss ratio, ε, given by

$$\varepsilon = 1 - \frac{S}{\lambda J}\ . \tag{3.30}$$

In this work it is also considered that B_{max} slots are allocated to a given MT set. A relevant parameter is the channel utilization ratio, ρ_b, defined as the ratio of the average number of slots used over the number of allocated slots (B_{max}) per RTT, which should always be below 1. ρ_b can be calculated using equation (3.31), which counts the total RA slots allocated and the average number of SA slots scheduled per RTT. The number of SA slots used for a given number of packets transmitted at the lth super-slot for P MTs is defined by the configuration parameters $n_l^{(P)}$ and the average value is influenced by the probability of having P MTs transmitting per epoch.

$$\rho_b = \frac{\left(Nn_0 + \sum_{l=1}^{R} \left(\mathbb{E}\left[X^l\right] \sum_{p=0}^{J-1} f_b(J-1, p, P_{tx})\, n_l^{(p+1)} \right) \right)}{B_{max}} \tag{3.31}$$

3.2.4 Energy Consumption

Multiple MTs transmit packets to the uplink channel, which arrive at the BS with an average reception power P_r due to the assumption of perfect average power control. A simplified energy model is proposed in this section, which considers only the transmission energy and neglects the energy consumption related to the circuit and algorithm complexity [CGB05].

The transmission power per packet, P_p, for each MT includes the transmission signal power, p_t and the amplifier's power consumption P_{amp}, where $P_{amp} = \beta p_t$. The transmission power per packet can be defined as

$$P_p = (1+\beta)p_t \ , \ \beta = \frac{\alpha}{\eta} - 1. \tag{3.32}$$

η is the drain efficiency of the radio frequency power amplifier and α is the Peak-to-Average-Ratio (PAR) which is dependent on the modulation scheme and the associated constellation size. The energy for each packet transmission E_p is then given by,

$$E_p = (1+\beta)p_t T_{on} \frac{E_b}{N_0}, \tag{3.33}$$

since $E_p = P_p \times T_{on}$. T_{on} is the packet transmission time for a total of M bits and E_b/N_0 is the bit energy over the noise ratio, measured at the satellite. These values are defined further ahead in section 4.1.3, as well as the remaining power transmission parameters.

The energy per useful packet, denoted by $EPUP$, measures the average energy necessary to successfully receive a packet at the satellite. The expression in equation (3.34) depends on the average number of slots that the MT transmits during an epoch, the success probability for the packets transmitted, and the packet's transmission energy E_p (equation (3.33)). Again, a Bayesian approach is followed to account the influence of the number of MTs transmitting per epoch.

$$EPUP = \sum_{p=0}^{J-1} f_b\left(J-1, p, P_{tx}\right) \frac{\mathbb{E}\left[tx\left(\Omega_{p+1}^{(R)}\right)\right] E_p}{1 - P_{err}\left(\Omega_{p+1}^{(R)}\right)} \tag{3.34}$$

3.2.5 Delay and Jitter Analysis

The packet transmission delay of SR-NDMA, $\tau(J, \lambda, \mathbf{n})$, includes an initial time until reaching the first group of RA super-slots, where the MT starts to transmit, the time in queue until a packet is being transmitted at the RA slots, and the total service time (including the propagation time and the possible retransmissions using SA slots).

3.2.5.1 Initial Time

The initial time until the MT starts to transmit is illustrated in Figure 3.6. Here, it is assumed an uniform distribution of the packet arrivals in the interval between RA super-slots, therefore, the expected initial time is $\mathbb{E}[\gamma] = T/2$.

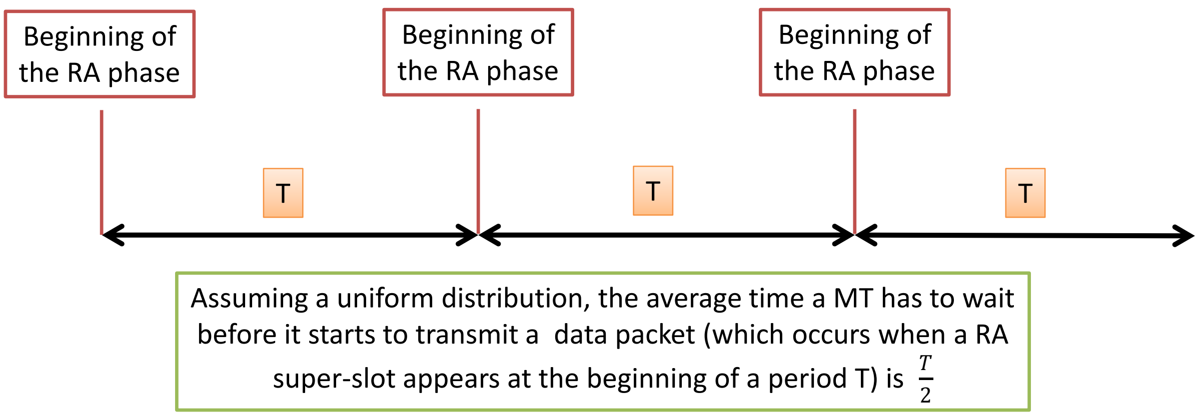


Figure 3.6: The time a MT has to wait in average to start its transmission.

3.2.5.2 Queuing delay until RA transmission

The time a packet spends in queue until the initial transmission in the RA phase depends on several factors. Firstly, it relies on N, the maximum number of packets that can be sent per RTT. So, in the RA phase the MT can send up to N different packets distributed over Nn_0 slots, as seen in Figure 3.7, and the remaining packets are queued waiting for future RA super-slots when the queue length is above N. When N increases, more packets can be sent at a time, decreasing the time packets spend in queue until they are transmitted.

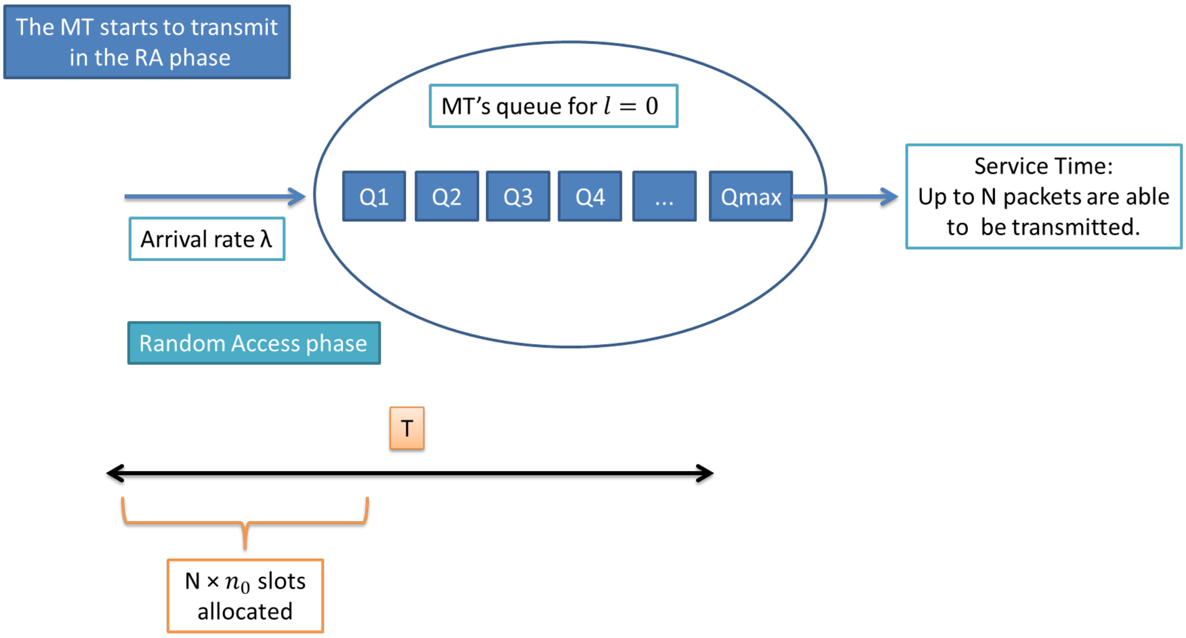


Figure 3.7: Random access, $l = 0$. The time a packet spends on queue depends on N.

The time in the queue at the RA phase, waiting for the first transmission has an expected value given by equation (3.35).

$$\mathbb{E}\left[\varpi\right] = \sum_{Q=1}^{MAX} \left(Pr\left(\xi = Q\right) \left\lfloor \frac{Q-1}{N} \right\rfloor \times T + \frac{1}{2} \times T \left(Q - \left\lfloor \frac{Q-1}{N} \right\rfloor \times N \right) \right) \tag{3.35}$$

3.2.5.3 Service Time

The service time depends mainly on which super-slot of an epoch the packet is correctly received, but is also affected by scheduling delay relatively to the successive retransmissions. The service time includes the epoch expected duration in equation (3.37) plus an additional delay due to scheduling constraints, due to the limits in the number of slots available per MT set (B_{max}).

The epoch duration is defined by the last slot with a packet transmission and is conditioned by the availability of enough SA slots in the channel to transmit the $n_l^{(P)}$ copies of the packets during the lth retransmission. A lower bound for the epoch duration can be obtained by ignoring any queueing that results from the B_{max} bandwidth constraint and considering only the propagation time. The epoch duration for an epoch where $\Psi^{(P,R)} = K^{(R)}$ is denoted by $d\left(\Psi^{(P,R)} = K^{(R)}\right)$ and shown in equation (3.36).

$$d\left(\Psi^{(P,R)} = K^{(R)}\right) \geq d_{prop}\left(\Psi^{(P,R)} = K^{(R)}\right) = \frac{T}{2} + T\left(\min_{l}\left\{\forall k > l, K_k^{(R)} = 0\right\}\right), \tag{3.36}$$

where, by definition

$$K_{R+1}^{(R)} = 0.$$

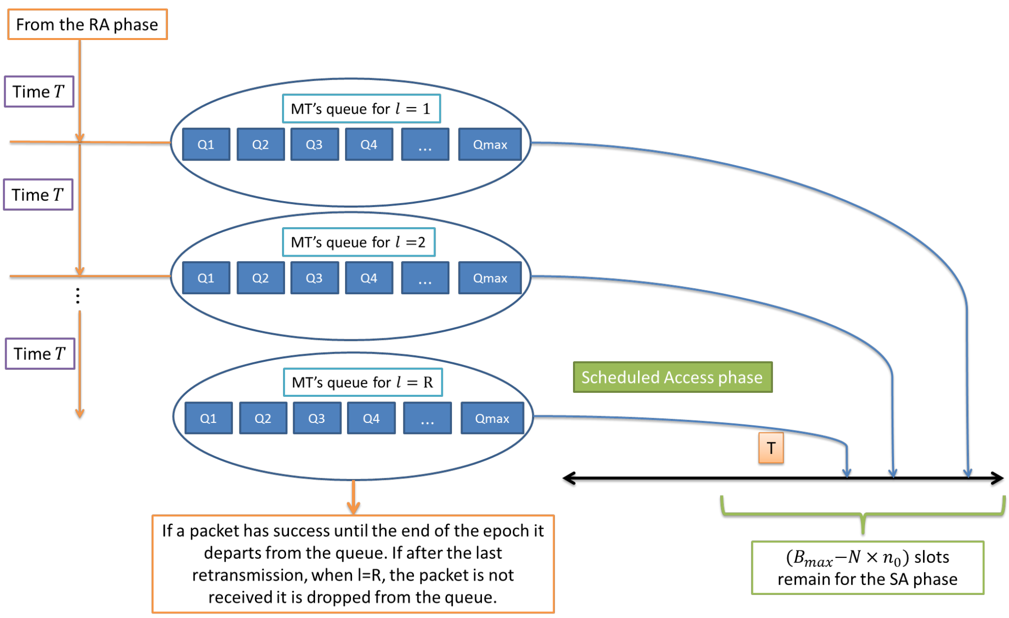


Figure 3.8: Scheduled access, $l = 1...R$. The time the remaining packets last in queue depend mainly on B_{max} and the optimization procedure for scheduling the slots.

In order to account for the bandwidth constrain that results from B_{max}, we have to consider that if packets fail to be received in the RA phase, additional slots are scheduled in the remaining slots, $B_{max} - Nn_0$, as seen in Figure 3.8. The number of retransmissions is equal or below R, so during the different retransmissions the packets suffer from distinct queue delays. The scheduling algorithm used by the satellite to allocate SA slots to the packets pending of further retransmissions gives priority to the older packets over the newer ones, to limit the total queueing delay a packet may suffer. If a packet is received with success during an epoch, it departs from queue. If a packet it not received with success at the end of the epoch (when $l = R$), it is assumed a QoS constraint that defines a bound delay, so the packet is also discarded from the queue.

The expected duration of the epoch due to propagation time (ignoring the queueing introduced by B_{max}), can be obtained using equation (3.37).

$$\mathbb{E}\left[d_{prop}\left(\Omega_P^{(R)}\right)\right] = \sum_{K^{(R)} \in \Omega_P^{(R)}} Pr\left\{\Psi^{(P,R)} = K^{(R)}\right\} d_{prop}\left(\Psi^{(P,R)} = K^{(R)}\right) \tag{3.37}$$

The service time, is then given by equation (3.38).

$$\mathbb{E}\left[ST\right] \approx \sum_{p=0}^{J-1} f_b\left(J-1, p, P_{tx}\right) \mathbb{E}\left[d_{prop}\left(\Omega_{p+1}^{(R)}\right)\right] \ . \tag{3.38}$$

3.2.5.4 Total delay and jitter

A lower bound for the delay can be defined adding all the components previously explained, and is given in equation (3.39).

$$\mathbb{E}\left[\tau(J, \lambda, \mathbf{n})\right] \geq \mathbb{E}\left[\gamma\right] + \mathbb{E}\left[\varpi\right] + \mathbb{E}\left[ST\right] \ . \tag{3.39}$$

An approximated value for the jitter is obtained by considering it the standard deviation of the delay, as shown in equation (3.40). The complete deduction of $\mathbb{E}\left[\tau^2\right]$ is given in Appendix A.

$$\sigma_j \approx \sqrt{\mathbb{E}\left[\tau^2\right] - \mathbb{E}\left[\tau\right]^2} \tag{3.40}$$

3.3 Optimization Procedure

Supporting QoS constrained traffic is a major challenge for a random access MAC protocol in a satellite network. For a given QoS class these constraints typically include the specification of an upper PER threshold (PER_{max}) and a delay bound (D_{max}) [AMCV06]. The objective should be to minimize the EPUP and to provide these guarantees supporting the total MTs load λJ and using only B_{max} slots per RTT. SR-NDMA's performance is influenced by the average E_b/N_0 measured at the satellite, by the values defined for the matrix parameter $\mathbf{n}$, the parameters N, R, Q_{max} and by the network load. This section illustrates how such optimization problem can be solved.

The number of RA super-slots per RTT, N, is greater or equal to $\lceil \lambda T \rceil$ ($\lceil x \rceil$ is the ceiling operation, which returns the maximum integer above or equal to x), to allow $\rho_R < 1$. N can be increased above this value to reduce packet drops and the queueing delay, leading however to some energy inefficiency since the number of SA slots is reduced.

A D_{max} delay bound introduces a limitation in the number of super-slots that can be part of an epoch. Ignoring the queueing delay, at the first optimization step, the dominant component of equation (3.39) is due to the product $R \times T$ in the service time (equation (3.37)), where T depends on the altitude of the satellite orbits. Its value is calculated in section 4.1.3. Therefore, R must satisfy

$$R \leq \left\lfloor \frac{D_{max} - \frac{T}{2} - \mathbb{E}\left[\varpi\right]}{T} \right\rfloor \leq \left\lfloor \frac{D_{max} - \frac{T}{2}}{T} \right\rfloor \ , \tag{3.41}$$

For example, when $R = 0$, all slots must be allocated as RA, N is selected to satisfy $\rho_R < 1$, and n_0 must be set to the highest value of $N \times \zeta_R^J$ that matches the available B_{max}, to minimize the EPUP, satisfying $\varepsilon < PER_{max}$.

For $R > 0$, for a given N such that $\rho_R < 1$, it is necessary to calculate the optimal matrix $\mathbf{n}^\star$ and $E_b^\star/N_0$. The optimization problem can be defined as:

$$\begin{aligned} \text{Minimize:} \quad & EPUP\left(\mathbf{n}^{\star(P)}, E_b^{\star}/N_0\right) \\ \text{Subject to:} \quad & \rho_b < 1 \ , \\ & \varepsilon < PER_{max} \ , \\ & \mathbb{E}\left[\tau\right] \leq D_{max} \ . \end{aligned}$$

The heuristic proposed is to initially calculate the minimum number of total slots (RA and SA), $\zeta_R^{(P)}$, that guarantee a PER value below the PER_{max} bound for a range of E_b/N_0 values and for all $1 \leq P \leq J$ using equation (3.41). Next, similarly to the S-NDMA optimization proposed in [GBD+12], for each P value, the state space defined by all combinations of E_b/N_0, RA and SA slots with R retransmissions and $\zeta_R^{(P)}$ slots is searched using an alternating minimization approach, to look for the $\mathbf{n}^{P\star}$ and E_b/N_0 configuration that minimizes the EPUP (3.34), satisfying the bandwidth, ϵ and QoS restrictions.

$\mathbf{n}^{P\star}$ cannot be used directly, because the optimal value of $n_0^{(P)}$ depends on P. Therefore, sub-optimal matrices were devised, where the n_0 value for all P is replaced by one of the $n_0^{(P)}$ of the optimal matrix. A second optimization iteration is done for each sub-matrix by reconfiguring the SA slots to maintain the $\zeta^{(P)}$ value, and searching neighbor values of the optimal $E_b^{\star}/N_0$, considering the P probability mass function distribution. The $\mathbf{n}^{\star(P)}$ and $E_b^{\star}/N_0$ values are defined by the sub-optimal matrix that leads to the minimal EPUP value for a given load. The optimal matrix depends on the B_{max} value, so different matrices should be calculated for different values of B_{max}.

Chapter 4

Simulation Results

4.1 System Considerations

This section presents a description of the SR-NDMA system configuration and the scenarios considered in the analysis that follows.

4.1.1 Satellite orbits and system's Round Trip Time

The LEO satellite considered is modelled after the Iridium constellation [1]. The constellation of 66 active satellites has 6 orbital planes spaced 30 degrees apart, with 11 satellites in each plane. The satellites are in low Earth orbit at a height of approximately 781 km with a covering angle, θ, that ranges from 0°to approximately 28°, represented in Figure 4.1. The maximum covering angle is defined by the Earth's shape and the satellite height. At 28° the MT looses the satellite coverage when it reaches the skyline.

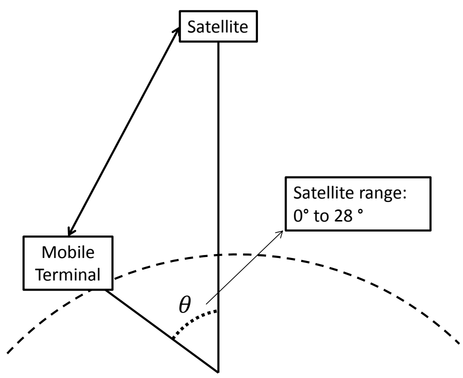


Figure 4.1: Satellite's covering angle range

[1] see section 2.1.2.1

In these conditions, we can define T based on the RTT of the furthest MT within one of the satellite's spots, which covers a total range with a radius of 200 km, corresponding to the 28°of the Earth's perimeter.

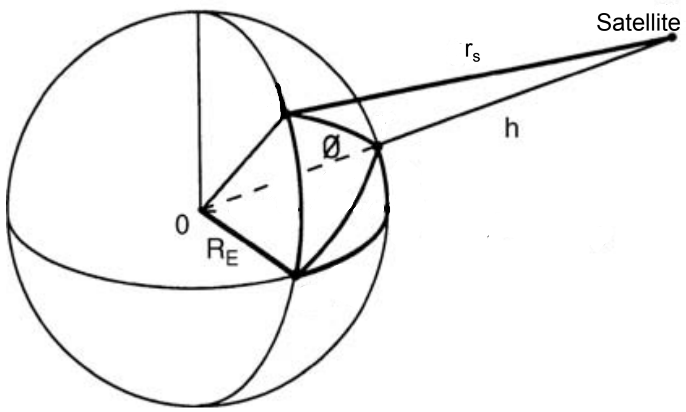


Figure 4.2: Satellite with θ displacement for RTT calculation purposes

Figure 4.2 illustrates the way the RTT was calculated for θ=28°, since this is the worst case scenario, in case a MT is in this position, this is the theoretical maximum angle that the satellite can cover. The average Earth's radius R_e=6371 km and the distance from the MT to the satellite h with θ=0° is 781 km. The distance from earth surface to the satellite with θ=28°, r_s is given by,

$$r_s = \sqrt{R_e^2 + (R_e + h)^2 - 2R_e(R_e + h)cos\theta} \tag{4.1}$$

Using equation (4.1), it was possible to conclude that r_s=3850.1 km when θ=28°. Finally, the RTT to that distance is calculated using,

$$RTT = 2 \times \frac{r_s}{c} \tag{4.2}$$

where c is the speed of light resulting in a RTT = 25.7 ms.

4.1.2 Handling very low power using CDMA

The long distance between satellites and MTs and the limited transmission power in the MTs introduces limitations for the design of SR-NDMA. Successful data reception can only be achieved when the energy received for one packet makes the total E_b/N_0 value (counting all copies received) above 0dB. Due to the huge path loss and the limited transmission power, this would force a number of data packet retransmissions extremely high (in the hundreds and even thousands), which would make the receiver extremely complex and impossible to implement in real-time. To resolve this issue, this dissertation adopts the solution proposed in [GBD+12], which combines SR-NDMA and CDMA. The slots are grouped into CDMA frames and the SR-NDMA receiver algorithm is applied to the CDMA frames. This decreases the number of packets handled by the receiver by a spreading factor, S_f. Besides the lower complexity, the noise power is reduced, due to the channel spreading factor gain. When the narrow band signal is transformed into a CDMA signal by a spread spectrum operation, the original channel bandwidth is replaced by a bandwidth S_f times higher.

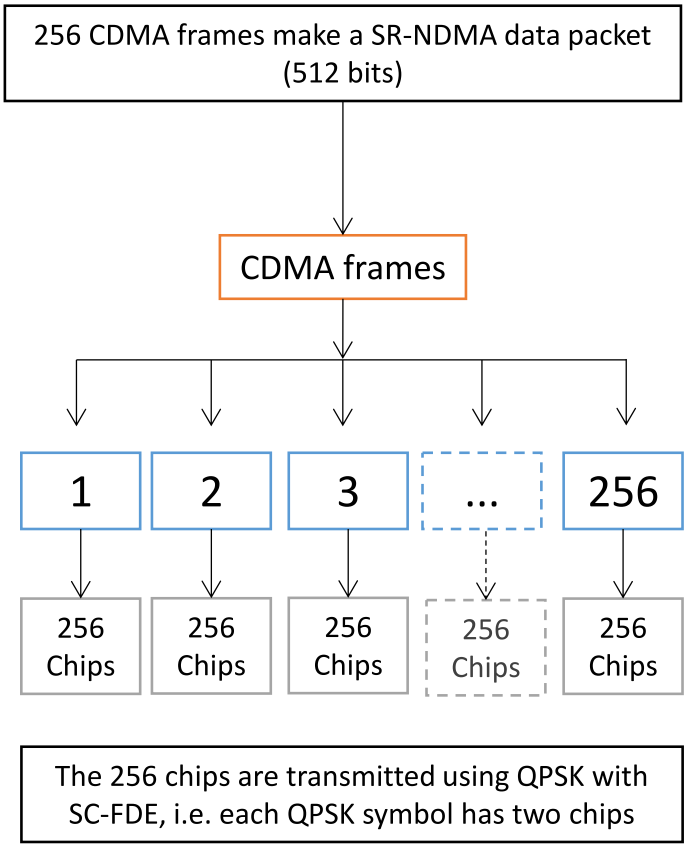


Figure 4.3: The system's data structure

In this work it was assumed that each MAC packet received is constituted by 512 bits, as seen in section 3.1.2. Before being transmitted, each packet is spread using a CDMA spread factor of 128, which means that each data packet has 512 $\times$128=65536 chips[2] as

[2] A chip sequence is a code assigned to each MT. To transmit a bit a MT sends a chip sequence.

seen in Figure 4.3. The MAC data packet that the receptor of section 3.1.2 receives, is constituted by 256 CDMA frames, so each data symbol of a CDMA frame is a QPSK symbol with two chips.

4.1.3 Transmission and System Parameters

A communication link between a MT and a satellite requires the definition of a set of radio frequency link parameters. The antenna's transmissions power of an handheld MT is a parameter that has some limitations, due to people safety concerns. Therefore, it was decided to fix the transmitted power, p_t (as defined in section 3.2.4), with a value of 2.5 Watts, aligned with current handheld mobile terminals. The power received in the satellite, P_r, can be obtained given the transmit antenna gain in dB, G_t, the receive antenna gain in dB, G_r and the free space path loss, L_{fs}.

The free space path loss is given by equation (4.3) [Ipp08],

$$L_{fs} = 20\log(f_c) + 20\log(r_s) + 92.44 \ , \tag{4.3}$$

where f_c is the carrier frequency in Ghz and r_s is the distance from the satellite to the MT in km.

According to [Ipp08], the received power at the satellite, in dBs, is given by

$$P_r = 10\log(p_t) + G_t + G_r - L_{fs} \tag{4.4}$$

The objective is to guarantee a bit energy E_b over the noise spectral density N_0 ratio between the MT and the satellite, not smaller then -2dB. This guarantees that the E_b/N_0 at the transmitter, in equation (3.34) only needs to be 2dB higher than the value at the receiver. This value is possible due to the adopted solution of using CDMA to spread the signal. If CDMA was not used, this value would only be possible using huge amounts of packets copies or increasing the energy of the MT making the system impractical. CDMA manages to garantee this difference without the need to increase the copies of the packets (making the receiver impossible to compute in real time) or to increase the energy at the

MT to inept values.

The E_b can be obtained by using equation (4.5),

$$E_b = \frac{P_r \times T_{on}}{M} \tag{4.5}$$

where M is the total number of bits in the data symbol (as seen in section 3.1.2), and T_{on} is the transmission time of a data block (as seen in section 3.2.4). The N_0 is given by equation (4.6) [MB02].

$$N_0 = \frac{k \times T_{temp} \times B}{S_f} \; . \tag{4.6}$$

k is the Boltzmann's constant in joules per kelvin, T_{temp} is the receiver system noise temperature in kelvins, B is the bandwidth in Hz and S_f, the aforementioned spreading factor.

It was assumed a bandwidth B with 40Mhz. The carrier frequency considered, f_c was of 1616 Mhz, which is situated in the L band [3]. Finally the transmission time T_{on} is given by

$$T_{on} = \frac{S_f \times M}{2 \times B} \tag{4.7}$$

Using equation (4.7), and considering a transmission power of 2.5 Watts to guarantee a E_b/N_0 between the MT and the satellite of -2dB, we obtain a transmission time of 819μs. The gains of the antennas plus coding, $G_T + G_S$, needed to achieve these specification is 64.4 dBs, and were calculated using equation (4.4) by considering a distance from the MT to the satellite of 781 km[4].

The number of slots in a RTT, T, is calculated using the RTT of the furthest MT divided by the transmission time, resulting in $T \approx 32$ slots. Therefore, for the simulated system B_{max} will be equal or below 32 slots per RTT.

The severely time dispersive channel of [GBD+12] was considered for the simulations,

[3] This frequency belongs to the Iridium satellite system, used for voice communications
[4] This distance to the satellite is when the MT has an angle θ of $0°$

with rich multipath propagation and uncorrelated Rayleigh fading[5] for each path and MT.

To cope with channel correlation for different retransmissions, the Shifted Packet technique of [DMB+09] was considered, where each transmitted block has a different cyclic shift (see section 2.5.2.1). A MT set composed by $J = 5$ MTs is also considered.

Commentary

The combined gains of the antennas required is challenging, but in practice it can be obtained. For example, if the modeled system uses an 8 state turbo codes[6] with uncorrelated fading, a gain of 20 dB could be obtained.

More gain can be achieved if the antennas of the MT and the satellite are both directional (a gain of 15 db could be reached in each). Another gains could be achieved if a proper channel model is used, i.e. instead of using a Rayleigh fading channel model, a Nakagami channel could be used, which is more adequate to satellite systems, improving the gain in the simulations. The channel model considered in the simulations presented in this dissertation is pessimistic.

A more complex receiver like Iterative Block- Decision Feedback Equalizer (IB-DFE)[7] also improves the overall gain of the system. So in spite of being a high combined gain of the antennas, it is possible to obtain these values.

Another approach would be to increase the spreading factor, which would diminish the noise, and the specification of an E_b/N_0 between the MT and the satellite of -2dB would require lesser gains of the antennas. But that would turn the transmission time bigger, resulting in a lower number of slots per RTT. In turn, the B_{max} would also decrease, limiting the number of slots that could be used per RTT. This, as we will see in section 4.2.2, will increase substantially the EPUP. Therefore, there is an important trade-off between the spreading factor and B_{max} that must be achieved.

[5] Rayleigh fading models assume that the magnitude of a signal that has passed through a communications channel will vary randomly, or fade, according to a Rayleigh distribution. Rayleigh fading is viewed as a reasonable model for tropospheric and ionospheric signal propagation.

[6] A class of high-performance forward error correction (FEC) codes

[7] A turbo equalizer scheme implemented in the frequency domain

4.2 Performance Analysis

In this section, the system performance is analyzed for SR-NDMA, considering the PER, throughput, EPUP, delay and jitter. As described above, the RTT considered is 23.4 ms (calculated in section 4.1.1) with a slot transmission time, T_{on}, of 819μ (obtained in section 4.1.3), making T=32 slots. MTs transmit uncoded data block with 512 bits using QPSK constellation and a spreading factor of 128.

For the scenario proposed, $R = 2$ was calculated from D_{max} using equation (3.41), while assuming the size of the MTs queue equal to 25 (Q_{max}=25). The QoS requirements considered, were of a video telephony traffic [AMCV06], which specifies a $PER_{max} \leq 1\%$, a maximum delay for a satellite link of D_{max}=100 ms (which corresponds to 122 slots). The voice quality degrades significantly when jitter consistently exceeds 30 ms (36 slots in this case). This is considered the maximum jitter requirement. The packet loss rate condition $\epsilon < 1$, although not explicitly shown, is guaranteed not to occur due to the slot optimization procedure followed.

The optimization algorithm was run considering a limited E_b/N_0 at the receiver between -4 and 12 dB. According to the specification of section 4.1.3, the E_b/N_0 at the transmitter has to be at least 2dB higher than the values of E_b/N_0 at the receiver. Meaning that if the receiver needs 12dB, the MT has to use at least 14dB.

4.2.1 n⋆ calculation

When a PER_{max} bound is defined, it is necessary to calculate the minimum number of packet transmissions $\zeta_R^{(P)}$, that achieve it. It is clear that H_k^T in equation (3.4) is not affected when R is set to zero or a value above zero, as long as the total number of slots with transmissions($\zeta_R^{(P)}$ for a single super-slot) does not change , i.e. $\zeta_0'^{(P)}$=$\zeta_R^{(P)}$. From equation (3.15), $\zeta_R^{(P)}$ could be obtained as the minimum value of $\zeta_0'^{(P)}$=$n_0^{(P)}$ that satisfies the condition,

$$\zeta_R^{(P)} \approx \min_{n_0^{(P)}} \left\{ PER_P \left(\Psi'^{(P,0)} = [P] \right) \leq PER_{max} \right\} . \tag{4.8}$$

Figure 4.4 illustrates the calculation of $\zeta_R^{(P)}$, considering PER_{max}. It shows that the

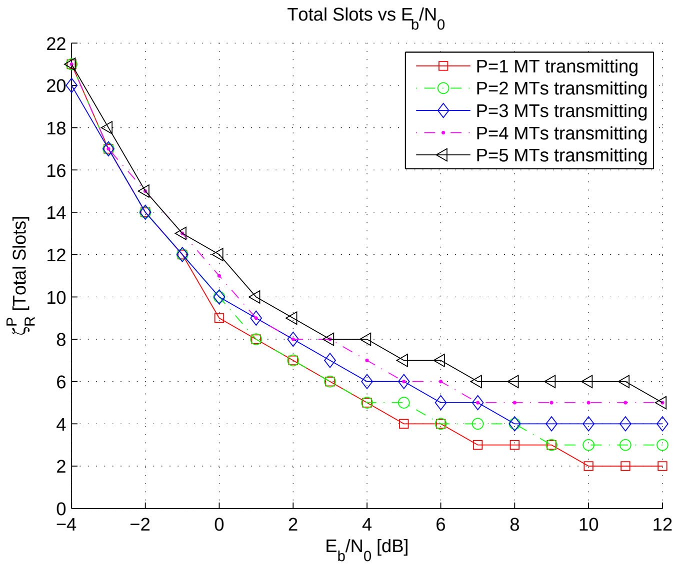


Figure 4.4: ζ_R minimum that satisfies PER $\leq 1\%$ over E_b/N_0

PER condition can be satisfied for all values of E_b/N_0, requiring the allocation of more slots for lower E_b/N_0 or higher P values. The number of slots allocated per MT decreases for higher P making SR-NDMA more bandwidth efficient.

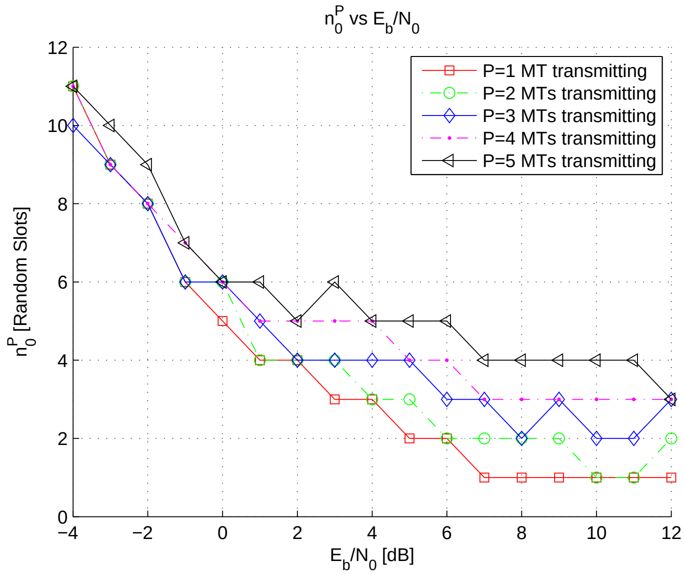
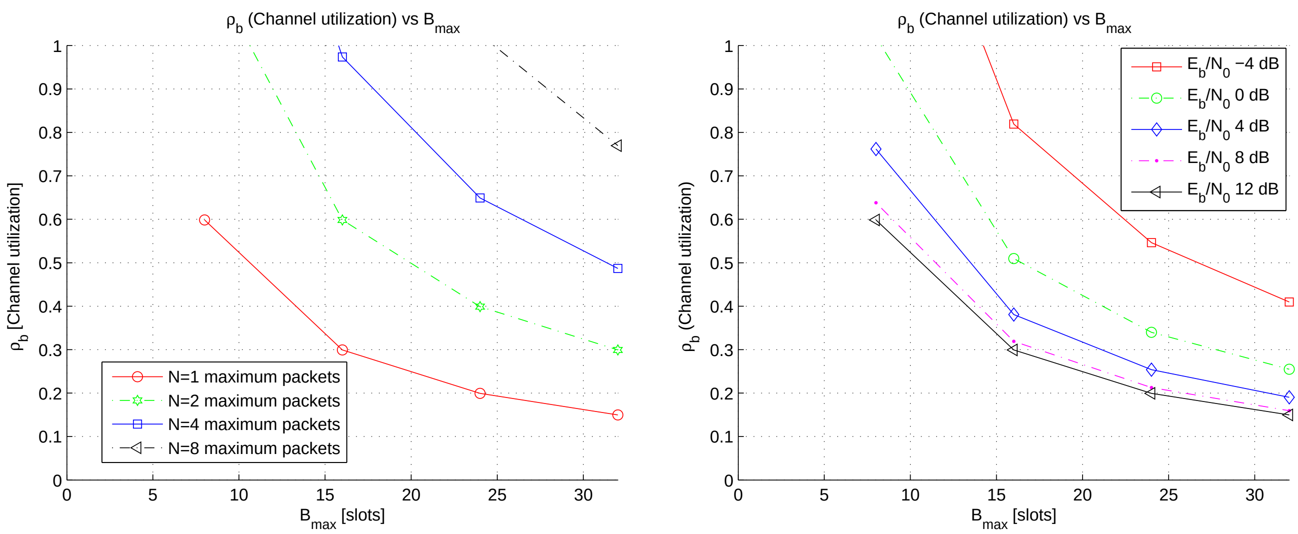


Figure 4.5: Optimal $n_0^{(P)}$ values over minimum E_b/N_0

In a second step, the optimal $\mathbf{n}^{P\star}$ is calculated for the different P values. Figure 4.5 depicts the optimal $n_0^{(P)}$ values obtained, showing that it has a significant variation with P for higher E_b/N_0 values, where the packet copies are mainly used to separate the packets transmitted concurrently. Near and below 0dB, $n_0^{(P)}$ has a small variation with P since the copies are mainly used to reduce the PER.

From this set of values it is possible to calculate the sub-optimal matrices and select the combination of E_b/N_0 and $\mathbf{n}^\star$ which minimize the EPUP and satisfy the QoS constraints for the B_{max} value considered.

4.2.2 B_{max} parameter

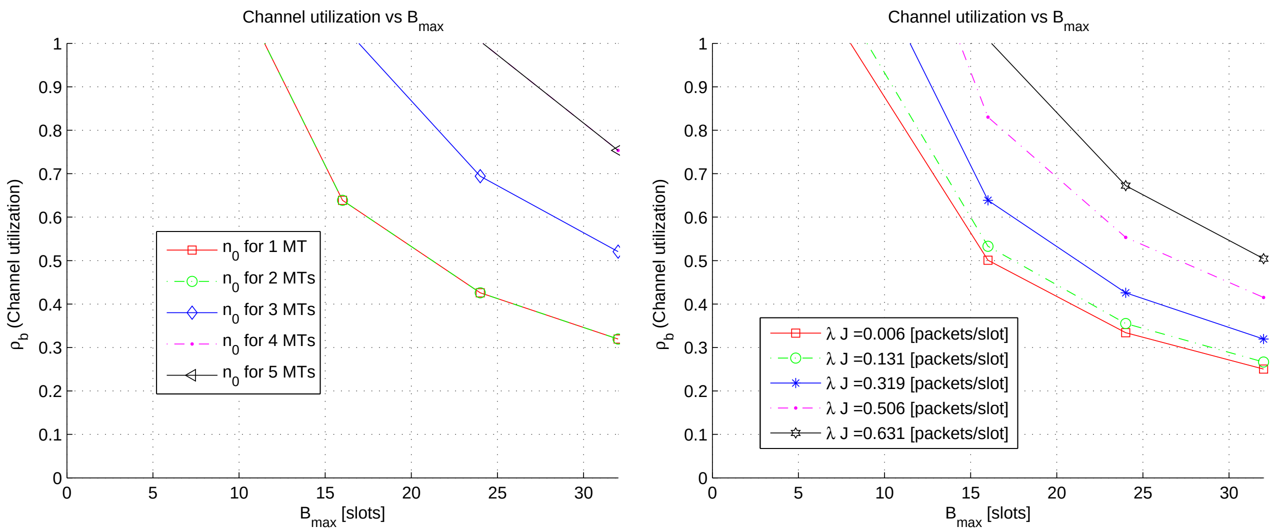


(a) E_b/N_0=12 dB, λJ =0.25 packets/slot, optimal $n_0^{(P)}$ for 4 MTs (b) N=1, λJ =0.25 packets/slot, optimal $n_0^{(P)}$ for 4 MTs

Figure 4.6: ρ_b (Channel utilization) over the allocated slots B_{max}. - 4.6(a) various N - 4.6(b) various E_b/N_0

The number of slots allocated to a set of MTs per RTT, B_{max}, depends on the number of MTs connected to the satellite. For a small number of MTs below the MPR capacity, a single set of MTs is used; otherwise, the slots in T have to be shared by different sets of MTs. The maximum energy efficiency is achieved when a single set of MTs use all slots in T, corresponding to B_{max} = 32 slots. Figure 4.6(b) depicts the variation of ρ_b over B_{max} for different values of E_b/N_0 for the optimal configuration that satisfies the QoS requirements.

It is shown that an higher channel utilization is required for smaller E_b/N_0 values, i.e., more packet copies are required to satisfy the same PER_{max} requirement. Therefore, the minimum transmission powers (associated to lower E_b/N_0 values) can only be used when B_{max} is high. Otherwise, to make a better utilization of the available band, the energy has to be raised. The ideal value of E_b/N_0 should be the one that is closest to the limit of the band utilization, but does not surpasses it, allowing a better energy efficiency (i.e. lower EPUP).



(a) N=8, E_b/N_0=12 dB, λJ =0.25 packets/slot

(b) N=8, optimal $n_0^{(P)}$ for 4 MTs, E_b/N_0=12 dB

Figure 4.7: ρ_b (Channel utilization) over the allocated slots B_{max}. - 4.7(a) various $n_0^{(P)}$ - 4.7(b) various λJ

Figure 4.6(a) shows the effect N in the channel utilization when B_{max} varies. An increasing N increases the channel utilization, since more RA slots are allocated in the available band, B_{max}, reducing the SA slots that can be used.

Figure 4.7(a) depicts the channel utilization for different $n_0^{(P)}$. The channel utilization has the tendency to increase for $n_0^{(P)}$ of higher number of MTs, since an increase in RA slots occurs. Also, the total load, λ J, also increases the channel utilization since the probability of more MTs transmitting increases, meaning that more SA slots are needed leading to an increase in ρ_b (Figure 4.7(b)).

4.2.3 N parameter

N controls the number of RA super-slots that can be used by a MT per RTT. The figures in this section represent how the N parameter influences the delay , EPUP and throughput, when B_{max} is fixed. The E_b/N_0 used is variable: for the various λJ and N values considered, it is used the lowest E_b/N_0 that verifies the optimization constraints. The $n_0^{(P)}$ is chosen according to the E_b/N_0 value, considering the optimal values for 5 MTs (according to Figure 4.5) , when B_{max}=32 slots. As expected, Figure 4.8 shows that N

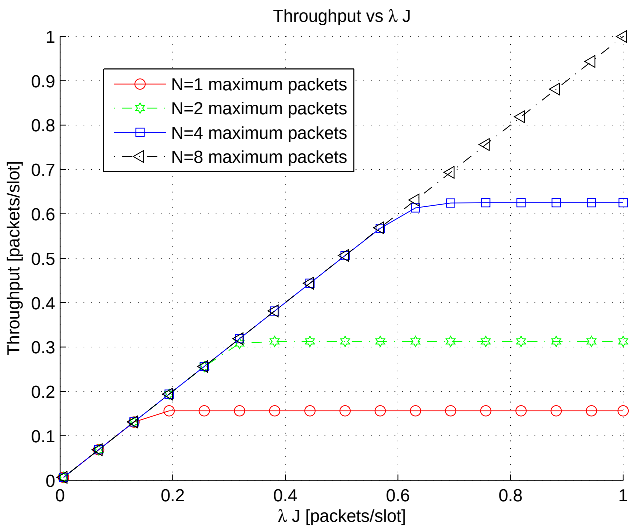


Figure 4.8: Throughput over the total load λJ for different values of N, $B_{max} = 32$ slots, optimal $n_0^{(P)}$ for 5 MTs

limits the maximum throughput in the SR-NDMA channel to NJ/T. Thus the system is saturated for a total load above this limit, leading to a discard rate of MAC data packets above PER_{max}.

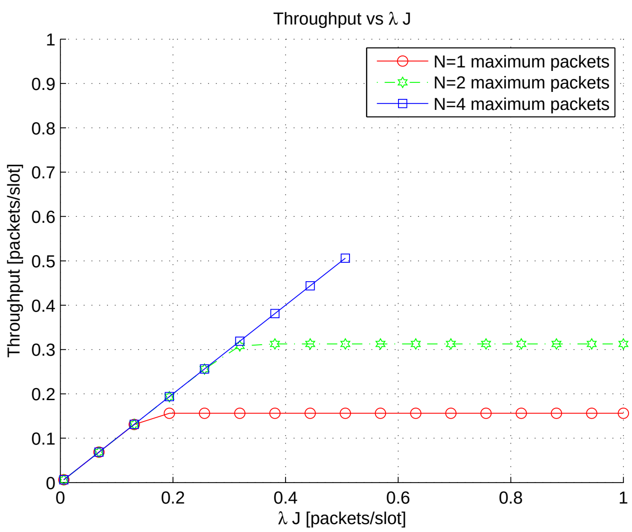
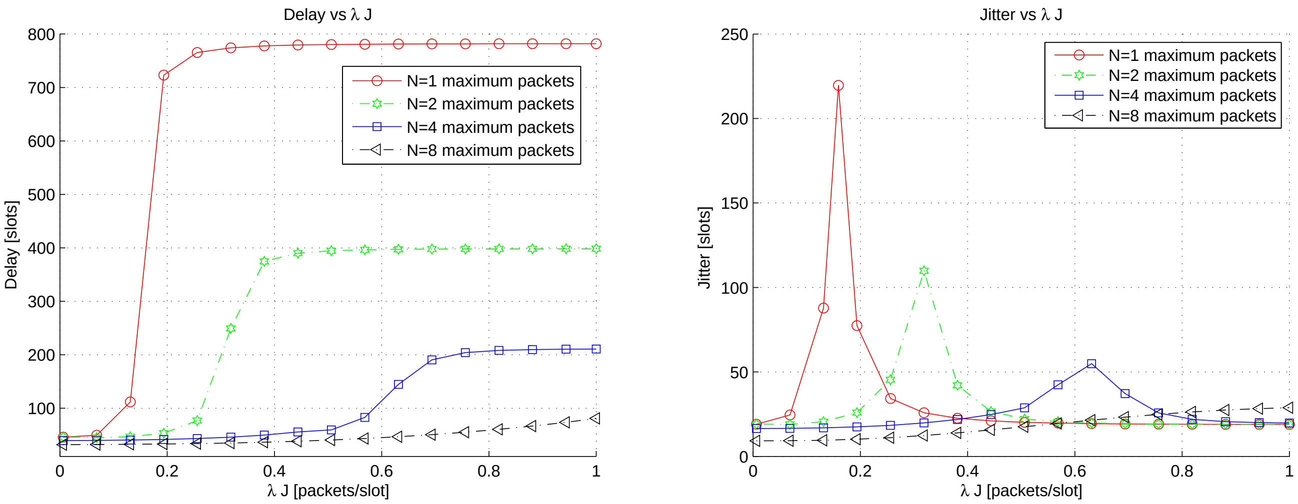


Figure 4.9: Throughput over the Total Load for different values of N, $B_{max} = 16$ slots, optimal $n_0^{(P)}$ for 5 MTs

The maximum achievable throughput for each value of N is also highly affected by B_{max}. When the B_{max} value is reduced to 16 slots, the maximum throughput for N=4 is represented in Figure 4.9. According to Figure 4.8 it should be 0.63, but this value is not achieved because, after 0.53 the ρ_b condition is not verified again, since the maximum E_b/N_0 was limited to 12dB. For N=8 the system is impracticable - values higher than 12dB would have to be used in order to verify the ρ_b condition.



(a) Delay over the total load λ J, $B_{max} = 32$ slots, optimal $n_0^{(P)}$ for 5 MTs

(b) Jitter over the total load λ J $B_{max} = 32$ slots, optimal $n_0^{(P)}$ for 5 MTs

Figure 4.10: Delay and Jitter over the total load λJ for different values of N

Figure 4.10(a) depicts the average delay for the packets that are not discarded at the queue. It shows that the total delay is below, $D_{max} = 122 slots$, and almost independent of N while the system is far from saturation. The saturation delay decreases for higher N values, as $\mathbb{E}[\xi]$ tends to Q_{max}/N.

Figure 4.10(b) depicts the jitter variation with the total load, showing that an increase of the N parameter reduces the level of the jitter. The peaks in each curve occur when the maximum throughput is achieved. After that, the jitter value drops for the saturation throughput, as according to Figure 4.10(a).

The jitter, when the systems approaches the saturation load, and for N equal to or below 4, surpasses the maximum jitter of 30 ms (36 slots). For all loads, except in those situations, the jitter requirement is guaranteed.

The results above show that increasing N increases the network capacity, but also increases the average energy spent per packet received. Figure 4.11 depicts $(EPUP/E_p)(E_b/N_0)$, which considers the energy at the receiver, removing the dependence on the path loss and other propagation issues [GBD+12]. This figure shows that when N is increased until

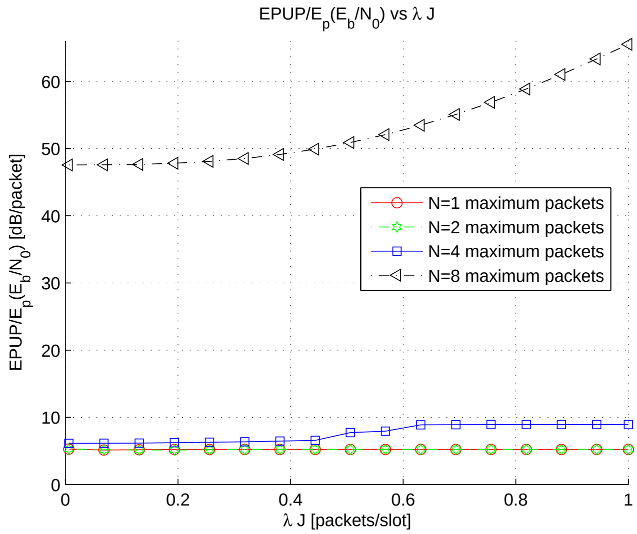


Figure 4.11: EPUP over the Total Load for different values of N, $B_{max} = 32$ slots, optimal $n_0^{(P)}$ for 5 MTs

$N = 4$ the EPUP does not change significantly. However, for $N = 8$ it increases continuously, because the slots left for SA retransmissions ($B_{max} - n_0N$) start to be too short, requiring higher E_b/N_0 values to satisfy the $\rho_b < 1$ condition. Therefore, there is an optimal value for N, which is mainly defined by the B_{max} value and by the average load.

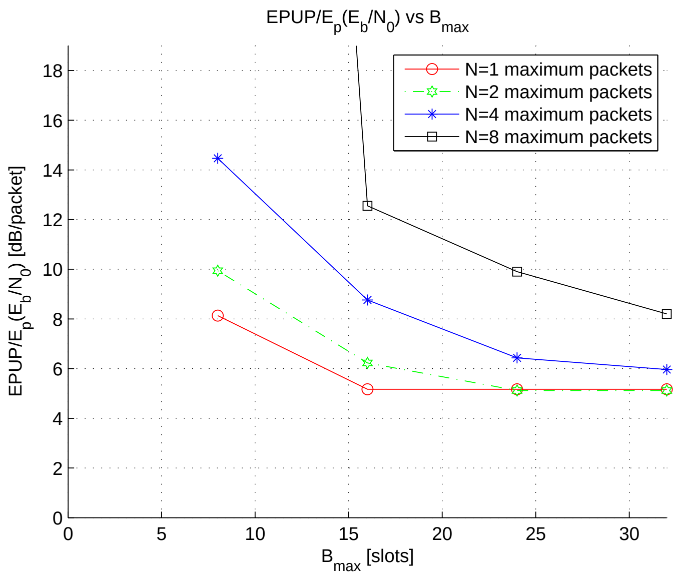
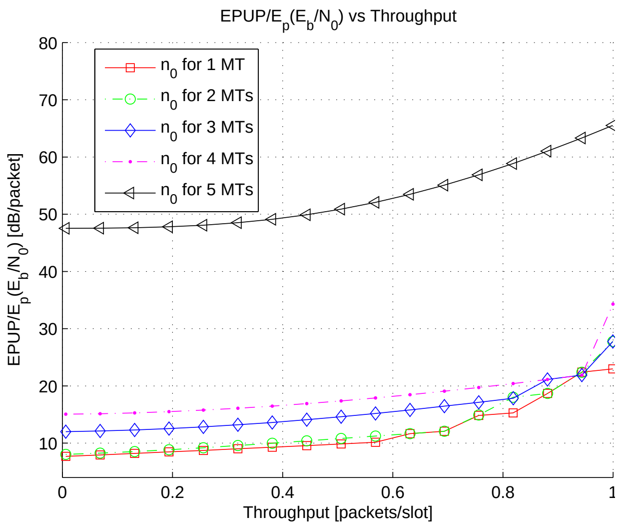


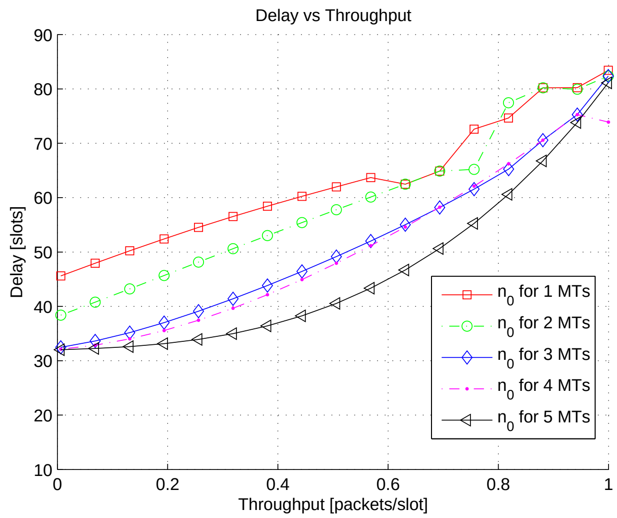
Figure 4.12: EPUP over the total allocated slots per RTT (B_{max} slots) different values of N, optimal $n_0^{(P)}$ for 1 MT, $lambda$J =0.13

Figure 4.12 demonstrates what is the effect on the EPUP when the B_{max} value increases for various values of N parameter. It shows that the EPUP decreases when the number of slots allocated per RTT is increased. For the smallest B_{max} values, it is needed an higher E_b/N_0 to verify the $\rho_b < 1$ condition, as seen in Figure 4.6(b), leading to higher EPUP values. A larger N value leads to more waste of energy, for a given B_{max} value, due to the reasons explained in the previous paragraph.

4.2.4 $n_0^{(P)}$ parameter



(a) EPUP over the throughput , $B_{max} = 32$ slots, N=8

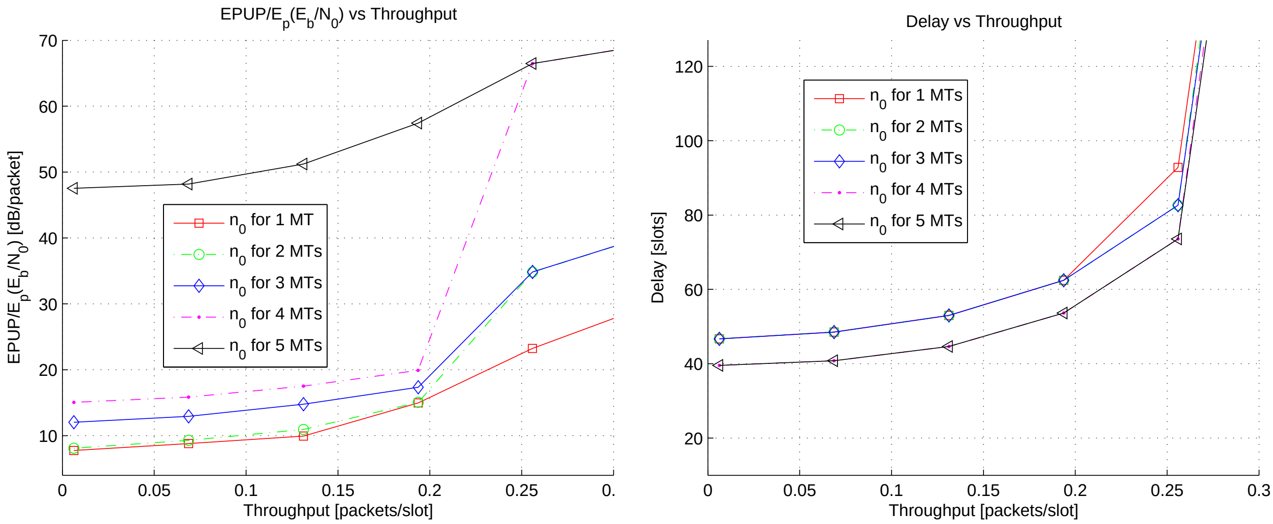


(b) Delay over the throughput $B_{max} = 32$ slots, N=8

Figure 4.13: First example of EPUP and delay over the throughput for different $n_0^{(P)}$

This section analyzes the effect of the $n_0^{(P)}$ parameter for fixed values of B_{max} and N, using the optimal E_b/N_0 defined by the algorithm in section 4.2.1. Figures 4.13(b) and 4.13(a) depict respectively the delay and the EPUP measured for different values of throughput with $B_{max} = 32$ slots and $N = 8$ packets/RTT. Given that all setups satisfy the PER_{max} requirements and use the same N, they carry the same maximum throughput. However, a higher $n_0^{(P)}$ value increases the use of RA slots and reduces the number of SA slots required, leading to less energy efficiency. The figures expose a configuration trade-off for SR-NDMA: n_0 can be used to maximize the energy efficiency (setting the optimal $n_0^{(P)}$ for 1 MT), minimize the delay (setting the optimal $n_0^{(P)}$ for J MTs), or using an intermediate value that balances delay and energy efficiency.

For this specific scenario, for all configurations the maximum delay is not surpassed ($D_{max} = 122$ slots), so the only criteria chosen to be minimized is the energy consumption. The best choice in this particular scenario would be to use the optimal $n_0^{(P)}$ of 1 MT for all throughput, since the delay is met in all alternatives and the optimal $n_0^{(1)}$ value is the one which leads to less energy consumption.



(a) EPUP over the throughput , $B_{max} = 16$ slots, N=2 (b) Delay over the throughput $B_{max} = 16$ slots, N=2

Figure 4.14: Second example of EPUP and delay over the throughput for different optimal $n_0^{(P)}$

Another scenario can be shown to exemplify the trade-off between the use of different $n_0^{(P)}$ values. In Figures 4.14(a) and 4.14(b) a scenario was considered where N=2 and $B_{max} = 16$ slots. Two possible options in this case would be to have throughputs smaller than 0.26 using $n_0^{(P)}$ for 1 MT, or between 0.26 and 0.28 using the $n_0^{(5)}$ values. The choice to use an $n_0^{(P)}$ considering an intermediate value of MTs is to still guarantee that the D_{max} condition is verified in the required interval. For a throughput larger than 0.28 the delay is exceed, so the maximum throughput is limited by the maximum delay in this case. These options guarantee the delay requirement and the use of the minimum energy at the same time.

Finally, Figure 4.15 shows the effect on the EPUP when different allocated B_{max} slots are available. In the scenario of the figure, the tendency is to waste more energy when a $n_0^{(P)}$ for more MTs is used. This effect is particularly visible for lower B_{max} values.

For larger B_{max} values, it becomes slightly indifferent which $n_0^{(P)}$ to use, meaning they all verify the $\rho_b < 1$ condition for the same E_B/N_0.

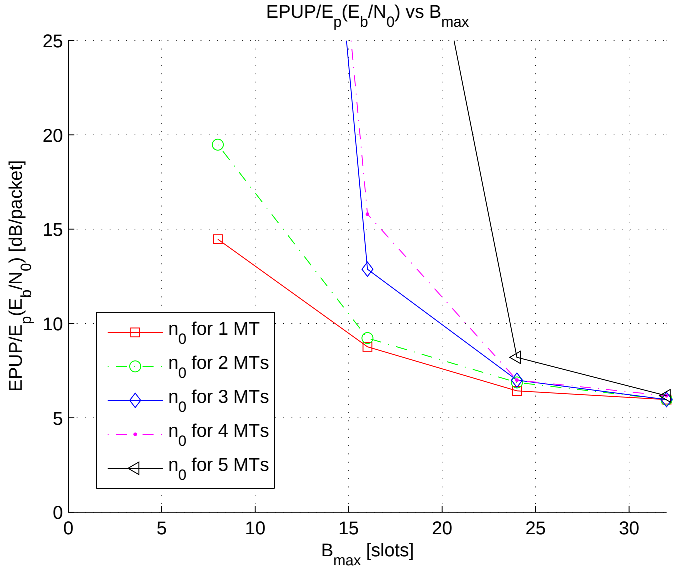


Figure 4.15: EPUP over the total allocated slots per RTT (B_{max} slots) for different optimal $n_0^{(P)}$, N=4, λ J =0.13

4.2.5 S-NDMA and SR-NDMA comparison

The models developed for the SR-NDMA protocol are in many ways different from the developed model of S-NDMA. For example, the delay models are not comparable, but an obvious delay gain of a RTT is achieved. The S-NDMA is a DAMA protocol and a request over a control channel to the satellite has to be used to make the proper reservations, increasing the total delay at least by a RTT. SR-NDMA, being a random access protocol does not require a pre-reservation of the slots, lowering the total delay.

The main criteria where S-NDMA has better results than SR-NDMA is the use of allocated band, which is related to an increase in the energy efficiency. The satellite allocates only the optimal exact number of slots for the transmitting MTs, not wasting any resources.

Figure 4.16 shows an EPUP comparison between the S-NDMA and SR-NDMA protocols, when different optimal $n_0^{(P)}$ values are used for SR-NDMA. It is shown that in this case, when the optimal $n_0^{(P)}$ for less MTs are used, the energy efficiency drops. The SR-NDMA only achieves the same EPUP of S-NDMA when the $n_0^{(P)}$ of 5 MTs is used, allocating the exact optimal number of slots, since in this scenario P=5 MTs transmit.

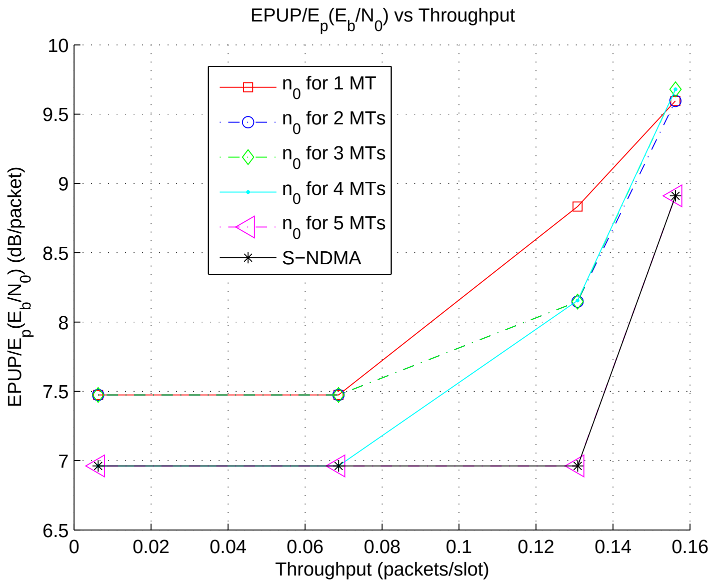


Figure 4.16: EPUP comparison between S-NDMA and SR-NDMA when 5 MTs transmit, with N=1 and B_{max}=8 slots

Chapter 5

Conclusions

5.1 Final Considerations

This dissertation proposed SR-NDMA, a protocol designed to provide QoS soft guarantees for scenarios with a high RTT, such as satellite networks. SR-NDMA combines an initial random access with additional scheduled accesses for packet transmissions, avoiding the delay degradation due to an initial reservation phase that exists for pure demand assigned multiple access (DAMA) protocols, such as S-NDMA. For the scenarios explored in section 4.2, S-NDMA would use one RTT for slot reservation. This leads to a increase of one RTT to the total delay, when comparing to SR-NDMA. The trade-off was, a slight decrease in the bandwidth and energy efficiency. An energy and bandwidth efficiency degradation occurs due to the random access phase, since some RA super-slots may be empty and the transmission parameters and slot allocation in this phase would be fully optimized to the number of MTs that are accessing in S-NDMA.

The SR-NDMA performance is very dependent on the receptor used as well as on the slots and transmission power configuration. As long as the optimization process is run offline, to produce a set of configuration tables indexed by B_{max}, D_{max}, J and the total load, the SR-NDMA system proposed can be used in run time by the satellite to configure the random access and scheduled access slots and the transmissions parameters for each individual MT.

The results for SR-NDMA system configured with optimal parameters show that SR-NDMA is capable of satisfying demanding QoS requirements, adapting to the bandwidth available and minimizing the energy consumption. However, to achieve these results, the system parameters chosen in section 4.1.3 are quite stringent. For instance, the combined gains in the antennas of the terminal and the satellite are 51.39 dB. For a practical implementation, state of the art technology would have to be used. For instance, the developed models consider an uncoded system; a coded one with powerfull FEC such turbo codes and directional antennas at both the MT and satellite would facilitate this requirement. Alternatively, the data rate could be decreased, increasing the spreading factor.

Another option would be to turn the system into less a demanding one, by using more powerful terminals with top mounted antennas. The transmitting power in the terminal could be increased (for example to 8 watts or more). Nevertheless, SR-NDMA can be a valid option for the satellite component of future integrated telecommunication networks.

5.2 Future Work

This dissertation addressed the QoS requirements of multimedia traffic, but considered simplified traffic models (Poisson) to allow an analytical performance evaluation. Future work includes the validation of SR-NDMA performance for more realistic traffic models, and considering non-real time data traffic, which may include more packet retransmissions. This would include extending the current model to consider G(Generalized) traffic in the queueing models (e.g. G/D/N) and the transmission of lost packets in multiple epochs.

The channel model could be replaced by a more adequate one for satellite communications, such as a Nakagami fading channel model. A more powerful receiver could also be used, such as IB-DFE, to tolerate a lower received power at the satellite improving the overall system efficiency, at the cost of additional complexity at the receiver.

A comparison between typical random access protocols (e.g. ALOHA and S-ALOHA), used in satellite communications with SR-NDMA would be very interesting in terms of relative performance.

The future work includes also the submission of a journal article including the work developed in this dissertation and further work, which will be developed in the context of a research scholarship.

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Appendix

Appendix A

Second Moment Of The Time In Queue

$$\mathbb{E}\left[\tau^2\right] \equiv \mathbb{E}\left[(\gamma + \varpi + ST)^2\right] \equiv$$

$$\mathbb{E}\left[\gamma\right]^2 + \mathbb{E}\left[\varpi\right]^2 + \mathbb{E}\left[ST\right]^2 + 2 \times \mathbb{E}\left[\gamma\right] \times \mathbb{E}\left[\varpi\right] + 2 \times \mathbb{E}\left[\gamma\right] \times \mathbb{E}\left[ST\right] + 2 \times \mathbb{E}\left[\varpi\right] \times \mathbb{E}\left[ST\right] \equiv$$

$$\equiv \frac{T^2}{3} + \sum_{Q=1}^{MAX} \left(Pr\left(\xi = Q\right) \left(\left\lfloor \frac{Q-1}{N} \right\rfloor \times T \right)^2 + \frac{1}{2} \left(Q - \left\lfloor \frac{Q-1}{N} \right\rfloor \times N \right) \right) +$$

$$\sum_{p=0}^{J-1} f_b\left(J-1, p, P_{tx}\right) \left(E\left[d_{prop}\left(\Omega_{p+1}^{(R)} \right) \right] \right)^2 +$$

$$2 \times \sum_{Q=1}^{MAX} \left(Pr\left(\xi = Q\right) \left(\left\lfloor \frac{Q-1}{N} \right\rfloor \times T \right) + \frac{1}{2} \left(Q - \left\lfloor \frac{Q-1}{N} \right\rfloor \times N \right) \right) \times$$

$$\left(\frac{T}{2} + f_b\left(J-1, p, P_{tx}\right) \mathbb{E}\left[d_{prop}\left(\Omega_{p+1}^{(R)} \right) \right] \right) +$$

$$2 \times \frac{T}{2} \times f_b\left(J-1, p, P_{tx}\right) \mathbb{E}\left[d_{prop}\left(\Omega_{p+1}^{(R)} \right) \right]$$