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Departamento de Electrónica, Telecomunicações e Informática

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VoIP Service Performance Evaluation over 3GPP networks

Avaliação de Desempenho do funcionamento de serviços VoIP sobre redes 3GPP

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Dissertação apresentada à Universidade de Aveiro para cumprimento dos requisitos necessários à obtenção do grau de Mestre em Engenharia Electrónica e de Telecomunicações (Mestrado Integrado), realizada sob a orientação científica da Professora Dra. Susana Sargento, Professora auxiliar do Departamento de Electrónica, Telecomunicações e Informática da Universidade de Aveiro e do Mestre Miguel Almeida, Engenheiro da Nokia Siemens Networks.

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palavras-chave

UMTS, HSPA, HSDPA, HSUPA, VoIP, Performance, Monitoring, OSS, Reporting, KPI

resumo

A gestão de conteúdos orientados ao utilizador tem-se vindo a revelar uma questão de extrema importância para os operadores, que embora não sejam os produtores e distribuidores da informação acedida, são no entanto parte interessada pois em última análise é a sua insignia que deve assegurar o acesso. Os modelos de negócio desenvolvidos actualmente antevêm a distribuição destes conteúdos assegurando o cumprimento dos parâmetros de QoS. Com a evolução da distribuição de serviços sobre as redes IP, seguindo a tendência da perspectiva “All-over-IP”, os ISPs necessitam cada vez mais de ter conhecimento acerca da forma como estes serviços e os seus utilizadores influenciam a utilização dos recursos da rede.

A monitorização de desempenho requer estratégias eficientes e optimizadas com múltiplas implicações ao nível da segurança/privacidade. Cada serviço possui características específicas que o podem tornar mais ou menos resistente a determinadas condições da rede. O objectivo deste trabalho é relacionar a informação relativa à sessão de um determinado tipo de serviço baseado em IP, com as condições de desempenho na entrega do serviço por parte da rede. O desafio é analisar diferentes tipos de informação, por um lado a informação de sessão foca-se nos eventos gerados durante o seu ciclo de vida, enquanto a informação de Performance Management (PM) da rede foca-se primordialmente no comportamento e capacidade da rede em suportar a entrega do serviço, a um grande número de assinantes, relevando portanto a utilização das métricas de QoS.

A proposta deste trabalho é definir uma série de ferramentas como relatórios e indicadores de desempenho, em que baseado na informação cross-layer, se possa descrever uniformemente o desempenho do serviço.

keywords

UMTS, HSPA, HSDPA, HSUPA, VoIP, Performance, Monitoring, OSS, Reporting, KPI

abstract

The management of user oriented contents is becoming of extreme relevance for network operators, which while not being the producers of the consumed data, are the ultimate insignia for the assured delivery. The business models being currently applied envision the assured delivery of multimedia services with the assurance of Quality of Service. By evolving towards the delivery of services over IP networks undergoing the “all-over-IP” perspective, the Internet Service Providers (ISP) needs to be aware of how the behavior of these services and users influences the network resources usage. Performance monitoring requires efficient and optimized strategies with multiple implications at the security/privacy levels. Each service has specific characteristics which may make it more or less resilient to some network performance issues. The scope of this work is to relate session information with the underlying network service delivery performance. The challenge is to analyze different kind of information, session information focus is event driven tracing the entire life-cycle of each event and network Performance Management (PM) information focusing on the behavior and ability of the network to support service delivery to a large number of subscribers, thus focusing on overall QoS metrics. The proposal is to define use cases that can be implemented to ease this analysis while defining general Key Performance Indicators (KPI) based on cross-layer information, to uniformly describe the service performance.

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Acronyms

3GPP 3rd Generation Partnership Project
ALCAP Access link control application part
AM Acknowledged mode
APN Access point name
ARQ Automatic repeat request
BER Bit error rate
BLER Block error rate
CAPEX Capital Expenditure
CCCH Common control channel (logical channel)
CCH Common transport channel
CCH Control channel
CDMA Code division multiple access
C-NBAP Common NBAP
CORBA Common Object Request Broker Architecture
CPICH Common pilot channel
CRNC Controlling RNC
CS Circuit Switched
CTCH Common traffic channel
DCA Dynamic channel allocation
DCCH Dedicated control channel (logical channel)
DCH Dedicated channel (transport channel)
DL Downlink
DPCCH Dedicated physical control channel
DPDCH Dedicated physical data channel
DPI Deep Packet Inspection
DRNC Drift RNC
DSCH Downlink shared channel
E2E End-to-End
E-AGCH E-DCH absolute grant channel
E-DCH Enhanced uplink DCH
E-DPCCH Enhanced dedicated physical control channel
E-DPDCH Enhanced dedicated physical data channel
EM Element Manager
EPC Evolved Packet Core
E-RGCH E-DCH relative grant channel
E-UTRAN Evolved UTRAN

FACH Forward access channel
FDD Frequency division duplex
FDMA Frequency division multiple access
FER Frame error ratio
FM Frequency modulation
FP Frame protocol
FTP File Transfer Protocol
GGSN - Gateway GPRS Support Node
GMM GPRS Mobility Management
GPRS General Packet Radio Services
GTP GPRS Tunneling Protocol
GTP-C GPRS Tunneling Protocol – Control Plane
GTP-U GPRS Tunneling Protocol – User Plane
HARQ Hybrid automatic repeat request
HLR Home Location Register
HSDPA High Speed Downlink Packet Access
HS-DPCCH Uplink High-Speed Dedicated Physical Control Channel
HS-DSCH High-Speed Downlink Shared Channel
HSPA High Speed Packet Access
HSS Home subscriber server
HS-SCCH High-Speed Shared Control Channel
HSUPA High Speed Uplink Packet Access
HTTP Hypertext transfer protocol
ID Identity
IETF Internet engineering task force
IMS IP multimedia subsystem
IMSI International mobile subscriber identity
IP Internet Protocol
ITU International telecommunications union
KPI Key Performance Indicator
L2 Layer 2
LTE Long-Term Evolution
MAC Medium access control
MBMS Multimedia broadcast multicast service
MM Mobility management
MME Mobility management entity
MMS Multimedia message
MOS Mean opinion score

NAS Non Access Stratum
NBAP - Node B Application Part
NE Network Element
NRT Non-real time
OBS Operational Business Software
O&M Operation and maintenance
OFDMA Orthogonal frequency division multiple access
OPEX Operational Expenditure
OSS Operations support system
OVSF Orthogonal variable spreading factor
PBCH Physical Broadcast Channel
PC Power control
PCB Printed Circuit Board
PCCC Parallel concatenated convolutional coder
PCCCH Physical common control channel
PCCH Paging channel (logical channel)
PCCPCH Primary common control physical channel
PCH Paging channel (transport channel)
PCPCH Physical common packet channel
PCRF Policy and Charging Rules Function
PDCP Packet data converge protocol
PDN Public data network
PDP Packet data protocol
PDSCH Physical downlink shared channel
PDU Protocol data unit
PHY Physical layer
PLMN Public land mobile network
PM Performance Management
POC Push-to-talk over cellular
PRACH Physical random access channel
PS Packet switched
PSCH Physical shared channel
PSTN Public switched telephone network
QAM Quadrature amplitude modulation
QCIF Quarter common intermediate format
QoE Quality of Experience
QoS Quality of Service
QoS Quality of service

QPSK Quadrature phase shift keying
RAB Radio access bearer
RACH Random access channel
RAI Routing area identity
RAN Radio access network
RANAP - Radio Access Network Application Part
RB Radio bearer
RF Radio frequency
RL Radio Link
RLC Radio link control
RMC Reference measurement channel
RNC Radio network controller
RNS Radio network subsystem
RNSAP RNS application part
ROHC Robust header compression
RR Round robin
RRC Radio resource control
RRM Radio resource management
RSN Retransmission sequence number
RSSI Received signal strength indicator
RSVP Resource reservation protocol
RT Real time
RTCP Real-time transport control protocol
RTP Real-time protocol
RTSP Real-time streaming protocol
RU Resource unit
SAE System architecture evolution
SAP Service access point
SAP Session announcement protocol
SCCP Signaling Connection Control Part
SC-FDMA Single carrier frequency division multiple access
SCH Synchronization channel
SCTP Simple control transmission protocol
SDD Space division duplex
SDP Session description protocol
SDU Service data unit
SEQ Sequence
SF Spreading Factor

SFN System frame number
SGSN Serving GPRS support node
SHO Soft handover
SIB System information block
SIC Successive interference cancellation
SID Silence indicator
SINR Signal-to-noise ratio where noise includes both thermal noise and interference
SIP Session initiation protocol
SIR Signal-to-interference ratio
SLA Service Level Agreement
SM Session management
SMLC Serving mobile location centre
SMS Short message service
SN Sequence number
SNR Signal-to-noise ratio
THP Traffic handling priority
TMN Telecommunication Management Network
UE User Equipment
UL Uplink
WCDMA Wideband CDMA, Code division multiple access
XML Extended Markup Language

1. Introduction

It is a privilege to be part of the Telecommunication industry nowadays, with all the great developments in the mobile networks, making it possible to ensure the continuous growth in respect to market shares and transmitted traffic. Cellular networks, with improved broadband access, are being deployed worldwide, with its increasingly high bit rates and throughputs, which enable new service support capabilities while assuring improved quality of experience for the end user.

The 3GPP mobile networks and their evolution (WCDMA, HSPA and HSPA Evolution) are an important stepping stone for this great achievement, making possible for millions of people, to access a true and universal wireless broadband service. These networks characteristics and features allows an always-on connectivity state of mind, deployed everywhere which in turn enables the innovative potential of new services, applications and business models.

For the next cellular networks generation (4G), 3GPP has a new technological proposal for both radio access (3GPP Long Term Evolution - LTE) and core networks (Evolved Packet Core - EPC). This proposal promises to fulfil the demands for even higher bit rates, seemingly mobility, ubiquitous deployment and "organic" growth.

It becomes easily predictable that these networks will become highly complex to operate, maintain and optimize, representing a great Performance Management challenge. Typically these networks produce thousands of Gigabytes of self performance monitoring information, on a daily basis per operator, imposing the need to have a good and scalable Operation Support System (OSS), well organized and with extended KPI and Reporting support.

> *"Today's "next-generation service providers" are required to manage a much more complex set of products and services in a dynamic, competitive marketplace. As a result, these service providers need next-generation OSS solutions that take advantage of state-of-the-art information technology to address their enterprise-wide needs and requirements. Next-generation OSS help service providers maximize their return on investment (ROI) in one of their key assets—information. OSS ultimately helps enable next-generation service providers to reduce costs, provide superior customer service, and accelerate their time to market for new products and services."* [1]

1.1.Motivation

As a Performance Management Engineer, integrated in an OSS product development team, a great part of this Thesis work is related with the organization of the PM information (Measurements and Counters) and with the definition of specific PM content (Key Performance Indicators (KPIs) and Reporting Use Cases). Driven by the innovative spirit of my company, I started to analyze mechanisms and techniques that can be used to relate information between Network performance and Service behavior. My main motivation for this work is to propose a selection of KPIs and Reporting Use Cases that can be used by the operator to monitor the root cause of a particular IP-based service failure.

Current network performance tools content is still much focused in network performance only, providing much less information about services behavior and Quality of Service (QoS) performance. The PM tools content is updated for each system release, which occurs in average once per year, and the aim is to extend the performance monitoring capabilities to the newly introduced features. These features aim to expand network capabilities and increase network capacity and performance.

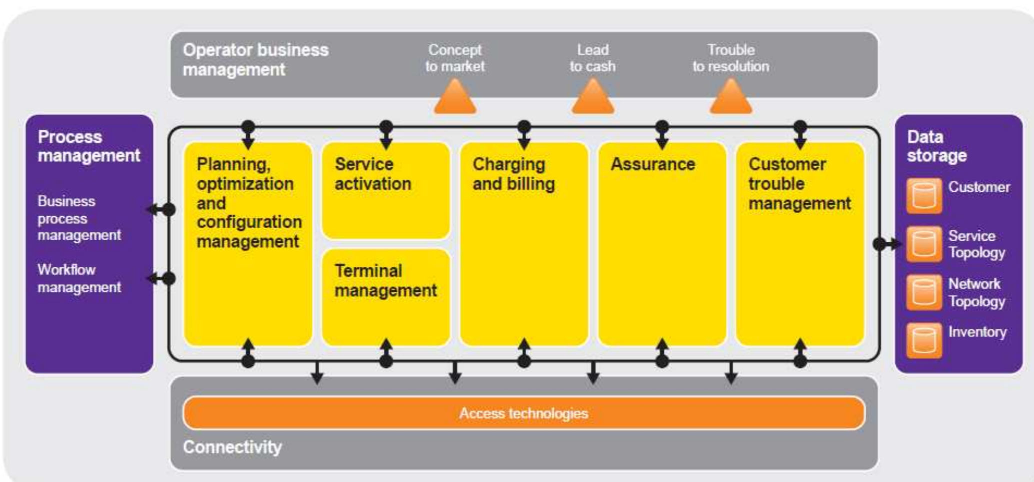


Figure 1 - OSS: Modular software architecture and process automation approach

While this approach has proven to be correct for a long time, currently we are assisting in the push to go beyond and start monitoring IP-based Services performance. Currently, there are no analysis mechanisms that enable the capability to check if a given network is optimized for IP-based Services delivery. Figure 1 shows a schematic approach to define a possible Next Generation OSS (NGOSS), providing means and functionalities to integrate both network and IP-based service information at the same place and same time. This allows addressing several use cases using the reporting mechanisms proposed by this work:

- Monitor Service Level Agreements fulfillment;
- Answer IP-based service subscriber complaints;
- Monitor and optimize operator own services and network;
- Forecast future service delivery requirements and network capacity expansion.

This work proposes a new approach in performance monitoring that relates the IP-based Service QoS performance with the network performance. This allows identifying which layer is contributing for the lack of performance and what are the corrective measures to take.

1.2. Objectives

IP Converged networks are becoming a reality, with more and more networks extending the IP delivery support. Cellular networks have been definitely contributing in a decisive way for this accomplishment, with the latest evolutions (HSDPA, HSUPA, HSPA+ and soon LTE) offering a great coverage of cellular broadband access. The effective support of IP connectivity through mobile broadband opens the opportunity to offer new bouquet of diverse and rich multimedia services, ranging from entertainment based services like Gaming and Mobile TV to some more business oriented services like Machine-2-Machine (M2M) communication and Web Conferencing.

This new reality brings new challenges to network operational, maintenance and optimization procedures, since the shift from Circuit Switched (CS) Voice optimized networks to all kind of IP-Services optimized networks, increases the number of optimization variables and thus the process complexity. The questions to be answered are the following: Do we have all the tools in place that can tell us how current network performance is influencing a particular IP session performance? Can we relate network and service statistics in a way that indicate to us if a particular network is suitable for that service delivery? Can we tell, prior to a new service deployment, if the network is suitable as it is? And if not, what should be the improvements to be done?

This work main objective is to answer these questions by proposing a reporting concept that relates information from two distinct but inter-dependent layers: Network Performance and IP-Based Service Performance. The intention is to select for each layer the most important KPIs and define analysis use cases that propose a workflow for performance monitoring and identify the correct measures to apply whenever needed. In the end, the results will show how these layers influence one another.

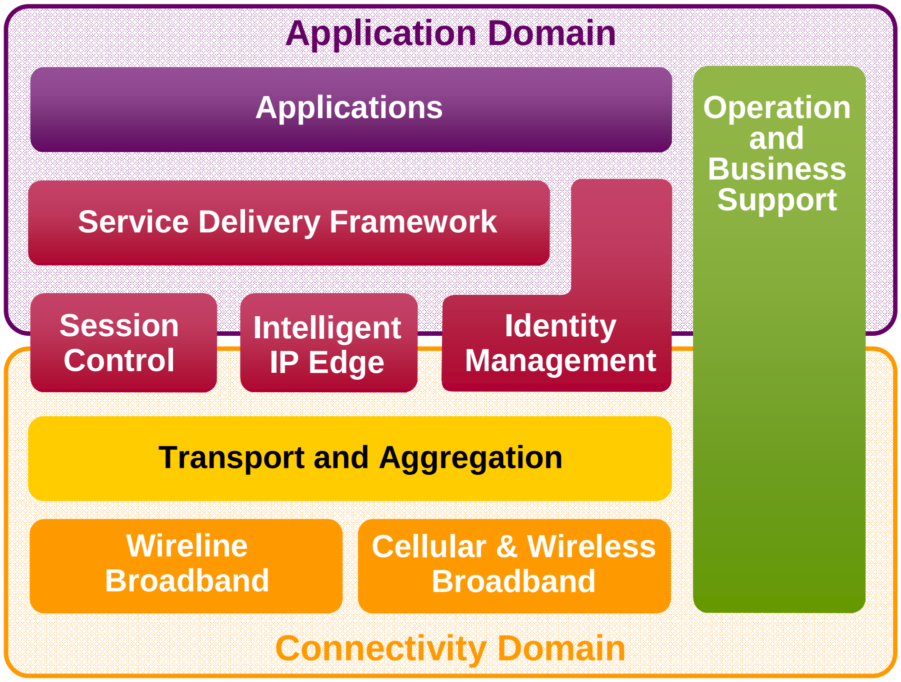


Figure 2 - IP-based logical architecture

Figure 2 describes a logical architecture for IP-based service delivery and, as it can be seen, the Operation and Business Support area is transversal to both Application and Connectivity domains. The reporting solution described in this work is included in this Operational Business Software (OBS) area and will focus on the 3GPP Cellular Broadband technologies details.

1.3. Contributions of the Thesis

The work developed accomplishes the major part of the proposed objectives and, as a result, its main contributions are:

- The proposal of a new PM Reporting feature that allows enabling IP-based QoS statistics monitoring related with Network Performance Monitoring. This feature will be integrated in Nokia Siemens Networks Reporting products commercial releases;

- The evaluation of real time IP-based Service (VoIP and IPTV) over HSPA, being valuable for the service key statistics identification;

- The work develop within this thesis, particularly the identification of the network performance monitoring architecture, layers and KPIs is part of a published article:

    - Miguel Almeida, Rui Inácio, Susana Sargento, "Cross Layer Design Approach for Performance Evaluation of Multimedia Contents", 2nd International Workshop on Cross Layer Design (IWCLD 2009), Mallorca (Spain), June 2009.

1.4. Organization of the Thesis

The present thesis is organized as follows:

- Chapter 2 provides an overview about this work related areas such as Network Management and 3GPP Networks. This work main goal is to provide a new concept of network and IP-based service performance monitoring to be applied to 3GPP operational support, and therefore, this chapter highlights all the relevant matters for the development of this study.
- Chapter 3 introduces the details related with this work concept proposal, describes the proposed architecture, identifies the performance data sources, introduces the data modeling processes, details the relevant Key Performance Indicators and presents the new reporting concept describing some relevant use cases.
- Chapter 4 discusses the simulation environment and simulated results for a VoIP service running on top of a 3GPP network combining UMTS R99 and HSPA elements.
- Chapter 5 presents the final conclusions taken from this work and identifies some possible future work.

2. Background

This chapter introduces the several concepts that are fundamental for the understanding of this work. Section 2.1 discusses the Network Management thematic detailing the Telecommunication Management Network concept, providing an insight about its objectives and structure, as well as briefly describes some of the most used performance monitoring techniques. Section 2.2 introduces the cellular packet switching networks, detailing its architecture, network elements, functionalities and procedures. Section 2.3 introduces in some detail the 3GPP UMTS High-Speed Packet Access technology, detailing the enhancements introduced when compared with UMTS R99 WCDMA technology. This section first provides a brief introduction to UMTS R99 data transmission functionalities, then detailing the HSDPA and HSUPA technological improvements, and the main characteristics that enable HSPA as a suitable technology for real-time IP-based service delivery. Section 2.4 provides a brief introduction to the 3GPP Long Term Evolution proposal that aims to be the main 4G technology. The last Section 2.5 summarizes the most important contents mentioned in this chapter.

2.1.Network Management

"An essential question when designing the network architecture is how to manage and monitor the network's overall operation, and remove flaws from the network when they occur. In general, network management can be perceived as a service that employs a variety of methods and tools, applications, and devices to enable the network operator to monitor and maintain the entire network. For a typically centralized mobile network, network management means the ability to control and monitor the entire network from a specific location and, possibly, remotely. The rapid universal growth of mobile networks has made the role of the network management a key feature that should be carefully taken into account early in the design process of network architecture. In particular, the basic functionality of networks should provide operators with the ability to control and maintain their networks and services." [2]

Several efforts, from different standardization organizations, have been developed to standardize a common network management model. The International Standard Organization (ISO) proposes a management framework [3] that proposes to control and monitor the use of network resources by defining a model which provides data storage and processing capabilities that can relate the performance of the different functional parts of the network. This model defines that the ultimate management decision is of Human beings responsibility, although the responsibilities may be delegated to the system automated processes. The OSI management functional areas are:

- Fault Management;

- Accounting Management;
- Configuration Management;
- Performance Management;
- Security Management.

The OSI Model defines functional mechanisms for each of these areas, being these mechanisms and related objects generic to more than one of these areas.
Important to mention is that the OSI model is the reference of some network management protocol such as Common Management Information Services/Common Management Information Protocol (CMIS/CMIP) [4] and Simple Network Management Protocol (SNMP) [5] [6].

International Telecommunication Union – Telecom Standardization (ITU-T) also defined a network management model, the Telecommunications Management Networks (TMN) [7]. This has become the dominant network management concept within the telecommunications world, serving as a reference for all the OSS system vendors.

2.1.1. TMN Logical Model

This work assumes that all the referred Operational Support Systems are based on the Telecommunications Management Networks (TMN) concept. The TMN concept was defined by the International Telecommunications Union – Telecommunications Service Sector (ITU-T) as an infrastructure to support management and deployment heterogeneous operating systems and telecommunication networks. TMN proposes a framework of software applications and procedures for operating and managing networks in a flexible, scalable, reliable, inexpensive to use and easy to enhance process. The following layers are defined:

Business Management Layer (BML) – high-level layer that focus on providing mechanisms and procedures that allow to: define business goals, do business planning, product planning, service planning, business agreements, etc. This layer is close related with financial metrics (like budgeting, revenue, OPEX, CAPEX), with legal arrangements and security issues.

Service Management Layer (SML) – this layer provides information that allows coordinating the interaction between services and service providers, interfaces with partner companies, customers and vendors. SML is responsible for collecting and analyze QoS metrics to insure that Service Level Agreement (SLA), between network operator and service provider, is being fulfilled.

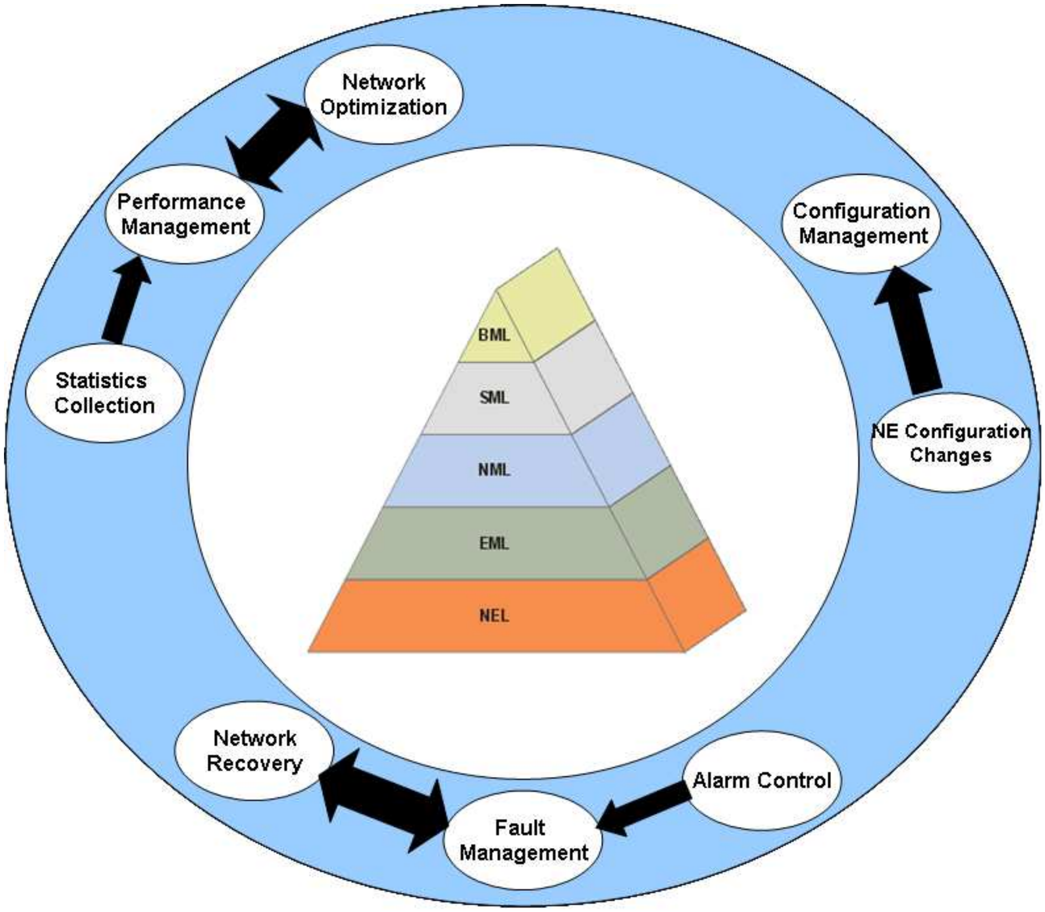


Figure 3 - TMN Logical Model and some basic Network Management functions

<u>Network Management Layer (NML)</u> – this layer offers all the functions and tools required for Network Elements (NE) management. This layer is divided in three logical inter-related areas: Fault Management (FM), Performance Management (PM) and Configuration Management (CM). NML is responsible to provide an End-to-End (E2E) perspective of the overall operator network, abstracting the heterogeneity and complexity of all technologies that are part of it.

<u>Element Management Layer (EML)</u> – this layer provides all the mechanisms and resources to implement in the Network Element Layer (NEL) the decisions made at upper layers, mainly at NML layer. EML is responsible to inter-connect NML layer with all the technology diversity nature of NEL layer. Typically, each technology (and thus its NEs) that composes the overall operator network has a corresponding Element Manager (EM) that is responsible to inter-connect and manage NEL components.

<u>Network Element Layer (NEL)</u> – this layer includes all the physical elements that compose a telecommunication network and that are responsible to implement all of its functionalities and protocols.

2.1.2. Performance Management Techniques

Performance Management covers a wide range of areas and functions; therefore, different techniques must be deployed. These techniques range from drive-tests, on-line monitoring of real-time performance metrics, to trend monitoring based on historical and aggregated data.

For each of these techniques there is a software tool type that is most adequate for the specific technique requirements. For instance, a drive-test tool must be adequate to collect the signaling messages that are transferred in the network radio interface in order to trace the network failures. This type of analysis is very specific, detailed, geographically limited and expensive, therefore its occurrence tends to be scarce, and is typically performed only when PM counters do not provide enough detail.

Other tools that rely mostly on PM data generated and collected from network elements, tend to have a wide and frequent use. This type of data can be used to perform on-line network monitoring that is characterized by the small data granularity which, in case of failure, enables the ability to act immediately upon the problem. Another important analysis performed using PM counters is the trend analysis, which can serve the intents of both Network Planning and Optimization Engineers, Marketing, Administration and Business Development people.

A PM Reporting tool is the key application in providing all this information, as its capabilities allow manipulate the network PM data to directly answer all these use cases. The PM data can be more or less summarized and relations can be established between performance indicators in a way that logically can provide answers for question coming from a multitude of different perspectives.

Although existing performance techniques work very well for network performance monitoring, the same is not applied to service layer QoS monitoring. The currently available systems depend on the metrics collected by the network elements in their self-monitoring procedures, which is a problem when it comes to monitor IP-based services QoS performance. The challenge of monitoring IP-based services over Cellular networks is that in the IP service layer path there are no network elements to retrieve performance data. One of the innovative factors of this work is to propose the usage of new IT techniques, like deep packet inspection, that extend the scope of the performance data gathered from the network to focus on IP-based service layer statistics.

This new data sources associated with performance management techniques that historically have proven to be effective will drive to a broad bouquet of monitoring solutions. For instance, there are already some tools that provide the ability to run drive tests at the application layer, which enables the capability to collect and analyze Quality of Experience statistics (QoE). In the other hand, there are already some solutions that address the on-line monitoring of IP-based Service Sessions that allow to detect almost instantaneously degradations in the session QoS performance.

However these solutions do not answer the questions that drove this work, mainly related with how network and service layer performance influencing each other behavior. This work proposes a new reporting approach that relates, in a cross-layer fashion, the service metrics with the network layer indicators.

2.2. Packet Switched Networks

Although Universal Mobile Telecommunication System (UMTS) Core Network (CN) can be constituted by both Packet-Switch (PS) and Circuit-Switched (CS) parts, this work focuses on PS Core Network only. The figure 4 illustrates a basic UMTS Packet Switched Network, representing Radio Access Network, Packet Switched Core Network, and additionally a service usage proposal.

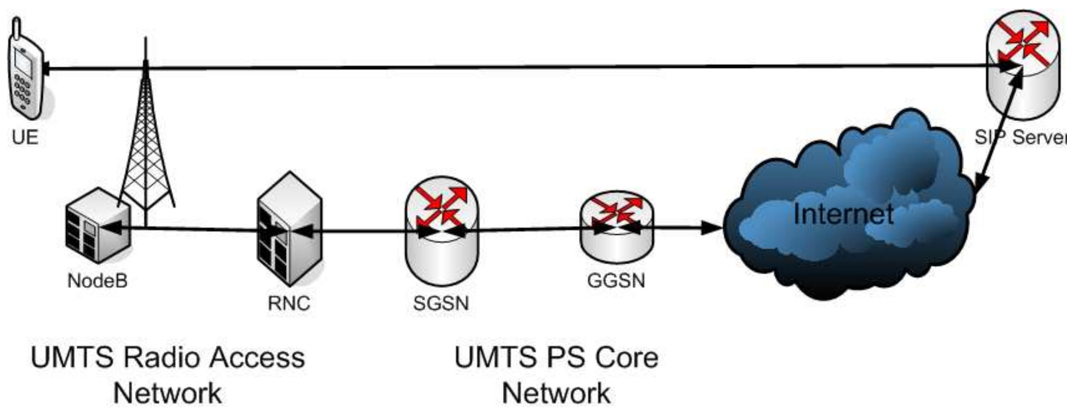


Figure 4 - UMTS Packet Switch Network Architecture

The next section describes the 3GPP UMTS Packet Switched Network architecture proposed by [8], its functional parts and network elements. There is an additional note that is related with the fact that this work is based on the 3GPP UMTS Rel6 Technical Specifications.

2.2.1. Network Architecture

<u>UMTS Terrestrial Radio Access Network (UTRAN)</u>

The UMTS Terrestrial Radio Access Network implements and controls all the access procedures to the network physical media – Radio Spectrum. Its main responsibility is to provide all the necessary conditions for User Equipments to connect to the network and establish logical links towards the Core Network. This logical link is known as Radio Access Bearer (RAB), which is used by RAN to create a connection context between User Equipment (UE)

and CN. RAB implements, at RAN side, all the E2E QoS call related requirements, defined by CN.
In order to be capable of supporting RAB QoS requirements for each call, RAN has the responsibility to manage all the radio resources available and system capacity usage. As it is commonly known, the radio resources are scarce which makes Radio Resource Management (RRM) function as one of RAN's most important features.
UTRAN is constituted by two major network elements: Node B and Radio Network Controller (RNC).

Node B

Node B is the radio access point of UTRAN; its main responsibility is to implement the Uu radio physical interface, allowing that UE links to the network and enabling data transmission from and towards the network. Node B interfaces RNC through Iub interface.

Radio Network Controller (RNC)

The RNC is the brain of UTRAN. RNC is the responsible element for the control of radio resources.

Functions that are performed by the RNC include:

- Iub transport resources management;
- Traffic Management of common and shared channels;
- Macro diversity combining/splitting of data streams transmitted over several Node Bs;
- Soft Handover control;
- Power Control;
- Admission Control;
- Channelization Code allocation;
- Handover Control;
- Packet Scheduling.

RAN Interfaces

There are three major transmission interfaces internal to RAN part: Iu, Iur and Iub interfaces.

Iu interface is responsible to connect RNC to corresponding Core Network element (SGSN). This interface is used to carry both Control and User Plane information. The control plane protocol stack consists of Radio Access Network Application Part (RANAP) and is used to control the Radio Network Layer. The Iu User Plane part is responsible for the user data transmission between RAN and CN.

Iur interface is responsible for the communication between adjacent RNCs. This interface was initially designed to support inter-RNC soft handover; however, more functions were added afterwards, like the support of inter-RNC mobility, dedicated and common channel traffic transmission and global resource management support.

Iub interface is responsible for the communication between RNC and Node B; it supports both Control Plane and User Plane protocol stack. The control plane protocol stack consists of Node B Application Part (NBAP). The Iub User Plane part consists of transmitting Frame Protocol Packets that transport MAC Layer Protocol Data Unit (PDU) related to radio dedicated and shared channels.

UMTS Packet Switched Core Network (UMTS PS CN)

Packet Switched Core Network plays a central role in UMTS system by providing means for subscriber authentication, mobility management, session management, packet routing, charging, etc. In order to implement these functionalities, there were defined two major network elements: Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN).

Serving GPRS Support Node (SGSN)

SGSN is the UMTS PS CN dedicated node for authentication and mobility management. This node serves as point of connection between the access network and the packet routing node by creating a mobility management context, which keeps track of UE Routing Area of specific cell, allowing to always delivering the packets to the UE regardless of its mobility. Its task includes:

- Packet transfer between RNC and GGSN;
- Mobility Management (attach/detach and location management);
- Logical Link Management;
- Authentication;
- Charging functions.

Gateway GPRS Support Node (GGSN)

GGSN is the UMTS PS CN gateway towards external IP networks such as public Internet, other mobile service provider's GPRS services, or enterprise intranets. GGSN maintains routing information necessary to tunnel the PDU data towards a SGSN serving a particular UE. To be able to route information from external networks towards the end user, GGSN holds information about the subscriber like: IMSI number, Packet Data Protocol (PDP) addresses, UE location information and serving SGSN information.

Data coming from SGSN in direction to an external network is converted by GGSN from GTP-U packets into IP packets and then transmitted. External access is accomplished through an entity called Access Point Name (APN) that is defined and implemented through GGSN. APN identifies the external networks or service servers that are accessed by subscribers via GGSN. For each GGSN there can be several thousands of defined APN.

2.2.2. Packet Session Establishment Procedure

This section explains all the fundamental steps for the establishments of a PS session, detailing the message flows that occur in this procedure. This session setup flow is composed by: Radio Resource Control (RRC) Connection Setup Procedure defined in [9]; GPRS Mobility Management control defined in [10]; Radio Access Bearer Assignment Procedure described in [11]; Radio Bearer Reconfiguration procedure defined in [9] and Radio Link Reconfiguration procedure described in [12].

This call flow description begins assuming that the UE is in RRC Idle Mode (UE is camped in a given cell). Figure 5 shows the control message flow for the call setup initiation process. UE starts to transmit a RRC connection request message to the RNC, through transport Radio Access Channel (RACH), which is encapsulated by the Physical Random Access Channel (PRACH).
This RRC connection request message is relatively small requiring only a single transport block transmitted using a 20ms Transmission Time Interval (TTI) and the RLC protocol in transparent mode. The RRC connection requested message can always be retransmitted, if necessary.

Once the RNC receives this RRC Connection Request message, it requests a Radio Link at the relevant Node B using NBAP signaling (NBAP: Radio link setup request/response messages). Node B starts to transmit a Dedicated Physical Control Channel (DPCCH) for the new Radio Link (RL). The RNC then reserves a set of Iub resources, using Access Link Control Application Part (ALCAP) Establish request/confirm messages.

Then, the RNC answers to the first RRC connection request message sent by the UE, using a RRC connection setup message, transmitted over Forward Access Channel (FACH). This RRC connection setup message is relatively large and requires seven transport blocks. These transport blocks have a size of 168 bits which are transmitted using a 10ms TTI and unacknowledged mode RLC. The block set size is typically defined such as two transport blocks can be sent per 10ms TTI. If the UE is in a poor coverage area, then it is possible that it did not receive one or more of the RRC connection setup message transport blocks. If it is the case, then the RNC is required to retransmit all messages from layer 3 (RRC layer).

Once the UE receives the RRC connection setup message, it attempts to achieve air-interface synchronization using the DPCCH transmitted by the NodeB. Once the UE achieves the air-interface synchronization, it starts to transmit in uplink on DPCCH, allowing the Node B to achieve the air-interface synchronization. Having achieved the air-interface synchronization, the Node B informs the RNC by sending a NBAP Synchronization Information Message. While this happens, the UE answers the RNC RRC connection setup message, by sending a RRC connection setup complete message. The RRC connection setup complete message and all the subsequent RRC signaling is transmitted using acknowledged mode RLC, and any necessary retransmission can be relatively rapidly completed in layer 2. At this point, the UE has established a RRC connection and the first phase, in the packet session establishment procedure, is completed.

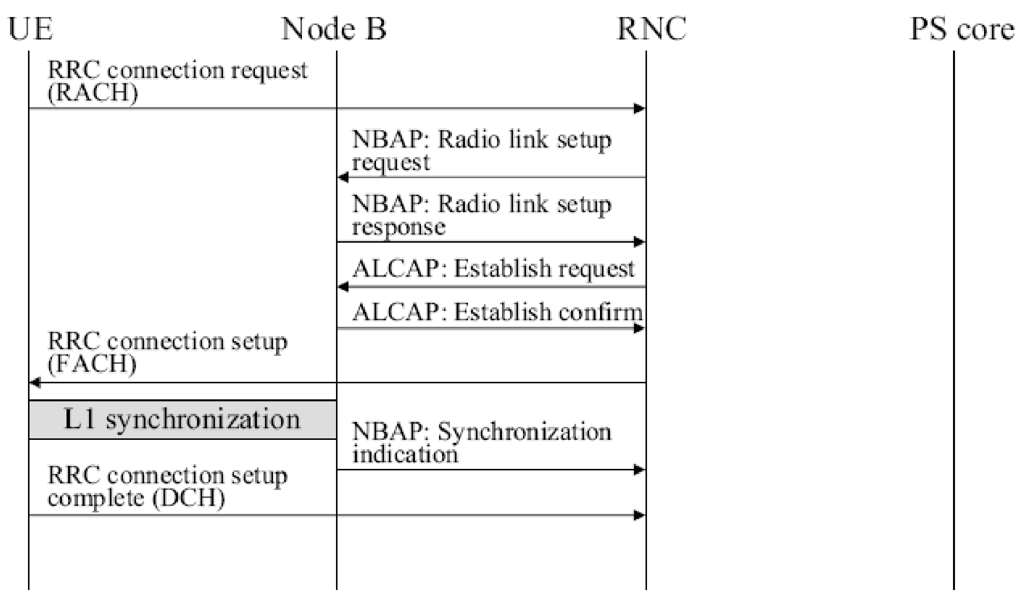


Figure 5 - Radio Resource Control (RRC) Connection Setup

In the above example, the RRC connection setup procedure used a dedicated channel (DCH). The alternative is to use RACH/FACH for the RRC connection setup (including RRC connection setup complete message). In this case, RACH/FACH would be used as well, for the transmission of the subsequent GPRS mobility and session management signaling. DCH allocation, base station and Iub resource reservation takes place during the radio bearer reconfiguration.

The second phase of a Mobile-Originated (MO) Packet-Switched (PS) data session establishment procedure involves GPRS Mobility Management (GMM) signaling with the core network. Figure 6 shows the GMM control message flow exchanged between the involved NEs.

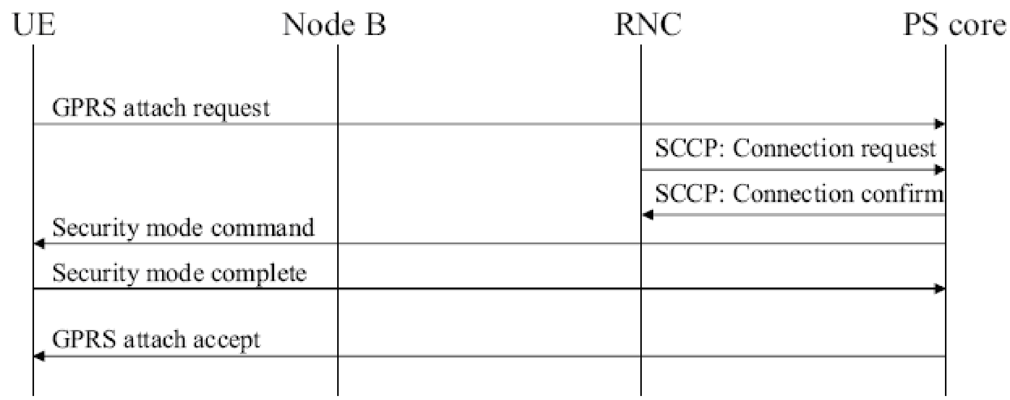


Figure 6 - GPRS mobility management

The UE sends a GPRS attach request message to the PS core through the RNC. The RNC relays the content of the message to the core network using a RANAP Initial UE Message. The RANAP message is combined with a Signaling Connection Control Part (SCCP) connection request message which is utilized to request the Iu signaling resources. The core network answers to the RNC with a SCCP connection confirm message to acknowledge that a signaling connection has been established across the Iu interface. The core network then completes the security mode procedure. There may be also a requirement to complete the authentication and ciphering procedure, which can be configured if it is needed for all the connections, or just for a specific percentage of them. Once the security mode is completed, the core network is able to send the GPRS attach accept message, becoming the UE registered for packet-switched services.

The third phase of Mobile Originated Packet Switched data session involves GPRS Session Management (GSM) and RAB assignment signaling messages, which are shown in Figure 7.

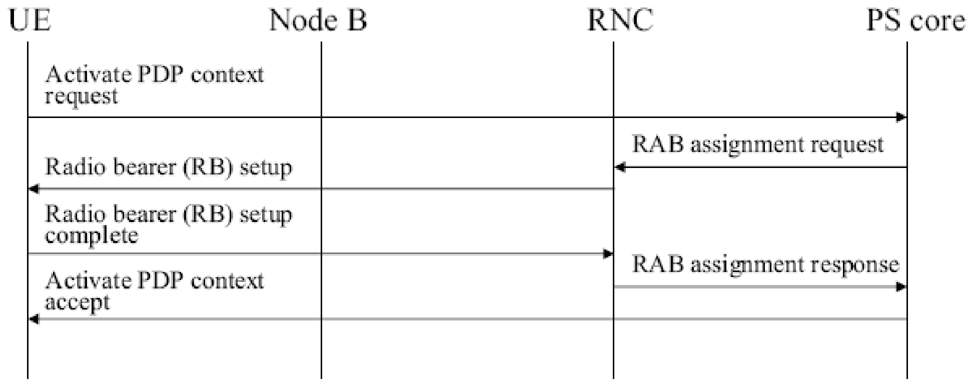


Figure 7 - GPRS session management

The UE starts to send to the Core Network a PDP Context Request message. This message defines and communicates to the Core Network what should be the call QoS requirements for the packet session, which can be done using one of the following ways: one way is to explicitly specify within the message, these QoS requirements; the other way is to indicate in the message, that the UE QoS requirements defined in Home Location Register (HLR) must be applied. Also, this message requires the assignment of an IP address and specifies the APN to which UE wants to connect. In response to this message, the Core Network sends a RAB Assignment Request to the RNC. The RNC runs the Admission Control (AC) algorithm and sends to the UE a Radio Bearer (RB) setup message, specifying a set of physical, transport and logical channel configurations. At this stage, the RNC may act in one of the following two ways: it can define a finite user plane bit rate to this session, even before it has any knowledge about the traffic that is supposed to transmit; or alternatively the RNC assigns zero bit rate to the user plane, waiting to receive a Capacity Request before assigning the appropriate bit rate to the packet session. This avoids the assignment of high bit rates to session when transferring small amounts of data. The UE answers to the RNC by sending a Radio Bearer Setup Complete message, and the RNC follows it by informing the Core Network with a RAB Assignment Response message. By this time the UE has established a Radio Link (RL) to the Node B, a Radio Bearer (RB) to the RNC and a RAB to the Core Network. The Core Network completes this stage by sending to the UE an Activate PDP Context Accept message, which includes all the QoS attributes as well as the IP address assigned to the UE. This Activate PDP Context Accept message ends with the third phase of packet session setup procedure.

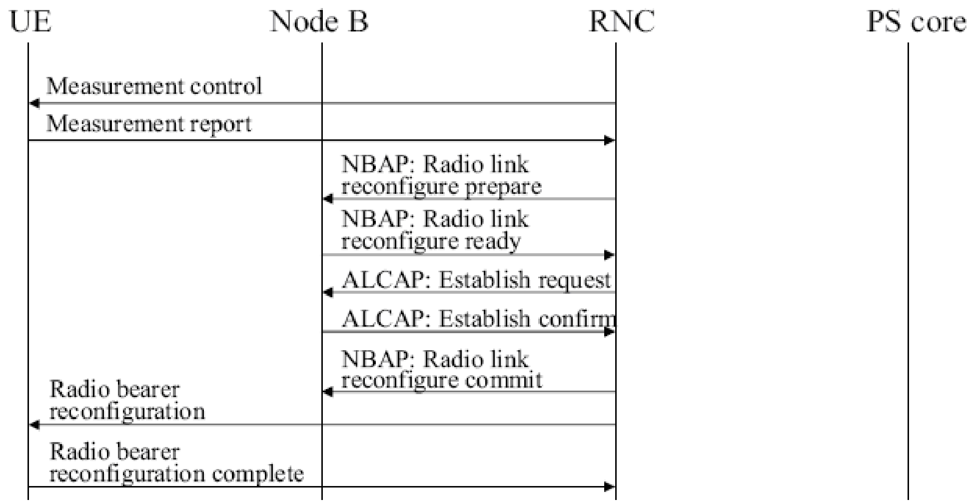


Figure 8 - Radio bearer reconfiguration for resource allocation

The fourth phase of the packet session setup procedure involves RB and RL reconfigurations. The RNC starts this phase by sending a Measurement Control message to the UE, which defines a threshold for traffic volume that can trigger an uplink capacity request according to 3GPP measurement reporting event 4a. There is a similar threshold in the RNC for the downlink that allows it to generate a downlink capacity request. In the signaling shown by the Figure 8, the UE initiates an uplink data transfer by sending a capacity request to the RNC, using the Measurement Report message. This report contains information about the amount of data that is waiting to be transmitted, which is used by the RNC to assign the appropriate bit rate. The RNC generates a capacity request from the measurement report, and assuming that it is accepted, the RNC instructs the Node B to reconfigure the existent Radio Link, using a NBAP: Radio Link Reconfigure Prepare message. This message contains all the required information, for the Node B to reconfigure the Radio Link in such a way that it can be capable of supporting a packet-switched data session. The Node B answers this with a NBAP: Radio Link Reconfigure Ready, indicating to the RNC that the reconfiguration is ready but not applied yet. Then, the RNC reserves a second set of Iub resources, using the same approach as in the first phase of this procedure, i.e. using ALCAP: Establish Request and Establish Confirm messages. Once the Iub resources are reserved, the RNC sends a NBAP: Radio Link Reconfiguration Commit message containing the Connection Frame Number (CFN), instructing in this way the Node B to apply the radio link configuration which includes the packet-switched bearer. The CFN has to be defined in such a manner that it guarantees that the UE applies the new configuration at the same time as the Node B. The CFN is communicated to the UE within the Radio Bearer Reconfiguration message sent by the RNC, which implies that the CFN must be set so it occurs after the UE receives this message. So, the CFN value should be a function of the Signaling Radio

Bearer rate, i.e. a shorter activation time can be used with the 13.6-kbps than with the 3.4-kbps signaling bit rate, and thus this must be taken into consideration. Plus, some margin should be allowed for L2 retransmissions and processing time, thus it is common that the RNC includes an additional parameter that defines a configurable time offset for the CFN.
Once the CFN occurs and the new configuration has become active, the UE answers with the Radio Bearer Reconfiguration Complete Message. This message and the subsequent signaling are transmitted across the air-interface using the 3.4 kbps bit rate, even if the 13.6 kbps bit rate had been applied previously. The UE is now configured with a finite user plane bit rate towards the PS core network and is ready to transfer data.

2.3.High-Speed Packet Access

HSPA is a term used to denominate two major technological enhancements to UMTS RAN technology: High-Speed Downlink Packet Access and High-Speed Uplink Packet Access. These new technological proposals were pushed by the increasingly need of data throughput demand.

2.3.1. UMTS R99 before HSPA

Although there are major achievements in HSDPA and HSUPA enhancements, Packet data transmission is supported right from the initial deployments of UMTS networks. The following downlink radio channels are available since UMTS R99 [13]:

- **Dedicated Channel (DCH):**
  - Transmitted over the Downlink Dedicated Physical Channel (DDPCH);
  - Maximum bit rate of 2 Mbps;
  - Can be used for any kind of service that requires QoS guarantees;
  - Suitable for medium or large amount of data transmission;
  - Reserves enough code capacity for the support of connection's required maximum bit rate;
  - Reserved codes are dedicated to a specific connection only;
  - Supports Fast Power Control;
  - Supports soft handover.

- **Downlink Shared Channel (DSCH):**
  - Uses dynamic variable spreading factor allocation;
  - Mainly designed to transfer bursty packet data;

  - Suitable for medium or large amount of data transmission;
  - Code resources are shared between all the connections using a time division approach;
  - Supports single-code and multi-code transmission;
  - Does not support Soft Handover;
  - Supports Fast Power Control;
  - It was removed from 3GPP specifications since the introduction of High-Speed DSCH (HS-DSCH) in UMTS Release 5.

- **Forward Access Channel (FACH):**
  - Transmitted over the Secondary Common Control Physical Channel (S-CCPCH);
  - Uses a fixed Spreading Factor;
  - Does not support Soft Handover;
  - Does not support power control;
  - Mainly designed to transfer bursty packet data;
  - Suitable for small amount of data transmission;

2.3.2. High-Speed Downlink Packet Access (HSDPA)

HSDPA was introduced in 3GPP Release 5 [14] and represents for WCDMA technology a major advance when compared with Release 99 capabilities. Although UMTS Release 99 (R99) already offers some methods for packet transmission over WCDMA air interface in downlink direction, the improvements introduced by HSDPA enhance substantially the downlink packet data throughput, which is fundamental for the support of rich multimedia content oriented services.

The required techniques to achieve higher data rates delivery and reduced delay are to implement Link Adaptation control closer to air interface and Physical Layer (L1) retransmission combining; HSDPA implements some architectural changes. A new user data transport channel is introduced, High-Speed Downlink Shared Channel (HS-DSCH), packet scheduling and link adaptation are conducted in the Node B according to the active scheduling algorithm and user priority handling mechanism. Channel quality is estimated by the Node B based on Channel Quality Indication (CQI) reports, power control, acknowledgment/non-acknowledgment (ACK/NACK) ratio and specific HSDPA user feedback.

HS-DSCH transport channel characteristics:

- Does not support soft-handover

- Does not support Fast Power Control;
- Uses a fixed Spreading Factor;
- Supports Adaptive Modulation and Coding (AMC) mechanism;
- Supports Multi-code operation;
- Supports Fast L1 Hybrid ARQ (HARQ) spectral efficient mechanism;
- Supports Node B scheduling.

WCDMA fundamental features include the variable spreading factor and fast power control; they are replaced in HSDPA proposal by Adaptive Modulation and Coding (AMC) technique, usage of multi-coding method and L1 Hybrid ARQ. HSDPA link adaptation and AMC techniques select the available coding and modulation combination scheme that requires higher Signal to Noise Ratio (SNR), for the user close to the Node B, i.e., with good interference/channel conditions in the short-term sense. This leaves to additional user throughputs for free, because in case of good air interface conditions the selected code and modulation can be less robust (when compared to require code and modulation scheme to address bad radio conditions), thus overhead is reduced which allows increasing payload bit rate transmission. To enable large dynamic range of HSDPA link adaptation and maintain good spectral efficiency, a user is allowed to allocate 15 multi-codes at the same TTI, which makes possible the usage of maximum allowed bit rate by one user only.
The Fast L1 HARQ mechanism allows that, in case of packet decoding failure in the UE, the packet is retransmitted by the Node B with identical information, or contains different bits encoding information. This redundant strategy allows diversity gains and improved decoding efficiency achievements.

The new HSDPA architecture does not impact the existing UMTS RAN elements organization the changes are mainly due to new enhanced functionalities. The R99 transport channels are terminated at RNC, which implies that the packet retransmission procedure, for a specific PS call, is located at the Serving RNC (SRNC). HSDPA architectural improvements propose a new shared downlink channel HS-DSCH, which implies the enhancement of Node B capabilities with the addition of HSDPA Medium Access Control (MAC) layer functionality, represented in Figure 10. The new MAC layer functionality implies the handover of control of packet retransmission from RNC to Node B, leading to faster retransmission, decreasing latency and retransmission delay.

The following figure shows the packet retransmission handling both in R99 and HSDPA systems, in the case where the Serving RNC and Controlling RNC are the same.

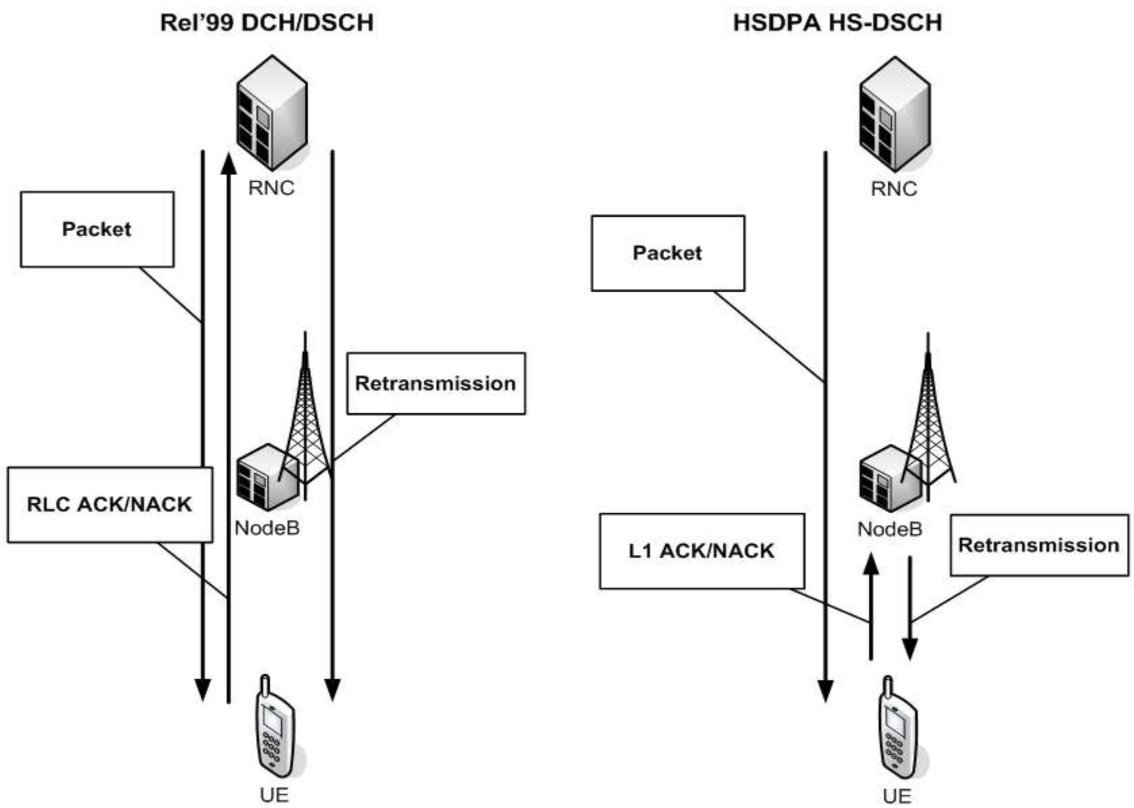


Figure 9 - Packet Retransmission Control: Rel99 versus HSDPA

In the HSDPA architecture, the RNC still retains the control of the Radio Link Control functions, which in case, for instance, of exceeding the number of maximum HS-DSCH retransmissions from Node B, it will take care of retransmitting the lost packet.

The new Node B MAC functionality is responsible for the HARQ functions, scheduling and priority handling. Ciphering is still done in the RLC layer, i.e. at the RNC, to ensure that the ciphering mask stays identical through the entire transmission/retransmission process, enabling this way the combining of retransmissions.

The type of scheduling to be implemented in the Node B is not defined by 3GPP standards, just few parameters are addressed, such as discard timer or priority scheduling indication that can be used by RNC to control the handling of an individual user. The scheduler type has a great impact on the system performance thus impacting the Quality of Service (QoS). This is a key factor for equipment vendor differentiation and influences the network operator choice in closing deals or not.

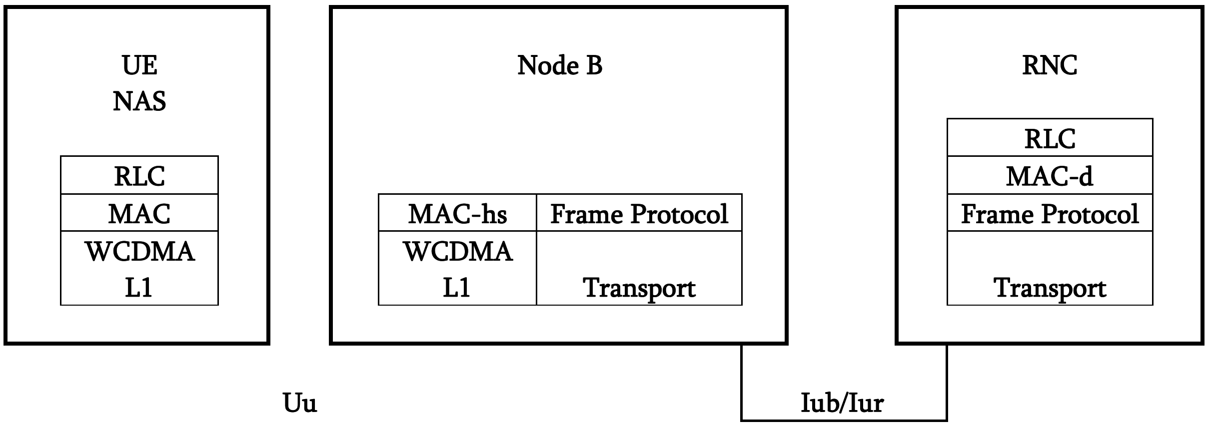


Figure 10 - HSDPA Protocol Architecture

As mentioned earlier, HSDPA proposed a new physical layer structure. The following three channels were introduced:

- High-Speed Downlink Shared Channel (HS-DSCH) – high speed transport channel used to carry the user data in downlink direction, with peak rates ascending to 14Mbps using 16QAM modulation and 21Mbps using 64QAM (introduced in Rel07);
- High-Speed Shared Control Channel (HS-SCCH) – this high speed channel is used to transmit physical layer control information, in order to enable HS-DSCH data decoding and to perform physical layer combining of retransmitted data on HS-DSCH;
- Uplink High-Speed Dedicated Physical Control Channel (HS-DPCCH) – this high speed channel is responsible for carrying the control information in the uplink direction, such as ARQ acknowledgments (ACK/NACK) and downlink Channel Quality Information (CQI).

High-Speed Downlink Shared Channel main features are:

- TTI of 2 ms which allows achieve a short round-trip delay retransmitting the data between UE and Node B;
- High-order modulation schemes of 16QAM and 64QAM;
- Fixed Spreading Factor (SF 16):
- Multi-code transmission and code multiplexing;

The HSDPA Spreading Factor is equal to 16 which imply that there are 16 channelization codes available. Since at least one channelization code must be reserved for common channels, the maximum number of codes available for HS-DSCH and associated (Uplink) DCH channel is 15. Depending on UE capability, there can be allocated for the same user for a given TTI 5, 10 or 15 codes.

Code multiplexing allows that different UEs use the same HS-DSCH (or same channelization code) channel to transmit data at different TTI, which contributes to improved spectral efficiency.

HS-DSCH modulation can be originally of 16QAM or, since Rel07, of 64QAM. The use of these modulation schemes, associated with multi-code technique, increases significantly the available user data rates.

2.3.3. High-Speed Uplink Packet Access (HSUPA)

The HSUPA [15] main goal is to improve capacity, data throughput and reduce delay in UL dedicated channels. The main idea behind the enhancements introduced by HSUPA is to achieve the defined goals by applying similar techniques to those introduced by HSDPA: Node B scheduling and Fast Physical Layer retransmission techniques.
The fundamental change introduced by HSUPA is the proposal of a new Enhanced Dedicated Channel (E-DCH) that implements several innovative techniques when compared with DCH. The main differences introduced by E-DCH are:

- Extensive multi-code operation support;
- Fast Physical Channel Hybrid Acknowledge Request (Fast L1 HARQ) ;
- Fast Node B scheduler.

HSUPA basic functional principle is the following:

- Node B uses UE specific feedback to estimate the UL data rate transmission needs;
- The Node B scheduler instructs the UE about the UL data rate, scheduling algorithm and prioritization scheme to be used;
- Data retransmission from UE towards Node B is initiated according to Node B feedback, using ACK/NACK messages.

With the introduction, at the Node B, of an additional retransmission procedure, a new MAC layer functionality is introduced to implement this procedure and additionally take care of uplink scheduling functionality. In the RNC side, Radio Link Control layer retransmission is still kept for hypothetical case of the new L1 retransmission procedure fails, thus needing that RLC retransmission procedure acts in replacement.
Retransmission process with HSUPA is dealt at two different layers, Physical and MAC. The physical layer packet combining process is handled by Node B where the soft buffers and CRC mechanism are located. In other hand, the MAC layer packet re-ordering is handled by RNC.

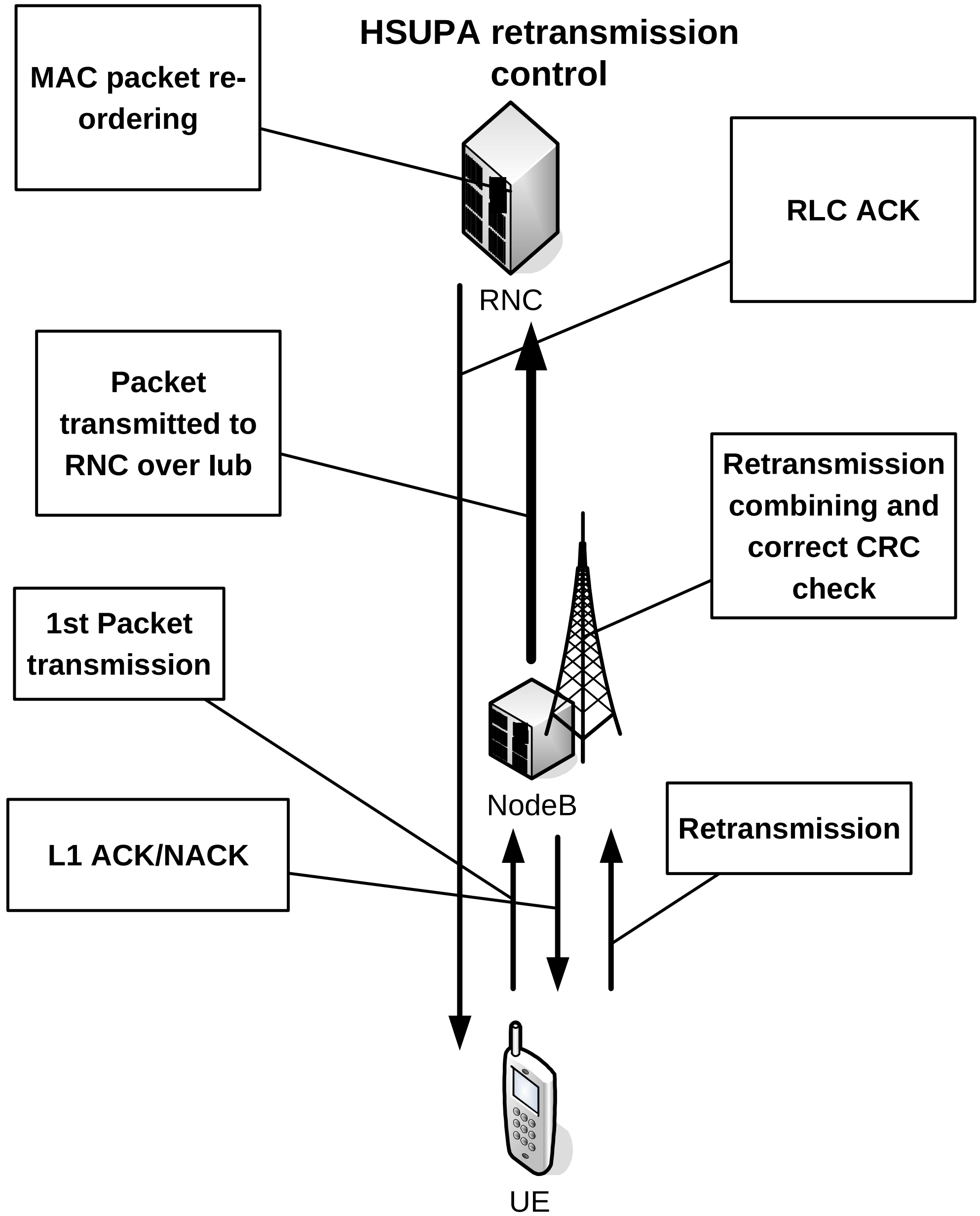


Figure 11 - Packet Retransmission Control: HSUPA

As shown in Figure 11 the HSUPA packet retransmission control flow is described as follows:

- UE transmits the packet over the radio interface, using the data rate specified by the HSUPA Node B scheduler;
- Node B checks the received packet integrity. In case of erroneous packet transmission, Node B instructs the UE to retransmit the packet through a Fast L1 Non-ACK message;
- UE retransmits the same packet, indicating that it is a retransmitted packet and the retransmitted packet redundancy version;

- The last two steps will repeat until a successful packet retransmission occurs in any of the active set cells or the maximum number of retransmissions is reached. In case of successful packet transmission from UE to Node B, this is forwarded to RNC;
- Since E-DCH supports soft-handover mechanism, there is a high probability that the RNC receives the same user data from several different Node B, thus imposing the need to implement a MAC packet re-ordering mechanism to process the incoming data;
- After the packet correct reception on RNC side, RNC sends a RLC ACK message to the UE, closing this way the RLC retransmission cycle.

Figure 12 shows the introduction of the HSUPA MAC functionalities in the protocol architecture. The new Node B MAC functionality (MAC-e) is supposed to handle the Hybrid Automatic Repeat reQuest (HARQ) retransmission, scheduling and priority handling functionalities. The UE has also a new MAC functionality (MAC-es/e) responsible for uplink retransmission, scheduling and E-DCH Transport Format Combination (TFC) selection. The RNC MAC-es functionality is responsible for packet re-ordering, which avoided changes at already existent MAC-d layer. The packet re-ordering mechanism is needed due to the fact that E-DCH supports soft-handover, and thus, packets arriving to the RNC at different sequence order from different stations have to be handled by this mechanism.

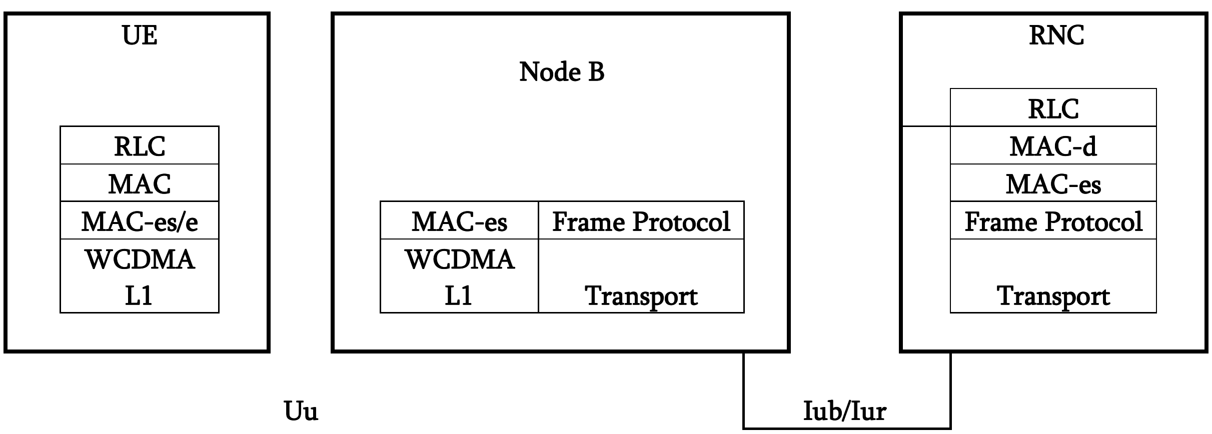


Figure 12 - HSUPA Protocol Architecture

As mentioned before, the HSUPA proposes a new transport channel, E-DCH that co-exists with the R99 DCH channels. The parallel usage of both E-DCH and DCH transport channel occurs every time an AMR voice call is established in parallel to a high-speed uplink data transmission call. At this stage, the parallel usage of E-DCH and DCH control part (DPCCH) is always present to carry pilot and downlink power control commands in the uplink direction. The E-DCH introduction has impacts on the physical layer architecture with the introduction of several UL and DL channels.

The channels introduced in UL are:

- Enhanced Dedicated Physical Data Channel (E-DPDCH) – this physical channel maps the E-DCH transport channel for user data transmission on radio interface;
- Enhanced Dedicated Physical Control Channel (E-DPCCH) – this is the control channel that provides feedback to the Node B about the data sent through E-DPDCH.

The DL channels are:

- Absolute Grant Channel (E-AGCH) – provides information to UE about the transmission power level that must be adopted in the next TTI. The E-DCH power level must always be higher than the DPDCH (physical channel associated with DCH) power level;
- Relative Grant Channel (E-EGCH) – indicates the UE to whether increase, decrease or maintains its E-DCH transmission power.
- HARQ Acknowledgement Indicator Channel (E-HICH) – channel used by Node B to provide feedback to UE about the HARQ Acknowledge/Non-Acknowledge messages.

The main differences between Rel99 DPDCH and E-DPDCH are:

- Rel99 DPDCH may use 10, 20 or 40 ms TTI, while E-DPDCH may use 10 or 2 ms TTI. The possibility of using a 2 ms TTI contributes for decreasing the round-trip time, but can result in excessive control information exchange if UE is on the cell edge;
- While in the early HSUPA releases the modulation is unchanged when compared with Rel99 modulation scheme (BPSK), the release 7 introduces the possibility of using 16QAM, which also allows to support even higher data rates, in UL, when good radio conditions occur;
- In the earlier HSUPA releases, since the modulation scheme remains unchanged, the increase in UL data rates is achieved with the extensive use of multi-code transmission. The Rel99 practical usage scenario is of a single SF4 code in UL transmission, which results in a peak user data rate of 384 kbps. In the E-DPDCH case, the data rates are a result of multi-code combination and the introduction of SF2. The combination of two SF4 channels with another additional two SF2 channels provide the ability to reach the maximum physical layer bit rate of 5.76 Mbps;
- E-DPDCH is responsible for carrying user data only and relies upon DPCCH to carry UE feedback about pilot quality, and on E-

DPCCH to carry the HSUPA-related physical layer control information.

E-DPCCH carries in a single message three different types of information:

- 7 bits for E-DPDCH-related rate information;
- 2 bits for retransmission related information that can either indicate that it is a new packet transmission or a retransmission of previously sent packet. In the case of a retransmission, the transmission redundancy version is also indicated;
- 1 bit that informs the Node B about the UE need to increase the corresponding data rate. This is implemented in a form of a Happy Bit, if the bit is set to "happy" state this means that the UE is satisfied with the current bit rate that is being served, and thus there is no need for the scheduler to increase it.

These are the most important physical channels due to the fact that they are directly responsible to implement E-DCH and its innovations.

The HSUPA Node B scheduler operation details can be described as follows:

- The Node B scheduler senses the medium and takes measurements of its condition, for instance the noise level (interference at the Uplink) to decide if there is capacity to allocate more traffic to a specific cell, or to decide if current user's data rate must be lowered to decrease uplink interference;
- The scheduler also monitors the received UE specific feedback, the Happy Bits, taking into knowledge what are the users that could transmit at a higher data rate. It also relates the happy bit information against its own MAC-e buffer availability and uplink power headroom availability, which allows the scheduler to decide if there is available capacity to increase the data rate for each user requesting higher throughput;
- The scheduler makes also use of defined user priority configured at the RNC, to decide what should be the data rate adjustment to be applied to a specific user or set of users. It can be the case that there is the request from a specific user to increase the data rate (Happy Bit set as "unhappy") and the available capacity to provide it, but the user priority handling configuration does not allow such upgrade.

This mode of operation implies that the RNC instructs the Node B what is the maximum data rate allowed for a specific service subscription/user profile. For the same subscriber there can be different priorities depending on the type of service that is being accessed. This is done making use of MAC flow identifiers.

It should be noted that there are traffic types that do not make use of scheduled transmission. This is the case of Signaling Radio Bearer (SRB) related information or, for instance, VoIP service. Both of types of data require a very limited delay or jitter value and lower throughput, which means that scheduling this kind of data would just imply a degradation of service delivery, for instance due to UE delayed measurement reports. For this kind of services, the RNC provides permanent grant that the Node B scheduler cannot change. Thus, the non-scheduled transmission operates like R99 DCH (not using Node B scheduler capabilities), with the significant difference that it still takes advantage of the Fast L1 HARQ mechanism.

2.4. LTE/SAE Evolution

The 3GPP Rel8 proposes an evolution for 4G cellular networks, both in Radio Access Network and Core Network side. The new technology is known as 3GPP Long Term Evolution/System Architecture Evolution [16] (LTE/SAE), and proposed both enhancements in Radio Access side with an Evolve-UTRAN (E-UTRAN), and in Core Network with Evolved Packet Core (EPC) architectures. The Figure 13 [17] shows the LTE/SAE deployment scenario describing the inter-relation between the existing 3GPP technologies.

The requirements for these technological evolutions are:

- Optimize the packet switched domain;
- Decrease the Round-Trip Time (RTT) between UE and Server below 30 ms and access delay below 300 ms;
- Increase the peak bit rates in DL to 100 Mbps and in UL to 50 Mbps;
- Guarantee a good level of secure and mobile communications;
- Improve the terminal power consumption efficiency;
- Provide frequency allocation licenses flexibility splitting the allocation share into 1.25, 2.5, 5, 10, 15 and 20 MHz. Also allow to license frequency share adjacent to WCDMA spectrum;
- Improve capacity when compared to UMTS Release 6 HSDPA/HSUPA system.

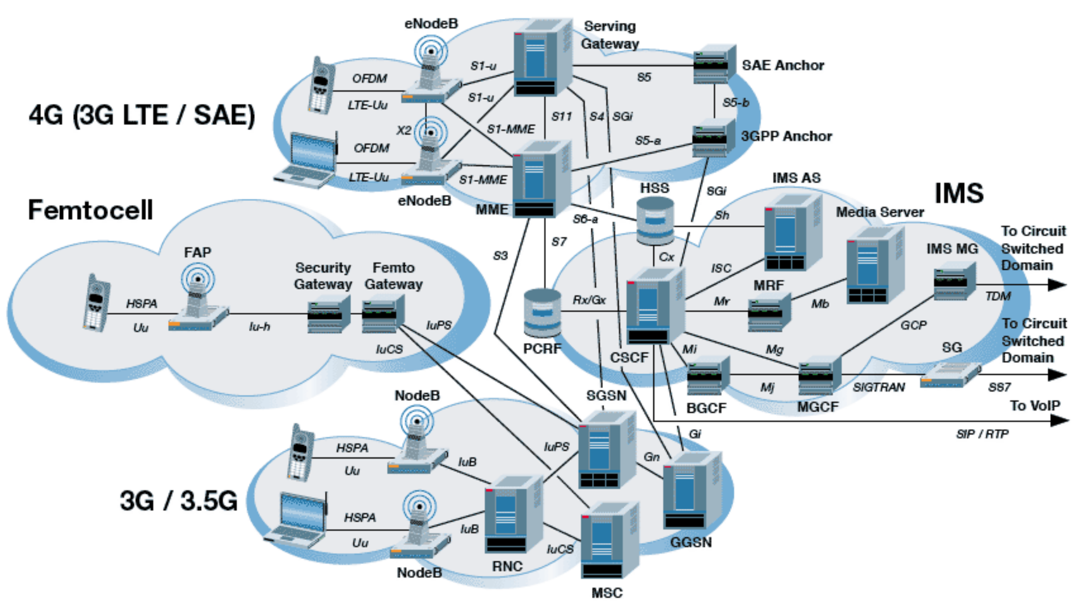


Figure 13 - LTE/SAE deployment scenario

The new enhancements proposed in this technology evolution are:

- Use of Orthogonal Frequency Division Multiple Access (OFDMA) and advanced multiple antenna techniques over the air-interface;
- Distributed (as opposed to centralized) architecture for RAN;
- IP-based Radio and Core Network;

The OFDMA together with multiple antenna techniques increase significantly the spectral efficiency, while OFDM is highly scalable. Therefore, going from 1.25 MHz or 5 MHz to a 20MHz channel bandwidth is achievable, thus offering higher peak data rates. This efficient air-interface allows many users to experience data rate in excess of 1Mbps, being the peak data rate target 100 Mbps in a 20MHz spectrum in downlink. The 3GPP access is based on OFDMA in downlink direction and Single Carrier-Frequency Division Multiple Access (SC-FDMA) in uplink.

A distributable RAN architecture allows reduced latency, since most dynamic decisions are made locally in the evolved-Node B (e-Node B). Since the Radio Resource Management decisions are taken close to the air-interface and the UE, therefore the required QoS parameters of a multimedia application can be managed is a more efficient manner. The proposed IP-based RAN results in an easy-to-scale network.

A fully packetized IP-based CN will replaced 3G CS and PS networks, providing high scalability and reduce cost. It also simplifies the introduction of new services via the IP Multimedia Subsystem (IMS). A very important feature introduced by the IP-based architecture is the capability to implement seamless mobility between different RAN technologies without impacting the session logical layer, which is managed by Mobile IP protocol. This enables the ability, in handover control, to always select the RAN from session level perspective.

<u>LTE/SAE Architecture</u>

The 3GPP LTE places the full radio functionality in the base station, so Evolved UTRAN (E-UTRAN) architecture consists of an evolved Node B (eNodeB). The interface between the eNodeBs is known by X2, which provides the means to support handovers. Figure 14 introduces a LTE architecture overview representing all the network elements and interfaces.

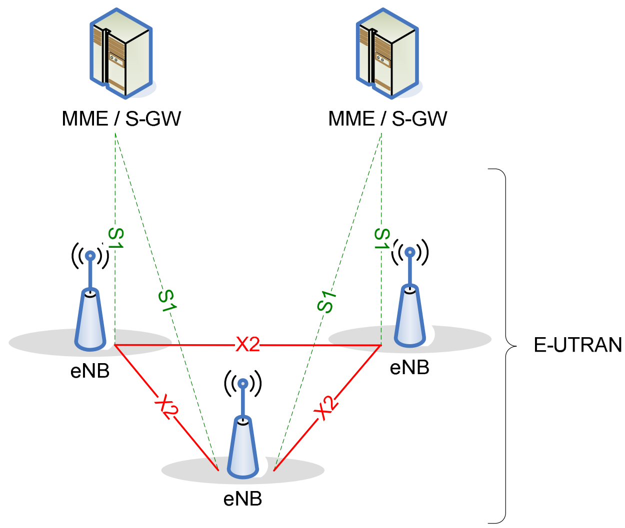


Figure 14 - LTE Architecture

The eNodeB new functionalities, when compared with HSDPA/HSUPA Node B, are RLC, RRC and PDCP layers.

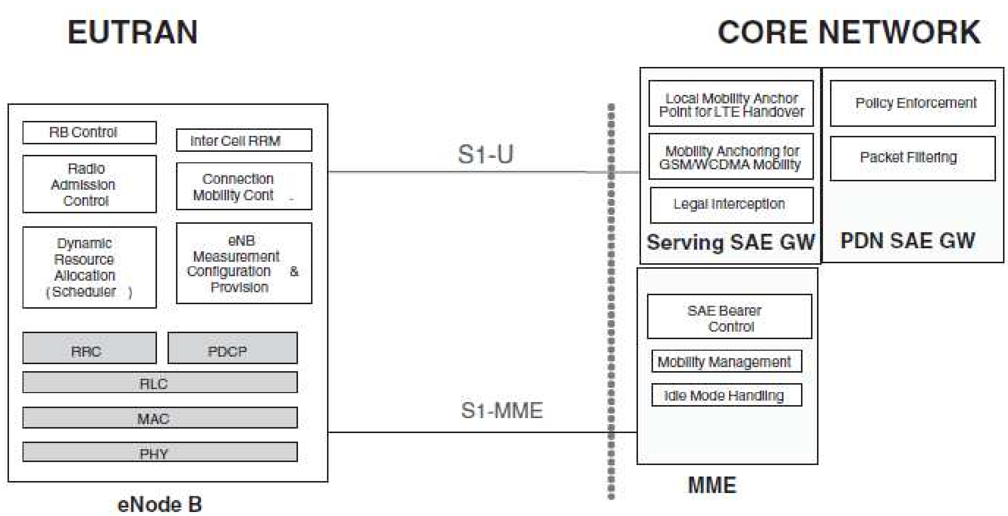


Figure 15 - LTE/SAE Protocol Architecture

From Figure 15 it can be depicted that the eNodeB connects to the new core network via a S1 interface, where S1 can have two distinct implementations in the core side: S1-MME and S1-U. The S1-MME connects eNodeB to the Mobility Management Entity (MME) and the S1-U to the Serving SAE Gateway (SGW) and Public Data Network SAE Gateway (PGW). The MME is the element responsible for handling the control plane signaling, specially the mobility management and idle-mode handling. In other hand, the SGW and PGW are responsible for processing the user plane data, as for mobility management inside LTE and between 3GPP RATs. In the new SAE core architecture, the interface that connects MME and SGW/PGW is S11 interface. There are two additional elements on the core side, Home Subscriber Server (HSS) and Policy and Charging Rules Function (PCRF). HSS covers the functionalities that HLR has for both 2G and 3G networks, and contains user specific information on service priorities, data rates, etc. The PCRF is related with the appliance of QoS and Charging policies.

This new flat network all-IP architecture provides high data rates, is scalable for increasing data volumes and adds capacity to support the increasing number of users, which in turn enables its cost efficiency.

LTE/SAE Performance Monitoring

The new flat All-IP architecture introduced by LTE and SAE advantages are not only related with network capability enhancements, but it brings also great and new things for the Performance Management field. The simplified user plane protocol stack, the evolution towards IP transmission links and the IP End-to-End native network brings new light to Service QoS performance and Network performance relation. Much of the IP-based Service metrics defined in this work, that in 3G are spread around a multitude of data sources, will be available on LTE/SAE as simple PM data which simplifies the access to this information.

Therefore, this means that this thesis work proposal is very much aligned with the innovation trend, and it can be also envisioned that, with few adjustments, it can be extended and simplified for the LTE/SAE support.

2.5.Summary

This chapter introduced the Network Management thematic describing some of the most used performance monitoring techniques and introducing the Telecommunications Management Network (TMN) Model that was used as the basis concept for this work. TMN Model proposes a layered network management approach, detailing the following layers: Business Management Layer (BML), Service Management Layer (SML), Network Management Layer (NML), Element Management Layer (EML) and Network Element Layer (NEL). This chapter also introduced the Packet Switched Network architecture, its network elements and functional parts: UTRAN and CN. The UTRAN is composed by the following elements and interfaces: RNC, Node B, UE, Iur, Iub and Uu. While the CN part is composed by SGSN, GGSN and Iu, it also details with the packet session establishment procedure as it is crucial for this work to understand the signaling process behind UMTS network access.

An important part of this chapter is dedicated to describe the UMTS Radio Access Technologies that enable the capability to transmit efficiently real-time multimedia user data. This chapter compares the UMTS R99 features with the HSDPA and HSUPA key features, highlighting this way the reasons for this 3GPP evolution path and how higher bit rates, less delay and latency, just to name few indicators, are achieved. HSDPA and HSUPA make use of techniques like: L1 Hybrid ARQ Fast Retransmission, Multi-Coding, Adaptive Modulation and Coding and Node B MAC Scheduling.

The last section of this chapter introduced the new 3GPP technological proposals: UMTS Long Term Evolution (LTE) and Service Architecture Evolution (SAE). These proposals main objective is to cope with 4G network requirements and the key are: Flat Architecture with less elements in the data path, thus reducing the latency; OFDMA, multi-antenna and scalable bandwidth allow a more efficient air-interface increasing the capacity and improving user experience; All-IP architecture simplifies the User Plane protocol stack and enables key features like seamless mobility between different RAN technologies, using for this the Mobile IP protocol.

3. Performance Management

Performance statistics play a central role in all network operational, maintenance, troubleshooting, planning and optimization procedures. Each performance statistic is fairly unique and is always related with a specific network parameter or event. There are many parameters, protocols and events that can be monitored, and thus, there are many related statistics that can be related between each other, which leaves to an infinite number of possible permutations. In this highly complex scenario, the challenge is to identify which are the most relevant statistics that must be monitored and analyzed in case of a particular network failure. This is the challenge that Performance Management Systems try to address.

This chapter is divided in six sections. Section 3.1 provides detailed description of the proposed Performance Management Architecture proposed by this work, and also details the fundamental elements and their functionality and purpose. Section 3.2 describes in detail all the performance management data sources both for network and IP-based service layer. The Section 3.3 introduces the ETL process as part of Performance Management cycle. Section 3.4 presents the network and IP-based service object model detailing the database structure for PM data storage. Section 3.5 provides a detailed description about the proposed KPIs included in this work. The last section, 3.6, describes the proposed reporting use case, which combines network and service statistical data.

3.1. Architecture Overview

This highly demanding network scenario requires a high performance, well organized, optimized and comprehensive PM system that can be used to assist network operator to take accurate actions in the least time possible. Figure 16 shows the proposed Performance Management system architecture, providing a complete overview from performance statistics sources to the mediations, adaptations, data storage and reporting applications.

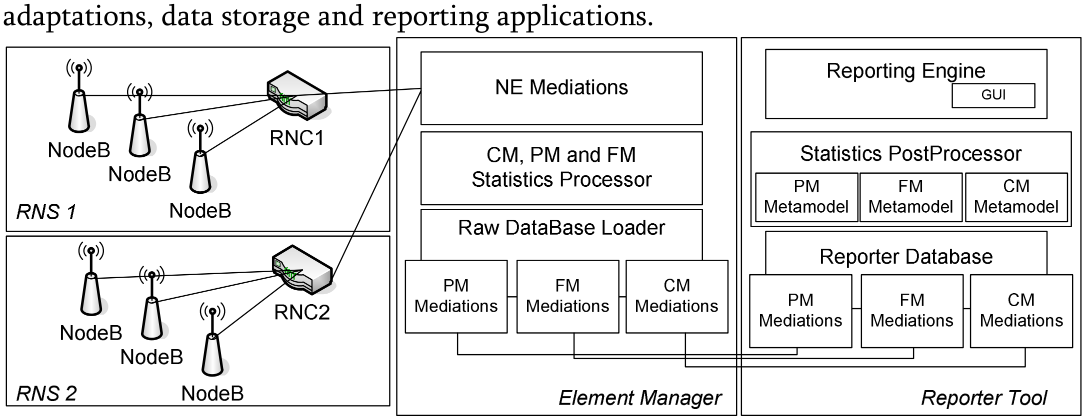


Figure 16 - Performance Management: System Architecture

Performance Management Concepts

There are two conceptual elements that are transversal to the PM System Architecture: Performance Measurement and Performance Counter. A third important conceptual element that can be derived from those elements is the Key Performance Indicator.

Network Elements

Each network element monitors its performance through the Performance Mediation. A subset of that module is responsible for the communication with the Element Manager. That interface is divided into three types of primitives relating to the type of data which is to be transported; Performance Management data (PM), Configuration Management data (CM) and Fault Management Data (FM). The first is intended to present metrics related with the continuous operation of the equipment, while the second indicates the configuration setup, including information such as topology and capabilities. FM is a more urgent type of data as it indicates critical issues to be evaluated. In figure 15 it is only present NodeB and RNCs, but any type of network nodes could be included beyond the Radio Network System (RNS).

Element manager

The NE Mediation manages the interactions between the Element Manager Module and the several Network Elements in the network. It is responsible for the collection of the Performance and Fault Management functions existing in each of the elements of the network. The NE Mediation Module implements the Extraction part of an ETL procedure.

The statistics Processor is responsible for converting all the diverse data gathered from the NEs according to a structured and generic meta-model, which will be briefly discussed ahead. This particular module processes Configuration, Performance and Failure Management Information.

The Raw Database loader is responsible for providing interfaces for the access relating to data storage management features. It is another function of an ETL procedure the upload of the gathered information into the raw databases. This module includes interfaces for mediation of the interactions between the EM and the analysis tools that evaluate the collected data. These interfaces answer to the Reporting Tool for requests related to CM, PM and FM data.

Reporting Tool

Reporter Database is a data warehouse designed for the coherent integration of diverse data sources, dimensioned to optimize the data discovery and reporting. This data repository modulates all the network topology into a hierarchal object structure, which provides the capability to analyze the entire network. This analysis can focus on the correlation of different parameters that can be Configuration, Performance or Fault Management related. A possible use case would be to assess what kind of configuration optimizes better the performance of the network, by improving network capacity and reducing its faults. This analysis can be extended in time and from different network perspectives, as historical and object data aggregation is possible. Moreover, the database primitives allow for the storage management and provide all the data access information to other layers.

This database is modulated based on NE specific metadata, which defines the Object Class (OC) structure, and for each OC all the PM Measurements, and related list of PM Counters and aggregation rules. The FM metadata is generic for all the OC, defining a list of failures that can occur.

Statistics Post-Processor is a software component that plays a decisive part on the reporting process. It is responsible for the entire object and time aggregations, which enhances the analysis capabilities, allowing the time trend analysis and drilling through the network objects, allowing a great diversity of network analysis. The aggregation rules are all defined through metadata specific for each NE, and provide information on how PM counters must be aggregated in time or in OC dimensions. For instance, these rules can define that a specific counter that is collecting information about the failed allocations of a specific cell resource, must be summed in time domain and in object domain, which results that this counter is counting, for the original object level, sequentially in time all the failure events, and when aggregated to a greater order object (Node B for instance), will accumulate all the contribution of the related cells.

Reporting Engine is the mind behind Reporter Tool. It is responsible for the database queries, it processes the results and displays them in a defined format. It provides all the data visualization capabilities, offering different pre-defined models and allowing the user to create their own. These pre-defined visualization models are important, because they allow the manipulation of data in different dimensions and the drilling through them, providing this way, different reports for different types of end users and even for different type of analysis, starting from a unique data set. Related to these models, there is an important reporting component, this is the Key Performance Indicators (KPI) set. A KPI is a data aggregation that provides fundamental information for the understanding of network behavior. It can be as simple as selecting a particular important counter, or as complex as implementing an arithmetic calculus, enhancing this way the data visualization capabilities. KPIs are defined in configuration components and can be either calculated on the fly by reporting engine or pre-calculated and store in Reporter Database.

The Automated Knowledge Discovery model is another important part of this reporting engine and provides the very important feature of automatic data monitoring, searching for patterns in the network behavior for the sake of forecasting upcoming events, such as Operation & Maintenance and Optimization needs.

3.2.Metadata

The metadata describes all the elements that are part of the information model, from the modeled network elements and its functional parts, to the performance measurement and counters. Every technology must have a set of metadata information that describes all the relations between all the monitored NEs, protocols, procedures and resources. Metadata is divided into three logical parts: Configuration Management (CM), Performance Management (PM) and Fault Management (FM).

3.2.1. Configuration Management (CM)

Configuration Management metadata is responsible for the mapping between the different NE present in the network and their components into a coherent and structured Object Class model. This way, CM metadata is used to identify objects with the same properties and to maintain possible occurrences of an object in the object class hierarchy.
Two types of objects can de defined for this model, Managed Objects (MO) and Reference Objects (RO). Managed Objects refer to objects that are directly related elements present in the network that can be managed, configured, manipulated, which are obviously the NE elements and their components e.g. a Node B and its Cells. Reference Objects refer to virtual reporting dimensions, i.e. elements that are virtually created to ease the network analysis by dividing and grouping the network into smaller segments thus reducing the analysis complexity. This Reference Objects are created and stored in to the Reporter Database by the Statistics Post-Processor module, using the CM metadata info.

Managed Objects have the following data structure:

- MOC ID – the numerical object class ID, that unequivocally identifies the Object Class within the object tree;
- MOC Vendor – identifies the vendor of the network resource;
- MOC Version – identifies the network resource version;
- OC Name – object class name that is displayed in the Reporter application, e.g. "NodeB";
- MOC Abbreviation – abbreviation used for the OC, for example "NB";
- MOC Time Stamp – the time when MO was last modified in the Reporter Database;
- MOC Description – description of the object class and the network resource that it maps;
- MOC Parent ID – the numerical number OC ID of the parent object class, if applicable. In case the OC is the root object class, this field is filled by default with the value "1".

Reference Objects have the following data structure:

• ROC ID – – the numerical object class ID, that unequivocally identifies the Object Class within the object tree;

• ROC Name – object class name that is displayed in the Reporter application, e.g. “Local Cell Group”;

• ROC Abbreviation – abbreviation used for the OC, for example “LCG”;

• ROC Time Stamp – the time when MO was last modified in the Reporter Database;

• ROC Description – description of the object class and the network resource that it maps;

• ROC Reference ID – since the Reference Object is always associated to one or more MOC instance, this field is used to store each object ID that is related to this reference object. For the LCG example, ROC Reference ID should be equal to each Cell MOC ID related that is part of it.

3.2.2. Performance Management (PM)

Performance Measurement Metadata is responsible for defining, for each network element, all the PM measurements and Counters and relating them to the CM data, i.e. to the OC structure. As network element represents a specific role in the network, there will be a different set of measurements/counters for each NE. The number of measurements and counters needed to monitor a specific NE is dependent on the NE complexity, ranging with the number of functionalities. A PM measurement is a logic representation of a NE functionality that defines a set of counters that monitors the network performance behavior. A PM Counter is the fundamental element of the performance monitoring process, as it provides detailed information ranging from specific procedures until group functions. As Counters are the basis of PM, from its values and monitoring purposes, there can be developed different kinds of aggregations such as KPIs and Reports, that allows to level the analysis from the most detailed information where there can be thousands of counters, through hundreds of KPIs and tens of reports for each object class, and ending with only one overview report containing only the most abstract and descriptive data. This way, different kinds of users and analysis can be satisfied with only one tool.

3.2.3. Fault Management (FM)

Fault Management Metadata defines the mapping between all the NE components and the fault events that describe system failures that can be either hardware or software driven. FM metadata thus relates OC with incoming network failure notifications. These failures are categorized and ranked by severity, which can range from debug to emergency state.

3.3. Data Sources

The novelty of this work is the proposal to relate IP-based Service Performance statistics and Network Performance statistics. For this purpose, several data sources were identified to provide statistical information from both Service and Network layers. This section describes in detail all the identified data sources.

Network Performance Management Statistics

These statistics provide detailed information about the overall network performance and its layers. As described in 4.1 this information is organized in PM Measurements which are logical organizational sets of Performance Counters.

As it is widely known, 3GPP Networks can be divided into main logical networks: Radio Access Network (RAN) and Core Network (CN).

For RAN sub-system comprising RNC, NodeB and UE elements, there are about a hundred performance measurements, corresponding to almost six thousand performance counters. This work assumes that the entire content of RAN PM information is available through the Performance Management system as described above. However, since this information is organized per network layers, the majority of these measurements/counters provide such detailed information describing events that can be monitored through the entire stack, and therefore a selection had to be made in order to provide the most meaningful information only.

The following **RAN PM measurements** were selected:

- Packet Call – provides information about packet calls from the mobile and the end user's perspective. Packet call sessions are active data transfer periods, and one RAB can contain multiple packet calls. This measurement monitors all User Plane dedicated and shared channels (DCH, HS-DSCH or E-DCH) and provides detailed information about:

Packet Call attempts detailed per channel
Packet Call allocations
Packet Call channel switches
Packet Call setup failures
Packet Call releases (normal and failure releases)

- Service Level – provides RRC and RAB connections related information, allowing monitor the network behavior from UE perspective. RRC and RAB procedures are divided in three phases; Setup, Access and Active phase. In case of network failure in any of these RRC or RAB phases, there is a bad experience by the end user.

- Traffic – this measurement provides information about the DCH, HS-DSCH and E-DCH allocation performance. It provides information about:

Channel allocations detailed per traffic class (CS and PS)
Requested bit rates (initial, upgrades and downgrades)
Channel switches
Channel allocation durations
Channel allocation setup failures per cause
Allocated channel drops per cause

- Cell Resources – provides information related to cell resources usage and availability, allowing to monitor in detail:

Total transmitted (DL) and received (UL) power
Traffic load estimates about the balance between RT and NRT traffic
Common channel usage rate: RACH (UL) and FACH (DL)
Code tree usage
NodeB hardware resources (Channel Elements – CE)
Average power per radio link
Cell availability duration

- Cell Throughput – provides information on the amount of data that is transferred from the SRNC MAC-d to the Iub interface. It enables easy monitoring of the distribution of HSDPA and DCH data volume, which is very useful for capacity optimization purposes.
- HSPA at NodeB – provides information about the HSPA functionality performance at the NodeB. It is mainly used for optimization and troubleshooting purposes.
- RNC Capacity Usage – provides information about the RNC resources availability and actual usage. It details the number of RRC connected users, Iub and Iu-PS licensed capacity usage, providing the ability to optimize RNC resource usage and to forecast the need to expand capacity.
- DSP Resources – provides information about the resource allocation of each RNC's DSP detailed per Service Type. This measurement allows to identify the number of calls per service

type (HSDPA, HSUPA, CS Voice, CS Video, etc.), which is important to evaluate the network traffic model.

- Iu-PS Performance – provides statistical information about the Iu-PS traffic load and GTP-U (GGSN Tunneling Protocol – User Plane) throughput between RNC and PS Core.
- RCPM RLC – provides information about the performance of user data transfer on RLC layer, for connection types that uses Acknowledge Mode (AM RLC), which are:

Signaling Radio Bearers
PS NRT (PS Background and PS Interactive) user data connections, which includes HSDPA and HSUPA
PS RT (PS Streaming) user data connections, which also includes HSDPA and HSUPA connections

The scope of RCPM RLC measurement can be seen in the following figure:

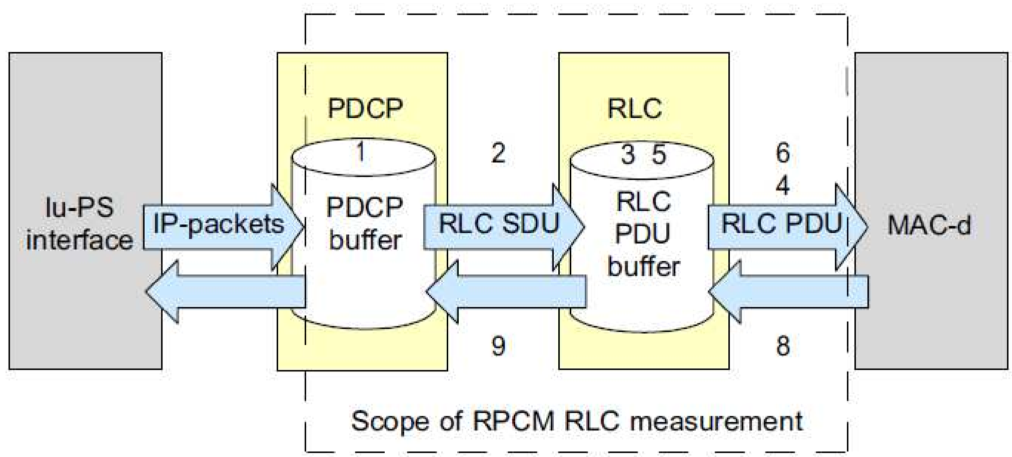


Figure 17 - Scope of RCPM RLC Measurement

As it can be depicted from Figure 17, this is a very important measurement, since it provides information about RLC UP data delivery performance that can be directly related with the Iu-PS throughput.

For CN sub-subsystem, it is only considered the Packet Switched part, which is composed by two major NEs: Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN). Since these nodes functionalities are very different and complementary between them, a separate analysis for each of these nodes is required.

Figure 18 introduces the PS Core Control-Plane protocols relating them with the NEs functionalities and interfaces.

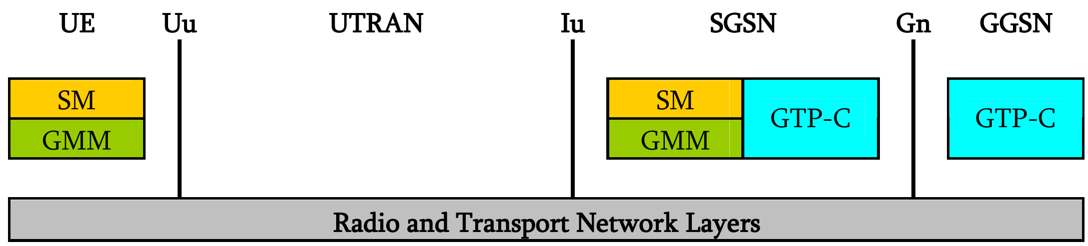


Figure 18 - PS Core Control-Plane Protocols

<u>SGSN PM Measurements:</u>

- Iu Mobility Management – provides detailed information about GPRS Mobility Management (GMM) related procedures performance. It includes detailed information about the following procedures:

Attach procedure details: attempts, successes and failures per cause
Attach durations detailing average, maximum and minimum duration
Detach procedure details: attempts, successes and failures per cause
Routing Area Update details attempted, successful, failed RAU and duration.
Inter-system Handover details attempted, successful, failed Handovers and duration

- Iu Session Management – provides detailed information about Session Management performance. It includes detailed information about the following items:

PDP Context Activation details: attempts, successes and failures per cause
PDP Context Modification details: attempts, successes and failures per cause
PDP Context Deactivation details: attempts, successes and deactivation reason
PDP Context QoS downgrade detailed per reason
PDP Context QoS upgrade rejections detailed per reason
Event Duration for each following event type: PDP Context Activation, Modification and Deactivation

- Iu Data Measurement – provides detailed information about the amount of data transmitted through Iu interface. It includes the following details:

Both GTP-U DL (SGSN –> RNC) and UL (RNC -> SGSN) throughputs, sent packets and amount of dropped data
Both GTP-U DL and UL throughputs, sent packets and amount of dropped data, detailed per associated priority queues
Both DL and UL throughputs, sent packets and amount of dropped data, detailed per traffic class, bearer class
DL and UL transmitted data detailed per PDP context providing the average throughput per PDP and the peak value occurrence

- User Management Measurement – provides detailed information about the User status. It includes the following details:

Attached Users details: average and peak number of attached users
Access Control details: the denied access requests per reason
PDP Contexts: details the average and peak number of active PDP contexts per priority classes. It also provides PDP activation rejections details
Active Subscribers: details the average and peak number of subscriber with active PDP Contexts

- CDR Measurement – provides detailed information about Call Detail Records controlled in SGSN. It provides the following details:

CDR generation: provides the number of generated CDR per subscriber type and the number of discarded CDR per reason
CDR duration: details the open CDRs average duration

- Iu RANAP Measurement – provides detailed and important information about the signaling changed between RAN and CN:

RAB Events: details the number of attempted, released and failed RAB establishments. It also details the RAB establishment failures per cause
Active RAB: details the average and peak number of active RABs

<u>GGSN PM Measurements:</u>

- General Indicators – provides detailed information about Packet Switched load metrics measured at dedicated data processors. It provides the following statistics:

CPU load due to Packet Switching: details the average and

maximum values
Total Throughput: details the average and maximum throughput value for the entire GTP set. Throughput values can be of Packet per second and Mbit per second
Packet Size: details the average and maximum packet size in bytes
Erroneous GTP packets: details the number of erroneous packet for both DL (GGSN to SGSN) and UL (SGSN to GGSN) traffic directions

- Session Management – provide detailed information about PDP Context. It provides the following statistics:

PDP Context Events: details the amount of, PDP create, update and delete messages received from SGSN. It also details each of these events per cause code
Active PDP Context: details the amount current active PDP Contexts. It also details the average and peak number of active PDP Context
PDP Context Duration: details the average and maximum active PDP context duration
Dynamic IP Address Usage: details the current number of allocated dynamic IP addresses. It also provides the average, maximum number and usage rate of dynamic IP addresses

- GTP Data per Access Point Network (APN) – provides detailed information about GTP transmitted data related with a specific APN. It provides the following statistics:

GTP Packets: details the number of sent and received packets during the period duration
GTP Data: details the sent and received data volume of during the period duration
GTP Error Free Bytes: details the sent and received error free data volume during the period duration

- Traffic QoS at GGSN Level – provides detailed information about transmitted traffic, detailed per QoS traffic class. It provides the following statistics:

Throughput: details the DL and UL throughput per QoS traffic class
DL Dropped Data: details the amount of dropped data in DL direction, detailed per QoS traffic class
Active PDP Contexts: details the number of active PDP Contexts per QoS traffic Class
Rejected PDP Context: details the number of rejected PDP

Context activation requests per QoS traffic Class, due to lack of resources

- Traffic QoS at APN Level – provides detailed information about transmitted traffic, related with a specific APN, detailed per QoS traffic class. It provides the following statistics:

Throughput: details the DL and UL throughput per QoS traffic class
DL Dropped Data: details the amount of dropped data in DL direction, detailed per QoS traffic class

- Service Specific – provides detailed information about transmitted traffic, related with a specific service (HTTP, WAP, FTP, P2P, etc.). It provides the following statistics:

Throughput: details the DL and UL throughput per Service
Dropped Data due to limited Bandwidth: details the amount of dropped data per Service

Call Detail Record

Call Detail Record is a file generated by the system for every call that is established and describes detailed billing information. Although originally the CDR was designed to describe call details for billing purposes, its information can be used to trace the call at the business level and retrieve service assurance relevant information.

This information complements the Performance Management information by extending the network behavior analysis to the service/subscriber scope, providing the ability to propose new analysis scenarios, e.g. to assess if network is accurate in the service delivery or which services are more suitable for that network considering its traffic model and user behavior.

The following fields were identified as being important from service assurance point-of-view:

- **recordOpeningTime** – this field provides information about the time when the CDR was created and thus the session creation time.
- **changeTime** – the time when this record was closed and therefore the session ended.
- **accessPointName** – this field indicates the type of external Packet Data Network (PDN) that was accessed by user during the session period.
- **ggsnAddress** – this field provides the control plane IP address of the serving GGSN.

- **sessionId** – this field provides the PDP context session identifier. Together with the **ggsnAddress** and **userLocationInformation**, it allows to identify the complete network data path.
- **servedIMSI** – this field provides International Mobile Subscriber Identity (IMSI) of the served party, the user.
- **servedPDPAddress** – this field provides the IP address related with the PDP context activated for the served IMSI.
- **accessType** – this field indicates which is the Radio Access Network type that served the connection in RAN side. This field can assume the following values:

    - 1 (GTP_RAT_TYPE_UTRAN): 3G (UMTS) RAN;
    - 2 (GTP_RAT_TYPE_GERAN): 2G (GSM(GPRS) / EDGE) RAN;
    - 3 (GTP_RAT_TYPE_WLAN): WLAN.
- **userLocationInformation** – this field provides information about the end user location during the all CDR life cycle. The location information consists of the following parts:
    - geographic location area
    - location area code (LAC)
    - cell identity (CI) /service area code (SAC)

<u>Deep Packet Inspection</u>

"Deep Packet Inspection" (DPI) is a computer networking term that refers to devices and technologies that inspect and take actions based on the contents of the packet (commonly called the "payload") rather than just the packet header." [18]

Deep Packet Inspection can be used for a wide range of applications and a common use can be described by an inspection point where packets pass and are inspected in order to search for protocol non-compliances, viruses, spam and intrusions. Although DPI wide spread use is related to security applications, it allows pre-configurations of analysis criteria in order to decide what actions to take on the packet. This can be used to extend its capabilities for other purposes, rather than security, like statistical collection. From this work perspective, the ability to collect IP based protocol statistics is the most relevant DPI functionality. It will enhance the service/session performance monitoring capabilities by providing means of collecting information about SIP, RTP and other protocols that can be considered relevant depending on the IP service that is under analysis scope.

<u>Session Initiation Protocol (SIP)</u>

SIP is a signaling protocol used to establish, manage and release sessions in an IP based network. SIP delivers session description information sent by the session initiator (caller) to all the other involved parties (callees). A SIP session can be of several forms raging from a simple two-way telephone call to a multimedia conference session. SIP was chosen by Internet Engineering Task Force (IETF) – RFC3261 – to be the standard for VoIP sessions control.

SIP is considered in the scope of this work as being very important to understand the session context from the control perspective. Thus, the following information is considered from [19]:

- Session name and purpose;
- Involved parties: Caller and Callees details (e-mail address, IP address, etc.);
- SIP Session Performance:

SIP Session Activation Delay – provides information about the duration of session establishment procedure
SIP Session Disconnect Delay – provides information about the duration of session deactivation procedure
SIP Duration Time – provides information about the total session duration
SIP Session Activation Success Rate – provides information about the probability of successful session establishment occurs
SIP Session Termination Rate – provides information about the probability of a normal session release occurs
SIP Active Drop Rate – provides information about the probability of an abnormal session release occurs
Hops per Request – provides the total number of hops in the SIP data path

<u>Real-Time Transport Control Protocol (RTP)</u>

"RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. RTP does not address resource reservation and does not guarantee quality-of-service for real-time services. The data transport is augmented by a control protocol (RTCP) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. RTP and RTCP are designed to be independent of the underlying transport and network layers." [20]

RTP Control Protocol main function is to provide, to all session involved parties, feedback about the quality of the transmitted data. RTCP control packets are periodically transmitted, using the same mechanisms as the RTP data packets, to all the session participants. The feedback provided by each of the session participants is related with flow and congestion control functions of other transport protocols, being very important for adaptive encoding and also to identify problems in data transmission path. This information can be used to identify, in case of transmission faults, if the source of the problem is local or global. These mechanisms provide, to service providers, the ability to use this information as an important source for monitoring the service performance.

According to [21], the following information can be retrieved from inspecting RTP and RTCP packets:

- Packet transmission performance provides information about:

the number of packets sent and packets lost, discarded packets
jitter and network round-trip delay
Mean Opinion Score (MOS), R-Value, Echo, Noise and Distortion Level
Burst/Gap Metrics like: Mean Burst Duration, Mean Burst Density and Mean Gap Density

3.4. Extraction, Transformation and Loading Process

"Unanticipated delays can make the data warehouse project appear to be a failure, but building the ETL process should not be an unanticipated delay. The data warehouse team usually knows that the ETL process consumes the majority of the time to build the data warehouse. The perception of delays can be avoided if the data warehouse sponsors are aware that the deployment of the data warehouse is dependent on the completion of the ETL process. The biggest risk to the timely completion of the ETL system comes from encountering unexpected data-quality problems." [22]

The Extraction, Transformation and Loading (ETL) process is a fundamental part of any data warehousing system, which remains true for the telecom OSS systems operation. ETL is crucial to data collection, conversion and population in PM database.

As it can be depicted from the name itself, ETL is a process that includes three major procedures: PM data collection (Extraction), data conversion from multiple diverse formats into a recognizable and unique OSS model format (Transformation), and loading the data into OSS database tables (Loading).

The Extraction phase is responsible for establishing communication between OSS system and with each and every statistical data source, configured in the managed network, and for the collection of the generated data to a designated folder within the OSS system. This process has to cope with the requirement of collecting enormous amounts of data, from hundreds of elements, within a very limited and short period of time.

The Transformation phase handles with the data stored in the collect performance measurement files, by converting this original data model to a single OSS meta-model format. This process is fundamental to the integration of multiple data formats, generated by the different statistical data sources, into a single, analysis optimized data model. The multiple data formats produced by the network can be due to different network technology elements and diverse monitoring purposes, such as session billing, session/service monitoring, etc.

The Loading process is simply related with all database storage procedures. This ETL process is very challenging due to the fact that it has to handle massive amount of data in a small period of time. The demand is to have the PM data available in the system's database, as quick as possible, and therefore, this process has to be highly optimized to achieve large performance. This is something to have in mind when designing a network object and information model.

3.5. Object Model

From Performance Management perspective, the Object Model purpose is to describe all the Managed Object (MO) in a Network, i.e., it is used to locally represent the network elements functionalities, its interfaces and relations with other elements. The MO is a network entity that can be not only monitored, but can also be managed via OSS system.

The UMTS Object Model used in this work represents, from and end-to-end perspective, each network element, its interfaces and relation with other elements. For the sake of simplicity, this OM is a simplified shorten version of real world implementation, focusing only on the objects that are relevant for this work goal.

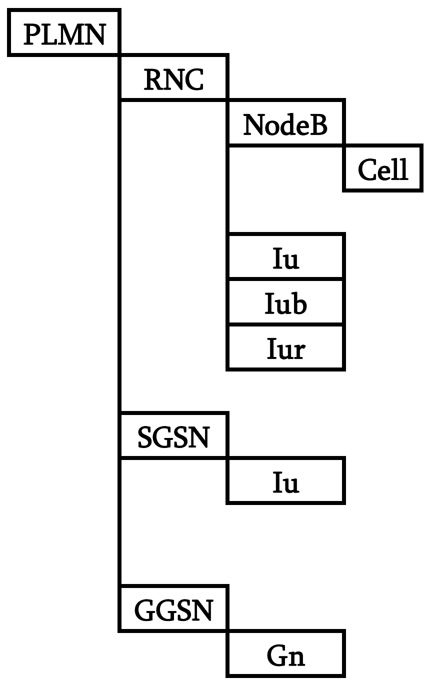


Figure 19 - UMTS Network Simplified Object Model

Public Land Mobile Network (PLMN) corresponds to the network operator infrastructure identification, which is unique in the entire world. From the PM point-of-view, it is used as the root object class that relates all the sub-networks and its network elements between each other. This object class provides means for creating an End-to-End performance monitoring abstraction layer, allowing creation of KPIs and Reports that relates information from several parts of the network.

RNC, SGSN and GGSN relationship is provided through the PLMN entity, and each one represents a vast set of logical functionalities. RNC is responsible for controlling the Radio Network resources, elements and functions. The RNC object class represents all this complexity by providing some important measurements and being an object parent of several other object classes, like NodeB, Cell, Iu, Iub and Iur manageable entities. RNC Object Class will typically be instantiated by one or two tens of objects in a UMTS network; depending on the operator's network dimension and traffic demand the number can be higher. The typical amount of NodeB objects is of several hundreds per RNC, once more depending on the network dimension. There are typically three cells per NodeB, which means that the number of cells per RNC can be in the order of a thousand. SGSN, GGSN and its interfaces are also represented. Each of these Object Classes will be instantiated by far less objects than the RAN side elements. SGSN number of objects is typically less than ten, since several RNCs connect to one SGSN. GGSN number of objects is even less than SGSN number, since a GGSN can connect more than one SGSN.

IP-based service elements and related metrics have an independent and generic Information Model. There are two fundamental reasons why the IP-based Service Layer should be modeled in an independent layer:

- The first reason is related with IP-based Services nature. IP layer is independent from relying transmission network layer, which allows applying the same concept defined in this work to other use cases. This means that no references to underlying network should exist in the mode definition;
- The second reason is related with the nature of PM data. While in network performance management approach the monitor focus is the network element performance, the Service layer objective is to provide network element performance, and also individual session detailed information, which means that the amount of records increases significantly.

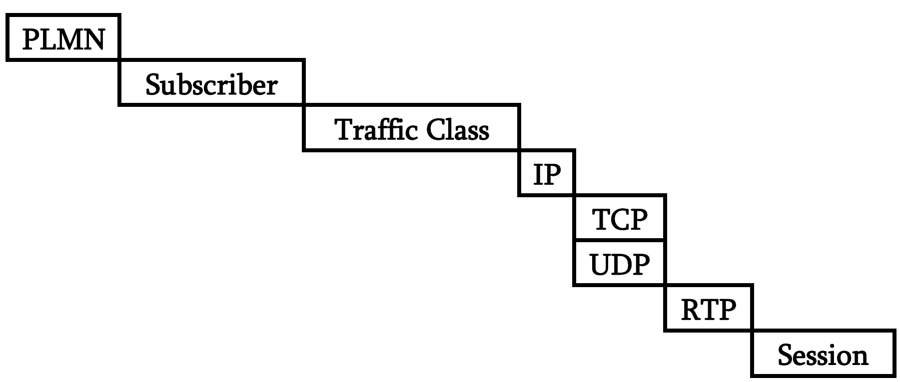


Figure 20 - IP-based Service Information Model

There is a business fundamental relation, from operator perspective, between the provided service and the network infrastructure. Operator requires that OSS systems provide the ability to bill, monitor and manage all kind of traffic and services delivered through its network. This work proposes an approach to enable such kind of feature, which is to relate the service specific information with the PLMN specific information. So, in order to enable cross-layer IP-based service monitoring capabilities, the corresponding service information model must be related with the PLMN object class. This also enables the capacity to report each element performance correlated with session performance specific information taken from the same instant.

The Subscriber entity must be directly related with PLMN object class, since this way it can be translated the business fundamental relation that exists between operator infrastructure and service user. There is also a logical explanation for this relation and has to due with the concept that all network activity description is an abstraction of the overall subscriber activity, since the total network activity is a sum of each subscriber activity, profile and behavior.

The remaining of the model describes the information hierarchy defined to describe each individual session, i.e., each session is related with a UMTS traffic class and related QoS parameters; in the other hand, for every single session there are collected IP, TCP or UDP and RTP metrics.

3.6. Key Performance Indicators

Key performance indicators (KPIs) are a set of selected indicators used for measuring the current network performance and trends. KPIs highlight the key factors of network monitoring and warn in time of potential problems. KPIs are also used for prioritizing the corrective actions.
A KPI can simply be equal to a Counter or to an arithmetic abstraction of counters that can be applied to monitor a certain part of the network, functionality or protocol.

ITU-T has defined in the standard E.800 some network monitoring KPI classes that are applied in this work:

- Accessibility – can be defined as the probability that a service can be obtained within specified tolerances and other given operating conditions when requested by the user.
- Integrity – can be defined as the degree to which a service is provided without excessive impairments, once obtained.
- Retainibility – can be defined as the probability that a service, once obtained, will continue to be provided under given conditions for a given time duration.
- Mobility – serves to monitor the performance mobility aspects (e.g., handover, location updates) and the degree of impact on network overall behavior.
- Usage – serves to evaluate the network usage and capacity performance (e.g., traffic volume, CPU usage, etc) and their impact on overall network behavior.

All the KPIs explained in this chapter are based on the earlier described measurements and counters. These KPIs were selected to provide all the necessary information about end-to-end network performance and ability to deliver IP-based services. All UTRAN related KPIs are a result of my daily work and were defined and proposed by me. The Core Network KPIs selection results from a research work on available defined KPIs. All the Service Layer KPIs were defined for the purpose of this work.

UMTS Network functionalities can be split using several different approaches. In this work the network analysis is done by dividing the network like in the following:

- Radio Access Network and Core Network division: these are two very distinct and independent parts of the network as already mentioned in Chapter 2.2;

- Control Plane and User Plane: analyzing the network from an end-to-end perspective, there is also the possibility to split into these two functional parts. Network's control plane is responsible for all the signaling between elements and protocol procedures, and data plane is responsible for the end-user data transmission;
- Cross-Layer Stack: this organization of network's functionalities allows abstract information about protocol layers, network elements and procedures into logical layers. This approach provides the flexibility to shape the performance information according to the analysis goal.

3.6.1. Network Session – Control Plane KPIs

The Network session control plane part is split into independent phases: Accessibility and Retainibility. The Accessibility part focuses on the ability to access the network and establish a call. The Retainibility part is mainly related with the network capacity to retain a call until it is normally released.

In this section it is presented a list of the most important network session control plane KPIs. The way these are presented follows this logic:

- First step is to list the higher layer KPIs, i.e., those related with the Packet Session layer;
- Follows the network service layer KPIs that are related with the several signaling phases that compounds a packet session procedure;
- The third step lists the network service KPIs detailed per failure causes.

Packet Call Accessibility related KPIs:

<u>Packet Call Setup Success Ratio</u> – provides information about the probability to establish successfully a connection with the network. This KPI combines the probability to establish a RRC connection, followed by a Radio Access Bearer, which means that both RAN and CN performance influence its result.

$$PCSSR = \frac{rrcEstabSucc[all]}{rrcEstabAtt[all]} * \frac{rabEstabTypeSucc[all]}{rabEstabTypeAtt[all]} * 100\%$$

Where *all* refers that all the PS connection types must be included in the formula calculation. PS connection type refers to the traffic class associated with the call establishment and can be one of the following:

- PS Interactive;
- PS Background;
- PS Streaming.

This KPI can be further detailed per traffic type, as follows:

$$PCSSR(NRT) = \sum i \frac{rrcEstabSucc[i, j]}{rrcEstabAtt[i, j]} * \frac{rabEstabTypeSucc[i, j]}{rabEstabTypeAtt[i, j]} * 100\%,$$

i refers to the traffic class.

i=0 -> instantiates the Interactive traffic class counter;

i=1 -> instantiates the Background traffic class counter.

$$PCSSR(RT) = \sum k \frac{rrcEstabSucc[k]}{rrcEstabAtt[k]} * \frac{rabEstabTypeSucc[k]}{rabEstabTypeAtt[k]} * 100\%$$

k refers to the traffic class.

k=3 -> instantiates the Streaming traffic class counter;

Packet Session Setup Success Ratio – this KPI monitors the RAN ability to establish a packet session. This KPI result is only influenced by RAN performance as it provides information about the channel allocation probability for PS traffic. The HS-DSCH, E-DCH and DCH successful allocations are compared to all the allocation attempts. This KPI can also be detailed per channel type, providing individually the HSDPA PSSR, HSUPA PSSR and R99 PSSR, which is important to assess each channel contribution for the overall result. Figure 21 shows a graphical representation of this KPI results taken from a real live network.

$$PSSSR = \sum i, j \frac{transChannelAllocSucc[i, j]}{transChannelAllocAtt[all]} * 100\%$$

i refers to the type of channel allocation and *j* to the traffic class.

For instance:

i=0, j=0 -> instantiates the HS-DSCH/E-DCH channel for Interactive traffic class attempt counter;

i=1, j=1 -> instantiates the HS-DSCH/DCH channel for Background traffic class attempt counter;

i=2, j=2 -> instantiates the DCH/DCH channel for Streaming traffic class attempt counter;

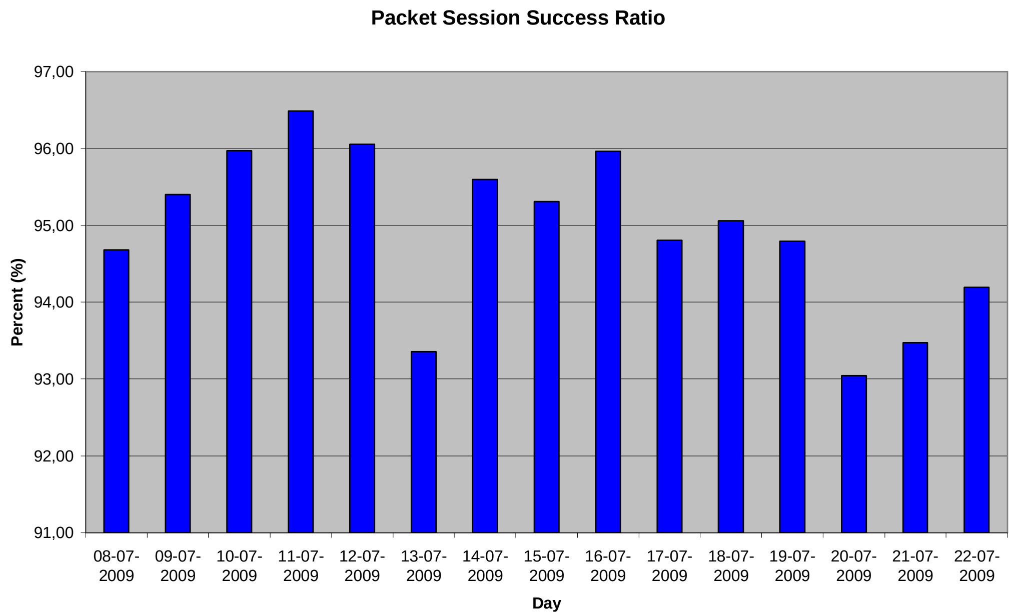


Figure 21 - Packet Session Setup Success Ratio daily evolution taken from a real live network

Packet Session Setup Failure Ratio detailed per Cause – this KPI group details the probability of call rejection due to a specific reason. It is used to identify the root cause of access problems. These KPIs can be influenced by causes originated both in RAN and CN, which are:

- Admission Control rejections, which indicates degradation of radio conditions mainly of DL codes, power and UL interference;
- Node B rejections mainly related with lack of hardware resources;
- DMCU rejections indicates the lack of RNC User Plane resources;
- Transport rejections mainly related with Iub congestion;
- Iu rejections are related with CN rejection;
- UE rejections related with user equipment problems;
- Other rejections reasons that are not contemplated in the earlier counters.

Figure 22 shows a graphical representation of this KPI set, with results taken from a real live network.

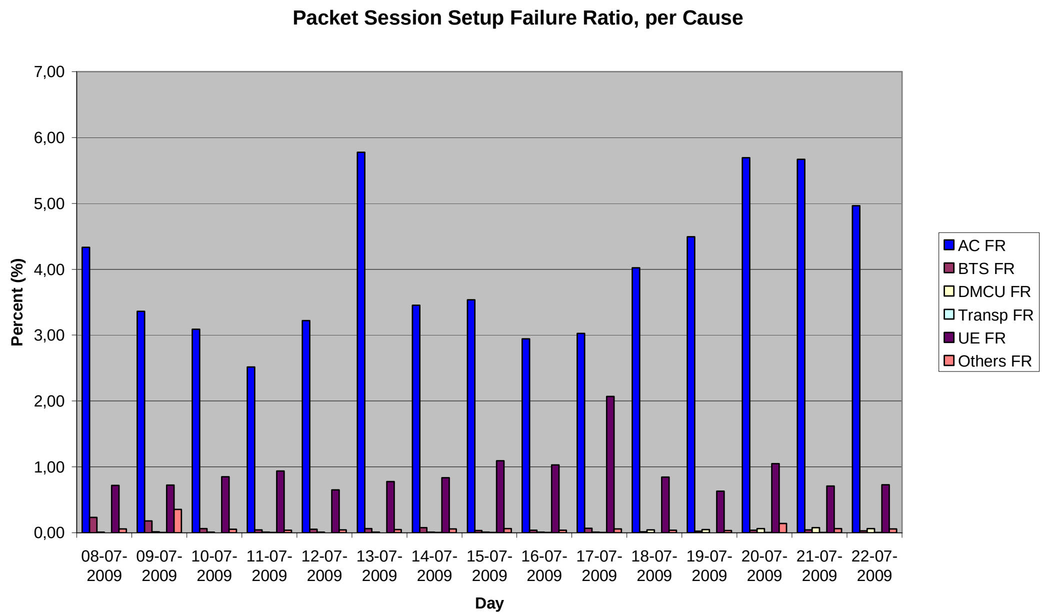


Figure 22 - Packet Session Setup Failure Ratio detailed per cause, daily evolution taken from a real live network

$$PSSSR(k) = \sum i, j, k \frac{transChannelAllocFailure[i, j, k]}{transChannelAllocAtt[all]} * 100\%$$

i refers to the type of channel allocation, *j* to the traffic class and *k* to the failure cause.

For instance:

i=0, j=0, k=0 -> instantiates the failure counter for HS-DSCH/E-DCH channel allocation, Interactive traffic class and Admission Control failure reason;

i=1, j=1, k=1 -> instantiates the failure counter for HS-DSCH/DCH channel allocation, Background traffic class and Node B HW failure reason;

i=2, j=2, k=2 -> instantiates the failure counter for DCH/DCH channel allocation, Streaming traffic class and lack of DMCU resources failure reason;

Packet Call Retainibility related KPIs:

<u>Packet Session Complete Success Ratio</u> – provides information about the network ability to retain a call until the successful call release. Combined with the packet call setup success ratio, this is one of the most important KPIs because it impacts directly the User Experience. Figure 23 shows a graphical representation of this KPI, using results from real live network information.

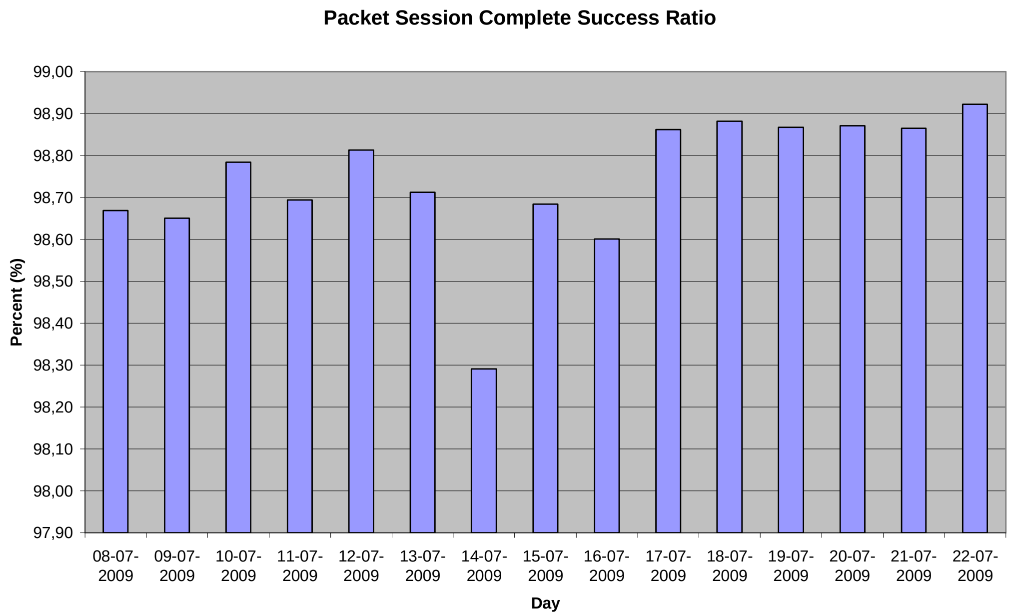


Figure 23 - Packet Session Complete Success Ratio daily evolution taken from a real live network

$$PSCSR = \left(1 - \frac{transChannelAllocDrop[all]}{transChannelActAlloc[all]}\right) * 100\%$$

Where *all* refers that all the Channel allocation, traffic class and failure cause combinations. This KPI first calculates the actual active channel allocation drops rate, and then subtracts that value to the unit in order to obtain the Packet Sessions that were normally ended.

<u>Packet Session Drop Ratio per Cause</u> – these KPIs provide detailed information about the reasons why a call was drop. The drop call reasons can be:

- Radio Link synchronization failure, indicate the source of the problem was the radio interface;
- Other failure than Radio failure,

$$PSCFR(k) = \sum i, j, k \frac{transChannelAllocDrop[i, j, k]}{transChannelActAlloc[all]} * 100\%$$

Figure 24 show a graphical representation of this KPI, using results retrieved from a real live network.

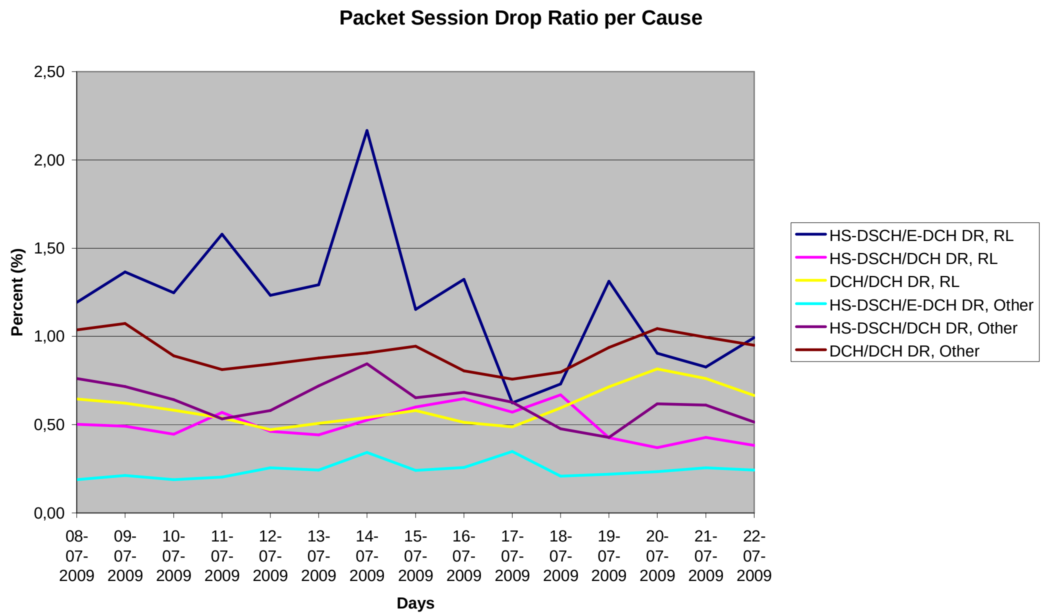


Figure 24 - Packet Session Drop Ratio, detailed per affected channel and release cause, daily evolution taken from a real live network

Service Accessibility related KPIs:

RRC Establishment Success Ratio – this KPI provides information about the probability of RRC connection is established successfully. It is an important KPI that is used to determine if RRC connection procedure is affecting negatively the overall packet call setup process.

$$\mathrm{RrcEstabSR} = \frac{\mathrm{rrcEstabSucc[all]}}{\mathrm{rrcEstabAtt[all]}} * 100\%$$

Where *all* refers that all the PS connection types must be included in the formula calculation. PS connection type refers to the traffic class associated with the call establishment and can be one of the following:

- PS Interactive;
- PS Background;
- PS Streaming.

RRC Establishment Failure Ratio detailed per cause – this KPI group provides information about the probability of RRC connection fails to establish successfully due to all the possible failure reasons. It is important to analyze this KPI group if the overall RRC connection establishment procedure is affecting negatively the overall packet call setup process.

$$\mathrm{RrcEstabFR(i)} = \frac{\mathrm{rrcEstabSucc[i]}}{\mathrm{rrcEstabAtt[all]}} * 100\%$$

Where *i* refer to the individual failure cause, which can be the following:

- Handover Control rejection;
- Admission Control rejection;
- Node B HW failure;
- Iub transport failure;
- RNC internal failure;
- Lack of UP RNC resources.

RAB Establishment Success Ratio – this KPI provides information about the probability of RAB establishment occurring successfully. It is an important KPI that is used to determine if RAB establishment procedure is affecting negatively the overall packet call setup process.

$$RabEstabTypeSR = \frac{rabEstabTypeSucc[all]}{rabEstabTypeAtt[all]} * 100\%$$

Where *all* refers that all the PS connection types must be included in formula calculation. PS connection type refers to the traffic class associated with the call establishment and can be one of the following:

- PS Interactive;
- PS Background;
- PS Streaming.

PDP Context Activation Success Ratio – this KPI focus on CN session management part and the ability to create a PDP context. This is a fundamental step of Packet Session setup procedure as it can prevent the session context setup and, for this reason, this KPI must be monitored when poor results are found in packet session setup KPI.

$$IuPDPContextActSR = \frac{IuPDPContextActSucc}{IuPDPContextActSucc} * 100\%$$

Service Retainibility related KPIs:

RRC Connection completion Success Ratio – this KPI shows the probability that an active RRC connection is normally terminated. It provides information about the network ability to retain successfully a RRC connection until the call is ended normally. All the RRC connection released due to a failure will impact the active call, leaving to a Call Drop.

$$\mathrm{RrcActConnCompSR} = \frac{\mathrm{rrcActConnComp}}{\mathrm{rrcActConn}} * 100\%$$

RRC Connection Drop Ratio per Cause – this KPI group details the RRC Drop probability per each root cause. In case RRC connection completion ratio presents a low value, these KPIs must be monitored to assess what is the reason of such problem.

$$\text{RrcActConnDropRatio(i)} = \frac{\text{rrcActConnDrop[i]}}{\text{rrcActConn}} * 100\%$$

Where *i* refers to the individual failure cause, which can be the following:

- Radio interface failure;
- Node B HW failure;
- User Equipment failure;
- Iu interface failure;
- Iur interface failure;
- RNC internal failure;
- Security Mode Control rejections.

RAB Complete Success Ratio – this KPI focus on the RAB Active phase of a call and provides information about the probability of an Active RAB be released normally. Figure 25 shows a graphical representation using real live network data.

$$\text{RabActConnCompSR} = \frac{\text{rabActConnComp}}{\text{rabActConn}} * 100\%$$

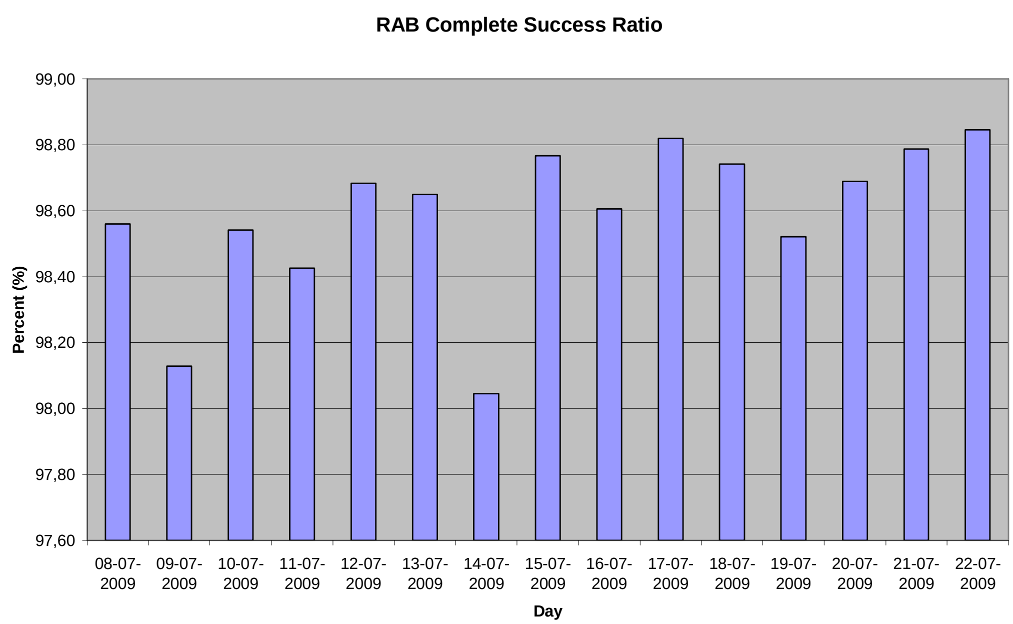


Figure 25 - RAB Complete SR, daily evolution taken from a real live network

RAB Drop Ratio per Cause – this KPI group shows the RAB Drop probability detailed per failure cause. In case RAB Complete SR presents a low value, these KPIs must be monitored to assess what is the root cause of such problem.

$$\mathrm{RabDropRatio(i)} = \frac{\mathrm{rabActConnDrop[i]}}{\mathrm{rabActConn}} * 100\%$$

Where *i* refers to the individual failure cause, which can be the following:

- Radio interface failure;
- Node B HW failure;
- User Equipment failure;
- Iu interface failure;
- Iur interface failure;
- RNC internal failure.

<u>PDP Context Normal Deactivation Ratio</u> – this KPI focus on the PDP Context active phase of a packet session and provides information about the probability of a normal release.

$$IuPDPContextNormalDeactivatioRatio[all] = \frac{IuPDPContextNormal\,\mathrm{Re}\,leases[all]}{ActiveIuPDPContext} * 100\%$$

Where *all* refers to the sum of the individual normal deactivation reason, which can be the following:

- UE deactivation;
- SGSN deactivation;
- GGSN deactivation.

3.6.2. Network Session – User Plane KPIs

The Network session user plane part is split into independent data path: GTP and PDCP path. The GTP data path is formed by the CN elements (GGSN and SGSN) and the RNC. These elements are responsible for the handling of GTP packets transmission. The GTP path is established in the RAN side between RNC and UE. The elements that are part of PDCP path are: RNC, Node B, Iub, Cell and UE.

GTP data path related KPIs:

<u>GTP-u Throughput</u> – this KPI group provides information about the GTP data throughput that can be measured both in DL and UL traffic directions. These KPIs are applicable to all the elements part of GTP path: GGSN, SGSN and RNC. Figure 26 shows a graphical representation of this KPI using results taken from a real live network.

$$IuPS\_GTPu\_Thp(i) = \frac{8 * GTPu\_Data\_Volume[i]}{GranularityPeriod * 60 * 1000000} (Mbps)$$

Where *i* refers to traffic direction:

i=0 -> instantiates the DL traffic direction counter;

i=1 -> instantiates the UL traffic direction counter.

GranularityPeriod provides the measurement period in minutes, and therefore there is the need to convert its value to seconds by multiplying by a factor of 60 seconds.

GTPu_Data_ Volume[i] provides the amount of Bytes transmitted over the Iu interface between RNC, SGSN, and therefore, it must be multiplied by a factor of 8 to convert its value for bits.

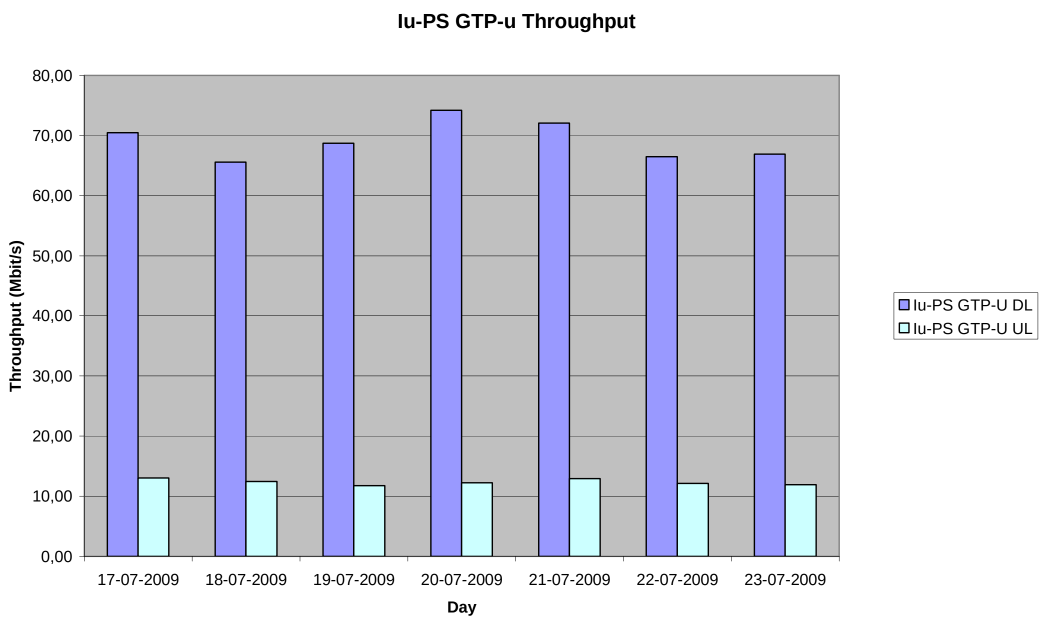


Figure 26 - RNC Iu-PS GTP-u Throughput, daily evolution from a real live network

GTP-u Traffic Volume – this KPI group provides information about the GTP data volume that can be measured both in DL and UL traffic directions. These KPIs are applicable to all the elements part of GTP path: GGSN, SGSN and RNC.

$$IuPS\_GTPu\_DataVol(i) = \frac{GTPu\_Data\_Volume[i]}{1000000} (MB)$$

Where *i* refers to traffic direction:

i=0 -> instantiates the DL traffic direction counter;

i=1 -> instantiates the UL traffic direction counter.

GTPu_Data_ Volume[i] provides the amount of Bytes transmitted over the Iu interface between RNC and SGSN.

GTP-u Throughput detailed per traffic/bearer class – this KPI group provides information about the GTP throughput that can be measured both in DL and UL traffic directions, adding additional detail about the corresponding traffic and bearer priority class. These KPIs are applicable to SGSN element in GTP data path.

$$\text{IuPS\_GTPu\_Thp}(i, j, k) = \frac{8 * \text{GTPu\_Data\_Volume}[i, j, k]}{\text{GranularityPeriod} * 60 * 1000000} \text{(Mbps)}$$

Where *i* refers to the traffic class, *j* refers to the bearer type and *k* to the traffic direction (DL or UL).

GTP-u Data Volume detailed per traffic/bearer class – this KPI group provides detailed information about the GTP-u traffic volume transmitted for each traffic class, bearer type association both in DL and UL directions. These KPIs are applicable to SGSN element in GTP data path. Figure 27 shows a graphical representation of this KPI set using results retrieved from a real live network.

$$\text{IuPS\_GTPu\_DataVol}(i, j, k) = \frac{\text{GTPu\_Data\_Volume}[i, j, k]}{1000000} \text{(MB)}$$

Where *i* refers to the traffic class, *j* refers to the bearer type and *k* to the traffic direction (DL or UL).

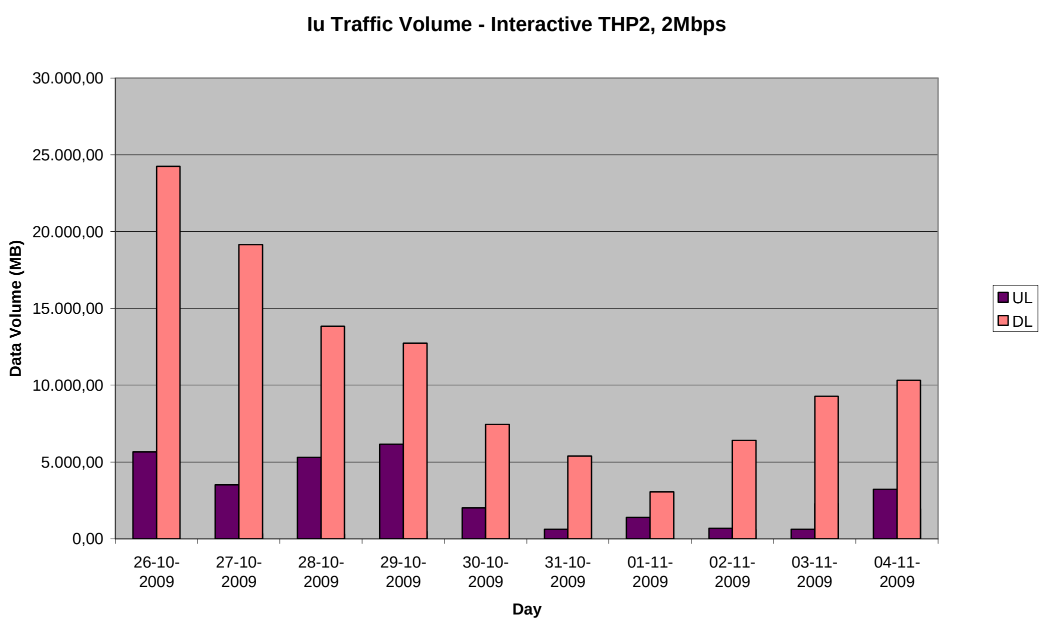


Figure 27 - SGSN Iu-PS GTP-u Data Volume detailed per traffic/bearer class, daily evolution from a real live network

GTP-u Data Discard Ratio – this KPI group provides information about the probability of GTP-u data is discarded during transmission. It is used to monitor the GTP-u path transmission performance and assess if eventually it is affecting the service delivery performance. These KPIs are applicable to all elements in the data path: RNC, SGSN and GGSN.

$$\text{IuPS\_GTPu\_DataDiscardRatio(i)} = \frac{\text{GTPuDiscardData}[i]}{\text{GTPu\_Data\_Volume}[i]} * 100\ (\%)$$

Where *i* refers the traffic direction, in DL or UL.

GTP-u Packet Drop Ratio – this KPI group provides information about the probability of GTP-u packet is drop during transmission. It is used to monitor the GTP-u path transmission performance and assess if, eventually, it is affecting the service delivery performance. These KPIs are applicable to all elements in the data path: RNC, SGSN and GGSN.

$$\text{IuPS\_GTPu\_PacketDropRatio(i)} = \frac{\text{GTPuDroppedPacket}[i]}{\text{GTPuTransmittedPackets}[i]} * 100\ (\%)$$

Where *i* refers the traffic direction, in DL or UL.

GTP-u Buffer Usage Ratio – this KPI provides information about the GTP-u buffer utilization ratio. It is useful to check this KPI if packet drop ratio is high, to check if probable cause is buffer congestion. This KPI is applicable to all elements in the data path: RNC, SGSN and GGSN.

$$\text{IuPS\_GTPuBufferUsageRatio} = \text{GTPuBufferUtilizationShare}\ (\%)$$

PDCP data path related KPIs:

PDCP PDU Data Volume – this KPI group provides information about the amount of transmitted data on the PDCP layer. These KPIs present the values detailed per traffic direction (DL or UL). These KPIs are applicable for RNC only.

$$\text{PDCP\_DataVol(i)} = \frac{\text{PDCP\_Data\_Volume}[i]}{1000000}\ \text{(MB)}$$

Where *i* refers the traffic direction, in DL or UL.

PDCP PDU Drop Ratio – this KPI group provides information about the probability of a PDCP PDU is dropped. These KPIs present the values detailed per traffic direction (DL or UL). These KPIs are applicable for RNC only.

$$\mathrm{PDCP\_PDU\_DropRatio(i)} = \frac{\mathrm{PDCPDroppedPDU}[\mathrm{i}]}{\mathrm{PDCPTransmittedPackets[i]}} * 100\ (\%)$$

Where *i* refers the traffic direction, in DL or UL.

PDCP Buffer Occupancy Ratio – this KPI provides information about the RNC average buffer capacity allocation. This KPI applies only for DL traffic. This KPI is applicable for RNC only.

$$\mathrm{P}\mathit{DCPBufferOccupRatio} = \mathrm{P}\mathit{DCPBufferAllocatedCapacity}\ (\%)$$

PDCP Transfer Delay – this KPI provides details about the average PDCP SDU transmission delay, that is the time difference between the arrival time of the first packet of a SDU received from the upper layer (RRC or PDCP), and the arriving time of the last PDU containing data from that SDU. This KPI is applicable for RNC only.

$$\mathrm{P}\mathit{DCP\_TransferDelay} = \mathrm{P}\mathit{DCPTransferDelay}\ (\mathrm{ms})$$

Amount of transmitted MAC PDU Data – provides the amount of data transmitted at the MAC layer. These KPIs detail the traffic direction, DL or UL. This KPI is applicable for RNC, Node B and Cell.

$$\mathrm{MAC\_DataVol(i)} = \frac{\mathrm{MAC\_Data\_Volume[i]}}{1000000}\ (\mathrm{MB})$$

Where *i* refers the traffic direction, in DL or UL.

MAC PDU Drop Ratio – this KPI group provides information about the probability of a MAC PDU is dropped during its transmission. These KPIs present the values detailed per traffic direction (DL or UL). These KPI are applicable for RNC, Node B and Cell.

$$\mathrm{MAC\_PDU\_DropRatio(i)} = \frac{\mathrm{MACDroppedPDU}[\mathrm{i}]}{\mathrm{MACTransmittedPackets[i]}} * 100\ (\%)$$

Where i refers the traffic direction, in DL or UL.

Amount of transmitted FP frames – provides the amount of Frame Protocol frames transmitted over Iub. These KPIs detail the traffic direction, DL or UL. These KPIs are applicable for RNC and Node B.

$$\text{AmountTransFPframes(i)} = \textit{Amount}\text{FP\_PDUs[i]}\,\text{(Nr)}$$

Where i refers the traffic direction, in DL or UL.

FP Drop Frames detail per channel – this KPI group provides information about the probability of a FP frame is dropped during its transmission. These KPIs present the values detailed per related transport channel, HS-DSCH, E-DCH or DCH (in the case of being the HS-DSCH return channel). These KPIs are applicable for RNC and Node B.

$$\text{FP\_FRame\_DropRatio(i)} = \frac{\text{FPDroppedFrame}[\text{i}]}{\text{FPTransmittedFrames[i]}} * 100\,(\%)$$

Where i refers to the radio transport channels associated with the transmitted FP frames.

i=0 -> instantiates the HS-DSCH transport channel counters;
i=1 -> instantiates the E-DCH transport channel counters;
i=2 -> instantiates the DCH transport channel counters;

MAC-hs Data Volume – provides the amount of correctly received MAC-hs data volume at the Node B. This KPI is applicable for Node only.

$$\text{HS - DSCH\_DataVol} = \frac{(\text{RecMAChsBits - DiscMAChsBits})}{8 * 1000000}\,\text{(MB)}$$

HSDPA active Cell throughput – provides the average active HS-DSCH MAC-d throughput, calculated as the HSDPA MAC-d throughput at BTS divided by the active HS-DSCH transmission time. The active time refers to the scheduled TTIs, i.e. when sending data. This KPI is applicable Cell only. Figure 28 shows a graphical representation of this KPI using results from a real live network.

$$\text{HS - DSCH\_ActCell\_Thp} = \frac{(\text{RecMAChsBits - DiscMAChsBits}) * 500}{\text{ActiveTransmissionTTI} * 1000000}\,\text{(Mbps)}$$

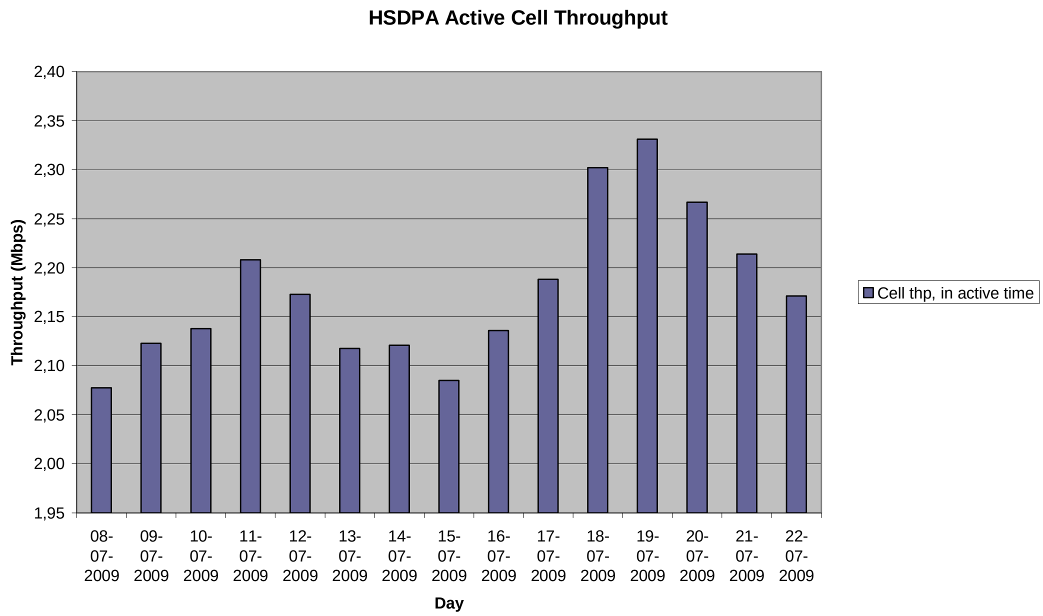


Figure 28 - Average HSDPA Active Cell Throughput, daily evolution taken from a real live network

HSDPA throughput per session – This KPI provides information about the average HSDPA throughput per Session. This KPI is applicable for Cell.

$$\mathrm{HS\text{-}DSCH\_Session\_Thp} = \frac{(\mathrm{HS\text{-}DSCH\_ActCell\_Thp})}{\mathrm{AverageHSDPAUsers} * \mathrm{ChannelUsageRatio}} \ (\mathrm{Mbps})$$

MAC-hs Discarded PDU – this KPI provides information about the amount of discarded MAC-hs PDUs. This KPI is applicable for Node B and Cell.

$$MAChsDiscardPDU = \mathrm{DiscMAChsPDU}\ (\mathrm{Nr})$$

MAC-hs efficiency – this KPI provides information about the HSDPA Successful transmission ratio, this is the ratio between successful received MAC-hs PDU and the total amount of transmitted MAC-hs PDUs. This KPI is applicable for Cell.

$$MAChsEfficiency = \frac{\mathrm{MAChsTransPDUSucc}}{\mathrm{MAChsTransPDU}} * 100\ (\%)$$

MAC-hs BLER per Cause – this KPI provides information about the HS-DSCH Block Error Rate detailed per error cause. This KPI is applicable for Cell.

$$MAChsBLER(i) = \frac{\mathrm{DiscMAChsPDU}(\mathrm{i})}{\mathrm{MAChsTransPDU}} * 100\ (\%)$$

Where *i* refers to the error cause:

i=0 -> instantiates the Maximum Number of Retransmissions counter;

i=1 -> instantiates the T1 Timer counter;

i=2 -> instantiates the Other Reason counter.

MAC-es Data Volume – provides the amount of transmitted MAC-es data volume over the Cell. This KPI is applicable for Node B and Cell.

$$\text{E - DCH\_DataVol} = \frac{(\text{EDCH\_DataVol})}{1000000}\,(\text{MB})$$

HSUPA active Cell throughput – provides the average active E-DCH MAC-es throughput, calculated as the total amount of MAC-es transmitted data at Cell level divided by the active E-DCH transmission time. The active time refers to the scheduled TTIs, i.e. when sending data. This KPI is applicable for Cell.

$$\text{EDCH\_ActCell\_Thp} = \frac{\text{EDCHtransmData} * 8}{\text{ActiveTransmissionTTI} * 1000}\,(\text{Mbps})$$

HSUPA throughput per session – This KPI provides information about the average HSUPA throughput per Session. This KPI includes retransmitted data. This KPI is applicable for Cell.

$$\text{EDCH\_Session\_Thp} = \frac{\text{EDCHtransmData} * 8}{\text{EDCHTransmTime}}\,(\text{Mbps})$$

MAC-es Discarded Frames – this KPI provides information about the amount of discarded MAC-es frames. This KPI is applicable for RNC and Cell.

$$MACesDiscardFrames = \text{DiscMACesFrames}\,(\text{Nr})$$

MAC-es efficiency – this KPI provides information about the HSUPA Successful transmission ratio, this is the ratio between successful transmitted MAC-es frames and the total amount of transmitted MAC-es frames. This KPI value decreases with the need to increase the number of retransmission, for instance due to radio conditions degradation. This KPI is applicable for RNC and Cell.

$$MACesEfficiency = \frac{\text{MACesTransFramesSucc}}{\text{MACesTransFrames}} * 100\,(\%)$$

MAC-es BLER per Cause – this KPI provides information about the E-DCH Block Error Rate detailed per error cause. This KPI is applicable for Cell.

$$MACesBLER(i) = \frac{\mathrm{DiscMACesPDU(i)}}{\mathrm{MACesTransPDU}} * 100\ (\%)$$

Where *i* refers to the error cause:

i=0 -> instantiates the HARQ failure counter;
i=1 -> instantiates the Buffer Overflow;
i=2 -> instantiates the Other Reason counter.

3.6.3. Service Session – Service Performance Indicators

The KPIs described in this session can also be found in [23].

The Service Performance Indicator (SPI) is mainly affected by the on-path links, nodes performance and the End-to-End conditions. It shows the overall performance of a network according to the quality of the contained service delivery.
The Revenue Factor (RF) has a direct implication on the evaluation of the network adequacy to the services being provided. The under-performance of voice should reflect a larger impact on the Network Adequacy Indicator (NAI) to the services being delivered since it is a very profitable service.

Network Adequacy Indicator – this KPI provides information about the ability of a specific network to support the delivery of a given service. This formula can be applied for any network and service combination.
It is composed by a Revenue Factor (RF) that is an operator configurable factor that translates the revenue importance of a specific service in the overall operator business.

$$NAI(i) = \frac{\sum_{i}^{TotalServices} \mathrm{RF(i)} * SPI(i)}{\mathrm{RF(i)}}$$

Where *i* refers to the type of service instance, for instance:

- VoIP;
- Video;
- Gaming;
- IPTV;
- Browsing;
- etc.

Service Performance Indicator – this KPI provides a weighted average between, the importance of a given KPI to a certain service, and the rating that the same KPI achieves. The SPI depends on the type of service, for instance when considering Voice, different parameters are addressed when compared to Video, as well as different relevance.

$$SPI(i) = \frac{\sum_{i}^{AllKPIs} k_i(KPI) \times KPRV_i(KPI)}{\mathrm{TK}(\mathrm{i})}$$

Where *i* refers to the type of service instance, like the following:

- VoIP;
- Video;
- Gaming;
- IPTV;
- Browsing;
- etc.

The Key Performance Rating Value (*KPRV*) reflects the importance of each KPI to the overall service performance calculus. $K_i(KPI)$ is the rating value according to the KPI performance. Table 1 illustrates a proposal of KPRV for several KPIs and Service combinations.

$\mathrm{TK}_{(i)}$ provides information about the overall sum, for each service, of all identified *KPRV*. $\mathrm{TK}_{(i)}$ is given by the following formula:

$$TK(i) = \sum KPRV(i)$$

Where *i* refers to type of service instance, like the following:

- VoIP;
- Video;
- Gaming;
- IPTV;
- Browsing;
- etc.

KPI:	KPRV By Type Of Service		
	Voice:	Video:	Browsing:
E2EDelay	5	2	3
Reliability	4	4	3
Discard Rate	3	3	2
Jitter	3	2	1
Waiting Time	2	1	2
Bit Rates	1	5	2
TK	18	17	13

Table 1 - KPRV value proposal

The KPRV establishes a weighted relation between each Service and KPI pair. The analysis of Table 1 shows that the importance of a specific KPI depends on the monitored service. For instance, the E2E Delay performance has more impact on Voice services then for Video and this fact is reflected in the KPI classified weight that is higher for Voice than for Video, and also is the KPI with higher weighted value within Voice classification. In the other hand, the most important factor impacting Video performance is the network capability to provide the required Bit Rate; this is reflected in the highest weighted value for Video classification. In the case of available Bit Rate KPI, for Voice service, it is the least important factor to be considered in overall performance evaluation.

Table 2 defines the service rating value (R Value) according to defined threshold for each KPI.

R	Delay (ms)	Jitter (ms)	Discard Rate (%)	Reliability (%)	WT (s)
5	<50	<0.1xDelay	<0.001	>99.999	<0.01
4	[50;100]	[0.1D;0.5D]	[0.001;0.01]	[99.99;99.998]	[0.001;0.1]
3	[100;150]	[0.5D;0.8D]	[0.01;0.1]	[99.9;99.999]	[0.1;1]
2	[150;300]	[0.8D;D]	[0.1;1]	[99.9;99]	[1;10]
1	[300;600]	[D;2D]	[1;10]	[90;99]	[10;100]
0	> 600	>2D	>10	<90	>100

Table 2 – Rating Value according to service behavior defined thresholds

Table 2 establishes, for each Service KPI, a rating value R according to the threshold rank defined to classify the network measured performance. For instance, in the case the network measured Delay is lower than 50 (ms), the rating value is equal to 5 and 0 if the measured result is higher than 600 (ms). In the other hand, a network Reliability performance value higher than 99.999 (%) corresponds to a rating value R equal to 5, in turn a measured result lower than 90 (%) corresponds to a R value equal to 0.

Applying to the defined SPI formula, the values defined in table 1 for KPRV and in table 2 for R value, it results in a final formula of the following form:

$$(SPI_{voice}) = (E2EDKPRV_1 \times R_1 \quad \cdots \quad BRKPRV_1 \times R_m)/\begin{pmatrix} TK_{voice} \\ \vdots \\ TK_{voice} \end{pmatrix}$$

In Section 3.7 the Service Performance Indicator Analysis reporting use case provides an example of this KPI classification usage.

3.7.Reports

A report is a logical association of KPIs and PIs that aim to provide means to take conclusions in the end of the analysis. The reports proposed in this work extend the current available reports, by combining data that not only describes the end-to-end network behavior, but also relates the network layer statistics with the IP-based Service layer statistics.

IP-based Service Report Use Cases

Packet Loss Analysis

The first reporting use case is designed to provide an analysis flow for IP-based Service Monitoring and detect problems related with service layer only, i.e., problems that are not originated by network performance degradation. This report group is called "Service Awareness".

Please note that the data shown in the following reports is not real and was produced for this Use Case demonstration.

The first report begins to detail the session context in a dashboard fashion. The following figure shows how the initial dashboard of "Session Description Report" is organized.

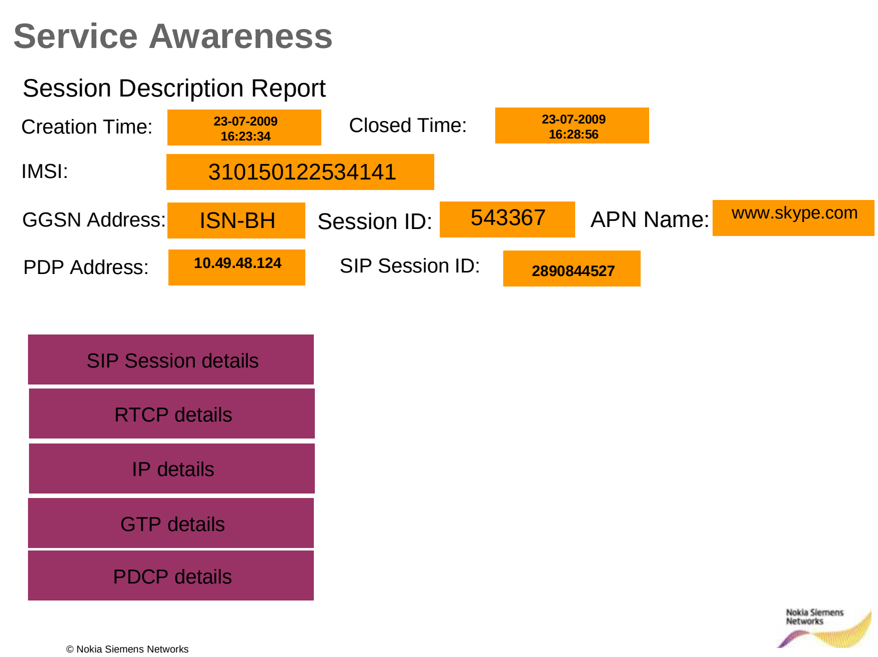


Figure 29 - Service Awareness – Session Description Report, Initial Dashboard

User can click on one of the following buttons, and browse for more details:

- SIP Session details;
- RTCP details;
- IP details;
- GTP details;
- PDCP details.

This use case is defined to analyze sessions that were identified as problematic. Therefore, it is assumed that the Performance Engineer will analyze the performance of each one in order to detect where the problem resides.

The “SIP Session Details” dashboard presents the VoIP call information that is under analysis, and additional SIP performance KPIs that provide detailed information about the server performance, mainly:

- Average SIP session initiation time;
- SIP session Activation Success Ratio;
- SIP session Termination Ratio Vs SIP session Drop Ratio.

For each monitored event, the information shown corresponds to a time range starting before the event occurred and ending after it. For instance, the “Average SIP session initiation time” graphic shows the server performance on a time interval that includes the call initiation event occurrence instant.

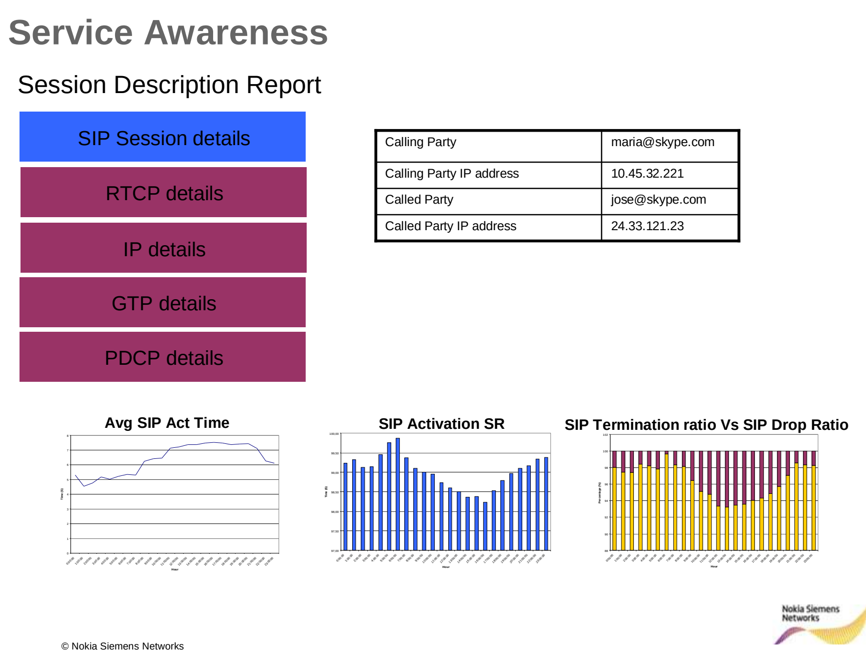


Figure 30 - Service Awareness – Session Description Report, SIP Dashboard

The next logical step is to analyze the RTCP monitoring layer, since it provides more detailed information about the session performance. The “RTP session Description Report” provides a set of KPIs and graphics that allow analyzing in detail the session performance for all the participants. The first two graphics

show the relation between the amount of lost packets when compared with the amount of sent packets. The first graph details the absolute values of sent and lost packets, and the second one details the actual relative “Packet Drop Ratio” value. The third graphic provides information about the average session End-to-End Delay and average Jitter. In the other hand, the fourth and fifth graphics provide information related with the user experienced sound quality detailed by the “Voice Signal Power”, “Noise Level” and “Distortion Level”.
The detailed analysis of this dashboard would indicate that the session was affected by the “Packet Loss Ratio” effect that, for one of the participants, is frequently above 2% along the session duration, which affects significantly the MOS calculated for this session, which also can be seen in the RTCP dashboard.
So, this would imply to analyze the next lower layer (IP layer) performance in order to assess where the problem source is.

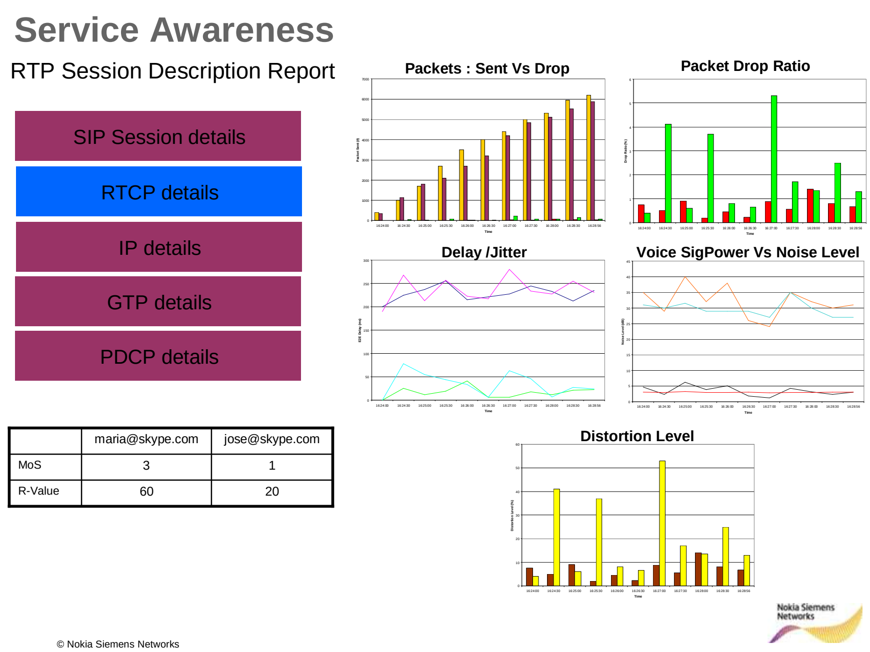


Figure 31 - Service Awareness– Session Description Report, RTP Dashboard

The analysis of the “IP session Description Report” dashboard enables to conclude that the IP layer shows the same pattern behavior as shown at the RTP layer. This allows concluding that the problem source is due to any specific RTP layer factor.

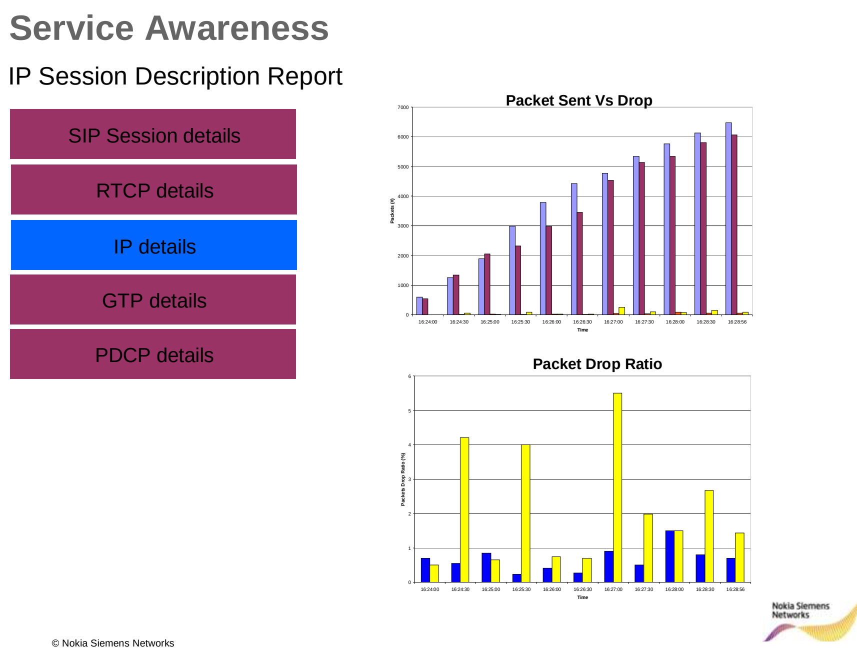


Figure 32 - Service Awareness– Session Description Report, IP Dashboard

The next logical step to take is to continue detailing the analysis towards the underlying network layers, to detect where the problem root cause is.

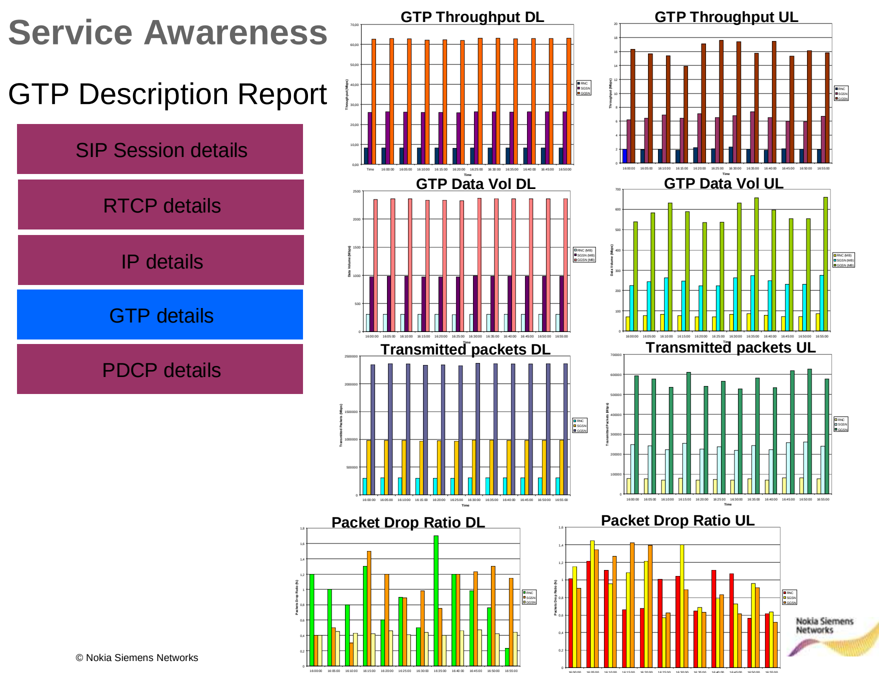
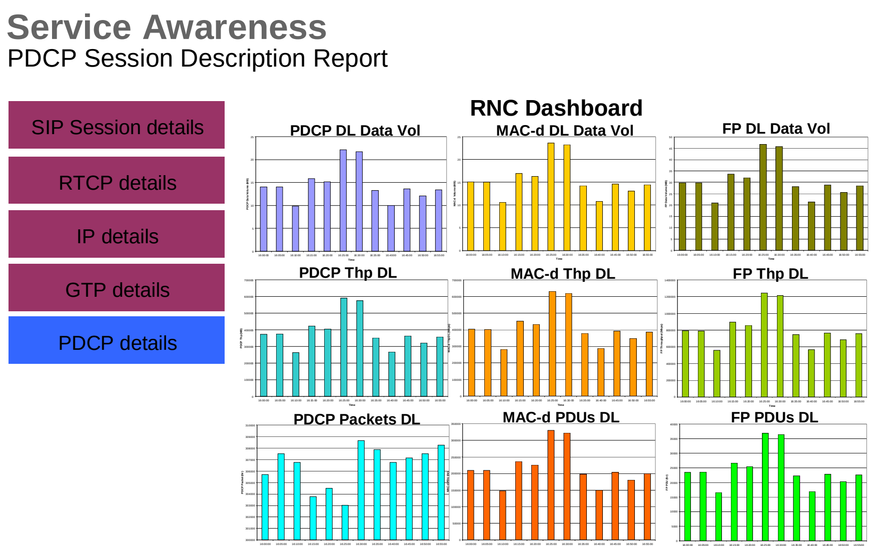


Figure 33 - Service Awareness– Session Description Report, GTP Dashboard

The next layer is GTP-u monitoring layer represented by the “GTP Description Report”. This report focus on the GTP-u interfaces usage and efficiency, providing detailed information, for both DL and UL traffic direction, about the

GTP-u Throughput, GTP-u Data Volume, GTP-u Transmitted Packets and GTP-u Packet Drop Ratio.
From the analysis flow described above, this session experienced some packet loss related problems that were identified both in RTP and IP layer. So, it is important to assess if the same pattern occurs at the GTP-u Layer.
After analyzing the proposed report, one would confirm that the Packet Drop Ratio, both for DL and UL, is below the 2% limit, which let one conclude that at least at the CN user plane side there is no apparent problem.
This conclusion leads the analysis to the last step, the PDCP layer report.

The PDCP layer begins with a detail RNC user plane report that provides information about the Data Volume, Throughput and Transmitted Packets, both for DL and UL direction and for each protocol layer involved in the transmission of PDCP packets from RNC to UE and vice-versa. There is also a detailed DMCU (RNC Data dedicated CPU) Load which provides information about the available RNC User Plane processing capacity and the UL PDU Drop Ratio. By analyzing this graphic, it can be concluded that the Iub UL is not the source of the problem under analysis.



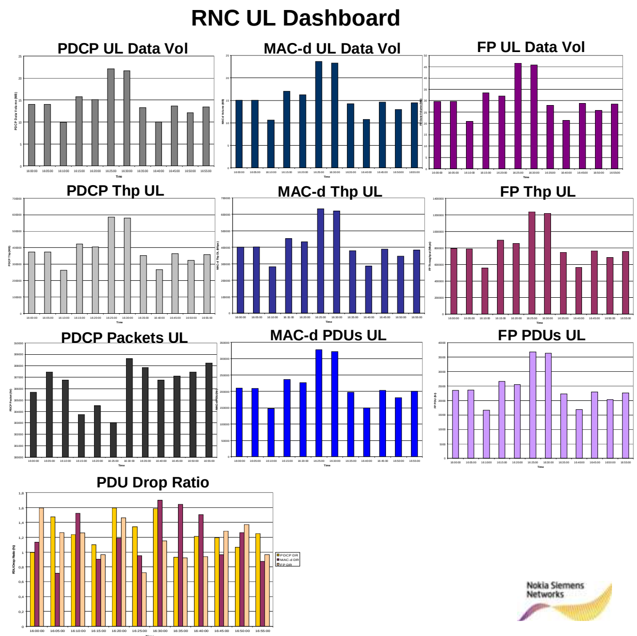


Figure 34 - Service Awareness– Session Description Report, PDCP - RNC Dashboard

The analysis continues with the Node B dashboard, where it can be found metrics related with Frame Protocol and MAC-d PDU that provides information about these layers performance from Node B perspective. There are two new graphics that provide information, from Node B perspective, about the amount of MAC-hs and MAC-es PDUs transmitted over the radio interface. The bottom graphics provide information about the MAC-hs and MAC-es efficiency detailed per each Node B Cell. A closer analysis to the “MAC-hs Efficiency” graphic highlights the root cause of this session’s problem: the Node B Cell 1 for some periods is performing poorly in the MAC-hs transmission, presenting efficiency lower than the minimum required 98%.

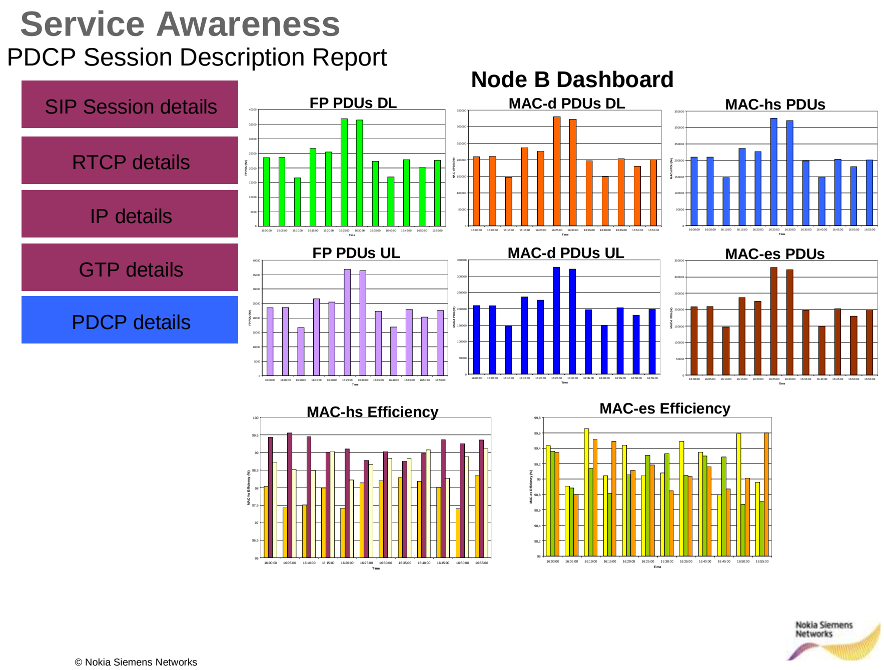


Figure 35 - Service Awareness– Session Description Report, PDCP – Node B Dashboard

The analysis of the Cell Dashboard confirms that a problem, related with MAC-hs efficiency, definitely exists in Cell 1 and analyzing in detail the “MAC-hs BLER, per Cause” graphic, it can be seen that the “Timer T1” cause is significantly contributing for such poor performance. The conclusion to take from this analysis flow is that the session serving cell HSDPA parameters has to be optimized, mainly T1 Timer parameter, has its current setting is affecting the VoIP service delivery.

This analysis flow may have other innumerous alternative analysis paths. In this case it was shown that the problem first identified was the packet loss ratio being higher than 2%, and the conclusion is that it was caused by a radio interface related bottleneck. However, this flow concept can efficiently be used to track other issues like delay, jitter, distortion, voice signal power or noise level effects.

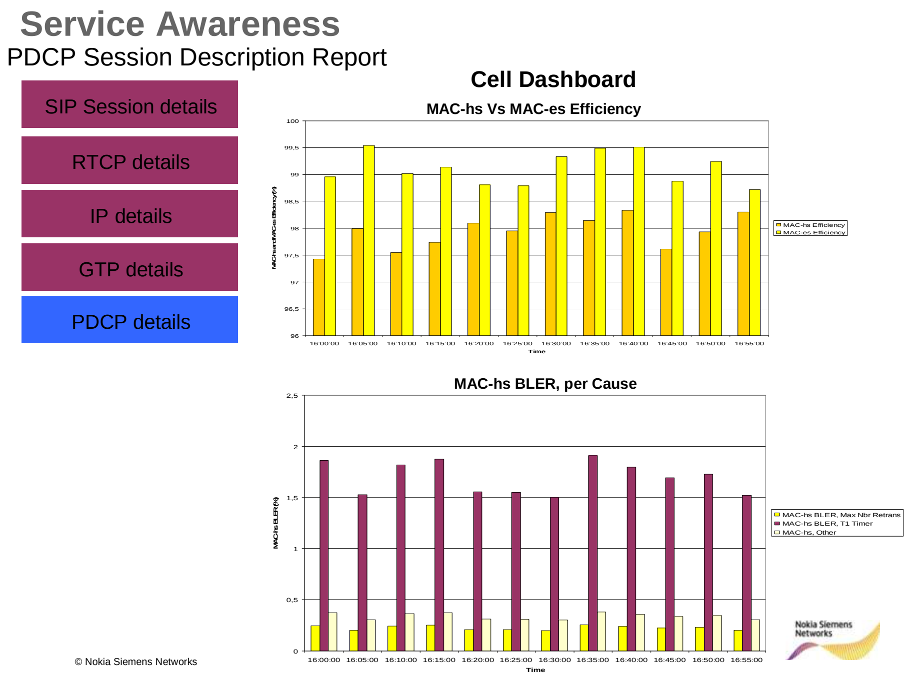


Figure 36 - Service Awareness– Session Description Report, PDCP – Cell Dashboard

Delay Analysis

This second use case is much simpler but yet powerful. The idea is to combine the service PM data with the CDRs data to provide a list of the Top-10 worst cells for session delays, i.e., the post-processor engine algorithm is to first identify the session affected by delay issues, then relate these results with the CDR data that provides for each session the Serving Cell IDs to setup a list containing the cells with more occurrences. This list serves as starting point for the session path analysis, beginning by the radio interface performance analysis towards the network end-to-end analysis and higher layers. Once more, the concept behind this analysis flow is that it should stop once the original problem source is detected, so if the analysis of a given layer does not provide enough information to take conclusions, analysis must continue in this case to the above layer.

Service Performance Indicator Analysis

This third use case takes advantage of the SPI defined metrics to classify the network ability to support a specific service. For instance, the network can be classified according to the measured performance while delivering VoIP service. This report makes use of the KPI ranking defined for Voice Service and combines the network R value, obtained from the average measured values for each KPI. The result is the network adequacy score for the service under analysis, which in this case, for VoIP the network the verdict would be that the network is good enough for VoIP support.

Service Performance Evaluation

VoIP report **SPI=3,7 -> Network is good enough for VoIP delivery**

KPI:	KPRV By Type Of Service		
	Voice:	Video:	Browsing:
E2EDelay	5	2	3
Reliability	4	4	3
Discard Rate	3	3	2
Jitter	3	2	1
Waiting Time	2	1	2
Bit Rates	1	5	2
TK	18	17	13

SPI_{voice} Network Score=3,7

E2EDelay=5*3

Reliability=4*5

Discard Rate=3*5

Jitter=3*3

Waiting Time=2*2

Bit Rates=1*5

TK=18

R	Delay (ms)	Jitter (ms)	Discard Rate (%)	Reliability (%)	WT (s)
5	<50	<0.1xDelay	<0.001	>99.999	<0.01
4	[50;100]	[0.1D;0.5D]	[0.001;0.01]	[99.99;99.9999]	[0.001;0.1]
3	[100;150]	[0.5D;0.8D]	[0.01;0.1]	[99.9;99.999]	[0.1;1]
2	[150;300]	[0.8D;D]	[0.1;1]	[99.9;99]	[1;10]
1	[300;600]	[D;2D]	[1;10]	[90;99]	[10;100]
0	> 600	>2D	>10	<90	>100

Figure 37 - Service Performance Evaluation - Network Classification for VoIP delivery

3.8.Conclusion

This chapter introduces the fundamental concepts developed in this work. It describes in detail an architectural proposal to cope with Performance Management system requirements. This architecture is divided in three fundamental parts: Network infrastructure composed by all system NEs; Element Manager responsible for managing the NEs and collecting the network PM data; Reporting tool responsible for data modeling and aggregation, statistical post-processing and data visualization engine. This architecture provides all the mechanisms and interfaces to allow PM data management end-to-end functionality, from the NE until the report generation for post performance analysis.

This chapter further details the adopted information model and its components described in network and service model metadata. Metadata is composed by Configuration Management (CM) information, which describes the network elements configuration detailing the topology, hierarchy, attributes and characteristics. CM information is stored in Object Class (OC) tree, where each object instance describes a network entity. Performance Management (PM) information is composed by measurements and counters that provide actual insight about network and service behavior. Each measurement is related with an OC and is composed by a set of counters that can describe protocols or procedures performance, hardware resource usage, processor load, etc. Fault Management (FM) information provides indications about system faults that can be of hardware or software type and are categorized by severity.

This chapter also describes the several data sources that are available both at network and service layers and enable the monitoring capabilities. These data sources are detailed per network sub-system, and it is also explained the kind of statistics that can be retrieved and their purpose. There are several data sources used in this work: PM data for RAN and CN; CDR data for session context description; and Deep Packet Inspection statistics that provide IP-based performance.

ETL process is also mentioned in this chapter. This is a very important process in the overall proposed Performance Management system, since it is responsible for implementing all the data collection, conversion and storage procedures, a cycle which has a high performance demand. There is often the requirement for ETL process to cope with a period of fifteen minutes, meaning that in this period the system must collect, convert and store thousand of PM data Mbytes. Extraction process is responsible to collect thousands of PM files from the entire network, using different collection mechanisms such as FTP, CORBA, etc. The Transformation process is responsible to convert different types and data formats into a single integrated and model coherent data-warehouse. The transformation process must have the ability to convert from different type of files and formats like XML, ASCII and often has even to handle with proprietary formats. Loading is the process responsible for storing all the data contained in the PM data converted in to the data-warehouse defined model.

This work proposes a simplified model for both network and service PM data, which is used to define and create the database structure that is used to load the CM, PM and FM data collected from all the identified sources.

The KPIs section provides information about the heart of this project, defines the fundamental performance indicators that must be used to establish relations in the identified layers and the impact that they have in each other. There are proposed KPIs for Network and Service layer, organized per system functional part: (ex. RAN, CN, Service, etc.); and per functional logic: Control Plane and User Plane.

Finally, this chapter describes some of the most important reporting use cases designed within this work. This is the part where the relation between layers is described and the analysis logic behind each report proposal. The proposed reports cover both reactive and proactive monitoring.

4. Simulation

This chapter presents the simulation related information. Section 4.1 introduces the simulation objectives and describes the simulation environment setup. Section 4.2 presents the obtained simulation results. Section 4.3 presents the simulation result analysis and conclusions.

4.1. Simulation Environment

The simulation work within this thesis served to assess the VoIP over HSPA network as a feasible scenario, by validating the service performance metrics. Furthermore, there is also the objective to compare the HSPA performance against UMTS R99.

The OpNet was chosen as the simulation tool to create a 3GPP Packet Switched network containing both R99 and HSPA elements and run the different simulation scenarios, comparing the results obtained from each other. Three simulation scenarios were implemented, representing different periods of the week with different traffic loads and usage.

Across the different scenarios the network topology is the same with varying traffic model.

The deployed simulation network model is composed by the following elements:

- 1 GGSN;
- 1 SGSN;
- 4 RNCs;
- 3 Node B HSPA capable;
- 1 Rel99 Node B;
- 22 HSPA capable UEs;
- 6 Rel99 UEs.

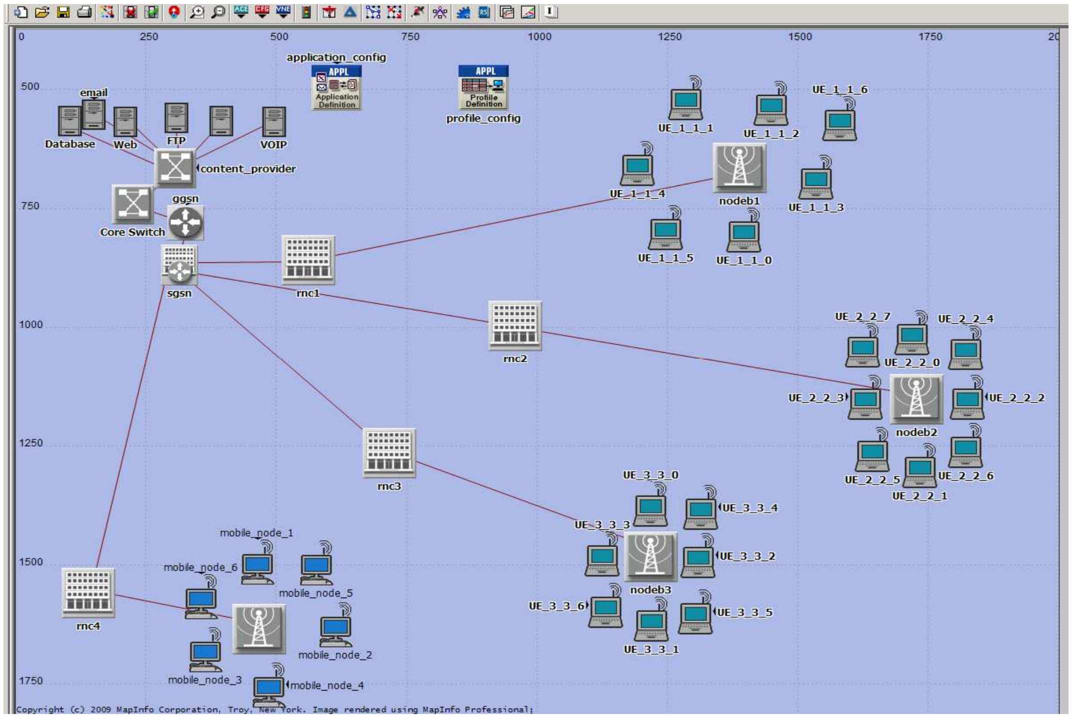


Figure 38 - OpNet – Modeled HSPA Simulation network

Since the OpNet does not offer any default HSPA model or NE, it was necessary to create the model that replicates HSDPA and HSUPA features, and both UEs and Node Bs were changed.
The changes performed to the Node Bs in order to enable HSDPA features can be seen in the Fig. 37.

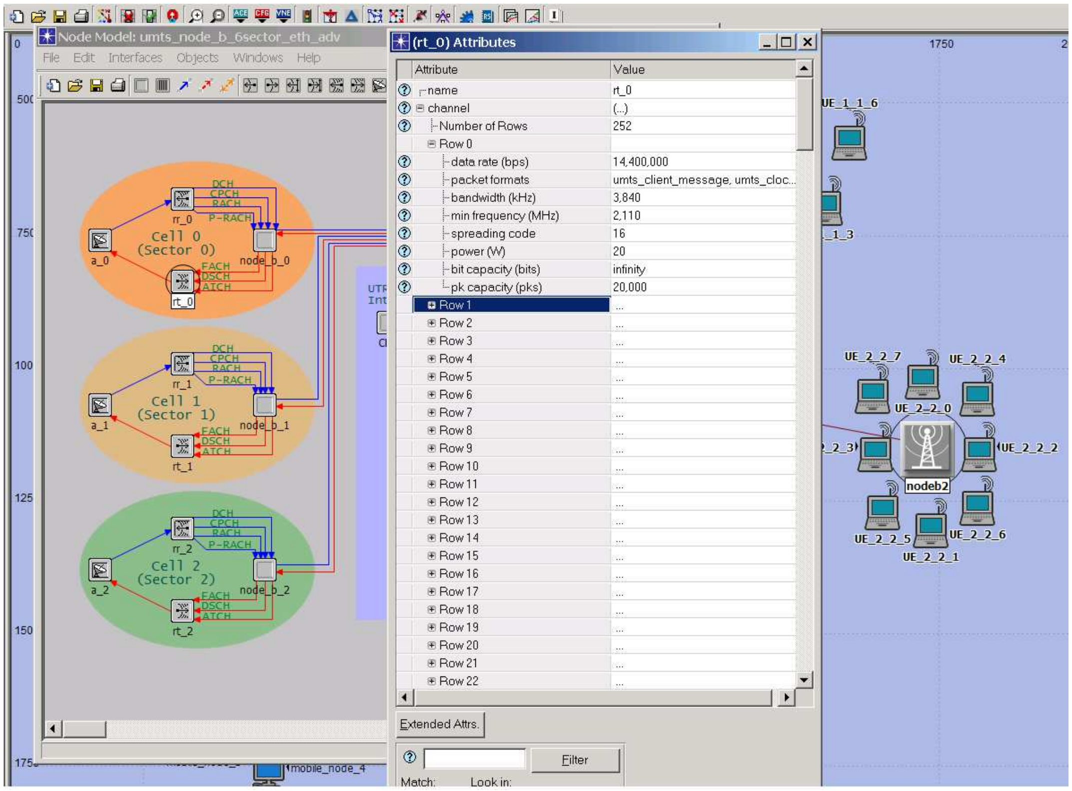
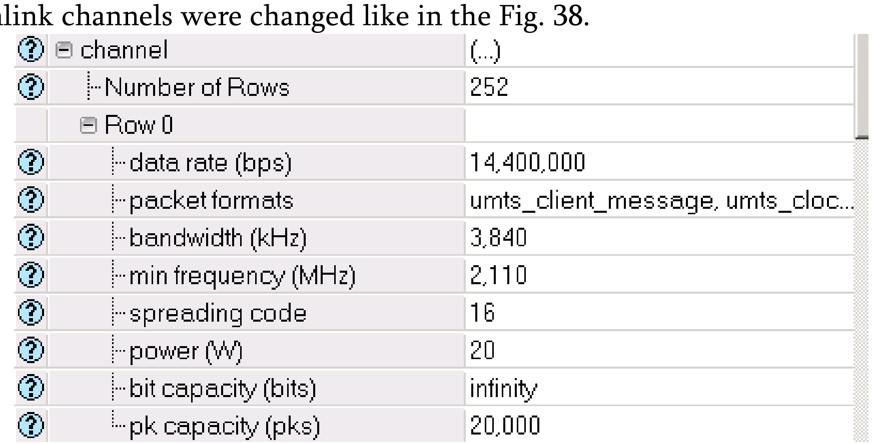


Figure 39 - OpNet – Node B HSDPA model change description

For each Node B Cell, the Modulation Scheme was set to 16QAM, and the downlink channels were changed like in the Fig. 38.



channel	(...)
Number of Rows	252
Row 0	
data rate (bps)	14,400,000
packet formats	umts_client_message, umts_cloc...
bandwidth (kHz)	3,840
min frequency (MHz)	2,110
spreading code	16
power (W)	20
bit capacity (bits)	infinity
pk capacity (pks)	20,000

Figure 40 - OpNet – Node B HSDPA channel change description

For the necessary UE changes to cope with HSDPA air-interface requirements the modulation scheme was set to 16QAM also and the downlink channels were configured like described in Fig. 39.

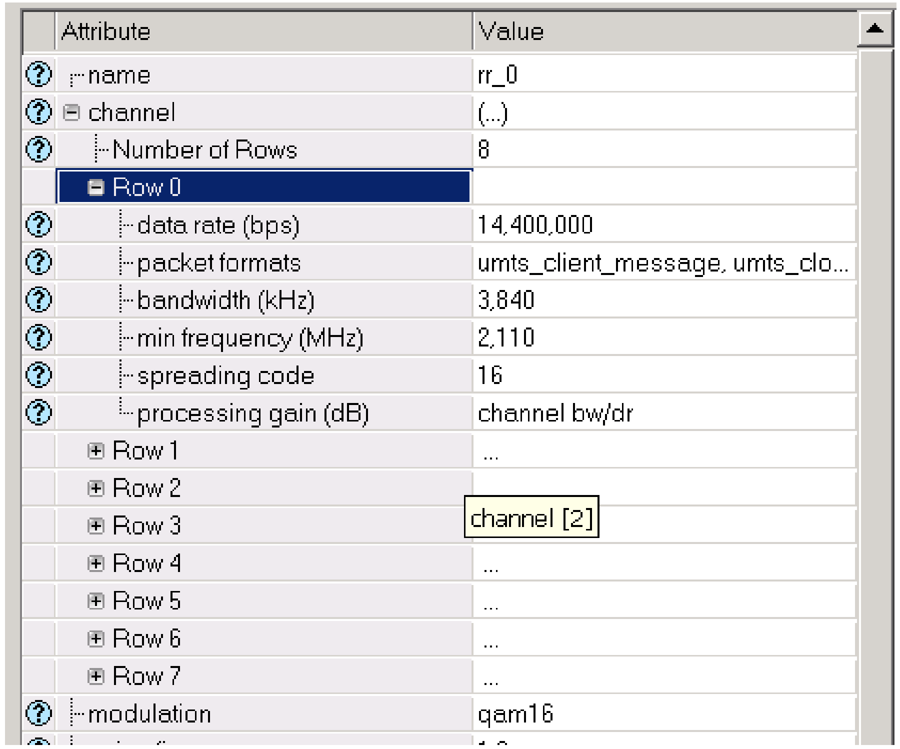


Figure 41 - OpNet – UE HSDPA change description

For the HSUPA model, it was also necessary to perform some changes in the Node B and UE uplink channel and modulation schemes.

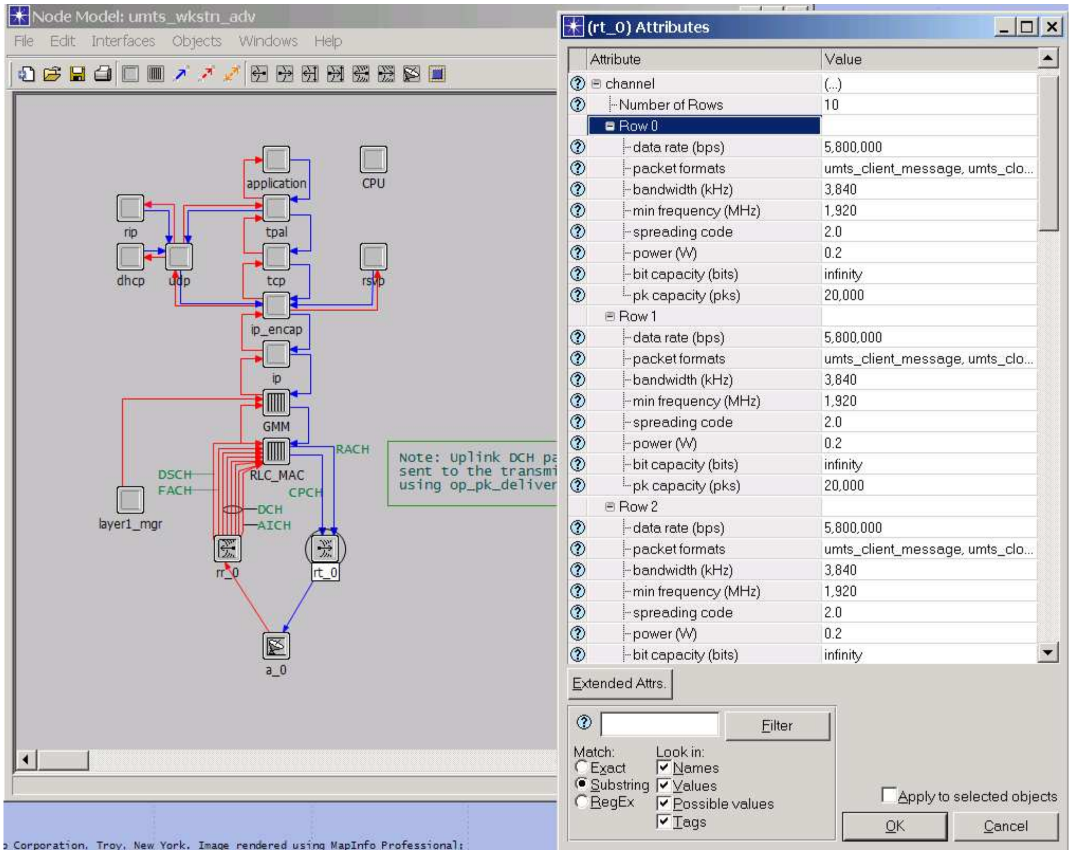


Figure 42 – OpNet - UE HSUPA changes description

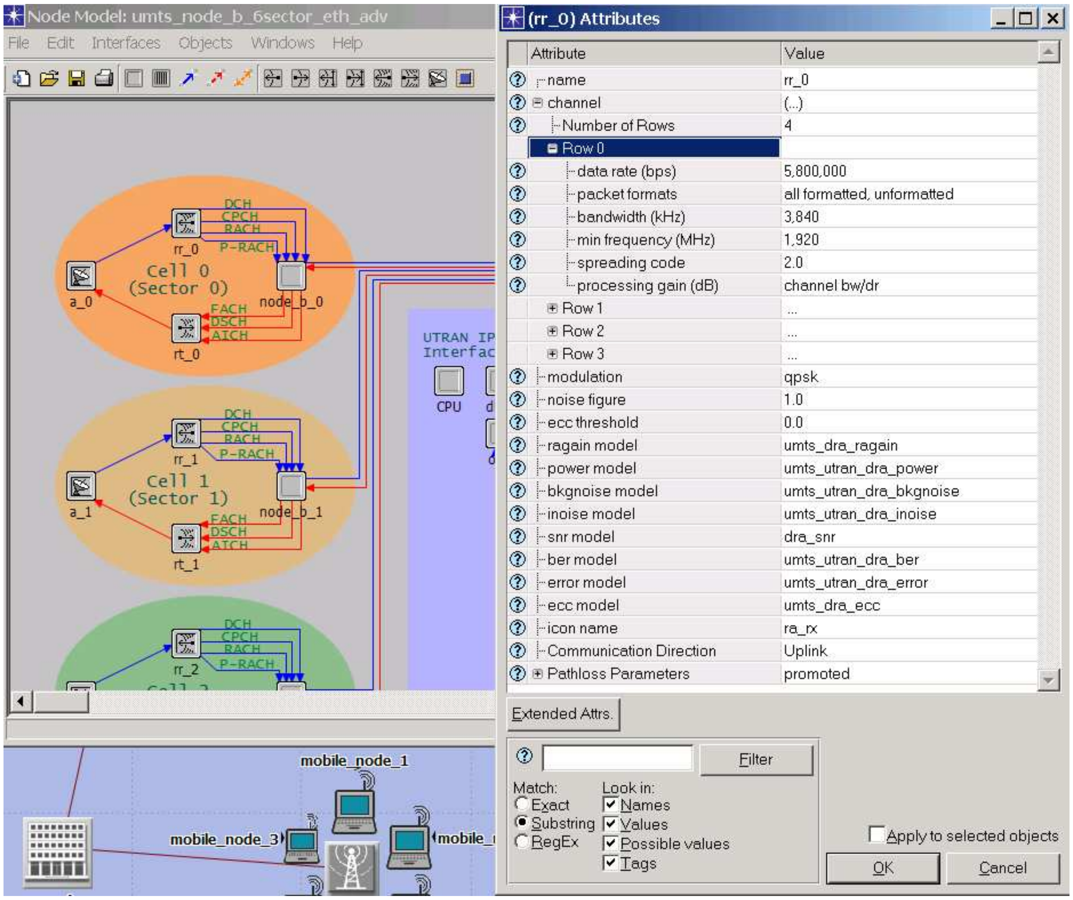


Figure 43 – OpNet -Node B HSUPA Change description

4.2.Results

Simulations were run with 500 values per statistic and an update interval of 500000. The simulations duration was one hour. All the graphics shown in this section have a confidence interval of 95%.

The following three simulation scenarios were deployed:

- Scenario 1: Light Load
- Scenario 2: Medium Load
- Scenario 3: Busy Hour

Each VoIP flow has the following characteristics:

- G.711;
- Throughput 120 kbps;
- 250 packet/s;
- Class of service: Interactive Voice.

<u>Scenario 1</u> – the 6 VoIP flows were created, between 3 UEs (1 UE per each HSDPA RNC) and a VoIP server.

Average E2E Delay Uplink: 100 ms	WT < 10s

Average E2E Packet Jitter Uplink: 50 ms	Reliability > 99.9%
Average E2E Delay Downlink: 100ms	Discard Rate = 0 %
Average E2E Packet Jitter Downlink: 45 ms	Bit Rate always available
GGSN GTP throughput DL = 480bps	GGSN GTP throughput UL = 340bps

Table 3 - Scenario 1-Simulation Results

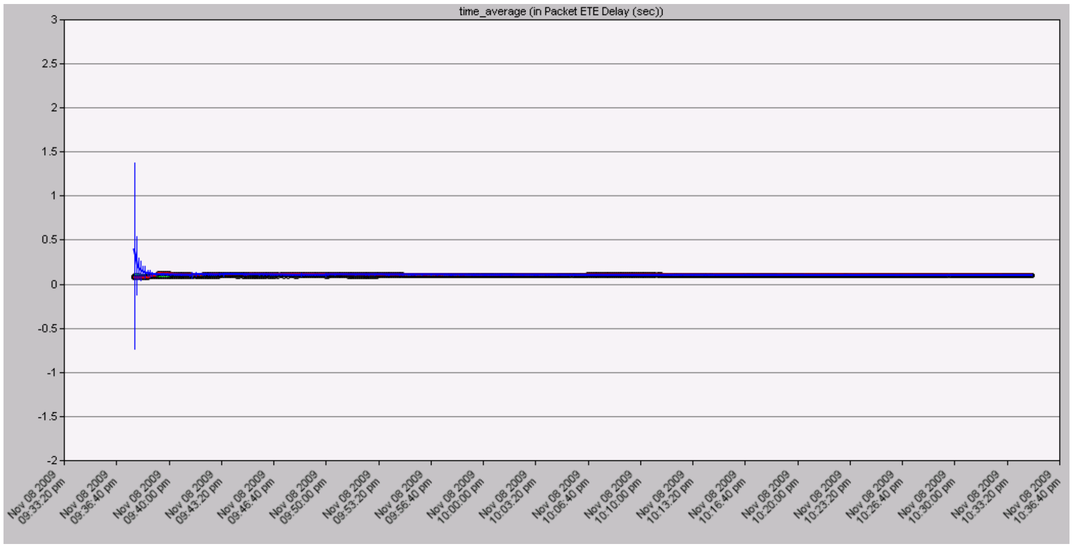


Figure 44 – Scenario 1-E2E Delay - Uplink

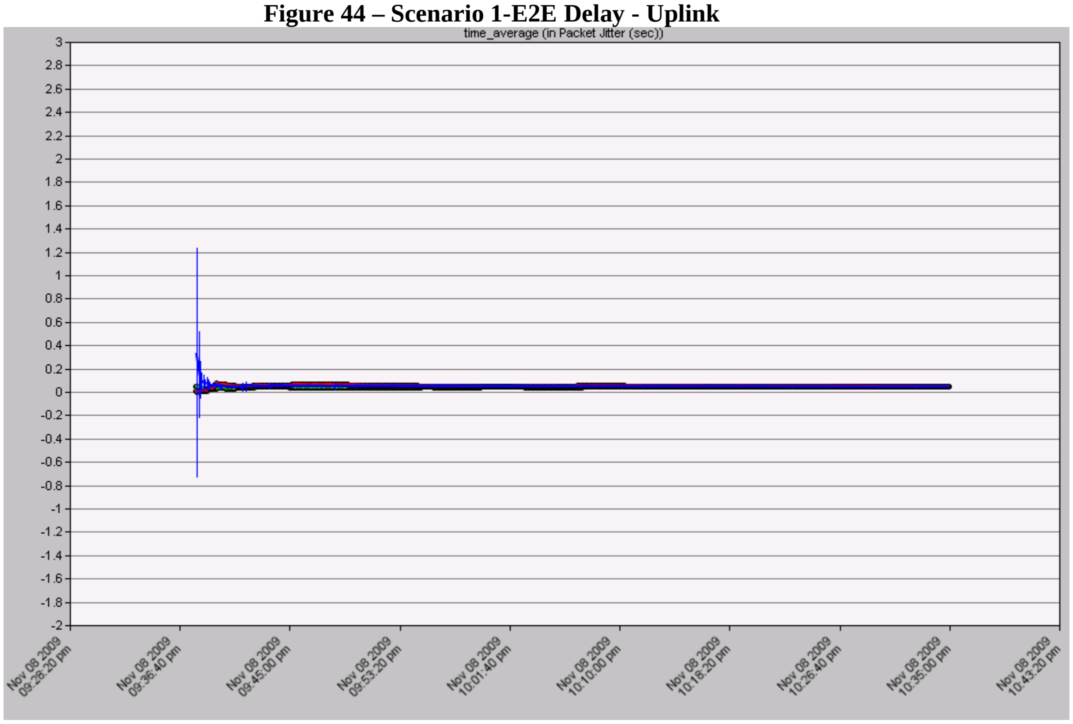


Figure 45 – Scenario 1-E2E Packet Jitter Uplink

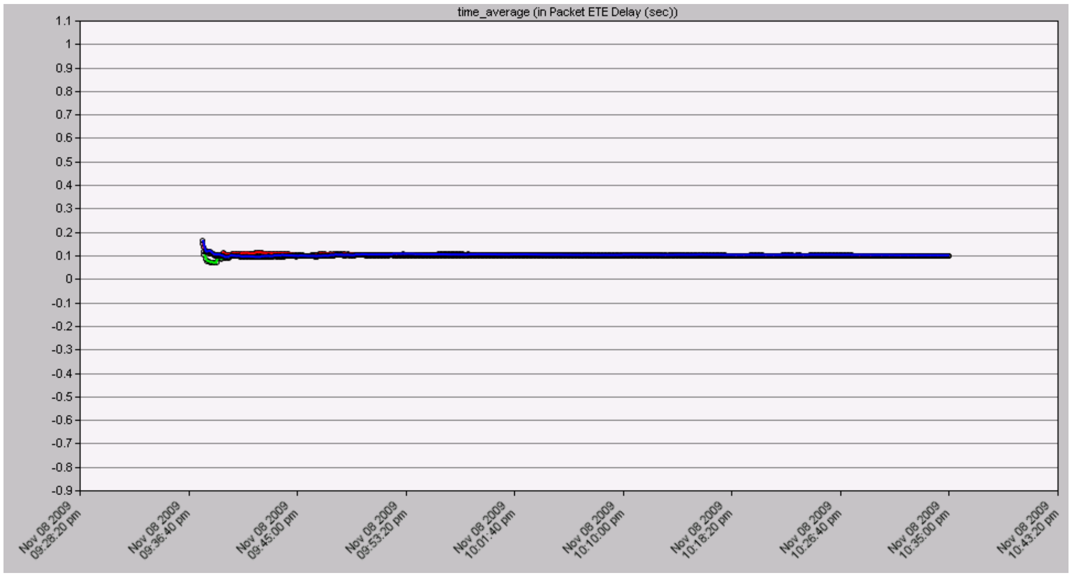


Figure 46 – Scenario 1- E2E Delay Downlink

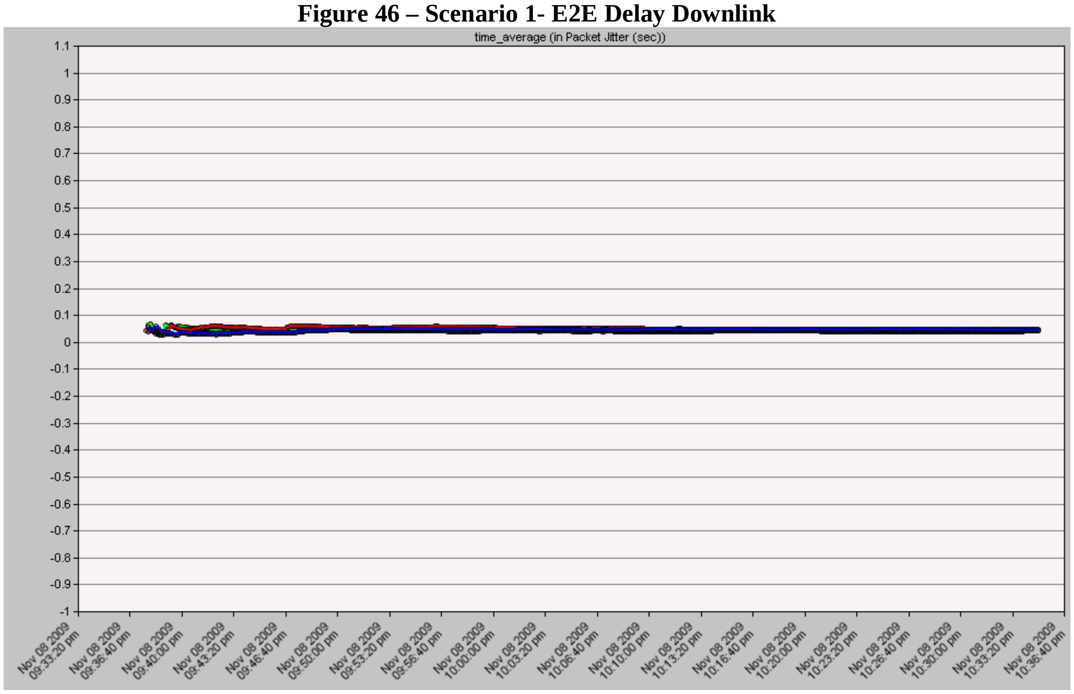


Figure 47 – Scenario 1-E2E Packet Jitter Downlink

<u>Scenario 2</u> – the 24 VoIP flows were created, between 12 UEs (4 UEs per each HSDPA RNC) and the VoIP server.

Average E2E Delay Uplink: 100 ms	WT < 10s
Average E2E Packet Jitter Uplink: 50 ms	Reliability > 99.9%
Average E2E Delay Downlink: 95ms	Discard Rate = 0 %
Average E2E Packet Jitter Downlink: 45 ms	Bit Rate always available
GGSN thp DL = 1800 Kbps	GGSN thp UL = 1400 Kbps

Table 4 - Scenario 2-Simulation Results

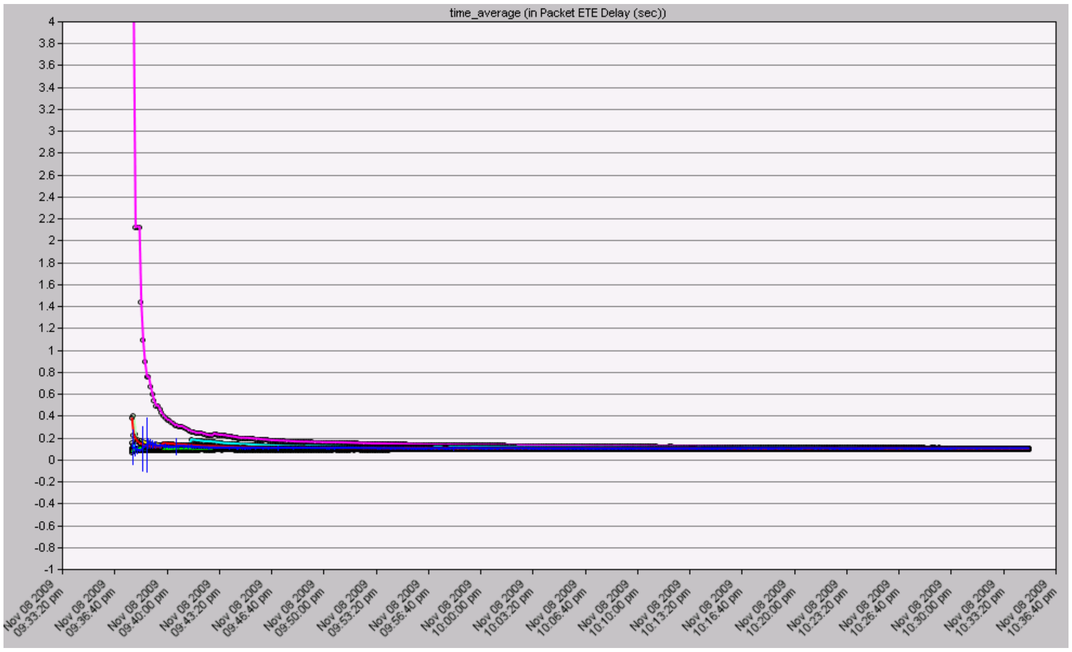


Figure 48 - Scenario 2-E2E Delay Uplink

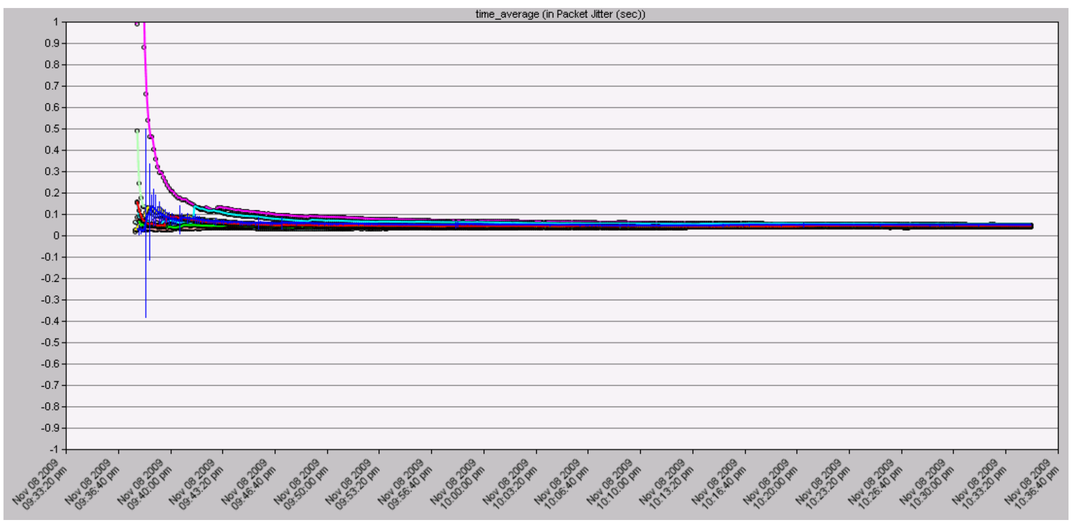


Figure 49- Scenario 2-E2E Packet Jitter Uplink

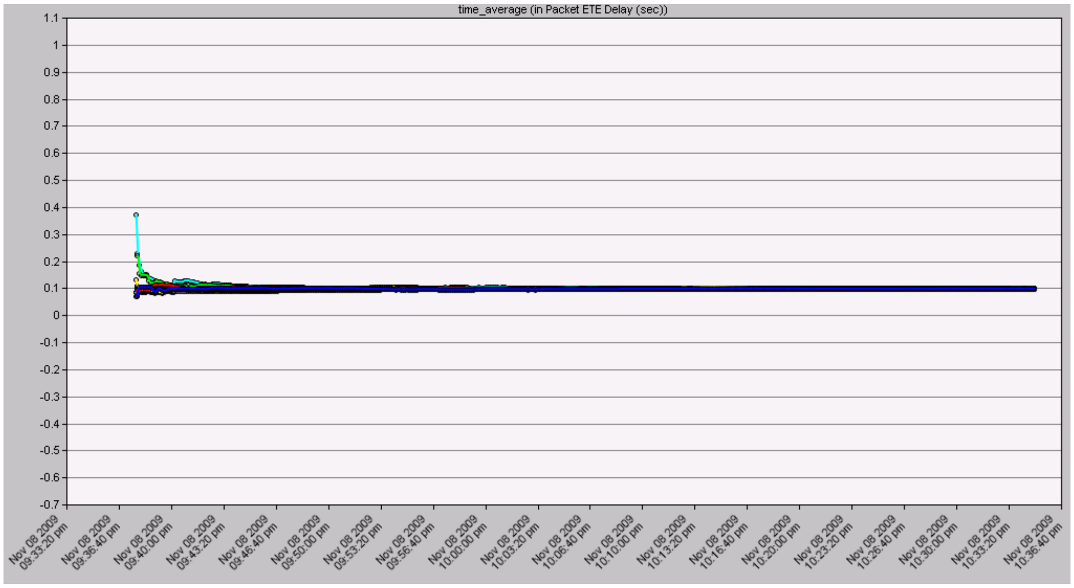


Figure 50 - Scenario 2-E2E Delay Downlink

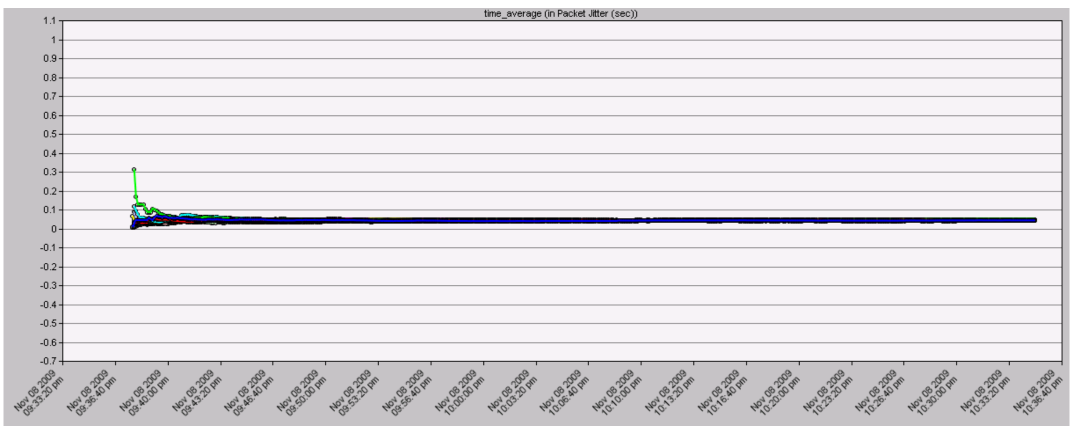


Figure 51 - Scenario 2-E2E Packet Jitter Downlink

<u>Scenario 3</u> – the 42 VoIP flows were created, between 21 UEs (7 UEs per each HSDPA RNC) and the VoIP server. This is the Busy Hour case.

Average E2E Delay Uplink: 100 ms	WT < 10s
Average E2E Packet Jitter Uplink: 50 ms	Reliability > 99.9%
Average E2E Delay Downlink: 95ms	Discard Rate = 0 %
Average E2E Packet Jitter Downlink: 40 ms	Bit Rate always available
GGSN throughput DL = 3300 Kbps	GGSN throughput UL = 2300 Kbps

Table 5 - Scenario 3-Simulation Results

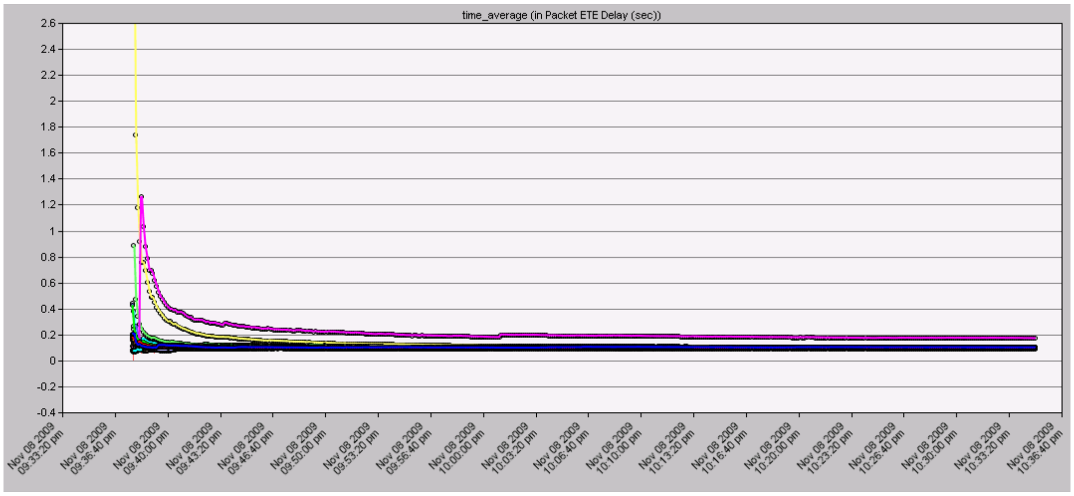


Figure 52 - Scenario 3-E2E Delay Uplink

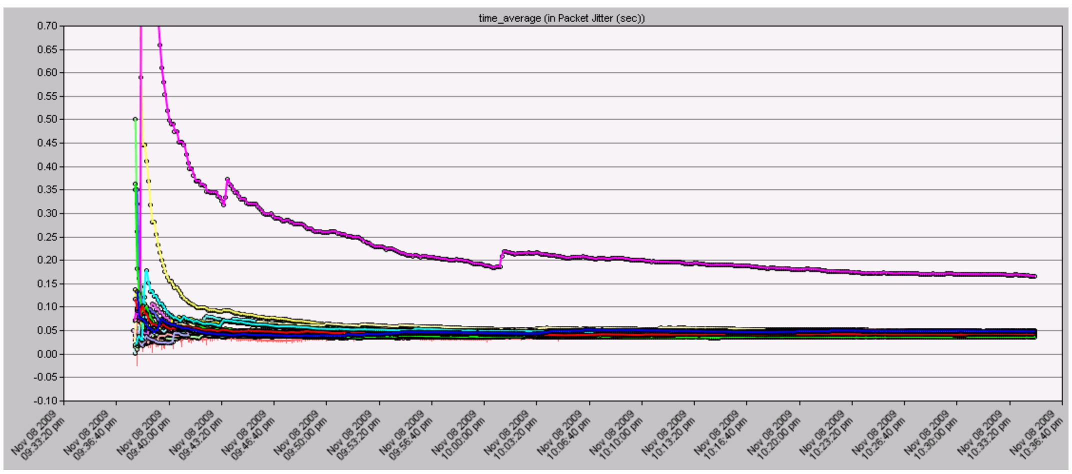


Figure 53 - Scenario 3-E2E Packet Jitter Uplink

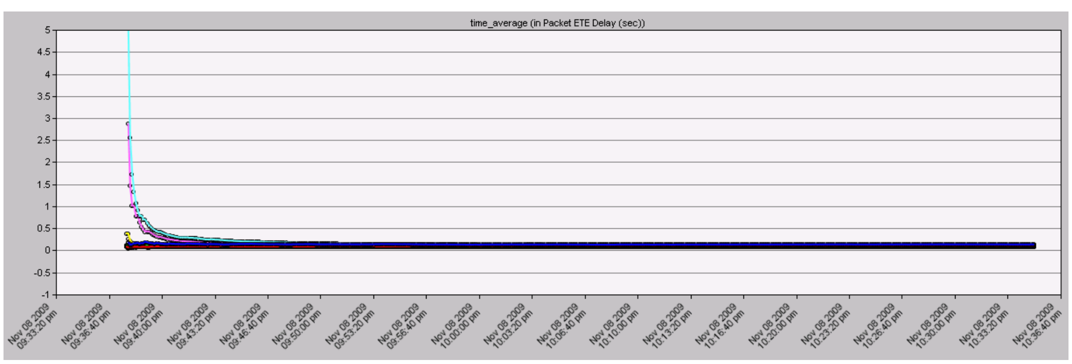


Figure 54 - Scenario 3-E2E Delay Downlink

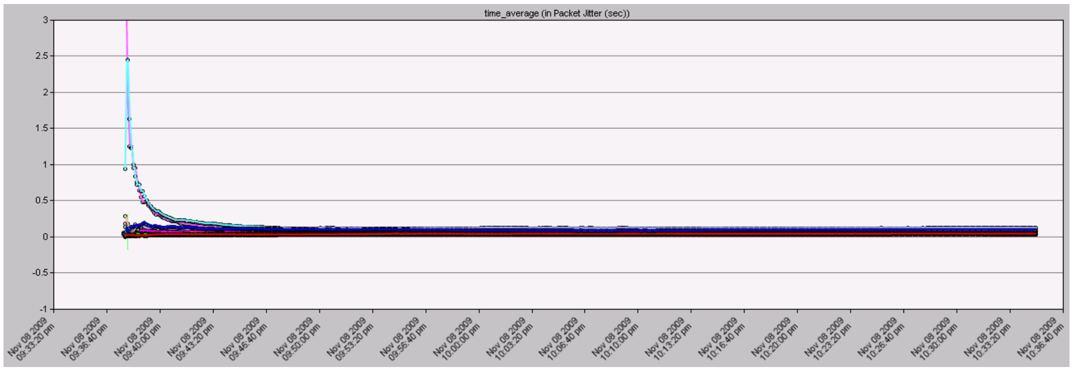


Figure 55 - Scenario 3-E2E Packet Jitter Downlink

4.3. Conclusion

This chapter describes the simulation work done within the thesis scope introducing the development environment and presenting the collected results.
The main goal of this simulation work is to demonstrate that the HSPA system is suitable for VoIP service delivery; therefore it was created a simulation network in OpNet that is able to model the most important features of HSPA, like throughput, modulation and spreading factor. There were created three different load scenarios representing different hours of a day, ranging from light load occurring for instance at dawn and a busy hour load occurring at the working hours.
The E2E Delay collected results show an average, between the three scenarios, of 100 (ms) for the Uplink direction and 96.67 (ms) in Downlink direction, which are values consistent with the values estimated in [25]. So the obtained results, throughout the three scenarios, are consistent with the expectations thus proving that the HSPA system is capable of delivering VoIP service.

Applying, to these results, the *"Service Performance Indicator Analysis"* defined in Section 3.7:

- Average E2E Delay throughout the three scenarios is 100 (ms), which implies that *R* (defined in Table 2) is equal to 4, being this KPI the most important to classify VoIP performance the KPRV (defined in Table 1) is equal to 5;
- Reliability is above >99.999, so R=5 and the KPRV is 4;
- The jitter is 0.5 of the delay, so R is 4 and the KPRV is 3;
- The available bit rate is always enough to support the service and there is always a surplus of bandwidth, so it is consider R value equal to 5;
- The WT is above 1 second and bellow 10 seconds, so R=2 for the KPRV of 2.

The HSPA network achieves a score of 3.94 out 5 while delivering the VoIP service.

The results obtained in this chapter also serve as proof of concept for the work developed in this thesis, by classifying the underlying network performance according to the delivered service demands. Due to the fact that this approach is technological agnostic it can be used to relate network performance with service behavior for every kind of underlying packet network, fixed or mobile, and be even extended to analyze different types of IP-based services.

The cross-layer approach proposed in this Thesis divides the analysis scope into two inter-dependent layers: Network layer and IP-based service layer; it is also validated by these results thus relating the network performance with service delivery QoS and providing mechanisms to classify the adequacy of a specific network to deliver a specific service, perform troubleshooting, forecast analysis, predict faults and the need for capacity improvements.

5. Conclusion

5.1. Final Conclusion

This work proposes a new and innovative performance monitoring concept which combines IP-based service metrics with network performance metrics analysis. This new performance management concept results from the achievement of all the objectives designed for this thesis.

The first part of this work was related with the IP-based services flow in 3GPP UMTS networks, the NEs present in the data path and their functionality. It was also investigated the elements specifically related with VoIP traffic and session management. This first part of the work had a great influence in the tasks related with the PM data source identification.

Another important step was related with the study of QoS support in 3GPP UMTS networks to understand, for instance, how a VoIP flow is handled in end-to-end data path and especially on the RAN side. This was a crucial step for the development of this work, mainly for the KPI design, selection and reporting definition.

The third step of this work was the identification of the PM data sources and their format. This task was in the critical path, and therefore, it can be classified as the most important part of the entire study. The identified data sources and the provided metrics are the very heart of this work and, without them, this entire work would not be possible. The extension of network PM metrics with other data types like CDRs and IP-Service statistics provided by Deep Packet Inspection mechanisms is a key factor for this work proposal success.

The next phase was the design and selection of the most important KPIs and the definition of the reporting concept. In this task, it was identified the most important set of KPIs for the several horizontal layers that describe both IP-based Service and 3GPP UMTS Network protocol stack, and the vertical layers that related with the functionality part: Control and User Plane. This work proposes a comprehensive set of both IP-based service KPIs and Network PM KPIs that enables end-to-end performance monitoring capabilities.

The last step of this work analysis is the report concept definition that aims to provide means of relating the IP-based Service Performance Layer with the underlying Network Performance Layer: in the particular case of this work is 3GPP HSPA network. This reporting concept combines both the IP-based session oriented information, QoS performance statistical metrics with a new approach for describing the end-to-end network performance view, leading to an innovative troubleshooting, optimization and planning tool proposal. This new reporting concept will be integrated in the Nokia Siemens Network solution portfolio.

The new performance monitoring concepts introduced by this work are a very important step towards the All IP Mobile Network Performance Management area, which provides the means and tools for Network Operators and Service Providers monitor the performance of their sub-systems and also the SLA agreements. This will allow setting a new trend in Operational Support Systems, Performance Management and Service Assurance solutions.

This work also provides some simulation results which show that the VoIP over HSPA networks is a feasible scenario, thus proving the effectiveness of this work, since this will become a frequent scenario in the near future.

5.2.Future Work

As future work, the performance monitoring concept introduced in this thesis can be extended, mainly to detail the support of mobility scenarios, both geographical mobility and inter-RAT mobility.

Moreover, since the objective of this work was the proposal of a generic performance monitoring concept, this work can also be extended to support other network technologies and analysis scenarios, such as multi-technology reports.

Finally, the proposed report use cases can be implemented and tested within a real-network premise, in order to be productized and integrated in the Nokia Siemens Network portfolio.

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