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Technical Report

**Satellite Earth Stations and Systems (SES);
Satellite component of UMTS/IMT 2000;
Detailed analysis of the packet mode for the SW-CDMA
(Family A)**

Reference

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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Satellite Earth Stations and Systems (SES).

1 Scope

The present document evaluates the possibility of using packet access mode for satellite. The objective of the present document is to design and demonstrate the realistic feasibility of the packet access mode transmission over satellite and its applications, which eventually will lead to the specifications for this type of access.

Packets are relatively small units of data that can be routed through a network based on the destination address contained within each packet [7]. Breaking communication down into packets allows the same data path to be shared among many users in the network. For the mobile user, the support for packet-switching means that a persistent link is not needed. The same broadcast channel can be shared among a number of users at the same time. The user's modem recognizes the packets intended for its user. As data such as e-mail arrives, it is forwarded immediately to the user without a circuit connection having to be established.

According to [8], in UMTS four different traffic classes can be defined:

- conversational;
- streaming;
- interactive;
- background classes.

The last two can be considered as packet data traffic. Conversational and streaming classes are assumed to be transmitted as real-time connections over the air interface.

As an example of this traffic, one can imagine a packet session during which one or several packet calls can be generated., so that the packet constitutes a bursty sequence of packets. The burstiness during the packet call is a characteristic feature of the packet transmission. For example, in a web-browsing session a packet call corresponds to the downloading of the document. After the document is entirely received by the terminal, the user takes a certain time to study the information. This interval is called "reading time".

The following parameters describe the characteristics of the packet data traffic:

- session arrival process;
- number of packet calls per session;
- reading time between packet calls;
- number of packets within a packet call;
- time interval between two packets inside a packet call;
- packet size.

The properties that are typical for non-real-time packet services from the air interface point of view are listed below:

- Packet data is bursty. The required bit rate can change rapidly from zero to hundreds of kilobits per second.
- Tolerates longer delay than real-time services.
- Packets can be retransmitted.

The methodology followed to create the present document was to include six contributions [9], [10], [11], [12], [13] and [14] from four projects:

- ATB;
- SATIN;
- 3GNetSim; and
- GAUSS.

A description of each project is given in clause 4, whilst the reader can find more detailed descriptions of the simulations in the annexes of the present document.

In WCDMA there are three types of transport channels that can be used to transmit packet data: common, dedicated and shared. Each of the contributions will choose the physical layer that best adapts to the particular scenario of each project, and this is what is presented in the present document. Convergence between the different proposed solutions will be needed.

Another issue the reader has to take into account is that each of the contributions is focused in a different layer aspects. The ATB project explains the optimum physical layer for the satellite environment in the first contribution and the multicast feasibility in the second contribution, SATIN project also explains the problematic with layers 2 and 2+ and GAUSS project describes the RLC for a particular application.

The first contribution from ATB titled "SW-CDMA Packet Access" explains the "packet access" in the unicast mode (point to point), although the concept "packet access" is a more general and includes multicast and narrowcast transmissions (point to multi-point). For the scenario of these simulations it has to be noted that the GEO satellite constellation case has been considered as baseline as it is considered to represent the most challenging configuration for the analysis of the packet mode. However, results are considered applicable also to other satellite constellations.

The second contribution from the ATB consortium was an investigation on the feasibility of packet access for point to multi-point communications. Specifically the submission investigates:

- large block interleaving and RS coding;
- medium block interleaver with CRC;
- hybrid short Carousel and FEC with interleaving; and
- narrowcasting.

These two contributions from the ATB focus on the technical aspects of this type of access. ATB Phase I activity had the objective of investigating strategies for packet support in SW-CDMA, analysing the techniques currently proposed for supporting the packet mode of T-UMTS W-CDMA 3GPP air interface (Release 99 and the still on-going Release 5) and when necessary to adapt them taking into account the specific satellite environment.

More information about this project can be found at the ESA telecommunications web page: http://www.telecom.esa.int/telecom/www/object/index.cfm?fobjectid=617

SATIN is an IST project focused on the particular implications of the IP-based packet mode on the S-UMTS design, for multicast and broadcast transmissions. The - dictated by the UMTS core network - requirements for the S-UMTS access network will be a fundamental drive for the SATIN design paving the way for full integration with the T-UMTS for efficient delivery of a series of packet based services.

Efficient support of Internet-based applications to mobile/nomadic users is a key feature of the 3G Networks. In light of the shortage and the high cost of the T-UMTS spectrum, the operators are looking into the provision of integrated broadcast/multicast services through hybrid broadcast-UMTS systems. S-UMTS could play an important role in the efficient delivery of some UMTS services to which it is better suited. These services include broadcasting and multicasting applications such as audio/video, e-newspaper, live stock exchange data, etc.

The project's main objectives are summarized into the following lines:

- to determine the potential role of satellites in UMTS and Service Delivery;
- to define potential S-UMTS architectures efficient for the support of the IP-based packet mode;
- to suggest an optimized - with reference to the IP-based packet mode - layer 1 and 2 design.

More information about this project can be found at its web page: http://www.ist-satin.org

The contribution from GAUSS is focused on the applicability of the packet access, which synergistically integrates navigation and communications, for providing enhanced location-based services (highly reliable, near real-time two-way communication between Mobile Users and Service Centre/Provider).

The main concept which the GAUSS project is based on, envisages the communication and navigation system components fitting within the general framework of S-UMTS and GALILEO (GNSS-2, Global Navigation Satellite System - Phase 2). The technological issues of such a concept relies on the development of a Demonstrator, which integrates existing and available facilities with ad-hoc designed components. The former ones constitute the ground segment, the latter ones include the advanced user terminal and the innovative services and applications.

GAUSS system is capable to provide location-based services, for safety-of-life applications in the transport sector (emergency assistance, fleet and freight transport management on road and inland waterway, intermodality, dangerous goods transportation and containers tracking).

These services are characterized by exchange of small data packets, from mobile users towards a service provider and vice versa, carrying precise position data and application relevant information. Furthermore, the GAUSS provided services are characterized by high quality performances as required by safety location-based applications, in terms of navigation (accurate positioning and integrity information) and communication (high reliability, availability, guaranteed time response and coverage, multicasting and broadcasting communication supported).

More information about this project can be found at its web page: http://galileo.cs.telespazio.it/gauss/

Clause 4.3 contains a description of the ESA Project "3GnetSim". Due to the time schedule of this project and the aim to finalize the present document by December 2003, the clause contains no quantitative results of simulation runs, which would allow to draw conclusions about the concepts being pursued within the project. It is therefore recommended to reopen the document in order to include these results, which are expected to be available by April 2004.

More information about this project can be found at its web page: http://telecom.esa.int/telecom/www/object/index.cfm?fobjectid=11675

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3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

broadcatching: communication capability which denotes an unidirectional distribution from all users connected to the network a single source Whilst *broadcasting* generally consist of a *one_to_ many* flow of information, *broadcatching* connotes a *many_ to_ one* gathering of information

broadcasting: communication capability which denotes unidirectional distribution from a single source to all users connected to the network

bursty traffic: type of traffic which consists of periods during which a different amounts of data are transmitted, preceded and succeeded by periods of random duration during which no transmission is made

carousel techniques: transmission where the same data is automatically sent a number of time, to have time diversity

interleaving: signal processing function which scrambles a sequence of data into a new sequence of data

NOTE: Recovery of the original data at the receiver is possible by using the reverse scrambling approach, called de-interleaving.

multicast: communication capability which denotes unidirectional distribution from a single source to a number of specified destinations, usually without obtaining acknowledgement of receipt of the transmission

NOTE: In a network, a technique that allows data, including packet form, to be simultaneously transmitted to a selected set of destinations.

narrowcast : communication capability which denotes unidirectional distribution of traffic data from a single source to a very small number of specified destination access points, which makes possible the unidirectional transmission of reception acknowledgements from the small group of receivers to the single source of information

unicast: communication capability which denotes the transmission of data between a single sender and a single receiver over a network

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

2G/3G	2nd/3rd Generation
3GnetSim	3rd Generation Network Simulator
3GPP	3rd Generation Partnership Project
AAL5	ATM Adaptation Layer 5
AC	Admission Control
ACK	ACKnowledgment
Acpt	Acceptance
AI	Acquisition Indicator
AICH	Acquisition Indicator CHannel
AM	Acknowledged Mode
AOA	Angle Of Arrival
API	Access Preamble Indicator
APN	Access Point Name
ARQ	Automatic Repeat reQuest
ASC	Access Service Class
ASYM	ASYMmetrical
ATB	Advanced S-UMTS Test Bed
ATM	Asynchronous Transfer Mode
AWGN	Additive White Gaussian Noise
B/M	Broadcast/Multicast
BC	BroadCast domain
BCCH	Broadcast Control CHannel (logical control channel)
BCH	Broadcast CHannel (transport channel)
BE	Best Effort
BLER	BLock Error Rate
BMC	Broadcast/Multicast Control
BM-IWF	Broadcast/Multicast Inter Working Function
BM-SC	Broadcast/Multicast Service Center in MBMS
BO	Buffer Occupancy
BPSK	Binary PSK
BS	Base Station
BW	BandWidth
CAC	Call Admission Control
CAI	Channel Assignment Indicator
CB	Cell Broadcast

CBC	Cell Broadcast Centre
CBE	Cell Broadcast Entity
CBMC	Control BMC
CBQ	Class Based Queuing
CBR	Constant Bit Rate
CBS	Cell Broadcast Service
CC	Call Control
CCCH	Common Control CHannel
CCPCH	Common Control Physical CHannel
CCTrCH	Coded Composite Transport CHannel
CDI	Collision Detection Indicator
CDMA	Code Division Multiple Access
CK	Ciphering Key
CN	Core Network
COM	COMmunication
CONT	CONTent delivery
CPCH	Common Packet CHannel
C-PHY	primitives for the Control of the configuration of the PHYsical layer
CPICH	Common PIlot CHannel
C-Plane	Control Plane
CRC	Cyclic Redundancy Check
CRLC	Control RLC
CRNC	Controlling RNC
CRNTI	Control RNTI
CS	Circuit Switched
CTCH	Common Traffic CHannel
CTCH-BS	Common Traffic CHannel Block Set
DCCH	Dedicated Control CHannel (logical channel)
DCH	Dedicated CHannel (transport channel)
Delay-EDD	Delay Earlier Due Date
DL	Down-Link
DPCCH	Dedicated Physical Control CHannel
DPCH	Dedicated Physical CHannel
DPDCH	Dedicated Physical Data CHannel
dRoD	dynamic Rate on Demand
DRODCH	Dynamic Rate On Demand CHannel
DRR	Deficit Round Robin
DRX	Discontinuous Reception
DSCH	Down-link Shared CHannel
E_b/N_t	Energy per bit over Noise power density ratio
E-BIA	Enhanced Internet Access
E_c/N_o	Energy per chip over Noise power density ratio
EDC	Error Detection and Correction
EDD	Earlier Due Date
ESA	European Space Agency
EVC	EVent Code
FACH	Forward Access CHannel
FBI	FeedBack Information
FCFS	First Come First Served
FCS	Fast Cell Selection
FDD	Frequency Division Duplex
FEC	Forward Error Correction
FER	Frame Error Rate
FIFO	First In First Out
FL	Forward Link
FSM	Finite State Machine
FSS	Fast Satellite Selection
GAUSS	Galileo And UMTS Synergetic System
GEO	GEOstationary orbit
GGSN	Gateway GPRS Support Node
GNSS	Global Navigation Satellite System

GoS	Grade of Service
GPRS	General Packet Radio Service
GPS	Global Positioning System / Generalized Processor Sharing
GSM	Global System for Mobile communications
GSN	GPRS Support Node
GSO	Geo-Stationary Orbit
GTP	GPRS Tunnelling Protocol
GTP-C	GPRS Tunnelling Protocol for Control plane
GTP-U	GPRS Tunnelling Protocol for User plane
GW	GateWay
HARQ	Hybrid ARQ
HC	Handover Control
HFN	Hyper Frame Number
HLR	Home Location Register
HPPICH	High Penetration Page Indicator CHannel
HSDPA	High Speed Data Packet Access
HSPA	High Speed Packet Access
HS-PDSCH	High Speed-Physical Downlink Shared CHannel
HSS	Home Subscriber Server
ICH	Indicator CHannel
ID	IDentifier
IE	Information Element
IETF	Internet Engineering Task Force
IGMP	Internet Group Management Protocol
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ISCP	Interference Signal Code Power
IST	Information Society Technologies
Kbps	Kilo bit per second
L1	Layer 1
L2	Layer 2
L3	Layer 3
LA	Location Area
LC	Load Control
LEO	Low Earth Orbit
LI	Low Interactivity
LLr	Linked List of addresses of packets to be re-emitted
LMMSE	Linear Minimum Mean Squared Error
LMS	Least Mean Square
MAC	Medium Access Control
MAC-b	Medium Access Control broadcast
MAC-c	Medium Access Control control
MAC-d	Medium Access Control dedicated
MAC-hs	Medium Access Control high speed
MaxSFN	Maximum System Frame Number
MBMS	Multimedia Broadcast Multicast Service
Mbps	Mega bits per second
M-commerce	Mobile commerce
Mcps	Mega chip per second
MEO	Medium Earth Orbit
MFTP	Multicast File Transfer Protocol
MIB	Master Information Block
MLD	Multicast Listener Discovery (protocol)
MLP	MAC Logical channel Priority
MM	MultiMedia / Mobility Management
MS	Mobile Station
MSB	Most Significant Bit
MT	Mobile Terminal
MU	Mobile User
MUI	Message Unit Identifier
NACK	Negative ACKnowledgement

NAS	Non Access Stratum
NAV	NAVigation
NI	Non Interactive
NL-HPA	Non Linear High Power Amplifier
NP	Narrowcast Protocol
NRT	Non Real Time
nrt-VBR	non real time Variable Bit Rate
NW-RRC	NetWork Radio Resource Control
O-QPSK	Offset QPSK
OVSF	Orthogonal Variable Spreading Factor
PA	Paging Area
PAD	PADding
PAM	Pulse Amplitude Modulation
PBR	Peak Bit Rate
PC	Power Control
PCCC	Parallel Concatenated Convolutional Code
PCCH	Paging Control CHannel (logical channel)
P-CCPCH	Primary Common Control Physical CHannel
PCH	Paging CHannel
P-CIPCH	Primary Common PIlot CHannel
PDCP	Packet Data Convergence Protocol
PDP	Packet Data Protocol
PDSCH	Physical DSCH
PDU	Protocol Data Unit
PG	Processing Gain
PGPS	Packet Generalized Processor Sharing
PHY	PHYsical layer
PI	Page Indicator
PICH	Paging Indicator CHannel
PIM	Protocol Independent Multicast
PLMN	Public Land Mobile Network
PMM	Packet Mobility Management (in GPRS)
PN	Pseudo Noise
PRACH	Physical Random Access CHannel
PS	Packet Switched / Scheduler
PSC	Primary Synchronization Code
P-SCH	Primary Synchronization CHannel
PSK	Phase Shift Keying
PVT	Position, Velocity and Time
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature PSK
RA	Routing Area
RAB	Radio Access Bearer
RACH	Random Access CHannel
RAN	Radio Access Network
RANAP	Radio Access Network Application Part
RASA	Random Access Spread Aloha
RAT	Radio Access Technology
RB	Radio Bearer
RBAM	Radio Bearer Allocation and Mapping
Req	Request
RF	Radio Frequency
RL	Return Link
RLC	Radio Link Control
RM	Resource Management
RNC	Radio Network Controller
RNP	Radio Network Planning
RNS	Radio Network Sub-system
RNTI	Radio Network Temporary Identity
RR	Round Robin
RRC	Radio Resource Control

RRM	Radio Resource Management
RS	Reed Solomon (Encoder/Decoder)
RSCP	Received Signal Code Power
RSSI	Received Signal Strength Indicator
RT	Real Time
RTD	Research and Technological Development
RTP	Real Time Protocol
RTT	Round Trip Time
Rx	Receive
SABP	Service Area Broadcast Protocol
SAP	Service Access Point
Sat	Satellite
SATB	SATellite Trial planned for ATB system in Phase 2
SATIN	SATellite UMTS IP-based Network
S-AUC	Satellite AUthentication Centre
SC	Service Centre
SCCH	Shared Control CHannel
SCCP	Signalling Connection Control Part
S-CCPCH	Secondary Common Control CHannel
SCFQ	Self Clocked Fair Queuing
SCH	Synchronization CHannel
SDU	Service Data Unit
Sf	interface between Satellite and gateway
SF	Spreading Factor
SFN	System Frame Number
SGSN	Serving GPRS Support Node
SHCCH	SHared Common Control CHannel
SI	Status Indicator
SIB	System Information Block
SICH	Signalling Indication CHannel
Sii	air interface between IMR- SAT
SIR	Signal to Interference Ratio
SIS	Signal In Space
SMS-CB	Short Message Service Cell Broadcast
SN	Sequence Number
SNIR	Signal to Noise plus Interference Ratio
SNR	Signal to Noise Ratio
SOS-M	Emergency Message
SP	Service Provider
SRB	Signalling Radio Bearer
SRI	Satellite Radio Interface
S-RNC	Satellite RNC
SRNC	Serving Radio Network Controller
SSC	Secondary Synchronization Code
S-SCH	Signalling Radio Bearer
SSDT	Site Selection Diversity Transmission
STFQ	Start Time Fair Queuing
Sui	air interface between IMR-UE, SAT-UE, IMR- SAT
S-UMTS	Satellite Universal Mobile Telecommunication Systems
SW	SoftWare
SW-CDMA	Satellite Wideband CDMA
T	Terrestrial
TB	Transport Block
TBS	Transport Block Set
TCH	Traffic CHannel
TCP	Transport Control Protocol
TCTF	Transport Channel Type Field
TDD	Time Division Duplex
TEID	Tunnelling Endpoint IDentifier
TFC	Transport Format Combination
TFCI	Transport Format Combination Indicator
TFCS	Transport Format Combination Set

TFI	Transport Format Indicator
TFS	Transport Format Set
TM	Transparent Mode
TMGI	Temporary Multicast Group Identifier
TPC	Transmit Power Control
Tr	Transparent
TrCH	Transport CHannel
TSTD	Time Switched Transmit Diversity
TSTP	Time STamP
TTI	Transmission Time Interval
T-UMTS	Terrestrial UMTS
TV	Virtual spacing Time
Tx	Transmission
UBR	User Bit Rate
UDP	User Datagram Protocol
UE	User Equipment
UL	UpLink
UM	Unacknowledged Mode
UMTS	Universal Mobile Telecommunication System
UNI	UNIdirectional
Upd.	Update
U-Plane	User Plane
URAN	UMTS Radio Access Network
U-RNTI	Utran RNTI
USRAN	UMTS Satellite Radio Access Network
UT	User Terminal
UTRA	Universal Terrestrial Radio Access (3GPP)
UTRAN	UMTS Terrestrial Radio Access Network
UW	Unique Word
VoD	Video on Demand
WCDMA	Wideband COMA
WFQ	Weighted Fair Queuing
WLAN	Wireless Local Area Network
WRR	Weighted Round Robin

4 Project descriptions

4.1 ESA/ATB

4.1.1 ATB packet access

This contribution summarizes some of the activities on packet access performed in the frame of Phase I of the ESA contract "Advanced S-UMTS Test Bed" (ATB, ESA Contract No. 15208). The results here reported are then the outcome of the work performed by several companies in the ATB team under the ESA guidance.

ATB Phase I activity had the objective of investigating strategies for packet support in SW-CDMA and then validating the proposed solutions through the development, in Phase II of the above contract (currently on-going), of a comprehensive hardware Test Bed allowing to test such solutions in a context faithfully representative of a real S-UMTS environment.

The approach followed is to analyse the techniques currently proposed for supporting the packet mode of T-UMTS W-CDMA 3GPP air interface (Release 99 and the still on-going Release 5) and when necessary to adapt taking into account the specific satellite environment. The GEO satellite constellation case has been considered as baseline as it is considered to represent the most challenging configuration for the analysis of the packet mode. However, results are considered applicable also to other satellite constellations.

This contribution will concentrate on the work performed in order to define, assess and optimize new operational modes for the specific support of point-to-point packet access. The new packet and multicast operating modes studied in the frame of ATB are expected to boost up data transmission efficiency and hence can be particularly helpful in increasing the appeal of future S-UMTS systems. To this end, the new Test Bed envisaged in ATB will further develops the remarkable testing and validation capabilities offered by the ROBMOD Test Bed (RTB), by incorporating the new packet and multicast supporting features allowing to satisfactorily experiment those new modes.

This contribution will in particular report on the trade-offs, definition and simulation of point-to-point packet operating modes for both up and downlink. The work has concentrated more on the up-link due to the higher difficulty to extend UTRA devised packet mechanism to the satellite environment on the reverse link direction as compared to the forward link direction.

4.1.2 ATB investigation on multicast/narrowcast

This contribution summarizes the activities on multicast and narrowcast in the frame of Phase 1 of the ESA contract "Advanced S-UMTS Test Bed" (ATB, ESA Contract No. 15208).

Multicast, broadcast and narrowcast are all applications which fit well for satellite applications due to the inherent one-to-many structure of satellite communication systems. This contribution reports on the trade-offs, definition and simulation of multi-cast and narrowcast for Satellite-UMTS. Specifically the submission investigates:

- large block interleaving and RS coding;
- medium block interleaver with CRC;
- hybrid short Carousel and FEC with interleaving; and
- narrowcasting.

The Large block interleaving study showed that to achieve an codeword error rate of less than 10^{-6} an interleaver of 10^5 codewords and a RS(255,175) code is needed to cover all elevation angles. If the requirements on the minimum elevation angle (dependent on position), the maximum error rate (dependent on the application), or the vehicle speed, were raised then the interleaver size could be reduced. The memory needed for such an interleaver is approximately 50 Mbytes (two required).

The medium block interleaver with CRC was also thoroughly investigated. It was found that even for a vehicle travelling at 120 km/h the average outage is in the order of one second. The histogram of the outages showed that there are more small outages, however, compared to the errors caused by the longer outages, this is insignificant. This result was confirmed through simulation results. The technique therefore was seen as not being useful as it does not consider the worst outage conditions.

An investigation was performed into a combined short Carousel and FEC scheme for multicast. The idea here was to see if implementing a fast carousel scheme together with the FEC scheme could reduce the interleaver size but maintain the performance and to some degree the throughput. Poor performance results were obtained which showed that short carousel techniques bring little performance improvement over satellite elevation, as well as that coding implemented within carousel blocks brings little gain, as well the code rate reduces the overall throughput substantially.

Finally narrowcast with ARQ was investigated through simulation. Results were determined for different code rates, data rates, and vehicle speeds. Performance was plotted in terms of overall delay needed to make sure a given packet was decoded reliably. Other plots showing the repeat requests and the NACQ messages were also plotted. From the information gathered a final parameter selection could be determined.

4.2 IST/SATIN studies/project description

SATIN, a project bringing together seven partners from five different EU member countries, commenced its activities in January 2001 with the following specific objectives:

- Identify the suitable service scenarios for S-UMTS through market and business analysis.
- Propose a closely integrated S/T-UMTS architecture considering the trade-offs related to the system components (satellite, terrestrial elements, terminal).

- Define/optimize the S-UMTS access scheme in packet mode.
- Verify key issues of this access scheme (Layer 1+ Layer 2/2+ elements) via simulation.

SATIN completed successfully its work in 2 years and 3 months. Within this period, 6 technical reports and 2 reports on dissemination and standardization activities were delivered.

At the initial stage of the project, potential business/market approaches that could lead to a profitable S-UMTS were investigated via market and business analysis. The study clearly indicated the need to address the mass, consumer market and suggested the delivery of *broadcast/multicast* multimedia services as the way to achieve this.

At the second stage of the project, two main system architecture scenarios were considered, reflecting two radically different approaches with reference to the terrestrial mobile network (T-UMTS): the *service complement* or close co-operative approach and the *geographical complement* approach. In the service complement approach, SATIN proposed a new architecture that involved the use of IMR (Intermediate Module Repeater) collocated with UMTS Node B.

In response to the outcome of the market and business study, the service complement approach was maintained in SATIN. At service level, the system complements the service offer of T-UMTS providing an efficient overlay multicast/broadcast layer. This efficiency was confirmed via a comparison study between T and S-UMTS networks for delivery of multicast broadcast services at the late stage of the project where it was shown that the satellite delivery was more cost effective than cellular for such MBMS services provided the ARPU is less than about 10 €/users At network level, the selected architecture features close integration with T-UMTS implying significant benefits in terms of R&D and system development cost reduction. Two scenarios, one *baseline* and one *optional* were defined. In the baseline scenario the return link is provided via terrestrial networks, while in the optional scenario a low-rate satellite return link is available.

The third stage of the project initially focused on the requirements identification for adaptation of the UMTS specifications for multicast/broadcast delivery via satellite; the apparent, subsequent step was the detailed definition of the respective packet based access scheme, targeting maximum commonalties with the UTRA FDD air-interface.

At the final stage of the project, the defined access scheme was validated via simulation analysis and a capacity evaluation was made. The feasibility of the SATIN architecture was established.

Throughout the project duration, dissemination and standardization activities were carried out in full swing and contributions were made to ETSI and ITU standardization working groups. SATIN was also present at several conferences including VTC, AIAA, PIMRC, GLOBECOM and IST Mobile Summit.

Overall the project successfully established an integrated S/T-UMTS architecture with the satellite part delivering the MBMS services. The process of standardization was started via ETSI-SES where initial technical documents have been accepted and will be continued via a follow on project MoDiS. Terrestrial mobile operators have been alerted to the system and are showing interest.

4.2.1 Main constraints and requirements on SATIN access

The SATIN access scheme is based on the following major constraints and requirements:

- No Packet Data Protocol (PDP) messaging between User Equipment (UE) and Satellite Radio Network Controller (Satellite-RNC).
- Compliance with the current core requirements of the Multimedia Broadcast/Multicast Service (MBMS) as specified within the on-going 3GPP Release 6 framework, in particular use of same low layer radio bearer for both the Broadcast and Multicast services.
- Need for dynamic allocation of radio resources with service on-demand.
- No real time interaction between UEs and Satellite-RNC.
- Return link capabilities: 3 kbps to 8 kbps with respect to user equipment capabilities via GEO satellite.
- The design of the data transport in the Return Link shall be optimized for providing the required QoS of short and bursty broadcast/multicast service requests.
- Restricted use of feedback/loop mechanisms (due to high round-trip delay constraint).

The resulting main access scheme features:

- Choice of transport channel type for the satellite forward link: FACH transport channel (versus DSCH).
- Choice of transport channel type for the satellite return link (applicable to optional architecture scenario): RACH transport channel (versus DCH).
- Choice of RACH procedure type: message ramping should be employed, in which no acquisition indication feedback is used.
- BMC protocol sub-layer is retained for adaptation as means to support broadcast/multicast traffic in the forward link (in both the baseline and optional architecture scenario).
  As broadcast/multicast data transfer should be operated in the Packet Switched (PS) domain [15] and not in the Broadcast (BC) domain, one assumes in SATIN that future *broadcast/multicast MBMS RAB* (handled by RANAP protocol) is compatible with the operating of point-to-multipoint radio bearer provided by BMC/RLC (whether traffic is scheduled or not in the BM-SC/CN, i.e. traffic volume and capacity reservation are not necessarily required).

The following main needs for adaptations and possible optimizations have been identified for further study and definition:

- Service notification and radio resource set-up for a set of common traffic channels (supporting Broadcast/Multicast bearers of BMC entities), building upon the use of the configuration of the common channels of the cell/spotbeam, made available to UEs via SIB messaging on BCCH.
- Enhancement of transport dynamics of cell broadcasting, i.e. fixed versus variable Transport Block size (Release 99 specifications assuming variable number of Transport Blocks only).
- Evaluation of random access procedure performance, considering both slotted and non-slotted Aloha operating modes.
- Admission and Load control performance optimization, considering throughput and interference-based strategy.
- Packet scheduling optimization, considering *multilevel priority with exhaustive service* and *Weighted Fair Queuing* schemes.
- Cell search procedure optimization considering W-CDMA and SW-CDMA.
- High order modulation and advanced turbo/layered coding schemes.
- IMR presence impact on radio propagation channel.

4.2.2 SATIN key issues

This clause will highlight the issues deemed as critical for an effective S-UMTS packet mode implementation, hence requiring particular care in the specification, simulation and evaluation phases of the air interface development.

4.2.3 Layer 2 and L2+ key issues

- **QoS handling** that address possible alternatives in the QoS mapping and QoS ranking (priority handling of the multiple services/flux based on QoS parameters, e.g. traffic priority).
- **Radio resource management**, that encompasses admission control and load control, which interact with the Packet Scheduler element and more particularly the scheduler of multicast flux in the Forward Link (assuming a given QoS per multicast address/service group).

- **Radio resource allocation** which performs the assignment of the radio resources, in conjunction with the packet scheduling process: the latter decides the optimal bit rates and the length of the allocation for the different broad/multicast services, each of them being assigned one transport channel and one channelization code "at a time" (and assuming a given QoS per multicast address/service group). One suggests to base packet scheduling on combination of code division and time division scheduling and to assess different queuing schemes (e.g. Multilevel priority with exhaustive service, Weighted Fair Queuing) in terms of fairness and effective throughput across L2/L2+ stack. QoS providing should be achieved with minimum processing demand on the user terminals and, in case of the baseline scenario without real-time interactive communication with the user (no RL via satellite).Possible alternatives in the setting of Radio Bearer attributes for providing the requested QoS may also be addressed (e.g. Real-Time services, Variable bit rate vs. Constant bit rate).

- **Optimization of RACH-like mode** using Slotted ALOHA in a GEO context:

    - identification of suitable, i.e. "optimized' RACH-related parameters to be used in relation to the RACH performance;

    - increment of signatures available (using higher order of Hadamard codes to extend the signature sets).

  All these enhancements need to take into account:

    - B/M traffic characteristic (packet size) and QoS trade-off (delay versus throughput);

    - acquisition procedure schemes: preamble ramping, message power ramping.

- **QoS differentiation on the traffic over the RACH:** this looks at the performance of the B/M traffic when there is some sort of signalling prioritization on the RACH.

4.2.4 Layer 1 key issues

- Forward link Transport/Physical channels for MBMS

The FACH channel is selected to be the most suitable channel for the SATIN access scheme. The DSCH could prove to be an appropriate channel, but the operation would require substantial modifications of the T-UMTS standards, whereas the FACH can be used with only minor modification(s) in the MAC header content.

- Layered Coding

Layered coding schemes (softly degrading schemes) perform decoding up to a level depending on the channel Signal to Noise Ratio (SNR) and may be used to achieve rate adaptivity (where we take advantage of good SNR conditions to increase the information rate) or protection adaptivity (where we transmit extra coded bits that are employed in order to fight fading at the cost of an extra complexity and/or delay).

The receiver will be able to use only the necessary layers that lead to the required BER, which means that the receiver will fit to the channel conditions and modify the error correcting capability (and the corresponding complexity and delay) according to the channel variation. The feature that makes layered coding schemes suitable for broadcast transmission is that the encoder transmits at the same information rate, while each user decoder decides its level of decoding independently, as a function of its own channel conditions.

- Turbo coding

Standard and enhanced turbo coding and decoding techniques shall be addressed for comparison purposes with the proposed advanced layered coding techniques.

- High Order Modulation

High order modulation techniques are bandwidth efficient and, hence, the system capacity can be increased using such schemes. However, they are difficult to implement and need more power depending on the order. Moreover, non-linearity in the payload amplifier impose restrictions on higher order modulation schemes.

- Synchronization procedure

W-CDMA and SW-CDMA cell search procedure shall be investigated and contrasted in terms of performance and complexity. SATIN environment appropriate modification could be introduced to reduce receiver complexity and mean time to acquisition.

- Random access procedure

The method of preamble power ramping with fast acquisition as employed in the T-UMTS is not applicable and is replaced by a one-shot acquisition technique with message power ramping. Modifications to the preamble structure that decrease the collision risk and access delay, and increase the throughput shall be investigated.

- Paging procedure

Appropriate paging procedures aimed at reducing power consumption and system transmission overhead is investigated in the proposed scenario, namely in the case of MBMS supported by common transport/physical channels (e.g. FACH).

- Power control

Traditional power control procedures are not applicable to MBMS delivered via common channel (e.g. FACH), hence procedures based, for example, on worst case or cell/beam statistic algorithms are investigated and described.

- Channel propagation models in the presence of IMR

The IMR presence, aiming at improving system coverage in indoor and urban environments, considerably changes the radio propagation channel characteristics: the same satellite signal is repeated by every IMR in the coverage area introducing a supplementary multipath propagation not experienced in other environments. This new feature must be considered both in the evaluation of radio interfaces adopting the terrestrial repeater concept, and in the conformance tests of the terminals. To this aim, two reference radio propagation models have been proposed to be used for performance evaluation of radio interfaces.

4.2.5 Service provision principles for MBMS

SATIN service access is based on the main phases of MBMS provision [15], as illustrated in Figure 1:

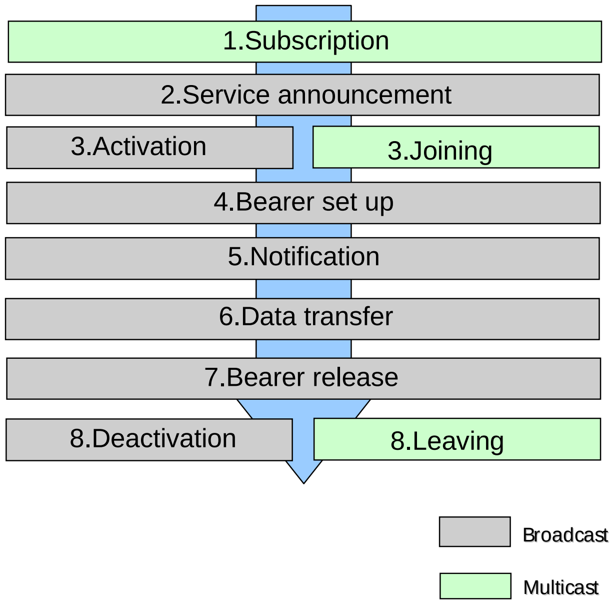


Figure 1: SATIN service provision phases

4.3 ESA-3GNetSim

4.3.1 Objective

The main objective of the 3GnetSim is to design, develop and validate a modular and efficient radio access network simulator for third generation mobile satellite systems to carry out simulation runs with a view to optimize the simulated system. Based on a thorough analysis of the simulation results that activity shall yield contributions to the standardization of the higher layer protocols involved.

3GnetSim is based on the OPNET modeller and will cover in particular the capability to perform a thorough analysis of Satellite UMTS radio access networks as well as to study the interaction of the S-UMTS radio access networks with the terrestrial UMTS IP based core networks.

4.3.2 Target system architecture

The target scenario of the simulator shall include GEO satellites with multiple spot beams. In order to reduce the bandwidth consumption in the feeder links, several GW stations shall be foreseen. The hierarchy principle in UMTS, which assigns to each Node B exactly one RNC and to each RNC exactly one SGSN implies a fixed assignment of each spot beam to exactly one GW station. Thereby, for each geographical area being covered by a group of spot beams, a GW station for this group and thereby for the corresponding area will be required. This would allow sharing of a satellite by different e.g. national operators, if the GW stations would be connected to different core networks. See Figure 2 for an illustration of the chosen target architecture.

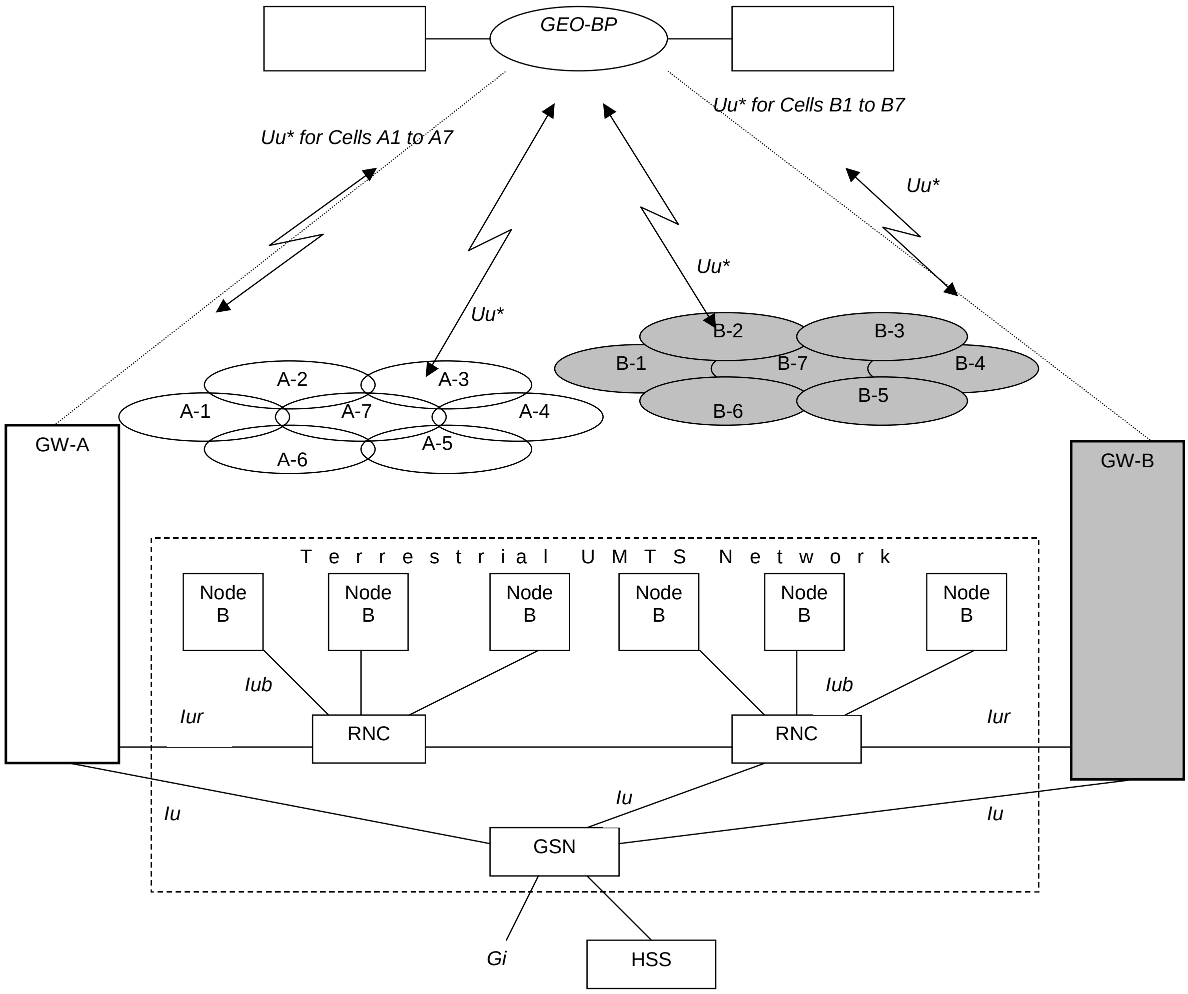


Figure 2: Target Architecture of the 3GNetSim

4.3.3 General concept of the simulator

The 3G Network Simulator to be developed shall cover a broad spectrum of simulation scenarios. The addressed simulation scenarios determine the required simulator functionality, ranging from detailed simulations of the radio access network (PHY, MAC, RLC, RRC), Iu protocols and CN procedures to the detailed modelling of user and application scenarios. Moreover the simulation scenario determines at which degree of accuracy a certain function has to be modelled. E.g. for the verification of the handover procedure the corresponding protocol has to be modelled accurately, whereas for simulations of overall system throughput it is sufficient to model the handover algorithm in an abstract way.

In the following, some scenarios are described and the main aspects to be considered therein are identified.

Analysis, verification and optimization of protocols

In the 3G Network Simulator Project S-UMTS protocols have to be developed, analysed, verified and optimized. For this purpose, it is necessary perform the following tasks:

- Test whether a designed protocol actually works. For this a comprehensive set of test cases has to be defined. Such test cases include extreme system constellations, which impose difficult conditions for the functionality of the protocol. E.g. it has to be verified that an RRC connection can be set-up also at the cell edges or in shadowed locations.

- Determine the probability of protocol failure, e.g. what is the probability that power-up or power-down commands of the closed loop power control are received incorrectly.

- Optimization of the protocol parameters, e.g. how to set the thresholds and hysteresis for handovers, such that the signalling overhead of the complete handover procedure is minimized.

- Optimization of protocol strategies, e.g. which retransmission scheme yields a minimal average delay or a minimum delay jitter.

The crucial characteristics of this type of simulator applications are that:

- The protocols of interest have to be modelled in detail.

- Only a very restricted set of protocols has to be modelled, depending on the simulation goal, e.g. for the analysis of the handover procedure protocols for RAB setup and release have to be included, but e.g. the ARQ protocol is not relevant and need not be modelled.

- The functionality of the protocol itself is addressed, but not its influence on the overall network performance.

- For C-plane protocol simulation, user traffic is not relevant at all. For the U-plane protocols it is sufficient to consider traffic on bearer level only, but not on application level.

Although the influence of the protocols on the overall network performance is not addressed in this type of simulations, the opposite direction, i.e. the influence of the network load on the protocol performance has to be regarded in some of the simulation scenarios, e.g. for simulations on the functionality and performance of the handover procedure. For this a module, which generates virtual system load and interference can be applied.

Simulations of per link performance figures

In this type of simulations the performance of a single peer-to-peer communication link is simulated, where peer-to-peer need not necessarily mean user-to-user but may also refer to the communication between e.g. the UE MAC and the URAN MAC. The goals of this type of simulations are e.g. to:

- determine the attained QoS of a single connection, e.g. on end-to-end application level or on bearer level;

- determine the Performance Profile of a transport channel in the Uu interface;

- optimize algorithms and protocols, such that attained QoS of a single connection is maximized, e.g. optimization of the packet scheduling such that the required delay can be attained.

The crucial characteristics of this type of simulator applications are that:

- in general only the relevant protocols of a single peer-to-peer connection have to be modelled in detail;

- for the link of investigation not only the U-plane, but also the relevant C-plane protocols have to be modelled;

- the performance of a single link, but not the overall network performance is of interest.

Although the determination of overall network performance is not of interest in this type of simulations, the opposite direction, i.e. the influence of the network load on the single link performance has to be regarded in some of the simulation scenarios, e.g. for the optimization of scheduling strategies. For this a module, which generates virtual system load and interference can be applied.

Evaluation and optimization of RRM algorithms

The goals of this type of simulations are to:

- optimize RRM algorithms such as Call Admission Control, Load Control, Handover- and Diversity Control in terms of overall system performance figures such as overall throughput, overall blocking probability, etc.

The crucial characteristics of this type of simulator applications are that:

- Only user traffic is simulated, signalling is regarded only in terms of additional load.
- Only those parts of the algorithms and required protocols are modelled in detail, which are subject of the specific optimization task and have an essential impact on the system performance figure, with respect to which the algorithm is optimized. All other system aspects are modelled only on an abstract level.
- Since the algorithms are optimized with respect to overall system performance figures a comprehensive network containing a large number of parallel users has to be modelled.

Simulation of overall network performance figures

The goals of this type of simulations are:

- Simulation of overall network capacity, signalling and traffic load, bottlenecks.
- Get measures on offered traffic, blocked traffic, and throughput.
- Simulation of call blocking, handover failure, etc.
- Efficiency improvements due to broadcast.
- Overall roaming patterns and performance; inter and intra-segment.
- The performance of certain algorithms (RRM) is obtained under various traffic conditions.

The crucial characteristics of this type of simulator applications are that:

- The entire network is covered, including all S-UMTS cells and users with realistic mobility profiles.
- Realistic terminal types and traffic mixes are simulated.
- Thousands of calls are generated over an extended period of time.
- Interactions of different users is accounted for both within a cell and between cells (resource contention).

In order to appropriately deal with these - in several aspects - contradictory requirements, two simulators will be implemented:

- A *Protocol Simulato*r: This simulator models the protocols and the Uu and Iu procedures in a detailed and comprehensive way. For added productivity the "OPNET Radio Module", the "OPNET UMTS Model" and the "OPNET IP Multicast Model" in addition to the baseline OPNET Modeller will be applied. In the protocol simulator the relevant protocols and algorithms are modelled in a detailed way, such that they can realistically and reliably be analysed, verified, evaluated and optimized. Due to the high modelling accuracy only a few parallel communication links can be simulated in detail simultaneously. Additionally, background load is modelled to take into account the effects of a large number of virtual users. The Protocol Simulator will be applied for:
    - the analysis, verification and optimization of protocols;
    - simulations of per link performance figures;

and

- A *Network Performance Simulato*r: This simulator models aspects such as traffic generation, terminal properties, application characteristics, user mobility, network resources, and signalling and traffic load at a high level of abstraction. It is thus capable of simulating many links in parallel, i.e. a large number of users and a correspondingly high traffic volume to obtain reliable statistics, i.e. on overall throughput, call-blocking and coverage issues. Only the baseline OPNET Modeller product is required. The Network Simulator will be applied for:
    - the evaluation and optimization of RRM algorithms in terms of system performance figures;
    - the simulation of overall network performance figures.

4.4 GAUSS project

GAUSS is a Research and Technological Development project co-funded by European Commission, within the frame of the Information Society Technologies (IST) V Programme. It is a two-year project, starting from December 2000, and successfully completed.

GAUSS objective was to design and demonstrate the feasibility of a system providing high quality Location-based services, from the integration of Satellite Navigation and Communications, within the contexts of GALILEO and the UMTS technology.The GAUSS solution addresses two classes of users profiles, belonging to info-mobility and inter-modality (road-river) markets. It supports highly reliable, near real-time two-way communication between Mobile Users and Service Centre/Provider.

GAUSS developed a system, which integrates advanced communication and precise navigation, for providing reliable and effective location-based services oriented to transport and mobility applications (freight and fleet management, road safety and info-mobility, emergency assistance, dangerous goods transportation control, inter-modal transport). The GAUSS system combines off-the-shelf and new developed technologies, using S-UMTS 3GPP compliant point-to-point and broadcasting packet-based communication and GNSS1-EGNOS accurate positioning. It has an open architecture ready to operate with the future GALILEO and UMTS, and further exploitation scenarios.

4.4.1 GAUSS Services

GAUSS main rationales rely on the feasibility of a solution for the provision of high quality Location-Based services, characterized by:

- high integrity/continuity positioning and guarantee of performance for navigation;
- highly reliable and available communications, featuring small latency with low-rate data transmission;
- very large population of Mobile Users with respect to the number of Service Centres/Providers;
- wide coverage areas, easily accessible from satellite.

The main concept which the GAUSS project is based on, envisages the communication and navigation system components fitting within the general framework of S-UMTS and GALILEO (GNSS-2, Global Navigation Satellite System - Phase 2). The technological issues of such a concept rely on the development of a Demonstrator, which integrates existing and available facilities with ad-hoc designed components. The former ones constitute the ground segment, the latter ones include the advanced user terminal and the innovative services and applications.

The services considered for GAUSS are based on two-way communication between Mobile Terminal (Mobile User) and Service Centre/Provider(s), at low data rate transmission of small data packets as typically required by Info-Mobility and Inter-Modality oriented applications. Messages from Mobile Terminal always carry very accurate positioning and timing information (Position-Velocity and Timing - PVT) data, messages from Service Providers carry service provisioning or acknowledge (ACK).

GAUSS Location-Based services are characterized by very demanding requirements in terms of global coverage, continuity, high integrity/continuity positioning and guarantee of performance for navigation (1 m to 5 m accuracy for positioning), communication reliable, near real-time, small latency featuring, with time response (time elapsed between the instant of message sending and the relevant answer reception) guaranteed (ranging from 25 s to 60 s). These services are characterized by bursty and unbalanced traffic in the RL, generated by a large number of Mobile Users towards a relatively small number of Service Providers, and vice versa from the Service Providers towards widely geographically sparse Mobile Users (i.e. greater amount of traffic in the return link with respect to the forward link). Both asynchronous and synchronous communications are supported, based on:

- broad-casting (i.e. data distribution from a SP to MUs);
- broad-catching (i.e. data collection from MUs to a SP);
- point-to-point schemes.

Resource access is based on CDMA (Code Division Multiple Access), according to the UMTS standard.

5 Conclusion and recommendation of the projects

Table 1 highlights the different key issues/founding covered in the projects. The ISO layer approach was used as a general framework.

Table 1: Key Issues of the different Projects

	ATB	Satin	3GNetSim	Gauss
		MBMS with no or little interactivity	Client/Server request response models	Location based services: GAUSS 53 byte cells (up to eight concatenated cells) on CORBA (Common ORB Architecture) -IIOP (Internet Inter Orb Protocol) at the Gateway Station
Layer 6: Presentation				
Layer 5: Session		IGMP (see note)	Unicast: 3GPP-based (on demand PDP context) Multicast: MBMS 3GPP Release 6 draft (IGMP and MBMS Session)	
Layer 4: Transport		UDP/TCP	UDP/TCP	UDP (GAUSS cells encapsulated into UDP fragments)
Layer 3: Networking/RRM	Dynamic Resource Assignment for RL packet transmission on DRODCH	3GPP-based (RANAP) (see note 2)	3GPP-based: U-plane: IP C-plane: RANAP/RRC	IP (U-Plane)/RRC (C-plane)
Layer 2+: RLC	Combined FEC/Interleaving for multicasting without return channel	3GPP-based (PDCP/BMC/RLC/MAC) (see note 3)	3GPP-based RLC, improved (delay reduced) FEC/Interleaving scheme for MBMS	PDCP (U-plane only)/BMC(U-plane only)/RLC/MAC (see note 4)
Layer 2: MAC	Combined DRODCH (see note 5) **/** RASA (see note 6) for RL packet transmission		3GPP-based MAC. DRODCH/RASA for RL Packet Access. Modified DSCH for FL Packet Transmission.	
Layer 1: PHY, transmission	SW-CDMA: (DSCH+SSCH), FACH, RACH, dRoD (see note 7)	W-CDMA+ (see note 8) (FACH transport for Broadcast/Multicast using group signalling in MAC frames).**/** SW-CDMA (RACH transport)	Simulation of SW-CDMA	SW-CDMA

NOTE 1: Restricted to Join/Leave functionality.
NOTE 2: Adapted version of the 3 GPP (Rel.5) standard.
NOTE 3: Adapted version of the 3 GPP (Rel.5) standard.
NOTE 4 PDCP (L2, for UDP/IP header compression - user plane only).
BMC (L2,for broadcast/multicast message handling - user plane only).
RLC/MAC (L2, for both user plane and control plane).
RRC (L3, control plane only).
NOTE 5: DRODCH: Dynamic Rate on Demand Channel.
NOTE 6: RASA: Random Access Spread Aloha.
NOTE 7: DROD: Dynamic Rate on Demand.
NOTE 8: Adapted version of the 3 GPP (Rel.5) standard.

5.1 ATB conclusions on unicast

Based on the studies performed in the frame of the ESA ATB study, a solution for packet access in SW-CDMA can be proposed. Particular emphasis was given to a GEO scenario. However the solutions proposed were also suitable for other constellation scenarios, like LEO.

For the unicast case in particular, open loop power control strategies seems best suitable for both up-and down-link accesses. Hence, no up and down-link DCHs are strictly required to be associated with MTs operating in packet mode.

Based on such finding, some modification on the 3GPP down-link packet access based on the DSCH have been proposed. In particular, a Shared Control CHannel (SCCH) was introduced for supporting the signalling needed for the DSCH operation. This solution although not compatible with a closed loop power control strategy, as that used for dedicated channels, appears the best compromise between access complexity and performances. The resulting DSCH access still support advanced features of CDMA like satellite diversity in both the conventional maximal ratio combining form and in the Fast Satellite Selection (FSS) operating mode.

As far as up-link is concerned two efficient solutions have been analysed:

- spread Aloha;
- dRoD.

Both slotted and non-slotted spread-Aloha are good alternatives for packet access if power control is sufficiently good. Anyway power control is critical for all kind of reverse link access based on asynchronous CDMA technique. It appears that the slotted variant gives no significant advantage in terms of throughput at least when there is no strict timing synchronization already available by external means.

Even a loose synchronization in slotted spread Aloha may bring a reduction of the miss-detection probability for a given false alarm probability and hence lower criticality of the acquisition process. In fact, in case of slotted spread-Aloha preamble detection has to be checked only in a window equal to the time uncertainty around the nominal position of the slot boundaries and not full time. On the converse, however, slotted spread Aloha requires a larger number of codes to make code collision negligible with respect to the unslotted variant.

A compromise in this respect between slotted and non slotted spread-Aloha may be represented by a solution in which the possible starting time of RACH bursts is quantized. However, the time period between two successive access slot boundaries is much less than the RACH burst length. This may help a little in detection probability without the need to increase significantly the codebook size. For example we may have nominal access slot size equal to a time slot (2 560 chips) with the RACH burst length which may instead span multiple frames (e.g. 10 ms, 20 ms, 40 ms, 80 ms. plus preamble).

Efficiency of spread Aloha packet access would also decrease with a decreasing of the Processing Gain (P_G). At very high bit rate, in fact, the statistical advantage of spread-Aloha decreases. Hence it appears preferable to merge such access with a reservation system for the possible transfer of higher traffic volume at higher bit rate.

dRoD appears more suitable for heavy loaded system due to the intrinsic stability of such access scheme. An hybrid, adaptive scheme in which spread Aloha access is also allowed is anyway attractive because of the potential for lower packet latency and was thus recommended. This last solution is thus the one recommended as the basis for up-link packet access in the SW-CDMA system.

The study performed, due to the limited funding and time budget (phase I was scheduled to run 6 months only), could not assess all the possible access variants which could have positive impact on performances. In particular, several CPCH variants could be thought which however were not addressed due to the limited resources. Further, the possibility to exploit multiple satellite diversity for dRoD transmission was not addressed, although this is a promising feature at least in non GEO scenarios. In a GEO scenario, where multiple satellite diversity could likely be not available, a solution which could be explored to improve reverse link performances is that of a synchronous (or quasi-synchronous) approach to reduce self-noise interference. A synchronous option was preliminary investigated in the ATB study based on analogous solution proposed in UTRA WCDMA. An extension of such technique to dRoD, if feasible, could greatly increase the access effectiveness.

5.2 ATB conclusions on multicast/narrowcast

Multicast, broadcast and narrowcast are all applications which fit well for satellite applications due to the inherent one-to-many structure of satellite communication systems. This paper therefore has investigated the application of these techniques to satellite systems.

The goal of the analysis was to provide dimensioning information for the techniques to be implemented in the second phase of the ATB project. This clause investigated both:

- large block interleaving and RS coding;
- medium block interleaver with CRC;
- hybrid short Carousel and FEC with interleaving; and
- narrowcasting.

The **Large block interleaving** showed that to achieve an codeword error rate of less than 10^{-6} an interleaver of 10^5 codewords and a RS(255,175) code is needed to cover all elevation angles. If the requirements on the minimum elevation angle (dependent on position), the maximum error rate (dependent on the application), or the vehicle speed, were raised then the interleaver size could be reduced. The memory needed for such an interleaver is approximately 50 Mbytes (two required).

The **medium block interleaver with CRC** was also thoroughly investigated. It was found that even for a vehicle travelling at 120 km/h the average outage is in the order of one second. The histogram of the outages does show that there are more small outages, however, compared to the errors caused by the longer outages, this is insignificant. This result was confirmed through simulation results. Applying medium block interleaver technique the improvement in codeword errors is insignificant at high codeword error probabilities but provides a gain (for a constant codeword probability error) of 5° to 10° in elevation angle at higher angles.

The technique was seen as not being useful as it does not consider the worst outage conditions. The design of an outage correcting solution has to consider worst case outages, not just average outage size. For this reason this technique is not useful as the more powerful FEC with large block interleaving will still be needed to achieve reasonable performance.

An investigation was performed into a **combined short Carousel and FEC scheme for multicast**. The idea here was to see if implementing a fast carousel scheme together with the FEC scheme could reduce the interleaver size but maintain the performance and to some degree the throughput. Poor performance results were obtained which showed that carousel techniques bring little performance improvement over satellite elevation, as well as that coding implemented within carousel blocks brings little gain. The coding rate of such schemes is also a major consideration which affects performance, with rates of 0,343.

The Hybrid solution (Small block Carousel with FEC) does not seem promising due to the fact that there is not enough diversity such that the decoding can correct errors when they occur. A carousel system would have to repeat the data more often and have larger block sizes, however the delay of the system would increase substantially if this was to be performed. Carousel implemented at the application level (with large delays of up to hours) is a completely different implementation than the one tested here. This method of data update is more than likely very useful as long as the application can cope with the delay and potentially missing data.

Finally **narrowcast** with ARQ was investigated through simulation. Results were determined for different code rates, data rates, and vehicle speeds. Performance was plotted in terms of overall delay needed to make sure a given packet was decoded reliably. Other plots showing the repeat requests and the NACQ messages were also plotted. From the information gathered a final parameter selection could be determined.

These experiments were carried out for two distinct mobility models, namely pedestrian and vehicular. These experiments have confirmed the close dependencies between the data transmission rate, RS coding and the number of ARQs. These dependencies are highlighted through multiple simulation runs, and are qualitatively reported in Table 2.

Table 2: Qualitative conclusions on narrowcast simulations

Parameter	ARQ	Re-transmission
Transmission data rate (increase)	Increase	Increase
RS coding rate (decrease)	Decrease	Decrease

The impacts of the above dependencies result in the bandwidth usage and its efficiency on the both up and down links. Obviously, the optimum bandwidth utilization will depend on the applications and the sender of the narrowcast for both its date rate and the RS coding. The sender may dynamically adjust these parameters through the narrowcast's QoS descriptors by reflecting the dynamic network metric such as packet loss rate, and so on.

For the purpose of the SATB field trials (one GW and 2 MT) the setting as given below could be used with following justification. Figure 80 and Figure 86 show that a minimized total delay is obtained with RS(255,123). From Figure 81, Figure 82, Figure 87, and Figure 88 it can be derived that a good efficiency for the BW usage on FL and RL is obtained at user data rate of 64 kbits/s. The final recommended RS coder coding rate is RS(255,123) as given in Table 3.

Table 3: Recommended RS Coder Coding rate

Parameter	Value
RS coder	(255,123)

5.3 SATIN project conclusions

The SATIN project defines a new integrated role for satellite systems to deliver multicast/broadcast services to users. This makes use of the best attributes of the satellite systems and relives the terrestrial cellular systems from one of their biggest drawbacks - it is thus a win-win situation. In addition it makes much more efficient use of the total spectrum allocated to satellite and cellular usage. The business study defines the classifications of services appropriate to the satellite and terrestrial segments of the integrated system with MBMS being the set most appropriate for satellite delivery. An approximate dimensioning and business analysis was performed with user profiles for "direct users", "roamer" and "MBMS users". This showed that taking direct users alone was not a viable business case for satellite. The satellite system only becomes viable if the MSMS users are included. An additional comparative study between MBMS delivery via terrestrial and satellite was performed and it was shown for the limited scenario of terrestrial operation considered that the satellite delivery was economically more viable provided the MBMS additional ARPU was below about 6 €/user.

An integrated satellite/terrestrial system architecture for the delivery of broadcast/multicast service delivery via satellite was defined based on UMTS standards also introducing a new terrestrial elements called intermediate module repeater (IMR). Two architectural options were considered:

- The baseline, where the forward link is via satellite and return link via the terrestrial networks.
- The optional, where both forward and return links are via satellite but with the return link being low data rate (3 kb/s to 8 kb/s) and for signalling purpose only.

System components (satellite payload, terminal, gateway, IMR) were selected based on fundamental engineering trade-off exercises.

- GEO constellation was selected based on initial implementation cost and compared to non-GEO cost.
- Transparent digital processing payload with multiple beams was selected considering flexibility, connectivity and currently available payload technology.
- Satellite receive only terminal was considered for baseline case.
- Node B and RNC functionalities are included in gateway.

The requirements for a broadcast/multicast efficient S-UMTS access scheme stemming from the service and system architecture selection were identified:

- No connection over the air.
- No PDP messaging between UE and Satellite-RNC (GW).
- No feedback/loop mechanisms (power control, ARQ, etc.).
- Use of common channel to configure lower layers based on unidirectional signalling from S-RNC to UEs.
- Compliance with current core requirements of 3GPP MBMS.

The required modifications needed physical layer and the radio interface sub-layers (MAC, RLC, BMC and RRC) were detailed:

- FACH mapped to the S-CCPCH Physical Channel (with Spreading Factor up to 8) was selected for forward link data transfer following comparison with DSCH. RACH selected for return link data transfer for optional case following comparison with DCH.
- Enhanced transport dynamics of cell broadcasting and adapted Broadcast Multicast Control layer (new messages and primitives).
- Service notification and radio resource set-up for a set of common traffic channels building upon Cell Broadcast mechanisms.
- Variant of Radio Resource Control states for the terminal in the preferred case of *baseline configuration with parallel receiver architecture* (active/non-active modes in terms of level of connectivity).
- Introduction of high order modulation and advanced turbo/layered coding schemes. W-CDMA 3-step cell search procedure (preferable to SW-CDMA alternative scheme).
- Fixed Radio Bearer configuration (Mode A) based on some prediction of the traffic over some interval of time and dynamic Radio Bearer configuration (Mode B) with full involvement of AC have been considered for RRM in SATIN. Mode B allows higher flexibility in the resource utilization at the expense of extra interlayer- and over the air- signalling (reconfiguration messages towards the group users). Adaptations of the Multilevel priority Queuing and Weighted Fair Queuing disciplines were considered for PS.

Based on the above access scheme proposed adaptation, simulation scenarios were defined and the individual functionalities were investigated.

- The main difference is propagation channel induced by IMRs with the large delay spread.
- Two alternative layered coding structures (Structure A-8PSK and Structure B-QPSK) were investigated. It was demonstrated for both layered coding structures that as the channel became worse, the gap between the SCCC and CC code performance increased. It is shown that the single finger receiver could not cope with the multipath interference in both low and high speeds and the diversity introduced by the RAKE have to be exploited to improve the receiver performance for all the speeds. The diversity gain seemed to be independent of the speed, however the performance gain seemed to be depending on speed for moderate to high E_b/N_o values. Structure B always outperformed Structure A but the spectral efficiency is lower (half of that of Structure A).
- 8PSK and 16QAM were considered as candidates for higher order modulation schemes to increase the spectral efficiency and it was shown that approximately up to 2 fold capacity improvement (with 16 QAM) can be obtained in ideal channel conditions. However the higher order modulation schemes are more sensitive to non-linear amplifier effects in the payload. Predistortion techniques can be usefully applied in conjunction with high order modulation to reduce HPA impact. The impact of HPA non linearities in the SATIN system were investigated both for a turbo coding scheme over AWGN channel and for a convolutional coding scheme over Rice fading channel to see the system performance for different values of the IBO. For IBO = 5 dB and IBO = 3 dB. For IBO = 1 dB, results of FACH in AWGN channel pointed out that the impairment introduced can vary from about 0,8 dB to 1,2 dB for a system loading factor within [0,25,1] in the most critical case of IBO = 1 dB.
- The three-step cell search (T-UMTS approach) and the two-step beam search (S-UMTS approach) were evaluated on the different SATIN scenario and compared in terms of performance and implementation complexity. The activity led to the choice of the three-step procedure for reasons of efficiency, complexity, and harmonization with the 3GPP procedure. Notably, the ETSI SES S-UMTS group has also adopted this approach.

- Standard SW-CDMA RACH procedure has been investigated and a novel multi-dwell scheme proposed in order to obtain good performance and reduced system complexity. The Differential Post Detection Integrator scheme (sub-preamble words acquisition) was selected as the most suitable scheme for satellite environment. The detection probability of the preamble does not depend on the length of the signature but on the length of the preamble. With more signatures available, there is a reduced risk of collision and hence better system performance can be achieved (for cases with a tight network-wide synchronization). Nevertheless, the drawbacks of having longer signatures are increased complexity in the receivers as more matched filters are needed, and also a higher value of spreading factor needs to be employed.

- As regards performance of fairness between flows of same QoS class or physical channel utilization it makes little difference for each PS scheme (MLPQ and WFQ) whether the mapping is power-aware or not, the exception being apparently the provision of fairness that seems to be harder in the pure bin-packing case with MLPQ PS. It remains that fairness achieved with the enhanced MLPQ is satisfactory, hence the implementation of WFQ between flux of the same QoS is not deemed necessary. The nature of WFQ is curbed by TFCS selection process and the dynamic range of the flux in many multiplexing cases has not allowed the significant reduction of TF range which may be foreseen with WFQ, compared to MLPQ. The latter also obtains better score as regards the utilization and E_b/N_o requirements. Due to the amendment performed on original basic mapping approaches, the difference of potential for improvement on E_b/N_o requirements as regards the mapping alternative is not significant. Finally, whatever the PS scheme the optimization of the system capacity/throughput depends on the cautious trade-off between power saving and multiplexing effectiveness (i.e. TBS size thresholds that preserves high potential utilization of the physical channel).

- In the Fixed mapping approach the capacity is limited due to transport resources (FACHs) unavailability, while in the Ad-Hoc Mapping the capacity is determined by power-load limitations. Individual blocking probabilities due to Power-QoS related constraints depends on the QoS class. All services, except for the highest QoS class (Video streaming 256 kbps), benefit from the power saving technique that is used in the "power aware" mapping. The overall blocking probability for the Ad-Hoc mapping is smaller compared to the fixed mapping. Nevertheless, this cannot be directly translated to a higher capacity (in terms of throughput) due to the relative high blocking probability experienced by Video Streaming 256 kbps. The performance of the network is strongly sensitive to parameters like satellite antenna leakage, orthogonality and Admission Control threshold. Increasing the antenna leakage the overall blocking probability is also increased, while higher values for Admission Control threshold result into lower blocking probabilities. The impact of orthogonality is opposite to the impact of antenna leakage. The low E_b/N_o values considered for the AWGN channel result in a very low multiple access interference, hence there is low blocking probability due to power constraints. On the contrary the multipath IMR channel requires higher values for target E_b/N_o, which in turn results in higher interference and blocking probabilities. When the Ad-Hoc mapping is used and interference has very low effect on the network's performance (e.g. AWGN channel) the system achieves the best utilization of FACHs.

- The capacity analysis study demonstrates the feasibility of the satellite multicast-broadcast overlay network approach proposed and developed within SATIN, paving the way for its future development. The studies performed show in fact that the link budget can be closed with both the direct and the indirect cases considering aggregated bit rates up to 1 680 kbps.

Thus technically the system has been verified by simulations and will now be demonstrated in the follow on MoDiS project. The new architecture and S/T-UMTS system has been introduced into standardization process via ETSI-SES where technical documents have been accepted from SATIN. Additional documents will be submitted via MoDiS in order to pursue the full standardization of the system.

5.4 3GNetSim

As the 3GnetSim project was in the implementation phase when its contributions were provided to the Working Group, no conclusions about the performance of the envisaged concepts were possible at this stage of the project. It is expected that quantitative results from simulator optimization runs will be available in April 2004.

5.5 Conclusions from the GAUSS project

The GAUSS system design has an open and flexible architecture, based on the adoption of current standards and consolidated technologies. This represents one of the challenging aspects of the GAUSS system: flexibility and conformity to widely adopted standards, while reducing design and development risks and costs, guarantee compatibility with current and future NAV and COM systems, and capability to be fully integrated with available infrastructure for providing complementary added-value services.

Studies and simulations were done, to find a suitable RLC configuration for avoiding the new channel affect the RL performances, in terms of packet handling and useless retransmission avoiding mechanisms. A lot of effort was spent to design and implement the access scheme for the return link described above in order to provide a realistic real-time behaviour of the physical layer for the GAUSS services. Therefore maintaining an absolute time reference in the physical layer and the synchronization of the MAC layer to the physical layer is mandatory. This is a quite difficult task regarding the serial interface between the physical and MAC layer and the non-real-time operating systems involved. The primitives for interface between the physical and MAC layer considered in the GAUSS Demonstrator is compliant to the S/T-UMTS standard.

GAUSS successfully demonstrated integrated GNSS1 precise positioning based on EGNOS and satellite UMTS packet communication. The new technology with respect to the current state-of the art, developed within the project, was validated during the trial campaign, including the implemented broadcasting and multicasting communication of data packet compliant to 3GPP standard. Furthermore, GAUSS message flexible fixed-format is able to support a wider class of Location-based services, requiring small data packet exchange, as no modification is required to the GAUSS standard packet to accommodate the data required by other specific applications. This guarantees very promising future exploitations and capability to provide other complementary value-added services.

6 Conclusion and further work

The present document gathers the results presented in five technical contributions to the ETSI SES-UMTS WG. The reasonable step forward at this point will be to try to converge in a set of technical recommendations, that will eventually lead to a set of technical specifications for the packet access mode in the satellite environment.

The present document is mainly based on layers one and two, but some technical contributions in the upper layers will be needed, in particular, 3GnetSim will investigate upper layer issues.

Due to the different time schedules of the research projects, which contributed and will contribute to the present document, it became necessary to preliminarily finalize the present document in December 2003, though not all projects have finished and could provide the whole set results of their trials. In order to be able to consider especially the results of 3GNetSim in the technical specifications to be followed by the present document, it is recommended, to revise the present document when these results are available.

Annex A: SW-CDMA packet access

A.1 Forward link

The investigations included in this annex are done with enhanced features compared to the specifications in [31], [32], [33] and [34].

According to the UTRAN approach, the forward link packet access for S-UMTS will be mainly based on the DSCH (Down-link Shared CHannel). Clearly the FACH and DCH are also available for packet access, but the DSCH is considered the most efficient and flexible mechanism for supporting such kind of access although the FACH channel is also essential for transition of the MT to the connected mode.

Prior of the introduction of the DSCH mechanism, packet support in UTRA down-link was through the use of FACH for short and /or infrequent packets and through the fast set-up of a DCH for long or frequent packets. It was shortly recognized in 3GPP that the use of DCHs for packet has several shortcomings. In particular, it is very difficult to estimate the user resource requirements, as these can be continuously changing. Negotiating an a-priory data rate is thus not a smart choice. Furthermore, efficient interference management is made more difficult if a-priory data rate assignment is made. In fact, if a user does not use the assigned resources, inefficiency results and lower statistical multiplexing gain is achieved. Finally, to reduce packet delay, it is always preferable to assign the maximum available data rate to a single user at a time (obviously when the user packet queue is not empty) rather than subdivide the available capacity between multiple users. In the former case, having assigned the fattest possible data pipe to a single user when there is a clear link, the overall delay experienced by the user is in fact minimized.

Furthermore, due to the use of orthogonal channelization codes within a cell (or sector), the number of available codes is limited. In that case, if a dedicated channel were assigned to each user having active packet session open, the channelization codes would be rapidly consumed. This would typically happen before saturation of channel capacity. Packet sessions could have in fact a very low activity factor. The activity factor is moreover dependent on the channel data rate and is typically lower for higher data rates.

In R99 UTRA specs, each user of the DSCH also needs a down-link DCH carrying the physical control channel, including the signalling for implementing power control support, and the higher layer signalling for resource allocation. Further, the DSCH operation is typically associated also with an up-link DCH. The up-link DPCCH is in fact required for the down-link power control feedback signalling.

The use of an associated DCH has the drawbacks that additional power and code resources are required by the DCHs. These drawbacks are mitigated by the fact that the DCH SF is very high (256) and that in case no signalling data have to be transmitted on the DCH only the control part (the DPCCH) is actually transmitted. Furthermore, suitable timer will transfer the user to the CELL_PCH (see note) state in case of prolonged pause in the usage of the DSCH.

NOTE: CELL_PCH is one of the possible state defined by 3GPP for the UE. For a review of such states see doc. 3GPP TS25.331.

For satellite application anyway, fast closed-loop power control is not possible and is also not as needed as in terrestrial application due to the much more benign propagation environment. This fact, makes the need of the associated DCH less obvious, taking into considerations its drawbacks which, moreover, are likely more important in a satellite environment due to the larger satellite delay. Such delay would, in fact, suggest the use of less aggressive time-out for switching down the user to the CELL_PCH state. In fact, state transitions (from/to CELL_DCH to/from CELL_PCH) would require more time in satellite than cellular systems and this could impact on the perceived packet latency.

An adaptation of DSCH not relying on an associated DCH was hence studied in the ATB program. The resulting access and simulation results will be shown in the present document. Before that, however, the role of satellite diversity and a possible strategy of power control not relying on an associated DCH channel will be investigated.

A.1.1 Potential diversity advantages

One of the most controversial issues in a CDMA based system is whether exploitation of satellite diversity on the forward link is advantageous. This because diversity exploitation on the forward link requires allocating RF power resources on multiple satellites for transmission of a single channel.

Furthermore, instead of conventional maximal ratio combining diversity, different diversity strategy may be also conceived like an FCS-like diversity mechanism. FCS (Fast Cell Selection) is a mechanism being investigated in 3GPP for selecting the best Node B to be used for transmission in place of conventional maximal ratio combining diversity.

FCS is a mechanism similar to that already specified in R99 for supporting SSDT (Site Selection Diversity Transmission) in DCH transmission. SSDT is an optional advanced form of FL power control used in a macro-diversity environment.

According to SSDT a UE select each frame or multiple times (up to 5) per frames one of the BS in the current active set as the *best* serving BS. SSDT signalling of the *best* serving BS is typically done via the FBI field (in particular the S sub-field of FBI, the D sub-field being used for transmitting weight values for the BS Tx antennas in case antenna diversity is employed).

The selected BS will transmit both the DPCCH and the DPDCH towards the UE, whilst all other BS in the active set only transmits the DPCCH. Selection of the *best* BS in the active set is done through measurement of the RSCP (Received Signal Code Power) of the CPICH (Common Pilot Channel). Similarly to SSDT, with FCS only the *best* BS transmits the DSCH.

Extension of FCS (or even SSDT) to S-UMTS is questionable due to the satellite delay which may render the selection performed by the MT obsolete when received from the GW; it may only be useful to counteract shadowing without operating in multiple satellite diversity mode. Also no FBI field is presently included in the S-UMTS physical layer. Support of this feature would then require the addition of a new signalling field in the up-link DCH. Finally, multiple satellite visibility (and thus FCS) could not be available at all with satellite constellations like the GEO one, which might only be based on single satellite.

A modified FCS scheme can be however useful in S-UMTS. In this modified scheme, no ad-hoc signalling of the best path is sent by the MT, but, taking into account that a single GW is generally serving the MT, such a GW can decide, within the potentially active satellite set, packet-by-packet, which satellite to use for each single packet transmission based either on the last measurement report or the GW own measurements of the MT transmissions as received through the different available satellite paths (see note). Clearly such an approach is not as fast as the use of FBI signalling managed at the physical layer. Moreover, number of bits per measurement reports are higher than required with the FCS mechanism. However, this may be acceptable taking into account that here we are not counteracting multipath fading (which is not possible for loop delay reasons) but only the slower shadowing. Hence, it appears anyway wise to introduce a feature of Fast Satellite Selection (FSS) within the *active set* of paths of a given MT.

NOTE: The measurement reports actually reports MT measurements averaged over periods of time which are typically longer than the multipath fading correlation period. Hence they are not affected by multipath fading (apart perhaps the very slow one). On the other hand, measurements performed on the received MT transmission may be related to a short packet transmission (like RACH access). In this case, being the measurement affected by multipath fading, the GW should not rely only on the last measurement for deciding on the best path.

The active set of paths would be defined according to traditional procedures based on measurement reports from MT and information about the constellation geometry and user localization. Within such active set, the path (or paths) to be really activated in each transmission could be faster signalled before each actual transmission to the MT.

The above mechanism does not cope with mitigation of multipath effects. For this scope conventional diversity can be very useful as demonstrated in Annex . Hence the two mechanisms are not necessarily alternative and may coexist particularly when a LEO constellation is envisaged. Both options should then be supported for DSCH operation in S-UMTS.

A.1.2 Power control issue

Closed-loop power control in a satellite environment is not as effective as in terrestrial case due to the high round-trip delay. This is still more true in a GEO constellation where:

Transmission delay is almost the same for all terminals, notwithstanding they may be spread over a large area (e.g. a cell of 1 000 km). Hence, the near-far problem experienced in terrestrial systems will not have be as significant in satellite systems, particularly in GEO ones.

Channel fading is relatively mild. Due to the link power margin, which is generally quite limited, the communication is normally possible only if a direct line-of-sight path exists which make the fading Ricean in nature.

These two characteristics makes the gain achievable by power control less relevant in satellite systems with respect to the potential gain which could be provided instead in the cellular environment. Also only low signal dynamics can be tracked. This contributes to make an open loop control scheme, managed entirely at the RRC level, attractive taking into account the advantage that such a scheme does not require that each DSCH packet user also transmits an associated DCH carrier (on both the up and down link).

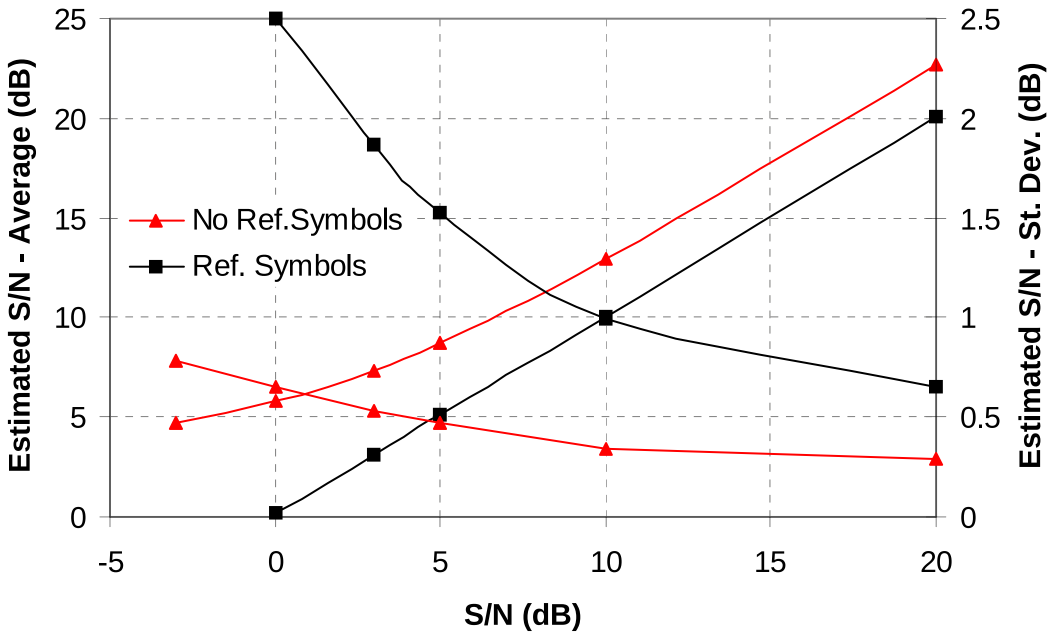
Furthermore, as far as the down-link is concerned, the synchronous nature of the CDMA access makes imperfect power control not as important as in the reverse link where interference is generally the most important factor as far as capacity limitation is concerned.

An efficient open loop power control mechanism for the DSCH carrier could be achieved based on the measurement reports from MT about the cell pilot channel (CPICH) SNIR. This is the same measurement also used for handover management (see note).

NOTE: SNIR measurement report is not foreseen by UTRA specs. Instead RSSI and RSCP measurements are foreseen. From these measurements the SNIR can be easily derived. However, for open-loop power control purposes, in order to reduce the signalling load, the direct reporting of SNIR, quantized with a small number of bits (e.g. 4) could be a better solution.

For open loop power control a good estimate of the down-link pilot channel SNIR is required. Figure 3 shows simulation results about two SNIR estimators respectively based on reference symbols (as those of CPICH) and modulated one (as in traffic carriers). It appears that SNIR estimation based on modulated symbols produces a biased estimation of the SNIR; hence it is not suitable for open loop power control. With reference symbols based estimators, instead, no bias results. From Figure 3, it appears that standard deviation of the estimator at 0 dB S/N ratio is about 2,5 dB. That result assumes that the SNIR is evaluated over 16 reference symbols. For longer estimation interval better performances are expected. For open loop power control on geostationary links, estimation interval in the order of about half a second (or longer) could be adopted. In such a case the SNIR standard deviation could be decreased, in principle, to less than 0,2 dB. We may thus assume that SNIR measurement on down-link pilots is very precise.

Further consideration on power control mechanization are deferred in next clauses.



NOTE: Number of symbols/frame is 60. One reference symbol out of 10 symbols is assumed for the case of estimator based on reference symbols.

Figure 3: S-curve of SNIR estimators: SNIR estimator based on reference symbols and SNIR estimator based on x4 phase non-linearity (QPSK modulation)

A.1.3 Adaptation of 3GPP DSCH to SW-CDMA

A.1.3.1 Trade-offs

From the physical layer perspective, the most important optimization to consider for the DSCH in S-UMTS is the possible decoupling of such channel from the associated DCH to avoid the overhead (both in power and in code usage) in maintaining such DCH. This has impact on power control issues and has already been discussed in the previous clause. A solution not relying on the use of DCH in conjunction with the DSCH was then selected.

Alternative approaches were also investigated which foreseen the use of a SICH (Signalling Indication CHannel) in place of the associated DCH. In this case, each user having an active packet session will be assigned resources on a SICH channel for both closed-loop power control and signalling indication purposes. Multiple SICH channels will be needed but the number of total SICHs will be anyway less than that of the associated DCHs which would otherwise be required. Different organization of the SICH could be envisaged but in each case a Shared Control CHannel (SCCH) for actual DSCH signalling is still needed as in the option with open-loop power control.

The SICH approach is such case would then provided essentially two advantages:

It would allow a closed loop power control strategy by guaranteeing a periodic transmission towards the user independently of the availability of data packets waiting for transmission.

It would allow some power saving by avoiding the requirement that users continuously monitor the SCCH, but just the SICH and only in the slots preassigned to them.

Some further discussion on SICH structure trade-offs can be found in clause 2. Anyway, this approach, was not considered further for implementation within the ATB test-bed, because it still requires additional code usage whilst providing only marginal advantages as far as power control and terminal power consumption. The decision may be reconsidered for standardization purposes.

It shall be observed that in absence of SICH, no maximal ratio combining of the SCCH is possible. Diversity operation with maximal ratio combining of the DSCH is instead still possible (see note 1) even though it make scheduling somewhat more difficult due to the constraint that the same data packet shall be simultaneously (see note 2) transmitted with the same characteristics (in terms of spreading factor) on multiple satellite paths.

NOTE 1: This because code multiplexing on the DSCH is supported. A given user can be thus allocated on the different satellite paths separate codes (with respect to other users) each one with the same SF.

NOTE 2: Actually almost simultaneously (a small Tx time offset may be present between different beams of the same satellite. Moreover transmission from the same GW to different satellite may be offset to partially compensate the path delay).

FSS strategy is also well feasible. As usual, in such a case, the network will maintain an *Active Set* list (to be signalled to the user) and select the *best path(s)* within such list which will be used for the DSCH transmission. Although not strictly required, we assume that the SCCH is transmitted on all paths where the DSCH will be also transmitted for signalling reliability purposes. Because, such paths are not known a priory (apart that the paths shall belong to the current *Active Set*), the user shall monitor the SCCH on all the satellite paths in its *Active Set*. The decoding of a single replica of the SCCH is anyway sufficient for recovering the information about what paths will be actually used for DSCH transmission. This information is clearly only needed if the FSS operating mode is enabled.

A.1.3.2 DSCH Access proposals

The following modifications to 3GPP DSCH are proposed:

Open Loop Power Control. Being no closed loop power control required then no associated Dedicated channels are required.

A Shared Control CHannel (SCCH) is suggested for supporting the DSCH related signalling.

From the physical layer point of view the SCCH will be similar to the FACH. I.e. frame structure and spreading strategies are the same as FACH. No time multiplexed control channel is however used being the transmission format fixed (and signalled on the BCH channel). Moreover, as far as coding is concerned, some modifications are suggested with respect to FACH which is inspired to work currently undergoing in 3GPP as far as the standardization of HSDPA is concerned: For HSDPA in fact, a set of shared control channels (comprising from one to 4 carriers) is also introduced for each HS-DSCH (High Speed DSCH). Each frame of this signalling channel is subdivided in groups of three slots where each group of slots support the signalling towards a single user. This group of slots is actually separately coded (including its associated CRC) (see note 1).

NOTE 1: Actually, in some proposal currently being investigated, each three slot group may be further subdivided in two parts as far as CRC and coding is concerned for faster retrieval of the signalling information.

Each slot group contains the following bits:

- 7 bits for the channelization code;
- 1 bit for the modulation format;
- 6 bits for the transport format identification;
- 3 bits for the HARQ process number (for supporting parallel Stop And Wait protocol processes);
- 2 bits for supporting the HARQ version number;
- 1 bit for indicating the new data indicator.

A 16 bit CRC or 2 CRC with 12 bits and 8 bits respectively (see note 1).

In our case, we do not need to support HARQ nor Adaptive modulation, hence only the 7 bits for the channelization code, the 6 bits for the transport format identification and a CRC would need to be supported. In addition a terminal identifier is also required. This information is not required in HSDPA because a signalling indicator (HI) is transmitted on a DCH for signalling the corresponding MT that the current group of slots on the shared control channel is for him.

However, if no DCH is used either an additional shared channel is used or a mobile terminal ID has to be also inserted in the SCCH. A short temporary identifier of 8 bit could be sufficient for that scope also taking into account that a given user may be assigned a given slot group number (modulo a TTI) and a given SCCH carrier (if multiple carriers are available).

The number of bits would then become 21 plus the CRC. With an 8 bit CRC, a total of 30 bits would results which also includes a spare bit (see note 2).

NOTE 2: If the FSS option is enabled, some additional bits could be required to signal the satellite paths which will be used. This option was not considered here.

The SCCH data rate would then be 30 bit/3 slots, i.e. 15 kbit/s. 3GPP standard rate 1/2 convolutional code would be used with independent code for each group of three slots (this implies that 8 tail bits has to be added to each signalling packet but puncturing can be used to maintain the code rate at 1/2).

Regarding the DSCH itself, we will assume a semi-static TTI which can be comprised from 1 frame to 8 frames as for DCH (the used TTI will be signalled on the BCH). Assignment on the DSCH shall be signalled in one of the time slot groups belonging to the TTI preceding the one where DSCH transmission will take place. Use of DSCH TTI shorter than one frame, can in principle be also supported (up to 3 slots), although the resulting packet latency reduction is probably not significant with respect to the satellite delay.

Diversity

As already mentioned maximal ratio combining will be exploited for DSCH. Variations in the active set handling have to be explicitly signalled. This could be done either via FACH or through the DSCH directly. The last solution has the advantage that the FACH is not required to be monitored during an active packet session and is proposed here. Handoff signalling is in that case transmitted in-band on the DSCH channel itself avoiding the need for the user to continuously monitor the FACH during the packet session.

Open Loop Power Control

The power control will be based on the MT measurements of the CPICH SNIR. This measurements will be reported back periodically to the GW during an active packet session. Anyway each time a packet is transmitted by the MT during the packet session (e.g. an ACK), a SNIR report is also appended. When no packet has been transmitted by the MT for a certain period, a time-out will expire which trigger a report transmission. For example in a GEO constellation a time-out reporting period of 1s could be selected. The GW will adjust its transmission power according to the reported SNIR and the target SNIR. The target SNIR can be adjusted to achieve the desired FER with a strategy similar to the outer loop of the closed loop power control. In particular, the target SNIR will be increased by the amount ε for each unsuccessfully packet transmission (no ACK received (see note 3)) or decreased by εP (P being the target frame error rate) for each successful packet (see clause A.1.3.3 for further details).

NOTE 3: A negative ACK event could also be foreseen when an allocation signalling has been correctly received on the SCCH but no subsequent DSCH packet has been successfully decoded. However, this negative ACK will not be considered being their transmission overhead not justified by the actual advantages produced.

A.1.3.3 System simulations

A simulation aimed at verifying the performances of the DSCH for packet access was performed. A single beam geostationary satellite has been considered. The beam has 1° beam width @ 2 dB with respect to centre of beam and boresight pointing @ 45° latitude on the same longitude as the sub-satellite.

It was assumed that only a portion of the beam power is dedicated to the DSCH channel. The remaining part was instead used for circuit services (or services anyway carried on physical channels other than the DSCH) and has not been explicitly simulated here.

If not mentioned otherwise in all the subsequent simulations we assume that the quota of beam power dedicated to DSCH is 50 % of the maximum beam power. Interference from adjacent beam is not dynamically accounted. However, a simple model has been assumed for such interference evaluation. According to such model the asynchronous interference power measured by an user could be at a level variable between -9 dB to +3 dB with respect to the power with which the same user can receive the DSCH assuming that such DSCH is transmitted at its maximum power level. The exact value would depend from the user position.

The maximum DSCH information data rate in all simulations which will be presented here was assumed equal to 512 kbit/s corresponding (after channel coding) to a SF of 4.

Different combination of data rates can be selected for the same TTI ranging from a single packet at the maximum data rate up to a number of N packets each with SF = N × 8. In our simulations the maximum SF was assumed equal to 128, corresponding to 16 kbit/s information rate. Clearly combination of packets having different SF can also be selected.

The ETSI web server traffic model as described in Annex but with a *Reading Time* of 10 sec was used.

Power Control

The open loop power control scheme has been implemented.

To summarize, the power control is based on the MT measurements of the CPICH SNIR which are reported to the GW either piggybacked with normal packets (or ACKs) or, if no packets has been transmitted for a while, standalone.

In particular for the simulations we have selected a time-out period for triggering the transmission of SNIR reports of 1 second. Clearly this time-out only applies to terminal having an open packet session. In real life the time-out period can be tied to the terminal operational state and even for MT involved in an active packet session such a time-out can be made depending on the real activity of the terminal. In other words if the MT is idle for a certain period, the reporting period can be increased to say 5 sec. If the idle period last even more than the packet session can be disconnected.

The GW will adjust its transmission power according to the reported SNIR and the target SNIR. A difference with respect to the up-link case (in addition to the fact that the CPICH SNIR is the preferred measurement here instead of the RSCP) is the fact that the measurements over which the GW will base its transmission power evaluation can be up to one second older (assuming a measurement report period of one second). Hence, the fading (even the shadowing induced one we are considering here) can have changed in the meantime. At this regard, it shall be mentioned that a correlation time constant of one second has been assumed for the channel model (see Annex) independently of the user speed (which however still influences the channel model state transition probabilities).

For each MT, the GW will store also a *margin* variable (equivalent to the one used in the reverse link simulations by each MT) which will be updated after every transmission.

In particular, if no ACK is received back by the GW after a transmission, the *margin* value will be increased by a value *up* (set equal to 1 dB in current simulations). If an ACK is instead received the value of *margin* can instead be decreased. In the simulations we have performed, we assumed that with the ACK the MT also attach (in addition to the CPICH measurement report) also the SNIR margin with which the packet was received. Based on such margin (referred here as *Delta*) the GW will update the *margin* variable of its open loop power control algorithm by reducing such *margin* of a quantity equal to *Delta/k* (operations are assumed in dB).

The parameter *k* and *up* can be tuned to get the desired packet error probability.

It shall be noted that the approach here considered is basically the same as the one illustrated for the up-link case. The values of *up* and *k* considered in the simulations which will be shown later on (if not otherwise mentioned) were 1 dB and 30 respectively.

Finally, as far as the bias in the power control a value of 2 dB for the error variance was assumed. It shall also be observed that the bias in the power setting of all the channels transmitted by the GW should be correlated. However we have considered the most conservative scenario in which the bias is totally independent for the various MTs.

Scheduler

The implemented scheduler, is very simple. It actually examines the various application queues (see note 1) and will select those from which data will be retrieved for transmission at the next TTI. Criteria for selecting the data are, in the order of importance:

- time of permanence in the queue;
- volume.

NOTE 1: We refer here as application queues the queue where data for a given MT are buffered before transmission.

We recall that to maximize the channel throughput, terminals having best propagation conditions have to be scheduled first. Hence request should also be sorted against the required power per bit to maximize throughput. However, such kind of scheduler is not fair as far as resource allocation is concerned. Hence we have preferred not to consider such a criteria apart for the check that if the conditions are so bad that the required power for the channel becomes excessive with respect to the available on-board power then the allocation is postponed.

The selected TTI for all the simulations was 10 ms. (one frame). If data remains in their queue for more than 4 sec they will be discarded.

Regarding the DSCH possible packet formats (i.e. the physical layer Protocol Data Unit length) which have been used in the simulations (taking into account the 10 ms. TTI) they are listed below:

- 320 bytes;
- 160 bytes;
- 80 bytes;
- 40 bytes;
- 20 bytes.

Simulation Results

Simulations were performed assuming a GEO spacecraft with the characteristics described before. As far as the maximum RF power allocated to the beam, we assumed 2 W (corresponding to an EIRP at beam centre of 45,7 dBW), of which up to 1 W can be allocated to the DSCH. Simulation results shown below actually suggest that a lower EIRP would have probably be sufficient. Efficiency of the DSCH is normalized to its maximum capacity, i.e. 512 kbit/s in the cases shown below.

Figure 4 and Figure 5 shows obtained results in absence of fading but with power control errors. It appears that the DSCH may actually be heavily loaded, at least 70 %, before delay becomes to increase rapidly. Also apparent is that the available RF power is not fully used even in high load conditions. At this regard, the distribution of DSCH used power is given in Figure 6 and Figure 7 for two different traffic loading conditions.

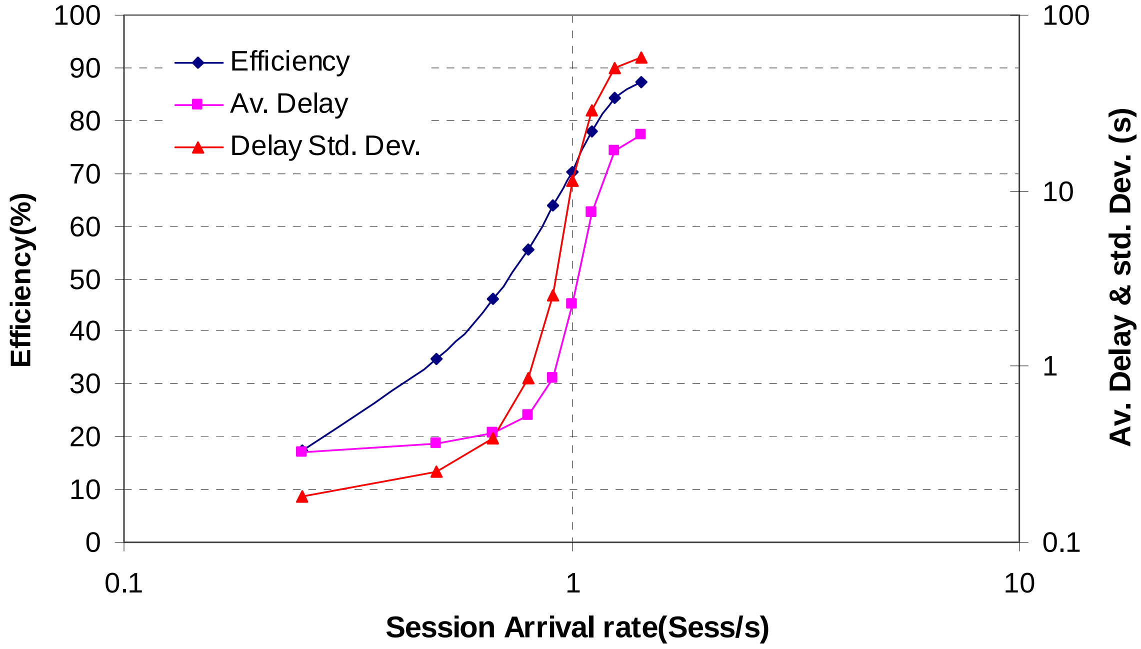
Another simulation result which should be noted is the fact that, notwithstanding that a *reading time* of only 10 sec was assumed in our simulations, the number of MTs having an active packet session is quite large. In fact, looking to Table 4, it appears that more than 300 MTs can simultaneously have a packet session open. This implies that, if a DCH would have been allocated to each MT as in 3GPP (see note 2), an excessive overhead would have resulted thus implicitly confirming the validity of our approach of renouncing to the closed loop power control.

NOTE 2: Actually in 3GPP the DCH is not maintained for the whole packet session duration, but the terminal will transition from the CELL_DCH state to the CELL_FACH state after a quite short time out (less than 1 sec). Further, transitions to CELL_PCH occurs after a time-out which is longer than the one for transition from CELL_DCH to CELL_FACH but anyway shorter than the average reading time in web browsing.

Figure 8 to Figure 10 show the same type of results as in the previous figures but will full simulation of the two-states channel model of Annex . It appears that the main differences with the case with no fading are in the increase of the delay (mean and standard deviation) under low load conditions. Under high load conditions the influence of fading is masked by the delay due to the congestion. The MT speed assumed was 3 m/s. To evaluate the impact of MT speed on performances the simulations have been repeated for two other different speed: vehicular user (70 km/h) and pedestrian user (3 Km/h) (see Figure 11 and Figure 12 respectively). In both cases the same target E_b/N_o has been maintained as before (i.e. 3 dB). This last assumption is not fully realistic in a fading environment. However, for the case considered where the satellite elevation is higher than 40°, the Ricean fading in the GOOD state is quite mildly. C/M ratio equal or in excess of 15 dB are in fact expected in a rural environment, making the differences between slow and fast fading not very significant. According to the physical layer simulations performed in the context of the ROBMOD project the difference between pedestrian and vehicular environment in Ricean fading with C / M = 15 dB is less than 1,5 dB assuming the use of a convolutional code and 40 ms. interleaving depth.

From the given figures it appears that the throughput achievable is not significantly depending on the MT speed. However the average packet delay is quite significantly impacted notwithstanding the fact that the scheduler is not privileging the MTs in good propagation conditions.

The same consideration about packet delay can be done also for the suburban environment results shown in Figure 14 and Figure 15 were some further worsening of the delay performances can be noted.



NOTE: SNIR estimation Error std. dev. at MT 0,5 dB; power control bias variance 2 dB, no fading.

Figure 4: Performance of DSCH with open loop power control

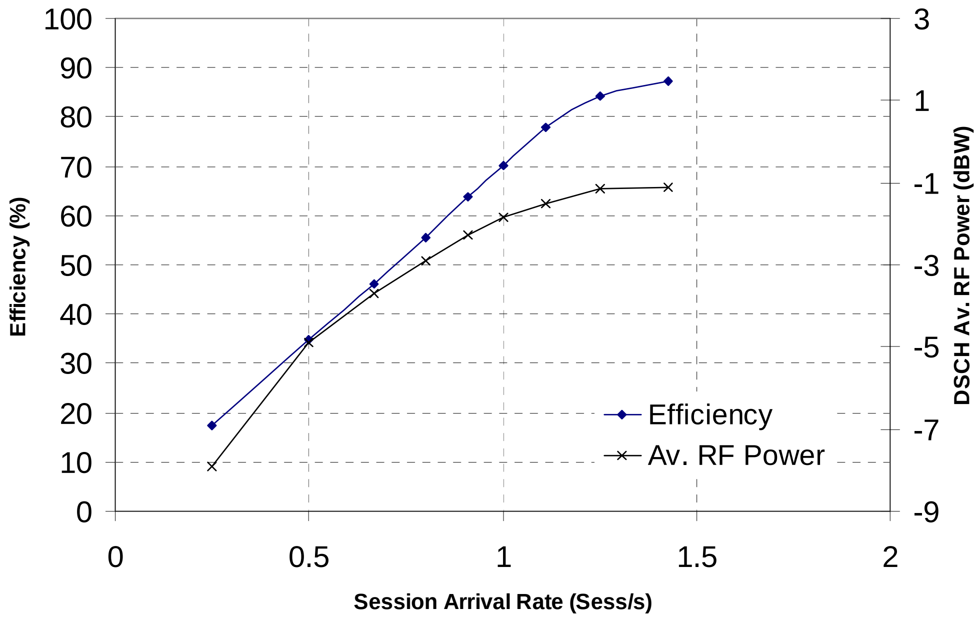


Figure 5: Average RF Power consumed by the DSCH in the same conditions as in Figure 4

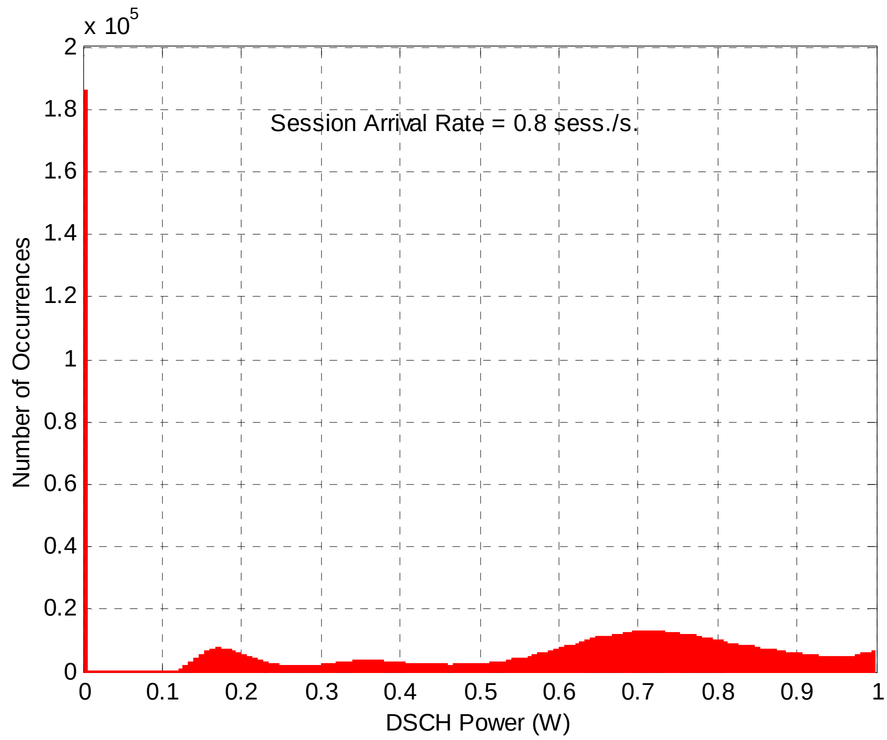


Figure 6: Distribution of the DSCH Tx EIRP (W) in the same conditions as in Figure 4 and session arrival rate = 0,8 sess/s

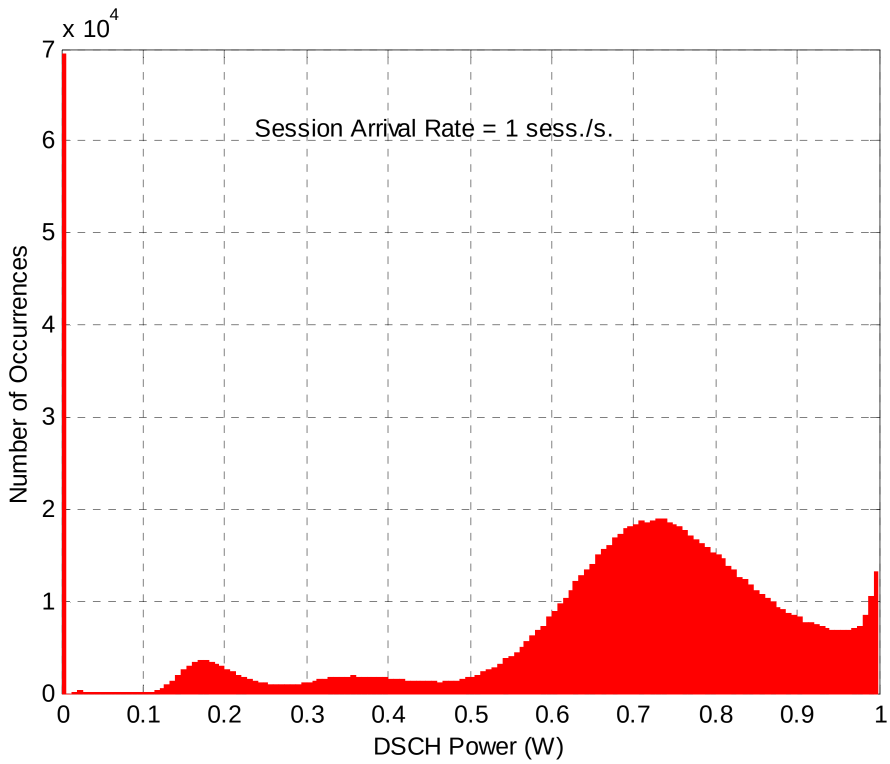
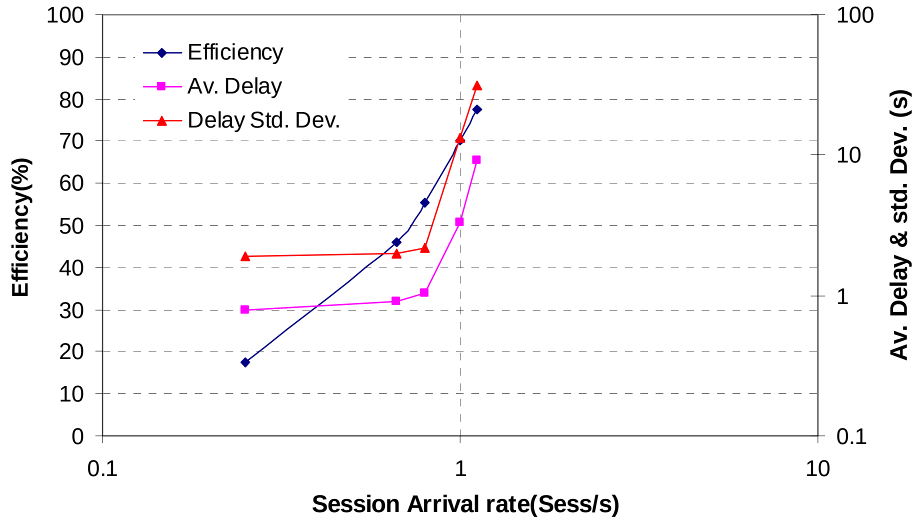


Figure 7: Distribution of the DSCH Tx EIRP (W) in the same conditions as in Figure 4 and session arrival rate = 1 sess/s

Table 4: Maximum Number of MTs with simultaneous open packet session

Session Rate	Max of MT
1,25	331
1,43	384



NOTE: SNIR estimation Error std. dev. at MT 0,5 dB; power control bias variance 2 dB. Two-states fading model, rural environment with mobile speed equal to 3 m/s.

Figure 8: Performance of DSCH with open loop power control

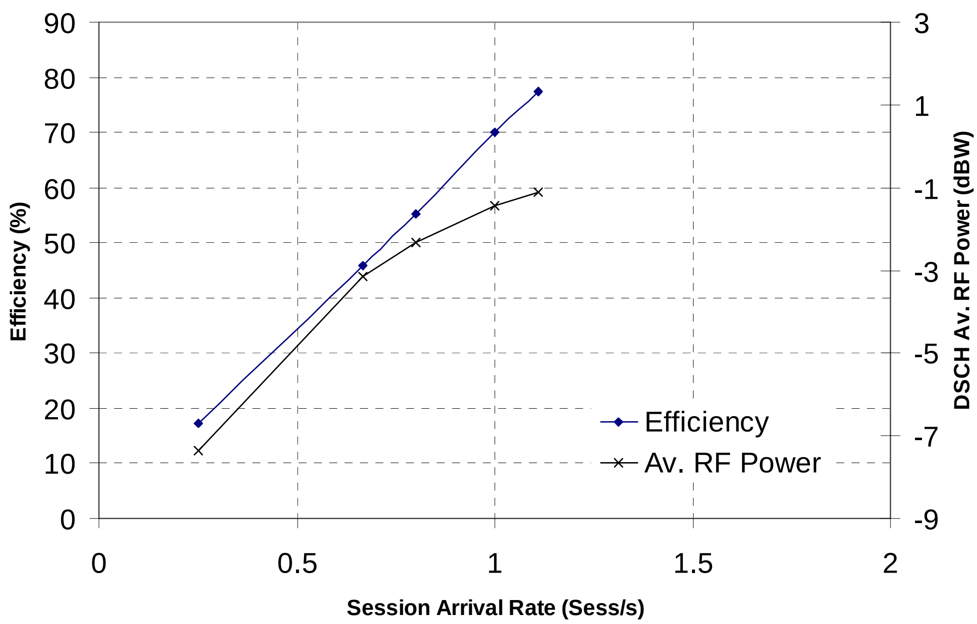


Figure 9: Average RF Power consumed by the DSCH in the same conditions as in previous figure

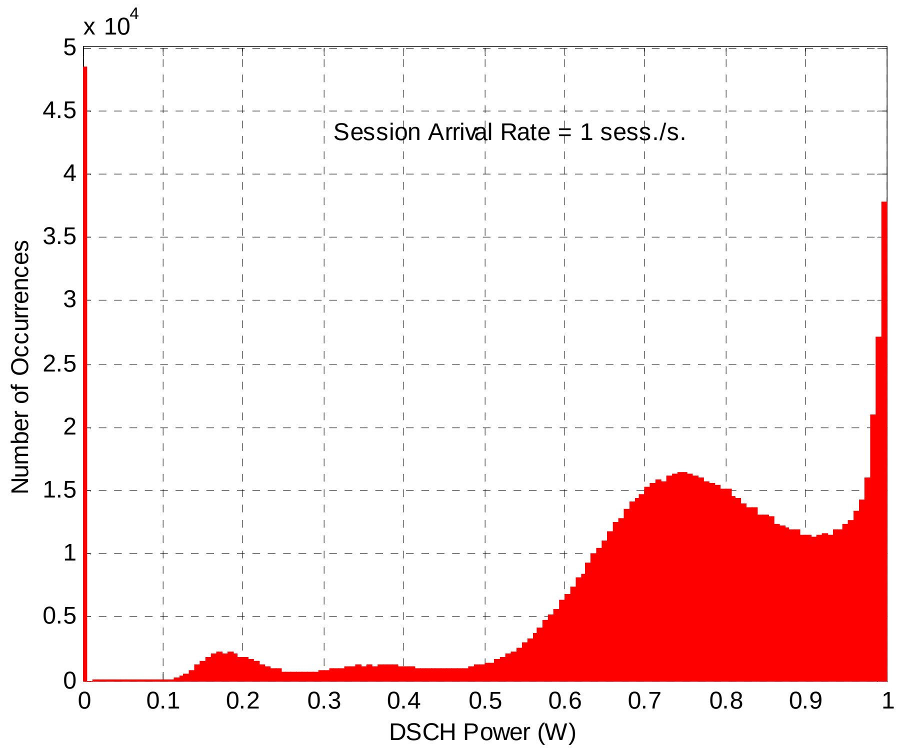
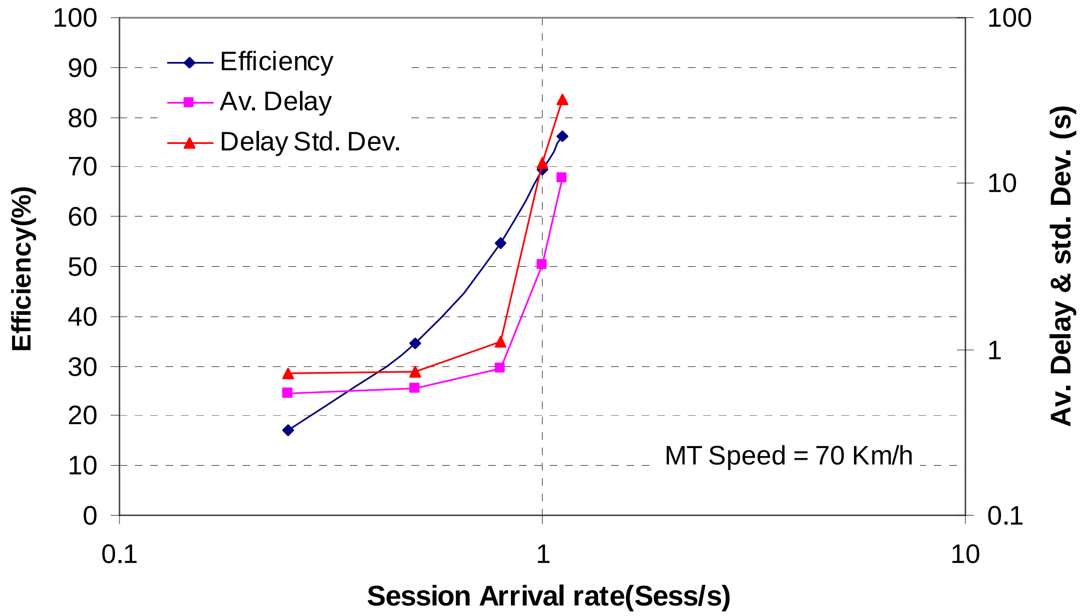
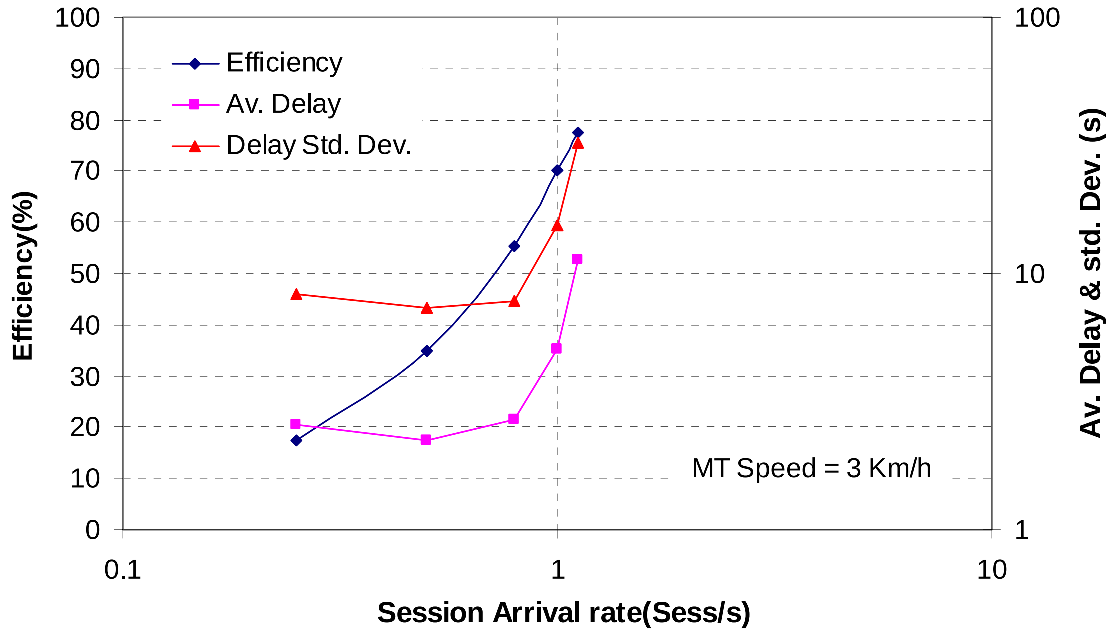


Figure 10: Distribution of the DSCH Tx EIRP (W) in the same conditions as in Figure 8 and session arrival rate = 1 sess/s



NOTE: SNIR estimation Error std. dev. at MT 0,5 dB; power control bias variance 2 dB, two-states fading model. Rural environment with mobile speed equal to 70 km/h.

Figure 11: Performance of DSCH with open loop power control



NOTE: SNIR estimation Error std. dev. at MT 0,5 dB; power control bias variance 2 dB, two-states fading model. Rural environment with mobile speed equal to 3 km/h.

Figure 12: Performance of DSCH with open loop power control

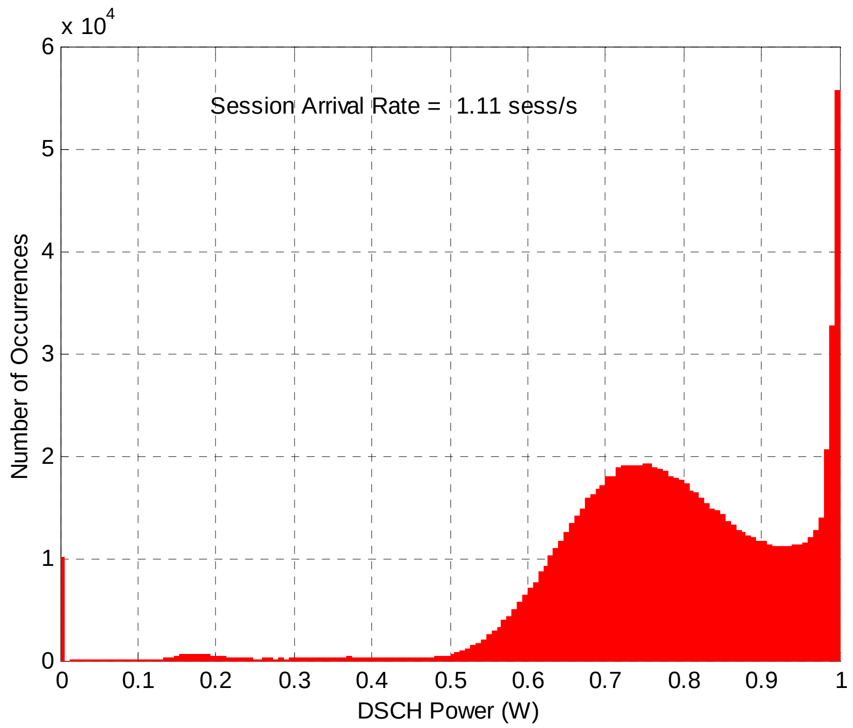
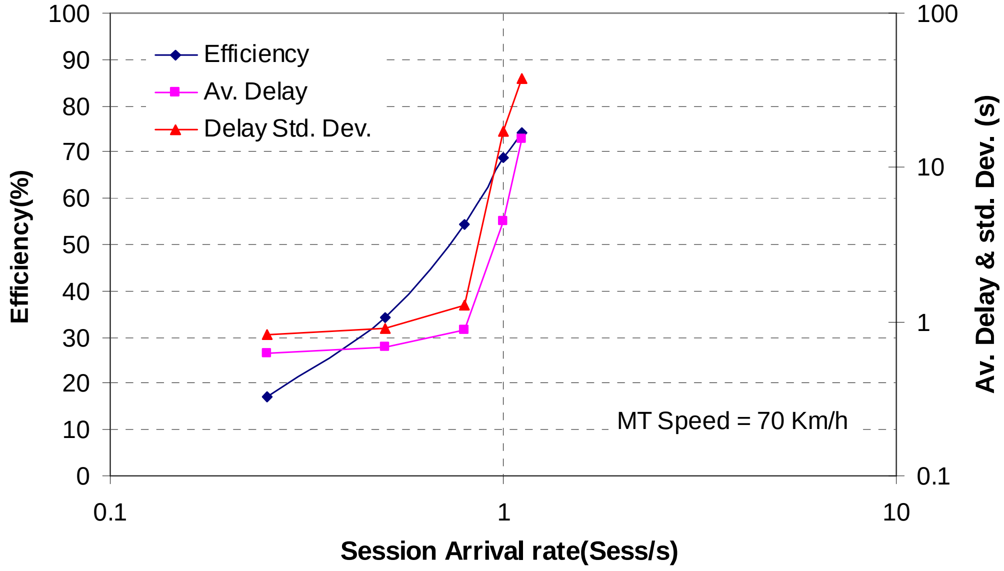
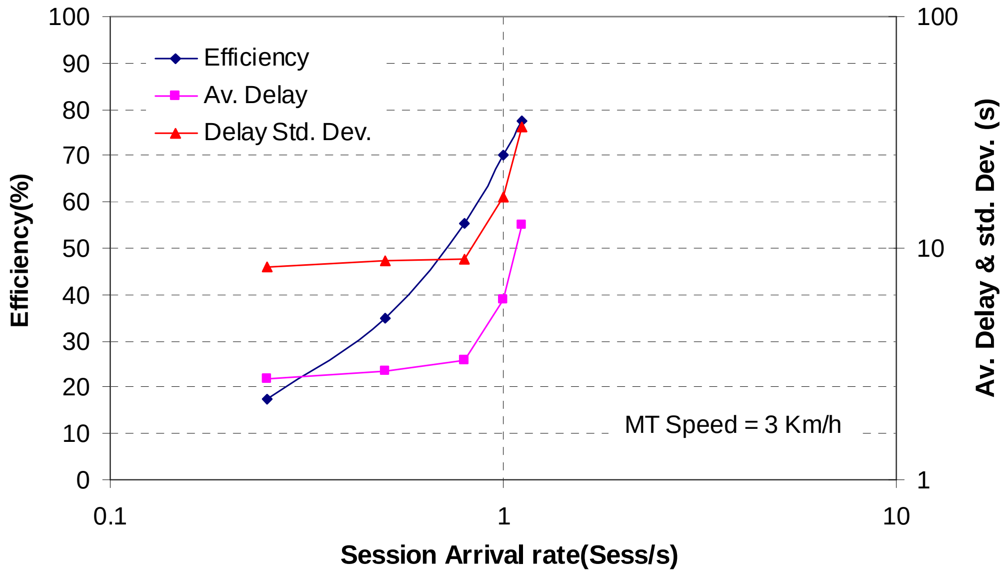


Figure 13: Distribution of the DSCH EIRP (W) in the same conditions as in Figure 12 and session arrival rate = 1,111 sess/s



NOTE: SNIR estimation Error std. dev. at MT 0,5 dB; power control bias variance 2 dB, two-states fading model. Suburban environment with mobile speed equal to 70 km/h.

Figure 14: Performance of DSCH with open loop power control



NOTE: SNIR estimation Error std. dev. at MT 0,5 dB; power control bias variance 2 dB, two-states fading model. Suburban environment with mobile speed equal to 3 km/h.

Figure 15: Performance of DSCH with open loop power control

A.2 Reverse link

For the reverse link packet access, three access techniques were investigated: Spread Aloha, a Common Packet CHannel (CPCH) variant and dynamic Rate on Demand (dRoD). The investigation was aimed mainly at the case of a GEO constellation.

A.2.1 Spread aloha access

A.2.1.1 Analysis

This clause discusses some background on spread Aloha access (slotted and non-slotted). The discussion is quite idealized being no thermal noise considered; further no power unbalance was present and it is assumed that correct packet demodulation happens with probability one as long as the average SIR in the packet exceeds the selected threshold. Below a simple theoretical analysis of the access will be performed.

Spread-Aloha has the potential of much better performances than conventional Aloha. As known conventional Aloha has a limiting efficiency of 18,38 %. Moreover, it tends to be strongly unstable in presence of a large user population. In practice, the limiting efficiency of conventional Aloha is difficult to attain in real life. The Slotted-Aloha variation doubles the limit efficiency. However, it requires strict time synchronization between users and does not improve the stability problem.

The spread-Aloha appears also somewhat more stable than conventional Aloha (for an analytical evaluation of the stability behaviour.

Moreover, it may not require synchronization like in slotted Aloha to achieve good efficiency. In fact, by using FEC coding and interleaving there is no significant difference between the slotted and unslotted variation. In fact, with coding and interleaving a packet will be lost only if the average interference over the whole packet is higher than the threshold. The instantaneous interference may well exceed the threshold without implying the loss of the packet (this would actually be the case without FEC coding and interleaving).

An analysis of slotted spread aloha efficiency was performed for the case of Poissonian distribution of packet arrival and an infinite user population with total aggregated traffic offer equal to *G* packet per time slot.

The unslotted case does not lend itself to easy theoretical analysis. However, some simulations have confirmed the intuitive result that no significant difference in throughput is to be expected (in the idealized case we are considering) provided that the Processing Gain (P_G) is sufficiently high. In most cases the only advantage of slotted spread Aloha with respect to the unslotted variant is some simplification in RACH burst detection due to the lower false alarm probability due to the fact that the burst has to be searched in smaller time windows with respect to the full asynchronous case.

The throughput, S, in packet per time slot is:

$$S = G\,P_G \sum_{k=0}^{\max Car-1} \frac{(G\,P_G)^k}{k!} e^{-G\,P_G}$$

where an infinite user population with total aggregated traffic offer equal to *G* packet per time slot was assumed. In the above equation *maxCar* represents the maximum number of users which can reuse the same time slot before the interference become excessive for correct demodulation and decoding. Finally, a perfect power control was assumed.

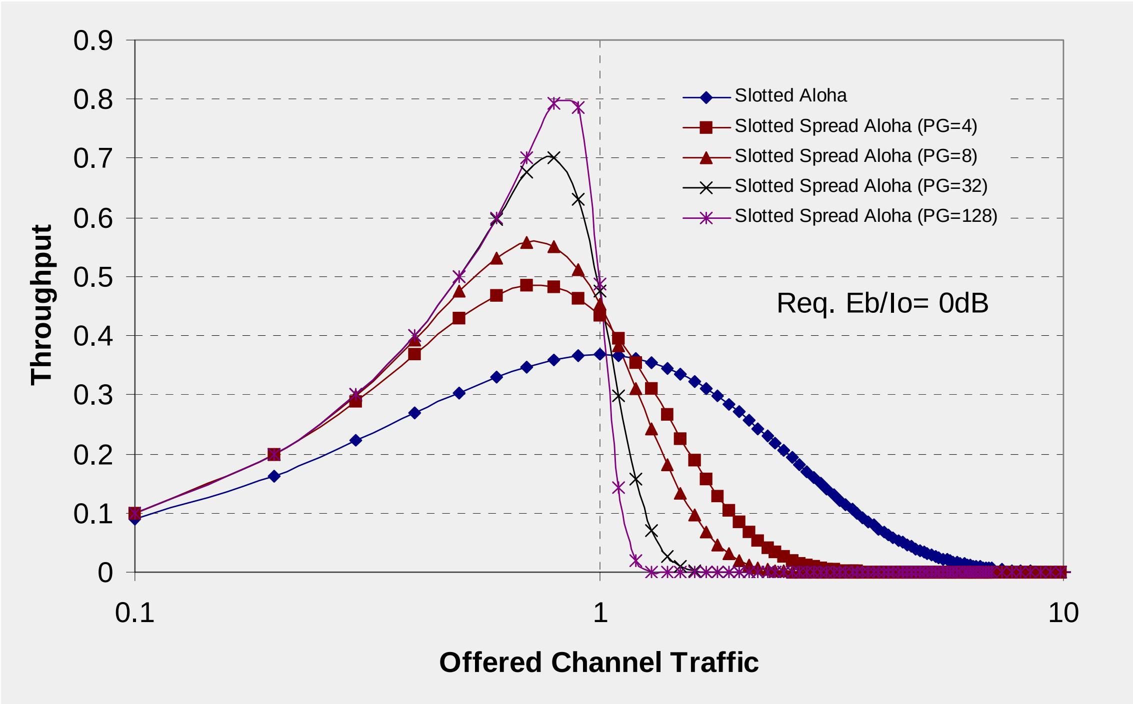
Clearly the throughput, T, expressed in bit/s is obtained by multiplying S by the number of bit per packet and dividing the results for the slot period. Doing that we get for spread slotted Aloha the normalized throughput, T/R_c:

$$T/R_c = G \sum_{k=0}^{\max Car-1} \frac{(G\,P_G)^k}{k!} e^{-G\,P_G}$$

where the normalizing factor R_c is the chip rate.

Figure 16 shows the achievable throughput T/R_c for various spreading factor in the hypothesis that $maxCar = P_G$ (this is quite simplistic and assumes that an S/N ratio of 0 dB is sufficient for demodulation).

It can be seen that, as the P_G increases, the normalized throughput also increases approaching the limit value of unity.



NOTE: The required E_b/I_o was assumed equal to 0 dB.

Figure 16: Throughput (normalized to chip rate) of slotted spread-Aloha versus the Processing Gain (PG)

Figure 17 shows the normalized throughput achievable with slotted spread-Aloha under more realistic assumptions of the required E_b/I_o. Also the case with P_G = 240 correspond to the 3GPP baseline RACH channel (R_c = 3,84 Mchip/s and R_b = 16 kbit/s).

Effect of code collisions

The previous analysis assumes that there is never collision between spreading codes (i.e. the number of spreading codes is assumed infinite).

For a finite number of available codes, there is the possibility of code collision. Let it be N_c the number of available codes. Each user selects a code for transmission in a random way.

The probability p_0 that a packet is successfully transmitted is now equal to the probability that no more than other *maxCar* - 1 users select the same slot for transmission and that none of the other carrier has selected the same spreading code. It can be shown that the normalized Throughput is then:

$$T / R_c = G \sum_{k=0}^{\max Car-1} \left[\frac{(G\,SF)^k}{k!} \left(\frac{N_c - 1}{N_c} \right)^k \right] e^{-G\,SF}$$

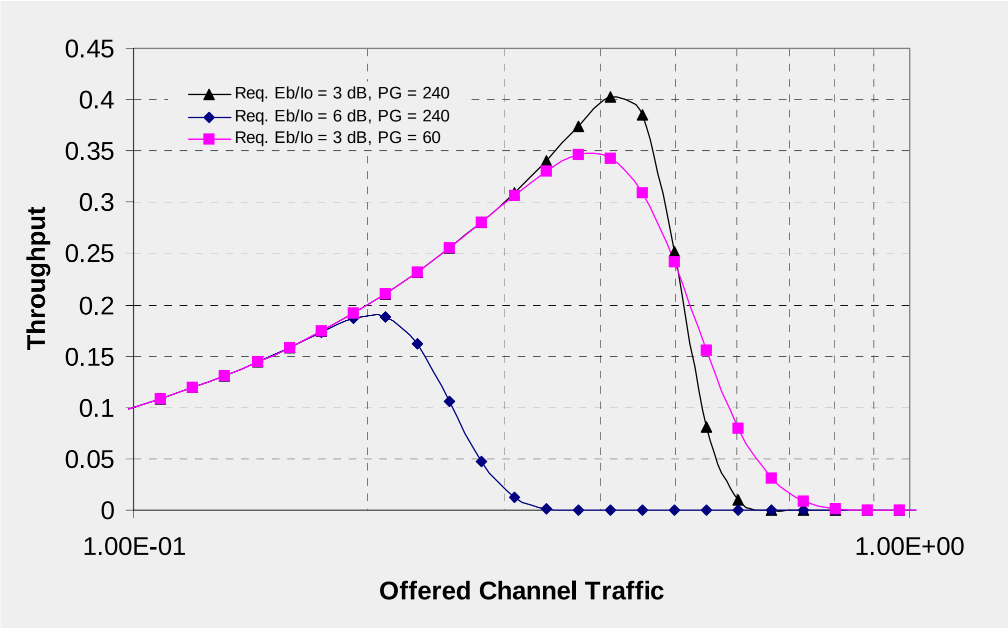


Figure 17: Normalized throughput of slotted spread-Aloha under different hypotheses of P_G and req. E_b/I_o

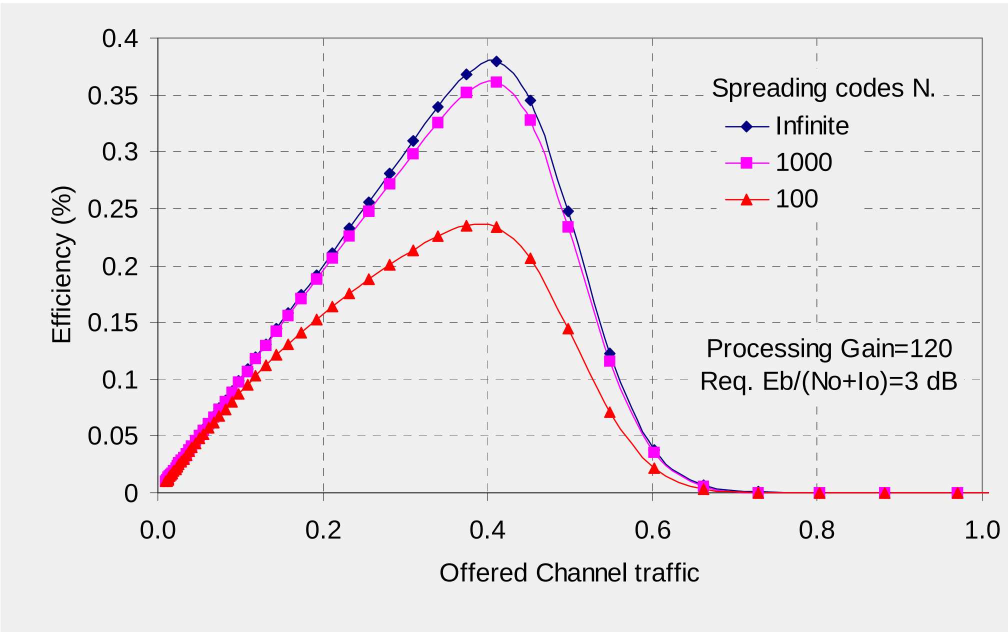


Figure 18: Normalized throughput of slotted spread-Aloha taking into account the possibility of code collision

Figure 18 shows the throughput results taking into account the code collision. With the selected parameter, the number of different codes to allocate appears quite high if impact of code collisions on throughput is to be minimized. Anyway, in practice, a lower number of codes with respect to what is indicated in the figure are likely required because the user synchronization is never perfect. Hence, even if two users select the same code for the same time slot, no actual collision could result due to the fact that the two users may have more than 1 chip period time offset.

Orthogonal slotted spread-Aloha

It shall be underlined that the analysis in preceding clauses applies to the case where spreading codes are random. In case of a synchronous CDMA system, orthogonal code may be used in the system. In such cases, when slotted Aloha is used, we will have an access scheme which we will call orthogonal slotted spread-Aloha. It shall be observed that efficiency in Orthogonal slotted spread-Aloha is the same as in conventional slotted Aloha. In fact, being codes orthogonal there is no limitation from interference; however, orthogonal codes are limited.

To make a comparison between orthogonal slotted spread-Aloha and slotted spread-Aloha with random codes, let us assume the following parameters:

- Chip Rate: 3,84 Mchip/s;
- Bit Rate: 32 kbit/s;
- Spreading Factor: 64 for the orthogonal case (see note) (may be even lower for the random case);
- Processing Gain: 120 (ratio between chip rate and data rate).

NOTE: This implies BPSK modulation and a FEC code with an approximate rate of 1/2.

Only 64 orthogonal codes are then available for the orthogonal slotted spread-Aloha case. Hence, the maximum efficiency achievable is equivalent to a number of simultaneous transmission for time slot equal to $1/e$ times the number of available code resources, i.e. $64/e$ = 23,5. The total achievable throughput is then about 753,4 kbit/s.

For the case with random codes, assuming no code collision (i.e. the use of a very high number of codes), negligible thermal noise and a required E_b / I_o = 3 dB, the results of Figure 18 (and in particular the curve referring to an Infinite number of spreading codes) can be used. It then results that the achievable throughput is 0,379 × 3,84 = 1,45 Mbit/s which is considerably higher then achievable in the orthogonal case. We can assume, as confirmed by simulation that spread Aloha (non-slotted) also approximately achieves the same throughput as slotted spread-Aloha.

It shall be however observed that throughput of orthogonal slotted spread-Aloha can double by using QPSK modulation instead than BPSK modulation. The throughput would in fact become 1,507 Mbit/s which slightly exceeds that of the non-orthogonal case. Anyway, the orthogonal access requires strict synchronization at chip level (not required in the other cases). Also QPSK modulation may be more critical than BPSK. Finally, any code rate reduction in the orthogonal case would translate in a reduction of the throughput, whilst it would be beneficial in the case of random code option due to the reduction of the required $E_b / (N_o + I_o)$. In fact, if the required $E_b / (N_o + I_o)$ could be decreased by 1 dB (i.e. to 2 dB), the non-orthogonal case throughput would increase to 1,85 Mbit/s. Hence the trade-off between orthogonal and non-orthogonal Aloha depends on the specific system characteristics. In particular it will strongly depend on whether the system is thermal or interference noise limited. For a single geostationary satellite case, with MT having low EIRP, the margins against interference may be quite limited and an orthogonal access could make more sense. Vice versa if a satellite constellation providing multiple satellite visibility is available, interference level will be anyway high (with omni directional MT) even if orthogonal codes are used. In such case use of non-orthogonal spread Aloha is clearly advantageous.

A.2.1.2 Simulations

The following parameters were used in the simulations:

- Web traffic model as described in Annex with average Reading Time equal to 40 s (see note).
- Required $E_b / (N_o + I_o)$ equal to 3 dB.
- No propagation delay.
- Time Out for ACK equal to 0,5 s.
- FIFO size equal to 50 segments.

NOTE: The model of clause 3 refer to traffic originated by a web server. For a reverse link a client web traffic model would have been probably more appropriate. We preferred to use anyway the server traffic model because it is more stressing as far as the system performances are concerned.

The FIFO size mentioned above refers to the FIFO constituting the interface between the mobile Application object (containing the traffic generator implementing the web traffic model) and the MAC object implementing the spread (slotted and non) Aloha protocol (see Figure 19).

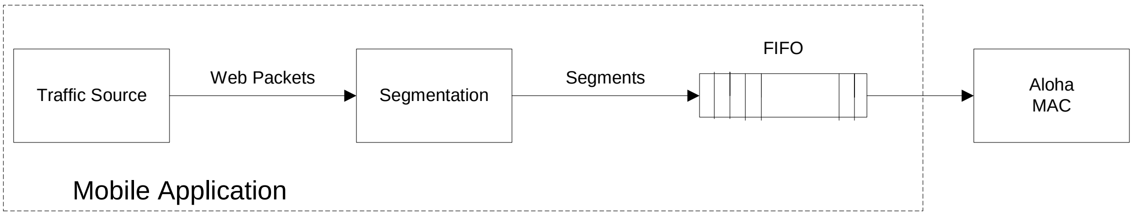


Figure 19: Block diagram of mobile application and MAC interface

The mobile Application object has been instead rewritten to include not only the new traffic generator but also to perform the source packet segmentation function. Traffic packets generated by the ETSI web model are in fact of variable length. This packets will be segmented into MAC packets of fixed length (40 bytes in simulations with MAC packet size of 10 ms). If data are not sufficient to fill a segment, dummy bytes are added. The segments are then sent to the FIFO. The FIFO size, if not otherwise mentioned was set to 50 segments. It shall be observed that the FIFO size actually impact the effective maximum source packet size. In fact, if a source packet size exceeds 2 000 bytes (for a size 50 FIFO and 40 bytes segments) then the additional bytes are actually lost because the FIFO is not able to buffer them.

Figure 20 shows the simulation results of spread Aloha with the above traffic model and the size 50 FIFO. Ten different RACH codes were assumed to minimize code collision. A bit rate of 32 kbit/s was assumed for the RACH channels which translate in a PG = 120 taking into account the chip rate value of 3,84 Mchip/s. Further all packets longer than 40 bytes were segmented in 40 bytes segments for transmission. The 40 bytes limit was compliant with a physical layer frame size of 10 ms. The number of different RACH codes to choose from at each access was 10.

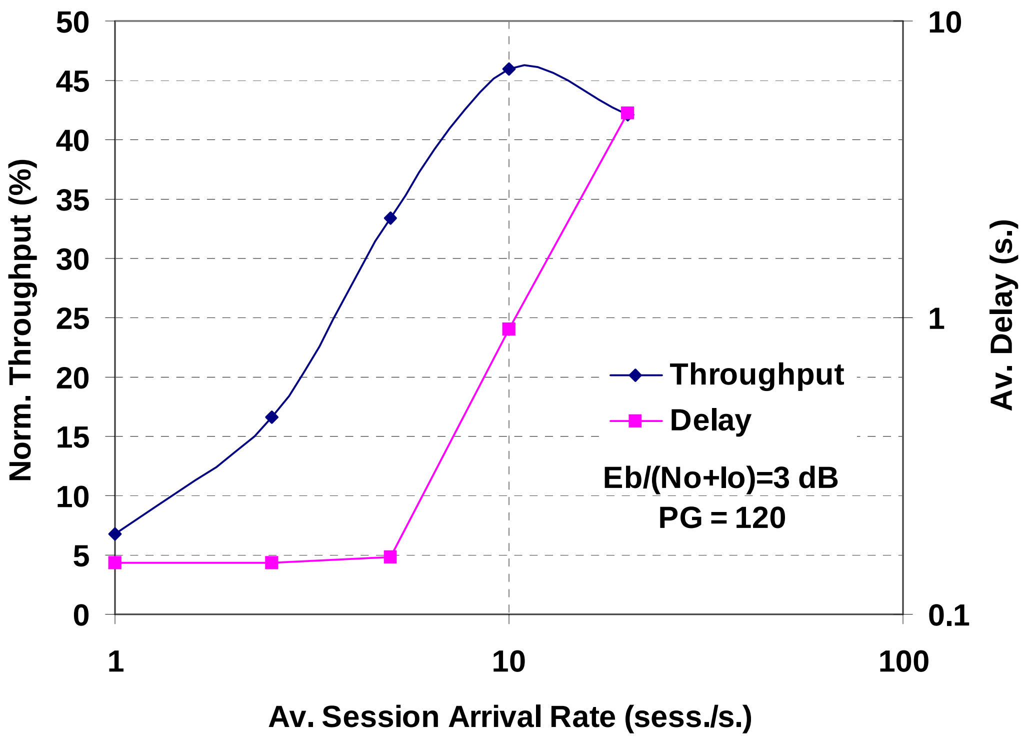


Figure 20: Spread Aloha performance with a more realistic traffic model

It appears that performances are somewhat better than with Poisson traffic model. This is quite encouraging for possible use of the spread Aloha access.

It shall be stressed that the above results do not include the effect of preambles for RACH acquisition. The effect of such preambles is twofold: on one hand, it obviously increases the interference level; on the other one, a miss detection probability has to be considered also.

Figure 21 shows the simulation results taking into account the presence of a preamble for burst acquisition. The preamble length was 12 888 chips and its transmission power was the same as the subsequent data part. It is assumed that the preamble is successfully detected if the average E_c/I_o across the full preamble is greater than -23 dB. At the 32 Kbit/s data speed considered it is equivalent to an E_b/I_o of -2 dB. With such a required E_c/I_o, misdetection probability is actually negligible. The reduction in throughput is only due to the greater interference produced by the preamble presence. Such a reduction is approximately equal to the ratio between the length of the preamble and the data part.

Figure 22 shows results in the same conditions as in Figure 21 but with a required E_c/I_o for correct preamble detection equal to -17,792 corresponding to an E_b/I_0 of 3 dB. Now the misdetection probability become quite significant under high load and the reduction in throughput is higher.

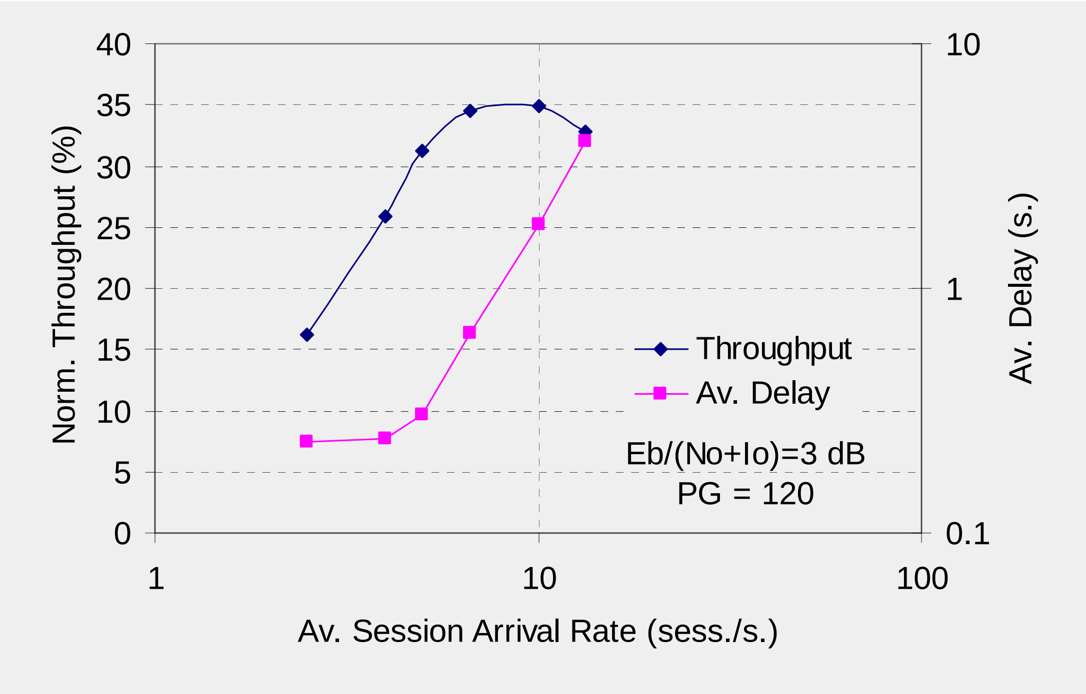


Figure 21: Spread Aloha performance with the same traffic model as in previous figure but with a preamble of 12 288 chips and required E_c/N_o for burst detection of -23 dBc

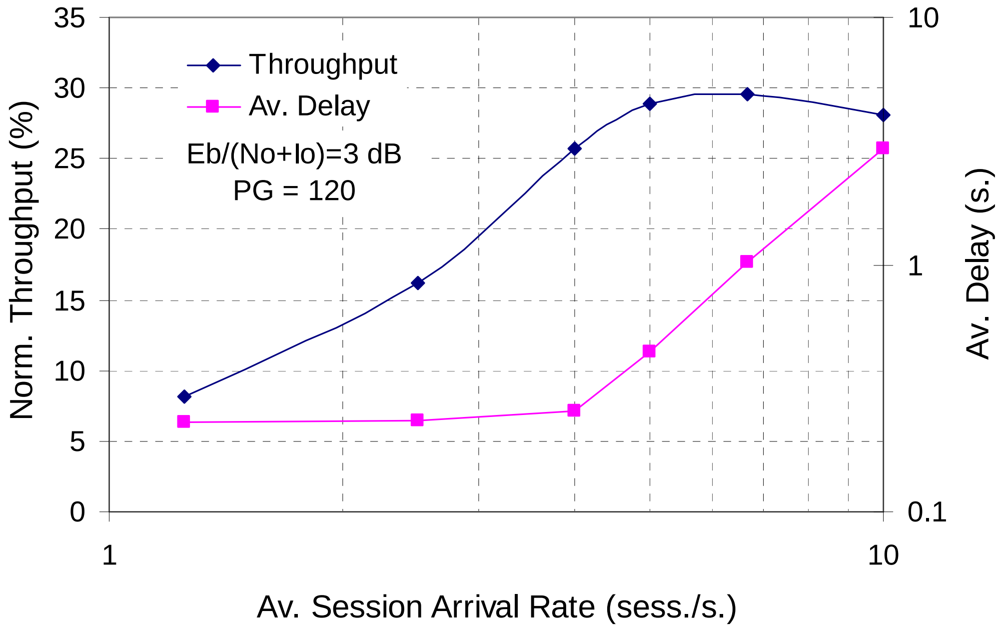


Figure 22: As previous figure but with required E_c/N_o equal to -17,79 dB

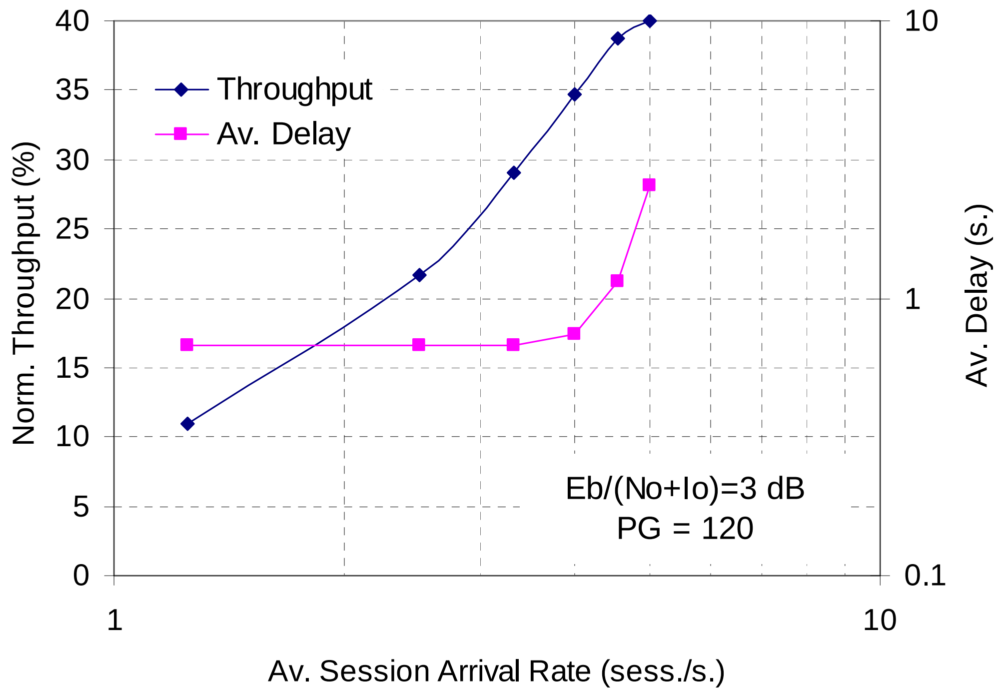


Figure 23: Spread Aloha performance with the web traffic model but with a preamble of 12 288 chips and required E_c/N_o for burst detection of -23 dB and max segment size of 80 ms

Figure 23 shows instead results with variable packet size. In particular each packet has been assumed to range in size from a minimum of one frame to eight frames (80 ms) plus the preamble. By comparing Figure 23 simulation results with those of Figure 21, it is apparent that some improvement of the throughput may be obtained with variable packet size. It is guessed, however, that such improvement is mostly due to the reduced impact of the preamble when longer bursts are used.

It can be observed that average packet delay has also increased with respect to the case of fixed segment size. This, however, is mostly a consequence of two factors:

first, the packet delay statistics is actually computed on MAC segments and not on application packets. Hence, segments which are transmitted without delay only experience the transmission time delay (propagation time has been assumed negligible): this implies that shorter segments also experience a lower delay.

second, having assumed a packet size of up to 8 frames, the maximum source packet supported with a FIFO of size 50 segments is now 16 000 bytes against the 2 000 bytes of earlier simulations.

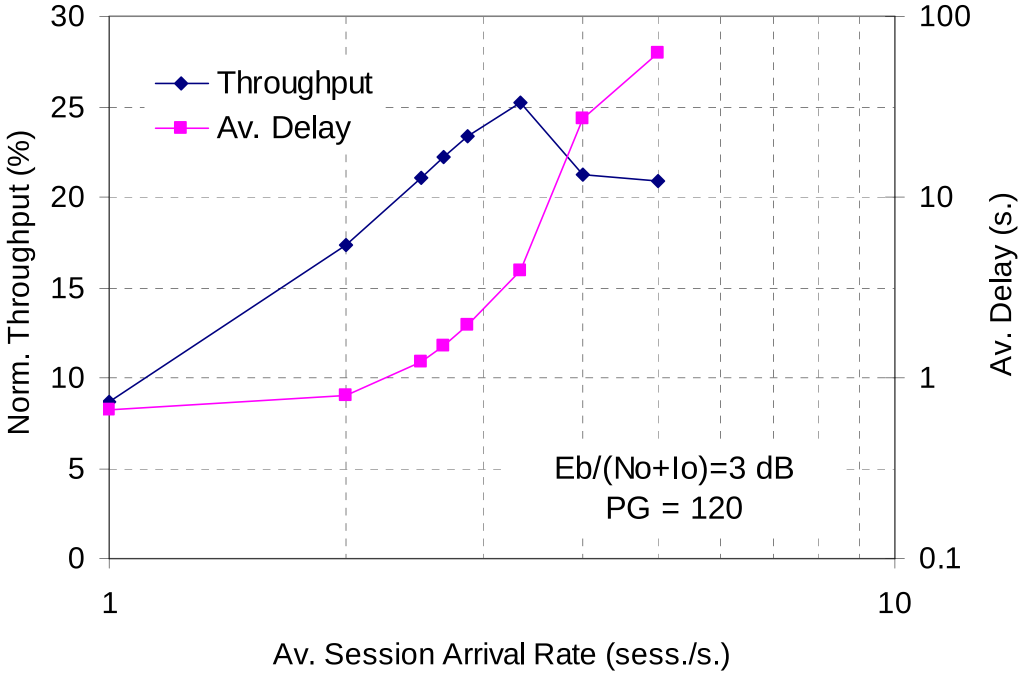


Figure 24: Simulation results in the same conditions as in previous figure but with power control error with 2 dB variance (lognormal distribution)

Figure 24 adds the effect of power control error to the simulation of Figure 23. In particular a lognormal distribution for the error has been assumed with a variance of 2 dB. Effects on throughput and delays are evident. It appears that the maximum achievable throughput actually decrease by about 2 dB (i.e. the variance of the power control error)

Spread Aloha in GEO satellite environment

A new set of simulations with spread Aloha was done taking into account the satellite delay. In particular the worst case scenario (from the point of view of delay) of a geo constellation was assumed (a one way link delay of 245 ms was considered). The same web traffic model as the one shown in previous clause was used. Only difference is the increase of the average packet separation in a packet call which has been increased by the satellite round trip delay (2 × 245 ms).

Clearly for such a scenario the approach followed up to now for the Aloha access, i.e. to wait for an ACK before transmitting a new packet, is too penalizing from the point of view of delay. Moreover, the maximum throughput achievable by a single MT would also be very low in such an hypothesis. What is needed here is the possibility to transmit more than a packet without waiting for the ACKs.

This has been implemented by parallelizing the Aloha protocol. In particular, four parallel Aloha protocol stacks have been considered. Hence up to 4 segments may be transmitted by a MT without waiting for the ACKs as shown in Figure 25.

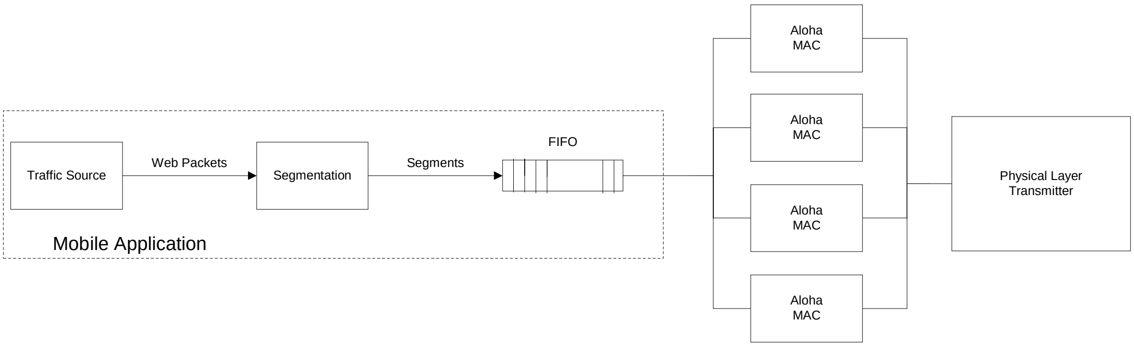
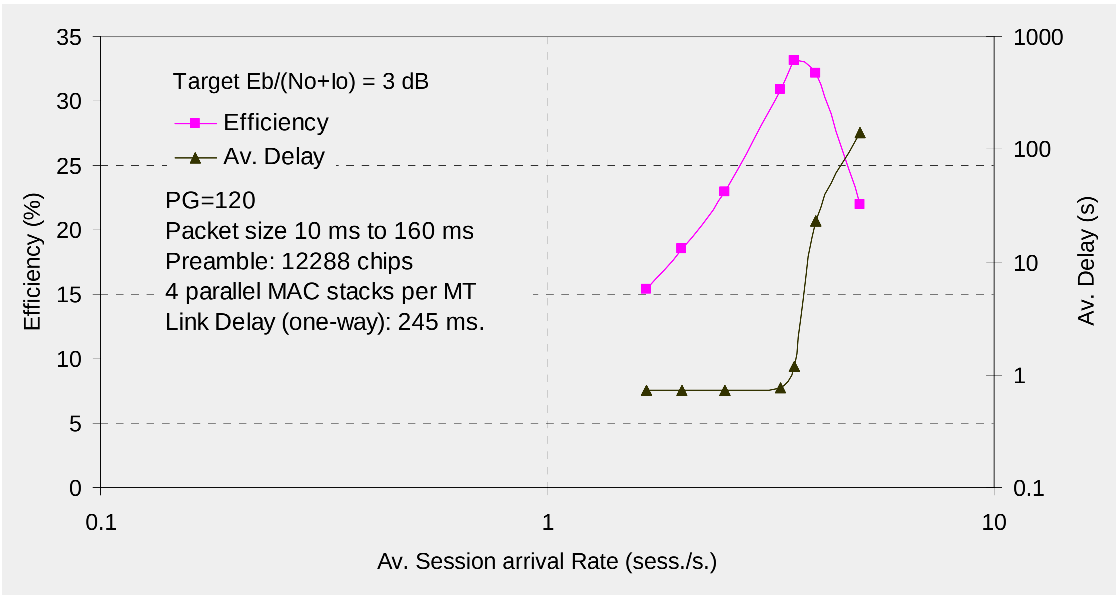


Figure 25: Block diagram of mobile terminal with 4 parallel MAC stacks

Clearly because only one transmitter is available at the MT, the 4 parallel MAC stacks have to be synchronized in order that they cannot simultaneously schedule a packet transmission on the physical layer.

Further, a variable segment size has been considered (from 10 ms up to 160 ms).



NOTE: Four parallel MAC stacks simulated.

Figure 26: Performance of spread Aloha with variable packet size and satellite link delay (245 ms)

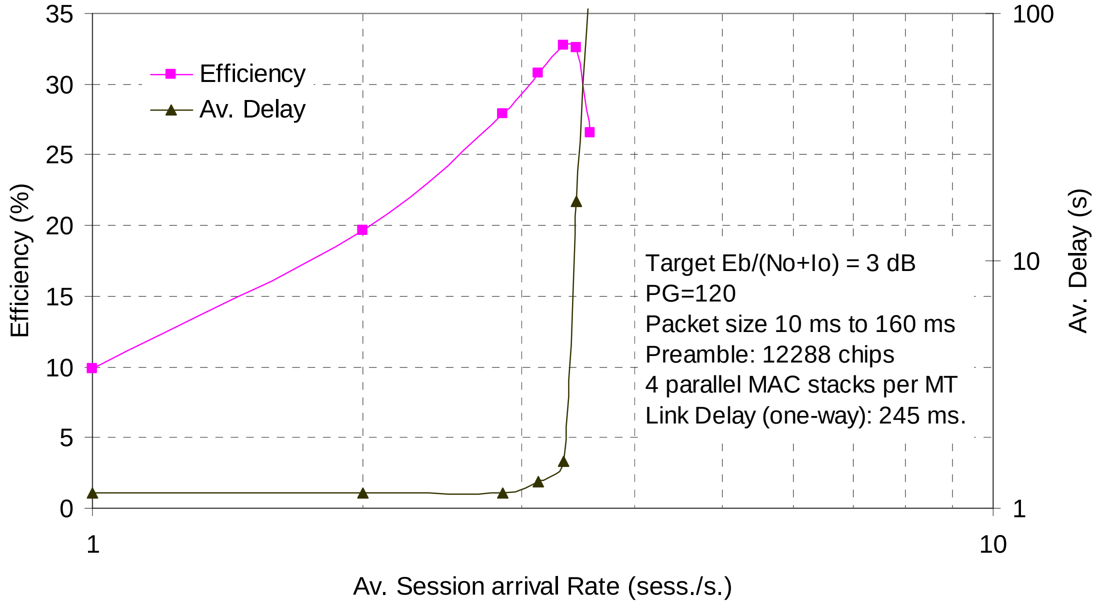
Results of simulation shown in Figure 26 suggest that the spread Aloha is still feasible on geostationary links provided that a facility to transmit more packets before blocking is considered. The average segment delay is below one s. up to the about the maximum throughput. After that it increases to very large values.

It shall be stressed that no optimization of access parameters has been performed. Hence the obtained results may probably be improved with the tuning of the access parameters.

The maximum throughput with respect to cases without delay has decreased a little but this may also be the consequence of the different segment lengths considered.

The results above been obtained with a FIFO size (see Figure 19) still equal to 50 segments as in previous results. To verify the impact of such FIFO on the performances (particularly the delay) its size has been increased to 120 segments. This new size is such that a source packet having the maximum length considered by the ETSI web traffic model (66,666 bytes) could be transmitted without truncation.

Results are shown in Figure 27. As expected the new FIFO size has small impact on the throughput but a large impact on the delay which now exceeds one second. Anyway, taking into account the large possible size of source packets, delay results appears still good.



NOTE: Four parallel MAC stacks simulated. With respect to previous figure a FIFO size of 120 segments was considered.

Figure 27: Performance of spread Aloha with variable packet size and satellite link delay (245 ms)

Finally the effects of power control errors were considered. The power control error was modelled with a lognormal random process. Figure 28 reports the obtained results in the same conditions as in Figure 27 but with power control error standard deviation equal to 2. As expected power control has a quite significant effect on the access performance (although throughput reduction is slightly less than the 2 dB value corresponding to the power control error variance). This is quite unfortunate because it implies that power control has to be quite accurate to avoid impact on access performances.

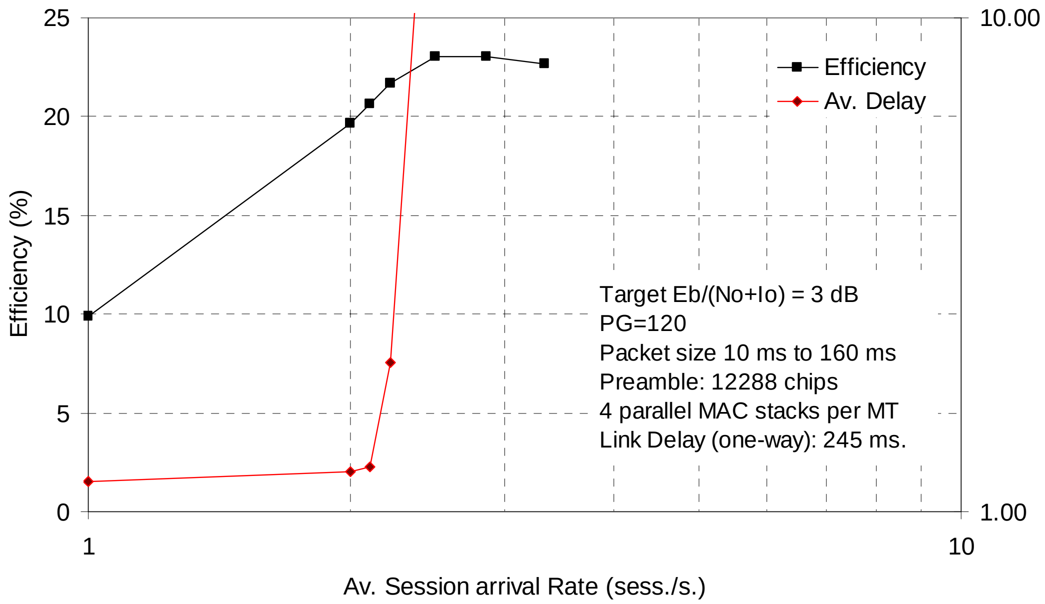


Figure 28: Simulation results in the same conditions as in Figure 27 but with effect of imperfect power control, power control error is lognormal with 2 dB variance

A.2.2 CPCH adaptation to SW-CDMA

The 3GPP CPCH has been adapted by Korean TTA for operation in their SAT-CDMA system. The main emphasis in SAT-CDMA is on LEO systems. Hence, a larger access slot size has been considered by TTA with respect to original 3GPP specifications, i.e. 20 ms (2 frames) against 1,333 ms (2 time slots) in 3GPP. Other differences in SAT-CDMA with respect to 3GGP are:

- the lack of any requirements of an associated down-link DPCCH for supporting power control but its substitution with a common control physical channel (CPCH CCPCH (see note 1)) shared by all user of the CPCH set;
- the Access Preamble (AP) and Collision Detection (CD) preambles are transmitted in pair (one after the other) in SAT-CDMA to reduce transmission delay.

NOTE 1: The CPCH-CCPCH is not subject to power control. This may be acceptable in satellite application where the attenuation dynamic is relatively low.

It shall also be observed that the preamble power ramp-up procedure of 3GPP is maintained although we expect that, with a sensible choice of the parameters, the ramp up iterations can be minimized to avoid impact on delay.

A terminal wishing to transmit data has to grab a CPCH by first transmitting an access preamble in one of the access slot. If an ACK is received then this implies the reservation for such a terminal of a CPCH. The reservation time cannot exceed a number of frames larger than a value NF_max which is set from the RRC layer. The terminal shall release the channel, even if it has more data to transmit, if such number of frames has been exceeded. In such case the random access procedure to grab a new channel shall restart.

Depending on the amount of data to be transmitted, a signature may be chosen for the transmitted access preamble which maps to a CPCH having a given a Transport Format (i.e. a transport block set size, NF_max, and TTI), spreading factor (i.e. data rate), scrambling and channelization codes.

It shall be observed that the CPCH approach does not lends itself very well for application in a GEO environment due the round trip delay. This makes, in fact, likely not appropriate to use a sufficiently large value for the NF_max parameter such that, most of the times, we can assume that the user is able to empty its packet queue with a single access. As a matter of fact, if this would be true the Gateway scheduler would not have the full information available for performing its duty. In fact, if the scheduler decides on a new allocation only based on the current interference load, a too conservative strategy would results because, before the new allocation could actually be effective, a significant time interval (in the order of 250 ms) would elapse and some users could have spontaneously terminated their packet transmission. To avoid such an inefficiency, either overbooking or the assumption that each allocation last a sufficiently short amount of time (i.e. a quite small NF_max) that the probability that the resource is released before actual lease expiration is minimum can be considered. The last approach was the one adopted here. However, it implies that a users which has to transfer a large amount of data needs to perform multiple access requests. With a reservation scheme, this is actually avoided, because the Gateway is informed of the real user needs and can perform the best possible channel allocation. Drawback with a classical reservation scheme is the longer duration of request packets with respect to the access probes of CPCH. However, this is mitigated by the fact that there is no need to perform multiple access probe transmissions for the uploading of very long packets.

Also, with a reservation scheme, it would be possible to extend the current allocation, if required, by in-band inserting of traffic volume information. For this reasons we have proposed to investigate the reservation scheme below dubbed as dynamic Rate on Demand (dRoD) for use in ATB.

The above considerations have been confirmed by some simulation results, although these are not conclusive due to the lack of any optimization of the access parameters.

In particular, in our simplified CPCH access simulator we assumed the availability of the following types of data rates:

- 128 kbit/s;
- 64 kbit/s;
- 32 kbit/s;
- 16 kbit/s.

Further, NF_max was put equal to 4 frames (40 ms). The access preamble length was assumed equal to 12 288 chips. If no ACK is received after an access preamble transmission (or if a NACK is received (see note 2)), the request is repeated after a random waiting time according to the classical Aloha protocol (with 1-persistency). The assumed maximum back-off was 1 s (although in case of overloading the GW can increase such a value).

NOTE 2: A NACK may be received because either the requested CPCH is already in use or because the system loading is too high for satisfying the request.

Finally, it was assumed that the users are perfectly synchronized. This however is not exploited to reduce interference via orthogonal spreading codes but only for simplifying the simulation. Random spreading codes are used as in the case of previous clause on Spread Aloha simulations. Perfect user synchronization with random coding was probably giving worst case performances because all transmitted access preamble are perfectly synchronized. Interference on the channel is thus not constant with time but interference peaks are experienced for the first 12 288 chips of every access slots.

On the other hand a loose access slot synchronization shall be available for the system to work. To avoid the problem of interference peaks at the beginning of each access slot, a randomization of the transmission starting time of the access preamble could be foreseen in order that access preambles are equally distributed within the access slots. This however has not been implemented in our simulator.

As mentioned already, a geostationary satellite was considered for our simulations. Due to the large round trip delay, and the low value for NF_max, a parallel MAC stack implementation was assumed. In particular a terminal may have up to 8 packets on the air before blocking.

Each protocol stack works according to the state diagram shown in Figure 29.

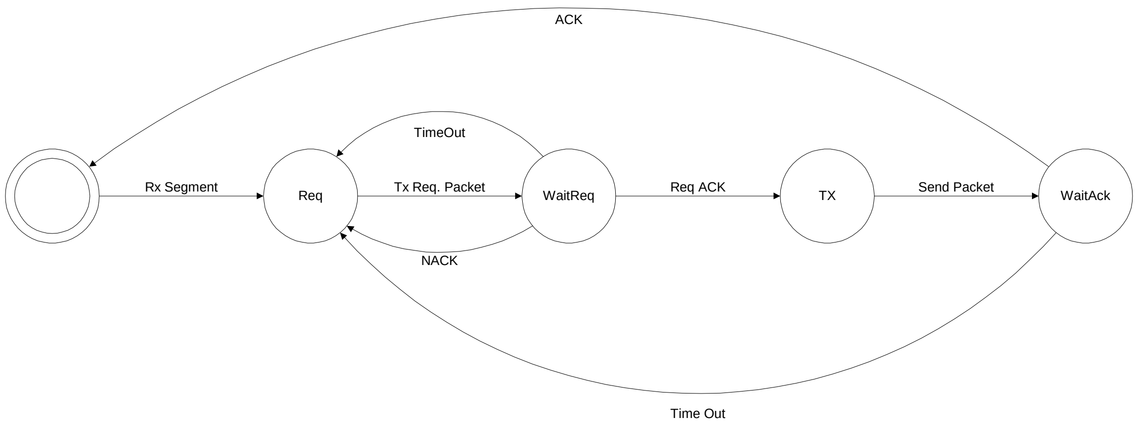


Figure 29: State diagram of CPCH MAC protocol

The processing starts when a new segment is inserted by the higher layer in the FIFO constituting the interface with the MAC. It is assumed that the segment has a size compatible with one of the available CPCH carriers. Based on the segment size, the MAC chooses what access preamble signature to use for the capacity request and stop waiting for an ACK. If an ACK is received the data segment is transmitted and another waiting is started for the data segment ACK. If such an ACK is received the MAC protocol restarts trying to read another segment from the input FIFO. If a time-out is experienced (either after request transmission or after data segment transmission), or a NACK is received, a random waiting time is waited and the whole process restarts with a new access preamble transmission.

Figure 30 shows the obtained results. The same web server traffic model as in spread Aloha was used with the only difference concerning the *reading time* which was shorten to 10 s instead of 40 considered before. This was done to further reduce the memory requirements of the simulation program. Perfect power control was assumed. It appears that performances are worst than for spread Aloha both in terms of efficiency and in terms of delay. The deterioration of the delay was expected due to the need to send request and waiting for an ACK before sending the data segment. More surprising was the deterioration also of the throughput.

Investigations should be carried out to understand if such deterioration of the performances was due to the excessive interference floor due to the requests. A clue to the reason of such bad performance comes from examination of simulation log file. It appears that under heavy load a lot of data segments are lost due to miss-detection of the preamble. At this regard preamble of data segments are transmitted with the same power as the message part of the packet. It is thus highly unlikely that segments are lost on high data rate carriers. More likely miss detection may probably affect the lowest data rate transmission (16 kbit/s in presented simulation results). In fact, the transmission power of data segments (including preamble was fixed for each considered data rate. On the other hand, the power of access preamble was adaptively changed during the simulation to maintain the access preamble miss detection probability at level of 10^{-3} at least up to a power level equivalent to that of a 32 kbit/s data carrier. Hence in case of high loading access preamble may be up to 2 times more powerful of 16 kbit/s data carriers.

Further, the high miss detection probability is also partly due to our approach of strict synchronization which makes the access preamble interfere with high probability with the data segment preamble. In such a case some improvement can be expected by randomizing the starting time of access preamble. Moreover, a longer data carrier preamble (for lower data rate carriers) may also help in recovering performances.

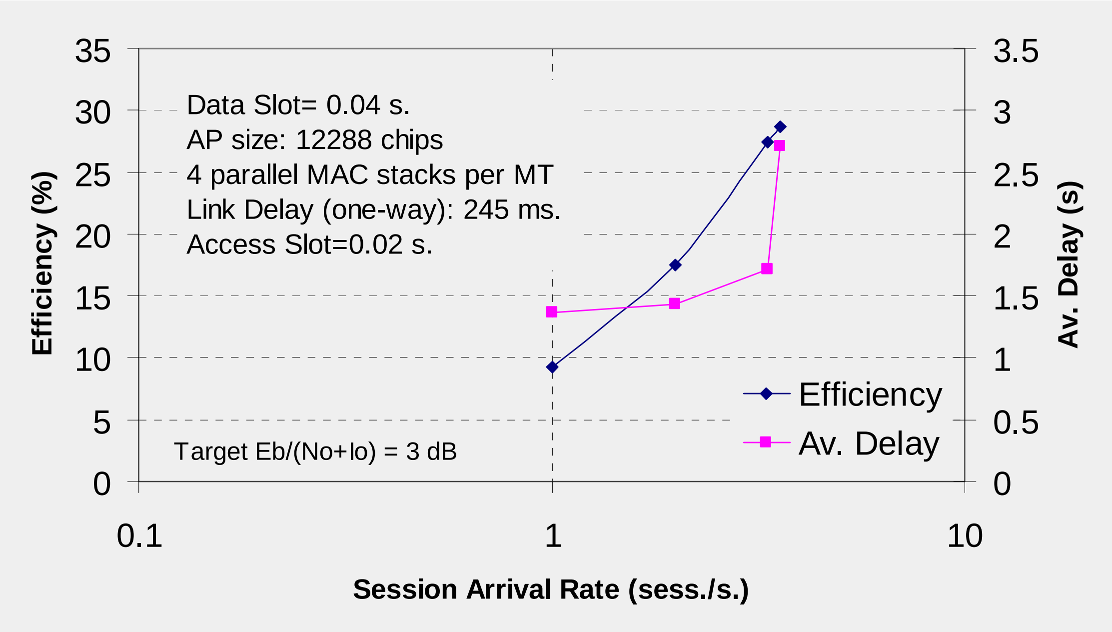


Figure 30: Performance of CPCH

In conclusions, we can state that the CPCH concept is certainly very attractive for low delay propagation environment. For GEO environment, however, its performance seem not attractive, at least with the selected access parameters.

A.2.3 Dynamic rate on demand

A.2.3.1 Access description

The dynamic Rate on Demand (dRoD) is a classical packet MAC protocol based on a reservation strategy. It appears best suitable to application in a GEO environment with respect to alternatives like the CPCH as discussion in the previous clause has highlighted.

In dRoD, a MT which happens to have packets in its input queue, sends a request for capacity to the GW. The request will contain the amount of data in the MT queue waiting for transmission together with parameters (like measured down-link attenuation) allowing the GW to allocate the appropriate resources (both in terms of data rate and allocation time) to the requesting MT. Following the allocation, the MT is able to transmit its data, if new data arrives in the MT queue whilst a transmission of other data is already going-on, a piggy-back request for new capacity is inserted in the packet being transmitted to minimize the need for use of random access for transmitting a new request. This would have the advantage of minimizing latencies in resource allocation as well as reducing the impact on system efficiency under heavy loading conditions.

To formalize a possible dRoD implementation could be based on the following algorithm (see also Figure 31):

Define the scheduler allocation cycle period (T_{ap})

Define an acceptable *maximum Noise Rise (MaxNR).* This value shall be defined taking into account the characteristics of the MTs (available EIRP and operational bit rates) and the maximum link attenuation which is desired to overcome.

Define the data rate for reservation request transmissions.

Measure at the GW the current *Noise Rise* (*currNR*) and broadcast the information to the MTs.

Based on *currNR* the GW will also select the access parameters which will be broadcast to every MTs. In particular, the GW will select:

- the threshold (in Bytes) under which use of random access is allowed for packet transmission. Also associated to such a threshold are the bit rates allowed for random access packet transmission.

The parameters of the random access (for both reservation request and packet transmission). In particular the maximum back off to be used for retransmission after a collision shall be defined. As far as the random access (both for reservation request or packet transmission) is concerned the same protocol as used in the Spread Aloha clause is used.

Based on the access parameter set by the GW, the MT having data in its MAC FIFO queue will transmit according to a Spread Aloha protocol a request for capacity or directly the data (see Figure 31).

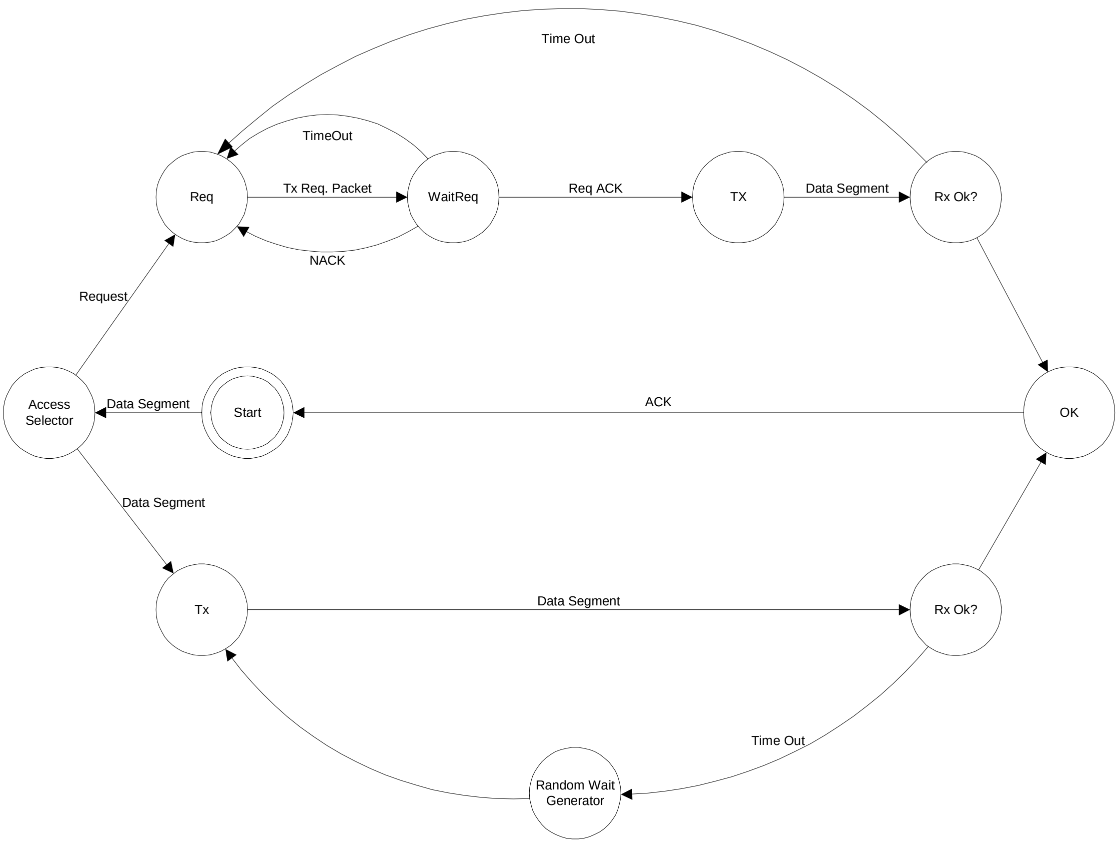


Figure 31: dRod state diagram representation

The request for capacity will be stored in a queue at the GW side. Every T_{ap} sec the GW scheduler is triggered and channel allocations are performed (see later clause on scheduling for the scheduling criteria) taking into account system loading, the current *noise rise* and the one which would results from the allocation. Also, a check on the required MT EIRP will be done (it would be useless to make an allocation which would require an excessive power at the MT due to bad propagation conditions, see also later clause on Power Control). GW scheduler allocations will be in terms of bit rate and period of time a given spreading code is allocated. Allocation time will be such as to allow the MT to transmit the amount of data which were signalled in its request. In case of lightly loaded system, longer than strictly needed time allocation could be done to allow the MT to possibly append additional requests to its packet or even further data segments.

The MT receiving an allocation will compute the required Tx power and if enough power is available transmits its data, otherwise a new request has to be send after a random wait time.

To test the performance of such a system a simulation program has been prepared where the whole reverse link channel is modelled.

Actually the program also models the RACH channels both for capacity requests transmission and for traffic transmission in spread Aloha (i.e. without exploiting the dRoD mechanism). In fact, one objective was also to evaluate how to exploit both dRoD and spread Aloha access to maximize channel usage as well as minimizing channel delay. A strategy based on the traffic volume was implemented at the user terminal for deciding which mechanism to select for channel access.

For the scope of this simulation work, no traffic packet segmentation has been performed. Also no piggy-backing of traffic requests on already on-going transmissions has been implemented. When a traffic packet arrives, a request for capacity containing the length of the packet waiting for transmission is transmitted according to the spread Aloha mechanism (see note 1). As far as the back-off mechanism considered for the random access, if the request is unsuccessful, i.e. no reply if got after a prefixed time-out (0,55 s in these set of simulations), the MT will generate a random waiting time uniformly distributed between 0 and *maxBackOff*. The *maxBackOff* value is established by the GW scheduler (see later clause on the scheduler) based on the system loading. The minimum value for *maxBackOff* was set equal to 0,5 s, whilst the maximum value can grows up to 40 s depending whether the average *noise rise* (see note 2) measured by the GW over the last one second exceeds the acceptable maximum *noise rise* value. When the maximum noise rise is exceeded, the GW broadcast a message disabling idle MT from starting a packet session.

NOTE 1: In order that a request can be send, no transmission shall be in progress. In such a case the request is deferred after the on-going transmission is finished.

NOTE 2: The *noise rise* parameter represents the ratio between the total noise density (i.e. thermal noise plus interference) and the thermal noise density only. In other words a *noise rise* value of 10 would indicate that the interference level is such that the total noise density is 10 times larger than the thermal noise density only.

A.2.4 Simulations

A.2.4.1 Simulations in a simplified system scenario

The simulation program also allows to model the link budget of a real satellite system. At this regard, both the RF parameters of the user terminals (Max EIRP and G/T) as well as of a GEO satellite can be defined. In particular for the GEO is also possible to define the beam pattern. This will also allow to simulate effects related to the different link gain experienced by the different users (whose position is randomly selected within the service area of the satellite). Further, effects of fading channel and imperfect power control are fully simulated. Detailed on these aspects will be given later.

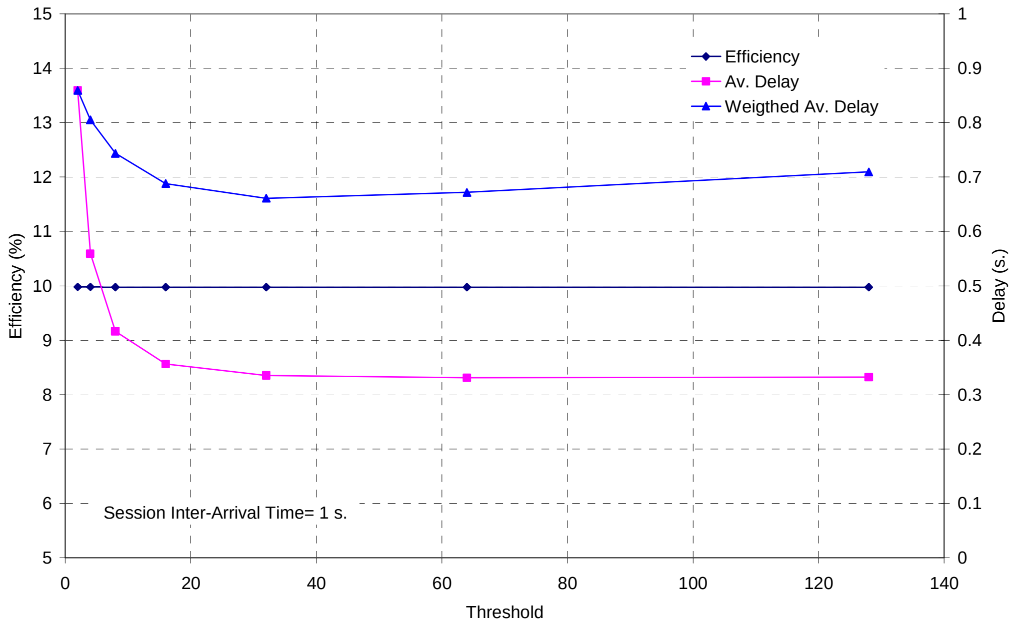
A first set of simulations was run in an ideal scenario without propagation and power control effects for testing the effectiveness of dRoD with respect to both Spread Aloha and CPCH. The results show that there is a slight improvement with respect to spread Aloha (capacity is approaching 40 % of the chip rate). Moreover, the access is fully stable. Also it was shown that power control errors have similar effects on throughput as in Spread Aloha and is almost equal to the power control error variance (in dB).

Some simulations was done with a mixed spread Aloha and dRoD access by setting a threshold packet length under which the spread Aloha access was used instead of the dRoD. Results showed some reduction in the maximum capacity. The delay for mildly loaded system (averaged per packet) however showed a quite significant decrease (see note).

NOTE: If the average packet delay is averaged according to the packet length the difference would instead become negligible. However, we think that the correct measurement here is the averaging per packet and not for byte.

Clearly an adaptive threshold could be set which depends on the system load. The GW could broadcast in such a case such a threshold which for high load could actually disable the spread Aloha mode leaving only the dRoD mode. The following three figures shows, in fact, the performances of the mixed scheme (both in terms of efficiency and average delay) in an ideal environment for different loading conditions and selected threshold. The curves labelled *Weighted Av. Delay* represent the average packet delay resulting from separately computing the average delay of packets transmitted with spread Aloha and with dRoD and then computing the overall average delay by weighting each average with the number of bytes totally delivered by the MTs. In the curve labelled Av. Delay. The overall average is obtained instead by computing the average of packet delay independently of its size and access mode.

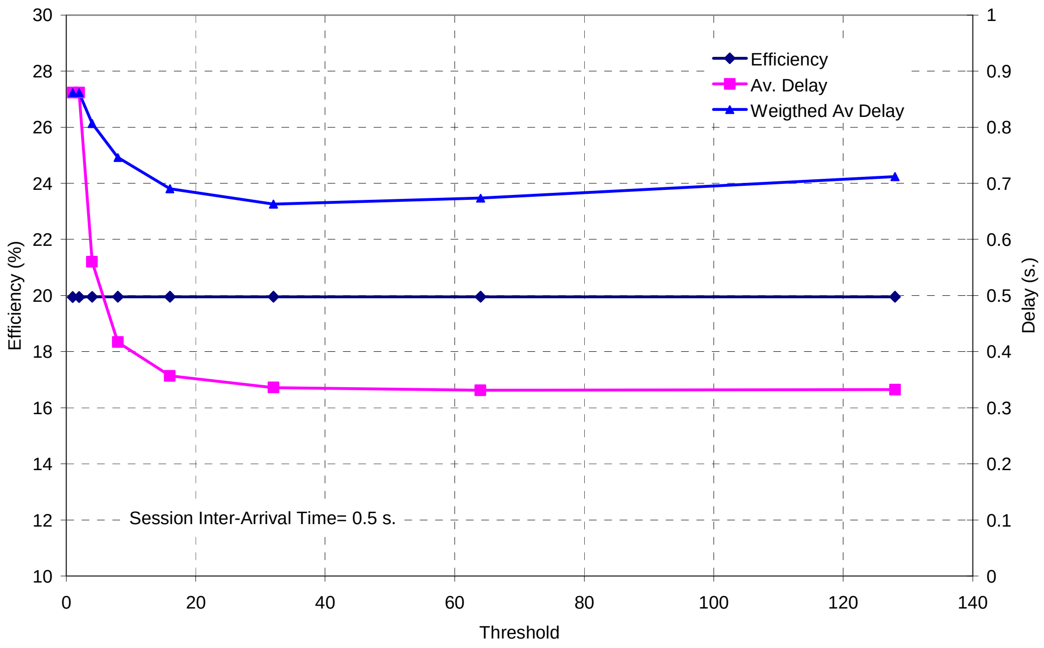
The results were obtained using always a fixed speed for the spread Aloha mode (32 kbit/s) without any parallelization of the spread Aloha protocol (one packet at most can be waiting for Ack). The threshold is related to the maximum packet size (in bytes) which can be still transmitted by spread Aloha through the relation MaxSize = threshold × 40.



NOTE: The weighted average delay curve weights the delay with the packet size. Ideal environment. Session inter-arrival time = 1 s.

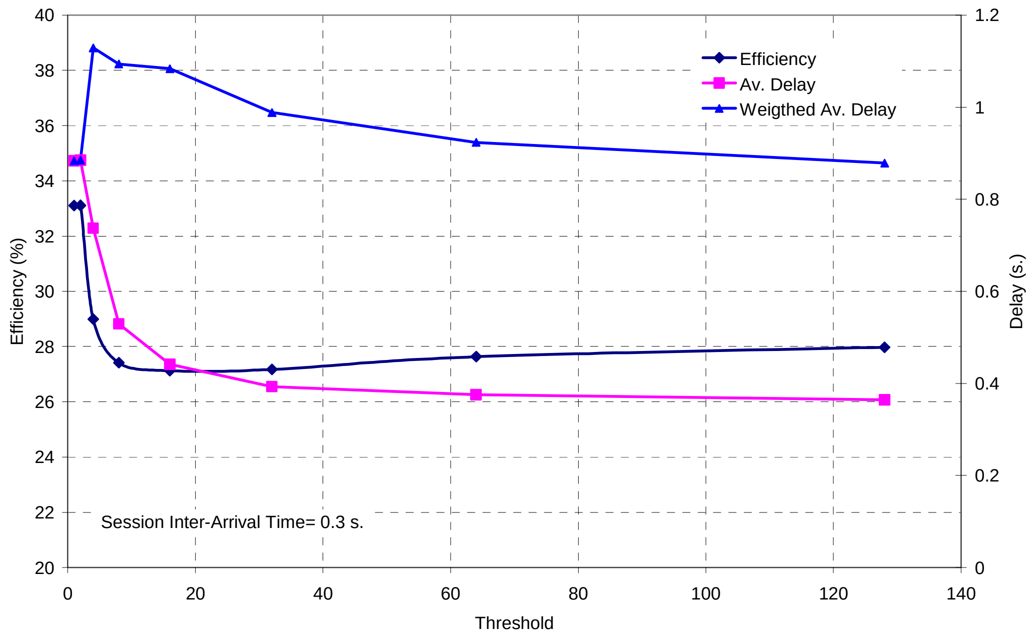
Figure 32: Efficiency and Average Delay with mixed dRoD/spread Aloha versus the packet threshold used for selecting the access

From the figures it appears that for low/medium loaded system the optimum packet size threshold is about 1 000 bytes to 1 500 bytes. Moreover, the performances (both in terms of efficiency and delay) are not very sensitive to the packet size threshold provided that it is set to at least 400 bytes. For highly loaded system (Figure 34) using of the mixed access produce a decrease of throughput although the delay is still decreased. This result however shall be carefully interpreted. In the simulations here reported, in fact, source packet are buffered in an input FIFO which may only contain one packet. If an additional source packet arrives before the old one has been taken in charge from the MAC for transmission, this packet is discarded.



NOTE: The weighted average delay curve weights the delay with the packet size, ideal environment, session inter-arrival time = 0,5 s.

Figure 33: Efficiency and Average Delay with mixed dRoD/spread Aloha versus the packet threshold used for selecting the access



NOTE: The weighted average delay curve weights the delay with the packet size, ideal environment, session inter-arrival time = 0,3 s.

Figure 34: Efficiency and Average Delay with mixed dRoD/spread Aloha versus the packet threshold used for selecting the access

Discarded packets are not included in the statistics about packet delay. However, the number of packed discarded is significant in the simulation of Figure 34 as witnessed by the drop of efficiency. Hence, we can conclude confirming our guess that dRoD access shall be preferred over spread Aloha for high loading and vice versa for low loading.

A.2.4.2 dROD Simulations in more realistic scenarios

Simulations taking into account the real satellite channel characteristics were also done. In particular a single satellite GEO system was simulated. To reduce the simulation CPU time a single beam system was considered. The results here obtained are thus quite optimistic, especially in presence of full frequency reuse due to the additional interference which would come from users outside the beam serving area.

System simulations have been conducted with a vehicular type of terminal with an EIRP of 14 dBW if not otherwise stated.

The simulations which will be discussed next introduces the following new features:

- unequal path loss (even in absence of channel fading and power control errors) due to variations both in range and in satellite beam gain towards the user directions;
- effects of fading;
- the effects of a more realistic strategy of user power management;
- modelling of open-loop power control.

Power Control

A strategy has been adopted to adapt the user transmitted power to the channel loading in order to transmit only the minimum required power.

The strategy is based on the following fact:

- the GW, after any allocation cycle, broadcasts the total noise rise which would result in that cycle as a consequence of the granted transmission allocations. Clearly such noise rise will not take into account the quota of traffic which is due to the random access. Also, because the bandwidth allocations have not all the same duration (see note 1) the resulting noise rise will not be constant with time. However, with the present scheduler, the maximum noise rise is always experienced in the first frame considered for allocation by the scheduler in that allocation cycle (see note 2). Hence, it is such value which is broadcast after each allocation cycle. It shall be observed that a user getting an allocation of several frames, may incur in some problems if during its burst transmission the interference level increases due to new allocations performed by the GW. This could be counteracted by the user if he continuously monitor the noise rise info broadcast by the GW. In our simulations we assumed, however, for simplicity in the simulation logic, that the user EIRP is not changed during a single transmission event. An EIRP margin is then considered by the MT to avoid problems due to a possible system load increase during its transmission.

NOTE 1: Allocation times can be multiple of one frame and is computed by the GW based on the user queue length which is transmitted by the user itself when the request was done and on the bit rate the GW assign to the request. In turn the bit rate is decided by the GW based on the amount of data to be transmitted but also on the EIRP requirement at the MT. In particular, the GW will estimate the MT fading condition and verifies that the required EIRP (inclusive of a safety margin) is lower than the MT maximum EIRP.

NOTE 2: This property depends from the fact that the scheduler here implemented will never decide for an allocation to start at a later frame with respect to the current allocation frame. If an allocation cannot start at the current frame every decision about its allocation is postponed to a later allocation cycle.

Every one second the GW also broadcast to every user the measured system load averaged over the last second (i.e. the actual measured noise rise averaged over a one second interval). This system load, being the result of an actual measurement, and not of an estimation based on traffic allocations only, also takes into account random access requests as well as errors in the setting of EIRP due to power control errors.

The GW also transmits to each user getting a bandwidth allocation, in addition to the allocation itself, also the SNIR measured on the user request. This can be used by the user to calibrate the open loop power control.

As far as the power control algorithm is concerned the following procedure has been implemented. In particular, the MT will estimate the down-link attenuation, *Att*, via the Receives Signal Code Power (RSCP) measurement on the CPICH and will assume that the up-link fading is the same as the down-link (see note 3).

NOTE 3: We also assume here that the up and down-link beam patterns are the same. In case the up and down-link beam patterns are different, the measured down-link attenuation has to be compensated for the different beam gains of the satellite towards the MT respectively in the FL and RL directions. This may however be taken into account with the *margin* variable introduced in the proposed algorithm.

$$Att = Nominal\ RSCP_{CPICH} - measuredRSCP_{CPICH}$$

The nominal $RSCP_{\mathrm{CPICH}}$ is the one a MT located at beam centre shall receive. The measured *Att* also takes into account loss due to the geographical user position (actual beam gain towards the user).

Based on that it will set up the Tx EIRP per bit, $EIRP_{bit}$:

$$EIRP_{bit} = \text{Nominal } EIRP_{bit} + Att + margin$$

As observed above an additional variable called *margin* is included in the EIRP computation. This *margin* variable is updated according to feedbacks (or lack of feedbacks from the GW). In particular, after every successful user transmission, the GW will evaluate the SNIR (see note 4) of the received packet and will feedback the difference between the measured RSCP and the target RSCP (referred below as *Delta*) to the user.

NOTE 4: We are here using the term SNIR as synonymous of the term $E_b/(N_o+I_o)$.

If the user will receive the feedback, then he will update the margin as:

$$margin = margin - Delta/k \qquad \text{a)}$$

where *k* is a factor which was set equal to 30 in the simulations. If no feedback is received then the user will update the margin as:

$$margin = margin + up \qquad \text{b)}$$

where *up* was set equal to 1 dB in our simulations.

In the simulation no updating of the target *RSCP* was performed (it was also not required because we assumed a fixed SNIR threshold of 3 dB for having a correct packet reception (see note 5)). In practice we think that such an update is not required. In fact, the procedure to update the *margin* variable at the MT will set a *margin* value which will produce the desired FER if the parameter *k* and *up* are ad-hoc chosen.

NOTE 5: This does not implies that packets having their average $E_b/(N_o+I_o)$ larger than the threshold are correctly received with probability 1, because the packet could go undetected if its preamble is below the threshold level for detection.

It shall be stressed that the values used for the power loop parameters in our simulations were not optimized. Also a variant of the *margin* equation a) could be better performing. We are referring here to a logic for updating the *margin* variable which simply decreases the value of *margin* at each correct on board reception by a factor *down*. In this case the ratio *down/up* should be selected to be equal to the desired target packet loss probability. Such a solution would have the advantage that no specific feedback on the SNIR measured at the GW is required but just the ACK (or lack of ACK). This solution was however not simulated (for lack of time being the program modifications absolutely trivial) because of the fear that the settling of the *margin* variable could be too slow when packets are transmitted too infrequently. Moreover, the fact that there could be a large time gap between successive packets, may further change the required SNIR and this cannot be fast tracked by such variant. Alternative a) with a smaller *k* factor could have some advantage in such cases.

It shall be finally observed that adaptation of the *margin* variable also compensate for calibration error of the power control (including error in the setting of the CPICH level at the GW). At this regard, it shall be observed that, the power control errors has been modelled in the simulation program through a lognormal distribution defined by its variance and its average value. The average value represents the calibration error. At this regard it shall be observed that the once the MT has computed the $EIRP_{bit}$ to apply to the next transmission, actually an error can be incurred in setting the HPA to the desired EIRP due to the hardware inaccuracy. This error as well the imperfect setting of the CPICH level are modelled through the average value of the lognormal distribution.

In practice in the simulation program it has been preferred to separate the generation of the random error from the calibration error. Hence two separate, zero-mean, lognormal distributions were used. The distribution used to generate random calibration errors, is shared by all MTs. Each MT at the beginning of its packet session will extract its calibration error which will be fixed for the overall duration of its packet session. As a consequence, two parameters are used by the simulator to represent the power control error: both parameters represent the variance of a zero-mean lognormal distribution, i.e. one for the random error and one for extracting the calibration error of each mobile terminal.

Scheduler

The scheduler implemented in the simulator is activated every 10 ms (i.e. each frame). Every request for traffic are thus stored in a list and when the scheduler is triggered, the list is examined and allocations are performed. The allocation algorithm is very simple. The list is examined in a FIFO order. Based on the number of bytes which are on the MT queue (as signalled in the request) a tentative bit rate is assigned. The tentative bit rate setting tries to allocate transmission slots preferably of the order of 10 frames (100 ms). A maximum bit rate of 128 kbit/s, as well as a lower bit rate of 8 kbit/s were however foreseen in the performed simulations. The range of permitted bit rates are those comprised between the above extremes and are in an integer power of two ratio with 8 kbits/s.

The power required for each allocation is evaluated. At this regard, a worst case approach is followed. In particular, a maximum noise rise, *maxNR*, is selected by the scheduler. This parameter represents the ratio between the total noise + interference received by the GW and the thermal noise only. In other words a *maxNR* value of 10 would indicate that the maximum loading that the GW will accept is corresponding to an RSSI (Received Signal Strength Indicator) which is 10 times the thermal noise floor. On the basis of the value of *maxNR*, the scheduler computes how much power is required at the MT for each allocation. If an allocation cannot be satisfied because otherwise the *maxNR* value is exceeded then the scheduler tries to reduce the bit rate for such an allocation. If the bit rate allowed is lower than 8 kbit/s, the request is deferred to a later allocation cycle. However, even if the allocation is deferred, a signalling message is anyway sent to the MT, informing that the request has been received and will be processed as soon as possible. The MT will then set a waiting timer to trigger after a predefined time-out (5 sec in present simulations). If an allocation is not received within the time-out period a new request is send. A deferred request is processed with higher priority at the next scheduling event. However, if a request is deferred for more than a given period (4 sec in current simulations) then it is discarded. No information is given of this event to the MT, because it will anyway retransmit its request if its waiting timer expires.

Before transmitting the assignments to the MTs, the scheduler will evaluate the real *NR* that the allocations would produce. In fact, if the *maxNR* value has not been exceeded the power required for each allocation can be recomputed to take into account the lower *NR*, thus further decreasing the *NR*. The *NR* resulting from this last step is broadcast to the users which can then compute the required EIRP for their transmission.

Simulation results

Figure 35 shows the results obtained via simulation (see Table 5 for a summary of simulation parameters). It appears that the maximum throughput in absence of power control bias, RSCP/SNIR measurement errors and fading will top at about 35 %.

In presence of RSCP/SNIR measurement errors and power control calibration errors (0,5 dB standard deviation for random error were considered plus a bias (random for each MT) with 4 dB variance. From Figure 36, it appears that the random bias is well compensated. Figure 37 refers to the same set of simulations as Figure 36 and shows the average MT EIRP as well as the noise rise experienced during the simulations.

As far as the noise rise is concerned, it shows that the noise rise is approaching the maximum value programmed for these simulation runs (i.e. 10) although does not reach it even for the maximum load.

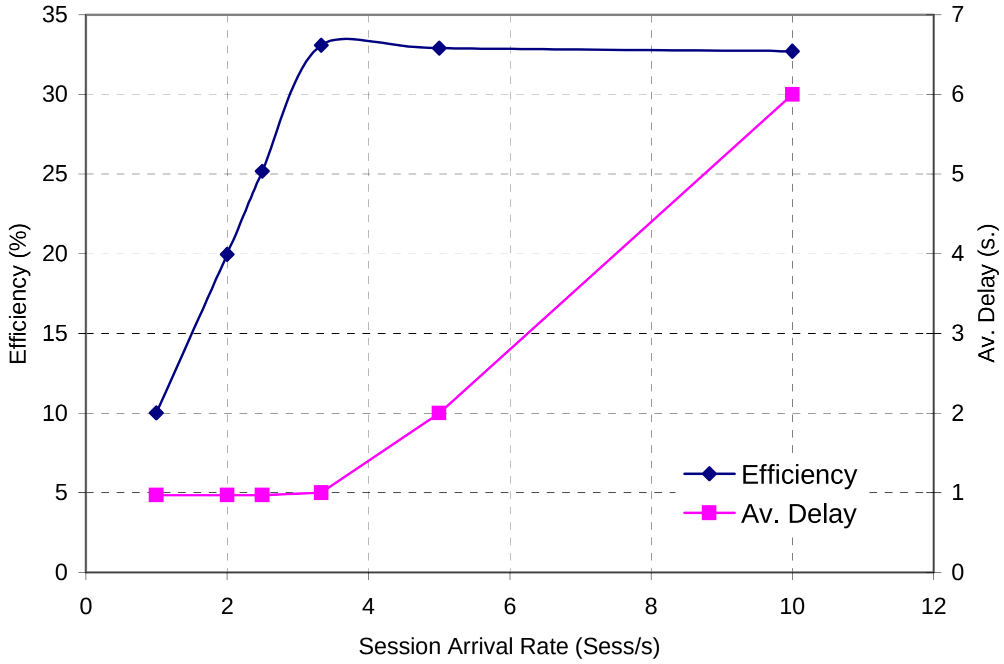


Figure 35: System simulator results without fading and with no SNIR measurement error and power control loop bias

Table 5: Summary of main simulation parameters

Traffic Model	ETSI Web Server Model with 10 s reading time
MT peak EIRP	15 dBW
N. of simulated satellite beams	1
Satellite Beam Peak Gain	42,7 dBi
Satellite G/T (centre of coverage)	15,7 dB/K
Coverage Area (Beam Width)	1 deg @ -2 dB

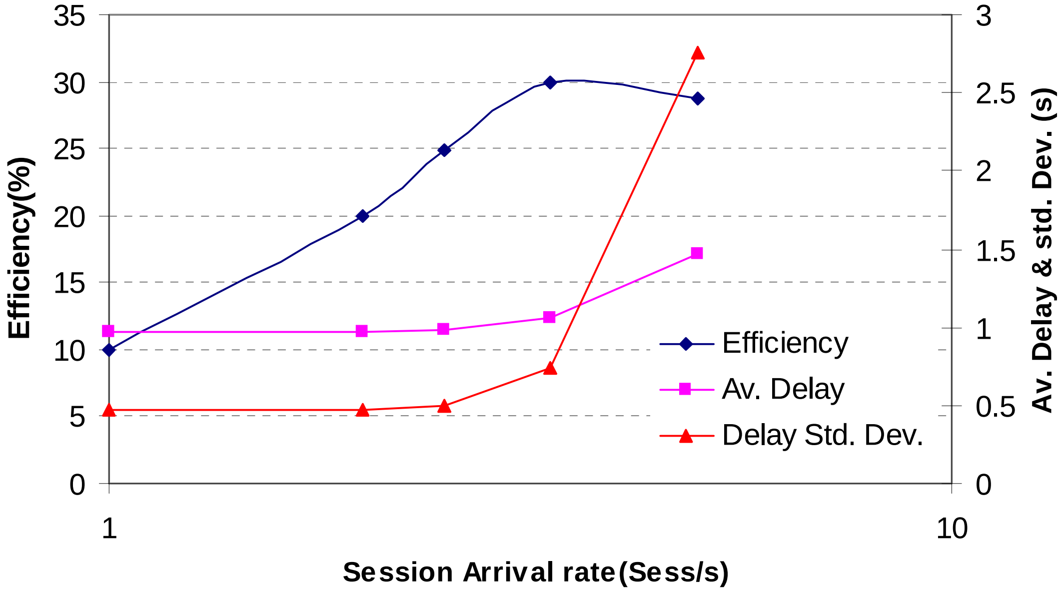


Figure 36: System simulator results with SNIR measurement error standard dev. equal to 0,5 dB and power control loop bias variance equal to 4 dB, no fading

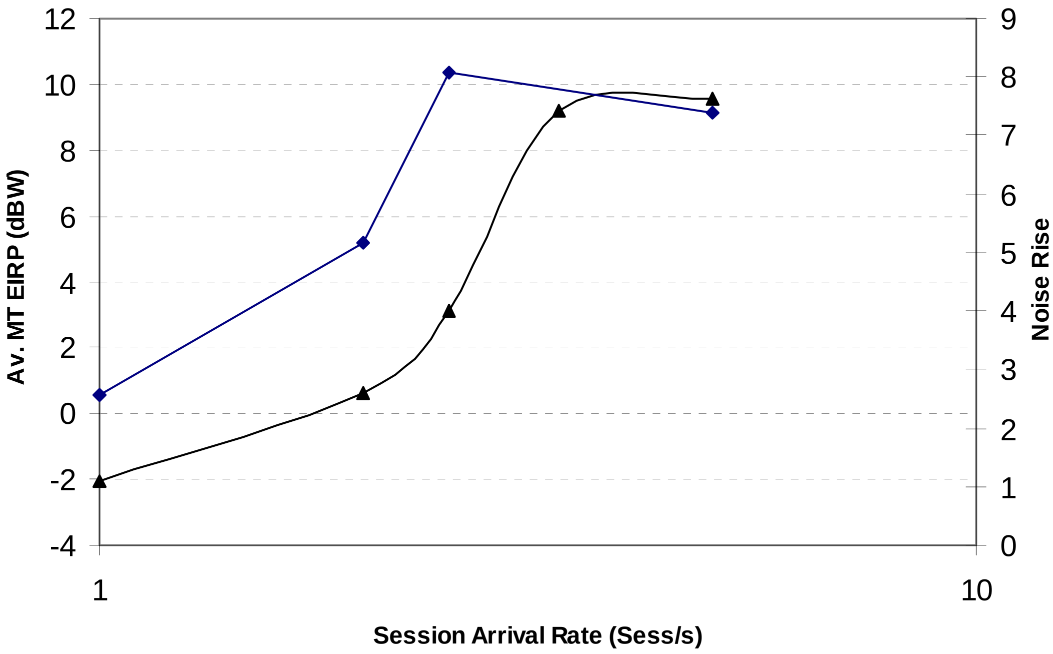


Figure 37: MT Average EIRP and noise rise in the same simulations as in previous figure

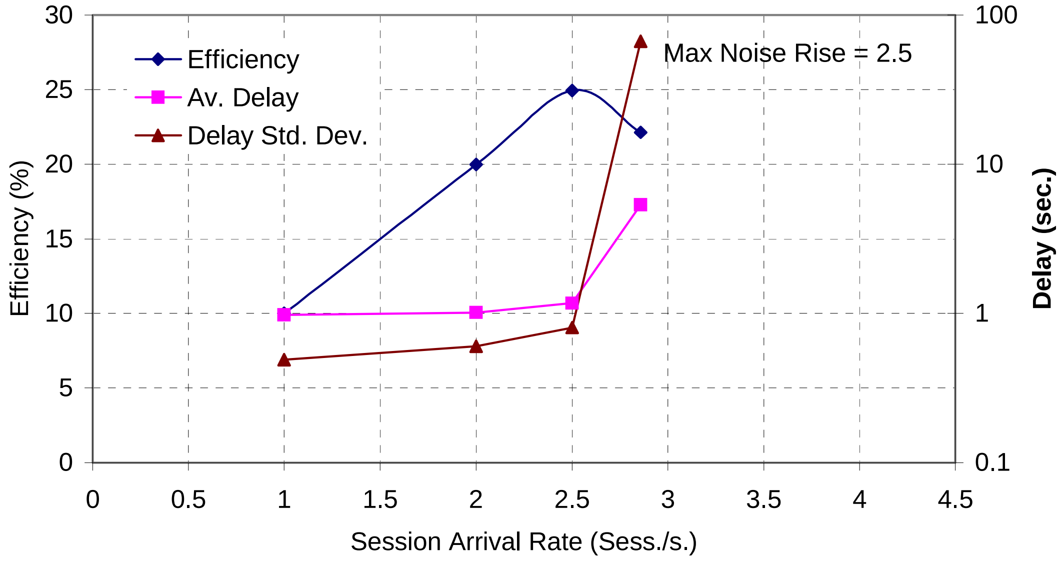
Finally simulations including the effect of the propagation channel were performed. Figure 38 to Figure 40 show the obtained results in a rural environment and user speed of 70 km/h. The fading characteristics are shown in Annex . In particular the results of Figure 38 presents the results obtained assuming a value for *maxNR* equal to 2,5. This value is significantly lower than the one assumed in previous figures (obtained with *maxNR* = 10). The reason to use such a lower value for the *maxNR* parameter lays in the fact that the MT maximum EIRP (15 dBW in this set of simulations) would not be sufficient to counteract the channel fading in presence of a too high noise rise. This would bring to a complete interruption of the service during the shadowed state and a consequently large average delay.

Obviously having reduced the maximum acceptable interference, the obtained maximum throughput is somewhat reduced with respect to the case without shadowing.

Figure 39 and Figure 40 show instead how the performance changes versus the *maxNR* parameter assuming a constant offered traffic (one and two new sessions per second respectively in the two figures).

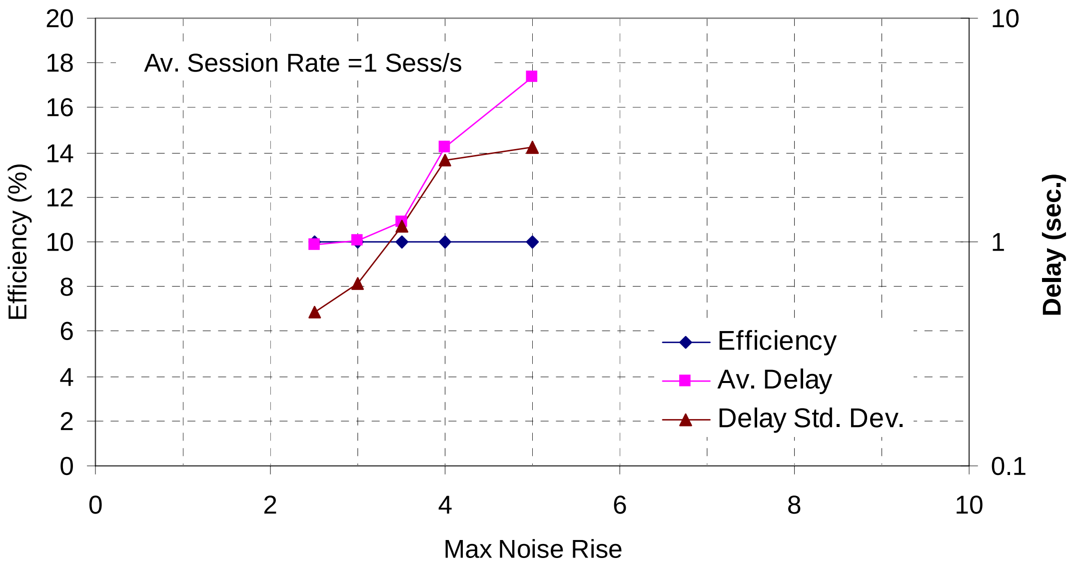
A slightly different scheduler was also tested in which the evaluation of the required MT power is performed on the basis of the actual interference scenario and not on the one corresponding to the *maxNR* parameter (see note 6). This kind of scheduler is expected to produce the same result as the original scheduler under heavy loading conditions (in that case in fact the allocated power is such that the actual noise rise is approximately equal to the value of *maxNR*) but it is also expected some small improvement in the delay performances under low loading conditions. In that case, in fact, the scheduler may decide to make an high speed reservation for an user even if propagation conditions are not optimum because the MT EIRP could still be sufficient due the low interference power.

NOTE 6: The estimation of the required MT EIRP is needed in order to avoid that the scheduler assign a bit rate to the MT which the user cannot support for lack of power. In such a case the scheduler will reduce the bit rate allocation or, in case no useful bit rate can be assigned due to a too strong MT shadowing, will defer the allocation to a later time.



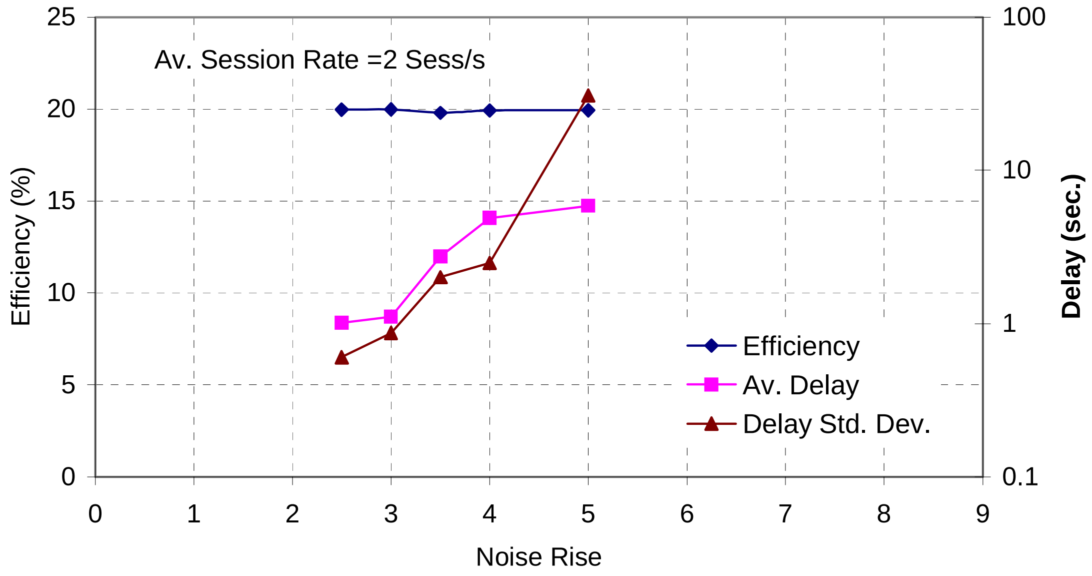
NOTE: Attenuation in the shadows state is lognormal with 8 dB average and 4 dB variance. Mobile speed is 70 km/h.

Figure 38: System simulator results in a rural environment with SNIR measurement error standard dev. equal to 0,5 dB and power control loop bias variance equal to 4 dB



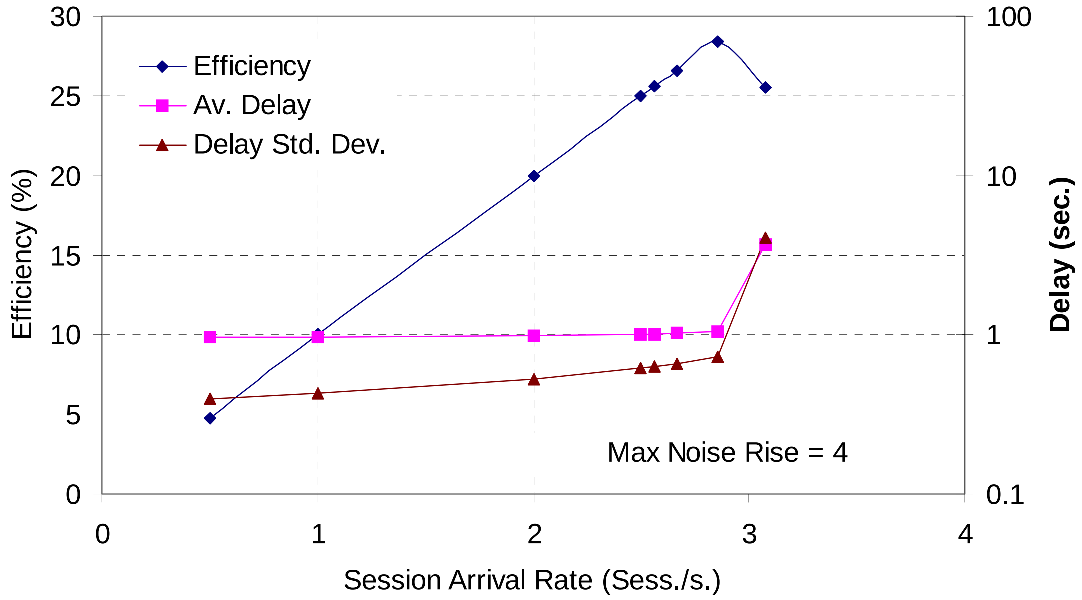
NOTE: The offered traffic is constant (an average of one new packet session per second).

Figure 39: Results in the same environment as in previous figure but with different values of the Max Noise Ratio parameter



NOTE: The offered traffic is constant (an average of two new packet sessions per second).

Figure 40: Results in the same environment as in Figure 38 but with different values for Max Noise Ratio parameter



NOTE: Attenuation in the shadows state is lognormal with 8 dB average and 4 dB variance. User speed is 70 km/h. A more optimistic scheduler is used instead of the one used in Figure 38.

Figure 41: System simulator results in a rural environment with SNIR measurement error standard dev. equal to 0,5 dB and power control loop bias variance equal to 4 dB

As it appears from Figure 41, the obtained results are slightly better than those obtained in Figure 38. Finally statistics about the actual MT EIRP and total Rx signal load at the satellite receiver inputs have been collected for the simulation point of Figure 41 corresponding to an input traffic load of 2,5 new packet sessions per second. Results are shown respectively in Figure 42 and Figure 43.

It appears that the distribution of MT EIRP is bimodal. This behaviour is due in our view to both the two state channel model and to the lower average power which is associated to the request transmission. Regarding the interference load at the satellite receiver input, it shall underlined that the thermal noise floor (measured in a bandwidth equal to chip rate) at the same reference point was -135,7 dB in the referred simulation.

Finally results in the same conditions as in Figure 41 but in a suburban environment are shown in Figure 44. No significant variation with respect to the rural case is noted due to the fact that sufficient EIRP is available in the MT to overcome, for most of the time, the channel attenuation.

It can be noted that some instability seems to be still affecting the dRoD access when realistic impairments are considered (see for example Figure 44). Our guess is that this instability can however be controlled by reducing the maximum acceptable noise rise at the GW as shown in Figure 45.

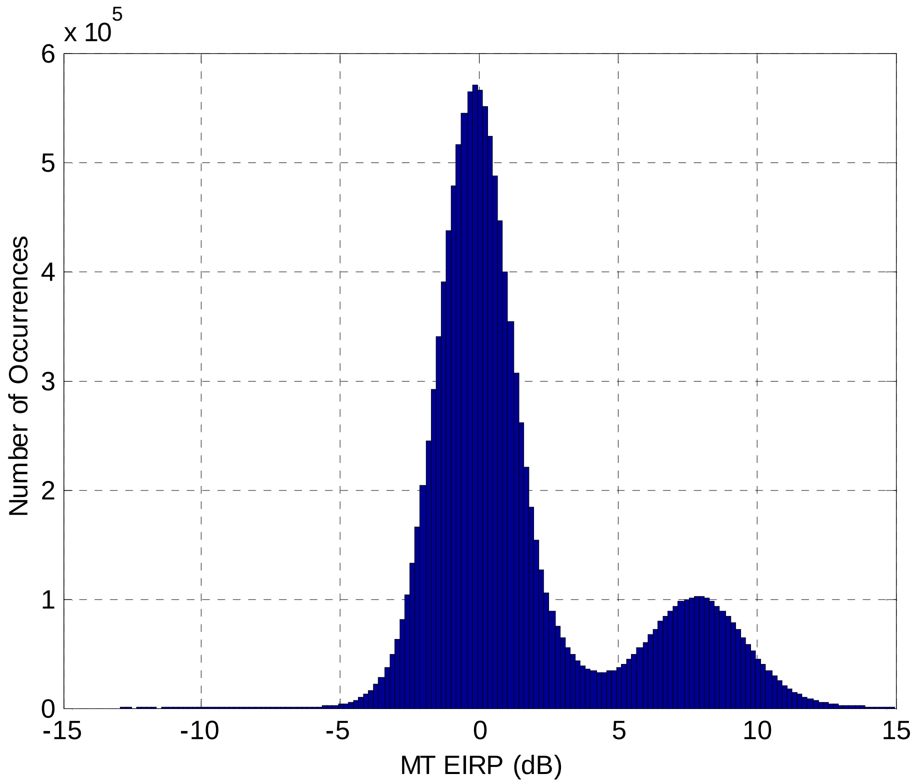


Figure 42: Histogram of MTs EIRP in the same conditions as in Figure 41 and taffic load corresponding to a session arrival rate of 2,5 sess/s

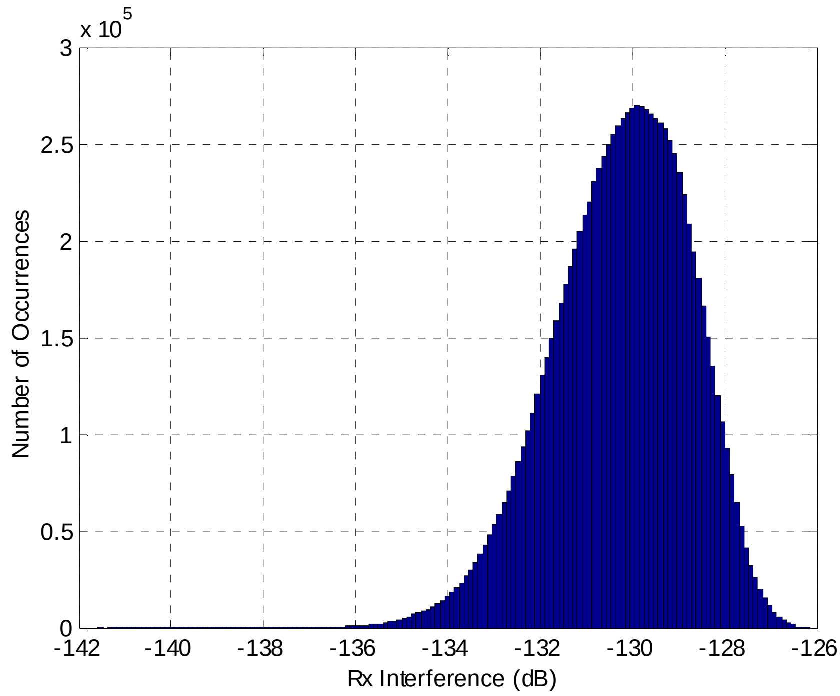
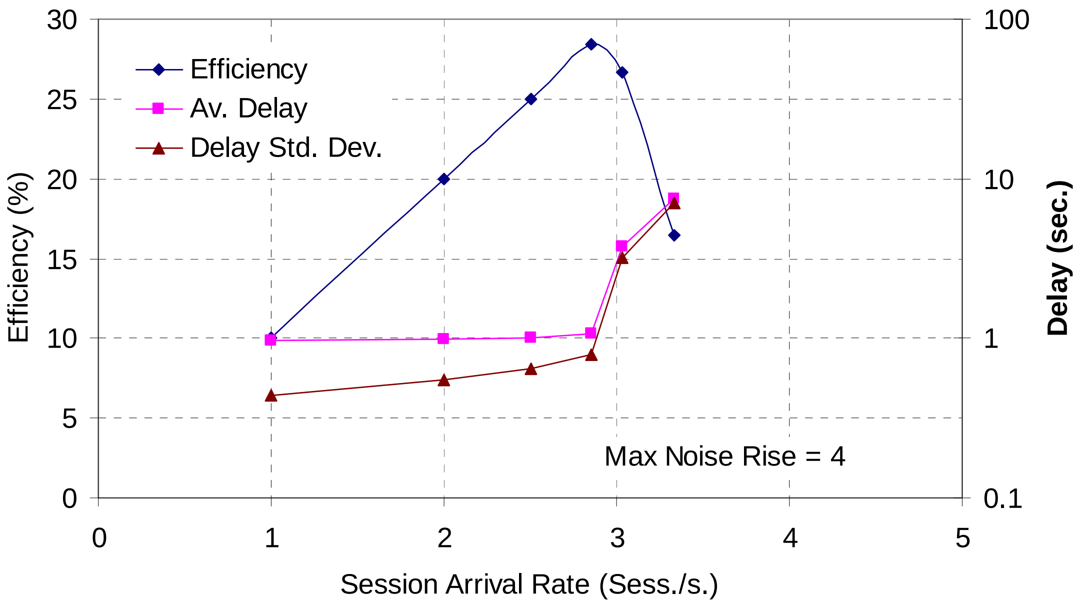


Figure 43: Histogram of total Rx signal power at the satellite receiver input in the same conditions as in Figure 41 and traffic load corresponding to a session arrival rate of 2,5 sess/s



NOTE: Attenuation in the shadows state is lognormal with 8 dB average and 4 dB variance. User speed is 70 km/h. The same scheduler as in Figure 41 was used.

Figure 44: System simulator results in a suburban environment with SNIR measurement error standard dev. equal to 0,5 dB and power control loop bias variance equal to 4 dB

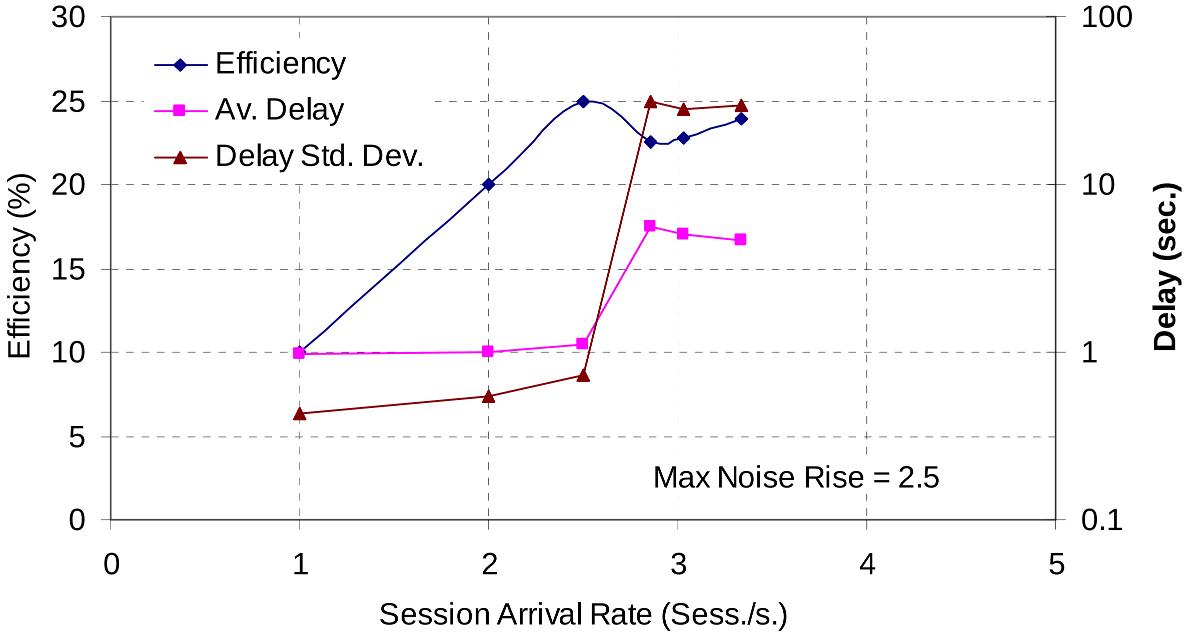


Figure 45: Simulation results in the same conditions as in previous figure but with lower maximum acceptable noise rise

A.2.5 Trade-off between alternative solutions

Simulation between three packet access alternatives were performed for the reverse link. It appeared that the CPCH-like alternative was less performing of both the spread Aloha and dRoD alternatives in a GEO test case. However, several variants are possible in the CPCH implementations which could improve its performance. We do not expect however, that the possible improvements could be such to exceed the performances of the other two access schemes. Regarding the relative performances of dRoD and spread Aloha, the conclusion of our work was that dRoD is better performing in highly loaded system whilst spread Aloha has better latency property in less loaded conditions. The best approach is thus the combination of the two schemes with an adaptive strategy favouring using spread Aloha for smaller segment length and dRoD for longer segments.

We recommend therefore such an hybrid scheme. In particular, a resource control strategy may be used in which the spread Aloha access parameters (persistence, back-off, max segment size) can be tuned according to the current system load. The parameters can then be chosen in order that spread Aloha is actually disabled (apart for request packets) when the system load exceeds a critical threshold.

Annex B: Satellite diversity assessment in packet access simulations

To assess the effectiveness of forward link diversity, the case of double satellite diversity will be first considered with the assumption that a clear path for both satellites is available. Fading will only be assumed Ricean with a C/M ratio of 10 dB. This assumption is somewhat pessimistic for relatively high satellite elevation angles which typically have C/M of at least 15 dB; anyway for lower elevation angles, or in presence of some tree shadowing, values as low as 5 dB to 7 dB may also be observed.

Simulations have thus been performed to compare performance of single diversity with double diversity in presence of signal unbalance between the two satellite paths with the scope of evaluating the best strategy from the point of view of the minimization of the required on-board power (for a given traffic load) or, equivalently, of the maximization of the capacity given the on-board available power.

Conventional double diversity, in fact, requires that for each user channel, two carriers are transmitted by the GW on two visible satellites. These two carriers will consume an equal amount of on board RF power on each of the two satellites. The number of transmitted carriers is thus the double with respect to a situation of single diversity where the best path is chosen (due to delay reason, the chosen path is the best on the average and not necessarily instantaneously). Hence if the RF power used by each of the two carriers corresponding to a user channel cannot be reduced by at least 3 dB with respect to the RF power consumed in the case of best path selection, then (conventional) double diversity would actually deteriorate the performance.

It shall be stressed that diversity may also increase asynchronous interference due to the increase in the carrier number. However, in case of satellite application where on-board power is fixed independently from the number of carriers, asynchronous interference level is actually independent from the diversity strategy (this may not be the case in terrestrial application where BS power limitations are often not relevant).

The above considerations have been formalized below where an estimation of the relative achievable capacity with single and double diversity (with equal path S/N ratio) is performed.

Let us assume that the same area is covered by two different satellites carrying a total of N_c^S channels. For simplicity when speaking of the satellites we will we will always implicitly refer to a single beam of the satellites.

Assuming each satellite carries half of that channels and that the available RF power in each satellite (PSAT) is equally divided between all carriers, then the ratio *C/I* is:

$$C / I = 2 / N_c^S$$

The assumption of using orthogonal codes within each satellite was taken. The above relation is justified by the fact that all signals are assumed having the same level and that the wanted carrier has $N_c^S/2$ interfering carriers (those from the other satellite). Taking into account that $C = E_b R_b$ and that $I = I_0 R_c$ where R_b and R_c are respectively the bit rate and chip rate we may write:

$$\frac{E_b}{N_0 + I_0} = \frac{\frac{E_b}{N_0}}{1 + \frac{E_b}{N_0}\frac{R_b}{R_c}\frac{N_c^S}{2}}$$

Given a required $\left[\mathrm{E_b} /(N_0 + I_0)\right]_S$ and taking into account that $E_b R_b\, N_c^S/2$ for a given satellite can be assumed constant and equal to maximum RF power received from that satellite, C_T, then the number of channels available in the system can be determined by solving the equation below:

$$\left|\frac{E_b}{N_0+I_0}\right|_S = \frac{\dfrac{2C_T}{N_0 R_b N_c^S}}{1+\dfrac{C_T}{N_0 R_c}}$$

It is:

$$N_c^S = 2\frac{R_c}{R_b}\frac{1}{\left|\dfrac{E_b}{N_0+I_0}\right|_S}\frac{1}{\left(1+\dfrac{N_0 R_c}{C_T}\right)}$$

In case instead, each carrier is transmitted in double satellite diversity, every satellite actually carries the double of carriers with respect to before. Following the same approach as before we have now:

$$N_c^D = \frac{R_c}{R_b}\frac{1}{\left|\dfrac{E_b}{N_0+I_0}\right|_D}\frac{1}{\left(1+\dfrac{N_0 R_c}{C_T}\right)}$$

where now $[\mathrm{E_b}/(N_0+I_0)]_D$ is the required SNIR value with double diversity for each single path. The ratio of carriers carried in single and double diversity is thus:

$$\frac{N_c^S}{N_c^D} = 2\left[\frac{[\mathrm{E_b}/(N_0+I_0)]_D}{[\mathrm{E_b}/(N_0+I_0)]_S}\right]$$

Double diversity will thus provide a capacity gain provided that it reduce the required $\mathrm{E_b}/(N_0+I_0)$ per path by at least 3 dB with respect to the single diversity case.

This is actually the case being the minimum possible gain when maximal ratio combining two equal S/N signal.

The analysis can be generalized for the case of diversity M as follows.

Obviously with M satellite in visibility the interference level is now (M-1) times larger as before. Hence if no multiple diversity is exploited we have:

$$\left|\frac{E_b}{N_0+I_0}\right|_S = \frac{\dfrac{MC_T}{N_0 R_b N_c^S}}{1+\dfrac{(M-1)C_T}{N_0 R_c}}$$

In case instead multiple diversity of degree M is exploited:

$$\left|\frac{E_b}{N_0+I_0}\right|_M = \frac{\dfrac{C_T}{N_0 R_b N_c^M}}{1+\dfrac{(M-1)C_T}{N_0 R_c}}$$

A gain is still available for the same reason why double diversity is advantageous with respect to single diversity.

The above results assume that full frequency reuse is exploited. This is a sensible choice when bandwidth is the system bottleneck. However, in cases the on-board power is the bottleneck, it may result convenient to reduce the frequency reuse factor if this may increase the overall capacity given the available on-board power.

For such cases let us assume that we divide the number of M visible satellites in F groups each operating on a different frequency band. Then we have:

$$\left|\frac{E_b}{N_0+I_0}\right|_M^F=\frac{\dfrac{FC_T}{N_0R_bN_c^{FM}}}{1+\dfrac{(M/F-1)C_T}{N_0R_c}}$$

The capacity ratio is then

$$\frac{N_c^{FM}}{N_c^M}=F\left[\frac{\left[\mathrm{E_b}/(N_0+I_0)\right]_M}{\left[\mathrm{E_b}/(N_0+I_0)\right]_M^F}\right]\frac{1+\dfrac{(M-1)C_T}{N_0R_c}}{1+\dfrac{(M/F-1)C_T}{N_0R_c}}$$

The factor F is fully compensated by the larger combining gain in the case of full frequency reuse gain. Such a gain ratio is in fact at worst equal to F. In a slowly fading channel, the combining process also provides significant additional diversity gain which may partly offset the increase of interference as apparent from the last multiplying ratio in the above equation.

The comparison above is not fair because bandwidth is not the same. To make the same bandwidth let us assume that even in case of full frequency reuse we use F frequency slots. In such case, because the on-board power is limited, the power per frequency slot is now F times lower.

The above equation would then become:

$$\frac{N_c^{FM}}{N_c^M}=F\left[\frac{\left[\mathrm{E_b}/(N_0+I_0)\right]_M}{\left[\mathrm{E_b}/(N_0+I_0)\right]_M^F}\right]\frac{1+\dfrac{(M-1)C_T}{N_0R_cF}}{1+\dfrac{(M/F-1)C_T}{N_0R_c}}$$

where in the above equation the value N_c^M has been increased by a factor of F to take into account the advantage from frequency reuse.

Assuming that no frequency reuse is done ($F = M$), the above equation is about:

$$\frac{N_c^{MM}}{N_c^M}\cong F\left[\frac{\left[\mathrm{E_b}/(N_0+I_0)\right]_M}{\left[\mathrm{E_b}/(N_0+I_0)\right]_M^M}\right]\left(1+\frac{C_T}{N_0R_c}\right)$$

In case of Gaussian channel, where no diversity gain is available it is always $N_c^{MM} > N_c^M$ (i.e. the case of no diversity/no frequency reuse could be preferable) with the difference between the two increasing with the available on board power (Figure 46). In such a situation using full frequency reuse is only negligibly penalizing when the on-board power is small. However, penalization grows to very high level for higher level of available on-board power.

For fading channels, where a diversity gain is available, a capacity advantage might be available if diversity is exploited, particularly in presence of slow fading where the diversity gain may be higher.

This conclusion is supported by Figure 47 which shows the factor of merit (i.e. the capacity ratio between the case without frequency reuse and the case with full frequency reuse and diversity exploited with diversity order equal to the number of available satellites) for different hypothesis of the parameter C_T/N_oR_c which is actually dependent on the on-board available power.

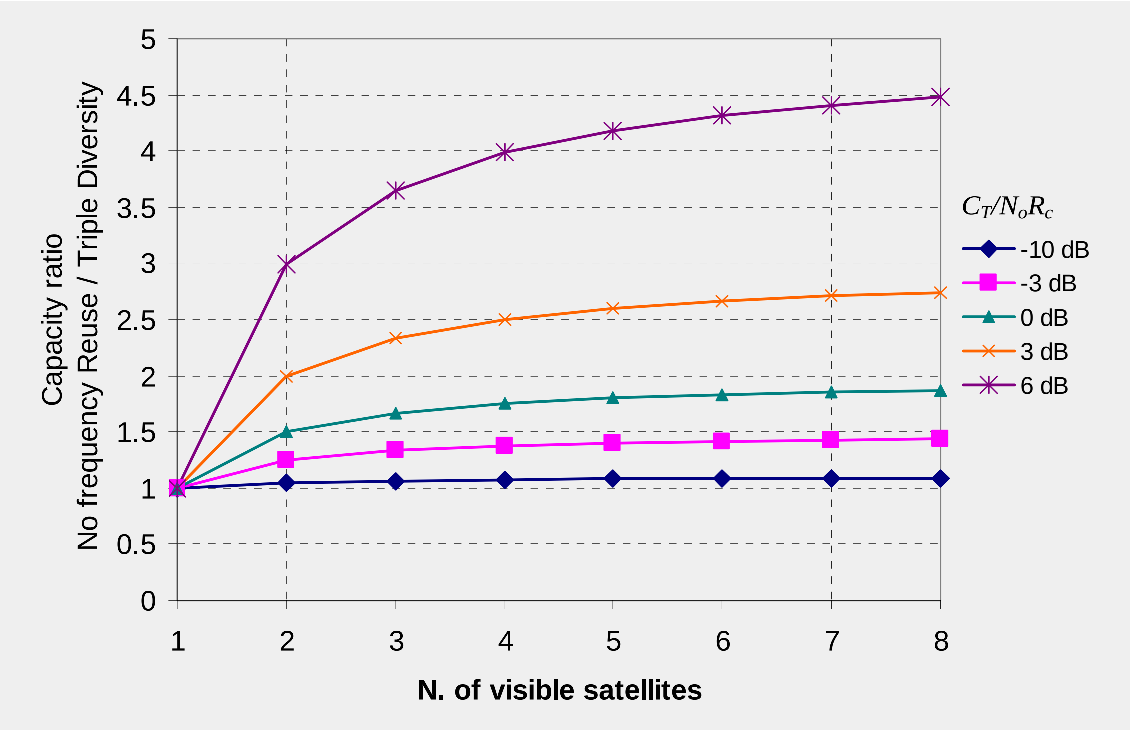


Figure 46: Capacity ratio (No frequency reuse/full diversity) versus the number of visible satellites in an AWGN channel (no diversity gain)

Figure 46 was drawn by assuming a diversity gain as reported in Table 6. Such a table approximately reflects the case of a slow Ricean fading with C / M = 10 dB.

Table 6: Diversity gain

Visible satellites	Diversity Gain (dB)
1	-
2	3
3	4
4	4,5
5	4,75
6	4,85
7	4,85
8	4,85

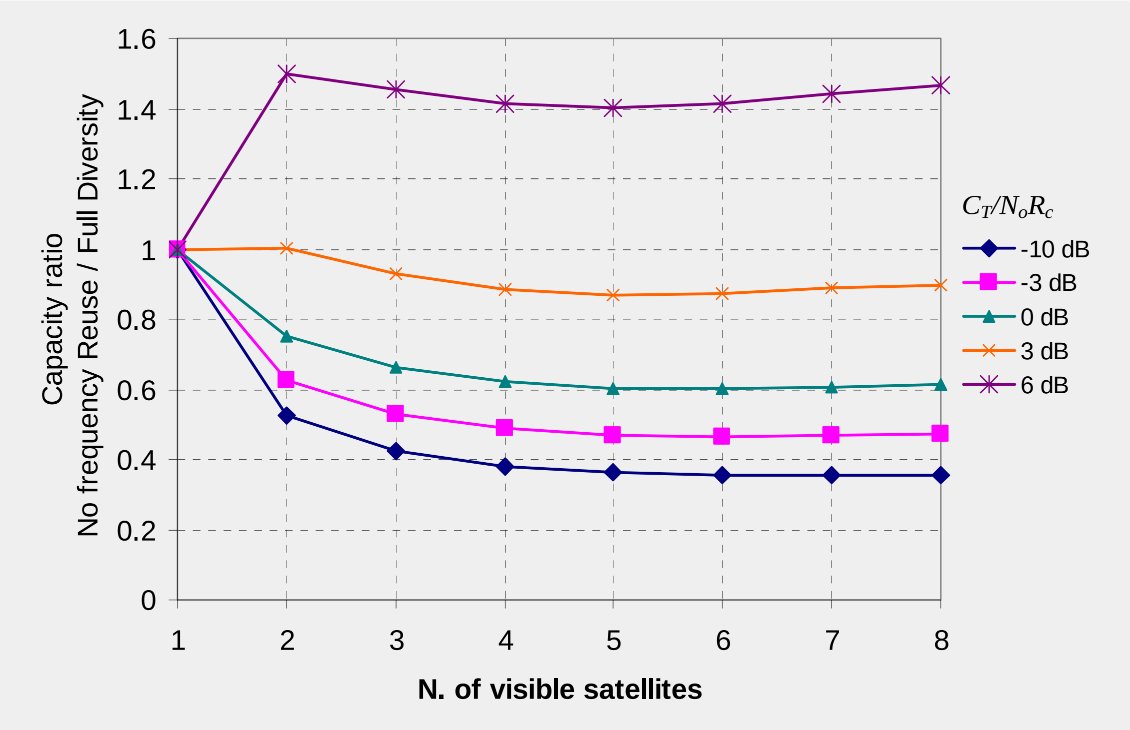


Figure 47: Capacity ratio (No frequency reuse/full diversity) versus the number of visible satellites

The results in Figure 47 assume that the user receiver is able to exploit all diversity path. This implies a very complex receiver as the number of visible satellites increases. Hence, in Figure 48 we have redrawn the comparison of Figure 47 by assuming that diversity may be at most triple.

Anyway, we recall that the results in the figure do not take into account the user blockage phenomenon which is an important consideration in a mobile environment (see for example figure 1 in [6]).

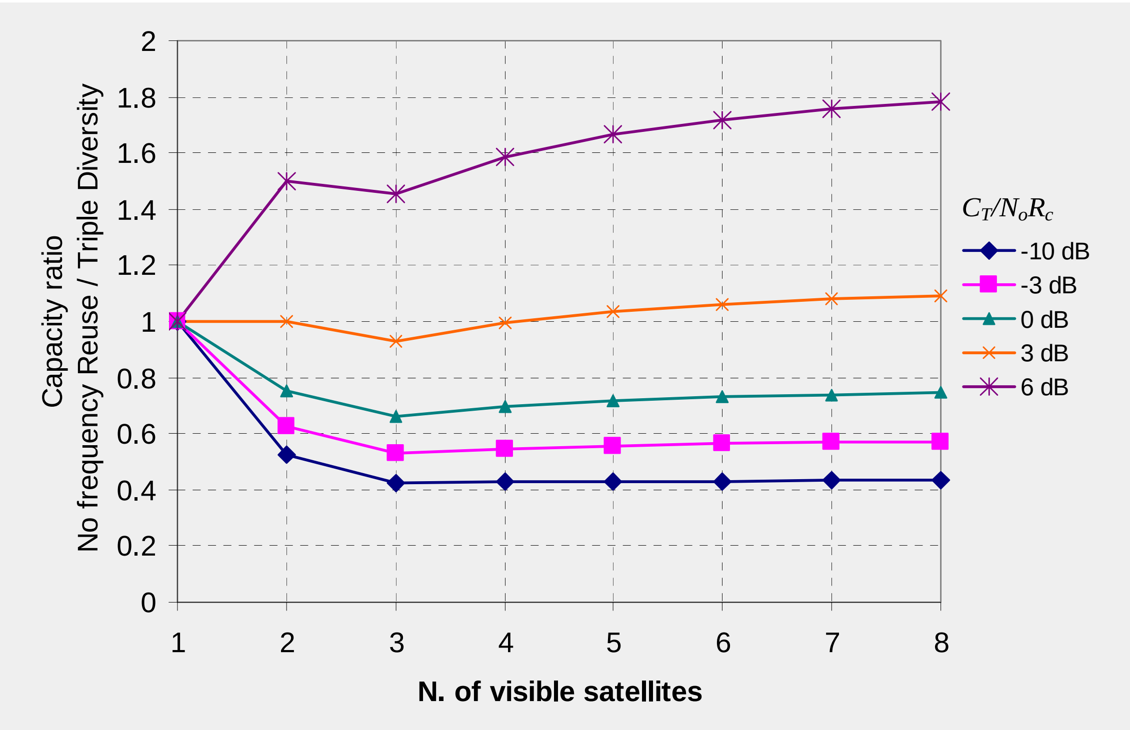


Figure 48: Capacity ratio as in previous figure but exploiting at most triple diversity

B.1 Simulation results

The analysis above shows that in case the two co-frequency satellites are received with the same strength, double diversity would always be better or equal to single diversity.

The result above has been analytically demonstrated but fully agrees with the intuition. In fact the interference and thermal noise level seen by a user, is a constant (assuming satellite transmit at constant power) independently from the diversity degree. With double diversity, anyway only half the power with respect to the single diversity is available on each path. This justifies the ratio 2 for the obtained $E_b / (N_o + I_o)$ on the two cases.

Clearly the obtained result cannot be extended to the case of unequal path attenuation. In such a case the simulation results shown below can be used to predict the best strategy.

The figure below show the performance comparison between single and double diversity for three different assumptions of user dynamic (corresponding to Doppler spread of 6 Hz, 24 Hz and 100 Hz) and different hypothesis on the S/N unbalance for the two paths in the case of double satellite diversity.

Obviously to perform a fair comparison it shall be taken into account that the following figure only shows (for double diversity) the required $E_b / (N_o + I_o)$ per satellite path (the first path for the unbalanced cases). Hence, for comparing power efficiency, the curves for the double diversity cases should be moved to the left by 3 dB.

Also with that correction it can be noted that double diversity is always advantageous with respect to single diversity with balanced links. With strongly unbalanced links, double diversity is instead penalizing with respect to a single diversity case where the best link is chosen. The threshold between the two regions is depending from the fading characteristics. For very slow fading, i.e. the 6 Hz Doppler spread, the unbalance threshold is about 6 dB as can be seen Figure 49. For Doppler spread of 24 Hz and 100 Hz the thresholds become instead slightly higher than 4 dB and slightly lower than 2 dB respectively (see Figure 50 and Figure 51).

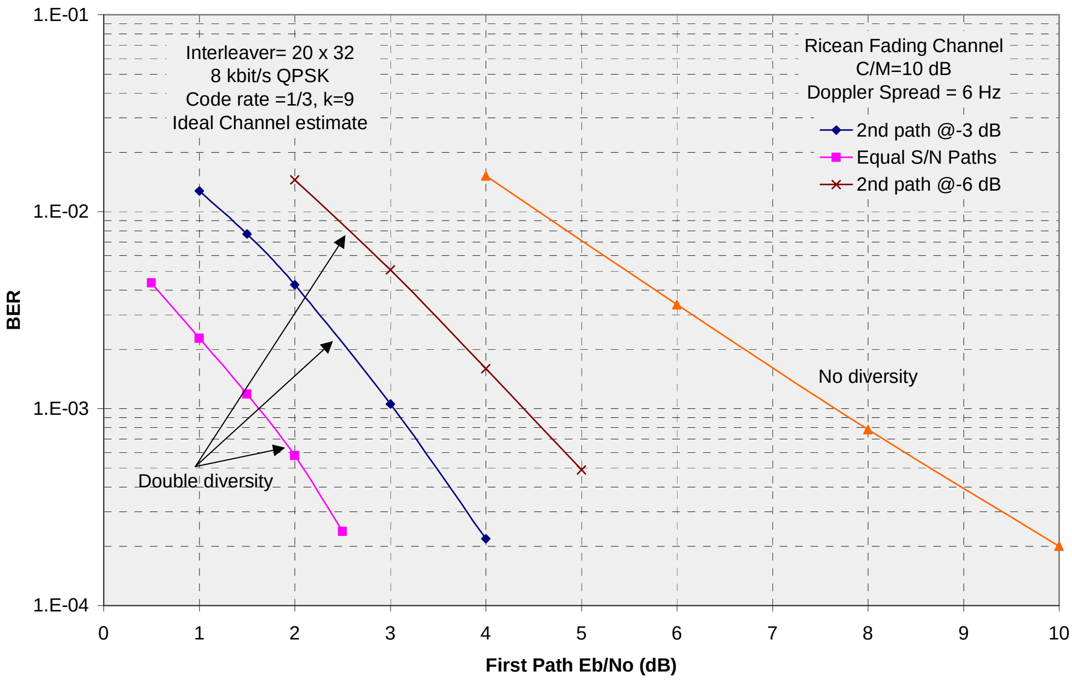


Figure 49: Comparison of single diversity and double diversity performance for Doppler Spread = 6 Hz

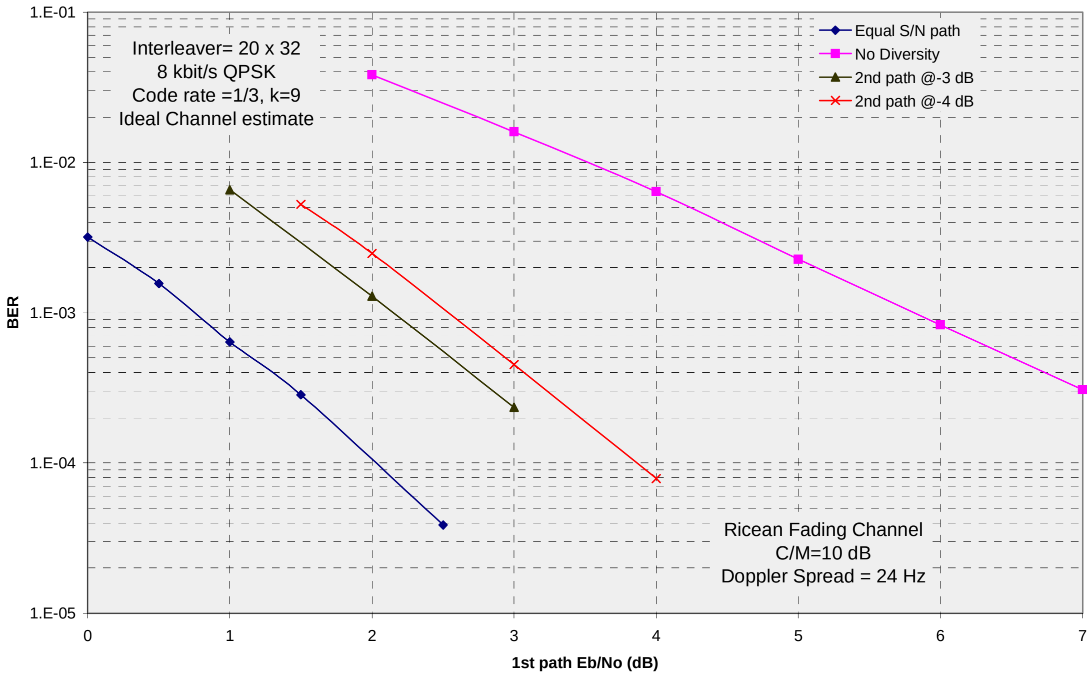


Figure 50: Comparison of single diversity and double diversity performance for Doppler Spread = 24 Hz

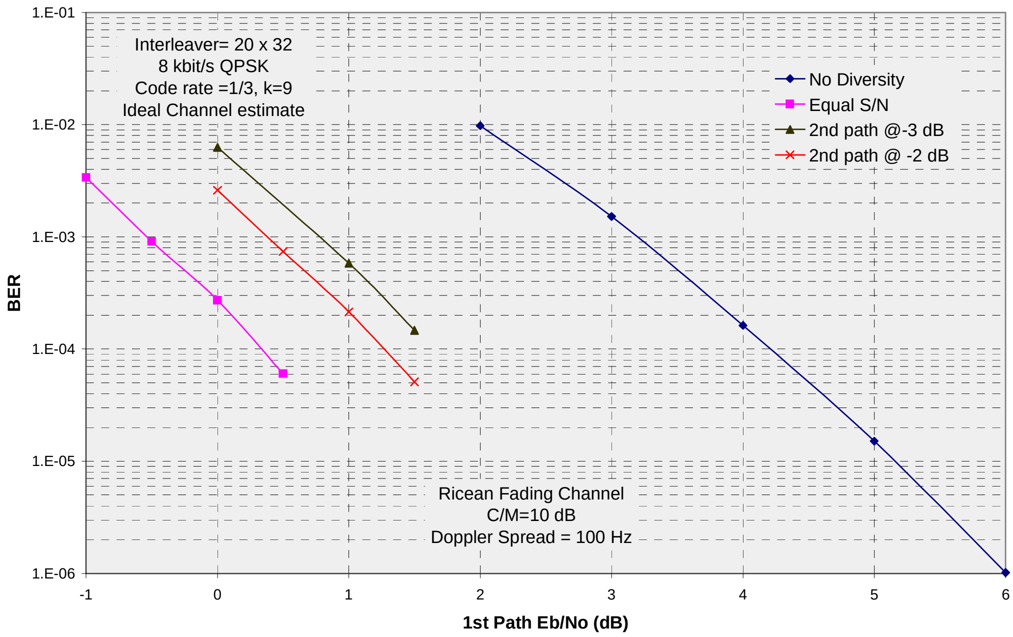


Figure 51: Comparison of single diversity and double diversity performance for Doppler Spread = 100 Hz

Annex C: Web traffic model in packet access simulations

It is well known that the Poisson distribution is not a good model for packet traffic. A more realistic traffic model was thus simulated to verify the sensitivity of random access to traffic model. In particular, the ETSI model for Web server traffic was implemented and tested. According to such model, new packet sessions are generated according to a Poisson distribution. However, within a packet session, the actual packet generation is not governed by the Poisson distribution. In our simulation program a new Mobile Station (MT) is generated and maintained alive for all the time the packet session is active. The full traffic model is described in detail in both [1] and [5].

Parameters of the model as suggested by ETSI in the Web browsing scenario were selected with the only exception of the *Reading Time* where a lower value for the average value parameter was selected to reduce the number of simultaneously open packet session. This was done in order to reduce the memory consumption of the simulation program. It shall also be stressed that the ETSI model is actually more appropriate as web **server** traffic generator and not as a web **client** traffic generator. This is reflected in the average packet length (see model parameters below) which is too high for a client scenario were packets mostly contain ACK and page requests. The model would in principle not be fully suitable for reverse link simulations were most of terminals are expected to be web clients and not servers. We used anyway said model also for reverse link because it is more stressing as far as the system performances are concerned. At this regard it is expected that the reduction of the reading time we operated with respect to the ETSI model is also conservative at least as far as the impact on the packet delay which will results from simulations.

The following parameters were used for the simulations:

- Number of packets in a packet Call: geometrical distributed with average value equal to 25.
- Number of packet calls in a packet session: geometrical distributed with average value equal to 5.
- Average packet gap in a packet call: geometrical distribution with average value equal to 125 ms.
- Average packet length: Pareto distributed with average of 480 bytes (max length is 66 666 bytes).
- Reading Time: geometrical distributed with average value equal to 40 s or 10 s.

Annex D: Channel model

A two states channel model is proposed for the system simulation work having a GOOD and BAD state. Two-state models were proposed by a number of authors. Most notably are Lutz and its co-workers [2] which performed several land mobile propagation measurement under ESA sponsorship.

In Figure 52 a picture of the channel model is shown [2].

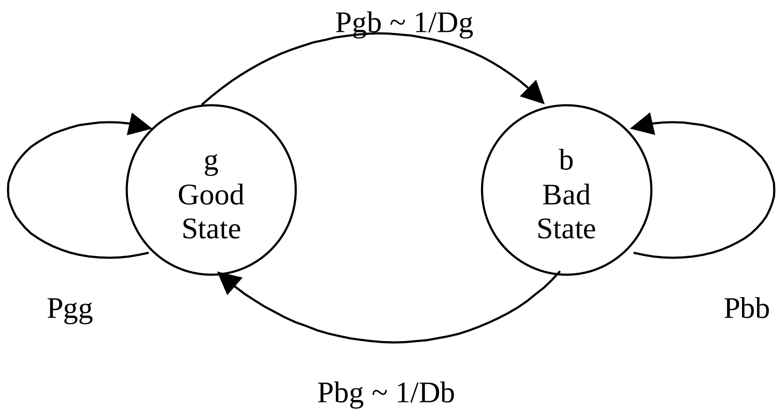


Figure 52: Two State Markov Model for the good and bad channel states of the land mobile satellite channel [2]

In the good state we assume no error occur and in the bad state we assume full errors occur. It is difficult to include errors in the good state as the turbo code and convolutional codes used in the physical layer receiver will produce bursty errors which are difficult to reproduce without implementing the full 3GPP chain and performing a full 3GPP simulation.

According to this kind of models the BAD state would correspond to shadowing or blockage of the satellite. The GOOD state would instead represent a state with direct line of sight with the satellite.

In the GOOD state a Ricean fading is typically assumed. The Ricean fading C/M will be depending on the elevation angle and typically range from about 7 dB to 15 dB with almost omnidirectional antennas. In the BAD state a lognormal fading is instead considered. The corresponding model is shown in Figure 53.

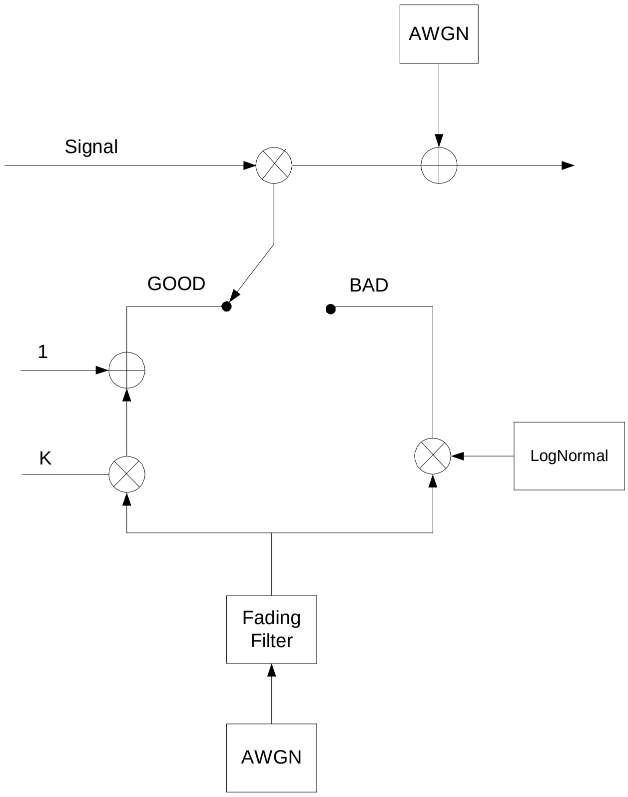


Figure 53: Two-states channel model

According to the measurement in [2] the characteristics of the lognormal fading in the BAD state depends on the environment. For the urban environment the average fading is 12,5 dB whilst for the rural/highway environment it is about 8,5 dB. The variances of the fading are about 4,5 dB in both cases. As far as the use of this model in our simulations, we have also introduced a time correlation between successive samples of the lognormal fading. In particular, a simple linear interpolator has been used to generate lognormal fading between time instants separated by the assumed correlation time (samples whose time distance are larger than the correlation time are assumed independent). The assumed correlation time considered in system simulations was 1 second. In practice in such system simulations no multipath fading was explicitly simulated but only the shadowing process according to the illustrated model. In particular, in the GOOD state, no attenuation was considered being the Ricean fading already taken into account through the required SNIR used in the simulations. Hence only the lognormal fading in the BAD state was explicitly simulated.

For the statistics related to state switching, i.e. to the availability or not of at least one satellite line of sight we can assume that satellite paths blockage probabilities are approximately independent. Making reference to paper in [3], single satellite path blockage probability, p_B, can be assumed to be a function of only the satellite elevation, ε:

$$p_B(\varepsilon) = \frac{1}{a}(90 - \varepsilon)^2$$

According to the measurements in [3], the normalization factor, *a*, is:

- a = 7 000 in urban area;
- a = 16 600 in suburban area.

We also assumed a slightly higher value for the parameter *a* for the rural environment (a = 24 000). To compute the transition probability between BAD and GOOD state, p_{bg}, the average distance *D*b (ε) in meters to exit from the BAD state can be introduced where:

$$D_b(\varepsilon) = \frac{1}{(1 - p_B(\varepsilon))^{\gamma}} \cdot \beta$$

where β is a scaling factor taken equal to 10 and γ is assumed equal to 1,1 for urban area and 3,2 for rural/highway areas.

The transition probabilities p_{bg} and p_{bb} are then easily computed taking into account the user speed, *v*, and data rate, *R* (we have implicitly assumed here a time unit equal to the bit period). In fact, the average number of time steps to exit from the BAD state is:

$$m_B = \sum_{k=0}^{\infty} k \cdot p_k = \sum_{k=0}^{\infty} k \cdot (1 - p_{bb}) p_{bb}^k = \frac{p_{bb}}{1 - p_{bb}}$$

where p_k is the probability to get out of the BAD state exactly after *k* time steps.

Because it is $D_b(\varepsilon) = m_B\, v/R$, then it follows:

$$p_{bg} = \frac{v}{v + RD_b(\varepsilon)} \qquad p_{bb} = 1 - p_{bg}$$

It is also easy to compute higher order statistics about the time to exit from the BAD state. As far as the variance to get out of the BAD state it is:

$$\sigma_B^2 = \sum_{k=0}^{\infty} (k - m_B)^2 p_k = \sum_{k=0}^{\infty} k^2 \cdot (1 - p_{bb}) p_{bb}^k - m_B^2 = \frac{p_{bb}}{(1 - p_{bb})^2}$$

From above it also appears that the probability to still remain in the BAD state after *r* time step, $P_B(r)$, is:

$$P_B(r) = \sum_{k=0}^{r} (1 - p_{bb}) p_{bb}^k = 1 - p_{bb}^{r+1}$$

The standard deviation of the distances in the bad state, assuming the above Markovian model, can be shown to be practically equal to the average distance.

The other transition probabilities, p_{gb} and p_{gg}, can be easily computed taking into account that the average distance *D*g (ε) in meters to exit from the GOOD state shall be equal to:

$$D_g(\varepsilon) = D_b(\varepsilon) \frac{1 - p_B(\varepsilon)}{p_B(\varepsilon)}$$

As already mentioned, in the GOOD state a Ricean fading is assumed. The Ricean fading C/M will be depending on the elevation angle. Anyway, the Ricean fading component will not be explicitly simulated in the system simulator but its effects will be taken into account by changing the required E_b/N_o.

D.1 Channel characteristics

In this clause we discuss and show the characteristics of the channel model. This is important as the design of the best receiver/decoder technique must be based on information about the channel characteristics. The channel conditions are taken from [2] where he has published two channels, one for urban and one for highway environments. Both models have been validated with channel measurements.

D.1.1 Lutz urban channel model

The urban area channel has bad and good mean distances as shown in Figure 54. This result is based on measurements and is a linear relationship. The model shows that the bad channel mean distance is larger than the good channel one for elevation angles under about 40°, while above this elevation the good distance is larger than the bad distance.

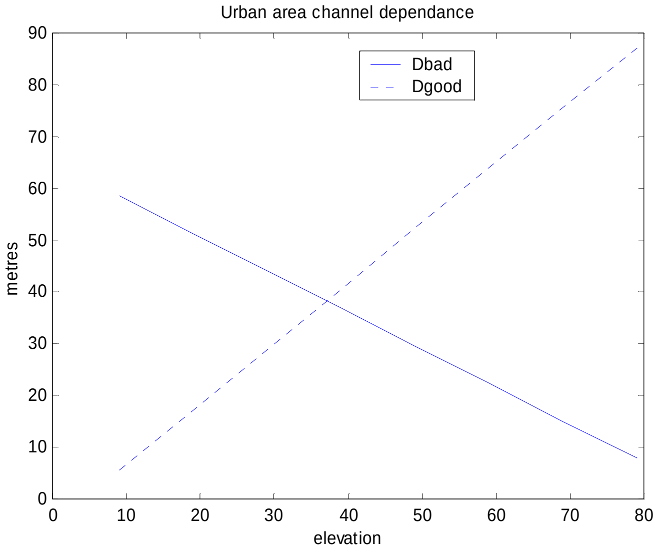


Figure 54: Lutz Urban Channel Model

This model can be represented as good and bad states as discussed before. These good and bad states can be converted into good and bad bits or frames if the speed of the vehicle is known and the data rate is known. A histogram of the frequency of bad frames (RS frames of 2 040 bits) vs. the size of the bad frames can be plotted to understand the characteristics of the channel.

In Figure 55 the linear plot of the histogram for the worst case elevation (9°) is plotted for the bad frames and their frequency. The data rate was set to 256 kbps and the speed of the vehicle was 3 km/h and 70 km/h. As expected the number of outages is significantly less for the higher speed. What is also observable is that the frequency of the outages is exponential, where the most outages occur for smaller frames (in this case for 3 km/h, less than 100 000 frames).

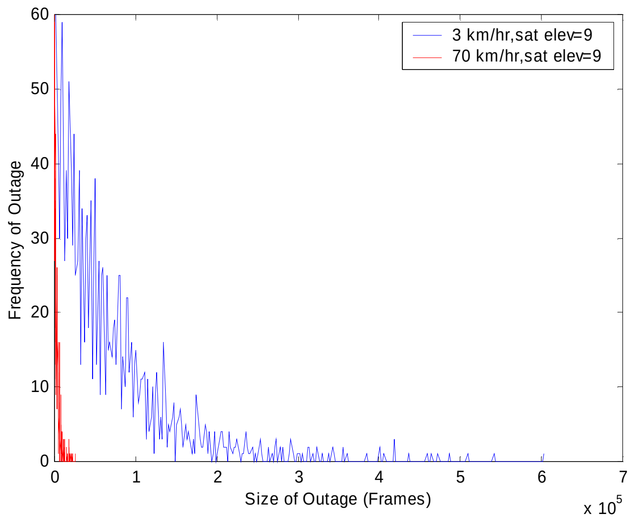


Figure 55: Frequency of Outages vs. Frame Outage size (in frames)

To get another view of these outages we plot the same data in Figure 55 on a log scale on both axis, we also include the statistics of the outages for the maximum elevation angle of 79°. This is shown in Figure 56. The results show very linear like characteristics, confirming the exponential assumption previously made. The results for 79° now show significant reduction in the outage size, which is approximately one order of magnitude.

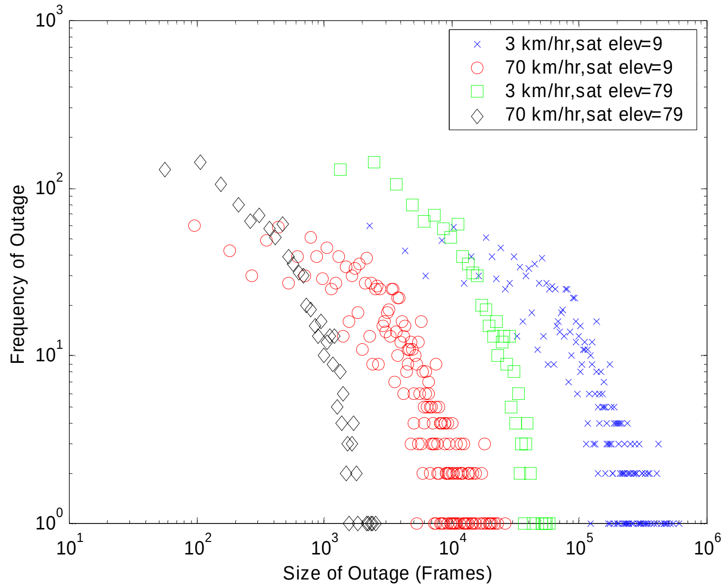


Figure 56: Frequency of Outage vs. Size of Outage (in Frames)

D.1.2 Lutz highway channel model

The Lutz highway channel model is shown in Figure 57. The numbers of metres in a bad channel and good channel are marked over a satellite elevation angle between sixteen (16)° and 55°. Note that the scale of metres is a logarithmic scale, unlike the previous channel, therefore after an elevation angle of 25° this channel is a very good channel with very little outages.

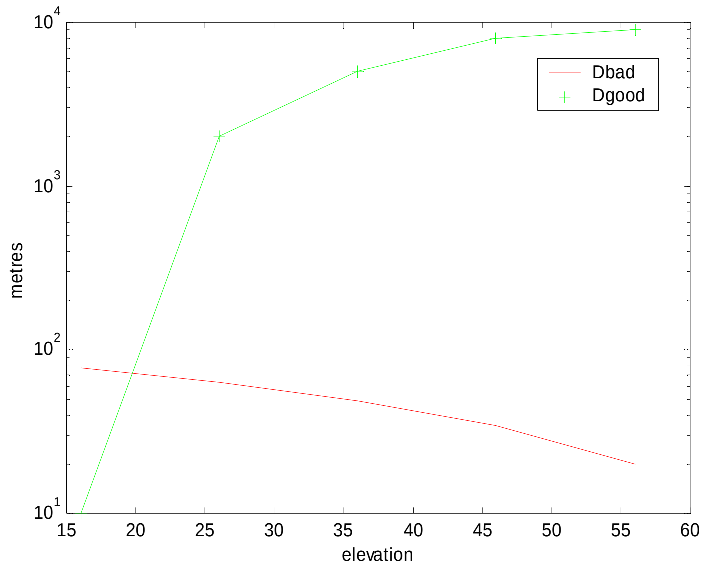


Figure 57: Lutz bad/good channel model for the highway

In Figure 58 we plot the same log distributions as plotted for the Urban channel. Note in this case that the frequency of the outages for the 56° elevation case are only ever equal to one, and the size of these outages is sometimes two orders of magnitude less than the size for an elevation angle of 16°. This reduction in outage frequency and outage size is to be expected for the channel characteristics shown before.

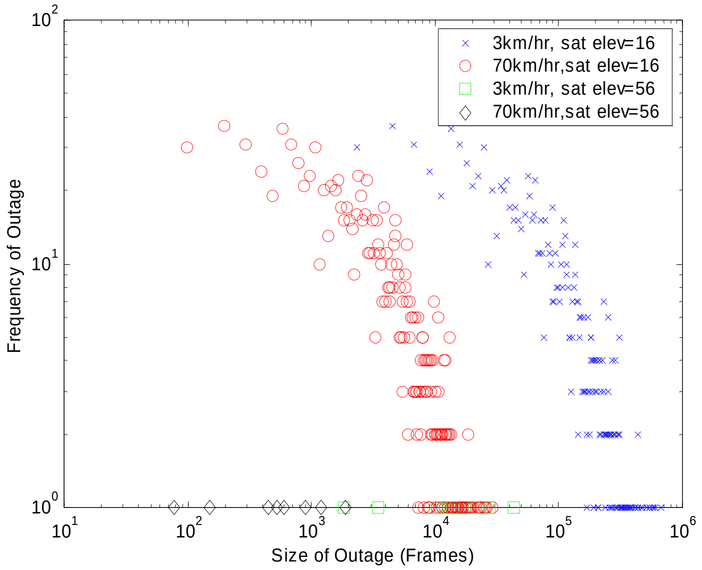


Figure 58: Frequency of Outage vs. Size of Outage (frames) for Highway model

Annex E: Investigation into multicasting and narrowcasting for S-UMTS

E.1 Assessment and optimization of S-UMTS for multicasting in SATB

This clause describes analysis to determine the requirements for multicasting in SATB, using S-UMTS as the physical layer. The aim is to determine key system parameters through numerical simulations for multicasting in SATB. These parameters include code rate, code size, interleaver size. A final system solution will be determined with supporting performance results.

E.1.1 Introduction

The purpose of this clause is to determine design parameters for reliable multicasting over S-UMTS. This task involves a large number of possible parameters and conditions that can occur in an S-UMTS system. To efficiently obtain results a number of system assumptions and channel model assumptions have been performed to simplify the task without compromising the results.

E.1.2 Objectives

The objective of reliable multicast using Satellite ATB is to provide a given QoS (packet error rate) over the satellite channel. Over wireline connections this is typically achieved by some form of automatic repeat request (ARQ) or Negative ARQ scheme. However over a satellite channel with many users the return channel does not have the required capacity and we can avoid a dedicated return link channel. For this reason the addition of forward error correction (FEC) and interleaving are used to specifically help.

The goal in this clause is to determine all the necessary parameters to determine the system requirements of the reliable multicast system.

E.1.3 System configuration

In Figure 59 a picture of the basic channel model is shown, here the data source is Reed Solomon (RS) encoded and interleaved before being sent to the physical layer transmitter. The modulated signal passes over the S-UMTS channel before reception through the physical layer receiver (which includes channel decoding). The data then is de-interleaved and RS decoded.

Possible extensions to this figure include carrousel transmission where the same data is automatically sent a number of times.

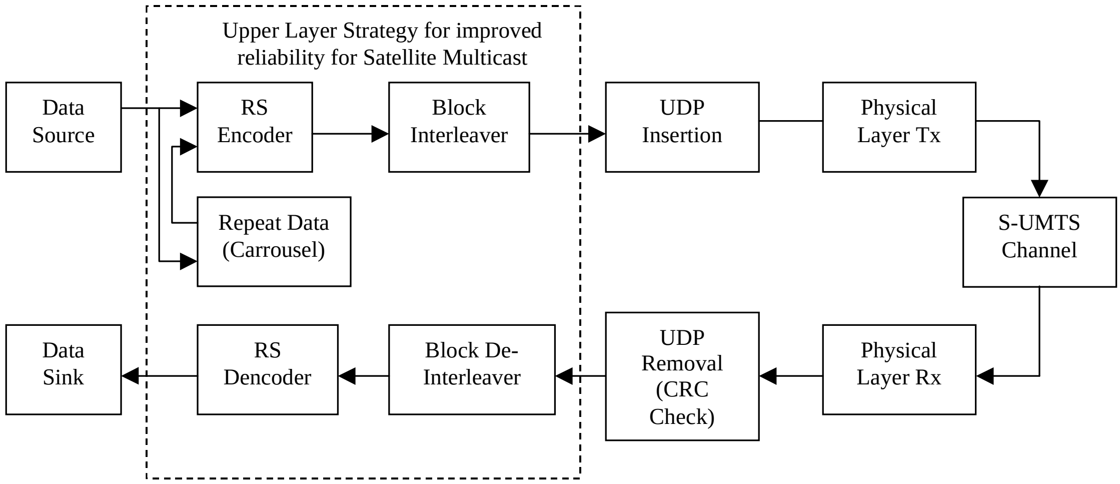


Figure 59: Block Diagram of Reliable Multicast with FEC

E.1.4 System assumptions

A number of assumptions have to be made to determine our system parameters. The following table lists the assumptions and provides a reason for this assumption.

Assumption	Reason
No ARQ is used	As stated in previous clauses of the present document the use of ARQ is seen as not workable due to the limited capacity of the system, An ARQ hybrid approach will be considered in a Narrowcast Study.
Channel is modelled as a two state Markov model with good and bad states of the land mobile satellite channel. When a connection is in a bad state then full errors are assumed.	This model is used extensively in the literature and there is measurement results to support the mean duration of the good and bad states, over elevation angle [2].
The system is "emulated" not fully implemented	Based on the known correction performance of the RS decoder and the channel outages the performance of the system can be found using emulation methods without implementing a highly complex full simulation.
Interleaver Type	Simple Block Interleaver, interleaver length determines how long the bad burst is.

E.1.5 Large block interleaving combined with Reed-Solomon FEC decoding

In this clause we investigate the performance of Reed-Solomon forward error correction (FEC) coding, along with the use of interleaving. The aim is to determine particular parameters for the implementation in the second phase of the ATB project. The other aim is to use these parameters to determine the performance (in terms of packet errors as a function of the satellite elevation angle.

E.1.5.1 System parameters to be determined

Table 7 shows the parameters to be determined in this clause.

Table 7: Parameters to be Determined

Parameter	Comment
RS Code Size	The code size determines the number of simultaneous errors that can be corrected.
RS Code Rate	The minimum code rate is determined when an infinite length interleaver is assumed and is therefore lower bounded by the probability of a "bad" state. The lower the code rate the better the performance of the code, however, this needs to be traded off against the overall throughput.
Interleaver size	The interleaver size determines the splitting of the bad sequences and indicates the maximum length "bad" sequence that can be de-interleaved. This size should be minimized as much as possible to minimize the memory space required by both the terminal and the gateway.

E.1.5.2 Important RS encoder/decoder characteristics

RS encoding takes a binary bit stream and encodes the data on the basis of symbols where there is B bits in a symbol. The RS frame then consists of N symbols, where K symbols contain the information data and N-K contain the parity data. This is illustrated in Figure 60. The maximum size of the RS encoded frame is 2^B-1 symbols and typically this is the value used. The rate of the code is determined by the number of information bits (K) in each encoded frame, where K can be between 1 and N-1.

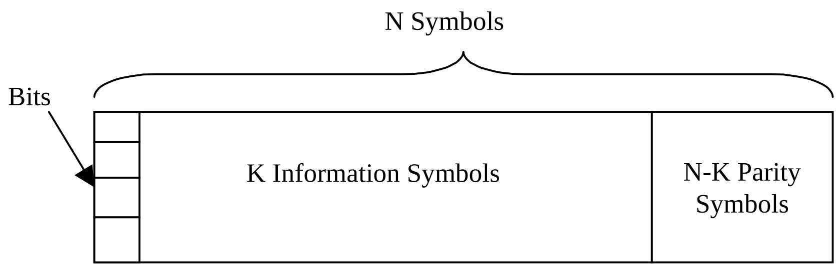


Figure 60: RS Encoded Frame

The maximum symbols that a RS decoder can correct in one frame is floor ((N - K + 1) / 2). If the position of the errors is known (i.e. erasure channel) then the RS decoder can correct N - K + 1 symbols.

Note that the RS decoder can correct any number of bit errors within the allowable symbols but if the bit errors are dispersed within the RS encoded block then they cannot be corrected. For example a RS encoded frame with 4 bits per symbol with a correct capability of 2 symbols can correct up to 8 bits as long as they fall only in two symbols. If these errors are spread throughout the RS frame then they cannot be corrected. i.e. RS decoding works well for burst errors.

The above point is very important for the design of interleavers for the RS decoding. This interleaver must have a minimum granularity of a RS symbol, not of a bit. If the bits were interleaved then the errors would be dispersed throughout the entire RS encoded frame and the RS decoder would fail the decoding process with a much higher probability.

E.1.5.3 Large block interleaver dimensioning

In this clause we investigate the use of large block interleavers in multicasting with two different Reed Solomon (RS) decoders. The purpose is to find the RS code rate and interleaver size, for a given channel performance. Additionally we show these results over a range of satellite elevation angles.

Reed Solomon N = 15

In this clause we assume the RS encoder is of size (N = 15,K), where we need to determine the K variable, and therefore the rate of the code. We also determine here the interleaver size based on this particular code.

From the channel model we now determine the K value for the RS code. If we assume an infinite length interleaver then the corrections that are needed is upper bounded by the number of bad states of the code. As the channel becomes worse (lower elevation) a lower code rate is needed to correct all the errors. This is exactly seen in Figure 61 where the code rate reduces for smaller angles. Here we plot the K value of the RS code for both an erasure channel and an error channel. It is also clear from these results that a larger K (higher rate) can be selected for the erasure channel than the channel with errors, based on the known RS correction capability for erasures and errors.

Note that we reduce the rate of the code in each case by a small tolerance value of 0,1. This is to allow a small difference between the lower bound on the RS rate and the actual rate of our code. This reduces the constraints on the interleaver design slightly.

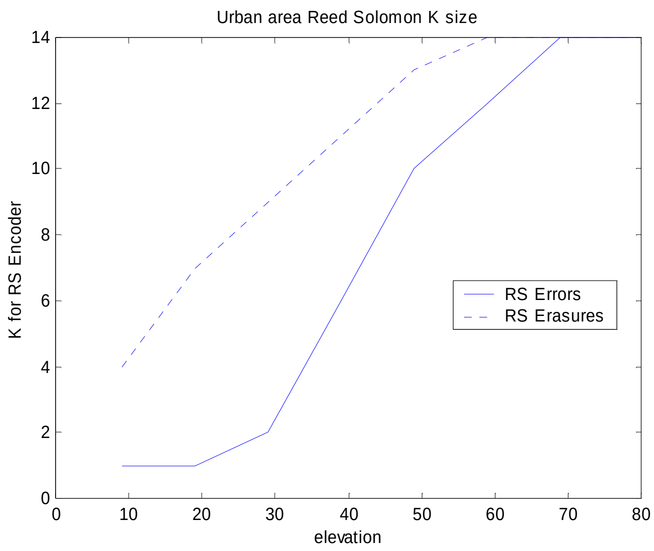


Figure 61: K Size of RS decoder (15, K) that is required for urban area environment

Based on knowing the average bad distance we can calibrate the results in terms of bits for a particular speed. In Figure 62 we can see the mean and Maximum "bad" bits generated by the channel as a function of the elevation angle. This is based on two vehicle speeds and a bit rate of 256 kbps, where the results are shown over elevation angle.

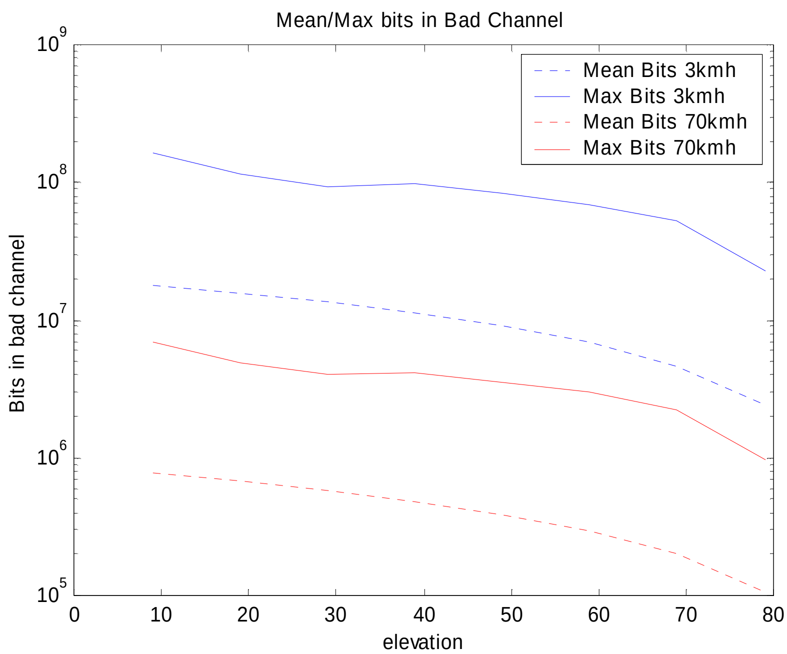


Figure 62: Mean and maximum bits in a bad channel over elevation

Based on this "bad" bit result we can compute the mean and maximum interleaver size, where the interleaver size dimension (int_size) refers to the number of code words in the interleaver. The size of the interleaver (int_size) is therefore equal to the "bad" bit size divided by the multiplication of the RS symbol size with the number of symbols that can be corrected in a RS frame. Note that the maximum interleaver size is determined from the maximum outage size that was experienced in this simulation run, it is plotted to give an idea for the difference between the mean and a possible maximum. As shown previously in the statistics of the outages, there is no fixed maximum size. This maximum size is used throughout this study to provide an idea of what this means for the system design and help with the interleaver and RS code dimensioning task.

We can now plot the required interleaver size for both the erasure channel and the error channel. For these simulations the rate of the RS code changes based on the elevation (as previously shown). The aim here is to therefore have an interleaver size that is roughly constant over all elevation angles. The results in Figure 63 show that to compensate or the mean error/erasure conditions the interleaver size is less than 5×10^4, while to cover all "bad" bursts an interleaver size of up to 10^7 is needed.

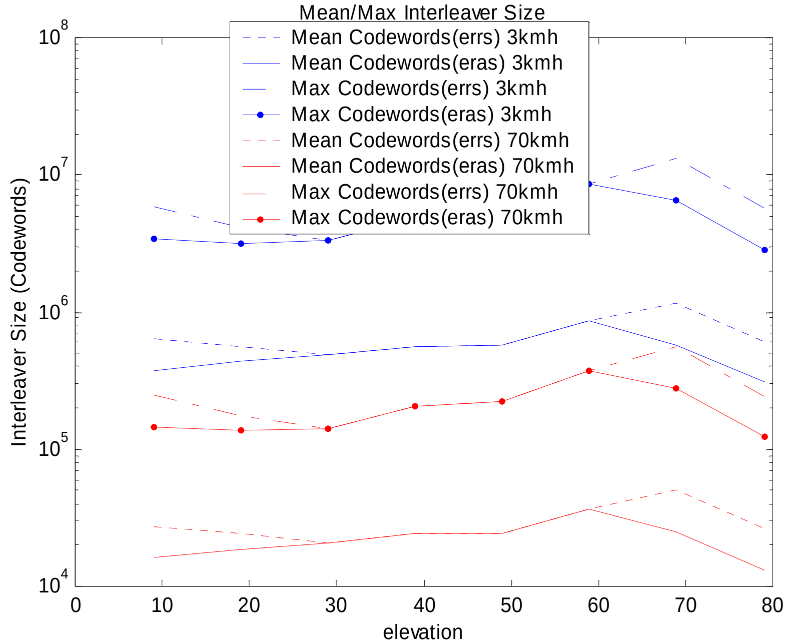


Figure 63: Mean and maximum interleaver size over elevation

Reed Solomon N = 255

In this clause we repeat the techniques discussed for the N = 15 RS code, but instead set N = 255. We see again in Figure 64 that the erasure channel can support a higher rate code for an equivalent elevation angle, as expected.

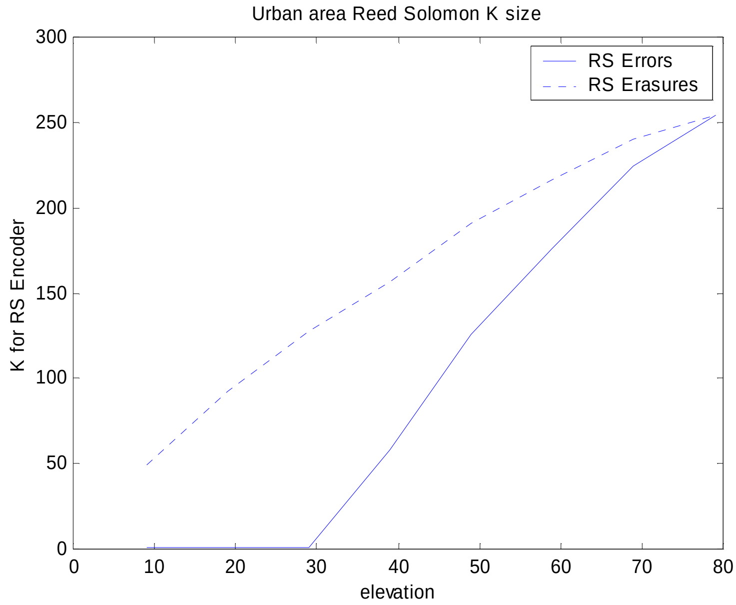


Figure 64: RS (255,K) K size required for urban area channel over elevation

The mean and maximum interleaver size is plotted in Figure 65. For 70 km/h the interleaver size is an order of magnitude smaller than that required for 3 km/h, where the interleaver size over elevation is relatively constant, noting that again the RS code rate is changing over elevation based on the plotted results for K in the previous figure.

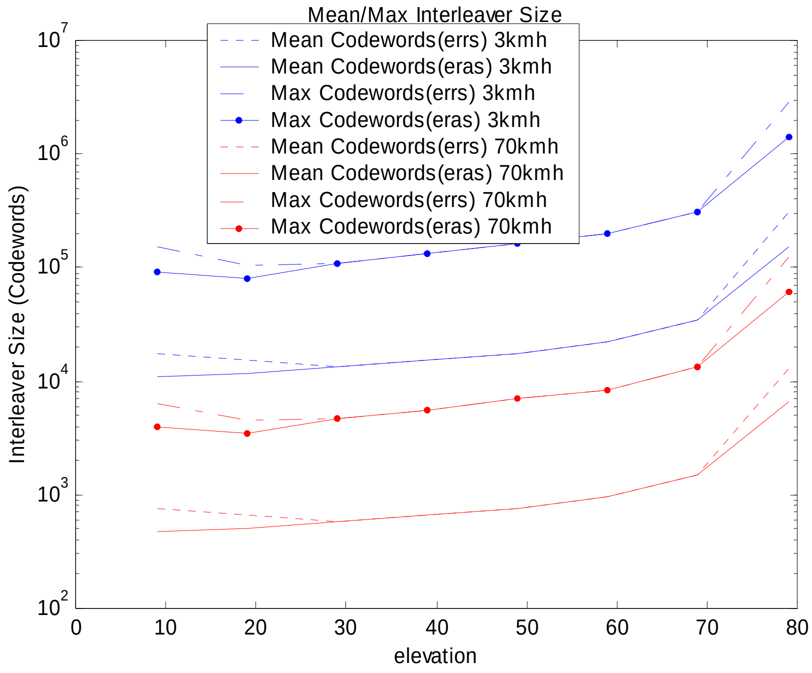


Figure 65: Mean and Maximum Interleaver Size over Elevation

E.1.5.4 Performance results (urban model)

In clause E.1.5.3 we plotted the code rate and interleaver size based on elevation angle. This was performed for different assumed speeds, with a data rate of 256 kbps. In the above results the code rate were changed over elevation which then produced interleaver sizes which were relatively constant over elevation. In practice it is expected that the code rate is constant (to maintain a given level of throughput) and the interleaver size is also fixed. Both of these parameters are picked to deal with the worst case environment, i.e. minimum elevation angle and tolerable error rate.

In this clause we utilize the results from the previous clause to determine code and interleaver parameters. Based on these selected parameters we determine the average packet error (RS frame) rates for the particular configurations.

The assumed channel conditions are shown in Table 8.

Table 8: Assumed Channel Conditions

Channel Condition	Value
Min. Elevation	50°
Max Elevation	90°
Vehicle Speed	3 km/h to 70 km/h

The coding and interleaving parameters selected are shown in Table 9.

Table 9: Coding and Interleaving Parameters

Parameter	Value	Comment
RS Code Size	255	Design for small Interleaver, therefore use a large RS code.
RS Code Rate	0,686	K = 175 (try to maintain a high rate)
Interleaver Size	10^2 to 10^5 codewords	Check performance with different interleaver sizes so the smallest interleaver possible can be selected

We use the Lutz urban channel model as previously described. We check the packet error (RS frame size) rate and plot this as a function of the elevation angle. The results can be seen in Figure 66. We selected a number of interleaver sizes to show the performance as the interleaver gets bigger. As expected the errors decrease for increasing interleaver size.

What is noticeable here is when the interleaver has no effect (very small interleavers) the probability of codeword error is the same for both speeds. This is as expected as the average amount of time in an outage is the same for both speeds, therefore the average codeword probability of error is the same.

As the interleaver size is increased the errors decrease more rapidly for the higher speed as the outages are shorter for this higher speed and a smaller interleaver is needed to spread the errors for correction by the decoder. This can be clearly seen with the 10^3 interleaver where the 70 km/h result becomes significantly better at high elevation angles. For interleaver sizes of 10^4 and greater the errors are totally corrected for the 70 km/h case but not for the 3 km/h case.

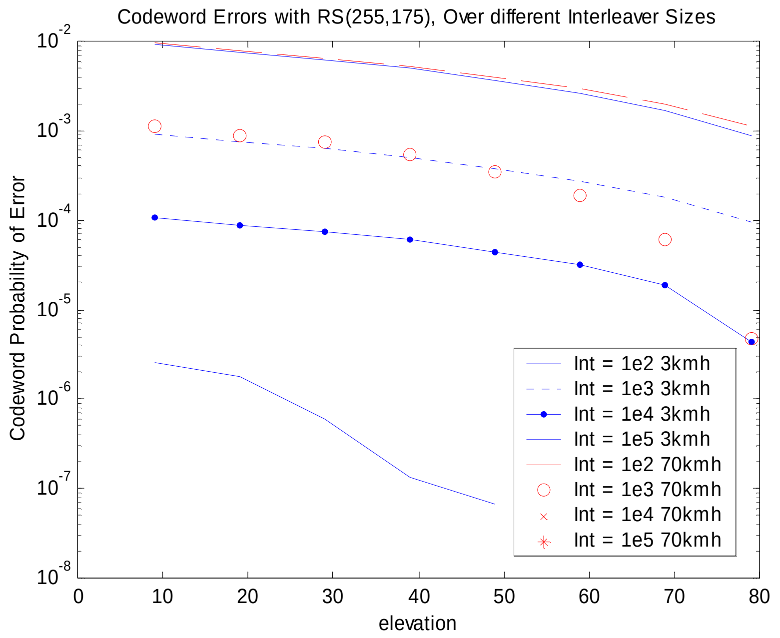


Figure 66: Codeword errors vs. elevation angle for urban channel model

What we see is, as expected, the errors approach zero as the interleaver size increases. From this plot an interleaver size can be determined, based on design constraints from code word probability and elevation angle. For elevations angles above 50° and interleaver size of greater than 10^5 will correct all errors for speeds of 3 km/h or higher.

Plots that are not shown (e.g. Int = 1e 4 70 km/h) is due to the fact that no errors were registered over the simulation interval tested.

E.1.5.5 Performance results (highway model)

In this clause we study the performance results using the Lutz Highway model. The parameters are the same as clause E.1.5.4. The Highway model is much better than the Urban model due to the a larger "good" distance and less "bad" metres.

The performance under this channel with the same conditions as the previous clause are shown in Figure 67. The performance at low elevation angles is comparable to the Urban model, however, as the elevation angle increases the performance improves much more rapidly than for the Urban model. Here the errors at 40° elevation start at 10^{-4} instead of 5×10^{-3} as for the Urban model. Due to the limitations in data from the model the performance can only be plotted over an angle elevation range of 16° to 56°.

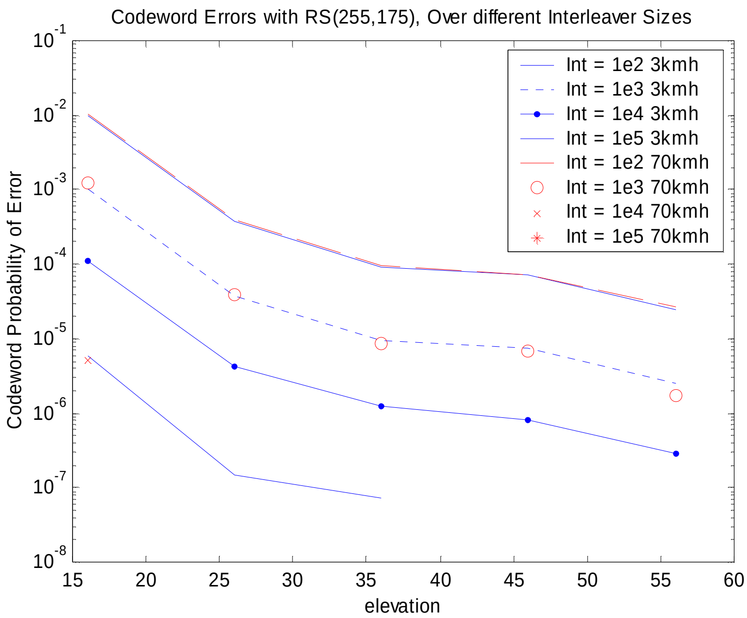


Figure 67: Codeword errors vs. elevation angle for highway channel model

In this example the maximum interleaver size to correct all errors for speeds is still 10^5 codewords, however the elevation angle is reduced substantially now to only 35°.

E.1.6 Medium block interleaver with CRC

The assumption here is that the satellite propagation channel consists of a number of impairments, namely:

1) medium/fast fading fluctuations;

2) short duration interruptions affecting a few packets (1 to 5 UMTS frames) (due to bridges, specular components, deep shadowing);

3) medium interruptions up to several minutes (tunnels, urban environment);

4) long interruptions up to several hours (underground parking, etc.).

The corresponding techniques to rectify these errors are:

1) physical layer FEC and interleaving;

2) Medium block interleaver with CRC;

3) RS FEC encoding and large block interleaving;

4) Carrousel techniques.

The idea is that frames plus the checksum frame are interleaved before being transmitted. At the receiver de-interleaving is applied and if one frame from the CRC has been identified to be an erasure then this can be regenerated.

E.1.6.1 Channel model investigation

Before simulating the technique just described we need to determine for the Lutz channel models if such short outages actually occur. We only consider the Lutz channel model in these examples but believe they are a good representation of the urban and highway channel due to the confirmation of this channel against channel measurements.

The average good and bad distance of the Lutz channel has been converted into the average Good and Bad time for 3 km/h, 70 km/h, and 120 km/h. The results for the Urban channel model are shown in Figure 68. This shows that the average bad time is significantly more than a 5 frames (50 ms) and that the smallest bad times only apply at high speeds (70 km/h or greater) or high elevation angles.

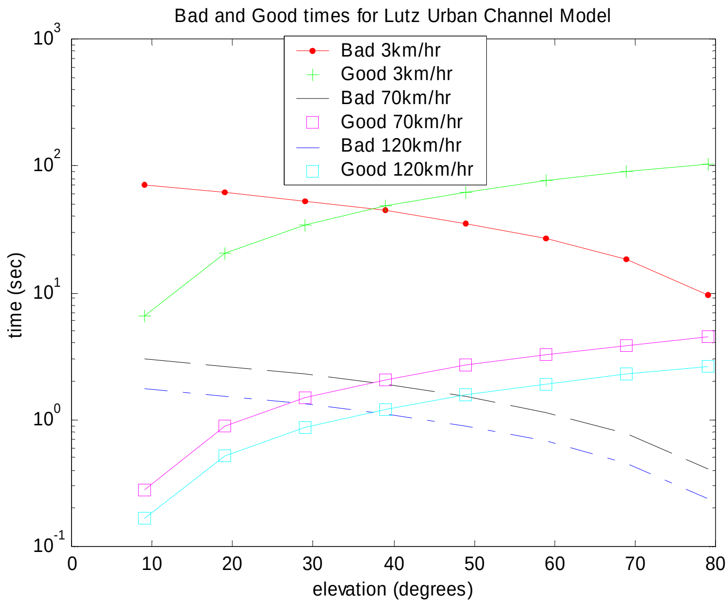


Figure 68: Average bad and good times for Lutz urban channel model

The same plot has been made for the Highway channel model shown in Figure 69. Here, for 120 km/h the average bad time is 1 second. This means that the average number of bad UMTS frames of 10 ms in a row is more than 100. This means if this technique was to correct all outages up to the average outage size then the CRC unit size would have to be a minimum of 100 frames, and therefore the total interleaver size for a rate 5/6 solution is 500 frames.

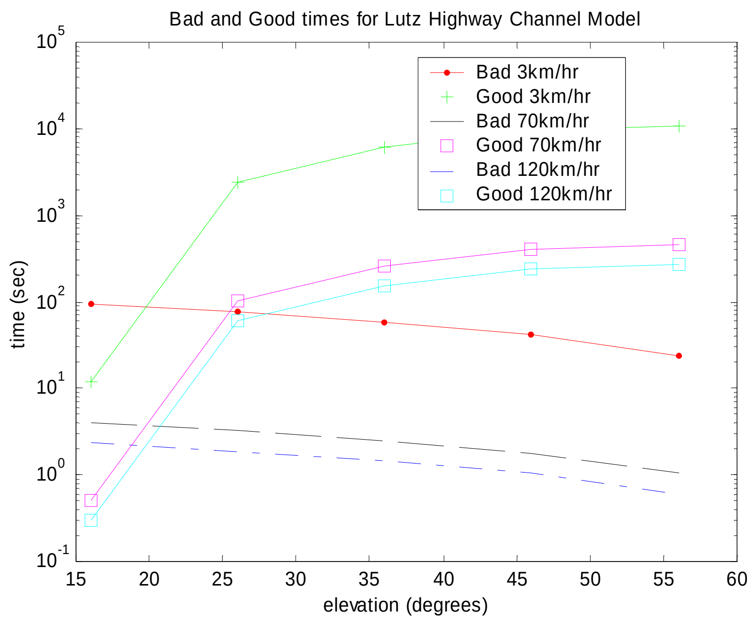


Figure 69: Average bad and good time for Lutz highway channel model

E.1.6.2 Channel characteristics and simulation results

To check the performance of this concept we used the Lutz highway model at 120 km/h and implemented an emulated CRC scheme to check the performance.

Firstly, however, we plotted the histogram of the frame error sizes against frequency. This is shown in Figure 70 for the lowest elevation case of 16°, where the single frame size is 1 second. The frequency of frames which are in error is significantly higher for one frame error, however, compared to the total number of frames in error this is not very significant.

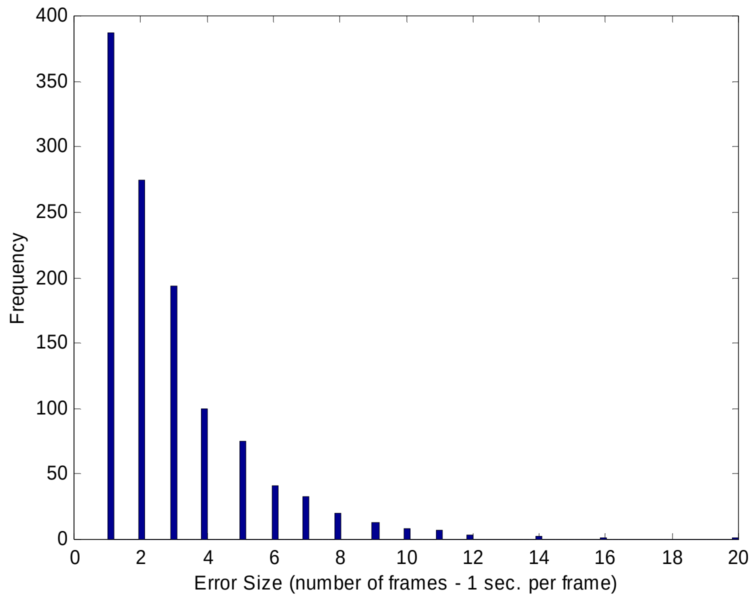


Figure 70: Histogram plot of frame error size

The codeword error rate performance improvement is shown in Figure 71. Here it can be clearly seen that the improvement in performance is nearly insignificant, especially at low elevation angles. This is as expected as the CRC scheme only corrects errors with one frame error, not outages with two or more frames.

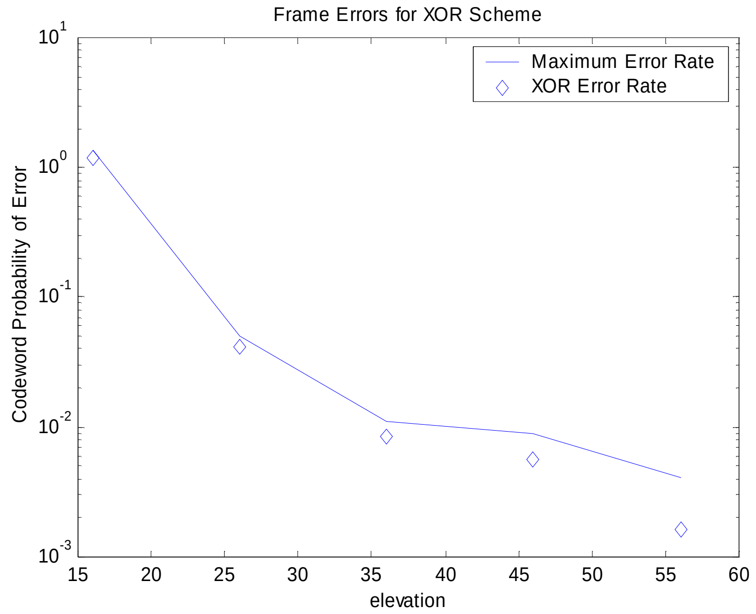


Figure 71: Codeword Probability vs. elevation angle for CRC Scheme

E.1.7 Carrousel transmitter with large and medium interleaver

Carrousel transmission takes advantage of time diversity by sending the same information block (potentially multiple times) at different times. The reliability of the data transmission is therefore improved for the outage channel of interest. Of course the trade-off for such an implementation is the reduction in throughput or rate. If the block is repeated then the rate is equal to 1/2, much like with FEC coding. This can also be thought of as a 3 dB increase in the signal to noise ratio.

The main parameters for Carrousel are:

- the size of the block for re-transmission;
- time between retransmissions;
- number of re-transmissions.

Carrousel techniques can be implemented at the application layer, where large repeat times and large block sizes can be considered. These techniques can also be considered in the upper layers where medium to small repeat times are utilized.

In this work we do not implement the carrousel alone as the performance gains are poor due to the lack of coding. We consider Carrousel for the upper layer together with FEC in clause E.1.8.

E.1.8 Combined hybrid implementation

The above results for the RS FEC with interleaving seem the most promising. The only problem with this technique is the size of the interleaver required. A 10^5 codeword interleaver for the RS255 equates to an interleaver of 25,5 Mbytes. This memory is required twice as one interleaver then can be emptied while another is being filled. This then means the total interleaver size is in the order of 50 Mbytes. The implementation of this interleaver is part of the LLC in the upper layers and would run in the kernel space of the operating system. According to network protocol results the largest expected memory that is allowed in kernel space of the Linux operating system is in the order of 10 Mbytes to 20 Mbytes. This means that the interleaver space currently required is approximately 2,5 to 5 times too big for what is expected to be possible with current operating systems.

For the given reasons above it is necessary to look at methods of reducing the memory requirements of the FEC/Interleaving approach. One approach is to implement a Carrousel technique before the FEC/Interleaving as shown in Figure 72. With such an approach the size of the interleaver needed is reduced as now the interleaver/FEC only has to provide reliable decoding every X frames, where X is the number of repeats by the Carrousel system.

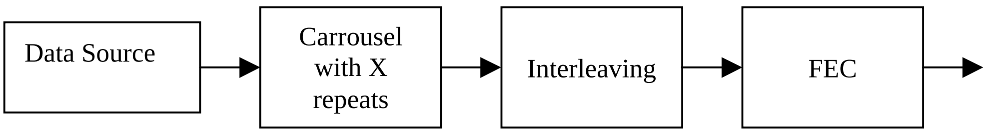


Figure 72: Hybrid Carrousel with FEC

E.1.8.1 Simulation results

In this clause the performance of the hybrid carousel plus FEC concept is analysed with the aim to reduce the interleaver size due to memory limitations in the gateway. Based on these selected parameters we determine the average packet error (RS frame) rates for the particular configurations.

The assumed channel conditions are shown in Table 10.

Table 10: Assumed Channel Conditions

Channel Condition	Value
Min. Elevation	50°
Max Elevation	90°
Vehicle Speed	3 km/h to 70 km/h

The coding and interleaving parameters selected are shown in Table 11.

Table 11: Selected Coding and Interleaving Parameters

Parameter	Value	Comment
RS Code Size	255	Design for small Interleaver, therefore use a large RS code.
RS Code Rate	0,686	K = 175 (try to maintain a high rate)
Interleaver Size/Carousel Block Size	10^2 codewords	Check performance with different interleaver sizes so the smallest interleaver possible can be selected
Carousel Repetitions	2	Minimize as the throughput is reduced

We use the Lutz urban channel model as previously described. The results in Figure 73 show the performance of a carousel system without FEC and interleaving and with FEC and interleaving, these results are at a speed of 70 km/h using the Lutz Urban model. The result is for a block size of 100 RS codewords (total of 204 kbits), where each block is sent twice (in succession), the RS code used was the RS(255,175) code. The received blocks were combined to minimize the error count from each repetition.

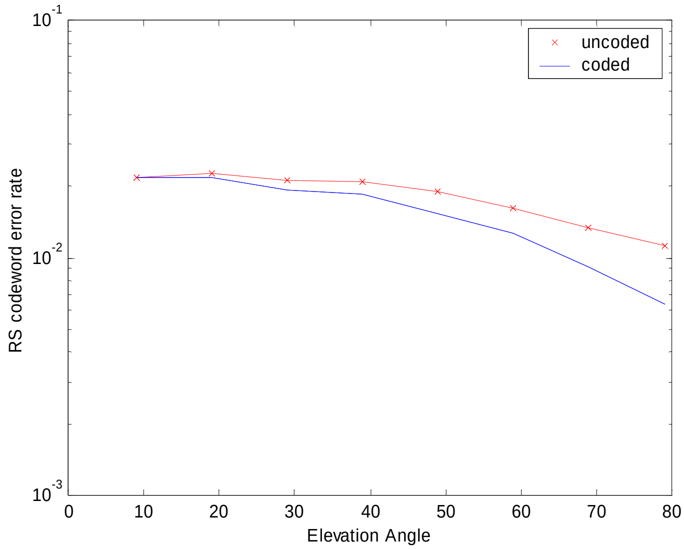


Figure 73: Performance of Carousel system over elevation

What can be seen is that the carousel uncoded result is very poor. Even with other interleaver sizes this was also observed. The carousel system brings some performance improvement over elevation angle but this is not really significant. With coding and interleaving the performance is slightly improved but not substantially (not to the level required).

The problem is that the coding performance can only be achieved if the interleaver block size contains approximately the good and bad time which match the code erasure correction capability. If the interleaver size is too small then the decoder has no chance of correcting the entire block, this is the reason the coded performance is not significantly better than the uncoded performance. Due to the poor results obtained for the short block carousel implementation this is not seen as a potential technique, especially when considering that the code rate of a coded/carousel scheme is approximately 0,343, therefore providing lower information bit throughput for the same bandwidth channel.

E.2 Narrowcast based on FEC and ARQ- a numerical study

E.2.1 Introduction

As already stated, the narrowcast is a routing technique that distributes a data stream to a smaller group of recipients. While, the FEC gives a degree of confidence for the reliable multicast, the narrowcast could give a 100 % reliability using a hybrid of FEC and ARQ techniques.

This study aims to demonstrate the efficiency and power of the FEC and ARQ combined technique within the mobile environment in particular S-UMTS. In this respect, the study uses the two states Markov channel model of Annex for assessing the narrowcast in terms of:

- number of retransmissions as result of ARQs;
- delay; and
- bandwidth requirements.

The study uses the simulation model of clause E.2.2 in order to assess the above narrowcast criteria through two distinct mobile speeds. This clause describes the proposed FEC and ARQ hybrid technique for the purpose of this study as well as the simulation model for the study. Clause E.2.3 presents simulation results for two distinct mobile speeds of 3 km/h and 70 km/h.

The channel model outage "bits" have been considered as follows. The channel model provides the statistics in number of metres spent in BAD state (D_{bad}) and the number of metres spent in GOOD state (D_{good}). From the known MT speed (S) we converted (D_{bad}) and (D_{good}) to time (time = speed / distance). With the data rate, time can be converted to bits (number of bits = data rate / time).

E.2.2 Simulator design and implementation

The simulator has been designed and developed in Matlab and is based on the channel model as described in Annex . The simulator has been implemented in a modular way, with a list of parameter available that can be easily changed from one simulation run to another. Table 12 gives these configurable parameters.

Table 12: Simulator parameters list

Parameter	Description	Value range
Environment	Urban/sub-urban	0/1
Speed	Terminal speed	3 km/h, 70 km/h
Rate	Data rate	512, 256, 128, 64 kbits/s
FEC coding rate	RS encoder parameters	(255,223), (255,123), (255,63)
File size	The size of the file that will sent to the MT before coding	2 000 kbits This file size has been selected to covers the wide range of potential multicast application. For instance a traffic map file size is around 200 Kbits, while an MP3 file size is a few hundred of Kbits, and a video clip file size is around a few Mbits.
Threshold	E_b/N_o	3 dB
Elevation	Satellite elevation	

To examine the effect of the different parameters in a narrowcast link from a single gateway to a number of mobile terminals, the simulation uses the two-state Markov channel model described earlier, with the assumption of perfect transmission in the non-blocked areas and a lognormal fading in the blocked areas.

The simulated Narrowcast Protocol (NP) is based on the combined FEC and ARQ. In this protocol, each of the addressed receivers sends an ARQ if the number of errors is larger than what it can successfully decode, thus having the correct data packet(s). The gateway filters the ARQs in order to decide whether there is a need for retransmission or not (i.e. at least one ARQ received after a full transmission). If there is a need for retransmission, then the gateway retransmits the whole file as is shown on Figure 74. The receiving mobile terminal therefore combines the results of consecutive transmissions until it correctly receives all the required data.

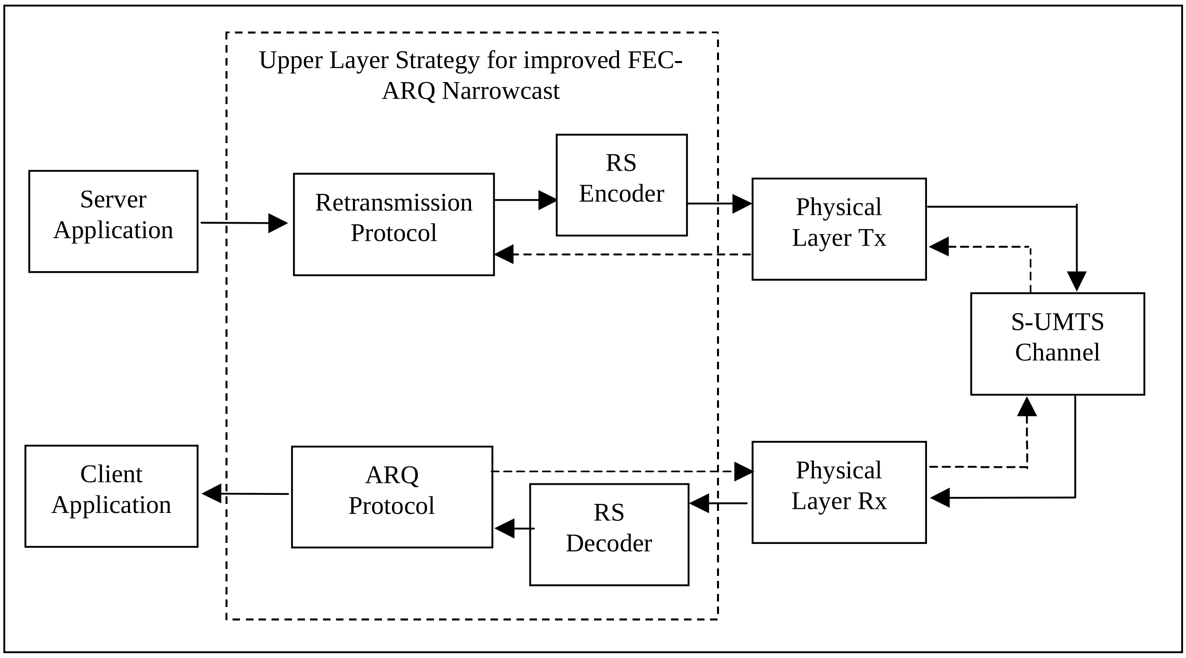


Figure 74: Block diagram for FEC-ARQ based Narrowcast Protocol (NP)

Figure 75 illustrates the behaviour of the above mentioned NP and hence the effect of its consecutive retransmission process for the 100 % reliable narrowcast - i.e. each and every receiver to correctly receive the complete datafile. This has been performed using a scenario that involves a narrowcast group of 100 terminals needing to correctly receive a copy of the entire file. Each dot on the graph indicates a lost packet.

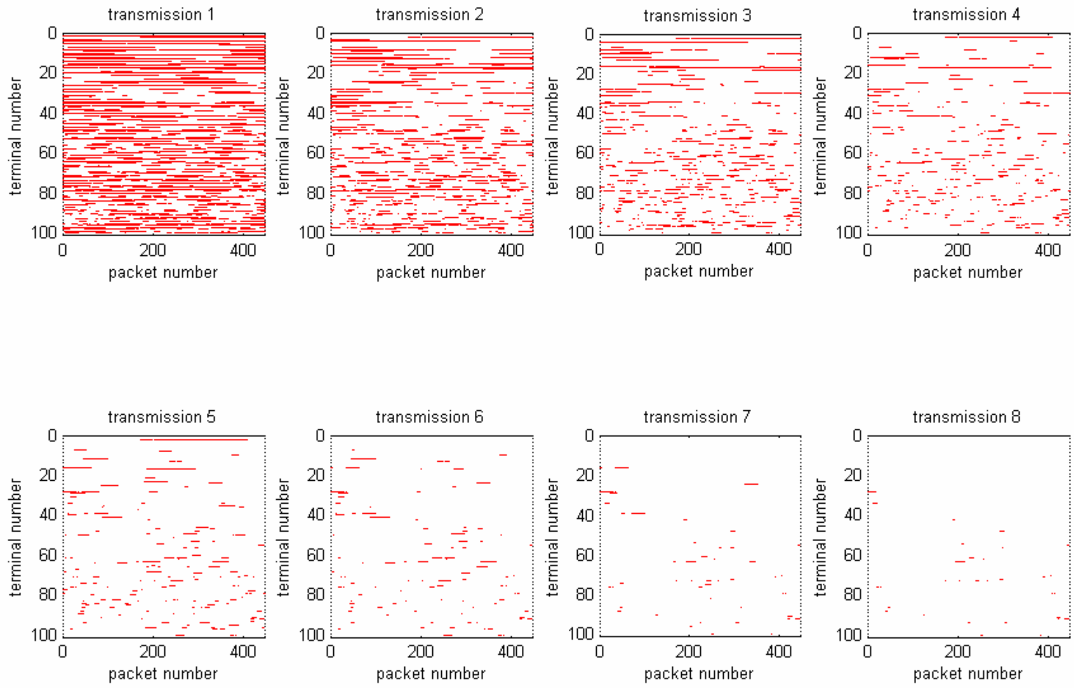


Figure 75: Receiver Performance over 8 re-transmissions

These tests results were obtained using a pedestrian MT (MT speed = 3 km/h) in urban area. The user data rate was set to 256 Kbits/s, and the total file size before RS coding is 2 000 kbits.

From the above graphs we can clearly see the improvement after each retransmission. After 9 re-transmissions, all terminals are able to correct the remaining errors and the file is received completely. As illustrated by the graphs from the top left to the bottom right, a certain number of terminals correctly receive the entire file via the first transmission while others have to use the ARQ process via the return channel. Through such an iterative process, the terminals will eventually get the correct narrowcast file. The histogram in Figure 76 shows the outcome of this protocol experiment.

The breakdown of the each retransmission hence histogram of the NP experiment in terms of the 100 recipient terminals is as follows: 3 terminals have received the file via the first transmission. 11 terminals required one retransmission, 18 two retransmissions, 29 three, 20 four, 11 five, 4 six, 2 seven, 1 eight and 1 nine retransmissions. In terms of the ARQ traffic on the return channel, the experiment reported 324 retransmissions in total, i.e. a 3,24 average per terminal. This corresponds to a high peak at the value 3 in the histogram.

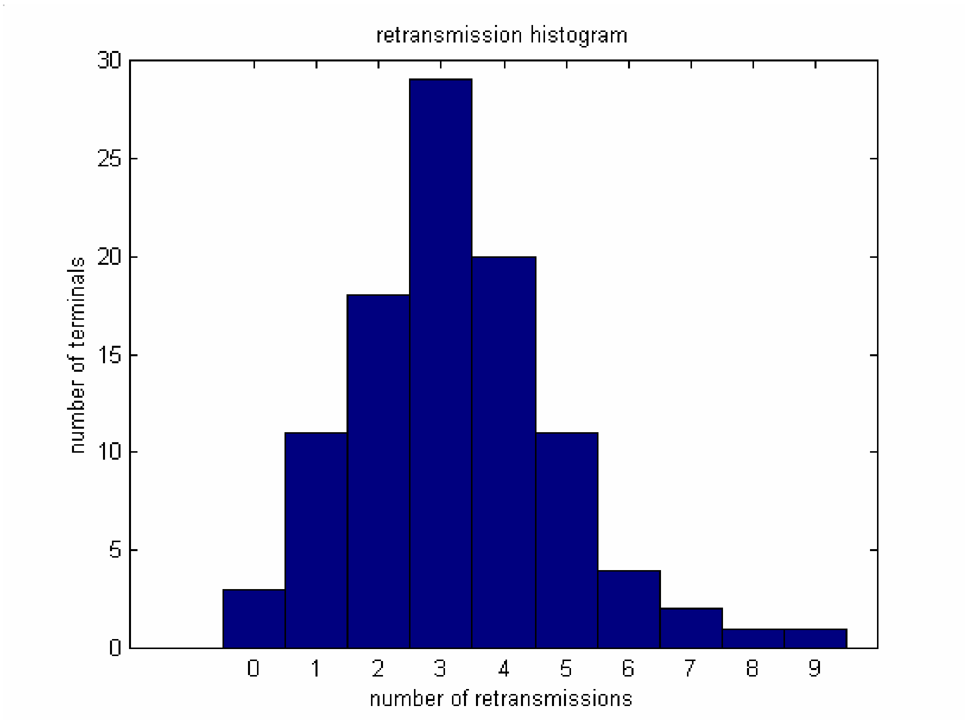


Figure 76: Number of terminals requiring retransmission vs number of retransmissions

E.2.3 Simulation scenarios

The NP simulator of clause E.2.2 has been used for two distinct mobility types hence scenarios:

1) pedestrian (speed range ~ 3 km/h);

2) vehicular (speed range ~ 70 km/h).

For each scenario, several simulation runs have been carried out with different parameter settings.

E.2.3.1 Pedestrian terminal in urban area

This NP simulation scenario has been carried out via various runs with the parameter set shown in Table 13.

Table 13: Parameters used for simulation scenario

Parameter	Value
Environment	Urban
Terminal speed	3 km/h
FEC coding	Variable: (255,223), (255,123), and (255,63)
Bit rate	Variable: 512,256,128, and 64 kbits/s
Size of the transmitted file	2 Mbits
Number of Terminals	Variable: 3 MT,10 MT,30 MT,60 MT,100 MT
Elevation	30°
Interleaving	None

Figure 77 reports on the NP experiment using FEC with RS(255,223) code for the four sets of bit rate of table 13.

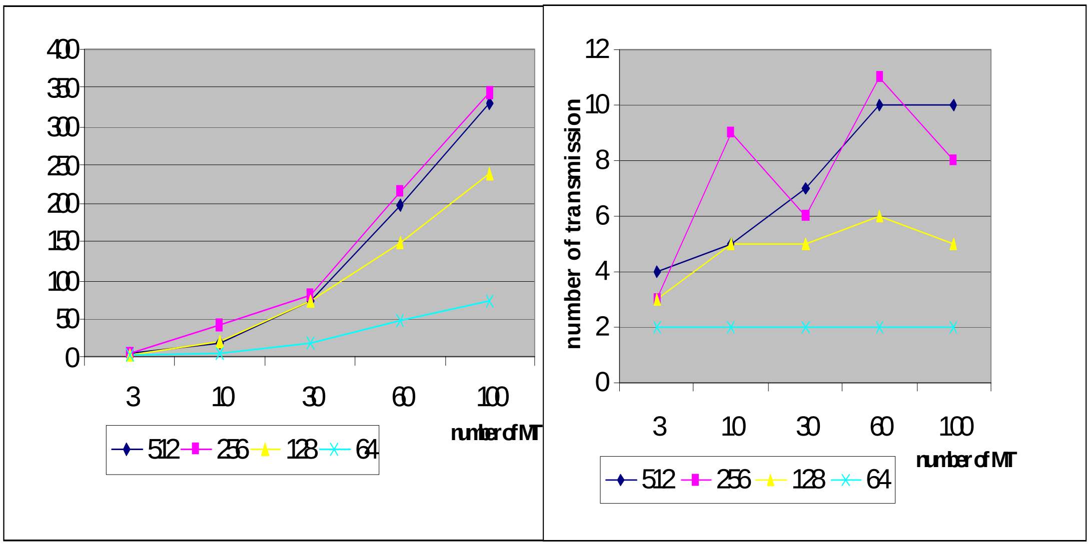


Figure 77: A - Average number of ARQ versus the number of MT
B - Average number of retransmission versus the number of MT

Figure 77-A shows the linear increase in the ARQ with respect to increase in number of terminals where the effect of the downlink transmission rate is patently clear. At 512 kbits/s transmission rate (pink curve with bold square), about 200 ARQs have been generated by a population of 60 MT. While at 64 kbits/s transmission rate (blue curve with x), less than 50 ARQs have been generated.

Figure 77-B shows the required number of retransmissions in view of the different bit rates. At 512 bits/sec (pink curve with bold square) an average of 8 retransmissions is required. While at 64 kbits/s (blue curve with x) mean required retransmission of 2 retransmission has been required.

Note on terminology as used in the clauses above:

- The number of re-retransmission = number of time the user data file is transmitted on the FL.
- The number of ARQ = number of ARQ re-transmitted on the RL.
- As the retransmission concept is dealing with missing data, the ARQ should be understood as a N-ACQ (Negative Acknowledgement).

Figure 78 shows the simulation result based on the same environment as above with the FEC coding rate set to RS(255,123).

Interleaving on the blocks has not been used, but different transmission data rates have been used.

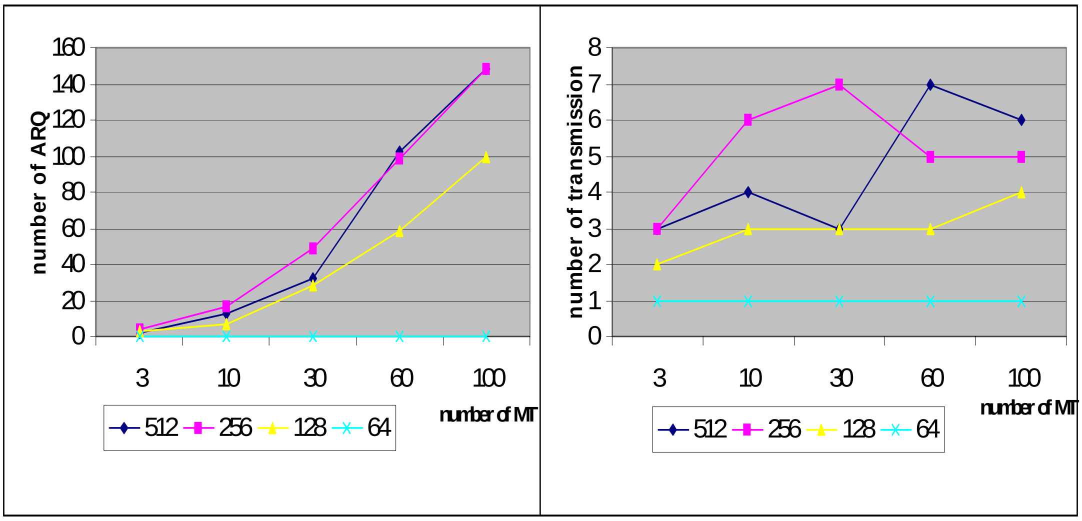


Figure 78: RS(255,123)

Figure 79 shows the simulation result based on the same environment as above with the FEC coding rate set to RS(255,63).

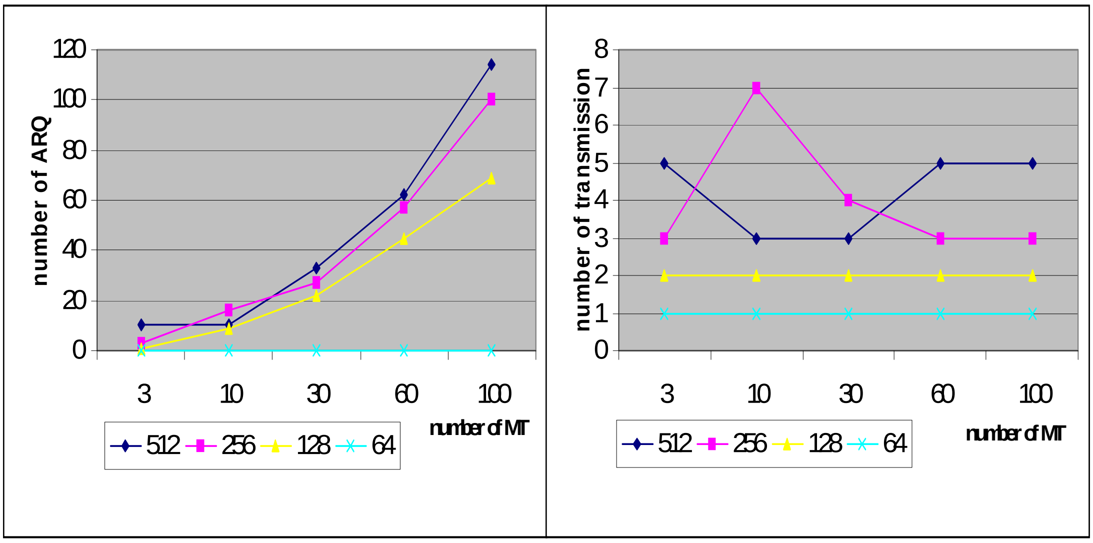


Figure 79: RS(255,63)

The direct relation between the decrease in the RS coding rate and the ARQ is patently clear from the above experiment (e.g. see for the rate 63/255, the throughput is reduced for the same channel symbol data rate). That is the higher number of redundant packets, the lower the number of generated ARQ thus less traffic on the uplink.

From the above simulation runs several statistics could be extracted and shown on the following figures.

Figure 80 shows the narrowcast transmission delay with respect to the different RS codes for various transmission rates. This is the total delay that includes consecutive retransmission and coding delays. This delay does however not include any delay due to the return link access for ARQ transmission, nor includes the satellite delay.

It should be noted that it is not needed to include a GEO satellite introduced (full round-trip time of about 500 ms as for a GEO). This is justified by the fact that on average we have about 8 re-transmission resulting into $8 + (8 - 1)) \times$ one hop delay $= 15 \times 250$ ms = 3,7 sec delay, while the transmission delays as indicated in Figure 80 is in (40,130) sec range.

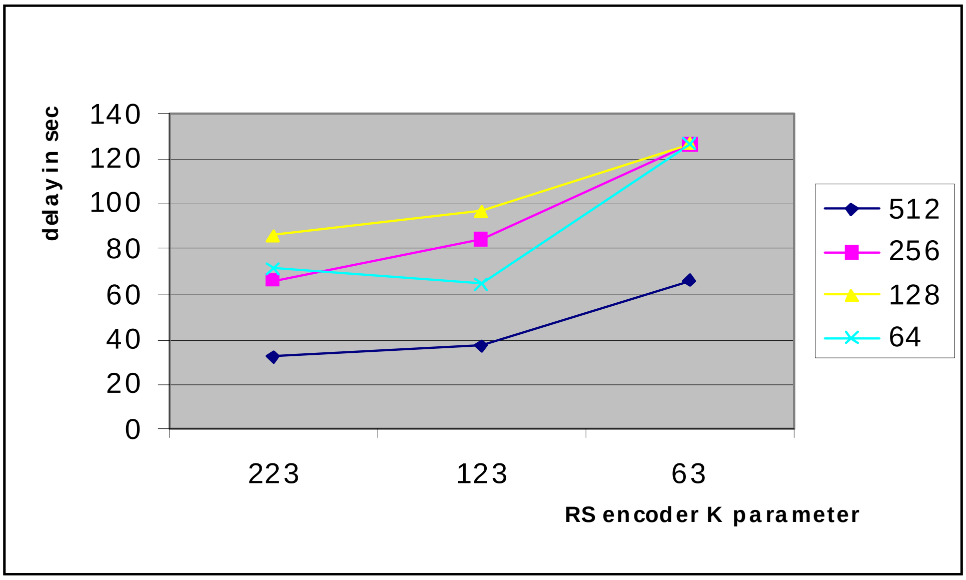


Figure 80: Total delay variation for different data and coding rates

The total delay increases significantly for decreasing RS coding rates (lower K parameter). This is justified by the fact that for lower coding rates the number of re-transmissions required is lower, while the coding overhead is higher and thus the total delay is longer.

Figure 81 shows the effect of the RS coding rate in view of the total bandwidth (BW) requirements on the forward link. Such a bandwidth includes:

- user data;
- coding overhead;
- re-transmission.

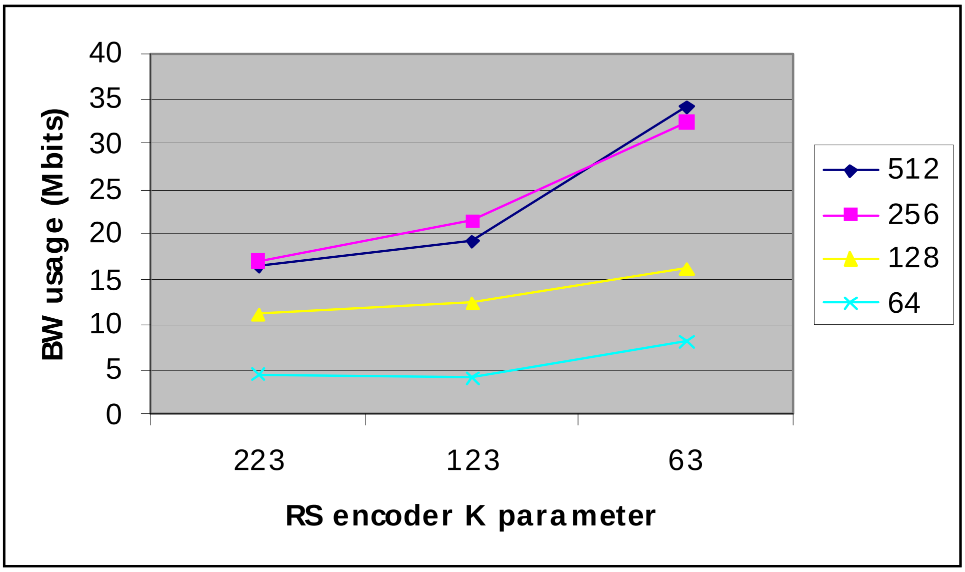


Figure 81: Total forward link BW usage for different data and coding rates

As the data rate increases the total BW usage increases. This is justified by the fact that at high data rate the data block is sent in a shorter time interval and it is possible for the entire block to be in error due to fading. For a slower data rate the time is longer for the entire block transmission and less errors are generated, thus requiring less re-transmissions.

In the S-ATB case the bandwidth of the channel is limited and the data rate of the application will have to be reduced if all ARQs (N-ACQs) are to be answered.

Figure 82 shows for different data rate the total return link traffic volume usage (assuming ARQ size = 50 bits) versus the RS coding rate.

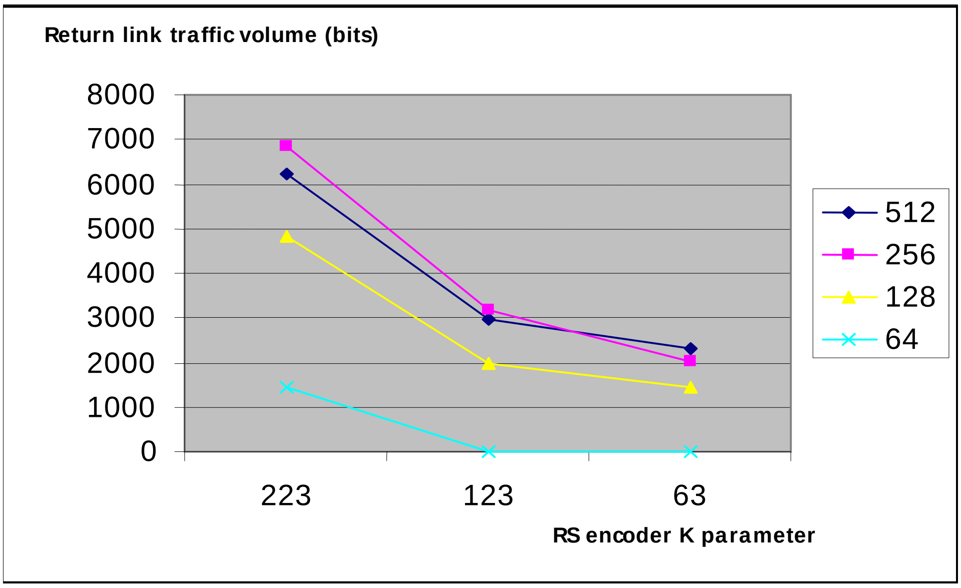


Figure 82: Return link BW usage

As the data rate increases the total return link BW usage increases. This is justified by the fact that at high code rates more errors are generated, thus more ARQ are generated.

E.2.3.2 Vehicular terminal in urban area

In this simulation run, the parameters setting is as shown in Table 14.

Table 14: Simulation parameter settings

Parameter	Value
Environment	Urban
Terminal speed	70 km/h
FEC coding	Variable: (255,223), (255,123), and (255,63)
Bit rate	Variable: 512,256,128, and 64 kbits/s
Size of the transmitted file	2 Mbits
Number of Terminals	Variable: 3,10,30,60,100 MT
Elevation	30°
Interleaving	None

Figure 83 shows the number of retransmissions and the number of ARQs (transmitted on the return channel) versus the number of terminals. An RS(255,223) FEC coding has been used.

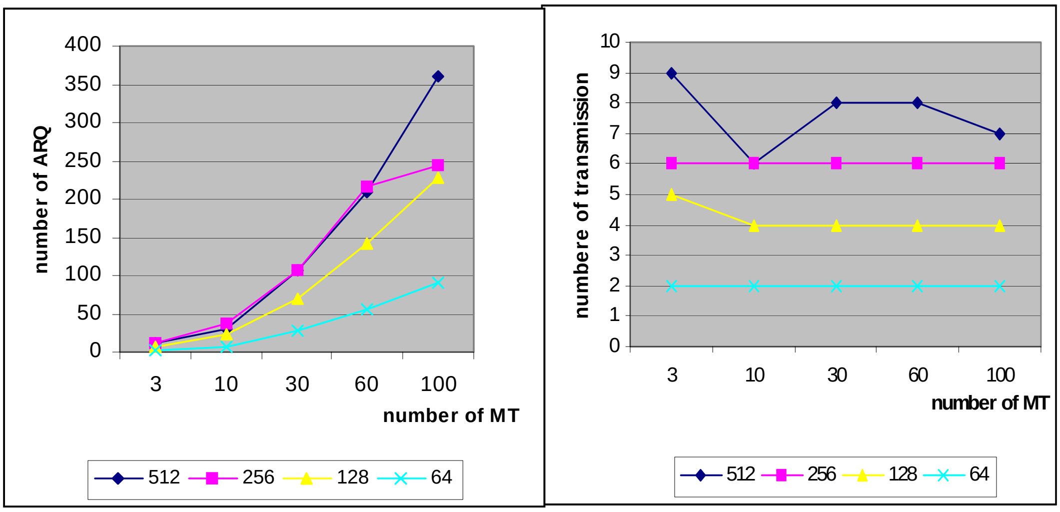


Figure 83: A - Number of Average ARQ versus the number of MT
B - Number of Average re-transmissions versus the number of MT

Figure 83-A shows that the number of ARQs transmitted on the return channel increases linearly with the number of MTs. The effect of the transmission bit rate on the number of ARQs is clear. At 512 kbits/s transmission rate (pink curve with bold square), about 200 ARQs has been generated by a population of 60 MTs. While at 64 kbits/s transmission rate (blue curve with x), less than 50 ARQs has been generated.

Figure 83-B shows the required number of retransmissions in view of the number of terminals. At 512 bits/sec (pink curve with bold square) an average of 3 retransmission is required, while this average is 2 at 64 kbits/s (blue curve with x). Figure 84 shows the simulation result based on the same environment as above, while the FEC coding rate has been set to RS(255,123).

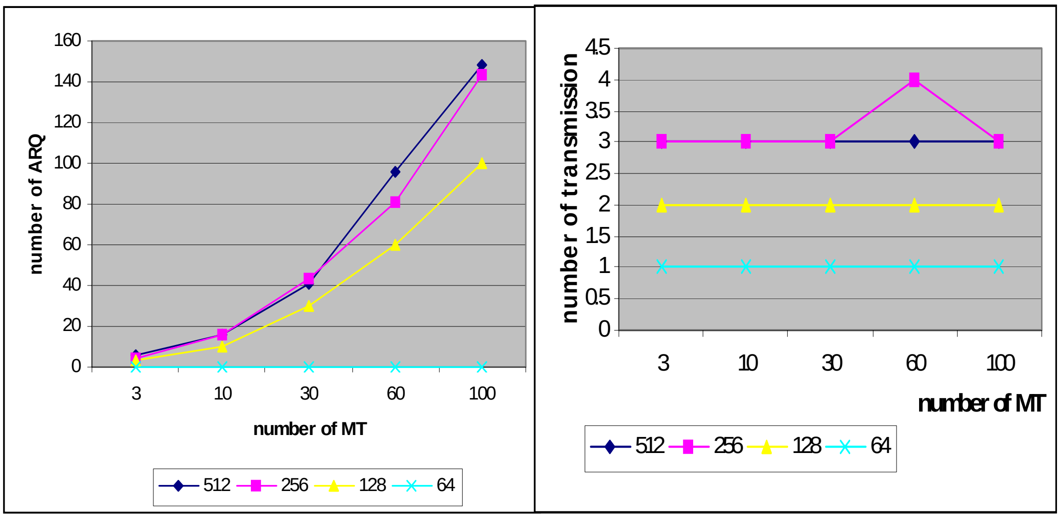


Figure 84: RS(255,123) Performance

Figure 85 shows the simulation result based on the same environment as above while the FEC coding rate has been set to RS(255,63).

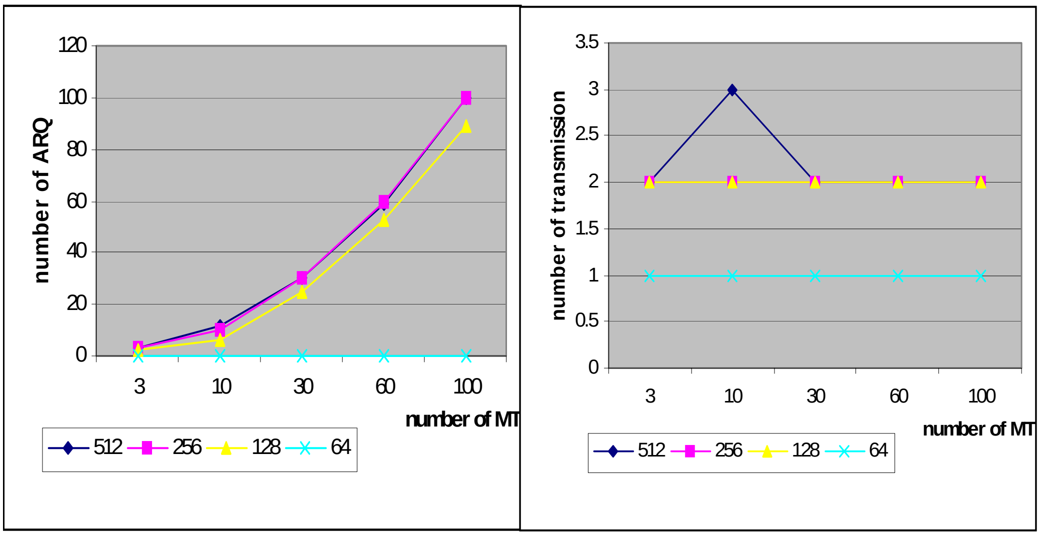


Figure 85: RS(255,63) Performance

Figure 86 shows the narrowcast transmission delay with respect to the different RS codes for various transmission rates. This is the total delay, which includes consecutive retransmission and coding delays. This delay does not however include any delay due to the return link access for ARQ transmission, nor includes the satellite delay.

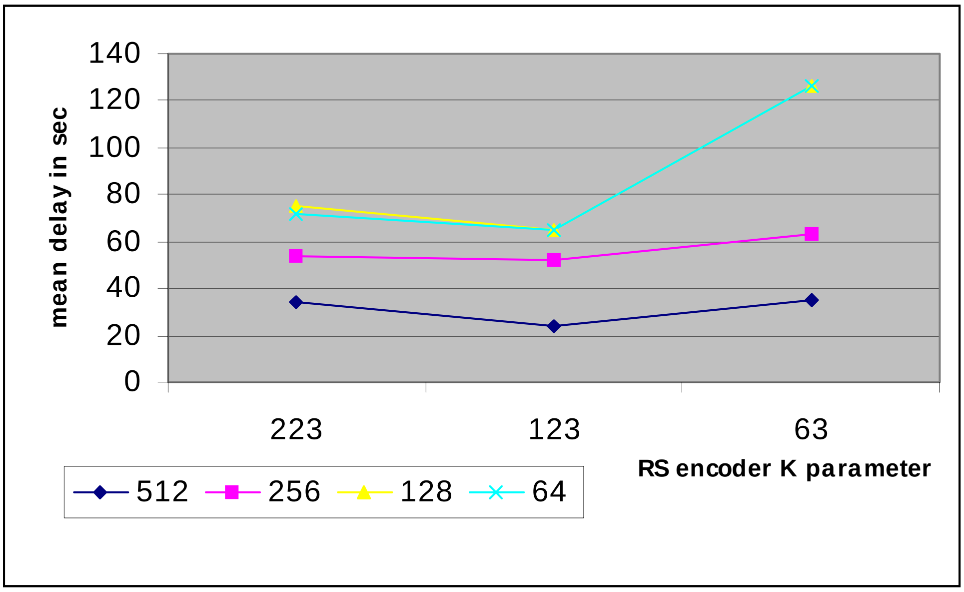


Figure 86: Total delay variation for different data and coding

The total delay increase significantly for increasing RS coding rates (lower K parameter). This is justified by the fact that for higher coding rate the number of re-transmission required is lower, while the coding overhead is higher and thus the total delay.

Figure 87 shows the effect of the RS coding rate in view of the total bandwidth (BW) requirements on the forward link. Such a bandwidth includes: user data, coding overhead, re-transmission.

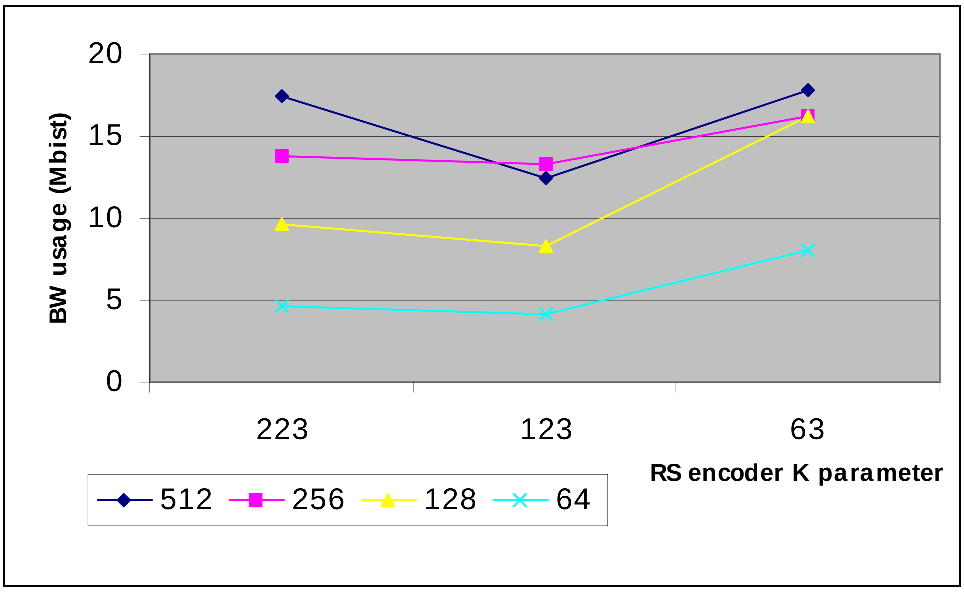


Figure 87: Total forward link BW usage for different data and coding rates

As the data rate increases the total BW usage increases. This is justified by the fact that at high data rate more error are generated, thus more re-transmission are required.

Figure 88 shows for different data rate the total return link traffic volume usage (assuming ARQ size = 50 bits) versus the RS coding rate.

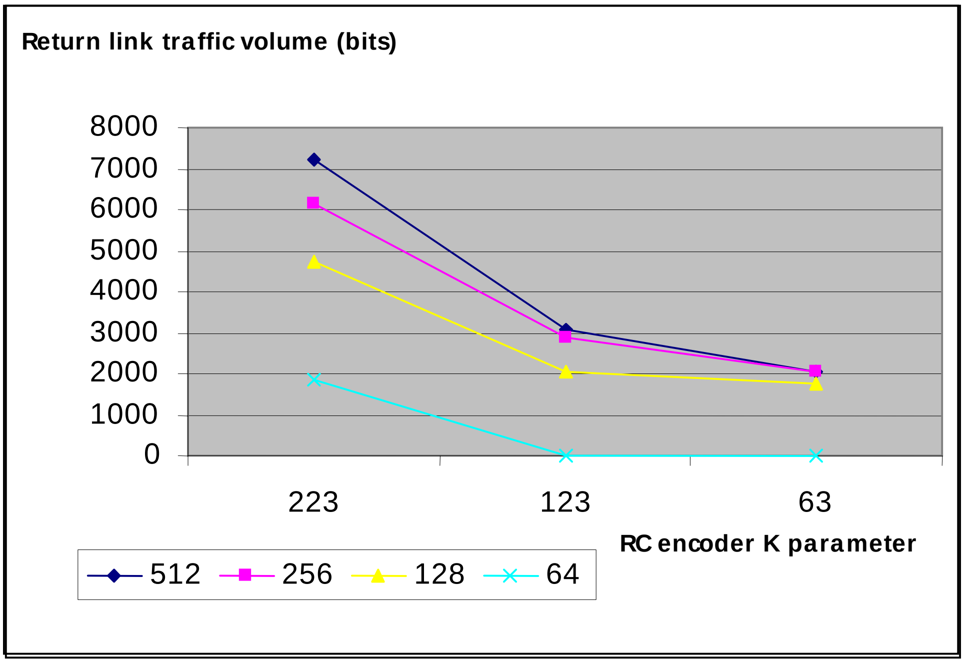


Figure 88: Total return link BW usage for different data and coding rates

As the data rate increases the total return link BW usage increases. This is justified by the fact that at high data rate more errors are generated, thus more ARQs are generated.

Annex F: IST/SATIN Access scheme definition

F.1 Layer 2 specifications and L2+ main features

F.1.1 RRM strategy and RRC interactions with lower layers

F.1.1.1 Introduction

The main function of Radio Resource Management (RRM) is to allocate physical radio resources when requested by the Radio Resource Control layer.

In the SATIN service scenario the RRM consists basically of Admission Control (AC), Load Control (LC) and Packet Scheduling (PS).

AC admits multicast user groups to set up or to reconfigure a Radio Access Bearer (RAB) only if these would not overload the system and provided the necessary resources are available. LC takes care that a system temporarily going into overload returns to a non- overloaded state. The main task of the PS is to handle all non-real-time-traffic e.g. allocates optimum bit rates and schedules transmission of the packet data, ensuring the required QoS in the terms of throughput and delays. The satellite RRM is located in the S-RNC at the Satellite Gateway.

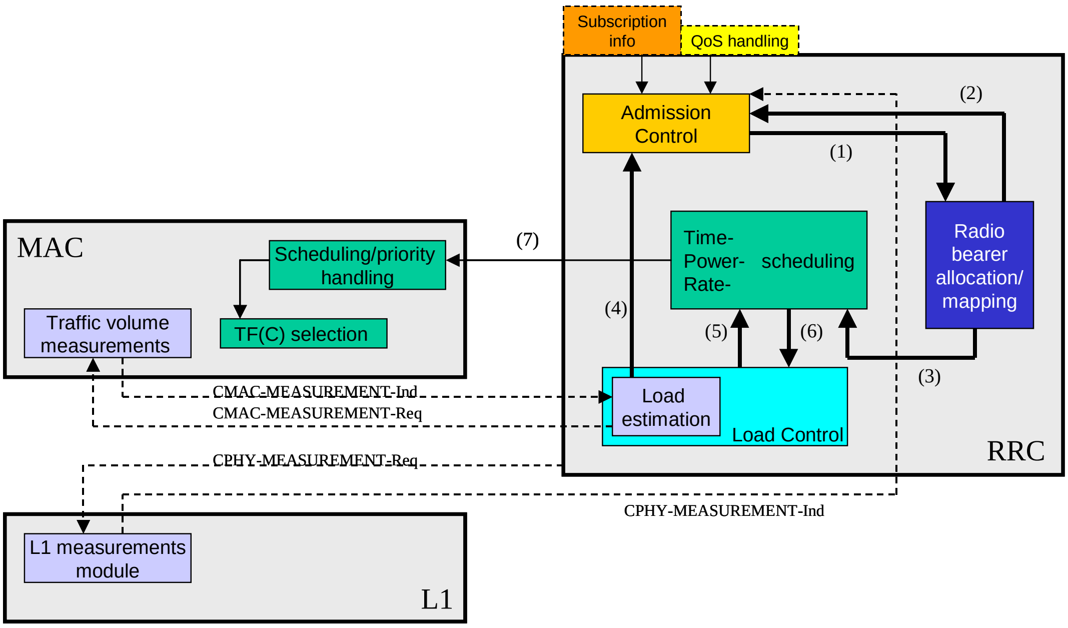


Figure 89: Model of the RRM entities in SATIN

Three main QoS service classes are considered in the SATIN architecture, the main distinguishing factor among them being how delay sensitive the corresponding application traffic is Table 15.

Table 15: Main characteristics of QoS service classes [15]

Traffic class	Streaming class	Interactive class	Background
Fundamental characteristics	Preserve time relation (variation) between information entities of the stream	Request response pattern Preserve payload content	Destination is not expecting the data within a certain time Preserve payload content

F.1.1.2 Radio Bearer Allocation and Mapping (RBAM)

General description

A mandatory step in the RRM procedure is the RB configuration, i.e. the number of transport/physical channels required and their mapping together with the actual TFCS for each physical channel. In general, this appears to be a task of a separate "block", a functional block within RRC that is in close connection with the other RRM functional blocks. This block within SATIN is called Radio Bearer Allocation and Mapping (RBAM).

Interaction of RBAM with other SATIN RRM functions

Within SATIN it is possible to identify two modes of operation:

Mode A: there is a fixed RB configuration over some interval of time, over which the traffic mix remains the same (for example in the order of 1h). This mode relies on some kind of prediction of the traffic that is expected over subsequent time intervals, which may be based on historical data drawn from measurements. This corresponds to the traditional dimensioning task, practised for years in telecommunication networks (e.g. telephone networks).

In this case the AC block functions within the constraints imposed from this mapping, which is the responsibility of the RBAM block.

Mode B: the RB mapping is drawn in an ad-hoc manner by the AC without any prior configuration. In this case the AC decides on the acceptance of the service request and then (re)-configures the bearer appropriately. The AC decision upon the acceptance or rejection of a service request is made on the basis of power or load constraints, without taking account of the per physical channel rate constraints. In effect, AC assumes that there is an infinite flexibility regarding the radio bearer mapping and allocation, i.e. the RBAM will re-map FACHs upon S-CCPCHs and reconfigure them, as far as a service request is accepted in terms of the extra load and power constraints it introduces.

The second option allows higher flexibility in the resource utilization at the expense of extra interlayer - and over the air - signalling (reconfiguration messages towards the group users). Given the absence of positive acknowledgement on behalf of the group users, the latter option is at disadvantage.

The RBAM function becomes more relevant in mode A and is described in the following.

RBAM strategy

The aim of the RBAM block in mode A is to perform some dimensioning of the system, on the basis of the traffic mix; the assumption is that there is some adequate description of the traffic mix in terms of -at least- arrival rate, duration and requested rate for each type of service. This is equivalent to providing an appropriate configuration of the bearers (number of FACHs, rates and mapping to S-CCPCH). The task may be split into more than one step and their actual context depends on the assumptions on the service characterization:

Option 1: It is assumed that all types of services are described by an arrival rate λ_i, duration μ_i and rate R_i.

Option 2: It is assumed that some types of services (classes) may be described in this way but there are other types of services that are scheduled by the network in a free manner, e.g. content that may be downloaded with whatever rate (and consequently duration) and may be scheduled with whatever arrival rate.

The steps relevant to the RBAM task are:

Step 1

Estimation of the necessary number of FACHs for the provision of the services under consideration. In option 1, the estimation involves all types of services, while according to option 2 the estimation concerns a subset of the services (e.g. RT services, both streaming and background). In the second case the residual capacity left after the FACHs computation for this subset of services is estimated.

Step 2

Mapping of the FACHs on the available S-CCPCHs and -possibly- derivation of the TFCS for each S-CCPCH.

In the following, these two steps are detailed further.

Step 1: Estimation of the necessary FACHs

This task may well rely on classical queuing theory (see for example [23]).

A finite range of rates R_i is assumed for the possible rates of services/flows. The exact, packet-level behaviour of the flows (CBR or bursty) is of no interest at this stage; we focus only on the mean rate (packet-level dynamics can be controlled by the TFCS passed to the packet scheduler).

For each possible rate *i*, different durations *j* are possible (again we assume some range of average values) and for each service type, i.e. combination of $\{\mu_{ij}, R_{ij}\}$ an arrival rate λ_{ij} is defined. To formulate this:

- Let S be an NxM matrix, where N are the possible service rates and M the possible service durations. No assumption is made for the flow burstiness (it might be CBR or VBR, but in the latter case the mean value is assumed), while the service duration may also be of any distribution having a rational Laplace transform [24]. Then each element s_{ij} corresponds to a type of service, i.e. combination of service rate and duration.

- Let also L be another NxM matrix, each element λ_{ij} corresponding to the arrival rate of the services of type s_{ij} and P_{bl} a 3rd NxM matrix corresponding to blocking probabilities targeted for each service, i.e. there is one-to-one correspondence between s_{ij} and Pbl_{ij}.

Then the estimation of the necessary FACHs for each s_{ij} can be derived according to the following ways:

a) From the m-server loss queueing system (Erlangian) (see for example [23]), for each service type s_{ij} separately,

i.e. application of the formula NxM times.

b) Via application of the recursive formula derived almost simultaneously in [24] and [25] over all types of flows s_{ij} requesting the same rate R_i, irrespective of the durations of the respective services. The formula is the extension of the Erlangian formula to the multiple services scenario and can be applied under the assumption that FACHs can be fully shared among classes requesting the same rates (i.e. as long as the TFCS provided can cope with possible discrepancies at the packet level) (see note 1).

NOTE 1: Application of the same formula over all services (of all rates) is not possible since the total number of FACHs is not available in full-shared mode.

In both cases the required number of FACHs is that number of servers (of rate equal to R_i) that will guarantee the target blocking probabilities Pbl_{ij}.

Step 2: Mapping of the FACHs on S-CCPCHs

A number of rules must be defined and now it is necessary to consider other attributes of the considered services, i.e. maximum/guaranteed bit rate, maximum burst size (see note 2), maximum transfer delay of service, and SDU size.

NOTE 2: Maximum burst size (related to the maximum time that a source may transmit at the peak rate) is not a standardised attribute within the TS 123 107 [17], yet it might hold some valuable information for making appropriate combinations of FACHs upon S-CCPCHs.

Strict rules or algorithms for performing this task are difficult to find. In any case, deriving the TFCS a priori, on the basis of traffic predictions is not too efficient. The TFCS should be broad enough to capture the packet-level dynamics of the services expected over some future time interval. The wider the range of services, the broader the TFCS should be with direct impact on the terminal processing requirements. Some rules might be:

- Try to combine bursty RT services with nRT services that do not have delay constraints on the same S-CCPCH. In this way the resource utilization can be improved by letting the BE, nRT services fill the S-CCPCHs whenever they are not fully occupied by RT data.
- If bursty RT services are to be combined with CBR RT services, make sure that the delay requirements (if any) of the latter can be fulfilled. This may be assured with appropriate TFCS definition or some generic checks, and is also dependent on the particular scheduling strategy, for example in case of two services, under strict-priority discipline, the higher priority service queue should not feature busy periods that are longer than the maximum transfer delay of the second-priority CBR service.
- The chosen TB sizes should be in line with the packet sizes expected from the applications, so that overheads (headers and padding) are minimum. The same reasons (i.e. minimization of the overheads and resource utilization efficiency) dictate TFs for each FACH, that can cover the full range of short-term rate variations.
- The E_b/N_o requirements of the services (FACHs) multiplexed on a single S-CCPCH should be similar in order to save power. Note that the actual E_b/N_o requirements are dependent on the actual utilization of the transport block sizes provided in the FACH transport format.

In option 2, instead of doing the dimensioning for background services in terms of the $\{\lambda, \mu, R,\}$ characterization and a desired blocking probability, we could estimate the residual capacity corresponding to FACHs from the residual rates per S-CCPCH, i.e. spread the FACHs over the available S-CCPCHs and fill each pipe (S-CCPCH) with one FACH of the residual rate. This is a more realistic scenario for background services of lower priority, the delivery of which might be triggered by network initiative rather than after an explicit request by a user group (i.e. delivery of a popular Web site that might be of interest to many users). In this case the network makes available these remaining FACHs to these services and AC negotiates the accepted rate iteratively (assuming that any offered rate will be finally accepted by the service). Further (softer) levels of priorities may be applicable here on the basis of the total transfer time of the flow, i.e. higher-rate FACHs may be allocated to the higher (among the low) priority streams.

Then going backwards, knowing the FACHs available for such low-priority services, the network might schedule the arrival rate of these services. Alternatively the network may program/schedule broadcast cycles of content that has been found to be popular. In that case the actual problem is related to the design of broadcast cycles and can be addressed along the lines of [26].

A generic approach for TFS and TFCS derivation

Building upon the transport channel description and capabilities as specified in TS 125 331 [20], TS 125 306 [27], 125 302 [28], 125 211 [29] a generic approach for deriving a candidate setting of TFS/TFCS is described hereafter.

TFS Derivation

According to its definition, a Transport Format (TF) consists of the following two parts:

- The dynamic part {TB_size, TBS_size}.
- The semi-static part {TTI, coding, CR, RM, CRC}.

We define TB_size and TBS_size as:

$$TB\_size = K * step ,$$

where *step* is the Min_TB_size (possible values 320, 160, 80, 40, 122, 244 etc) and *K* is the integer number of steps into one TB.

$$TBS\_size = TB\_size * N = (K * step) * N ,$$

where *N* is the number of TBs into one TBS.

The relationship of TBS_size and maximum bit rate and minimum bit rate and TTI can be given by the following relationships respectively:

$$Max\_TBS\_size = \frac{Maximum\,bit\,rate}{100} * \#\,frames\,per\,TTI$$

$$Min\_TBS\_size = \frac{Minimum\,bit\,rate}{100} * \#\,frames\,per\,TTI$$

where *100* are the radio frames transmitted in one second.

For the TFS selection we need those combinations of *K*, *N*, such that:

$$Min\_TBS\_size \le (K * step) * N \le Max\_TBS\_size$$

$$K * step \le 5000$$

TFCS Derivation

Let's assume that a number of transport channels are multiplexed on a S-CCPCH as shown in Figure 90.

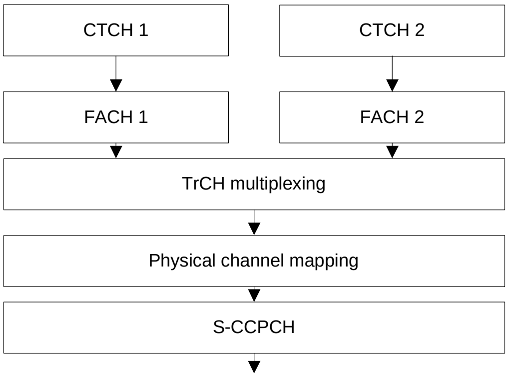


Figure 90: FACH multiplexing and mapping to S-CCPCH

We must then select the possible combinations that satisfy the following equation:

$$\sum_{i=1}^{I} \frac{(TFsize)_i}{(\#of\ \ frames\,per\,TTI)_i} \le 20 * \frac{256*15}{SF_{S-CCPCH}}$$

where i is the i^{th} TrCH and I is the total number of the TrCHs multiplexed on a single S-CCPCH.

The spreading factor *SF* for a S-CCPCH is equal to $256/2^k$, where k = [0...6]. The factor (20 × 256 × 15 / SF) comes from the fact that T_{slot} = 2 560 chips, 20 × 2^k bits per slot and 15 slots per radio frame.

In the SATIN scenario, we have $SF_{S\text{-}CCPCH}$ = 8.

In this case for a single S-CCPCH we have that Bits / frame = 9 600, which yields:

$$\text{Max\_TBS\_size=1440*1=1440 bits}$$

The relationship of TF_size, TB_size, CRC size, Rate Matching, Coding rate, can be given by the following equation:

$$(TFsize)_i = [(TB\_size)_i + CRC_i] * CR_i * RM_i * N / \# \text{ of frames per TTI}$$

Semi static

Dynamic

Therefore, for one S-CCPCH with SF = 8 we must select the possible combinations that satisfy the equation:

$$\sum_{i=1}^{I} \frac{[(TB\_size)_i + CRC_i] * CR_i * RM_i * N}{(\#of \;\; frames\; per\, TTI)_i} \leq 20 * \frac{256 * 15}{8} = 9600$$

where i is the i^{th} TrCH and I is the total number of the TrCH.

This derivation of admissible TFS/TFCS can be:

- refined, in particular considering a minimum bit rate supported by the TFCS (corresponding to the sum of the minimum bit rates of the services mapped onto the S-CCPCH);
- adapted/revised considering a different determination of Min/Max Transport Block Set sizes, since those actually characterize the range of "instantaneous transport rate" (over a TTI), and may not equal the service QoS requirement (guaranteed, maximum bit rate). Maximum instantaneous rate may be higher than the value of the (RAB) service attribute, e.g. in case of discontinuous transmission of NRT services on periodic transport basis (such "a la CBS" CTCH occasions period may be changed during the session, ideally without change in the TFCS so as to avoid S-CCPCH reconfiguration). The instantaneous maximum rate may also be set somehow lower than the maximum bit rate admitted (i.e. RAB attribute), provided adequate buffering for maximum burst size handling.

The valid set submitted for TFC selection shall comply with the requirements on maximum number of TFCs that can be stored in terminals, depending on the terminal class targeted. Complementary elements for appropriate mapping and TFCS determination (aiming at reduced set) should outcome from the packet scheduler process assessment via the analysis of the distribution of the selected TFCs.

F.1.1.3 Admission Control (AC)

General description

Admission control (AC) is responsible for accepting or rejecting new or re-negotiated connections aiming at preserving the required QoS/GoS while making efficient utilization of the network resources.

In UMTS, AC belongs to Radio Resource Management (RRM) and is a function of the Radio Resource Control subsystem. AC process in RNC involves RANAP and RRC layers through the following steps:

1) The Core Network (CN) requests from the Serving RNC to establish a RAB indicating QoS parameters.

2) According to QoS parameters the requested service is assigned a type of service. AC is performed according to the type of service.

3) Resources are allocated according to the result of AC.

4) Acknowledgement is sent back to the CN according to the result of AC. Sublayers are configured accordingly.

Admission and load control, together with the packet scheduler, ensure that the network stays within the planned condition.

Admission control is a process that guards the access to the radio access network. It prevents the "overallocation" of resources. The purpose of admission control is to ensure that there are free radio resources for the intended RAB with required SIR and bit rate. Admission control is performed when a group attempts to set up a new bearer. The control procedure checks whether the requested resources can be granted without causing network overload problems later on. The problem of AC in a W-CDMA system is a complex one, because a new RAB increases the overall interference level in the beam, and thus it has a direct effect on the quality of service of the other user groups, in the service area. The increased interference reduces the effective coverage of the satellite. Thus it may happen that a new RAB in the beam may drop another RAB, or even several RABs, near the beam boundary. Groups may accept the occasional blocking of their RAB attempts, but not the excessive dropping of existing RABs.

Interactions of AC and other SATIN RRM functions

The interaction between these AC and other RRM entities is shown in Figure 91. Their interfaces follow the numbering convention in Figure 89 and are detailed in Table 16.

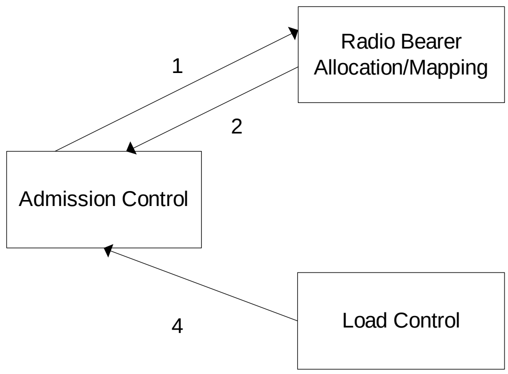


Figure 91: Interactivity between AC and other RRM functions

Table 16: Interactivity of AC with LC and RBAM

Interface #	Events	Parameters	Comments
1	Mode A (fixed RB) - AC notifies RBAM of accepted RAB and AC-derived mapping on pre-configured RBs Mode B (ad hoc RB) - AC notifies RBAM of accepted RAB mapping, and the resources to be allocated (RB configuration)	- RAB traffic characteristics (e.g. guaranteed rate, delay, etc) - RAB/CTCH/ FACH/ S-CCPCH mapping	In mode *A*, AC indicates to RBAM to which (pre-configured) RB the accepted RAB should be mapped. In mode *B*, AC determines the ad hoc RB configuration for the accepted RAB, and instructs RBAM accordingly.
2	Mode A (fixed RB) - RBAM notifies AC of the fixed RB configuration Mode B (ad hoc RB) - Interface not applicable	- FACH/ S-CCPCH mapping configuration	RBAM notifies AC only in the case where there has been a change in the RB configuration
4	LC notifies AC of specific load changes	- Load - Service ID and QoS parameters	LC notifies AC that specific changes have taken place and that the status has been updated (e.g. transition to overload state, drop of bearers)

Finite State Machine

This clause describes the internal working of the AC Functional Block by means of a Finite State Machine (FSM).

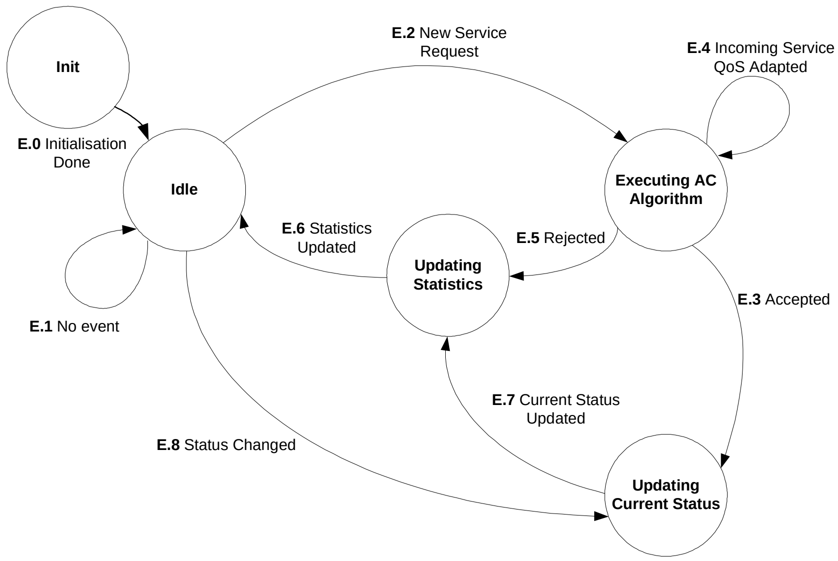


Figure 92: The Finite State Machine (FSM) of the AC functional block

Description of states

- **Init:** all functions to initialize the AC block are performed during this state.
- **Idle:** state at which external requests and notifications are expected.
- **Executing AC Algorithm:** during this state the AC algorithm is being executed and the relevant criteria are being examined, after the arrival of a new service request.
- **Updating Current Status:** during this state the view of the current status is updated to depict the changes following the acceptance of a service request or changes performed by the packet scheduler and/or load control.
- **Updating Statistics:** at this state the AC-specific statistics are updated.

Table 17: AC FSM State transitions

Event-name	Parameters	Description
E.0		All initialization tasks have been completed.
E.1		There is no service request; the system remains in the IDLE state.
E.2	QoS parameters, Service ID	This event indicates that a new service request has arrived.
E.3	QoS parameters, Service ID	This event indicates that the incoming service request has been accepted.
E.4		This event indicates that the AC algorithm execution will be repeated with adaptation of the required QoS.
E.5	Service ID	This event indicates that the service request cannot be admitted in the system due to lack of resources.
E.6		This event indicates that the necessary statistics were updated and the service requestor has been informed of the rejection.
E.7		This event indicates that the current status has been updated successfully.
E.8	Resource allocation configuration	This event indicates a change in the status due to actions taken by the RBAM or Load Control.

Admission Control Strategy

The admission control is the set of actions taken by the satellite network during the phase of service establishment or service re-negotiation to decide whether to accept or to reject service request. A new service request can be accepted only when there are adequate network resources available to guarantee the Quality of Service (QoS) of all existing and the requested services.

In the SATIN satellite WCDMA system all user groups share the common bandwidth and each new user group established increases the interference level of all other user groups affecting their QoS. Based on the baseline and optional service scenarios the possible AC schemes should follow the service characteristics. Assuming that Broadcast/Multicast services on the SATIN architecture require downlink only radio resources, the downlink case of a S-UMTS AC algorithm is considered.

Since the SATIN network is based on a W-CDMA layer 1 architecture, it is useful to analyse the limitations (criteria) that determine the admission or not of a new RAB. The criteria upon which the AC algorithm will grant the radio network resources should satisfy multiple conditions. First of all a feasible algorithm should be based on easily obtainable parameters. These parameters should cover a wide variety of QoS prerequisites and furthermore should be representative of the current status of the network. In a real satellite network the radio network planning and dimensioning normally provide parameters and threshold values easily measured.

For the proposed SATIN AC algorithm the following constraints are used:

- QoS constraints;
- power constraints;
- code constraints;
- rate constraints.

Each multicast service group has an individual QoS requirement, which, in the context of AC, is specified in terms of maximum BER and guaranteed transmission rate, mapped to a required E_b/N_o and a type of service, i.e. one of S_{max} service (traffic) classes. Each such class S_j ($0 < S_j < S_{max}$) comprises a number of subclasses that correspond to different information rates R_{ij} and required $(E_b/N_o)_{ij}$ values.

Each new RAB is assigned to a S-CCPCH according to available channels. A specific SF is assigned to each S-CCPCH and the mapping is decided upon service requirements and available resources.

Let n_i be the n_{th} user in the network, and n_i ranges from 0 to N_i. Let also the required (Eb/No) for the service *j* of the user n_i be denoted by γ_j. This value is common for any user n_i that wants to receive service *j*.

The achieved $\frac{Eb}{No}$ for each user n_i must satisfy the following condition:

$$\left(\frac{Eb}{No}\right)_{j,ni} = \frac{W}{u_j R_j} \frac{P^r_{j,ni}}{P_N + I_{oth_{j,ni}} + I_{own_{j,ni}}} \geq \gamma_j$$

where,

R_j is the rate of service *j*;

u_j is the activity factor of the service *j*;

$P^r_{j,ni}$ is the received power of *nth* user for j^{th} service;

$I_{own_{j,ni}}$ is the own beam interference;

$I_{oth_{j,ni}}$ is the interference from other beams;

P_N is the thermal noise.

The ratio $\frac{W}{u_j R_j}$ corresponds to the respective spreading gain.

We may write as P_{tot}^t the total transmitted power from each carrier for all services and groups and assume that this power is allocated to wanted data channels and overhead channels.

Let us write $P_{tot}^t = P_{wanted}^t + P_{overhead}^t = \beta * P_{tot}^t + (1-\beta)P_{tot}^t$

i.e. a fraction β of the total power is allocated to wanted data channels.

The own beam interference may be written as:

$$I_{own_{j,ni}} = \frac{(1-a_{ni}) \cdot P_{tot}^t}{L_{ni}}$$

where a_{n_i} the orthogonality factor due to multipath, which depends on the n[th] user location and L_{ni} the total path losses.

From now on we will use the notation $h_{ni} = \frac{1}{L_{n_i}}$ as the channel gain. So we rewrite:

$$I_{own_{jni}} = (1-a_{ni})h_{ni}P_{tot}^t$$

For a satellite CDMA system, if we assume a reuse factor of 1, the other beam interference is due to antenna leakage at the borders of the beam area.

Assuming that for all beams we have the same transmitted power, h_{ni} is the same as previously since the two beams are collocated on the satellite, and λ_{ni} is a factor depicting the antenna leakage to adjacent beams then:

$$I_{oth_{jni}} = h_{ni}P_{tot}^t \lambda_{ni}$$

The factor λ_{ni} takes values from 0, for frequency reuse factor greater than 1, up to - theoretically - 1 for antennas without leakage.

The received power P_{jni}^r is given by:

$$P_{j,ni}^r = h_{ni}P_{ji}^t$$

The thermal noise is given by:

$$P_N = n_o W$$

where n_o the noise spectral density of the users' mobile front-end. In the case that we have one service to one physical channel mapping the equation on the achieved $\frac{Eb}{No}$ for each user n_i and the previous equation yield the *QoS constraint*:

$$\left(\frac{Eb}{No}\right)_{jni} = \frac{P_j^t \cdot \frac{W}{u_j R_j}}{P_N / h_{ni} + \lambda \cdot P_{tot}^t + (1-a_j) \cdot P_{tot}^t} \geq \gamma_j$$

Another set of constraints is the power constraints. Assuming the case of fixed/ad hoc RB configuration, and FACH multiplexing over *M* S-CCPCHs, where F_k is the maximum number of FACHs for S-CCPHCH *k* we get:

$$\sum_{k=1}^{M} P_k^t + P_{overhead}^t \leq P_{total\ threshold}^t$$

If $P^t_{overhead}$ is written as $(1-\beta)P^t_{total\ threshold}$, which is the worst-case scenario.

Then $(\sum_{k=1}^{M} P^t_k) \le \beta P^t_{tot}$

$$\sum_{k=1}^{M}\left(\frac{\frac{\gamma_k}{W}}{\sum_{j=1}^{F_k} u_{jk} R_{jk}} \cdot (P^t_{tot} \cdot (1-a+\lambda) + n_o W / h_{ni}) \right) \le \beta P^t_{tot}$$

From this equation and the equation on the achieved $\frac{Eb}{No}$ for each user n_i we obtain:

where γ_k is the $\frac{Eb}{No}$ required to satisfy the most demanding of the services (in terms of BER) on S-CCPCH *k*, a is the average orthogonality of the beam, and λ is the average antenna leakage factor.

Assuming $P^t_{tot}(1-a+\lambda) >> n_o W / h_{ni}$

the inequality becomes:

$$\sum_{k=1}^{M}\left(\frac{\frac{\gamma_k}{W}}{\sum_{j=1}^{F_k} u_{jk} R_{jk}} \cdot (1-a+\lambda) \right) - \beta \le 0 \Rightarrow \sum_{k=1}^{M} \frac{\gamma_k}{SF_k} \cdot (1-a+\lambda) - \beta \le 0$$

which is the AC criterion.

In the case where $P^t_{tot}(1-a+\lambda)\ \ is\ \ not >> \frac{n_o W}{h_{ni}}$

then the AC criterion becomes:

$$(\sum_{k=1}^{M} \frac{\gamma_k}{SF_k}) - \beta / (1-a+\lambda+\frac{n_o W}{h P^t_{tot}}) \le 0$$

Another important constraint is that the total number of codes allocated to S-CCPCHs should be less than the maximum number of codes available at each time instant. This could be expressed either in number of codes or in % of available codes in the SATIN code tree.

The above algorithmic approach holds for both fixed and ad hoc RB mapping. However, specifically for the case of fixed RB mapping one additional constraint that needs to be taken into account is the rate constraint. If $R^{\max}_{jk}$ represents the allowable information rate for $FACH_{jk}$ that is mapped to the k^{th} S-CCPCH physical channel, A^{FACH} is the set of available FACHs (i.e. not reserved by active services) and $R_{new\_service}$ is the requested rate for the incoming service request, then the AC criterion will be satisfied if:

$$\exists FACH_{jk} \in A^{FACH} : R_{new\_service} \le R^{\max}_{jk}$$

Throughput-based approach to S-UMTS downlink admission control algorithm

In this approach, a new RAB is accepted if the resulting overall load would not exceed a predefined threshold. In other words, if n_{DL} is the downlink load factor before the admittance of the new RAB at the w S-CCPCH:

$$\eta_{DL} = \sum_{k=1}^{M} \frac{\gamma_k}{SF_k} \cdot (1 - a + \lambda)$$

then the new RAB will be accepted provided,

$$\sum_{k=1, \neq w}^{M} \frac{\gamma_k}{SF_k} \cdot (1 - a + \lambda) + \frac{\gamma_w}{SF_w} \cdot (1 - a + \lambda) \leq n_{DL\_threshold}$$

where $n_{DL\_threshold}$ is the load factor threshold set by the RNP process.

F.1.1.4 Load control

General description

Load control monitors, detects and handles situations when the system reaches near overload or an overload situation while RABs remain actives. Therefore, when somewhere in the network limited resources degrade service quality, load control brings the system back and restores stability seamlessly.

The purpose of load control is to maintain in an objective manner the use of radio resources of the network within the given limits. Load control measures a load factor, and, if the predefined load factor is exceeded, the network reduces the bit rates of those RABs whose service contract allows it to be done. Like admission control, load control attempts to manage the resource usage in the telecommunication network so that it will not become overloaded. The difference between these control procedures is that admission control is an on-off procedure, whereas load control is a continuous one. The admission-control process considers each resource request as a separate case, whereas load control manages the network as a whole. Admission control decides whether a new RAB can be set up or not; load control then monitors and manages the existing RABs.

If the admission control process could predict the changes in the amount of traffic and interference levels, it could decide whether a new bearer could be set up or not. However, in practice this is not always possible. There are services that can generate highly varying data rates, and the actual amount of traffic cannot be predicted when the service is requested. The interference level in the beam can change over time.

The load control and packet scheduler are tightly coupled. Since the packet scheduler does not guarantee the delays of the non-real-time connections, the load of non-real-time packet traffic can be controlled. If the load of real time users gets too high, the packet scheduler can decrease the load of the controllable non-real-time users.

Interactions of LC and other SATIN RRM functions

The interaction between LC and AC, PS is shown in Figure 93. Their interfaces follow the numbering convention in Figure 89 and are detailed in Table 18.

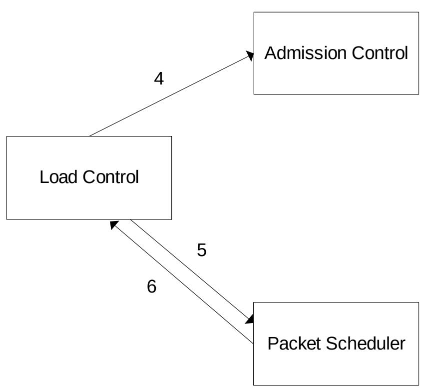


Figure 93: Interactivity between LC and other RRM functions

Table 18: Interactivity of LC with PS and AC

Interface #	Events	Parameters	Comments
4	LC notifies AC of specific load changes	- Load - Dropped RAB ID	LC notifies AC that specific changes have taken place and that the status has been updated (e.g. transition to overload state, drop of bearers)
5	LC instructs PS of specific action	- Rate - Priority/weight - Service ID	LC decides on the actions to be taken following a transition to a preventive/reactive congestion state and delegates the instruction to the PS
6	PS notifies LC that action was performed		PS acknowledges that specific action has been taken

Finite State Machine

This clause describes the internal working of the AC Functional Block by means of a Finite State Machine (FSM).

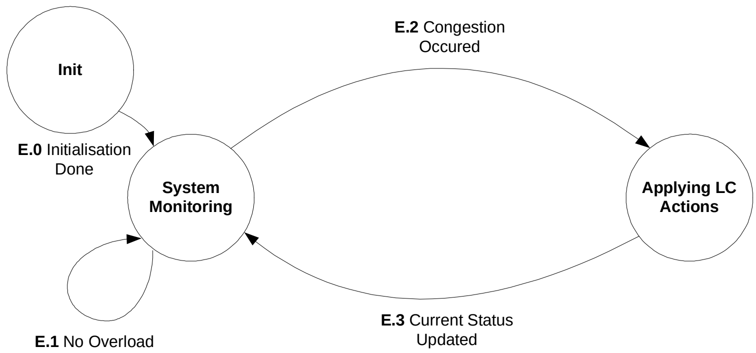


Figure 94: The Finite State Machine (FSM) of the LC functional block

Description of states

- **Init:** all functions to initialize the LC block are performed during this state.
- **System Monitoring:** during this state LC is monitoring the load of the system and is prepared to take actions should a threshold crossing occurs.
- **Applying LC Actions:** during this state LC, following a overload threshold crossing decides on the necessary actions to be performed by the PS so that the system may revert to a stable state.

Table 19: LC FSM State Transitions

Event-name	Parameters	Description
E.0		All initialization tasks have been completed.
E.1		No overload condition is being detected. The system remains in this state.
E.2	Amount of overload	This event indicates that the load threshold has been exceeded and appropriate actions should be taken to rectify the situation.
E.3	Updated system status	This event indicates that the necessary actions have been completed and the system status has been updated.

Load Control Strategy

The main function of load control can be divided into two tasks. In normal circumstances LC takes care that the network is not overloaded and remains in a stable state. To achieve this LC works closely with AC and PS. This task is called preventive load control. In very exceptional situations, however the system can be driven into an overload situation. Then overload control is responsible for reducing the load relatively quickly and thereby bringing the network back into the desired operating area defined by the Radio Network Planning (RNP).

The AC and PS functions perform together the preventive load control, LC working as mediator between these two functions. LC updates beam load status based on radio estimations and measurements provided by the AC and PS. If the beam is in the normal state, AC and PS can work normally. If the load exceeds the target but is less than the specified overload threshold, preventive load control actions are performed. AC admits only new RT bearers. The PS does not increase further the bit rate of the admitted NRT bearers. If the beam moves to an overload state, the PS starts to decrease the bit rates of NRT bearers taking into account the bearer classes and the priorities set by the operator within the same traffic class. In the most extreme case NRT and RT bearers might even be dropped.

A load control algorithm requires the estimation of a descriptive parameter, the load factor, depicting the availability of radio resources. In SATIN, a throughput-based load estimation is proposed.

Throughput-based downlink load control

In this case, a load factor can be estimated which will be given by the sum of the bit rates of all currently active RABs divided by the maximum possible throughput for the beam:

$$n_{DL} = \frac{\sum_{k=1}^{N} R_k}{R_{\max}}$$

where R_k is the bit rate of RAB k and N is the total number of RABs.

It is also possible to estimate the load factor as:

$$\eta_{DL} = \sum_{k=1}^{M} \frac{\gamma_k}{SF_k} \cdot (1 - a + \lambda)$$

In order to detect when the system has entered the preventive or the overload state, two thresholds need to be defined, respectively:

- $n_{preventive}$;
- $n_{overload}$.

where $n_{preventive} < n_{overload}$

Therefore,

if $n_{DL} \le n_{preventive}$

Normal State

else if $n_{preventive} \le n_{DL} < n_{overload}$

Preventive State

else if $n_{DL} \ge n_{overload}$

Overload State

The thresholds $n_{preventive}$ and $n_{overload}$ are set by the RNP process.

F.1.1.5 Admission and Preventive Load Control

In this clause an admission control strategy coupled with a preventive load control mechanism is proposed. It is an admission control strategy that implements a preventive load control algorithm aiming at determining the admissible set of TBSs that can be supported by the system, ensuring the required QoS. This combined strategy can be applied to cases where more than one FACH are multiplexed on S-CCPCH. Every time a new service request is examined for admission to the system, the algorithm looks for all eligible TFS (see note):

$$Min\_TBS\_size \leq (K * step) * N \leq Max\_TBS\_size$$

where,

$$K * step \leq 5000$$

NOTE: The equations quoted in this clause are detailed in clause F.1.1.2 "A generic approach for TFS and TFCS derivation".

The algorithm subsequently calculates the allowable combinations based mainly on the total information bit rate that may be carried by a S-CCPCH. The algorithm checks the possible TFCs (step 2) including the new session as if it has already been accepted, according to the following equation:

$$\sum_{i=1}^{I} \frac{(TFsize)_i}{(\#of\ \ frames\ per\ TTI)_i} \leq 20 * \frac{256 * 15}{SF_{S-CCPCH}}$$

In step 3, AC checks the allowable TFCs selected from step 2 against the AC criteria. AC provides the allowable combinations to RBAM, which in turn provides the corresponding TFCS to the MAC part of PS (it is possible that the resulting TFCS is a reduced set following the application of additional constraints). This selection can be seen as the fast part of the radio resource control dedicated to MAC, close to L1. Thereby the bit rate can be changed very quickly and with no need for L3 signalling.

If the AC criteria are satisfied then preventive LC takes over (step 4) and checks what would be the total load for each TFC if the new session had been accepted. If this load criterion is also satisfied (comparison is done between the sum of loads from all services and a load threshold normally
expected from the planning process) then the session is accepted and the selected TFCS is available to PS via RBAM. The above process would ideally ensure that the system would not enter a congested state (depending on how conservative or optimistic the selection of the load threshold value is).

The above procedures are depicted in Figure 95.

It should be noted that there is more than one approach for AC to decide whether to accept or reject the new request as previously described in the admission control strategy clause.

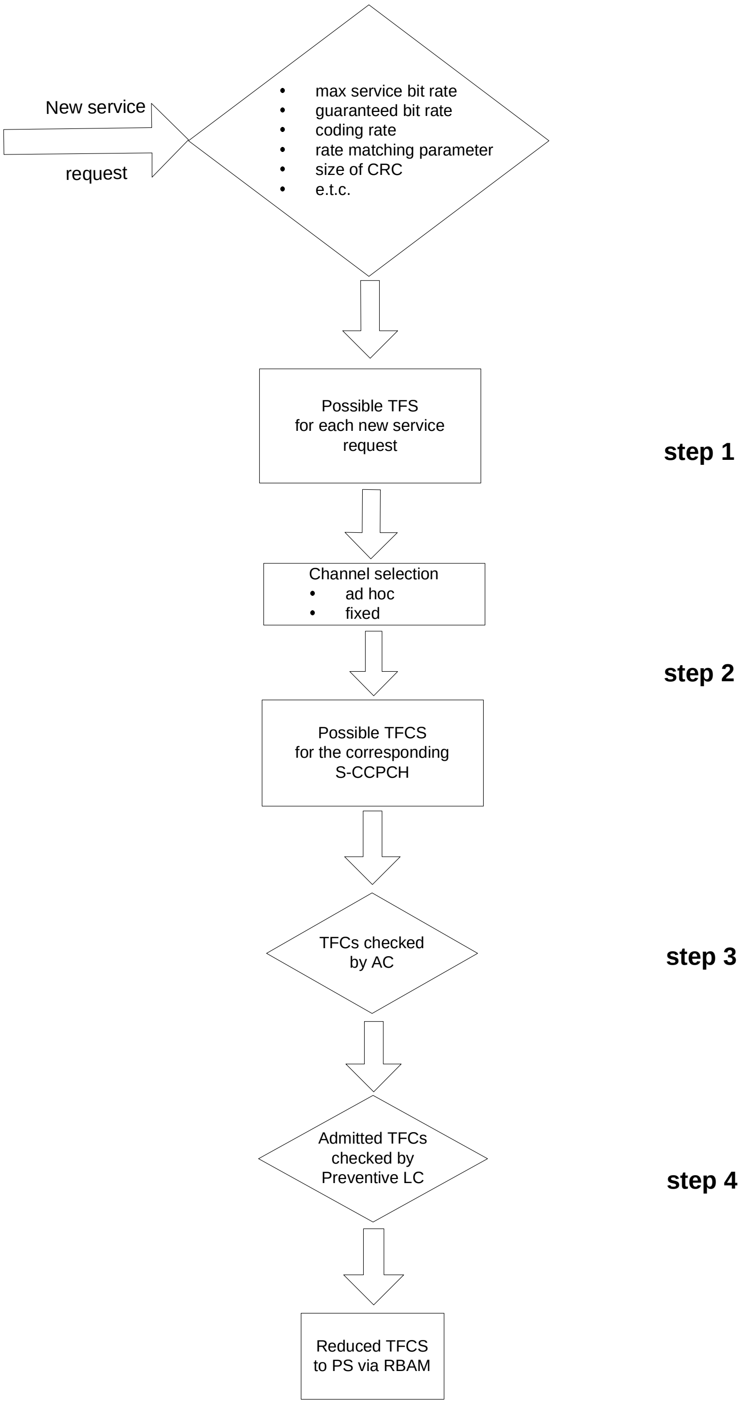


Figure 95: Admission and Preventive Load Control strategy

F.1.1.6 Packet scheduler

The packet scheduler is another critical entity of the SATIN radio interface. Given the absence of the power control mechanism, the PS becomes the main mechanism of fast resource allocation with a significant part in the fulfilment of QoS objectives for the transferred services.

General description

The main part of the scheduler resides - functionally - in the RRC entity, where most of the information for resource management is located. This part is aware of the current radio resource allocation at the forward link. There is another part at the MAC layer, instantiated by the priority handling and transport format selection functions. The respective functions are configured by the respective RRC module and turn its decision into action. There is one packet scheduling entity per beam, controlling the traffic - particularly its delay-tolerant portion - destined for it.

Interaction of PS with other SATIN RRM functions

The dependence of the scheduler on other blocks and functional entities of the radio interface are depicted in Figure 96. Their interfaces follow the numbering convention in Figure 89. In the following clause the interfaces and dependencies of the scheduler are identified.

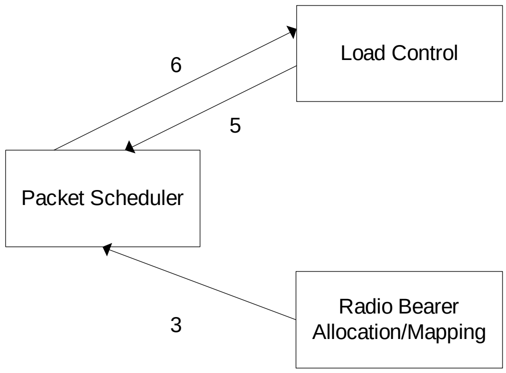


Figure 96: Interaction of PS with RBAM and LC

LC-PS interface

Given the availability of measurements and the Load estimation module, LC might reconfigure the PS. Instructions passed to the PS might be:

- Reduce/increase the rate of one or more CTCH/FACH pairs or FACHs.
- Change some priority /weight (temporarily).

PS-LC interface

The main requirement is that the PS and LC share knowledge about actions taken from the one or other module. In this context, exchange of some information/notification messages might be necessary.

Availability of -at least- two sets of measurements is assumed at the RRC:

- Traffic load (RLC/CTCH queues) provided by MAC layer, either in periodic or event-triggered mode.
- Physical layer measurements, like transmit power per code/physical channel. Its relevance is discussed below.

Both measurements are provided via standard primitives.

RBAM-PS interface

The mapping of the FACHs to S-CCPCHs, as well as the TFCS available to S-CCPCHs, are passed to the scheduler by the RBAM block.

The packet scheduler needs to know the traffic characterization of the stream accepted (guaranteed /effective rate and possibly delay requirements, e.g. max delay/jitter allowable).

The packet scheduler also needs to be aware of the QoS rank (in accordance with UMTS QoS traffic classes) of the service and the radio bearer (CTCH/FACH pair) to which the service is/was mapped to, in order to modify - if it is deemed appropriate - the priorities of the time scheduling function.

Packet Scheduler Strategy

The Packet Scheduler is responsible for the short-term resource allocation. It distributes the available resources, i.e. rates and power in the generic case, under the constraints imposed to it by the AC and/or the LC functions. Given the fixed SF of the code channels, the major task of the scheduler becomes the time scheduling of the different FACHs that have been mapped to a single S-CCPCH. The power transmitted per code channel is set to a specific level that can guarantee some QoS measure (usually of statistical nature) for all services (i.e. FACHs) carried from the code-channel. It may vary over intervals equal to the duration of a flow, but not at the level of Transmission Time Interval. In fact, the power scheduling check may only be made once, when the service is admitted or may even be taken into consideration once in the dimensioning of the system (number of code channels supported per beam) and not be repeated again in the more dynamic stages of RRM.

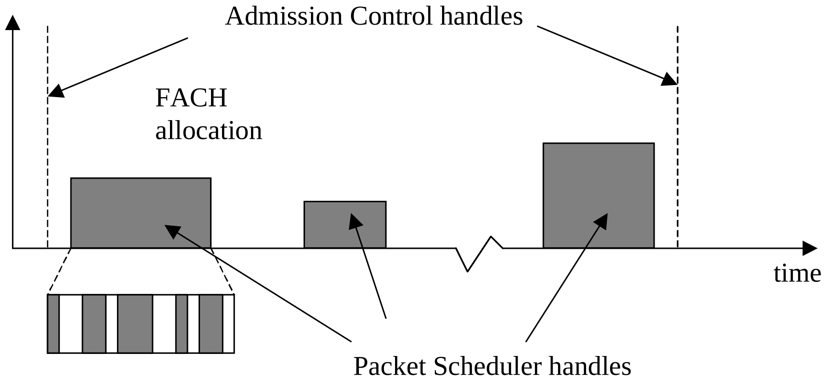


Figure 97: Admission control and Packet Scheduler timescales of action

Three steps can be identified in the Packet Scheduling procedure that is executed for each data traffic S-CCPCH periodically with a period dictated by the smaller TTI of the supported transport channels (Figure 98):

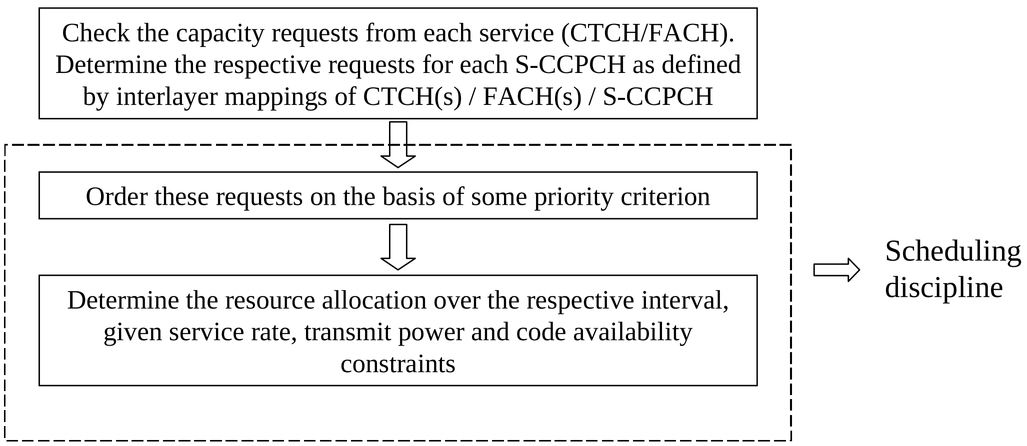


Figure 98: Packet scheduling procedure

Step 1: Identification of current capacity demand

The capacity demands are treated separately for each data S-CCPCH by the scheduler. In each resource allocation interval (TTI) the scheduler examines all the active code channels, i.e. physical channels that carry traffic at that specific moment and estimates the requested capacity demand per transport channel mapped to a physical channel.

Step 2: Prioritization of different demands

The demands identified in the first step are subsequently ordered according to some priority criterion. In selecting the respective criteria, the service attributes are considered (e.g. delay/delay variation tolerance, rate requirements, etc). Besides the higher-level traffic handling priorities defined by 3GPP on the basis of UMTS QoS classes, further differentiation of services may arise, e.g. on the basis of delay ranges that may be tolerated or bit rates that have to be guaranteed.

This prioritization may be more or less dynamic. In the former case the relative priority of the different channels may change in each resource allocation interval, depending for example on the maximum delay tolerated by a service and the number of packets buffered at the respective CTCH. In the latter, the priority may be more static, e.g. as long as there are packets of service x, they are always served first ("blind" preemption).

In the case of one-to-one mapping of FACH to S-CCPCH the prioritization function is redundant, i.e. from the S-CCPCH in question the scheduler will forward the maximum possible TB Set Size supported by the TF Set of the channel.

Step 3: Selection of TF(C)s for the specific resource allocation interval

Finally, on the basis of the two previous steps, the exact TFC is determined for each active (see note 1) physical channel (S-CCPCH). These TF(C)s are selected among the available TF(C) sets that are passed to the scheduler during the admission of a new service and its mapping on a specific bearer. This step has to take into consideration constraints in terms of service requirements (e.g. minimum guaranteed rate, maximum tolerated delay) as well as system-level constraints (load, transmit power per beam).

NOTE 1: The term active denotes a physical channel, upon which one or more services (flows) have been mapped. An active S-CCPCH may not carry data over a specific TTI, if for that specific TTI the respective CTCH do not have (buffer) any data, e.g. as a result of temporary traffic source idleness.

Effectively the steps 2 and 3 jointly constitute the scheduling discipline of the packet scheduler; they differentiate one from another and define the capability of each one to marry the service QoS requirements with an efficient system resource utilization.

In the following the basic principles of the scheduling function in the SATIN case are described. Then two scheduling disciplines, constituting direct analogs of scheduling disciplines in wired networks are presented.

Generic formulation of the scheduling problem

The scheduler treats independently at TTI level each physical channel. The exact number of physical channels at a specific time instance and the corresponding mapping of transport channels onto the code channels are defined by the RBAM (see clause F.1.1.2).

In formulating the scheduling function, the following notation is necessary:

i indexing notation for the physical (S-CCPCH) channels, $1 \le i \le M$

j indexing notation for the j^{th} FACH mapped to the i^{th} S-CCPCH, $1 \le j \le N_i$, where N_i is the number of FACHs mapped to the i^{th} S-CCPCH.

k indexing notation for the k^{th} TBSet size supported on j^{th} FACH of i^{th} S-CCPCH, $1 \le k(i,j) \le K(i,j)$, where *K(i,j)* is the maximum number of different TBSet Sizes supported per FACH. We assume that these TBSet Sizes are sorted in increasing order, namely for each FACH channel *j*, mapped to S-CCPCH *i*:

$$TBSet\ size(i,j,k) \quad \le TBSet\ size(i,j,k+1),\ 1 \le k \le K(i,j)-1$$

NOTE 2: TBSet size (i,j)* notation for the selected TBSet size.

Rmax maximum number of bits passed within a TTI from MAC to Layer 1, related to the (semi-static) exact RB configuration (SF, coding, rate matching etc).

l index of the possible TFCs that are made available to the PS from the RBAM block, $1 \le l \le L$. Each TFC *l* features a certain TBSet Size *k* for each one of the N_i FACH channels mapped to the S-CCPCH *i. L* is the maximum number of TFCs in the TFCS, obeying the limitations defined in TS 25 302 [28], TS 125 133 [30] and TS 125 306 [27].

Each TFC *l* corresponds to a certain number of bits R_l passed from the scheduler to the Layer 1. The task of the scheduler is to select within each TTI some "appropriate" TFC. The actual context of the term "appropriate" is dictated by the service QoS requirements and differentiates the one scheduler from the other. This differentiation is summarized in the term scheduling discipline, i.e. the way the semi-statically fixed capacity of each physical channel is time-shared among the different FACHs.

In the following, two of the possible disciplines are analyzed further. Both of them are adaptations of well-known scheduling disciplines that have been used for years in the context of wired networks. In fact, given the semi-statically fixed SF of the physical channels, which limits the relevance of code scheduling, the fast resource allocation task within the SATIN context is a direct adaptation of the respective concepts in the wired networks.

Multilevel priority with exhaustive service scheduling

This is effectively the adaptation of the non-preemptive priority discipline to the W-CDMA context. In our case a CTCH queue at the RLC level may carry one service/flow. This scheme favors the high priority classes, being able to assure minimum delay for its packets, at the potential expense of an increased delay for lower priority classes.

Depending on the mapping of different services (CTCHs) at transport and physical layer, each S-CCPCH carries one or more services. The N_i FACHs mapped to a single code channel are ordered from 1 to N_i, according to their priority. The usual convention is followed, i.e. a lower order number implies a higher priority.

The choice of the proper TFC for a given TTI includes some search over the possible TFCs envisaged within the TFCS of the code channel.

The scheduler first seeks to allocate the maximum TBSet size to the first FACH:

If the queued data (in bits) are more than the maximum supported TBSet Size (in bits) for this FACH in the TFCS, the selected TBSet size will be the maximum one available in the TFCS. Otherwise, the selected TBSet size is the minimum available in the TFCS that can serve the queued bits. In mathematical notation this may be expressed via the following condition:

$$if\ \ queuesize(1) > TBSet\ size(i,1,K(i,1))$$

$$TBSet\ size(i,1)^* = TBSet\ size(i,1,K(i,1))$$

else (1)

$$TBSet\ size(i,1)^* = TBSet\ size\ (i,1,n^*),\ n^* = \min\{l: TBSet\ size(i,1,l) \geq queuesize(1)\}$$

Out of the whole TFCS, a reduced TFCS is derived:

$$reduced\_TFCS = \left\{ \bigcup_{l\ \in\ TFCS} TFC^l : TBSet\ \mathrm{size}(i,1,l) = TBS\_size(i,1) \right\}$$

The procedure is repeated recursively for each one of the N_i-1 remaining channels, till the single feasible TFC is selected, namely for each FACH *j*:

$$if\ \ queuesize(j) > TBSet\ size(i,j,K(i,j))$$

$$TBSet\ size(i,j)^* = TBSet\ size(i,j,K(i,j))$$

else (2)

$$TBSet\ size(i,j)^* = TBSet\ size\ (i,j,n^*),\ n^* = \min\{l: TBSet\ size(i,j,l) \geq queuesize(j)\}$$

where the search over the set of Transport Format Combinations is limited to the ones of the reduced TFCS that came out of the previous step.

When more than one CTCHs, of the same priority, are multiplexed - at transport or physical layer (via FACHs) - on a single S-CCPCH, the channels may be served in round-robin mode.

Only after a higher priority channel is served, under the condition that (1) and (2) are satisfied and within the bounds set by the allocated TF(C)S, may a lower priority channel pass packets to the physical layer for transmission. Figure 99 provides an outline of the algorithm. Obviously this discipline has much in common with the non-preemptive priority discipline, which is common place in wired networks; and as a consequence, it inherits its advantages and disadvantages as well; namely, the discipline favours the higher priority channels, guaranteeing high rates and low delays to them at the expense of the lower priority channels, for which no delay bounds can be provided.

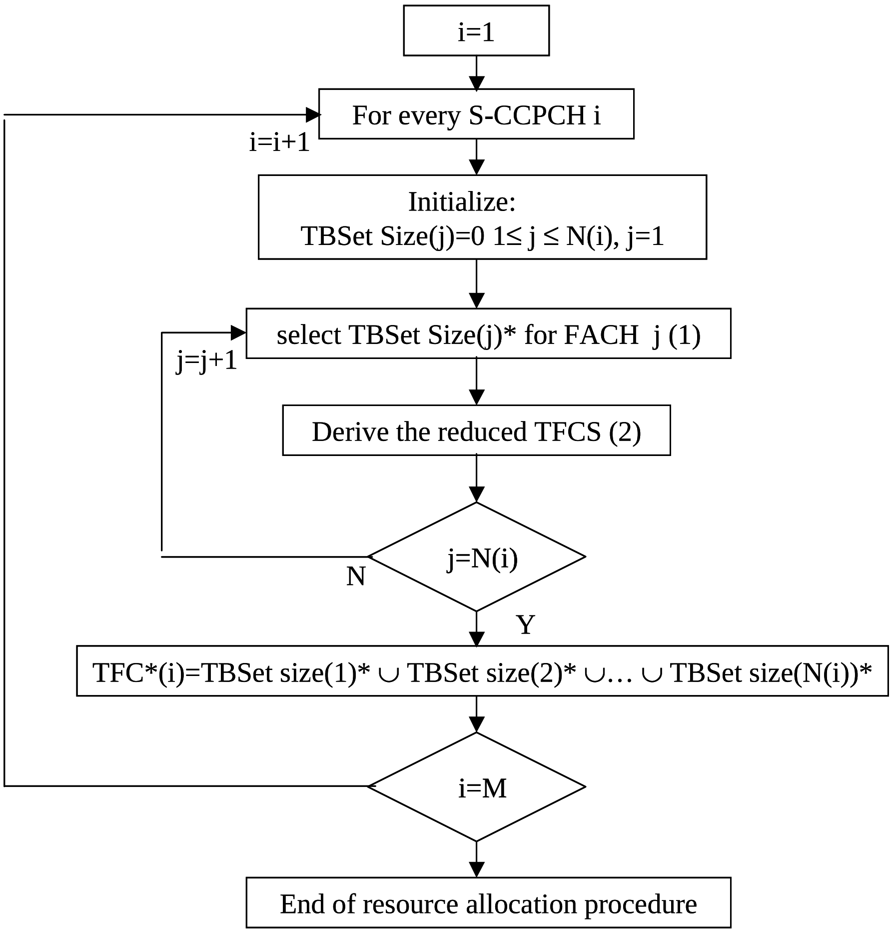


Figure 99: Outline of the SATIN-adapted version of multilevel priority with exhaustive service discipline

Weighted Fair Queuing (WFQ)-based scheduling

The main features of the WFQ algorithm are the following:

- WFQ guarantees a minimum bandwidth per bearer/flux or per set of bearers/flux grouped together for traffic handling purpose.
- WFQ provides a nice and efficient solution for simultaneously multiplexing real time services and high-speed data services. For real time traffic with time delay constraints, jitter is low, is bounded and is always less than the one generated by a simple FIFO queue model.
- The most significant feature of WFQ is fairness: redistribution of the free bandwidth is performed proportionally to the weight assigned to each connection; in other words, WFQ implies sharing available bandwidth between active bearers/flux in proportion to their rate parameters r_i.
- WFQ is work conserving: scheduler will never miss an opportunity to serve a packet, i.e. scheduler is kept busy as long as there are active radio bearers and buffer is not empty.

The proposed WFQ scheduler is more particularly based on the *Virtual Spacing* policy that uses the notion of *Virtual Time* (see [30]), and involves the following parameters:

r_i.: spacing rate or "weight" associated with bearer/CTCH i (T_i =1 / r_i),

corresponds to the share of the multiplex capacity allocated to the RLC queue;

$TSTP_i$.: Time Stamp associated with bearer/CTCH i,

tags the packets at their arrival and is used to order the scheduling;

TV: Virtual spacing Time of the system,

The Time Stamp of the last packet sent out of the queues,

i.e. last packet served or being served.

The weights are primarily set according to the rates of the services multiplexed (with respect to the bandwidth-sharing nature of WFQ). Such a distribution can be adapted whenever necessary: update of Radio Resource assignment (subsequent to service admission or new mapping, dynamic bearer control according to RLC buffer thresholds), dynamics of CTCH periods of allocation. The spacing rates modification is applied to the packets not served (i.e. new packets as well as packets stored).

The virtual spacing scheme is applied simultaneously to all active radio bearers of the multiplex and consists of three functional blocks:

- reception process;
- scheduler emission process: how to manage emission from the scheduler;
- re-emission process: what to do with a departing packet.

Reception process

When a packet arrives, it is stored in the packet memory buffer, as illustrated in Figure 100. The Time Stamp (TSTP) of the packet is computed and the address of the packet is written in the corresponding line of the scheduler.

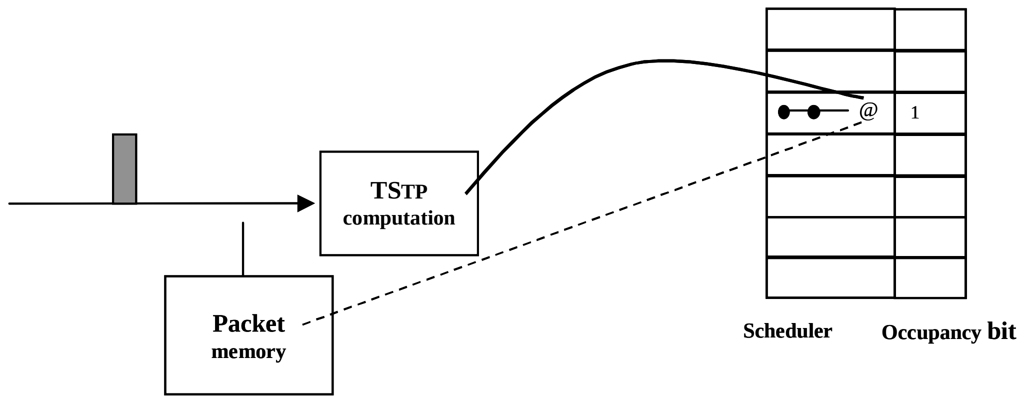


Figure 100: Virtual spacing: arrival of a packet

Thus the reception process is described by the following algorithm (Figure 101).

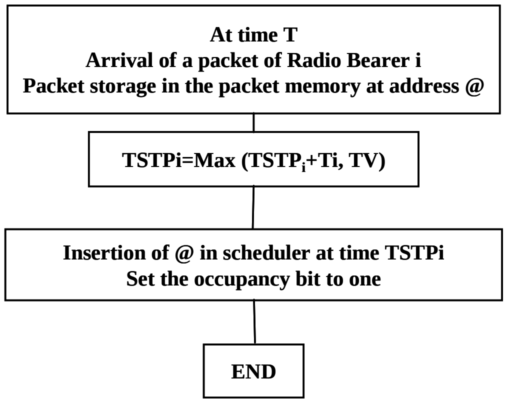


Figure 101: Virtual spacing: reception algorithm

Scheduler emission process

This process is described in Figure 102 and Figure 103. At each frame the scheduler emission process, i.e. delivery of packets to a global Linked List of addresses of packets to be re-emitted (LLr), is enabled/disabled by the status of this LLr: if the LLr is empty (or about to become empty), the lowest time (stamp) in the scheduler where there are packets to re-emit is "scanned". The linked list of address of packets to be re-emitted in that line of the scheduler is added to the global re-emission Linked List associated to the scheduler (LLr). After emission from the scheduler the occupancy bits are reset accordingly and the Virtual spacing time of the system (TV) is set to the Time Stamp of the scheduler line that has been delivered ($TV = TSTP_{new} = TSTP_{lowest}$).

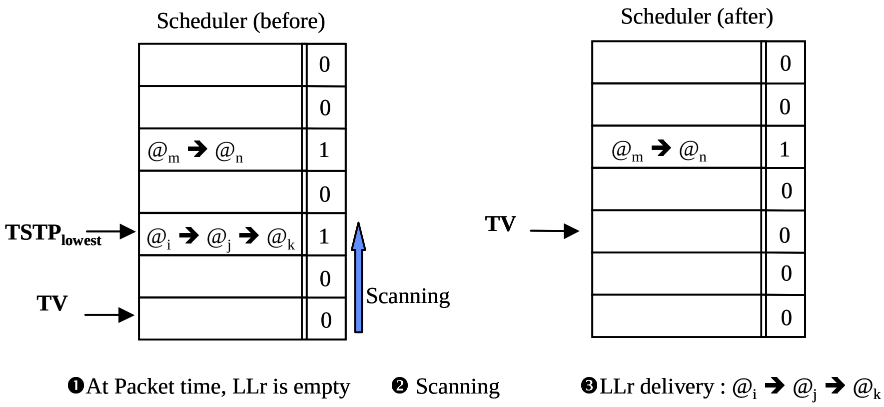


Figure 102: Scheduler emission process (description)

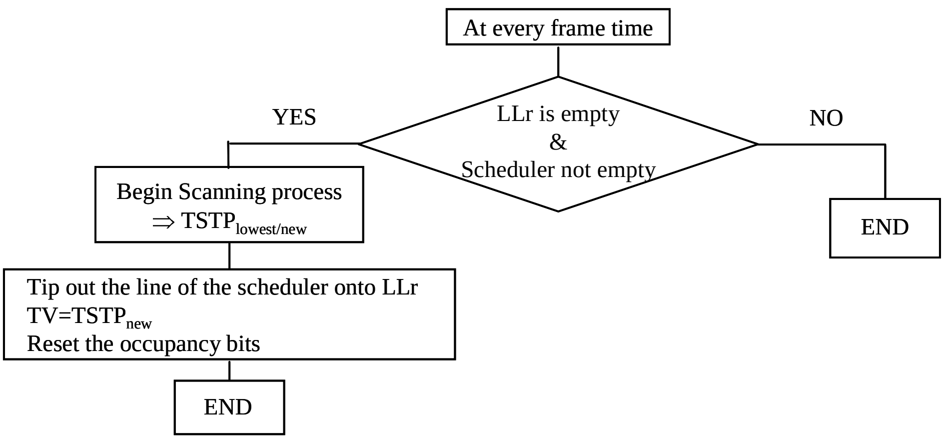


Figure 103: Scheduler emission process (algorithm)

The delivery process has to be done in such a way that no service slot (transmission interval) be ever lost due to any lack of synchronization between the processes.

The so-called scanning process or equivalently the Time Stamp sorting problem (see [30] for details) is the heart of the implementation issues as regard Weighted Fair Queueing policies in general, as the time constraint becomes stringent for high data rates (Mbits/s).

Re-emission process

There is a unique global re-emission Linked List (LLr) associated to a scheduler. The re-emission process is described below.

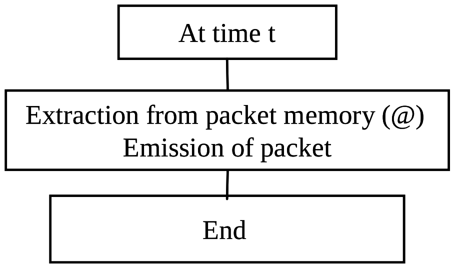


Figure 104: Re-emission process

Synchronization between processes

At each packet time (of the transmission channel), it is important that the processes be executed in the following order.

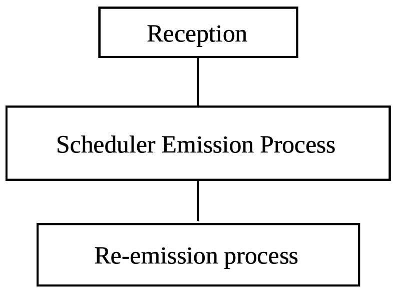


Figure 105: Synchronization between processes

F.1.1.7 RRC interactions with lower layers

RRC and BMC/RLC/MAC interactions are characterized by the following:

- UEs MAC, RLC and physical layers are configured based on unidirectional RRC messages from RNC; elements for lower layers configuration are notified to UEs in two steps: first System Information Broadcast allows reception of in-band service signalling (passed by BMC messages) onto a "system master CTCH/FACH" (to remain available to all UEs in the cell) which then provides service-to-channel mapping, hence allows protocol stack configuration for reception of any other CTCH of interest. Discontinuous reception functionality is combined with BMC resource allocation and scheduling.

- RRC functions are restricted to the following subset: establishment, reconfiguration and release of Radio Bearers, broadcast of information provided by the non-access stratum, (CN), broadcast of information related to the access stratum, integrity protection, CBS-related functions; and besides cell selection which need to be adapted to spot selection (the latter actually addressing upper layers and Layer 1).

- RRM strategy shall avoid reconfiguration signalling, and more particularly limit the need for *System Information change* (i.e. as regards common channel configuration). Possible change of the CTCH occasions (non-static Level 1 scheduling) and when necessary new service-to-channel mapping allows some adaptation of resource allocation with limited number of transport/physical channel reconfiguration. Basically rate adaptation is performed by TFC selection (variable Transport Block sizes as set in the TFCS of the physical channel configuration). Dynamic radio bearer control is performed in RRC, based on the traffic volume measurement reported by MAC: MAC reports to RRC the traffic volume status of each logical and transport channel, and therefore RRC can take proper action for new radio bearer configuration accordingly.

RRC interactions with BMC

RRC and BMC interacts by means of primitives.

RRC interactions with RLC

As it is done in the precedent clause, interactions between RRC and RLC are achieved by means of primitives.

RRC interactions with MAC

As it is done in the two precedents clauses, interactions between RRC and MAC are achieved by means of primitives.

F.1.2 BMC sublayer specifications

The SATIN BMC sublayer features a subset of the full BMC functionality described in [22].

F.1.2.1 Model of the SATIN BMC sublayer

Broadcast/Multicast Control (BMC) is a sublayer of L2 that exists in the User-Plane only. It is located above RLC. Figure 106 shows the model of the L2/BMC sublayer within the SATIN radio interface protocol architecture.

The BMC sublayer shall consist of one BMC protocol entity per broadcast/multicast service. Each BMC entity requires a single CTCH, which is provided by the MAC sublayer, through the RLC sublayer. The BMC requests the Unacknowledged Mode service of the RLC.

A BMC protocol entity serves those messages of the respective broadcast/multicast service at BMC-SAP, that are to be broadcast into a spotbeam (towards all users or certain group).

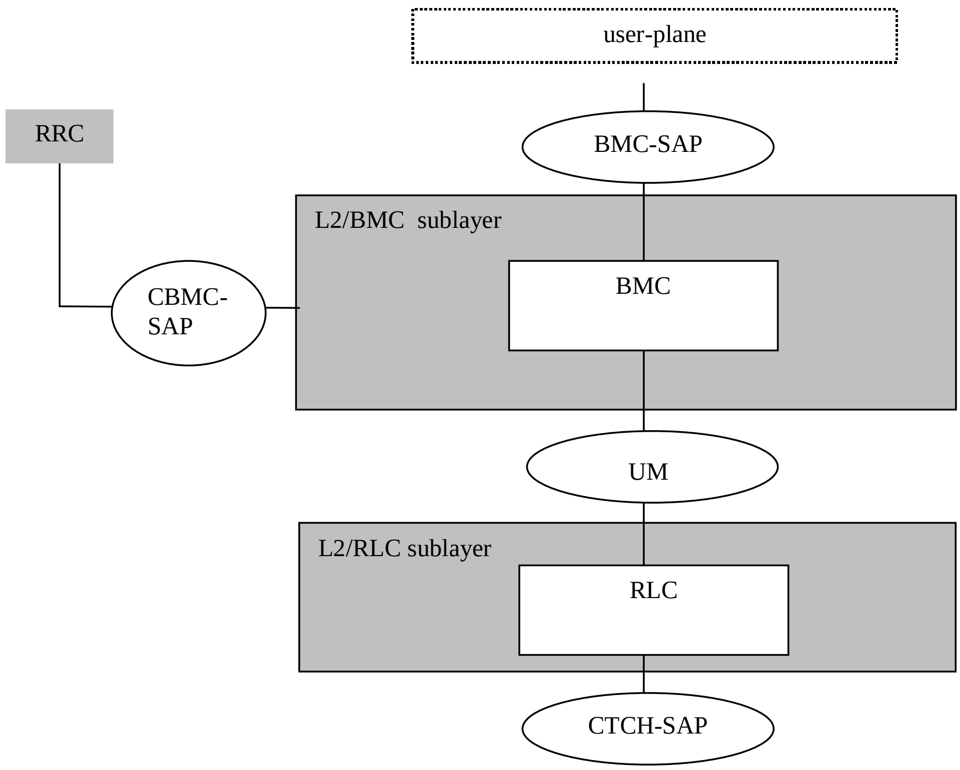


Figure 106: BMC protocol model

F.1.2.2 Functions

The functions supported by the SATIN BMC sublayer are:

- storage of BMC SDUs (Cell Broadcast Messages extended to MBMS packet bursts);
- traffic volume monitoring and radio resource request for broadcast/multicast services;
- scheduling of BMC messages;
- transmission of BMC messages to UE;
- delivery of BMC SDUs (Cell Multicast/Broadcast messages) to upper layer (NAS).

Four types of BMC messages are required: CBS message (which is used for service announcement, i.e. list of current and forthcoming services) and schedule message as specified in the procedures of BMC [22], and two new types of message from USRAN to UEs:

- *Notification message,* that consists of all service notification information elements for the cell/spotbeam: service-to-channel mapping, Level 1 scheduling parameters associated to each CTCH/S-CCPCH. It is generated/processed by the BMC peer-entities operating the global service control and notification (also referred to as system master CTCH/FACH) and further processed by UEs RRC.
- *Notification change message*, that informs the BMC peer-entity (and further the UE RRC) operating a given MBMS service of the forthcoming change of the associated notification information, and indicates which parameter(s) are updated and their respective new value(s) (new service-to-channel mapping and/or new Level 1 scheduling parameters) and the time for their applicability (System Frame Number).

Similar to BMC *Schedule Message,* the *notification* and *notification change* messages begin with their respective *message type* Information Element (IE), the coding of which is TBD among the values reserved for future use in [22]. Also a value of *message description type* (included in the message description IE of the *schedule message*) remains TBD among the values reserved for future use in [22]).

F.1.2.3 Services provided to upper layers

The BMC-SAP provides a broadcast/multicast transmission service in the user plane on the radio interface for common user data in unacknowledged mode.

The BMC sub-layer interacts with other entities as illustrated in Figure 106. The interactions with the upper layer/U-plane and the RRC layer are specified in terms of primitives where the primitives represent the logical exchange of information and control between the BMC sublayer and higher layers.

Three types of primitives are used as follows:

- REQUEST

This type is used when a higher layer is requesting a service from a lower layer.

- INDICATION

This type is used by a lower layer providing a service to notify its higher layer of activities concerning that higher layer.

- CONFIRM

This type is used by a lower layer providing the requested service to confirm to the higher layer that the activity has been completed.

The primitives defined below are for communication between upper layer and BMC, as well as RRC and BMC in the same protocol stack.

For the BMC sub-layer two sets of primitives are defined.

- **Primitives between BMC and upper layer (U-plane):**

  BMC - Generic name - Type: Parameters.

- **Primitives between BMC and the RRC entity:**

  CBMC - Generic name - Type: Parameters.

F.1.2.4 Services expected from RLC

The BMC uses the Unacknowledged Mode service of the RLC sublayer (see clause F.1.3).

F.1.2.5 Elements for layer-to-layer communication

Service Primitives between RRC and BMC

Primitives

The primitives supported at CBMC-SAP between RRC and BMC are shown in Table 20.

Table 20: Primitives between BMC and RRC

Generic Name	Parameters
CBMC-Measurement-IND	CB-Traffic-Volume
CBMC-Rx-IND	Action, DRX selection
CBMC-Config-REQ	CTCH configuration
CBMC-Notif-IND	Services notification
CBMC-Notif-REQ	Services notification
CBMC-Notifupdate-IND	Service-to-channel Mapping, Level 1 scheduling, Time of change
CBMC-Notifupdate-REQ	Service-to-channel Mapping, Level 1 scheduling, Time of change

CBMC-Measurement-IND

The CBMC-Measurement-IND primitive is used by BMC to possibly indicate the service traffic volume.

Primitive Type: indication.

Parameters:

CB-Traffic-Volume.

CBMC-Rx-IND

The CBMC-Rx-IND primitive is used by BMC to indicate to RRC whether CB message reception shall start or stop and indicate when CB messages of interest are arriving in the next CBS/MBMS schedule period.

Primitive Type: indication.

Parameters:

Action.

DRX selection.

CBMC-Config-REQ

The CBMC-Config-REQ primitive is used by RRC to inform the BMC about the setting of the CTCH configuration.

Primitive Type: request.

Parameters:

CTCH configuration.

CBMC-Notif-IND

The CBMC-Notif-IND primitive is used by the system master BMC (in UEs) to inform RRC about each service-to-channel mapping and associated Level 1 scheduling parameters.

Primitive Type: indication.

Parameters:

Services notification.

CBMC-Notif-REQ

The CBMC-Notif-REQ primitive is used by the RRC (in RNC) to inform system master BMC about the setting of each service-to-channel mapping and associated Level 1 scheduling parameters.

Primitive Type: request.

Parameters:

Services notification.

CBMC-Notifupdate-IND

The CBMC-Notifupdate-IND primitive is used by BMC to inform the UE RRC about the forthcoming change of the notification information for the relevant service providing.

Primitive Type: indication.

Parameters:

Service-to-channel Mapping.

Level 1 scheduling.

Time of change.

CBMC-Notifupdate-REQ

The CBMC-Notifupdate-REQ primitive is used by the RRC to inform BMC about the forthcoming change of the notification information for the relevant service providing.

Primitive Type: request.

Parameters:

Service-to-channel Mapping.

Level 1 scheduling.

Time of change.

Parameters

CB-Traffic-Volume

Expected CTCH transmission rate [kbps].

Value set: 0, 1, ... , 312.

Action

Start CBS/MBMS reception.

Stop CBS/MBMS reception.

DRX selection

List of absolute CTCH BS indices which are of interest and which should be received by Layer 1.

CTCH configuration

Current CTCH-BS index, $1 \leq i \leq 256$.

Timing of CTCH-BS sequence, i.e. CTCH occasion parameters of the non-static Level 1 scheduling: period of allocation N, frame offset with respect to SFN cycle.

NOTE 1: Background services featuring medium/high data rate (more than 64 kbps) and Real Time services should be provided with continuous occasions (N = number of frames in the TTI).

FACH identification.

Transport Format Set of the allocated FACH (TB size, TBS size, TTI).

Allocated CTCH transmission rate [kbps]: 0 ,1 , ... , 312.

Services notification

Complete set of service-to-channel mapping and associated CTCH occasion parameters (Level 1 scheduling: period of allocation N, frame offset with respect to SFN cycle), for all MBMS services broadcast in the cell/spotbeam.

Service-to-channel Mapping

Next service-to-channel mapping of relevance to service support.

Level 1 scheduling

Next Level 1 scheduling parameters (period of allocation N, frame offset with respect to SFN cycle) of relevance to service support.

Time of change

Time, indicated in terms of System Frame Number with respect to the SFN cycle) at which new service-to-channel mapping and/or Level 1 scheduling parameters of interest should be effective.

Service Primitives between upper layer (U-plane) and BMC

The primitives supported at BMC-SAP between BMC and upper layer (U-plane) are briefly described in Table 21. This set of primitives corresponds to the one specified in [22] in relation to UMTS Core Network.

Table 21: Primitives between BMC and upper layer

Generic Name	Use	Parameters
BMC-Data-REQ	Used by upper layer to request repeated transmission of CB messages	Message-ID, [,Old-Serial-Number], New-Serial-Number, Data-Coding-Scheme, CB-Data, [Category], Repetition-Period, Number-of-Broadcasts-Requested
BMC-Data-IND	Used to indicate received CB messages (i.e. CB Data) to upper layer	Message-ID, Serial-Number, Data-Coding-Scheme, CB-Data
BMC-Data-CNF	Used to indicate the complete broadcast of CB messages	Message-ID, Serial-Number
BMC-Congestion-IND	Used to indicate to upper layer (BM-IWF) that the BMC entity is congested	
BMC-Normal-IND	Used to indicate to upper layer (BM-IWF) that the BMC has recovered from a congestion situation and is operating normal	
BMC-Activation-REQ	Used to request CB message reception and to notify which CB messages are of interest and shall be delivered to the upper layer	Message-ID (n times)
BMC-Deactivation-REQ	Used to request stop of reception of listed CB messages. If no more CB messages are to be received, CB message reception shall stop	Message-ID (n times)
BMC-DRX-REQ	Used to command CBS discontinuous reception (CB DRX) if it were required by O&M system.	CB-DRX-Schedule-Period, Reserved-CB-Capacity
BMC-Error-IND	Used to indicate unsuccessful operations of the BMC entity requested	Cause
NOTE: [ ] Optional parameters.		

NOTE 2: MBMS packets or bursts can be formatted as CB messages (BMC SDUs not restricted to the sole CBS messages).

NOTE 3: Iu interface is subject to evolution within the MBMS context (i.e. necessary evolution of Iu PS Rel.5, still to be further defined) that should be adopted as part Rel.6. Nevertheless SATIN assumes the use of future multicast RAB (managed by upgraded RANAP) allowing service/traffic differentiation (QoS class) and its compatibility with the operation of point-to-multipoint radio bearer provided by BMC/RLC (whether traffic is scheduled or not in the CN, i.e. traffic volume and capacity reservation not necessarily required).

F.1.3 SATIN RLC sublayer specifications

The SATIN RLC sublayer features a subset of the full RLC functionality described in [19]. Two types of entities are envisaged: Transparent Mode (TM) and Unacknowledged Mode (UM) entities.

Figure 107 illustrates the respective entities in the SATIN RLC model.

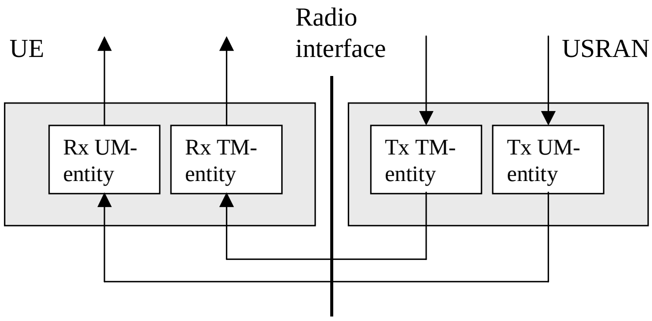


Figure 107: Overview model of the SATIN RLC sublayer

In general, the TM and UM entities can be configured to act as transmitting or receiving entities, in which case the entities transmit/receive RLC PDUs to/from lower layers respectively. In the frame of the elementary RLC procedures description the transmitting entity is called Sender and the receiving entity is called Receiver.

Within SATIN transmitting entities only reside at the USRAN side, while at the UE side, the TM-and UM- RLC entities act as Receivers. It should be noted that Figure 107 depicts the RLC entities at the UE that are related to the SATIN interface. It is expected that a SATIN terminal in the baseline scenario will be a fully T-UMTS compatible terminal, i.e. it will have the full RLC functionality for interfacing with T-UMTS and accessing services not provided by SATIN.

There is one transmitting and one receiving RLC entity for each transparent mode and unacknowledged mode service. Each RLC and TM entity exchanges data PDUs via use of logical channels.

The RLC sublayer is adapted according to the above simplification of the W-CDMA RLC sublayer as specified in 3GPP.

F.1.4 SATIN MAC sublayer specification

The SATIN MAC sublayer draws heavily from the respective 3GPP MAC architecture. In fact it is a combination of simplifications and slight modifications of the MAC protocol described in TS 125 321 [20] "Medium Access Control (MAC) protocol specification".

F.1.4.1 MAC layer in the SATIN baseline scenario

Architecture

The RRC is responsible for the configuration of the MAC layer via the respective service access points and the interlayer procedures.

MAC entities

The MAC entities retained in SATIN are a subset of the full set of entities defined in [20] and their functionality is different depending on whether they are at the user (UE) or the network (USRAN) side.

MAC-b entity

This is the entity responsible for the Broadcast CHannel (BCH). There is one MAC-b entity in each UE and one in the USRAN side for each satellite spot beam.

MAC-c entity

This is a subset of the full MAC-c/sh entity defined in T-UMTS and within SATIN controls access to the FACH transport channel.

Note that at the user side the full MAC functionality is expected. The terminal is making use of the terrestrial link for non-SATIN services. However, the functions related to services provided by the S-UMTS network are:

Demultiplexing on the basis of TCTF field

The entity detects logical channels multiplexed on a single FACH channel and demultiplexes them on the basis on the TCTF field of the MAC header.

There is one MAC-c for each UE.

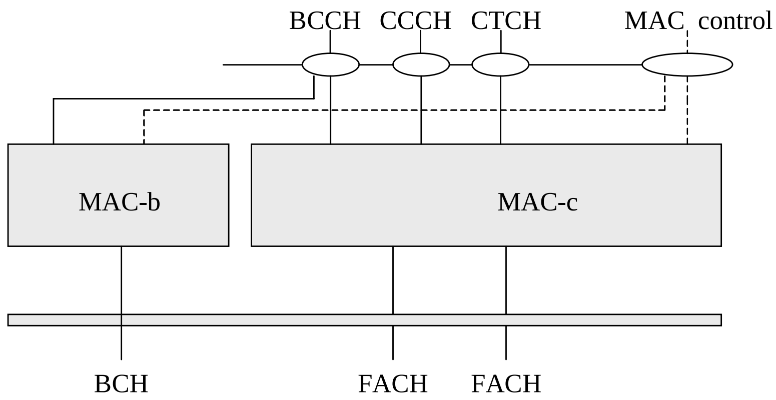


Figure 108: SATIN UE side architecture (satellite-specific part)

At the USRAN side the functions fulfilled by MAC-c are the following:

Scheduling -Priority handling

Different treatment is provided to flows of different classes of service. The differentiation is not in terms of individual users but in terms of multicast/broadcast groups. There is one-to-one correspondence between CTCH and B/M service (upper limit envisaged for the whole USRAN is 32).

Multiplexing on the basis of TCTF field

The entity inserts the appropriate TCTF field at the MAC header allowing the multiplexing of logical channels on the same transport channel.

TFC selection

The appropriate transport format set (combination) is selected each TTI for the FACH channel(s), potentially multiplexed on a single physical channel (S-CCPCH).

There is one MAC-c entity per satellite beam.

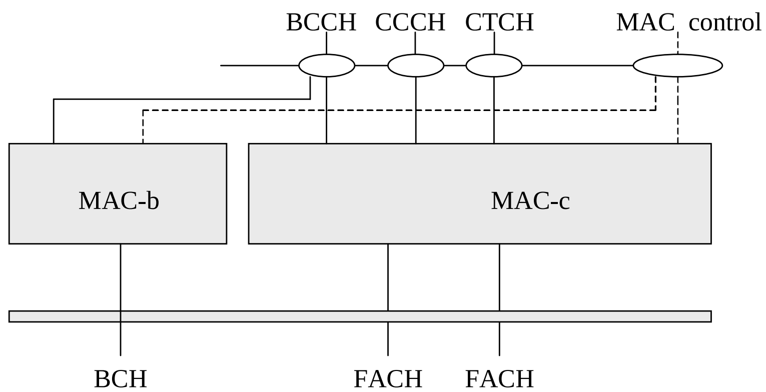


Figure 109: SATIN USRAN side MAC architecture

In comparison with the standard terrestrial network, there is no need for the MAC-d and MAC-sh entities at the USRAN side, since no dedicated channels are envisaged within the SATIN baseline scenario. The terminal however, as mentioned earlier, will feature full MAC functionality as described in [20] (depending on its class) that will allow it to receive the point-to-point services via the terrestrial network.

Channel structure

Transport channels

The transport channels (between MAC and physical layer) envisaged within SATIN are:

- Forward Access CHannel (FACH);
- Broadcast CHannel (BCH);
- Paging CHannel (PCH).

Logical channels

The MAC data transfer services are provided on logical channels. Depending on the information provided by logical channels they are separated into control and traffic channels. Within SATIN control-plane information is provided by:

- the Broadcast Control CHannel (BCCH); and
- the Paging Control CHannel (PCCH),

while the only traffic channel envisaged for the transport of multicast/broadcast services (user plane) is the Common Traffic CHannel (CTCH) channel.

Functions

The functions that the MAC layer performs within the SATIN context are:

- Mapping of logical channels (CTCH, PCCH, BCH) to transport channels (FACH, BCH, PCH).
- Selection of the Transport Format at TTI time level for each transport channel.
- (De)multiplexing of upper layer PDUs from/into transport block sets passed from/to the physical layer on the BCH, PCH and FACH channel(s).
- Measurements of traffic volume.
- Service differentiation to different services (oriented to all or groups of users) by means if dynamic scheduling and priority handling of the different flows (logical channels).
- Ciphering when the RLC transparent mode is adopted (e.g. for BCCH and PCCH channel).
- Identification of groups of users, when multiplexed on common transport channels (in the baseline scenario this MAC function is not relevant if 1 CTCH per FACH is the envisaged mapping).

Relation between MAC functions and transport channels

USRAN side

The functions applicable to each one of the possible mapping combinations between logical and transport channels at the USRAN side are given in Table 21a.

Table 21a

Associated MAC Functions	Logical Ch	Transport Ch	TF Selection	Priority handling between services/groups of users	Scheduling	Mux/ Demux on common transport channels
Downlink (Tx)	BCCH	BCH			X	
	BCCH	FACH	X		X	X
	PCCH	PCH	X	-	X	-
	CTCH	FACH	X	X	X	X

UE side

At the UE side, the supported functions are limited due to the lack of satellite uplink functionality; in fact, the MAC performs only demultiplexing functions. Basic error checking functions (like identification of invalid transport formats and subsequent discarding of a MAC SDU) are part of its tasks as well.

Table 21b

Associated MAC Functions	Logical Ch	Transport Ch	Mux/Demux on common transport channels
Downlink (Rx)	BCCH	BCH	
	BCCH	FACH	X
	PCCH	FACH	-
	CTCH	FACH	X

Primitives and PDUs are the same as in TS 125 321 [20], but only a subset of them is relevant because of the simplified MAC layer.

F.1.4.2 MAC layer in the SATIN optional scenario

For the SATIN optional case, everything as in the SATIN baseline described above is retained, but with added functions so as to enable the use of a direct return link over the satellite through the RACH. Described herein are the additions necessary so as to accommodate the use of the RACH.

Architecture

MAC Entities

The modified model for the MAC entity is given in Figure 110.

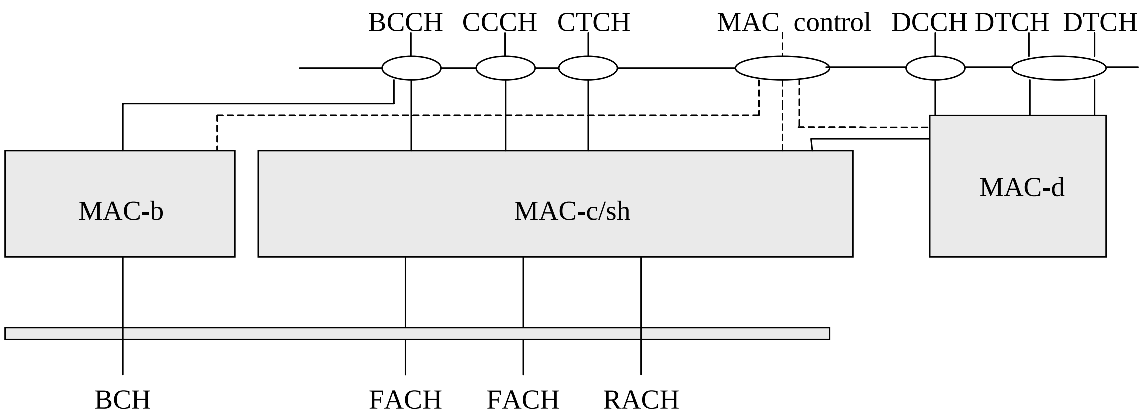


Figure 110: UE side MAC architecture for SATIN optional scenario (similar for USRAN side)

An uplink channel is added to the supported transport channels in order to support point-to-point services of limited interactivity. The use of logical channels of dedicated type introduces the need for the MAC-d entity and additional functions related to the MAC-sh entity.

MAC-d entity

When logical channels of dedicated type (DTCH) are mapped to RACH, MAC-d submits the data to MAC-c/sh via the illustrated connection between the functional entities. The additional functionality of MAC-d for supporting the multiplexing of logical channels of dedicated type over common transport channels is given below:

Transport Channel type switching

Transport Channel type switching is performed after an RRC command; if requested by RRC, MAC shall switch the mapping of one designated logical channel between common and dedicated transport channels.

C/T MUX

The C/T MUX is used when multiplexing several dedicated logical channels onto one transport channel is used. An unambiguous identification of the logical channel is included.

There is one MAC-d entity in the UE and one MAC-d entity in the USRAN for each UE that has one or more dedicated logical channels to or from the USRAN.

MAC-c/sh entity

The additional functionalities needed for the optional case are:

Add/read UE Id:

The UE Id is added for RACH transmissions to identify data to this UE.

(De)multiplexing on the basis of TCTF field at both directions

This function is now necessary for uplink channels (i.e. RACH) as well. In the uplink direction, it implies the mapping/multiplexing of uplink logical channels on the RACH channel at the UE side and the respective demultiplexing at the USRAN side.

ASC selection

Class is either determined by the RRC (upon an RRC CONNECTION REQUEST message) or selected by MAC. It assures that the messages sent on RACH (optional scenario) are sent with the respective {signature(s), slot(s), backoff} tuple, in effect conforming to the priority level related to the ASC.

Channel structure

Transport channels

The additional transport channel (between MAC and physical layer) envisaged within SATIN optional case is the:

- Random Access CHannel (RACH).

Logical channels

For the optional case, the additional control channels used for the transfer of control plane information are:

- the Dedicated Control CHannel (DCCH); and
- the Common Control CHannel (CCCH) - at the uplink direction.

while the additional traffic channel envisaged in both directions is the Dedicated Traffic CHannel (DTCH)

Functions

Additional function of the MAC functions with regard to the baseline case is the Access Service Class selection for RACH.

Relation between MAC functions and transport channels

The functions applicable to each one of the possible mapping combinations between logical and transport channels at the USRAN and at the UE side for the optional case are given in Tables 21c and 21d.

Table 21c: USRAN side

Associated MAC Functions	Logical Ch	Transport Ch	TF Selection	Priority handling between UEs	Priority handling (one UE)	Scheduling	Identification of UEs	Mux/ Demux on common transport channels
Uplink (Rx)	CCCH	RACH						X
	DCCH	RACH					X	X
	DTCH	RACH					X	X
Downlink (Tx)	BCCH	BCH				X		
	CCCH	FACH	X	X		X		X
	DCCH	FACH	X	X		X	X	X
	DTCH	FACH	X	X		X	X	X

Table 21d: UE side

Associated MAC Functions	Logical Ch	Transport Ch	TF Selection	Priority handling (one UE)	Identification	Mux/Demux on common transport channels
Uplink (Tx)	CCCH	RACH				X
	DCCH	RACH	X	X	X	X
	DTCH	RACH	X	X	X	X
Downlink (Rx)	BCCH	BCH				
	CCCH	FACH				X
	DCCH	FACH			X	X
	DTCH	FACH			X	X

F.2 Layer 1 specifications

F.2.1 Transport channels

In the following the transport and physical channel description and their association are considered for both the T-UMTS and S-UMTS air interfaces (see [29] and [31]) with reference to the delivery of MBMS in the SATIN scenario. In particular, the following satellite environment peculiarities shall be considered:

- Diversity operation, as used in UTRA, is not easily extendable to the satellite environment.
- Due to the increased round-trip-time, power control operations cannot be performed on a slot-by-slot basis but only on a frame-by-frame basis, and moreover they are not directly applicable to MBMS.
- The larger delays and delay variation among users lead to different timing offsets between channels, and make synchronization between them more difficult. As their physical operation is based on these timing relations, their exact feasibility in a satellite environment should be assessed.
- Implementation of new physical channels in order to support broadcast/multicast services, not provided in the current set of standard specifications, has to be taken into account.
- Framing details could change, accordingly.

F.2.1.1 Forward Link/Downlink

Dedicated channels

It is possible to use DCH in the Optional scenario, but providing packet data MBMS with DCH is highly inefficient due to the large overhead required to set up the RRC connection. In addition the DCH concept inherently contradicts the point-to-multipoint nature of MBMS.

Common channels

The common downlink transport channels are the same in T-UMTS and S-UMTS: Broadcast CHannel (BCH), Forward Access CHannel (FACH), Paging CHannel (PCH), and Downlink Shared CHannel (DSCH).

Broadcast channel

Since a maximum of 10 bits per slot is available for user information, the channel bit-rate is limited to 15 kbps. Hence, taking into account the channel coding rate, BCH is not suited to support broadcast-multicast services with bit rate higher than 5 - 7,5 kbps.

Forward Access and Paging Channels

From a higher layer point of view, an adaptation to be made concerns the user data identification/indication towards the targeted UEs. Currently, this is done by a UE-specific identifier in the MAC header, e.g. when DTCH is mapped onto FACH. When using FACH for MBMS, an identification of the receiver group must be introduced. This could be done e.g. by introducing an MBMS group ID in the MAC header.

FACH is applicable to the baseline scenario.

FACH is applicable to the optional scenario.

Hence, FACH is applicable to the SATIN environment and may be employed to support broadcast-multicast service with high data rate.

DSCH - Downlink Shared Channel

The main adaptation to be made regarding DSCH usage in S-UMTS is the possible decoupling of the DSCH from the associated DCH to avoid overhead (both in power and in code usage) in maintaining such a DCH. The satellite environment deals with larger cells (i.e. more users) and larger distances from satellite to mobile. This in combination with the nature of packet data traffic, including highly discontinuous data (e.g. for web browsing applications), proves DCH overhead a critical factor in DSCH operation. Obviously, when targeting MBMS, relying (solely) on dedicated channels for signalling is highly inefficient and further strengthens the need for an alternative to the DCH approach.

Possible improvements

In case of DCH shortage (or just by implementing a disconnection timer function), a straightforward solution could be to revoke the DCH of the MS with the longest idle time, meaning this terminal will go from CELL_DCH state to CELL_FACH state, freeing resources for UEs in need. This approach is an improvement on the code resources, but not in terms of power resource, i.e. interference generation.

Minimization of this power waste can be obtained if, on one side, the DSCH capacity assignment can be communicated either in-band (discarded by 3GPP) or via a common control channel (discarded by 3GPP, but still under discussion for use in HSPDA as a complement to DCH, i.e. the Physical Shared Common Control CHannel (PSCCCH) and power control is either not used or is based on measurements not requiring a DCH.

A possible solution, still providing closed loop power control support, is to employ both a Shared Control CHannel (SCCH) carrying the actual control information for the DSCH, and a new type of channel to support power control and to carry signalling indication for multiple terminals towards the SCCH, referred to as the Signalling Indication CHannel (SICH). Each user having an active packet session will be assigned resources on a SICH channel), i.e. is allocated a slot in the SICH. Multiple SICH channels will be needed but the number of total SICHs will be anyway less than that of the associated DCHs, which would otherwise be required. These SICH channels in their turn would point the UEs to look at the SCCH, and provide them the necessary transport format information.

When only open loop power control (or even no power control) is envisaged, which seems highly likely for MBMS, a DCH or a SICH is not strictly required. The power control support functionality of the SICH can be dropped, so that the SICH can be used exclusively for signalling indication towards the SCCH. Alternatively, the whole SICH can be dropped if a TFCI field is provided in the SCCH.

Applicability

For satellite MBMS the DSCH seems useful if, instead of dynamically allocating its resources to a *single user*, a *user group* (multicast group) can be addressed. Hence, some adaptation in the higher layer signalling for the resource allocation will be required.

Of course also in this case it is mandatory to avoid doing the signalling and control through dedicated channels, because of the reasons mentioned above in the ATB approach.

Basically, two options, based on the ATB approach, can be distinguished:

- DSCH + SCCH + SICH:
  - Advantages:
    - slightly greater capacity for SCCH (no TFCI field, strictly data);
    - beneficial for UE power consumption.
  - Disadvantages:
    - extra channel.
- DSCH + SCCH:
  - Advantages:
    - no extra channel needed.
  - Disadvantages:
    - higher power consumption;
    - no capacity increase.

In the Optional scenario, using a DCH to carry the control information is not impossible, but it is inefficient for the reasons mentioned above in the ATB approach. If no power control is considered, both options are candidates in exactly the same way as for the Baseline scenario. Open loop power control can be implemented in the optional scenario based on pilot SNIR measurements at the satellite or based on ACK messages. If some sort of closed loop power control must be implemented, only option 1 remains.

Forward link traffic channels comparison (FACH vs. DSCH)

Two transport channels were found to be suitable for the SATIN downlink traffic, i.e. delivering packet data, in a satellite environment, to user groups (satellite MBMS).

The FACH/S-CCPCH can almost directly be adopted from T-UMTS, a slight modification to be made, however not on a physical layer level, would be to include a user group ID in the MAC header instead of a single user ID.

The DSCH/PDSCH can also remain similar to the T-UMTS DSCH/PDSCH, but it cannot be operated in the same way, due to the need for both downlink and uplink dedicated channels to accompany the DSCH for signalling and control. An alternative approach would be to use a Shared Control CHannel (SCCH), possibly in conjunction with a Signalling Indication CHannel (SICH). This approach would hence impose modifications wrt the terrestrial UMTS standard. Fast power control, which is an essential feature of the DSCH in T-UMTS, is not feasible in the SATIN environment.

The biggest conceptual difference between both transport channels is the resource allocation approach. DSCH operation employs a separate signalling channel, while the FACH contains an ID in the MAC header.

In terms of available downlink data rate, both options are estimated to have similar capacity.

HPPICH -High Penetration Indicator channel

The concept of a high penetration paging service was conceived for a scenario where the downlink is directly from the satellite to the terminal, without the presence of IMRs. SATIN targets a full coverage of the mass market in terms of service complement. This means that IMRs will be mandatory wherever there is T-UMTS coverage. Through these IMRs the reception of the normal paging service should not pose any problems. Hence, the relevance of a high penetration paging service is in doubt for the SATIN case.

F.2.1.2 Return Link/Uplink

Dedicated channels

DCH can be employed to send large single data packets.

In the presence of multiple packet transmission the DCH can be inactive for a certain time interval, waiting for new packets.

Since an uplink DPCH operates jointly with a downlink DPCH for control purposes, and because of the large cells and therefore the large number of users to be addressed, the code resources would be insufficient to support the use of a DPCH.

Common channels

RACH - Random Access Channel

The adoption of the 3GPP slotted ALOHA random access scheme with 15 access slots per two radio frames for the S-UMTS is not feasible, since round-trip delay is in the order of several hundreds of milliseconds and delay variation is estimated to be 20 ms. One-shot acquisition, proposed in ETSI S-UMTS, is envisaged as an alternative.

CPCH - Common Packet Channel

The same impairments as with the RACH concerning power ramp-up procedure using multiple preambles.

Layer 1 collision detection is not efficient because large round-trip delay.

Fast power control is impossible because of large round-trip delay.

For the Optional scenario, this channel is not suitable.

F.2.2 Mapping of transport channels onto physical channels

Figure 111 summarizes the mapping of transport channels onto physical channels.

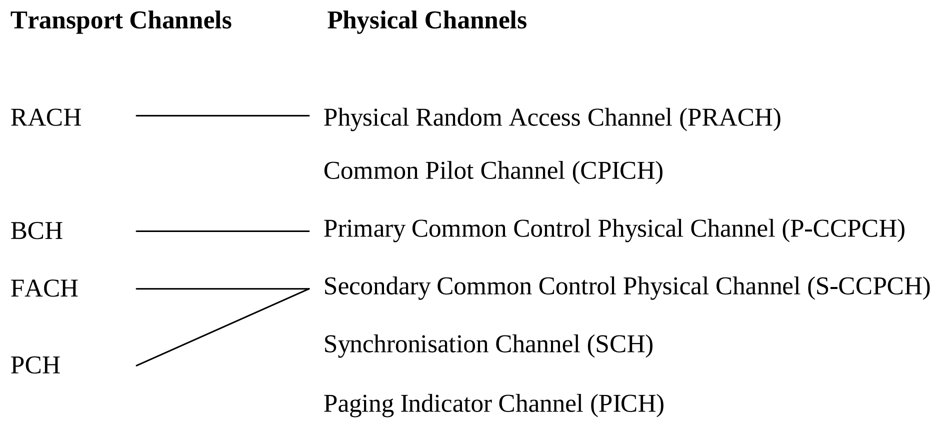


Figure 111: Transport-channel to physical-channel mapping

F.2.3 Timing relationship

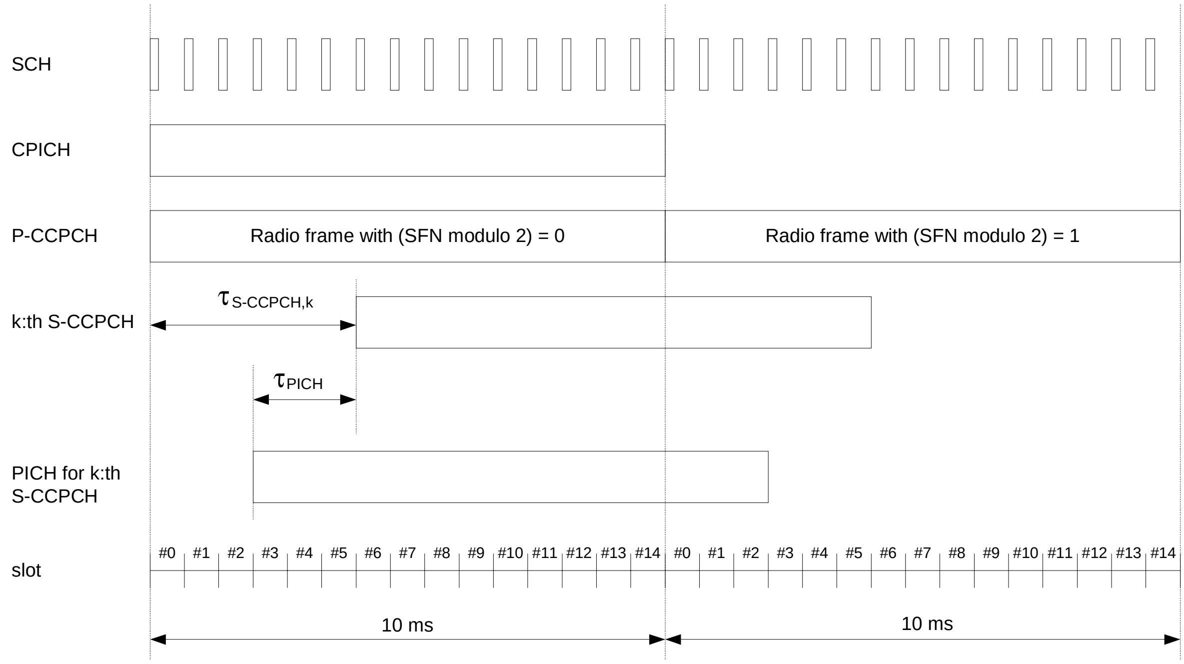


Figure 112: Frame timing of downlink physical channels

In Figure 112, the following applies:

- SCH, CPICH and P-CCPCH have identical frame timings.
- The S-CCPCH timing may be different for different S-CCPCHs, but the offset from the P-CCPCH frame timing is a multiple of 256 chips, i.e. $\tau_{S\text{-}CCPCH,k} = T_k \times 256$ chip, $T_k \in \{0, 1, \ldots, 149\}$.
- The PICH timing is τ_{PICH} = 7 680 chips prior to its corresponding S-CCPCH frame timing.

F.2.4 Higher order modulation schemes

The mobile satellite communication channel suffers from a lot of impairments such as multipath fading, Doppler shift and phase variations. Moreover, non-linearity cannot be neglected in a satellite environment since on-board amplifiers are usually driven into saturation to optimize the DC/RF conversion efficiency. To face this problem, a different kind of modulation schemes, more robust towards non-linear effects, could be taken into account. For example, an Offset-QPSK (O-QPSK) has the advantage that after a band limited filtering the transmit signal exhibits less amplitude variations than a QPSK signal. Anyway, the most effective impairment caused by non-linear device is mainly due to the presence of the multiplexed transmit signal that is characterized by high amplitude peak value forcing the amplifier to work on the saturation region.

High order modulation such as QAM (Quadrature Amplitude Modulation) will be investigated. The rationale being the analysis and design of bandwidth efficient coding and modulation schemes (for example, for turbo coding rate 1/2 and 16 - QAM, 2 bits per symbol are transmitted) over a mobile satellite channel. To this aim, a *pragmatic approach*, can be adopted: 16 - QAM can be used without redesigning the encoder and decoder; the input turbo decoder is made up of soft decisions associated to each turbo encoder output bit. The concept of the pragmatic turbo coded modulation is the use of a simple pair of turbo encoder/decoder so that various TCM schemes are obtained, by modifying a puncturing function after the encoding and the modulation signal constellation.

F.2.4.1 8-PSK

The 8-PSK constellation is a constant envelope one, as all PSK constellations. For that reason it is a natural candidate for satellite systems. Moreover 8-PSK is an option for SATIN if Layered Coding is employed. Its spectral efficiency is 3 bit/s/Hz.

The signal can be represented as:

$$
\begin{aligned}
s_m(t) &= \mathrm{Re}\left[g(t) e^{j2\pi(m-1)/8} e^{j2 f_c t} \right] \qquad \mathrm{m} = 1,2,\ldots 8, \quad 0 \le \mathrm{t} \le \mathrm{T} \\
&= \mathrm{g}(\mathrm{t}) \cos\left[2\pi \cdot f_c t + \frac{2\pi}{8}(m-1) \right] \\
&= \mathrm{g}(\mathrm{t}) \cos\frac{2\pi}{8}(m-1) \cos 2\pi \cdot f_c t - g(t) \sin\frac{2\pi}{8}(m-1) \sin 2\pi \cdot f_c t
\end{aligned}
$$

In the above equation g(t) represents the signal pulse shape and f_c is the carrier frequency.

In order to represent the 8-PSK signal as a two dimensional constellation we will express it as a combination of two orthogonal signals. It can be proved that an 8-PSK signal can be represented as:

$$
s_m(t) = s_{m1} f_1(t) + s_{m2} f_2(t)
$$

where

$$
f_1(t) = \sqrt{\frac{2}{E_g}} g(t) \cos 2\pi \cdot f_c t
$$

$$
f_2(t) = -\sqrt{\frac{2}{E_g}} g(t) \sin 2\pi \cdot f_c t
$$

and

$$
s_m = \left[\sqrt{\frac{E_g}{2}} \cos\frac{2\pi}{8}(m-1), \sqrt{\frac{E_g}{2}} \sin\frac{2\pi}{8}(m-1) \right] \quad \mathrm{m} = 1,2,\ldots 8
$$

E_g denotes the energy in the pulse g(t).

The constellation diagram of Gray-coded 8-PSK signal is shown in Figure 113.

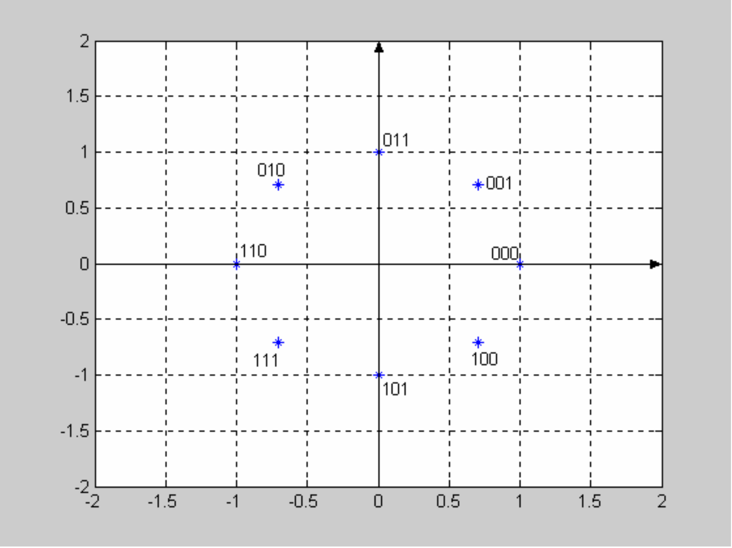


Figure 113: 8-PSK constellation

F.2.4.2 16-QAM

Quadrature Amplitude Modulation (QAM) is a modulation technique, which combines Pulse Amplitude Modulation (PAM) and Phase Shift Keying (PSK), in the sense that the carrier is now modulated both in amplitude and in phase. The most widely used approach is considering the QAM signal as a couple of orthogonal carriers, $\cos 2\pi f_c t$ and $\sin 2\pi f_c t$, each of which is modulated by an independent sequence of information bits. f_c denotes the carrier frequency. The transmitted signal waveforms, $u_m(t)$, have the form:

$$u_m(t) = A_{mc} g_T(t) \cos 2\pi f_c t + A_{ms} g_T(t) \sin 2\pi f_c t, \qquad m = 1,2,\ldots,M$$

where A_{mc}, A_{ms} are the sets of amplitude levels that are obtained by mapping k - bit sequences into signal amplitudes and g_T is the pulse with duration of T. In general, rectangular signal constellations result when two quadrature carriers are each modulated by PAM.

The geometric signal representation of the signals is in terms of two dimensional signal vectors of the form:

$$s_M = \left(\sqrt{E_s} A_{mc}, \sqrt{E_s} A_{ms}\right), \qquad m = 1,2,\ldots,M$$

E_s denotes the symbol energy. This approach also reflects the most efficient way to demodulate a QAM signal at the receiver, which consists splitting the received signal into two orthogonal signals and demodulating them as independent PAM waveforms. The two PAM waveforms are called inphase (I) and quadrature (Q) components.

16-QAM has a spectral efficiency of 4 bit/s/Hz.

F.2.5 Advanced Coding schemes

F.2.5.1 Impact of the Broadcast/Multicast services

The Broadcast /Multicast services provided by Satin might need a higher modulation scheme such as 8-PSK which is also an option for Layered Coding. In multilevel coding the encoder transmits at the same information rate and the decoder decides its level of decoding independently, as a function of its own channel conditions. According to that, an option for multicast and broadcast services is that the transmitter uses 8-PSK. Then each user decodes 3bits (8-PSK) or 2 bits (Q-PSK) taking different quality of service.

Satellite broadcast applications involve the transmission of information encoded with the same channel code over channels that may significantly vary, depending on the end-user location. The communication channels may follow a Gaussian, Ricean or Rayleigh distribution and with respect to temporal spread they are typically flat fading and may suffer shadowing. Even for point-to-point applications (not our case here), mobility implies that the same user may come across different channels during a session, for example when a vehicle goes under a tunnel.

S-UMTS supports 3G services that require a specific quality of service, i.e. BER upper-limits or a minimum received E_b/N_o for a considered channel coding in any channel fading condition. Moreover, in a GEO-based broadcast environment, feedback or retransmission of packets is not generally possible and may involve excessive delay. The communication system has to typically rely solely on Forward Error Correction (FEC).

Coding may be designed to address the worst-case fading scenario, which leads to unnecessary receiver processing complexity for the majority of users. Alternatively, coding may address an average fading scenario, which cannot provide a hard guarantee for the quality of service. Ideally, coding should allow a user with a good channel to recover the information with low complexity, while a user with a bad channel should still be able to achieve an acceptable BER at the cost of increased complexity or of some extra decoding delay and power consumption. However, in the case of broadcasting/multicasting systems, the transmitter can only employ a single channel code which is simultaneously received by all users; thus, the same channel code has to be able to provide the capability of soft trade-off in the processing complexity-achievable BER sense as a function of receiver decoding.

Moreover, even if the degradation due to severe fading cannot be completely avoided, it is desirable that the system performance follows a BER vs. SNR curve with smaller (smoother) slope than most codes naturally provide, or in other words, a softer performance degradation. For example, for turbo codes the BER curve is characterized by a very abrupt "waterfall" area, where the BER drops orders of magnitude within tenths of dB. It is undesirable for a user to observe such a huge performance difference (and a quality of service change) over so slight channel variation.

The layered coding schemes (softly degrading schemes) perform decoding up to a level depending on the channel Signal to Noise Ratio (SNR) and may be used to achieve rate adaptiveness (where we take advantage of good SNR conditions to increase the information rate) or protection adaptive ness (where we transmit extra coded bits that are employed in order to fight fading at the cost of an extra complexity and/or delay).

With these considerations in mind, it is necessary to consider for SATIN an appropriate coding scheme that includes several layers, each representing an additional error correcting stage. The receiver should be able to use only the necessary layers that lead to the required BER, which means that the receiver should fit to the channel conditions and modify the error correcting capability (and the corresponding complexity and delay) according to the channel variation. The feature that makes layered coding schemes suitable for broadcast transmission is that the encoder transmits at the same information rate, while each user decoder decides its level of decoding independently, as a function of its own channel conditions.

The layered coding scheme under consideration includes a convolution and a turbo code. The use of both codes yields a coding gain of around 4 dB (for BER = 4×10^{-4}) in AWGN compared to the use of only the convolution code and this may be even larger in the case of a Raleigh channel. This coding gain is comparable to the standard deviation (around 6 dB) of the log normally-distributed received E_b/N_o, due to the shadowing, in the absence of power control. Therefore, by using layered coding, the majority of users will manage to receive the broadcast signal with E_b/N_o very close to the required value and the standard deviation of their corresponding distribution will be much lower. In other words, the implementation of layered coding in SATIN will have similar effect on the system performance to that of power control, which is not easily implemented in a GEO system that involves large propagation delays.

The coding structure employed with layered coding can basically be thought of as a serially concatenated code, with one restriction: that the inner convolution code is systematic, or in other words, the outer convolutional code can be decoded separately. Serially concatenated codes can be used with any kind of modulation. Thus layered coding as well can be used with any kind of modulation.

F.2.5.2 Combining layered coding with high spectral efficiency

Figure 114 demonstrates the general structure of a layered code that has spectral efficiency k bits/sec/Hz, and employs a constellation of size $2^{k+n1+n2}$. The restriction imposed by the layered structure that the inner encoder is systematic implies that $n_1 \geq 1, \quad n_2 \geq 1$. That is, we cannot have an inner encoder of rate one.

With this structure, for example, we can achieve spectral efficiency of 2 bits/sec/Hz by using 16 QAM modulation as following: the outer encoder has k = 2 inputs and k + n_1 = 3 outputs, while the inner encoder is systematic with k + n_1 + n_2 = 4 outputs that are mapped to a 16 QAM symbol.

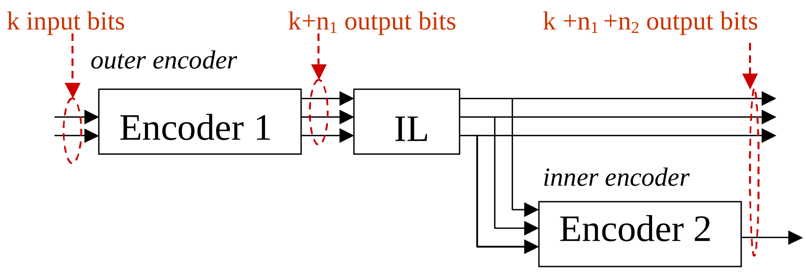


Figure 114: General layered coding structure

Two basic encoding structures to be examined are structure A (Figure 115), based on mapping onto 8PSK symbols, and structure B, based on mapping onto QPSK symbols, and are discussed in the following clauses.

Structure A -Mapping onto 8PSK symbols

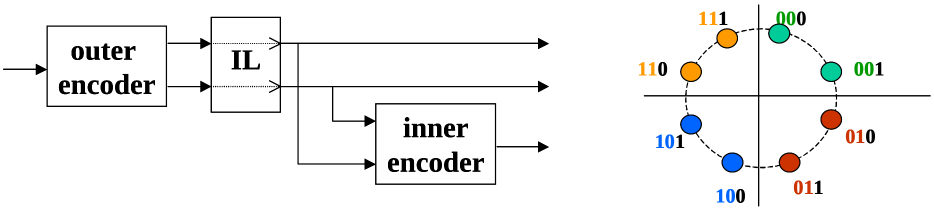


Figure 115: Layered Coding Structure employing 8PSK - Structure A

This structure is based on mapping the coded bits either onto an 8PSK symbol (Outer and Inner code - low BER option) or onto a QPSK symbol (Outer Code - high BER option).

Generally a non-uniform 8-PSK constellation can be described by a parameter θ, $0 \le \theta \le \frac{\pi}{8}$, as is depicted in the following figure. For the limit value $\theta = 0$ the constellation becomes identical to uniform 4PSK while for $\theta = \frac{\pi}{8}$ the constellation coincides with uniform 8PSK.

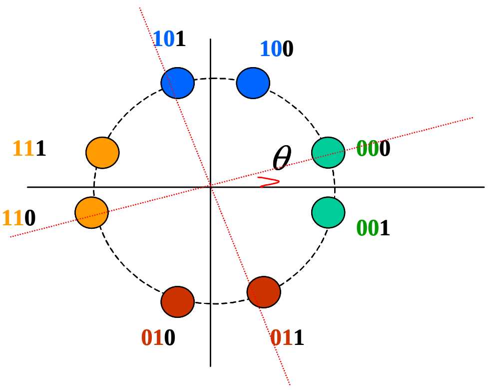


Figure 116: Non-uniform 8-PSK constellation

Table 22 lists possible generator polynomials for the encoder structure according to the respective angle of constellation.

Table 22: Encoder generator polynomials for encoder structure A

Memory elements	Type of encoder	Generator polynomials
8	Outer, feedforward, systematic	{0435, 0657}
3	Inner, feedback, $\theta = \pi / 8$	{013, 01, 07, 07, 00}
3	Inner, feedback, $\theta = 3\pi / 32$ *(Instead of 3π/8)*	{015, 02, 07, 02, 03}
3	Inner, feedback, $\theta = \pi / 16$	{015, 01, 03, 06, 03}

Structure B -Mapping onto QPSK symbols

Structure B (Figure 117) is based on mapping the coded bits either onto two consecutive QPSK symbols with different error protection (Outer and Inner code, low BER option) or onto one QPSK symbol (Outer code, high BER option).

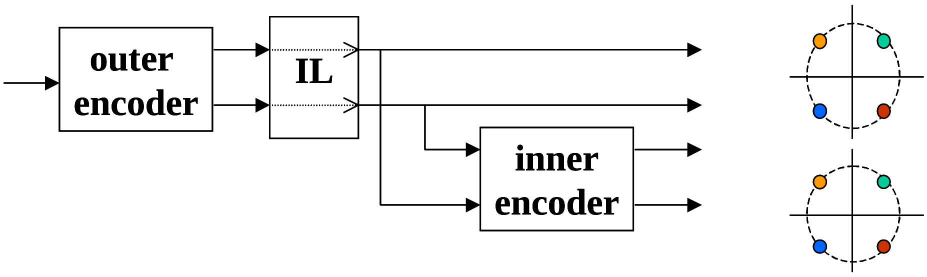


Figure 117: Layered coding structure employing QPSK - structure B

The interleaver employed is an extended spread semi-random interleaver that works at symbol level. The interleaver size shall be chosen according to the TB size.

The following table lists possible generator polynomials for the encoder structure.

Table 23: Possible generator polynomials for encoder structure B

Memory elements	Type of encoder	Generator polynomials
8	Outer, feedforward, systematic	{0435, 0657}
3	Inner, feedback AWGN	{013, 01, 03, 02, 06, 01, 01}
3	Inner, feedback Fading Channels	{011, 02, 05, 03, 06, 02, 01}

F.2.6 Physical layer procedures and operation

F.2.6.1 Cell-search procedure

During the cell search, the UE searches for a cell and determines the downlink scrambling code and common channel frame synchronization of that cell.

According to [36], in W-CDMA air interface, three downlink physical channels are provided for the initial cell acquisition: the P-SCH, the S-SCH, and the P-CPICH. The channel structures are reported in Figure 118 and Figure 119. P-SCH is provided for slot synchronization. It consists of a bursty repetition, at the beginning of each slot, of the same 256-chip pseudonoise sequence, the PSC. Once PSC synchronization is acquired, the slot boundaries are identified as well. The P-SCH structure is reported in Figure 118, PSC is denoted as ac_p. PSC is a generalized hierarchical Golay sequence [36], and is common to all the cells in the system. S-SCH is provided for frame synchronization and BS scrambling code group identification. S-SCH consists again of a bursty transmission, at the beginning of each time slot, of a 256-chips sequence, the SSC. In this case, however, SSC change from slot to slot and is repeated at a frame level. The macro-sequence of 15 SSC repeated at a frame level, is BS specific and identifies the particular scrambling code group used by the serving BS among all the 64 scrambling code groups. The 64 macro-sequences belong to a 16 level Reed Solomon code and they are chosen so that their cyclic-shifts are unique, i.e. a non-zero cyclic shift of any of the 64 sequences is not equivalent to any cyclic shift of any other of the 64 sequences, nor to any cyclic shift, less than 15, of the sequence itself [36]. In this way, once the macro-sequence is identified, the frame boundaries and the scrambling code group are identified as well. The S-SCH structure is reported in Figure 118; $ac_s^{i,k}$ indicates the SSC, being $i = 0, 1,\ldots,63$ the number of the scrambling code group, and $k = 0,1,\ldots,14$ the slot number. SSC are complex values with identical real and imaginary parts and are constructed from position wise multiplication of an Hadamard sequence and a sequence *z* defined in [36]. P-CPICH is provided for scrambling code acquisition. It is a fixed rate (30 kbps, spreading factor 256) downlink physical channel carrying a pre-defined symbol sequence. P-CPICH is always spread by the same OVSF channelization code and scrambled by the primary scrambling code of the serving BS. Once the pilot symbol sequence is correctly detected, the primary synchronization code is identified. The frame structure of the P-CPICH is reported in Figure 119. There is one and only one P-CPICH per cell and it is broadcast over the entire cell.

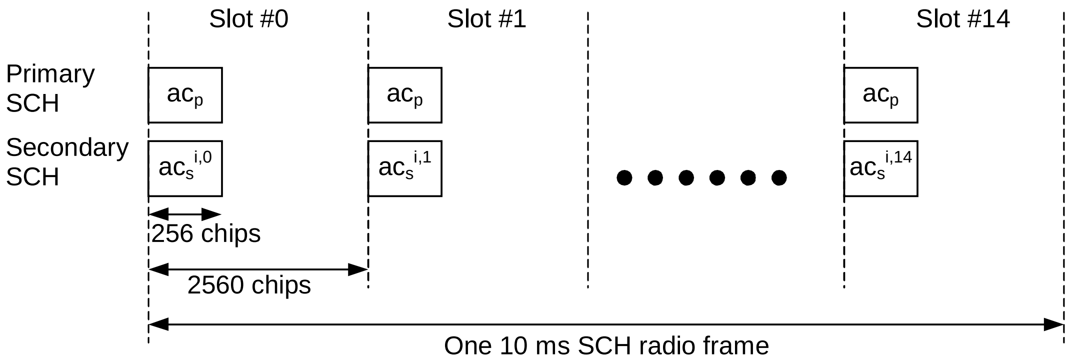


Figure 118: SCHs in W-CDMA air interface

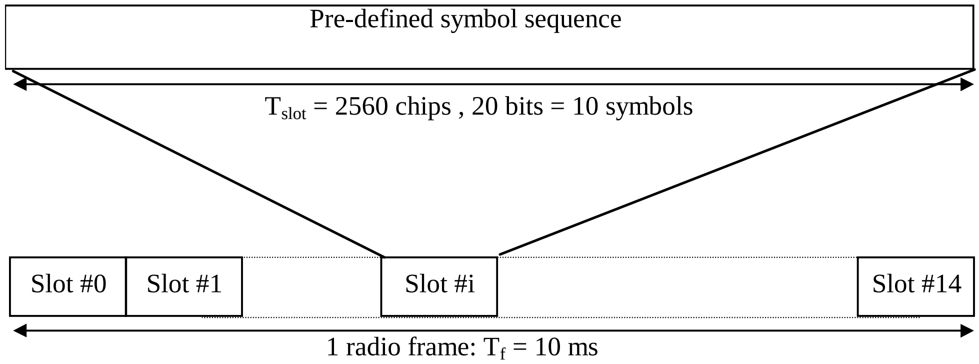


Figure 119: P-CPICH structure

According to the loading factors concept, the cell search performance is evaluated with different values of (α,β,γ).

F.2.6.2 Power control

Uplink power control

The only uplink power control relevant for SATIN involves the RACH procedure.

Optional SATIN scenario

The random access procedure for the optional SATIN scenario can happen in two distinct ways, depending on the whether the UE is within T-UMTS coverage:

1) The UE is within T-UMTS coverage, and will attempt to access the T-UMTS network, as this is the most power efficient option. The standard UTRA FDD random access procedure is carried out, refer to baseline SATIN scenario random access procedure/uplink power control.

2) The UE is out of T-UMTS coverage, and will attempt to access the S-UMTS network with the SATIN specific one-shot acquisition random access procedure straight to the satellite.

Only case 2 is discussed in this clause.

General

Below are some characteristics which are relevant for the power control of the RACH:

1) One-shot acquisition is targeted, i.e. aiming at a *high probability of success* at the first transmission attempt. Power ramping is thus still employed but the occurrence should be sporadic.

2) Message power ramping is employed instead of preamble power ramping.

3) Acquisition indication is communicated through the FACH.

4) Before the random-access procedure can be initiated, L1 receives a value for the initial PRACH transmission power from higher layers (RRC), as well as a value for the power ramp-up increment (ΔP_0), the satellite receiver noise and interference level and the CPICH TX power.

The transmit power of the UE shall be calculated by the following equation:

$$PRACH = L_{SAT\text{-}UE} + I_{SAT} + \text{Constant value},$$

where,

PRACH: Transmitter power level in dBm.

$L_{SAT\text{-}UE}$: Measured path loss in dB between satellite and UE, $L_{SAT\text{-}UE}$ = CPICH TX power - CPICH RSCP, CPICH TX power is broadcast on the BCH.

I_{SAT}: UL interference signal power level at the satellite receiver input in dBm, which is broadcast on BCH.

Constant value: This value shall be designated via Layer 3 message (operator matter).

Setting of PRACH control and data part power difference

The message part of the uplink PRACH channel shall employ gain factors to control the control/data part relative power as described above in the baseline SATIN scenario (T-UMTS procedure).

Downlink power control

The merit of downlink power control is limited as it is for MBMS. If at all power control would be implemented, it would have to be a *group* power control, no dedicated PC treatment can be given.

Power control involving physical multicasting (addressing a group of users with one channel) requires more complexity in processing compared to the dedicated case because essentially a group of users, all with different instantaneous link characteristics, has to be dealt with. A straightforward solution would be to adjust the downlink transmitted power to support the connection to the receiver with the highest power requirement among all receivers in the multicast group, but that would be inefficient in co-operation with the advanced coding schemes (clause F.2.5). Therefore, it would be beneficial for the power control to be statistically based on the link information coming from the UEs in the group.

Closed loop power control is not feasible because the absence of a 2-way dedicated connection between UE and RNC. FACH/RACH interactivity is not suited for closed loop power control. In addition, a broadcast/multicast system with a large number of active users per cell will not have the uplink capacity to process the power control messages that have to be frequently sent by each UE to support closed loop power control (every slot as in UTRA FDD or even every frame).

An open loop power control procedure is proposed, i.e. the Tx estimates the power level needed to offer the users the required service.

Basically 3 alternative approaches can be distinguished:

1) Uplink SNIR based downlink power control:

- Measure the UL pathloss on the RACH pilot bits ($L_{UE\text{-}SAT}$), which are in the control part of the RACH message, before initiating a random access attempt.

  $L_{UE\text{-}SAT}$ = RACH_Tx_power - RACH_RSCP (all in dBm).

- Each time a RACH is sent to the satellite, include RACH_TX_power to communicate this value to the satellite/GW.

- Compute the average of measured $L_{UE\text{-}SAT}$ values for UEs belonging to a certain region at the GW, with help of GPS position knowledge.

- Measure the UL interference at the satellite (I_{UL}) before initiating a random access attempt.

- FACH_Tx_power = average_$L_{UE\text{-}SAT}$ + I_{UL} + constant value (all in dBm).

NOTE: The constant value is connected to the SNIR target and is signalled from higher layers.

2) Uplink pathloss based power control:

- Measure the UL pathloss on the RACH pilot bits ($L_{UE\text{-}SAT}$), which are in the control part of the RACH message, before initiating a random access attempt.

  $L_{UE\text{-}SAT}$ = RACH_Tx_power - RACH_RSCP (all in dBm).

- Each time a RACH is sent to the satellite, include RACH_TX_power to communicate this value to the satellite/GW.

- Compute the averages of measured $L_{UE\text{-}SAT}$ and I_{DL} values for UEs belonging to a defined region at the GW, with help of GPS position knowledge.

- Measure DL interference at the UE (I_{DL} = USRA_Carrier_RSSI) before initiating a random access attempt and communicate to satellite/GW via the RACH.

- FACH_Tx_power = average_$L_{UE\text{-}SAT}$ + average_I_{DL} + constant value (all in dBm).

UL interference is not an accurate estimate of the DL interference (biased because of UL being asynchronous and DL synchronous), hence it is more accurate to measure the DL interference straight away. As a drawback, this alternative requires more capacity from the RACH for the I_{DL} signalling.

3) Downlink SNIR feedback to satellite/gateway:

- Measure the DL pathloss on the CPICH ($L_{SAT\text{-}UE}$) before initiating a random access attempt.

  $L_{SAT\text{-}UE}$ = CPICH_Tx_power - CPICH_RSCP (all in dBm).

  CPICH_Tx_Power is communicated over the BCH.

- Measure DL interference at the UE (I_{DL} = USRA_Carrier_RSSI) before initiating a random access attempt, and communicate to satellite/GW via the RACH.

- Each time a RACH is sent to the satellite, include ($L_{SAT\text{-}UE}$ + I_{DL}) to communicate this value to the satellite/GW.

- Compute the average of measured (L_{SAT-UE} + I_{DL}) values for UEs belonging to a defined region at the GW, with help of GPS position knowledge.
- FACH_Tx_power = average_(L_{SAT-UE} + $I_{DL)}$ + constant value (all in dBm).
- This is the most straightforward approach, however, also individual-user short-term/fast-varying effects are measured (e.g. multipath fading because of IMR presence) and this is not necessary.

Individual user link variations cannot and must not be tackled by the power control mechanism for broadcast/multicast traffic. The downlink power control in SATIN should moderate variations in the link quality that are common for all users, or a large part of the users, in a BC/MC group, i.e. slowly varying, large-scale variations caused by e.g. cloud variations, rain shadowing, atmosphere variations. Alternative 2 seems to be the best solution according to the above reasoning, provided the inclusion of both RACH_Tx_power and I_{DL} values into the RACH is not an issue. If it is, option 3 seems to be the most suitable alternative.

F.2.6.3 S-UMTS paging

S-UMTS air interface paging procedure is equal to the terrestrial one.

Baseline scenario and Optional scenario:

As regard to terrestrial paging information, no changes are expected respect to UTRA FDD air interface.

Vice versa, satellite-paging information may be delivered in two different modes:

- direct procedure via satellite link;
- indirect procedure via terrestrial link.

Direct procedure employs the PICH and the PCH of the satellite air interface. In this way, the S-UMTS link manages the satellite paging procedure and does not overload the terrestrial link. A mobile terminal that receives a broadcast-multicast service via S-UMTS has to hear paging information both in terrestrial and satellite band. Considering the large amount of users to be addressed by one PICH in a time-shared fashion (because of the large cell size), and the requirement on the DRX cycle to correspond to a realistic maximum response time, it can already be stated here that the maximum number of possible paging indicators (PIs) in a PICH frame will most probably have to be used, being 144 PIs in one PICH frame.

An indirect procedure involves the terrestrial-UMTS in order to support the satellite paging. The use of an indirect approach presupposes a hard synergy between UTRAN and USRAN because the paging messages generated by USRAN have to be delivered via terrestrial link: the terrestrial link will be overloaded with satellite paging information. A positive aspect of the indirect procedure is an efficient use of the satellite link resources; in fact entire paging procedure is passed on T-UMTS. Moreover, for the terminal, it is simpler to monitor only the terrestrial PICH respect to both terrestrial and satellite PICH.

F.2.6.4 RACH procedure

The two main approaches for the uplink access are Random Access Spread Aloha (RASA) and dynamic Rate on Demand (dRoD), as proposed in the ESA ATB project. The RASA approach is purely UE initiated and is based on the random access procedure as it is employed in UTRA FDD, using RACH/CPCH transport channels. The dRoD option is in essence based on a reservation scheme, so the gateway has the control in granting a channel for the UE to use (this could be a DCH). Selection of the appropriate approach is based on the amount of traffic to be sent, and more specifically, the length of the message it will be sent in. The maximum bearer information rate provided by the RACH during its transmission period may be in the order of 40 kbps. The large amount of overhead per radio frame (or double radio frame), mainly due to the preamble part as well as the high collision risk, indicates that it is not wise to transmit large amounts of packet data via the RACH. Closed loop power control is also not possible. However, if packet frequency is low and packets are small, then the RASA method using the RACH is adequate since it does not require signalling overhead for dedicated channel assignment.

The bulk of the uplink traffic in the SATIN case will most probably be bursty, short-lived packet data, so the RASA option seems to be the most relevant. The dRoD alternative can be kept as a fallback in case of large messages. It should be investigated if this option is relevant enough to be further studied in SATIN.

Note that for a higher random access efficiency it is advisable to use *slotted* RASA. The access slot timing is relative to the P-CCPCH. In UTRA FDD there are 15 access slots per two frames. This will be considerably less in a GEO satellite environment.

In a GEO satellite environment, the round-trip delay is approx. 500 ms. Multiple preamble transmission (access probing) by successively increasing UE power as well as fast access and collision detection indication on the forward-link are certainly *not feasible* in a satellite system. All these methods require immediate response by the network, which cannot be provided when there is a large propagation delay.

As a result, taken into account the long propagation delay for the satellite case, it is better to send the preamble together with the message part and ensure a high probability of success at the first transmission attempt (i.e. one shot acquisition). The UE set the Retransmission Counter to a maximum value and send the preamble and the message part. Only when the preamble is successfully detected, the message part will be decoded. If the message block (consisting of preamble and message part) fails to be detected and decoded, then the whole block is retransmitted with a higher power (message power ramping: the power step ΔP_0 is specified on BCH) and the retransmission counter is decreased. The procedure ends when the counter is equal to zero or an acknowledgement is received from the BS. Hence, preamble power ramping method is not employed and also with this method, no AICH channel is needed, as acquisition indication is received on the FACH channel.

Notably, in this proposal, only one access slot per radio frame can be supported, as compared to 15 access slots available per two radio frames in 3GPP for terrestrial-UMTS (in both cases, 16 signatures are available). The effect of this is that, when the access request increases, the collision risk increases as well and the RACH performance degrades. In particular, collision will occur when there is more than one request selecting the same signature.

Hence, it can be concluded that using a longer preamble (note that a longer preamble than the one used in the terrestrial is inherently needed in the satellite case due to the increased cell size in the satellite environment) scheme, as proposed by ETSI, on one hand enhances the synchronization and acquisition procedure performance, but on the other hand, it increases the risk of collision when the attempt rate is high.

Among the directions that can be looked at to reduce the risk of collision are increasing the number of access slots in relation with reducing the size of the preamble search window, and increasing the number of available signatures per access slot. These will surely introduce the issues of further trade-off between the collision risk and the preamble detection probability.

Slotted vs Unslotted Aloha

Within [47], it is unclear if a slotted or unslotted approach should be used. Nevertheless the comparison between slotted and unslotted spread aloha for CDMA packet access has been the subject for considerable research activity in the recent or not so recent past. The results are dependent on a large number of system assumptions, which makes it necessary for some caution in drawing conclusions. However, some relevant points can be clearly identified:

There is definitely not a significant throughput gain (or none at all, depending on spreading factor and code family) in imposing a slotted structure, because of the inherent robustness of a spread spectrum signal with respect to interference. This is not the case for a TDMA signal, in which the throughput nearly doubles for slotted aloha. However, differently from the TDMA case, both in the W-CDMA and in the SW-CDMA proposals interference is not completely avoided by the use of a slotted approach and the resource is shared by exploiting the pseudo-noise codes cross-correlation properties.

Code collision only happens when two preambles using the same pseudo-noise sequence arrive at the receiver within one chip period (i.e. 260 ns). Through a GEO satellite link to mobile users, as the one assumed in SATIN, this is a very unlikely event. This is the reason why a small number of codes may be sufficient for the RACH procedure, allowing for smart firmware solutions to resolve multiple instances of a code. Only if extremely tight timing requirements are imposed for a "strict-sense" slotted aloha (i.e. chip level accuracy), may the necessity for a larger number of codes arise. In this case, a feasibility investigation would be necessary.

There are some advantages in a "pragmatic" slotted aloha, namely a slotted structure requiring accuracy of around 5 μs to 10 μs, in terms of detecting the presence of a RACH burst. This is because there is no ramping and because threshold setting must be adapted in fading conditions to an estimate of disturbance level in order to optimize performance in Neyman-Pearson terms. However, these advantages may or may not justify the consequent complexity increase, both at system and terminal levels, that follows the choice of slotting the access.

Nevertheless as mentioned in TR 101 866 [47], a slotted structure may help to easily determine the round trip delay in the gateway and also since access request can only be sent within these access slots, the access requests are more randomly distributed and this will reduce the interference within access slots. Thus a reasonable approach for S-UMTS could be the definition of one access slot per radio frame [47].

Finally, transmission power for the RACH will be based on open loop power control. An adequate margin, to account for the open loop inaccuracy, should be included in the calculation of the RACH transmission power.

The physical random access procedure described is initiated upon request of a primitive from the MAC sublayer. Before the physical random-access procedure can be initiated, Layer 1 receives the following information from the higher layers (RRC):

- the preamble spreading code(s)/message scrambling code(s);
- the message length in time, either 10 ms or 20 ms;
- the available spreading factors for the message part;
- the satellite receiver noise plus interference level;
- the CPICH current transmit power level;
- the power ramp-up increment (ΔP_0);
- the nominal down-link centre of beam carrier frequency;
- the applied down-link centre of beam frequency correction;
- the available signatures for each Access Service Class (ASC);
- the maximum power ramping parameter;
- the set of Transport Format parameters.

At each initiation of the physical random access procedure, Layer 1 shall receive the following information from the higher layers (MAC):

- the Transport Format to be used for the PRACH message part;
- the ASC (Access Service Class) of the PRACH transmission;
- the data to be transmitted (Transport Block Set).

The physical random-access procedure shall be performed as follows:

1) Randomly select a signature from the available signatures for the given ASC. The random function shall be such that each of the allowed selections is chosen with equal probability.
2) Set the Retransmission Counter to the maximum power ramping parameter.
3) Set the PRACH transmission power according to the value previously calculated.
4) Transmit RACH using the selected signature, and pre-computed transmission power.

5) After a predefined interval (greater than the estimated satellite round trip time), if the UE does not detect an acquisition indication on the FACH channel with the selected UE ID, it:

   5.1) Randomly selects a new signature from the available signatures within the given ASC. The random function shall be such that each of the allowed selections is chosen with equal probability.

   5.2) Increases the transmission power by ΔP_0.

   5.3) Decreases the Retransmission Counter by one.

   5.4) If the Retransmission Counter > 0 then repeat from step 4; otherwise informs to the higher layer (MAC) and exit the physical random access procedure.

6) If a negative acquisition indicator corresponding to the selected signature is detected in the FACH, inform to the higher layer (MAC), and exit the physical random access procedure.

F.2.7 UE physical layer measurement abilities

Depending on e.g. the strategy for Radio Resource Management (handovers, power control, load control etc.) a subset of the existing UTRA FDD measurements (possibly in modified form).

The list of measurements, which the physical layer reports to higher layers, is described below.

F.2.7.1 SFN-CFN observed time difference

This measure is mandatory for UE.

Table 24

Measurement	SFN-CFN observed time difference
Source	L1 (UE)
Destination	RRC (RNC) for handover
Reporting Trigger	On-demand, Event-triggered
Description	Time difference between the SFN of the target neighbouring cell and the CFN in the UE.

This measurement is used in UTRA FDD for handover between neighbouring FDD cells. In SATIN there will be no such handover.

CFN_{Tx} is the connection frame number for the UE transmission of an uplink DPCCH/DPDCH frame at the time T_{UETx}, hence not applicable to both the Optional (RACH uplink) and the Baseline (uplink and downlink are separate systems) scenarios in SATIN.

This measurement is irrelevant for SATIN.

F.2.7.2 Observed time difference to GSM cell

This measure is mandatory for UE capable of handover to GSM.

Table 25

Measurement	Observed time difference to GSM cell
Source	L1 (UE)
Destination	RRC (RNC) for maintenance and handover to GSM
Reporting Trigger	On-demand, Event-triggered
Description	Time difference between a UTRA cell and a GSM cell.

Handover to GSM is not targeted within SATIN.

This measurement is irrelevant for SATIN.

F.2.7.3 P-CCPCH RSCP

This measure is mandatory for UE with TDD mode capability.

Table 26

Measurement	P-CCPCH RSCP
Source	L1 (UE)
Destination	RRC (UE, RNC)
Reporting Trigger	periodic or event triggered
Description	Received signal code power of the P-CCPCH

No TDD mode capability is envisaged, leaving this measurement irrelevant for SATIN.

F.2.7.4 Timeslot ISCP

This measure is mandatory for UE with TDD mode capability.

Table 27

Measurement	Timeslot ISCP
Source	L1 (UE)
Destination	RRC (UE, RNC)
Reporting Trigger	periodic or event triggered
Description	Interference Signal Code Power is the interference on the received signal in a specified timeslot.

No TDD mode capability is envisaged, leaving this measurement irrelevant for SATIN.

F.2.7.5 SIR

This measure is mandatory for UE with TDD mode capability.

Table 28

Measurement	SIR
Source	L1 (UE)
Destination	RRC (UE,RNC)
Reporting Trigger	Periodic, once every power control cycle, event triggered
Description	Signal to Interference Ratio

No TDD mode capability is envisaged, leaving this measurement irrelevant for SATIN.

F.2.7.6 GSM carrier RSSI

This measure is mandatory for UE with GSM capability.

Table 29

Measurement	GSM carrier RSSI
Source	L1 (UE)
Destination	RRC (RNC)
Reporting Trigger	Periodic, event triggered, on demand
Description	Received Signal Strength Indicator, the wide-band received power within the relevant channel bandwidth. Details are specified in TS 100 911 [48].

No GSM mode is envisaged within SATIN, this measurement is considered irrelevant.

F.2.7.7 UE Rx-Tx time difference

This measure is mandatory for UE with FDD mode capability.

Table 30

Measurement	UE Rx-Tx time difference
Source	L1 (UE)
Destination	RRC (RNC)
Reporting Trigger	On-demand, periodic, event-triggered
Description	Time difference between the UE uplink DPCCH/DPDCH frame transmission and the first detected path (in time) of the downlink DPCH frame from the measured radio link. Type 1 and Type 2 are defined.

No uplink-downlink dedicated channel relation. This measurement is irrelevant for SATIN.

F.2.7.8 SFN-SFN Observed time difference

This measure is mandatory for UE.

Table 31

Measurement	SFN-SFN observed time difference
Source	L1 (UE)
Destination	RRC (RNC)
Reporting Trigger	On-demand, Event-triggered
Description	Time difference between a specific reference UTRA cell and a target UTRA cell. Type 1 and Type 2 are defined.

Used in handover procedure. Irrelevant to SATIN.

F.2.7.9 Timing Advance (T_{ADV}) for 1,28 Mcps TDD

This measure is mandatory for 1,28 Mcps TDD UE.

Table 32

Measurement	Timing Advance (T_{ADV}) for 1,28 Mcps TDD
Source	L1 (UE)
Destination	RRC (RNC)
Reporting Trigger	On-demand, Event-triggered, Periodic
Description	Difference between the uplink transmission of the UE and the downlink reception.

No TDD mode in SATIN, irrelevant measurement.

F.2.7.10 Physical channel BER

Table 33

Measurement	Physical channel BER
Source	L1(Node B)
Destination	RRC (RNC)
Reporting Trigger	On-demand, Event-triggered, periodic
Description	The Physical channel BER is an estimation of the average bit error rate (BER) on the DPCCH of a Radio Link Set. This measurement applies to FDD mode only.

Irrelevant to SATIN because there is no use of DPCCH.

F.2.7.11 RX timing deviation

Table 34

Measurement	RX timing deviation
Source	L1 (Node B)
Destination	RRC (RNC)
Reporting Trigger	Periodic, event triggered
Description	The difference of the time of arrival of the UL transmissions in relation to the arrival time of a signal with zero propagation delay. This measurement is applicable for TDD mode.

Irrelevant because TDD mode capability is not envisaged in SATIN.

F.2.7.12 Timeslot ISCP

Table 35

Measurement	Timeslot ISCP
Source	L1(Node B)
Destination	RRC (RNC)
Reporting Trigger	periodic or event triggered
Description	Interference on Signal Code Power; it is the interference on the received signal in a specified timeslot. This measurement is applicable is applicable to TDD mode only.

Irrelevant because TDD mode capability is not envisaged in SATIN.

F.2.7.13 RSCP

Table 36

Measurement	RSCP
Source	L1 (Node B)
Destination	RRC (RNC)
Reporting Trigger	periodic or event triggered
Description	Received Signal Code Power is the received power on DPCH or PRACH or PUSCH. This measurement is applicable for TDD mode only.

Relevant to downlink power control procedure options 1 and 2.

F.2.7.14 Acknowledged PRACH preambles

Table 37

Measurement	Acknowledged PRACH preambles
Source	L1 (Node B)
Destination	RRC (RNC)
Reporting Trigger	Periodic, event triggered, On demand
Description	This measurement indicates the number of positive acquisition indicators transmitted per access frame on each AICH ***(replace by "on the FACH (S-CCPCH)")***. This measurement is applicable for FDD mode only.

Irrelevant to SATIN because of the different random access approach.

F.2.7.15 Detected PCPCH access preambles

Table 38

Measurement	Detected PCPCH Access preambles
Source	L1 (Node B)
Destination	RRC (RNC)
Reporting Trigger	Periodic, event triggered, On demand
Description	This measurement indicates the total number of detected access preambles per access frame on the PCPCHs belonging to a CPCH set. This measurement is applicable for FDD mode only.

Irrelevant to SATIN, CPCH is not used.

F.2.7.16 Acknowledged PCPCH access preambles

Table 39

Measurement	Acknowledged PCPCH access preambles
Source	L1 (Node B)
Destination	RRC (RNC)
Reporting Trigger	Periodic, event triggered, On demand
Description	This measurement indicates the total number of acknowledged PCPCH access preambles per access frame on the PCPCHs. where an access frame consists of fifteen access slots from access slot #0 to access slot #14. This measurement is applicable for FDD mode only.

Irrelevant to SATIN, CPCH is not used.

F.2.7.17 SIR

Table 40

Measurement	SIR
Source	L1 (Node B)
Destination	RRC (RNC)
Reporting Trigger	Periodic, event triggered
Description	Signal to Interference Ratio.

The same holds for SATIN.

F.2.7.18 PRACH/PCPCH Propagation Delay

Table 41

Measurement	Propagation delay
Source	L1(Node B)
Destination	RRC (RNC)
Reporting Trigger	Event triggered, periodic
Description	The one-way propagation delay as measured during either PRACH or PCPCH access. This measurement is applicable for FDD mode only.

Irrelevant to SATIN because of the different random access approach.

F.2.7.19 UTRAN GPS Timing of Cell Frames for UE positioning

Table 42

Measurement	UTRAN GPS Timing of Cell Frames for UE positioning
Source	L1 (LMU)
Destination	RRC (RNC-UE positioning)
Reporting Trigger	On-demand, Event-triggered, Periodic
Description	This is the absolute time reference measurement in respect to GPS Time Of Week for the transmission of a particular frame.

UE position knowledge (context) is envisaged in SATIN. This measurement is relevant.

F.2.7.20 SIR ERROR

Table 43

Measurement	SIR ERROR
Source	L1(Node B)
Destination	RRC (RNC)
Reporting Trigger	Periodic, event triggered
Description	Signal to Interference Ratio Error This measurement is applicable for FDD cells only.

The same holds for SATIN.

F.2.8 UE transmission and reception

TS 125 101 [38] frame can be kept, but must be adapted to the SATIN architectural definition and access scheme definition. Below is a list indicating which clauses/items will need adaptation:

General: power classes.

Frequency bands and channel arrangement: frequency bands, TX_RX frequency separation, channel number.

Transmitter characteristics: UE output power related items, power control related items.

Receiver characteristics: antenna diversity, reference sensitivity.

Performance requirements: modify for SATIN relevant channels, modify requirements involving power control, modify paging and acquisition related requirements according to the SATIN access scheme definition.

Measurement channels: based on the services targeted in SATIN, the channels selected, and the corresponding data rates, a convenient set of measurement channels needs to be defined.

Propagation Conditions:

This clause in [38] describes the propagation conditions used for performance measurements. The purpose is rather to test all functionality of the receiver baseband. They need not necessarily be realistic since testability needs to be taken into account. These propagation conditions are tailored for UTRA baseband testing (primarily simulating) and therefore differ from the more general purpose ITU propagation channels, which are specifically designed to evaluate the relative performance of candidate RTTs, refer to [39]. Nevertheless, the ITU propagation models in [39] are related to the propagation conditions in [38]. In [38] propagation conditions were derived from ITU models by redistributing the rays (taps) in such a way that they coincide with multiples of the chip period, and optimizing the different conditions so that each case tests separate aspects of receiver baseband functionality.

The specific aspects for a channel model which are different in the SATIN case with respect to the normal terrestrial case are due to the presence of the IMRs (introducing a high amount of artificial multipath) and to lesser extent the presence of the direct path from the satellite. These are the consequences:

1) Wider delay spread (reflected in multipath fading propagation conditions, case 2).

2) Larger number of (significant) signal components (reflected in multipath fading propagation conditions, case 5 and case 6).

3) Presence of a direct path from the satellite, preceding all other paths (reflected in multipath fading propagation conditions, case 5 and case 6).

The different propagation conditions, different cases and modifications to suit SATIN are discussed hereafter.

STATIC CHANNELS:

Static propagation condition

The propagation for the static performance measurement is an Additive White Gaussian Noise (AWGN) environment. No fading and multi-paths exist for this propagation model.

No changes wrt [38].

Multi-path fading propagation conditions

The multi-path channel is modelled by a tapped delay-line, with all taps having classical Doppler spectrum.

TS 125 101 [38] specifies 6 distinct multi-path fading propagation condition cases.

Table 44

Case 1, speed 3 km/h		Case 2, speed 3 km/h		Case 3, speed 120 km/h		Case 4, speed 3 km/h		Case 5, speed 50 km/h (see note)		Case 6, speed 250 km/h	
Relative Delay [ns]	Average Power [dB]	Relative Delay [ns]	Average Power [dB]	Relative Delay [ns]	Average Power [dB]	Relative Delay [ns]	Average Power [dB]	Relative Delay [ns]	Average Power [dB]	Relative Delay [ns]	Average Power [dB]
0	0	0	0	0	0	0	0	0	0	0	0
976	-10	976	0	260	-3	976	0	976	-10	260	-3
		20 000	0	521	-6					521	-6
				781	-9					781	-9
NOTE: Tap at 0 ns is Rice distributed.											

case 1:

Measures how well a receiver works in a flat fading environment. This model was originally derived from the ITU indoor channel model.

No change needed for SATIN.

case 2:

To test diversity gain in a receiver and long delays (largest tap delay = 20 μs; largest delay to be found in the ITU models). Note that all taps are at the same level (in order to test the gain easily). This model was originally derived from ITU pedestrian channel model and ITU high antenna vehicular model.

The 20 μs delay could be replaced by a longer delay because the IMRs introduce extra delay, refer to clause 2.4 in [46]. The extra delay spread is estimated at 24,2 chip times, i.e. 6,3 μs. When positioning the longest delay tap on a multiple of a chip period, 26 563 ns can be considered a safe new value taking into account the extra delay introduced by the IMR environment.

case 3:

To test demanding allocation of RAKE receiver to put fingers 1 chip apart. Derived from ITU vehicular model.

This model is relevant for SATIN in its present from, but it represents a situation where the UE is close to an IMR, refer to Figure 120. When a UE is a bit further away from the closest IMR, so when the UE is getting closer to the edge of a "cell" (i.e. the hexagonal cellular layout as reported in clause 2.4 in [46]), the composite delay spread profile is more spread out, and the number of significant signal components is larger, refer to Figure 121. To cover also this situation, a new case is added: case 5. Case 6 is case 5 for high speed mobiles.

NOTE: Figure 120 and Figure 121 are derived from the composite power delay profiles. Each ray is split into two rays, one sample to the left and one to the right. The power of these new rays is such that the sum is equal to the original power, and the power of each of the new rays is inversely proportional to the distance to the original ray. Finally the power of all rays on one sample are added up. This yields a model with a number of independent Rayleigh fading rays on the sampling instants, i.e. multiples of the chip period in this case.

case 4:

Propagation condition for the inner and outer loop power control testing.

Not expected to be relevant.

case 5:

Propagation condition for RRC connection mobility testing (only used in [30]).

Not expected to be relevant.

case 6:

Propagation condition for high speed mobility. Similar to case 3.

general remark:

The relative delays in case 1 and case 2 are based on the former UMTS chip rate of 4,096 Mcps (T_{chip} = 244,14 ns). As it is more convenient to implement models in simulations with the taps on multiples of the chip period, as is done in case 3. Tap positions on non-integer multiples of a chip period are accounted for in the moving propagation condition. Therefore it seems logical to adjust also cases 1 and 2 to the current UMTS chip rate of 3,84 Mcps (T_{chip} = 260,42 ns).

Table 45: Proposal for new multipath fading propagation conditions

Case 1, speed 3 km/h		Case 2, speed 3 km/h		Case 3, speed 120 km/h		Case 4, speed 250 km/h		Case 5, speed 120 km/h (see note)		Case 6, speed 250 km/h (see note)	
Relative Delay [ns]	Average Power [dB]	Relative Delay [ns]	Average Power [dB]	Relative Delay [ns]	Average Power [dB]	Relative Delay [ns]	Average Power [dB]	Relative Delay [ns]	Average Power [dB]	Relative Delay [ns]	Average Power [dB]
0	0	0	0	0	0	0	0	0	-3	0	-3
1 042	-10	1 042	0	260	-3	260	-3	260	-3	260	-3
		26 563	0	521	-6	521	-6	521	-9	521	-9
				781	-9	781	-9	1 042	-3	1 042	-3
								1 302	-3	1 302	-3
								1 562	-3	1 562	-3
								1 823	0	1 823	0
								2 083	0	2 083	0
NOTE: Tap at 0 ns is Rice distributed.											

case 1:

Adjusted to 3,84 Mcps chipping rate.

case 2:

Last tap adjusted to reflect the wider delay spread caused by the multiple IMR environment.

case 3:

No change. This case reflects the situation where the UE is close to an IMR. It is based on the chart displayed in Figure 120.

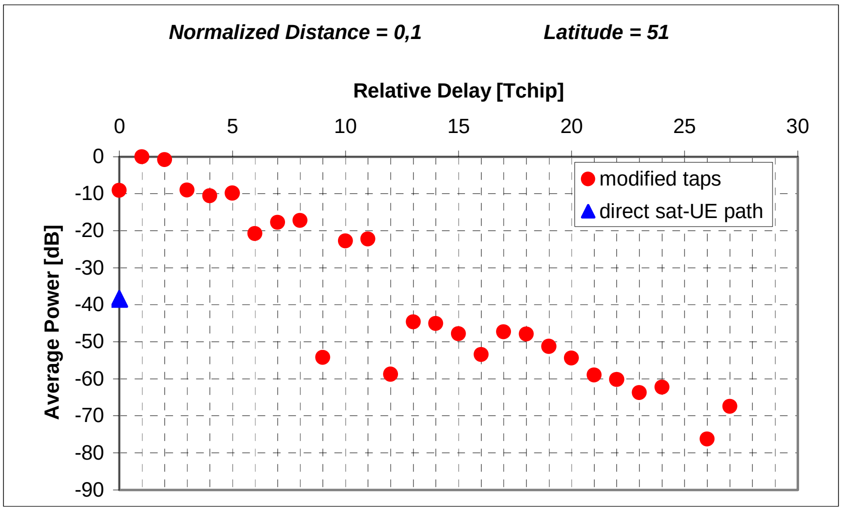


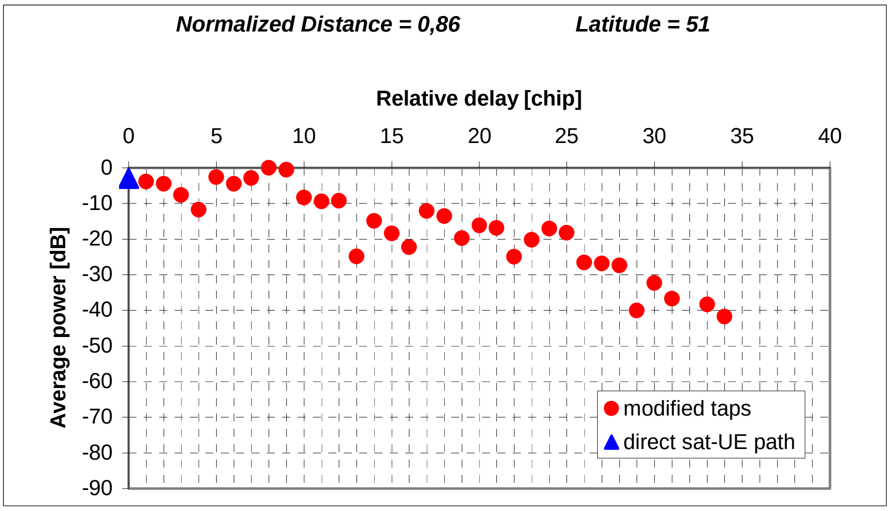
Figure 120: Power-delay profile when UE is very close to IMR (Low Power IMRs)

case 4:

No change. Same as case 4, but for high speed.

case 5:

Case 3 adapted to the situation where a UE is further away from the closest IMR, i.e. in between IMRs. Based on the charts displayed in Figure 121.



NOTE: in this case the tap representing the path straight from the satellite is significant. This tap is Rice distributed, as opposed to the other taps which are Rayleigh distributed.

Figure 121: power-delay profile when normalized distance is 0,86 (Low Power IMRs)

case 6:

Same as case 5, but for high speed.

DYNAMIC CHANNELS

Moving propagation condition

The purpose of a moving channel model is to test that the UE can track when the paths delay spread is varying. It is a non fading channel model with two taps, one static, Path0, and one moving, Path1. The taps have equal strengths and equal phases. The delay spread between the two paths is varying according to a sinusoid. Figure 122 shows the channel.

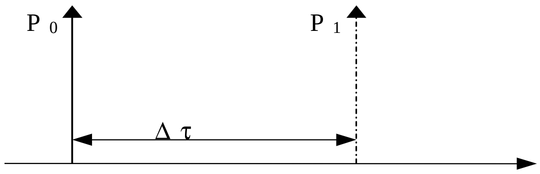


Figure 122: The moving propagation condition

$$\Delta\tau = B + \frac{A}{2}\left(1 + \sin(\Delta\omega \cdot t)\right)$$

The parameters in the equation are shown in Table 46.

Table 46

Parameter	Value
A	5 μs
B	1 μs
Δω	40×10^{-3} s^{-1}

These parameters are based on a speed of approximately 100 km/h.

Note also that this model is useful to identify possible performance problems when the tap delay is a random non-integer multiple of the chip period.

This model should not be modified for SATIN.

Birth-Death propagation condition

The purpose of a birth-death channel model is to test that the UE can find new paths when something suddenly happens in the environment. To be able to easily measure performance this channel is quite extreme. In the channel model the birth death model will be much faster than in reality. This will also be reflected in the performance on this channel.

This is how it is specified in [38].

The dynamic propagation conditions for the test of the base band performance is a non fading propagation channel with two taps. The moving propagation condition has two taps, Path1 and Path2 which alternate between "birth" and "death". The positions the paths appear are randomly selected with an equal probability rate and is shown in Figure 123.

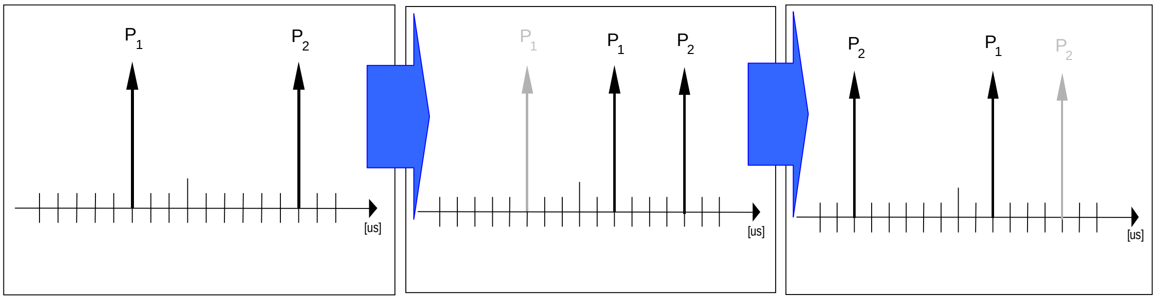


Figure 123: Birth-death propagation sequence

1) Two paths, Path1 and Path2 are randomly selected from the group[-5, -4 ,-3 ,-2 ,-1, 0, 1, 2, 3, 4, 5] µs. The paths have equal magnitudes and equal phases.

2) After 191 ms, Path1 vanishes and reappears immediately at a new location randomly selected from the group [-5, -4, -3, -2, -1, 0, 1, 2, 3, 4, 5] µs but excludes the point Path 2. The magnitudes and the phases of the tap coefficients of Path 1 and Path 2 shall remain unaltered.

3) After an additional 191 ms, Path2 vanishes and reappears immediately at a new location randomly selected from the group [-5, -4, -3, -2, -1, 0, 1, 2, 3, 4, 5] µs but excludes the point Path 1. The magnitudes and the phases of the tap coefficients of Path 1 and Path 2 shall remain unaltered.

The sequence in 2) and 3) is repeated.

This model should not be modified for SATIN.

F.3 Inter-layer procedures

F.3.1 UMTS access network level of connectivity and RRC states

The standard modes and RRC states are applicable to the SATIN case as well. The difference is that UEs camp on spot beams instead of cell. It has to be decided whether the UE stays in this mode to receive the multicast/broadcast content or it has to be in connected mode or in a mode different from idle and connected mode. It is important to consider how S-UMTS and T-UMTS modes are going to coexist. The mode definition also depends of the terminal type (Dual/Parallel etc.) as well.

There are two more cases considered in SATIN scenario excluding pure T-UMTS operational mode of the UE.

- B/M content delivery while the UE is in T-UMTS coverage. In this situation, RL (signalling only) is via T-UMTS and FL (B/M content delivery) via satellite.
- B/M content delivery while UE is in out of T-UMTS coverage. In this situation, both RL (signalling only) and FL (B/M content delivery) are via satellite.

These two cases are not defined as modes and their context has to be defined within SATIN with respect to the standard T-UMTS states (CELL_DCH, CELL_FACH, CELL_PCH and URA_PCH).

Requirements for SATIN are specified in two distinct clauses (F.3.1.1 and F.3.1.2): one for the baseline and another for the optional case.

F.3.1.1 Requirements on states in SATIN baseline case

Baseline configuration with parallel receiver architecture

This receiver can handle T-UMTS and S-UMTS downlink simultaneously.

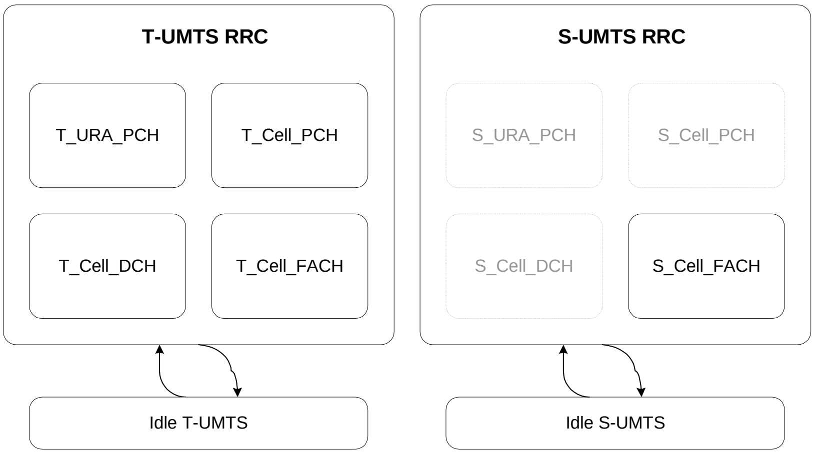


Figure 124: RRC states in baseline configuration with parallel receiver architecture

- S_URA_PCH/S_Cell_PCH

These states may be used as broadcast/multicast services alerting may be achieved by possibly adapted paging type 1 procedure.

- S_Cell_FACH

The same as in terrestrial except that RACH/CPCH will be handled by the terrestrial uplink.

- S_Cell_DCH

The use of a DSCH and DCH will not be possible in the baseline case since no synchronized feedback path is present.

Baseline configuration with reconfigurable receiver architecture

This receiver can be reconfigured from T-UMTS to S-UMTS. Because T-UMTS and S-UMTS downlink cannot run simultaneously and because the T-UMTS uplink always needs a downlink, 2 possible solutions exist:

1) Long term time-multiplexing between an S-UMTS downlink and a T-UMTS duplex connection (e.g. for Push and Store services).

2) Time-switched reception of both downlink modes in such a way that it is transparent for higher layers. This has to be further investigated.

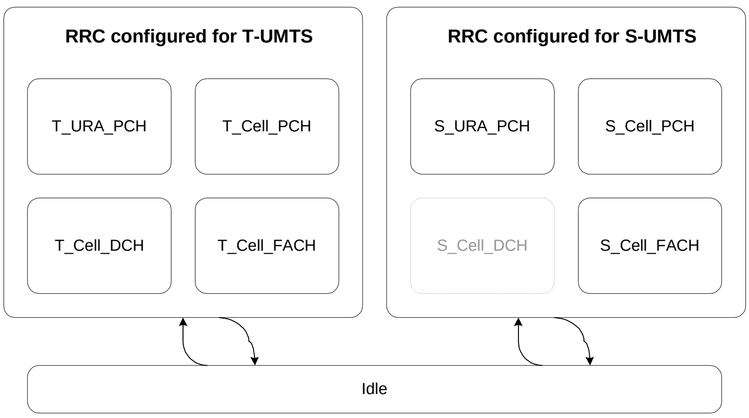


Figure 125: RRC states in baseline configuration with reconfigurable receiver architecture

- S_URA_PCH/S_Cell_PCH

The same as in terrestrial RRC.

- S_Cell_FACH

The same as in T-UMTS except that RACH/CPCH will be handled by the terrestrial uplink.

- S_Cell_DCH

The use of a DSCH and DCH will not be possible in the baseline case since no synchronized feedback path is present.

F.3.1.2 Requirements on states in SATIN optional case

These SATIN Optional states encompass requirements specified in the baseline case plus requirements specific to optional case (e.g. dedicated and/or shared channels, RACH, etc.)

Optional configuration with parallel receiver architecture

This terminal is able to establish a bi-directional link with both the terrestrial network and the satellite network at the same time.

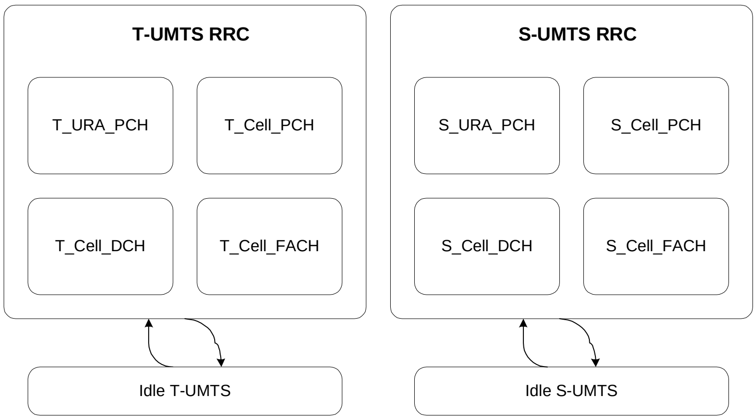


Figure 126: RRC states in optional configuration with parallel receiver architecture

- S_URA_PCH/S_Cell_PCH

The same as terrestrial RRC.

- S_Cell_FACH

The RACH can be used in an adapted version. CPCH is excluded because a fast power control scheme is not possible.

- S_Cell_DCH

The use of DSCH and DCH is possible, although providing MBMS with a DCH is highly inefficient.

Optional configuration with reconfigurable receiver architecture

This receiver can be reconfigured from T-UMTS to S-UMTS.

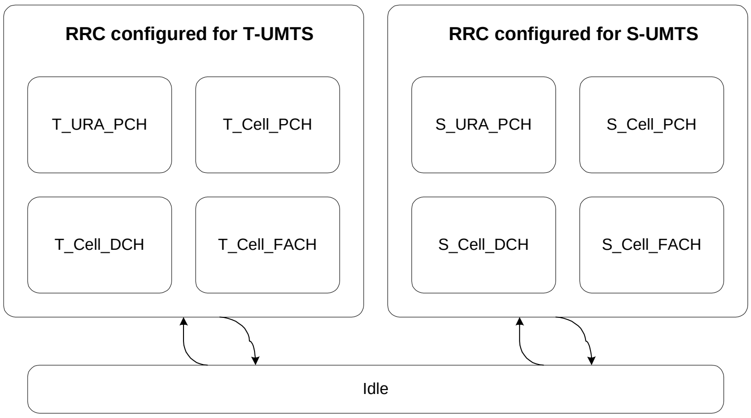


Figure 127: RRC sates in optional configuration with reconfigurable receiver architecture

- S_URA_PCH/S_Cell_PCH

The same as T_Cell_PCH

- S_Cell_FACH

The RACH can be used in an adapted version. CPCH is excluded because a fast power control scheme is not possible.

- S_Cell_DCH

The use of DSCH and DCH is possible, although providing MBMS with a DCH is highly inefficient.

F.3.2 Basic system procedures

F.3.2.1 Paging

The terminal needs to be alerted even for broadcast or multicast content delivery or any T-UMTS bound calls. In the broadcast mode, service discovery and announcement should be enough and UE may be capable to receive on-going or forthcoming transmission of multicast content based on service notification, without need for requesting the network. Now, multicast alerting should be a feature of SATIN access, may be achieved by paging Group ID instead of User ID -> impact to be checked, possible requirement for SATIN access scheme. On the other hand, the users may like to receive the calls while receiving the B/M content. In this case there are few possibilities:

- When UE receives B/M content and it is in the T-UMTS coverage: paging via T-UMTS.
- When UE receives B/M content and it is out of T-UMTS coverage: paging via satellite. But voice services are not supported within SATIN access; thus there will not be need for this scenario. However some users like to know who is trying to call them and they can call if they have the calling party number when they are eventually back to T-UMTS service area.

For the multicast case, it should be specified whether the alerting procedure comes as part of the multicast protocol or not. Again for the broadcast case, it should be decided whether alerting is necessary or not.

Based on the above decisions the paging procedures should be defined.

F.3.2.2 Camping on the cell

In SATIN case, there are two possibilities.

- Camping onto T-UMTS cell

  The procedure is same as normal T-UMTS case, but the situation can be different in the following sense:

  - When terminal switches from satellite broadcast services to T-UMTS services, the terminal has to do the camping with terrestrial network.

- Camping onto S-UMTS cell/spotbeam

- In that case, the procedure is also the same as in the normal T-UMTS case, but the situation can be different in the following sense:

  - The terminal camps on S-UMTS if it is outside the T-UMTS coverage (It always camps on T-UMTS network, when it is switched on, if the T-UMTS coverage is available).

  - When the terminal switches from T-UMTS services to satellite broadcast services, the terminal has to do the camping with satellite network.

How these procedures work, should be further investigated.

F.3.3 Radio resource set-up and release

F.3.3.1 RRC connection set-up

The main requirements with respect to RR set-up procedure are the following:

Baseline case

In SATIN baseline case, B/M delivery is a one way service and shall only be based on unidirectional signalling from the Satellite-RNC to the UEs: no connection request can be received by the Satellite-RNC and complete regular RRC connection set-up procedure prior to Radio Bearer (RB) set-up is not feasible, i.e. the signalling exchange is not possible, whether data link with dedicated or common channel be envisaged. A possible adaptation of the procedure would be to plan some time for the UEs (all or group) to get activated and configure their RLC, MAC and L1, after which SRNC assumes that connection set-up is completed. In the latter, the lack of feedback from UEs remains, which prevents from reliable data transfer. Though a connection-oriented communication has some relevance to multicast delivery, SATIN baseline access should fully take advantage of the broadcast nature of satellite, hence be based on connectionless radio links and on adapted cell broadcast features of UTRAN.

Optional case

However, there may be similar procedures as in the T-UMTS case, when the UE is out of T-UMTS coverage and wants to register for the B/M service. In this case an RRC connection has to be requested by UE. Since in SATIN only common channels are available (dedicated point-to-point are envisaged in optional scenario, but this will not be SATIN main focus), RRC connection set-up is only possible with common transport channel (RACH/FACH).

F.3.3.2 RRC connection release

Baseline case

Since there is no return link via satellite, the T-UMTS procedure is not directly applicable and the following adaptation has been considered. SRNC sends the connection release message and configures RLC/MAC on its own side, without waiting for confirmation message from UEs. UEs can do the same thing on their side as soon as connection release message is received. However a RRC connection is not required in SATIN baseline (refer to clause F.3.3.1 "Baseline case").

Optional case

T-UMTS common channel case is applicable based on the common channel RRC connection set-up procedure.

F.3.4 Radio bearer configuration requirements

F.3.4.1 Baseline case

In the SATIN baseline case, transport channel configuration shall be based on unidirectional messages from Satellite-RNC to UEs (no possibility for standard completion message from UE to SRNC). This indicates that there will not be any confirmation from UE about the completion of RB set-up/modification/reconfiguration. There are two possibilities:

1) Radio Bearer control without dedicated, and more particularly considering common traffic RBs of the cell, which can be established/maintained/released by RRC based on CBS, i.e. make use of the configuration of the common channels of the cell, made available to UEs via SIB messaging on BCCH. Still, there will need some complement/adaptation to allow the configuration for a set of CTCHs/FACHs with improved transport format dynamics.

2) NW-RRC allows for some time the UEs to configure the RLC, MAC and L1, and acts just as if the procedure was completed. This choice would not be more reliable and does not suit broadcast mode.

F.3.4.2 Optional case

For the optional case, the common channel (RACH/FACH) procedures in 3GPP can directly be applicable.

F.3.5 SATIN transport channel configuration

F.3.5.1 Baseline case

In the SATIN baseline case, transport channel configuration shall be based on unidirectional messages from Satellite-RNC to UEs, as it is done in CBS. No UE dedicated transport/physical channel configuration has been assigned, the UE uses the common physical channel and transport channel configuration according to the system information, contained by BCCH on BCH.

F.3.5.2 Optional case

Address additional requirements (i.e. complementary to baseline) for Transport Channel Configuration when dedicated or shared channels are used.

F.3.6 Physical channel configuration

F.3.6.1 Baseline case

In the baseline case, the number of S-UMTS RRC states depends on the chosen architecture, i.e. parallel receiver architecture or reconfigurable receiver architecture. In the parallel receiver architecture there is only one RRC state, namely S_Cell_FACH, hence, nor case C neither case D procedures are applicable. In the reconfigurable receiver architecture, the three RRC states envisaged are S_URA_PCH, S_Cell_PCH and S_Cell_FACH, thus, again case C and case D are not applicable.

F.3.6.2 Optional case

In the optional case, the S-UMTS RRC includes four states whatever architecture will be applied. Hence, transport channel type switching may be requested, i.e. case C and case D are applicable.

Annex G:
ESA/3GNetSim

G.1 Introduction

3GNetSim will study topics regarding Uu aspects (forward and reverse link power control issues for unicast transmissions and an optimized FEC and interleaving method for multicast transmissions, as well as studying the possibility of cell breathing supporting satellite systems). Some Iu interface aspects will be studied like the compatibility of T/S_UMTS, as well as the issues regarding MBMS. Also call admission control and congestion control will be object of study during this project. The different clauses of this annex describe these issues in detail.

G.2 Uu aspects

G.2.1 Enhanced open-loop power control

Power Control for DRODCH transmission is based on Return Link Measurement Reports. In particular, based on the received measurement reports, a UE adjusts its power to be used for DRODCH transmission once an allocation is assigned to it. Using this approach, the transmit power to be used in order to achieve the target E_b/N_o is determined by the UE a-priori to the actual transmission. During a complete DRODCH burst, the transmission power remains constant. Since RL measurement reports are not always available, e.g. for the transmissions of the first packets within a stream until the first RL measurement report arrives at the UE and also in the case of RACH transmission, a combination with FL measurements is proposed for setting the RL transmission power. In particular, whenever no RL measurement reports are available at the UE, it performs SIR measurements on the continuously transmitted S-CCPCCH in order to estimate the current link quality. This is possible for S-UMTS since due to the reduced effect of multipath-fading in the satellite channel forward and return link can be assumed to be correlated. In the 3G-Network-Simulator project a smart procedure for combining RL measurement reports and FL SIR measurements on the S-CCPCH will be analysed and optimized.

G.2.2 Cell breathing

In T-UMTS, cell breathing refers to the dynamic variation of a cell size (i.e. the range of the geographical area covered by a cellular transmitter), based on the amount of traffic currently handled in the considered cell. In particular, when a cell becomes heavily loaded, its size is shrunk. Subscriber traffic is redirected to a neighbouring, more lightly loaded cell, by dynamically expanding the size of this cell.

The goal is to have in each cell approximately the same load and interference. In this way overall system capacity can be improved. The dynamic variation of cell sizes can smartly be attained by dynamic variations of the CPICH power: An increased CPICH power induces that more users are handed over to the cell, i.e. its coverage region increases.

In S-UMTS the same effect can be achieved by adjusting the CPICH power for each beam. However, particular challenges arise for the S-UMTS system, since the Tx power at the satellite, which has to be smartly distributed between the supported beams, is the capacity limiting factor for the forward link. Moreover, the cell sizes are much sharper, because of the relatively steep slopes of the antenna patterns, which define the coverage area of a beam. This means that CPICH induced dynamic cell size extensions may be quite costly.

Within the 3G-Network-Simulator project the performance of cell breathing and algorithms for dynamic power distribution between the cells will be analysed and optimized. The achievable capacity increase will be studied and assessed by confronting it with the required additional costs.

G.2.3 Downlink Shared CHannel (DSCH) and signalling concept

As also outlined in [10] from the physical layer perspective, the most important optimization to consider for the DSCH in S-UMTS is the possible decoupling of DSCH from the associated DCH (as foreseen in T-UMTS). Such a decoupling would avoid the overhead both in power and in code usage when maintaining the DCH. However, if there is no DCH available in parallel an alternative approach for both the power control as well as the signalling needs to be defined.

According also to the proposal in [10], the following concept for DSCH transmission is considered in the 3G-Network-Simulator:

- Open Loop Power Control, based on periodic Forward Link Measurement Reports.
- A Shared Control CHannel (SCCH) is used for the DSCH related signalling.

However, in contrast to the [10] proposal, a different format for the Signalling Channel SCCH is applied and investigated in the 3G-Network-Simulator project. In particular, the proposed signalling format allows to address up to N = 15 UE on the SCCH simultaneously, using an SCCH Signalling-Block for each addressed UE. Hence, a varying number of UEs (from 1 up to N) can finally be served on DSCH in parallel, each one on a separate DSCH sub-channel. These sub-channels are dynamically assigned and selected relative to the root channelization code allocated to the DSCH pipe, hence allowing to support different rates for the served UEs.

Within the 3G-Network-Simulator project, the applicability of this proposed signalling scheme and the open loop power control mechanism for DSCH is analysed and optimized. Furthermore, scheduling schemes for efficiently allocating the DSCH resources can be studied.

G.2.4 Combined FEC/interleaving

Due to the long round-trip delays, especially in the considered GEO satellite scenario, error correction based on an ARQ scheme can induce rather high delays and wasted bandwidth. This is even more critical in the case of Multicast Services, for which a large number of participating users can be expected in the satellite case. In such a scenario, it is very likely that if an RLC SDU is multicast to several users, each user only receives parts of the RLC SDU. In RLC Acknowledged Mode, each multicast user may finally ask for the retransmission of a different part of the SDU. If pure ARQ is used, the sender will likely have to retransmit the entire RLC SDU again. As a result, the achievable throughput for such multicast applications may be rather low. AM based solely on an ARQ scheme may therefore proof to be quite inefficient.

To improve the performance of multicast transmission, the usage of a combined ARQ scheme with Forward Error Correction and Interleaving at the level of RLC PDUs has been proposed and designed within the 3G-Network-Simulator. However, expecting a rather large number of Multicast Users due to the wide coverage area, a scheme solely based on Forward Error Correction and Interleaving for MBMS is considered to be more applicable. However, the proposed FEC/Interleaving scheme applicable to both cases (RLC with ARQ and without ARQ).

The proposed FEC/Interleaving concept actually combines Forward Error Correction with Interleaving. The interleaving is not directly applied to the actual data as it is typically done e.g. using a Block Interleaver. Instead, user data is combined for encoding such that interleaving is introduced implicitly. This approach greatly reduces the interleaving delay typically involved. A more detailed description is presented in the subsequent paragraphs. Overall, the combined FEC/Interleaving scheme investigated in the 3G-Network-Simulator project is characterized as follows:

- Greatly reduces interleaving delay compared to regular Block Interleaving.
- Allows to optimize deinterleaving delay at the receiving side.
- Highly scalable to support different Interleaving Block sizes, as well as code-rates.
- In-band signalling for dynamic adaptation of Interleaving parameters.

Description

FEC is done *across* several PDUs. In particular, k symbols - each taken from one of k user PDUs along a given position - are encoded by adding (n-k) redundancy symbols using a systematic code (see note). Each of these redundancy symbols is then placed into one of (n-k) newly created redundancy PDUs. Figure 128 shows one FEC block, consisting of k user PDUs and (n-k) redundancy PDUs.

NOTE: In the 3G-Network-Simulator project, Reed-Solomon encoding is applied for the combined FEC/Interleaving scheme.

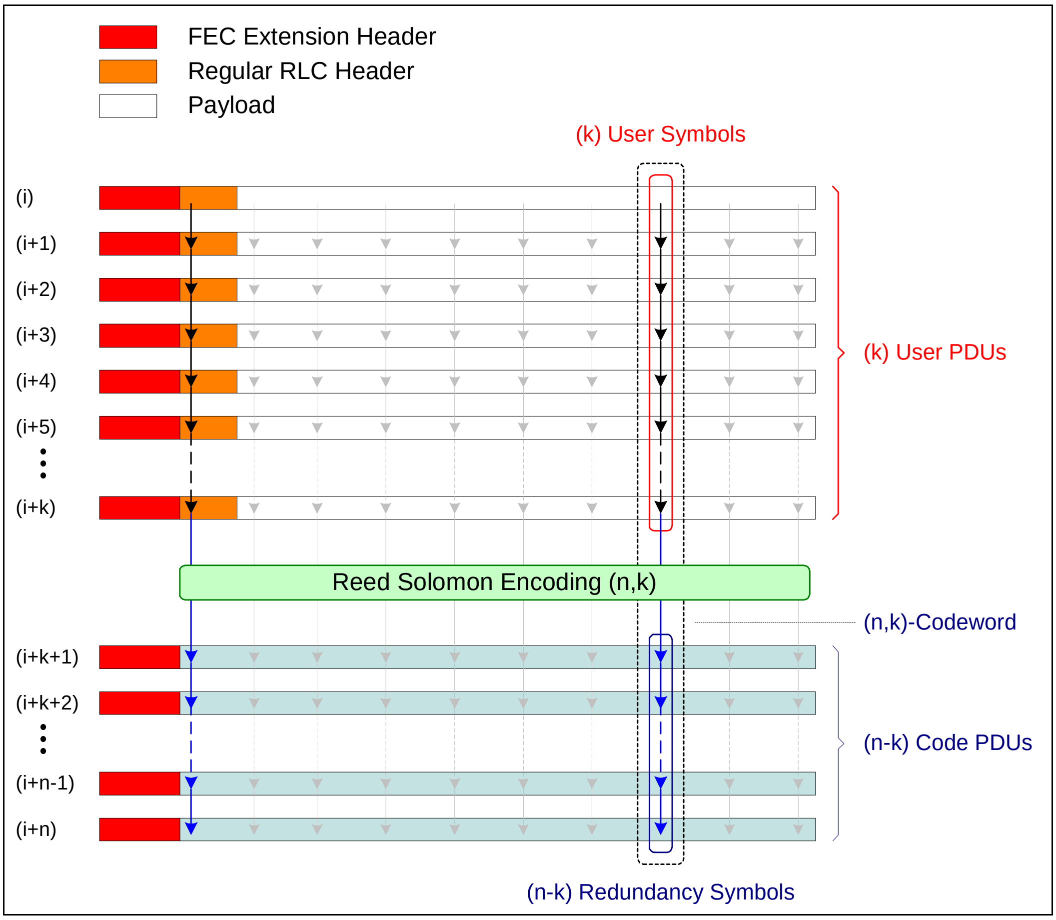


Figure 128: Combined FEC/Interleaving Scheme

The horizontal bars, numbered from (i) up to (i + n), represent the different PDUs of such a FEC block. As indicated by the dotted box, each code-word is formed vertically across different user PDUs, placing the code symbols across the additionally created redundancy PDUs. Most importantly, there is no need to buffer any of the user PDUs. In a typical interleaving scheme, no data can be transmitted until up to k PDUs have been received and stored into an interleaving buffer. Contrary to this, the proposed FEC/Interleaving scheme allows to transmit user PDUs directly without introducing any additional interleaving delay. It is sufficient to keep a temporary copy of the user data. Once all k user PDUs are transmitted, the redundancy PDUs are created at once based on the copies and transmitted to the receiving side. Only the transmission of these redundancy PDUs introduces additional delay, depending on the parameters n and k. However, for a typical configuration of n and k, this delay is much smaller than the interleaving delay faced using block interleaving.

Looking at the scheme illustrated in Figure 128, it can be easily seen that e.g. for Reed-Solomon encoding, up to (n-k) arbitrary PDU can get completely lost or up to (n-k)/2 PDU from a FEC block can be corrupted without causing the decoder to fail. Using the approach as depicted in Figure 128, an FEC code-word spans across several PDUs. Similarly, data loss due to a missing PDU is spread over several code-words, making it possible for the receiver to reconstruct the missing or corrupted data.

The parameters n and k are chosen such that the receiving side can reconstruct all PDUs that were lost during a interval (e.g. shadowing) of a desired length. In case a single FEC block is not large enough to robustly span over such an interval (depending on the maximum size n of a code-word, the PDU size as well as the transmission speed) the scheme can simply be cascaded.

For a regular block-interleaving scheme, de-interleaving introduces a similar delay as experienced during interleaving. In the proposed scheme, however, a delay reduction can also be achieved at the receiving side. As long as PDUs are received without errors and in sequence (utilizing e.g. CRC and RLC UM sequence-numbering), they can be forwarded to upper layers immediately. Similarly to the transmitting side, a temporary copy is kept for the recovery of possible subsequent errors within the same FEC block. Once a missing PDU is detected, any subsequent PDU cannot be forwarded as long as the missing data is not reconstructed. This either requires the reception of the complete FEC block, or at least the successful reception of k PDUs out of the block.

- As illustrated in Figure 128, an RLC FEC Extension Header is foreseen. Such a header allows for in-band signalling of FEC/Interleaving parameters to support varying size FEC Blocks. This may be required in case of bursty input traffic in order to keep a more continuous transmission.

Within the 3G-Network-Simulator project, the performance of the proposed FEC/Interleaving scheme for MBMS is analysed. Special attention is given to optimizing the FEC and Interleaving parameters and identifying their dependency on system and service parameters.

G.3 Iu aspects

The 3GNetSim simulator will implement T-UMTS signalling procedures for a specific set of satellite scenarios: GEO bent-pipe direct and indirect access scenarios (explain this) with an RNC dedicated to satellite segment. The target scenario of the simulator includes GEO satellites with multiple spot beams.

During the design of 3GNetSim the analysis two topics of Iu-interface and CN issues were further investigated.

- compatibility issue T-/S-UMTS at Iu Interface;
- MBMS issues.

G.3.1 Compatibility issue T-/S-UMTS at Iu interface

In [16] the applicability of the 3GPP UMTS system Release 99 standards for S-UMTS was analysed. During the design phase of the simulator the analysis in [16] has been confirmed so far regarding the influence of an S-UMTS system in the given satellite scenario to a standardized T-UMTS Iu interface. No additional requirements have been identified for RANAP and NAS signalling procedures.

Even though the basic signalling procedures are applicable, the parameterization is expected to be different due to the transmission characteristics of the air interface. Two technical specifications (TS 123 107 [17] and TS 124 008 [18]) give quantitative requirements based on the assumed transmission characteristics expected for a T-UMTS air interface. These TS are explained in the following clauses.

G.3.1.1 TS 123 107 - Quality of Service (QoS) concept and architecture

In [17] the framework for Quality of Service within T-UMTS is defined. The main purpose of the standard is to specify the list of attributes applicable to UMTS Bearer Service and Radio Access Bearer Service, as well as describe the Quality of Service architecture to be used in UMTS networks.

An S-UMTS system will not be able to fulfil all Radio Access Bearer Service attributes as specified for T-UMTS. The table below lists the attributes per UMTS Traffic class.

Table 47: Radio access bearer service attributes

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate (kbps)	tbd	tbd	tbd	tbd
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size (octets)	≤1 500 or 1 502	≤1 500 or 1 502	≤1 500 or 1 502	≤1 500 or 1 502
SDU format information	tbd	tbd		
Delivery of erroneous SDUs	Yes/No/-	Yes/No/-	Yes/No/-	Yes/No/-
Residual BER	tbd	tbd	tbd	tbd
SDU error ratio	tbd	tbd	tbd	tbd
Transfer delay (ms)	tbd	tbd		
Guaranteed bit rate (kbps)	tbd	tbd		
Traffic handling priority			1,2,3	
Allocation/Retention priority	1,2,3	1,2,3	1,2,3	1,2,3
Source statistic descriptor	Speech/unknown	Speech/unknown		

Using the 3GNetSim, suitable parameter ranges and values (marked *tbd* in the table above*)* for Radio Access Bearer attributes shall be determined.

G.3.1.2 TS 124 008 - Core network protocols, stage 3

In [18] the procedures used at the radio interface (Reference Point Um) for Call Control (CC), Mobility Management (MM), and Session Management (SM) are specified.

Due to the additional delay caused by the satellite air interface, default values for timers used for the various UMTS NAS signalling procedures (mobility management such as GPRS Attach, session management procedures such as PDP Context Activation) and RANAP signalling procedures need validation with respect to S-UMTS. Using 3GNetSim S-UMTS specific default values for the NAS signalling procedures can be validated. RANAP timers are considered as application specific by 3GPP. The results obtained by 3GNetSim will be documented to guide future implementations and requirement specifications for different satellite scenarios. In particular, the timers listed in the following three tables are subject to further investigation.

Table 48: GPRS mobility management timers

Timer Number	Cause of start	Entity
T3310	ACTIVATE REQ sent	MS side
T3350	DETACH REQ sent	MS side
T3330	ROUTING AREA UPDATE REQUEST sent	MS side
Tservice	SERVICE REQ (Signalling or Data) sent	MS side
T3313	Paging procedure initiated	Network side
T3322	DETACH ACCEPT received	Network side
T3350	ATTACH COMPLETE received RAU COMPLETE received	Network side

Table 49: GPRS session management timers

Timer Number	Cause of start	Entity
T3380	ACTIVATE PDP CONTEXT sent	MS side
T3381	MODIFY PDP CONTEXT sent	MS side
T3390	DEACTIVATE PDP CONTEXT sent	MS side
T3385e	REQUEST PPD CONTEXT ACTIVATION sent	Network side
T3386	MODIFY PDP CONTEXT REQUEST sent	Network side
T3395	DEACTIVATE PPD CONTEXT REQUEST sent	Network side

Table 50: RANAP timers

Timer Number	Cause of start
$T_{RABAssgt}$	RAB Assignment Request sent
T_{ignOC}	Reception of overload message
$T_{RELOCalloc}$	CN sent a Relocation Request
$T_{RELOCprepe}$	SRNC sent Relocation Required
$T_{RELOCcomplete}$	CN sent a Relocation Command
$T_{RELOCoverall}$	SRNC received Relocation Command

G.3.2 MBMS

MBMS (multimedia broadcast and multicast system) is standardized by 3GPP for Release 6. During the design of 3GNetSim current standardization drafts and suggestions (see [15], [44] and [45]) were analysed. Based on the state of the T-UMTS standardization Iu and CN procedures to develop an MBMS service for S-UMTS were defined. For any not yet standardized functionality appropriate procedures were specified.

G.3.2.1 Summary of Iu and CN MBMS procedures

Users wishing to join or leave a multicast group do so by an MBMS Multicast Service Activation/Deactivation. The activation procedure is based on IP Multicast protocols sent over a unicast PDP context. The architecture ensures that data paths are only established to RNCs that have multicast users located in them. The RNCs are notified of start and end points of data transfer by means of a signalling procedure (MBMS Session Start/Stop). A common MBMS Bearer Setup/Release procedure is used to set up RABs for broadcast and for multicast services.

MBMS attributes and parameters are stored in two contexts.

- The MBMS UE Context contains UE specific information related to a particular MBMS bearer the UE has joined.
- The MBMS Bearer Context contains all information describing a particular bearer of an MBMS service.

In case of a broadcast service the service is activated in the network by a manual configuration of the SGSN. For all UE that want to receive a broadcast service no signalling is required between the UE and SGSN. Therefore, in case of a broadcast service no MBMS UE Context is created. The service is initiated at the UE by user selection of an MBMS broadcast channel identifier.

G.3.2.2 Expected deviations to T-UMTS MBMS

- Due to the larger propagation delay compared to T-UMTS extra signalling to notify UE of imminent data transfer should be avoided. Standardized T-UMTS procedures are not yet available.
- The large propagation delay in S-UMTS makes the use of several ptp-bearers that are established instead of a ptm-bearer in case of ARQ to save resources impracticable. As a consequence it is not necessary at the RNC to keep track of the exact number of users in one cell. It is sufficient to know that there is at least one subscribed user in a cell. Any deviations to signalling procedures will be most likely at the Uu interface and not at the Iu interface or in the CN.
- Due to the large cell size and a higher expected number of subscribed users any approach discussed in 3GPP to detect active users in a cell by a procedure that wakes up a substantial number of users or keeping users in PMM-CONNECTED mode should be disregarded. Again this will most likely not change Iu and CN procedures but only information elements that are exchanged between the RNC and the SGSN.

As in the case of unicast timers for signalling procedures (session management and MBSM specific RANAP procedures) will be different due to the additional propagation delay.

G.4 User / traffic / application aspects

In addition to the capabilities for detailed simulation of protocols and Uu and Iu specific procedures at the level of individual users and applications, the 3GNetSim also supports simulations involving large amounts of traffic generated by sizable user populations distributed over extended geographical areas (such as the European region). These simulations operate at the level of traffic connections and traffic bursts, whereby lower-layer procedures and protocol actions are taken into account only in an abstract way. The main characteristics and issues addressed in these type of simulations are briefly highlighted in the following.

Call Admission Control

The main characteristics of the CAC scheme applied are:

- A load based CAC mechanism is employed.
- Pre-emption of already established, low priority connections is foreseen in order to free up resources to accommodate a new higher-priority call request.
- Forced handover to a neighbouring cell is foreseen to accommodate a new higher-priority call request.
- Bit rate reduction for low-priority calls is foreseen to accommodate a new higher-priority call request.

A number of issues and options/possibilities exist with respect to the concrete application of these mechanisms, in particular:

- Strategy to be applied for call pre-emption, e.g.: how to select calls for pre-emption, max. number of calls to be pre-empted, etc.
- Strategy to be applied for forced handover, e.g.: how to select calls for forced handover, max. number of calls to be handed over, etc.
- Strategy to be applied for bitrate reduction, e.g.: how to determine the amount of bitrate reduction, maximum allowed reduction, etc.

These issues will be studied in the simulations with the objective to optimize overall network performance and throughput while respecting application QoS and overall user satisfaction.

Congestion Control variants/parameters

The Congestion Control Scheme employed is load-based, and operates in close analogy to the CAC scheme. In particular, pre-emption, forced handover and bitrate reduction for low-priority connections are foreseen. In addition to the issues and options/possibilities mentioned for CAC, the following optimization issues are identified for congestion control:

- Criterion for congestion onset; length of measurement period for the load.
- Criterion for congestion termination.

For these issues, the same optimization criteria as for CAC apply.

QoS differentiation strategies

Each application must be assigned a priority level with respect to pre-emption, forced handover and bitrate reduction. Different priority classifications may result in different overall network performance, and this shall be studied in simulation runs.

Network dimensioning and configuration issues

The simulator will allow the user to evaluate overall network performance as a function of different application, user and mobility scenarios for a given network configuration. It is a key objective of this project to assess the overall network performance for different configurations and dimensioning of network resources such as network entities and links.

Annex H: Packet data transmission in the GAUSS System

H.1 The GAUSS system

The key element of the GAUSS defined system is the NAV and COM integration and the synergy achievable from the two technologies, based on criteria of providing the specified services with maximum efficiency.

Main design concepts are at the basis of the GAUSS solution:

- A common flexible format message for all the applications, based on 424-bit cells which are able to carry the required information:
    - Single-cell messages.
    - Multiple-cell messages (up to 8 concatenated cells) are supported.

GAUSS packet is the bit stream including the GAUSS cell plus the overhead for EDC and preamble for burst transmission.

- The open architecture based on current standard interfaces and protocols.
- The compliance to S-UMTS standards, suitably adapted and optimized to support the envisaged low bit-rate data services envisaged.
- New developments starting from the usage of consolidated and existing technologies.
- Tight harmonization between the Terrestrial 3G Mobile System and the Satellite components for supporting the integrated NAV/COM Services, with the perspective of deploying a single highly-integrated network, based on:
    - Common Core Network, but distinct Radio Access Networks.
    - And a future Single User Terminal (dual mode) for Satellite and Terrestrial access.

The radio access of the GAUSS system towards the external networks is conceived as a distinct RAN of the S-UMTS family, optimized (as here after detailed in clauses H.3 and H.4) for low-bit rate packet-based services. The standard interface towards the Core Network ensures compatibility and inter-operability with 3GPP systems. This is not against the common vision of the UMTS to be a system mainly aiming at the high bit-rate mobile service market. GAUSS has not to be intended as a competitor with other systems (UMTS itself), but a system capable of fulfilling effectively and efficiently the requirements of inter-modality and info-mobility applications compatible with the GAUSS concepts (small size messages, low bit rate). The background guideline of the designed architecture is that of pursuing an open standard for the communication services to be integrated with the navigation ones, rather that a proprietary standard.

In this framework, it has to be noted that the GAUSS architecture is compliant with the current assessed release 4 of the UMTS, and some optimizations are applied for taking into account the envisaged low-rate data transmission. More specifically:

- The radio access scheme designed to use as much of the work done for S-UMTS as possible, and appropriately adapted.
- The radio protocol layer designed to be compliant to the standard, and optimized to support the defined services.

H.2 The GAUSS data packet

At the application, GAUSS messages are formatted in data cells (the GAUSS cell) characterized by (see Figure 129):

- A common structure for all the applications.
- A standard 53-byte fixed-length format, 5 bytes used for the header and the remaining 48 bytes for information field. Contents of the packet data fields are depending on the specific applications (e.g. the PVT data auxiliary optional data).

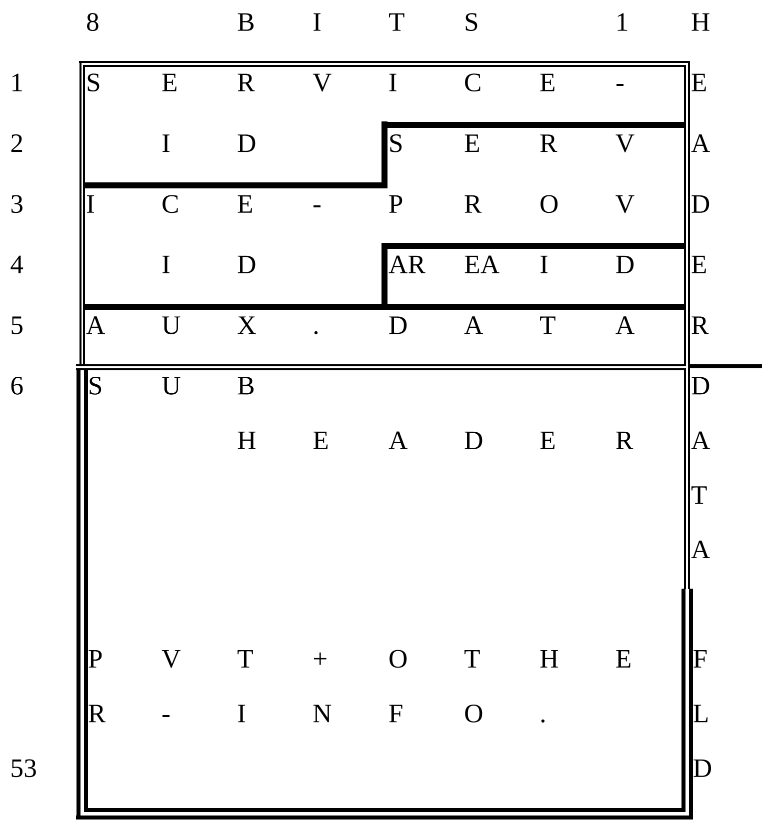


Figure 129: GAUSS cell format

Messages generally comply with the size of a single 424-bit GAUSS cell. Longer messages can be transmitted by fragmenting them into more cells (up to 8). The application has the task of reconstructing the originally sent messages, along with the check for completeness and the eventual request for selective repetition.

For this standard GAUSS cell, two different message formats has been defined:

- for broad-catching and point-to-point services (communications from MU to SP) (Figure 130);
- for broad-casting and point-to-point services (communications from SP to MU) (Figure 131).

Basic concepts that drive the above design are as it follows:

- criteria of defining a common message cell structure for all supported services;
- fulfilling the GAUSS requirements, based on small data packet communication;
- being compliant with the requirement of routing through a Terrestrial Network with minimum adaptation;
- standardizing to the maximum feasible extent the services for what is concerning the physical layer, leaving the applications the task of managing all specific requirements.

Criteria of flexibility

The GAUSS cell, while providing the specific GAUSS types of services, is definitively capable to support a wider class of Location-based services, requiring small data packet exchange at low bit rate transmission. As a matter of fact, no modification is required to the GAUSS standard packet (including the integrated NAV/COM messages used for all the applications), to accommodate in the messages the data required by other specific application.

Broad-catching and Point-to-point (MU→SP) Services Message				
HEADER	**bit**	**Subtotal bit**	**byte**	**Remarks**
Service ID	12			
Service Provider ID	16			
Service Provider Area ID	4	32	4	
Message Type	2			0 = broad-catching, 1 = 1 cell message point-to-point, 2 = multiple cell message (up to 8 concatenated cells) point-to-point
Spare	6	8	1	
Header Total		**40**	**5**	
APPLICATION DATA FIELD				
Sub Header				
User ID	21			Blank for user sensitive information
Retry message ID	3	24	3	Decrement numbering of attempts from 7 up to 0 or until an ACK is received. When ACK is received is set to 7 for the next message
Acknowledgement request	1			0 = no acknowledgement request, 1 = acknowledgement request
Cell numbering	3			Decrement numbering for up to 219 byte messages, segmented into eight cells. The last cell takes the number 0. Only the last cell contains the navigation data
Spare	4	8	1	
Message Sequence ID	8	8	1	Decrement Number from 7 to 0 (up to 8 concatenated cells). Messages requiring concatenated cells are identified by the same Msg Seq. ID
Sub Header Total		**40**	**5**	
Data Field				
PVT Data	96	96	12	IF Cell Numbering = 0 THEN PVT data, ELSE available for data field
Data	248	248	31	
Data Field Total		**344**	**43**	
CELL TOTAL		**424**	**53**	

Figure 130: GAUSS cell for MU→SP messages

Broadcasting and Point-to-point (SP➔MU) Service Message				
HEADER	**bit**	**Subtotal bit**	**byte**	**Remarks**
Service ID	12			
Service Provider ID	16			
Service Provider Area ID	4	32	4	
User group ID	2			0 = broadcast, 1 = individual message (point-to-point), 2 = group addressing (i.e. multi-cast)
User Channel ID for Up link	6	8	1	Channel ID numbering
Header Total		**40**	**5**	
APPLICATION DATA FIELD				
Sub Header				
User ID	21			▪ All blank for broadcast ▪ For User group ID = 1, ID of addressed Mobile User ▪ For User group ID = 2, ID of addressed group ID
Spare	3	24	3	
Acknowledgement request	1			0 = no acknowledgement request, 1 = acknowledgement request
Cell numbering	3			Decrement numbering for up to 232 byte messages, segmented into eight cells. The last cell takes number 0.
Spare	4	8	1	
Message Sequence ID	8	8	1	Decrement Number from 7 to 0 (up to 8 concatenated cells). Messages requiring concatenated cells are identified by the same Msg Seq. ID
Sub Header Total		**40**	**5**	
Data field				
Virtual Spot ID+ Cell Counter These fields (total 12 bytes) are available for data filed in case of point-to-point service, as no spot has to be specified				
# of spot	2			0 = global coverage
Centre of spot	41			geographical co-ordinates in arc seconds (accuracy to 10 m to 30 m)
± delta lat.	10			co-ordinate variation in arc minutes (range 10°, accuracy to 2 km)
± delta lon.	10			
Spare	1	64	8	
Cell counter	32	32	4	IF CM = 0 THEN decrement cell counter
Sub Total		**96**	**12**	
Selective addressing ID				
Data	64	64	8	IF Group ID > 1 AND User ID > 0 THEN available for user terminal synchronization algorithm, ELSE available for information data
Field for messaging	184	184	23	
Data Field Total		**248**	**31**	
CELL TOTAL		**424**	**53**	

Figure 131: GAUSS cell for SP→MU messages

H.3 GAUSS access and control subsystem

The access scheme proposed for GAUSS is compliant with the current assessed release 4 of the UMTS), appropriately adapted for effectively supporting the envisaged services, considering their nature:

- Highly-bursty low-rate communications.
- Transmission of structured short data packets.

The adaptations mainly affect:

- The radio access scheme designed to use as much of the work done for S-UMTS as possible.

The radio protocol layer designed to be compliant to the standard. The Upper Layer in the GAUSS Demonstrator has the same structure as the one described in 3GPP. On top of PDCP the IP protocol is used (for the user plane) while UDP, which is a transport layer connectionless protocol (layer 4 in the OSI model) is used for the communication with the Application Layer, which takes place through the 53 byte cells. These GAUSS cells are always encapsulated in TBs of fixed size, which are sent to the physical layer. Next figure presents the Channel mapping in the Return and Forward Links, where the new channels are shadowed.

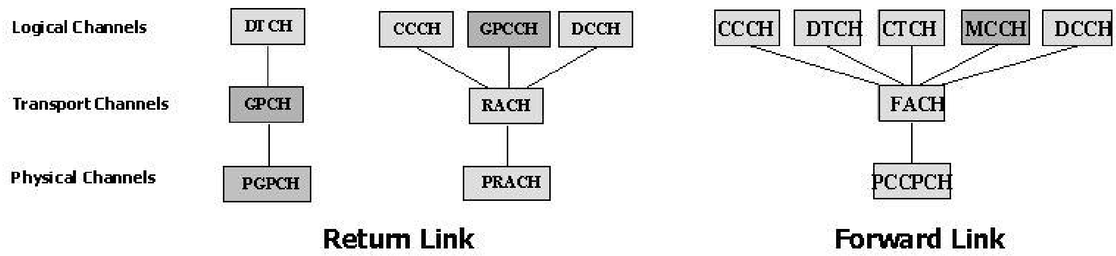


Figure 132: Channel mapping in the GAUSS Demonstrator Return and Forward Links

The main characteristics are the following:

- UMTS Release 4 compliant Access and Control System, with some compatible variants.
- A new physical channel in the RL: PGPCH (Physical GAUSS Packet CHannel) with the corresponding transport channel GPCH (GAUSS Packet CHannel), used for the RL data transfer (DTCH corresponding logical channel).
- Two new logical channels: GPCCH and MCCH.
- GPCCH (GAUSS Packet Control CHannel) used for asking resource allocation to the scheduler in the GW before transferring data.
- MCCH (MultiCasting CHannel) point-to-multipoint channel used for the transfer of user information in multicasting services.

The new physical channel in the return link, the PGPCH and the corresponding new transport channel the GPCH, are used for the return link data transfer, that was optimized for burstly low bit rate data transfer. GPCCH (GAUSS Packet Control CHannel) is the logical channel used for asking resource allocation to the scheduler in the return link data transfer. The return link data communication can be summed up in three main steps:

- A message is sent on the RACH channel in order to request resources. The message payload includes the number of bytes that is requested to be sent.
- The scheduler in the GW allocates the necessary resources and communicates back with an acknowledgement the time when data transmission must initiate.
- Data is transmitted through the PGPCH channel.

H.4 Study on RLC configuration for GAUSS system

As far as the Radio Link Control (RLC) procedures are concerned, the protocol specification used for GAUSS is compliant to that defined by 3GPP and applied in S-UMTS.

The upper layer protocols and the features of the RLC protocol are the same as specified by the standards. Furthermore, for the radio protocol, due to satellite communication, a high value of the round trip time is experimented and then feed-back delays are larger than T-UMTS. This possibly leads an acknowledged mode transmission to suffer from high delays in the reception of the acknowledgements. This problem is more critical in case of low rate data flow. Thus, an optimum RLC configuration is required in order to satisfy the timing requirements needed by GAUSS services. As the return link is characterized by a very low bit rate, this investigation had been carried out by means of simulations.

Some RLC configurations have been studied and compared in order to observe the effects of the high RTT on the transmission efficiency.

H.4.1 RLC services and functions

The RLC protocol provides mainly segmentation and retransmission services for both user and control data. The services provided by RLC to higher layers are:

- Transparent mode data transfer (TrM): no protocol overhead is added to RLC SDUs, only segmentation is performed.
- Unacknowledged mode data transfer (UM): no retransmission mechanism is used and therefore data delivery is not guaranteed.
- Acknowledge mode data transfer (AM): a retransmission mechanism by means of acknowledgements is used for error correction.
- Maintenance of QoS as defined by upper layers.
- Notification of unrecoverable errors.

The RLC functions implemented in the GAUSS System are:

- segmentation and re-assembly;
- concatenation;
- padding;
- transfer of user data;
- error correction;
- in-sequence delivery of higher layer PDUs;
- duplicate Detection;
- flow control;
- sequence number check;
- protocol error detection and recovery;
- ciphering;
- polling;
- status transmission;
- sDU discard;
- suspend/resume function;
- stop/continue function;
- re-establishment function.

H.4.2 RLC study in GAUSS scenario

As previously described the RLC protocol is the same as in T-UMTS. What has to be taken into account is the GAUSS scenario, which is characterized by typical satellite delays, low bit rate data flow and a small amount of data packets that lead to a simplified RLC configuration.

These study aims at finding an efficient RLC configuration for AM data transfer in the GAUSS scenario described above which is quite different from the T-UMTS one in which high bit rate, low round trip time and large amounts of user data are experimented. The preliminary results that will be presented are based on simulations of a peer to peer RLC transmission when a retransmission mechanism for error correction is used.

The simulations have been carried out for the Return Link only because it is the most critical one in terms of time of data transmission. The following assumptions are made:

- RTT = 500 ms;
- Max number of RLC PDUs = 8;
- Acknowledged mode transmission;
- User data PDUs on Return Link;
- Status PDUs (ack) on Forward Link;
- TTI (RL) = 1 440 ms;
- TTI (FL) = 80 ms.

The TTI of the RL considered in the simulations is actually different. The proposed access scheme for the RL defines a GAUSS super frame format divided into a data interval and a reservation interval which is subdivided into two random access intervals of equal length. The reservation procedure is based on a contention among access requests of different users, so there can be collisions and therefore these access requests cannot be satisfied. The data interval can carry one RLC PDU. The time period of the super frame is 2 seconds, therefore the actual TTI for user data PDUs is 2 s instead of 1 440 ms. Moreover in this study the contention mechanism is not considered and blocking probability of zero is assumed:

- TTI = 2 sec;
- Blocking Probability = 0.

H.4.3 A suitable RLC configuration

Two configurations have been considered, both TX driven, which means that the transmitter triggers the receiver to report on which PDUs have been correctly received and which need to be retransmitted by means of a Status PDU:

- Each time the transmitter sends a PDU a Poll bit is set, requesting a Status PDU from the receiver.
- The Poll bit is set every 8 PDUs.

The results are shown in Figure 134. Since no significant differences can be noticed if only 8 PDUs are transmitted, the simulation have been carried out with 100 PDUs in Figure 135.

For a given delivery time the curves show the probability that the PDUs are correctly received within that time. As it can be seen the second configuration gives better performance. This is due to the fact that the round trip time is very high, so it can happen that Status PDUs are not up-to-date and trigger the retransmission of already retransmitted PDUs, lowing the protocol efficiency. It is clear why the second configuration gives better performances: if the status rate is lower, the number of useless retransmissions is reduced.

H.4.4 A mechanism to avoid useless re-transmission

In order to avoid these useless retransmissions a simple mechanism has been studied. For better understanding the principle on which this mechanism is based on, in the next figure an example of useless retransmission is given:

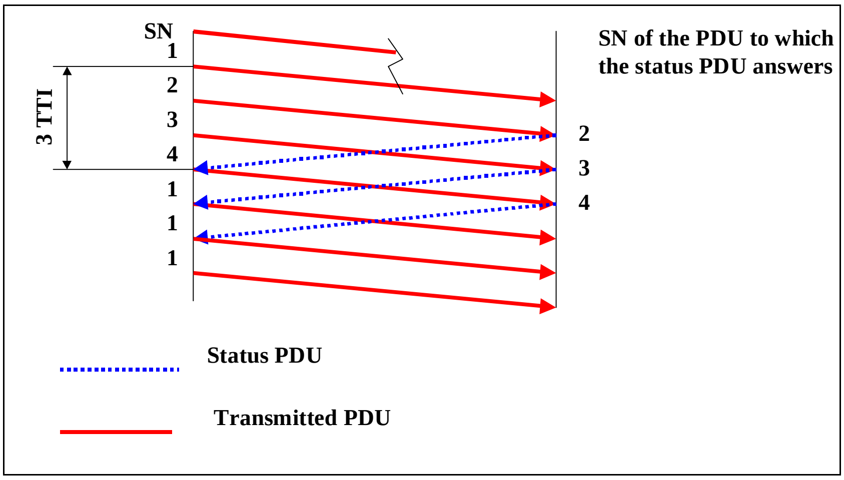


Figure 133: Mechanism for avoiding useless re-transmission

The PDU with Sequence Number = 1 (SN) is not correctly received and therefore the receiver does not answer with the corresponding Status PDU.

In the meantime PDUs, with rising SN, are transmitted before the detection of the first status PDU triggers the retransmission of the PDUs not correctly detected at the receiver side.

Between the transmission of the PDU with SN = 1 and its first retransmission, three PDUs have been correctly detected at the receiver side, and the receiver has answered with the same number of status PDU, each of them signals that the PDU with SN = 1 has not been correctly received yet.

Due to this, all these status PDUs will trigger the same retransmission, and obviously only the first one is useful.

Hence in this example the high RTT causes two useless retransmission of the same PDU, lowing the protocol efficiency.

These two useless retransmissions could be simply avoided by forcing the transmitter to block any retransmission of the first PDU for 3 TTI after the first one.

In order to achieve this, two new figures should be standardized:

- A Timer, started at the beginning of each transmission and incremented every TTI.
- A state variable, one for each PDU, in which saving the value of the Timer introduced, at the moment in which each PDU is transmitted.

The transmitter, in order to avoid useless retransmission should block any triggered retransmission if the difference between the values of the new figures is less than 3 TTI.

It is now interesting the comparison between the second RLC configuration described above, in which a Poll bit is sent every 8 PDUs, and the first one, with the mechanism to avoid the useless retransmission implemented.

As it can be seen in Figure 136, the RLC configuration with the Useless Retransmission Avoidance implemented seems to give better performances, especially for BLER = 10 % where we have more retransmissions, since we have a higher probability to have a lower delivery time.

If this improvement is already visible when only 8 PDU are transmitted, it becomes more evident with 100 PDUs (see Figure 137).

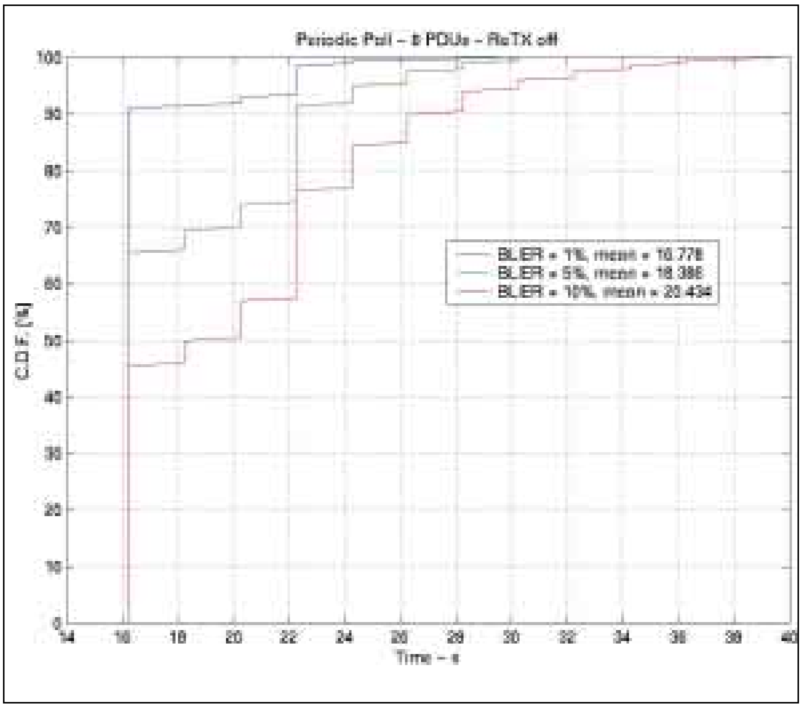
Defining the delivery time, as the time within all the PDUs are correctly delivered with a probability of th 95 % an efficiency gain of the RLC configuration chosen can be defined:

$$EffGain = \frac{TD_{95\%} - TD_{ref}}{TD_{ref}} \cdot 100,$$

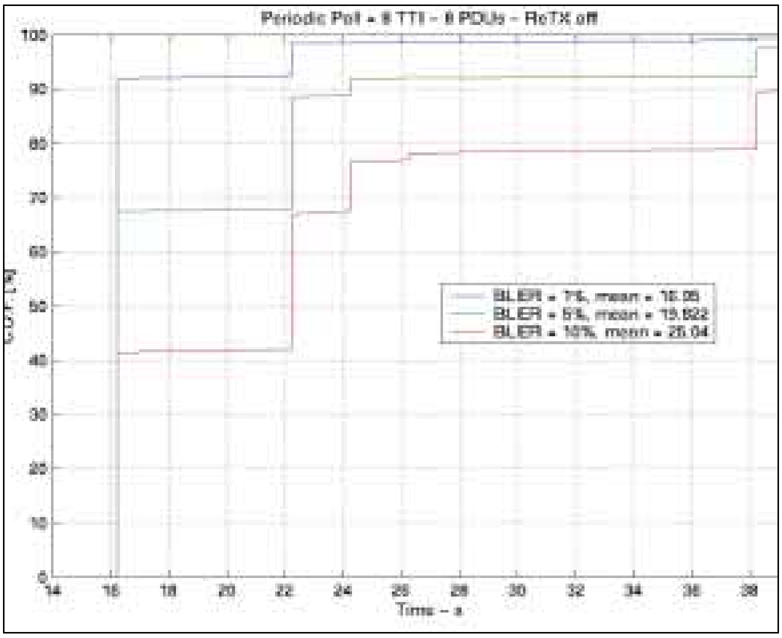
where Td_{ref} is the delivery time of the RLC configuration chosen as term of comparison (i.e. Poll every PDU without ReTx Avoidance). In Figure 138 the comparison of the Efficiency gains of the RLC configurations investigated so far is shown.

Three RLC configurations had been studied:

- Poll every 1 PDU, that can be considered as the basic RLC configuration, which is quite inefficient because of the useless retransmission.
- Poll every 8 PDUs, which leads to an efficiency gain of the 8 %, as reduces the number of useless retransmissions, and does not need any RLC modifications.
- Poll every 1 PDU and the retransmission avoidance mechanism applied, which leads to a better efficiency gain (12 %) but two new figures must be introduced, as stated before.



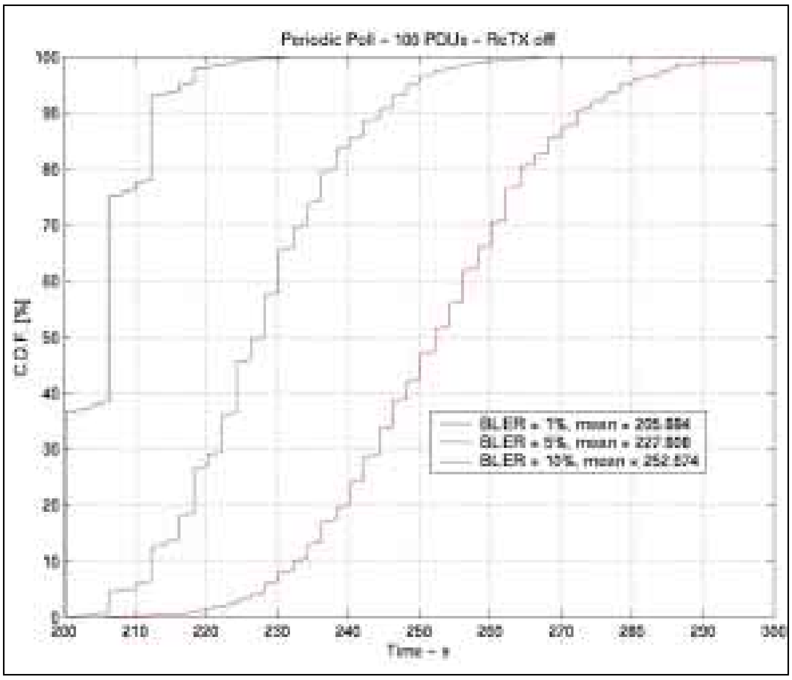
a) Poll every 1PDU



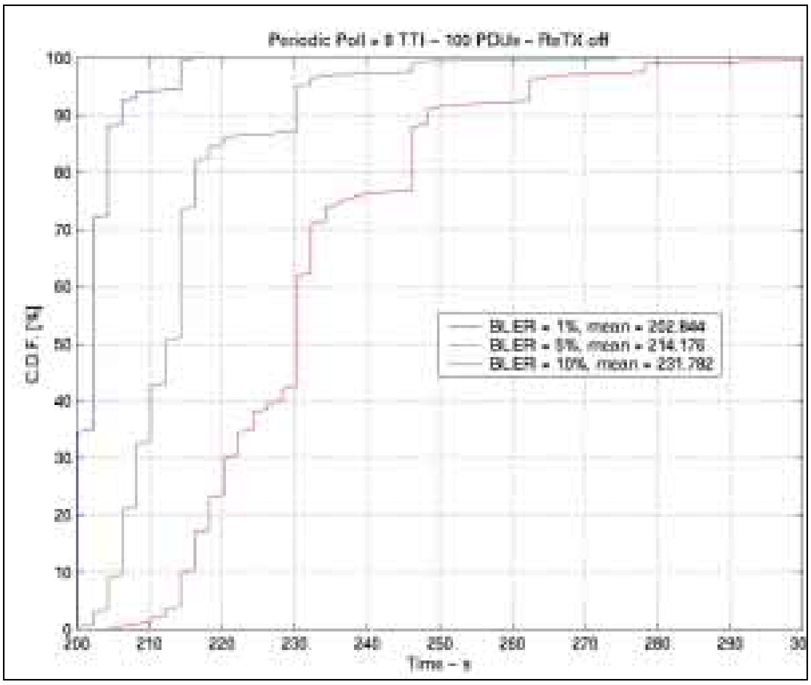
b) Poll every 8 PDUs

NOTE: Should you encounter problems to read this figure, please be informed that it is contained in archive tr_102061v010101p0.zip which accompanies the present document.

Figure 134: GAUSS RLC simulation results (8 PDUs transmitted)



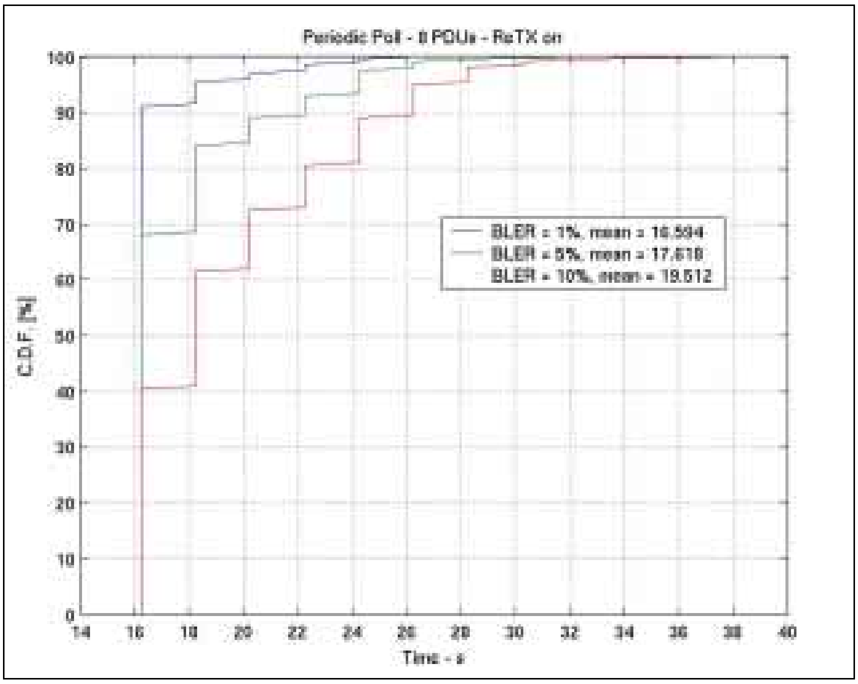
a) Poll every 1PDU



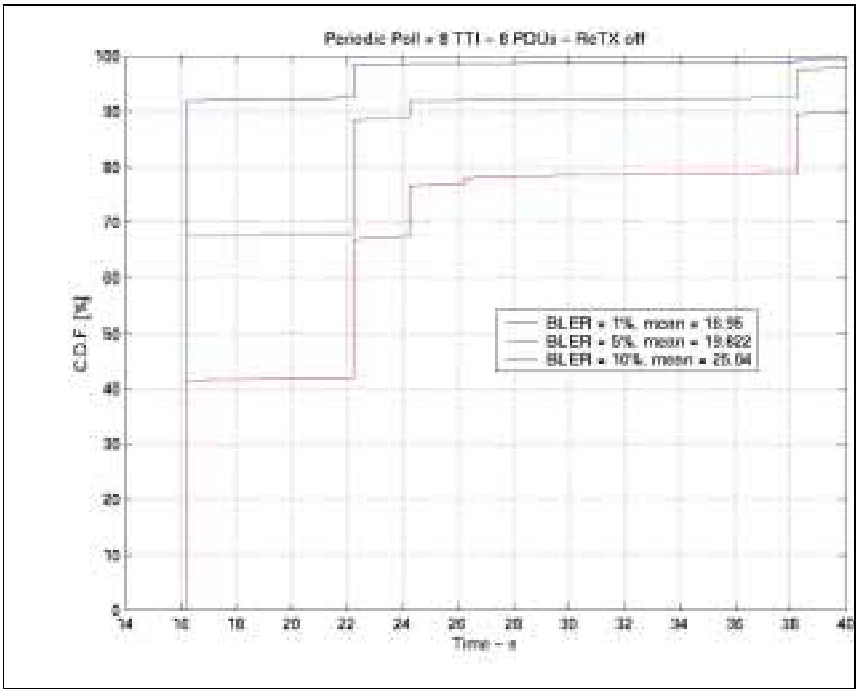
b) Poll every 8 PDUs

NOTE: Should you encounter problems to read this figure, please be informed that it is contained in archive tr_102061v010101p0.zip which accompanies the present document.

Figure 135: GAUSS RLC simulation results (100 PDUs transmitted)



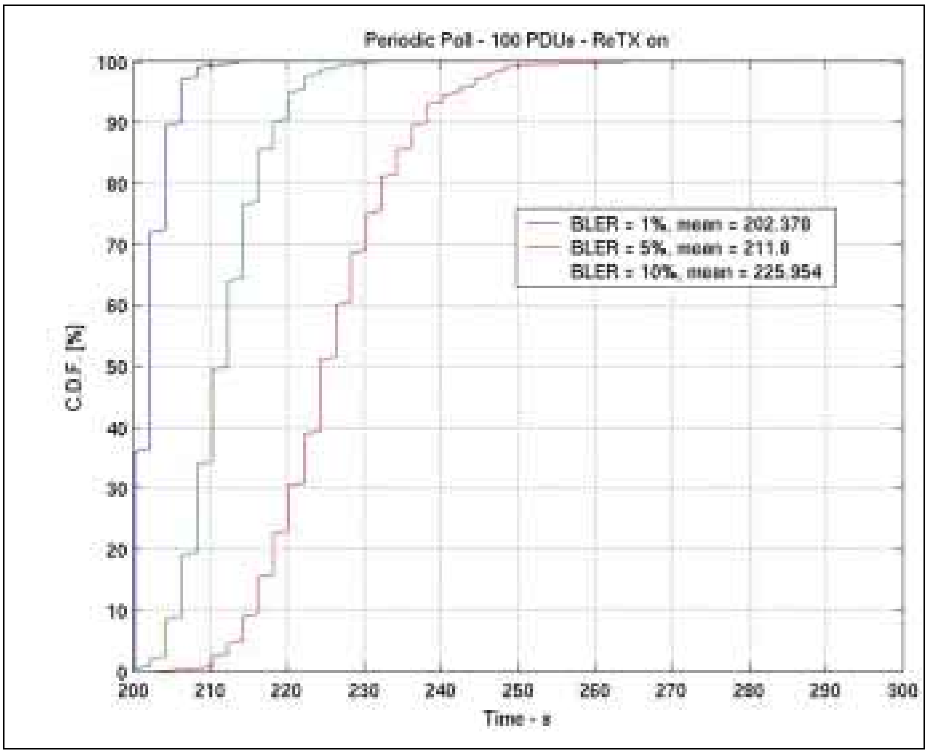
a) Poll every 1 PDU, ReTx avoidance



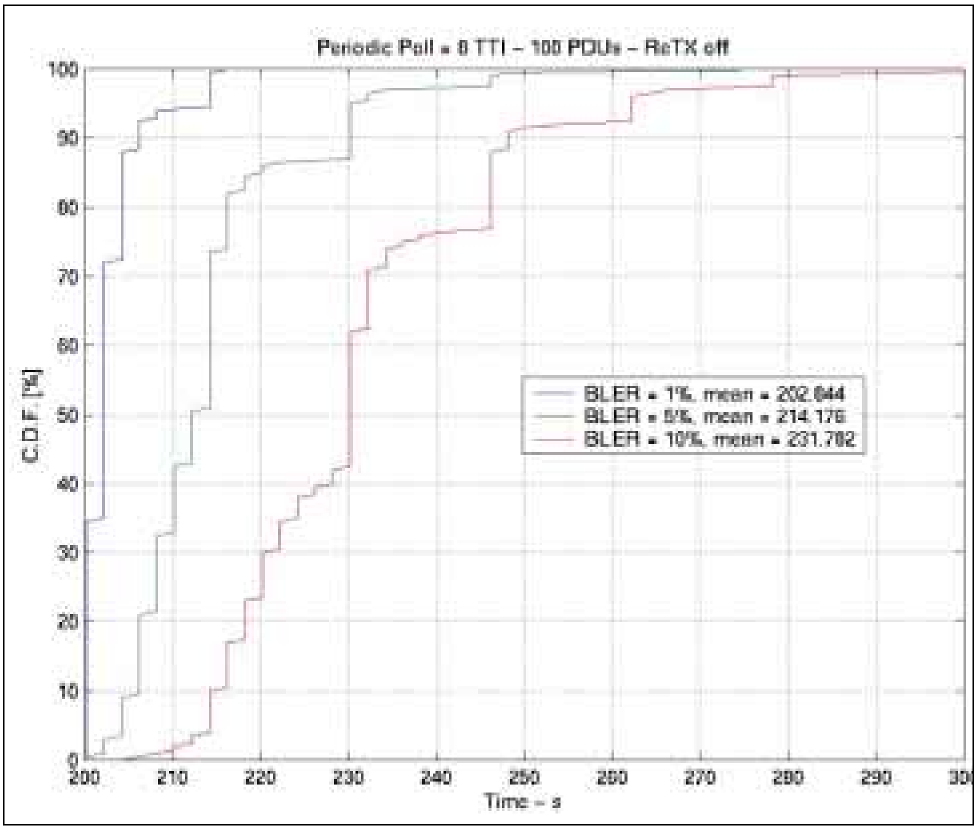
b) Poll every 8 PDUs

NOTE: Should you encounter problems to read this figure, please be informed that it is contained in archive tr_102061v010101p0.zip which accompanies the present document.

Figure 136: GAUSS RLC simulation results (8 PDUs transmitted)



a) Poll every 1 PDU, ReTx avoidance



b) Poll every 8 PDUs

NOTE: Should you encounter problems to read this figure, please be informed that it is contained in archive tr_102061v010101p0.zip which accompanies the present document.

Figure 137: GAUSS RLC simulation results (100 PDUs transmitted)

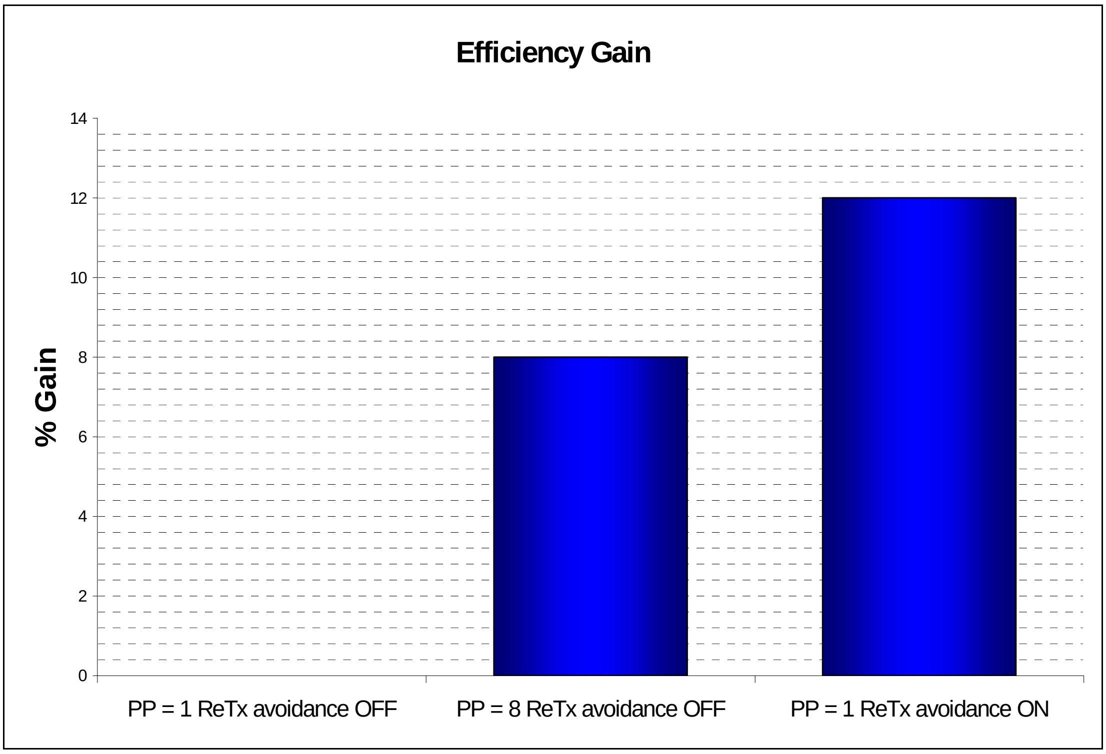


Figure 138: Efficiency gains

H.5 GAUSS forward link physical layer

According to the S-UMTS specifications, the physical layer of the forward link in GAUSS has been based on the Primary Common Control Physical Channel (P-CCPCH) used as a 2nd Secondary Common Control Physical Channel (S-CCPCH) in order to map on it the Forward Access Channel (FACH) transport channel.

Each TB 600 bits long, coming from the Medium Access Control (MAC) layer was formatted into four frames introducing an adaptation overhead which transforms the original TB into an encoded TB (ETB) of 736 bits. A Transmit Time Interval (TTI) of 1 280 msec was adopted obtaining 320 msec as physical frame duration. The resulting information bit rate is then ETB/TTI = 575 bits/sec. After applying a 1/3 code rate and inserting the Pilot Symbol (PS) and Frame Sync Word (FSW) a channel bit rate of 3,75 Kbps is achieved corresponding to a QPSK Symbol Rate (SR) of 1,875 Ksps. The UMTS standard specifies a 256 Spreading Factor (SF) for the P-CCPCH so the chip rate required to transmit the P-CCPCH is SR × SF = 480 Kcps. This chip rate was obtained, also according to the bandwidth available at the Gateway Station (GW), scaling the standard UMTS chip rate (3,84 Mcps) by 8.

The acquisition processing at the user terminal (UT) side is based on a matched filter exploiting the property of the Gold code imposed to the transmitted string. The first 256 bits of this code are correlated and submitted to a threshold process in order to obtain a synchronization whose precision is 1/2 chip. At the same time the acquisition also performs a coarse carrier frequency recovery within an error of ±1,875 KHz. The resulting mean acquisition time, estimated on many acquisition trials carried out at laboratory level, is about 300 msec.

The steady state at the end of the acquisition phase is controlled and assured by three loops working in parallel: The Delay Locked Loop (DDL), the Automatic Frequency Control (AFC) loop and the Automatic Gain Control (AGC) loop.

The DLL concerns the timing recovery and it is based on an early-late algorithm modifying the impulse response of the Square Root Risen Cosine (SRRC) input filter according to the delay found by the error detector. The AFC loop allows a carrier frequency tracking within a range of ±100 Hz when stimulated by a frequency step. Finally the AGC loop is used to force the signal amplitude to that necessary for the next de-spreading and de-scrambling algorithms.

The performances achieved for the P-CCPCH of the forward link are reported in Figure 139.

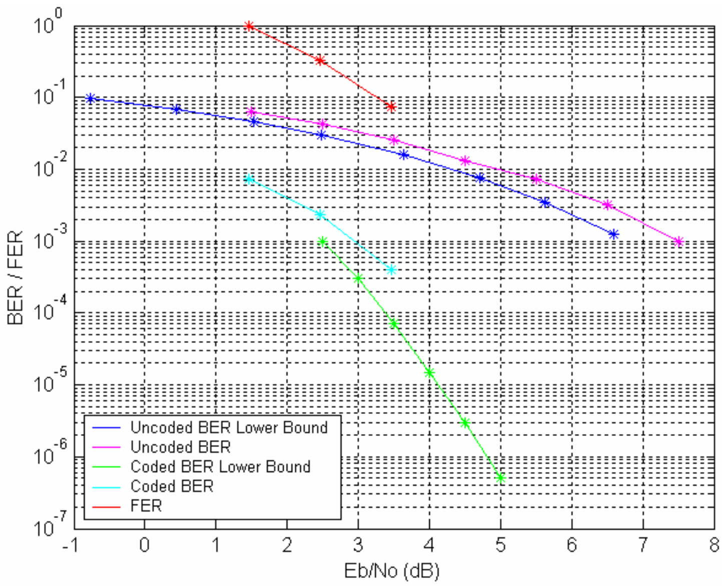


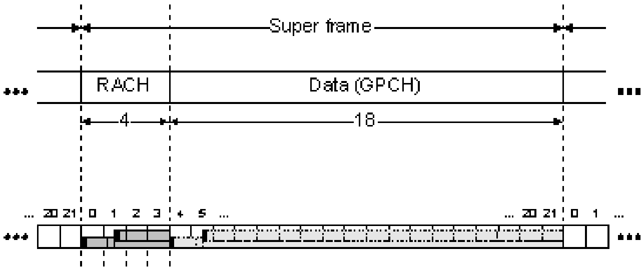
Figure 139: Performances of the forward link P-CCPCH

Both the uncoded and coded BER curves are shown. An implementation loss less than 1 dB was obtained (as it is possible verify looking at the theoretical lower bounds reported in the picture) including the fixed point approximation of the digital processing and the losses due to the analog processing (ADC, DAC and filters).

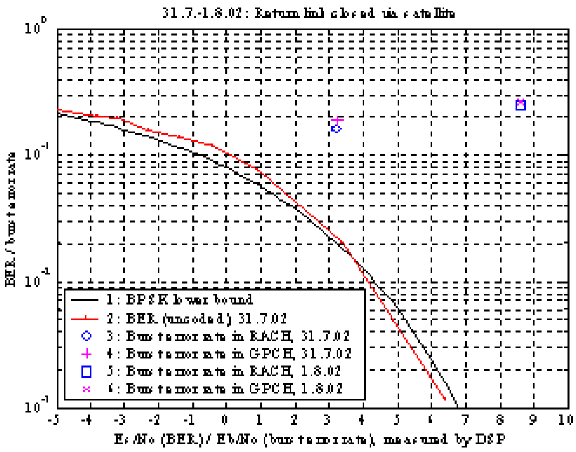
H.6 GAUSS return link physical layer

The design of Return Link of the GAUSS Demonstrator was specifically tailored to the needs of the GAUSS location based services, i.e. the return link had to support highly bursty, low rate data services. Therefore delay was a minor issue in the return link. Nevertheless all components in the link were designed to be close to real-time operation. Since for simplicity reasons the components from different partners of the project were connected with serial interfaces and non-real-time operating systems were used, a careful design of the communication via these serial interfaces and of the host software on the physical layer hardware was mandatory.

In the GAUSS Demonstrator two transport channels were provided: a random access channel (RACH), which is specified in the S-UMTS standardization documents and an additional GAUSS packet channel (GPCH) providing a non-continuous packet transmission. The time scale in the return link is divided in consecutive super frames as depicted below. During RACH periods the user terminals are allowed to issue requests for transmission capacity based on a slotted ALOHA scheme. The scheduler in the gateway station then allocates GPCH slots according to the request and transmits this information via the forward link to the user terminal. Now the user terminal can transmit its data on a contention-free basis. Since all mobile terminals have timing and positioning information from the GAUSS navigation subsystem, it is possible to synchronize the return link signals in such a way that they are (quasi) synchronous at the satellite receiver. Since the burst duration in comparison with the round trip delay is very low, closed loop power control inherently is not meaningful for the GAUSS services.



GAUSS RL Superframe



BER and packet loss rate with link is closed via satellite

NOTE: Should you encounter problems to read this figure, please be informed that it is contained in archive tr_102061v010101p0.zip which accompanies the present document.

Figure 140: GAUSS RL main characteristics

In the RL Super Frame, the data part is reserved by the GW as a response to a random access message. A GAUSS super frame is divided into a data interval of length T_D and a reservation interval of length T_{res}, which is subdivided into N_R random access intervals of equal length T_R. Whenever a MT wants to transmit a GAUSS message it randomly selects a random access interval and a spreading code out of a code set with N_C codes. Only a unique MT identifier, the number of bits to transmit and some QoS flags are spread with the selected code and transmitted in the selected random access interval. The scheduler in the GW allocates return link capacity, i.e. it decides in which time interval and which spreading code the MT has to transmit its data. The described random access scheme can be seen as $N_R * N_C$ parallel slotted ALOHA channels. This corresponds to a sparsely loaded slotted ALOHA system with low bandwidth efficiency. But if T_{res} can be chosen much smaller than T_D the inefficient used random access interval does not effect the bandwidth efficiency of the whole transmission.

The mapping of transport blocks to radio frames and the air interface is designed to be compliant to the S/T-UMTS standards. An additional physical channel is introduced on which the GW receiver searches for transmitted signal in a wide frequency range. This Synchronization Emulation CHannel (SECH) is invisible to the upper layers and emulates the tight coupling of the forward and return link in real systems where the UT is roughly synchronized to the GW. The GAUSS Demonstrator aimed at very low rate data services and operates on a chip clock, which is eight times lower than in S/T-UMTS. Therefore the symbol duration in GAUSS was very long and frequency offsets had to be compensated very carefully. This required very accurate frequency estimation and tracking algorithms in the GW receiver. In order to emulate additional users in the return link, a traffic generator was included at the GW, which was synchronized to the access scheme in the return link.

History

Document history		
V1.1.1	May 2004	Publication