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IP in the RAN as a Transport Option

in 3rd Generation Mobile Systems

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Abstract:

The purpose of this technical report is to consider the way in which IP can be applied in Radio Access Networks within 3rd Generation mobile systems. The mobile systems to be considered are the UTRAN being developed by 3GPP (3GPP-UTRAN) for both FDD and TDD modes and the RAN being developed by 3GPP2 for CDMA2000 (3GPP2-RAN). The application of IP may also be considered for other mobile systems, particular where a sharing of a common transport network between 3rd generation mobile and other mobile systems might be considered to provide benefits to operators.

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1 INTRODUCTION

1.1 Objectives of the MWIF Technical Report

This technical report, from the IP in the RAN group within MWIF, considers how IP networks and protocols can be applied as a transport option for Radio Access Networks within 3rd Generation mobile systems and the benefits which might be provided. Within this work IP is only considered as a transport option over the RAN internal interfaces and RAN interfaces to the core network, without any changes to the RAN architecture or radio control protocols.

The mobile systems to be considered are the UTRAN being developed by 3GPP (3GPP-UTRAN) for both FDD and TDD modes and the RAN being developed by 3GPP2 for CDMA2000 (3GPP2-RAN). The application of IP may also be considered for other mobile systems, particularly where sharing of a common transport network between 3rd generation mobile and other mobile systems might be considered to provide benefits to operators.

It is expected that this report should influence input to 3GPP, 3GPP2 and IETF to provide directions as to how IP can be applied within Radio Access Networks and to justify the benefits of IP as a transport option. The output of this work should be liaison statements from MWIF to various groups within 3GPP, 3GPP2 and IETF.

The scope of the work is only to consider the way in which IP can provide benefits to operators in terms of a transport network to support information transfers over RAN internal interfaces and RAN interfaces to the core network. The objective is to enable IP as an option for the Transport Network Layer supporting the Radio Network Layer protocols within 3GPP-UTRAN, 3GPP2-RAN and possibly other mobile systems.

There should not be any changes in the network architecture, functionalities or radio network layer protocols in the 3GPP-UTRAN (release 99 and release 4/5) or 3GPP2-RAN (3G-IOS version 4.0, cdma2000). The only exception is if it is identified that a minor change to the radio network layer might simplify the interface to an IP based Transport Network Layer then this might be suggested to 3GPP and/or 3GPP2.

1.2 Definitions

This document employs the following terminology:

- Must, Shall, or Mandatory - the item is an absolute requirement of the Technical Report (TR).
- Should — the item is highly desirable.
- May or Optional — the item is not compulsory, and may be followed or ignored according to the needs of the implementers.

1.3 Overview of the Technical Report

The purpose of this technical report is to consider the way in which IP can be applied in Radio Access Networks within 3rd Generation mobile systems. The mobile systems to be considered are the UTRAN being developed by 3GPP (3GPP-UTRAN) for both FDD and TDD modes and the RAN being developed by 3GPP2 for CDMA2000 (3GPP2-RAN). The application of IP may also be considered for other mobile systems, particularly where a sharing of a common transport network between 3rd generation mobile and other mobile systems might be considered to provide benefits to operators.

1.4 Release plan

It is the objective of the MWIF to provide timely industry direction for mobile wireless internet. In order to accomplish this, the MWIF will periodically release Technical Reports. The period in which Technical Report will be released will be frequent enough to meet the objective of timely industry direction.

This Technical Report is one of a series intended to specify the MWIF architecture. At the time of release of this report, the following MWIF Technical Reports are scheduled:

- MTR-001 MWIF Architectural Principles
- MTR-002 MWIF Architecture Requirements
- MTR-003 MWIF Layered Functional Architecture
- MTR-004 MWIF Network Reference Architecture
- MTR-005 MWIF Gap Analysis
- MTR-006 MWIF IP in the RAN as a Transport Option in 3rd Generation Mobile Systems
- MTR-007 MWIF IP OpenRAN Architecture

This report (MTR-006) will be released in multiple versions. Each version will include descriptions of the technical work performed during a particular stage along with the conclusions made.

1.5 MTR-006 Conclusion

The MWIF WG4 (IP in the RAN as a Transport Option in 3rd Generation Mobile Systems) focused on evaluating the applicability of IP in the RAN as a transport option. In addition to the background information, MTR-006 includes:

1. A technical assessment of the viability of IP transport in the RAN,
2. Applicable IP protocol stacks described in Chapter 8,
3. 3G RAN traffic models described in Chapter 9,
4. Simulation results for IP protocol delay and performance for several proposed IP protocol stacks described in Chapter 10.

As of the time of releasing MTR-006 to the MWIF Technical Committee, the technical study carried out by the MWIF WG4 is fully described in this report. This study has resulted in the following unanimous consensus:

1. IP in the RAN with careful design is a viable option, in particular, as it relates to delay, and bandwidth efficiency concerns.
2. Simulations conducted by several companies consistently demonstrated that the IP transport performance is equal or better than the present transport used in the RAN today. Based on the models described in Chapter 9, IP transport showed approximately 10% improvement over ATM/AAL2 (See simulation details in Chapter 10).
3. The bandwidth efficiency improvement of 10% for IP transport over AAL2 transport is a maximum value, which is only achieved over the last hop when no routing header is applied.
4. The simulations only considered IP transport over the last hop. Other scenarios, like the use of a managed network, should also be considered. This might be done analytically

1.6 Structure of this Report

Chapter 1 : **Introduction** to the content and purpose of this report.

Chapter 2 : **References** to relevant papers, specifications and reports

Chapter 3 : **Definitions, Symbols and Abbreviations**

Chapter 4: **Overview** of the work described in the report

Chapter 5 : **Radio Access Networks Architectures** as a description of the current RAN architectures within 3GPP, 3GPP2 and other mobile systems. This describes current interfaces within these mobile systems where IP transport networks might be applied.

Chapter 6 : **Transport Network Requirements** as the requirements placed on the transport network by the Radio Network Layer, operator requirements, etc.

Chapter 7 : **IP Transport Options** as the IP networks and protocols from standards (e.g. IETF) with brief comments as to the benefits they might provide to support transport over RAN interfaces.

Chapter 8 : **IP based RAN Transport Network** as various solutions to the way in which IP networks and protocols might be used to support transport networks within various RAN architectures.

Chapter 9 : **Traffic, Network and System Models** as a description of the models developed to support the evaluation and simulation of various IP networks and protocols.

Chapter 10 : **Performance of IP in RAN as Transport** as the results of the evaluation, simulation and comparison of various IP network and protocol options.

Chapter 11 : **Recommendations** from this work

Chapter 12 : **Future work**

Annex A: 3GPP Specification References

Annex B: 3GPP2 Specification References

Annexes C: Equations for Pareto Distribution

Annexes D: List of Microsoft Word Styles used in this document

Document History as changes made in each new version of this document.

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3 DEFINITIONS, SYMBOLS AND ABBREVIATIONS

This section provides definitions, symbols and abbreviations relevant to the application of IP as a transport option within the 3GPP-UTRAN, 3GPP2-RAN , IP work in IETF, and other mobile systems. Comments are provided on the alignment of definitions, symbols and abbreviations between different systems, e.g. 3GPP-UTRAN and 3GPP2-RAN. Abbreviations and definitions for the UTRAN are provided in TS 25.990.

3.1 Abbreviations

3G	3 rd Generation
3GPP	3 rd Generation Partnership Project
3GPP2	3 rd Generation Partnership Project-2
AAL	ATM Adaptation Layer
AAL2	ATM Adaptation Layer type 2
AAL5	ATM Adaptation Layer type 5
ALCAP	Access Link Control Application Part
A3	<i>See Definitions section</i>
A7	<i>See Definitions section</i>
Abis	<i>See Definitions section</i>
AMR	<i>See Definitions section</i>
ATM	Asynchronous Transfer Mode
BS	Base Station
BSC	<i>See Definitions section</i>
BTS	<i>See Definitions section</i>
CDG	CDMA Development Group
CDMA	Code Division Multiple Access
CFN	Connection Frame Number
CID	Content ID
CIP	Compressed IP
CN	Core Network
CPCH	Common Packet Channel
CRC	Cyclic Redundancy Check
CR-LDP	Constraint based LDP
CRNC	Controlling Radio Network Controller
cRTP	Compressed RTP
cTCRTP	<i>see Definitions section</i>

cUDP	Compressed User Datagram Protocol (UDP)
DCH	Dedicated Channel
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name Service/Server
DS0	Digital Signal Level 0 (64 kbps)
DSCH	Downlink Shared Channel
DRNC	Drift Radio Network Controller
DTX	Discontinuous Transmission
FACH	Forward Access Channel
FDD	Frequency Division Duplex
FEC	Forwarding Equivalence Class
FR	Frame Relay
GRE	Generic Routing Encapsulation
GSM	Global System for Mobile communications
ICMP	Internet Control Message Protocol
IETF	Internet Engineering Task Force
IOS	Inter Operability Standard
IP	Internet Protocol
IPSEC	IP Security (Protocol)
IS-IS	Intra-autonomous System to Intra-autonomous System
Iu	<i>See Definitions section</i>
Iu-CS	Iu-Circuit Switched
Iu-PS	Iu-Packet Switched
Iur	<i>See Definitions section</i>
L1	Layer 1 (physical layer)
L2	Layer 2 (data link layer)
L3	Layer 3 (network layer)
LER	Label Edge Router
LFI	Link Fragmentation and Interleaving
LSP	Label Switched Path
LSR	Label Switched Router
MAC	Media Access Control
MAC-c	MAC-control Channel
MAC-d	MAC-dedicated Channel
MAC-sh	MAC-shared Channel

MGCP	Multimedia Gateway Control Protocol
MPLS	<i>See Definitions section</i>
MTU	Maximum Transmit Unit
NBAP	Node B Application Part
O&M	Operations and Maintenance
OSPF	Open Shortest Path First
PAN	Personal Area Network
PCF	Packet Control Function
PDCP	Packet Data Convergence Protocol
PDSN	Packet Data Support Node
PID	Packet Identification
PPP	Point to Point Protocol
QoS	Quality of Service
RAB	<i>See Definitions section</i>
RACH	Random Access Channel
RADIUS	Remote Access Dial-in Service
RAN	<i>See Definitions section</i>
RANAP	Radio Access Network Application Part
RIP	Routing Information Protocol
RLC	Radio Link Control
RNC	<i>See Definitions section</i>
RNS	<i>See Definitions section</i>
RNSAP	Radio Network Subsystem Application Part
RRC	Radio Resource Control
RSVP	Resource Reservation Protocol
RTP	Real Time Protocol
SCCP	Signaling Connection Control Protocol
SCTP	Stream Control Transport Protocol
SDU	<i>See Definitions section</i>
SRNC	Serving RNS
SS7	Signalling System No. 7
TCP	Transmission Control Protocol
TCRTP	Tunnelled cRTP
TDD	Time Division Duplex
TFI	Transport Format Indicator

TNL	Transport Network Layer
ToS	Type of Service
TrCH	Transport Channel
TSG	Technical Specification Group
TTI	Transmission Timing Interval
UE	User Equipment
Um	<i>See Definitions section</i>
UMTS	Universal Mobile Telecommunications System
UDP	User Datagram Protocol
UNI	User-Network Interface
UP	User Plane
UTRA	Universal Terrestrial Radio Access
UTRAN	<i>See Definitions section</i>
Uu	<i>See Definitions section</i>
VPN	Virtual Private Network
WAN	Wide Area Network
WCDMA	Wideband Code Division Multiple Access
WWW	World Wide Web

3.2 Glossary of terms

A Interface: Interconnection point between BSC and Core Network in 3GPP2 IOS 4.0. Designated by Iu in 3GPP Specifications.

Abis: Interconnection point between BSC and BTS in 3GPP2. Similar to the Iub in 3GPP Specifications, but not standardized.

A3 and A7: Logical interface between two BSCs in 3GPP2 IOS 4.0. A3 and A7 interfaces are both TCP/IP connections over ATM. Identified as Iur in 3GPP Specifications.

Access Network: An access network comprises all functions that enable a user to access core network services. It can be used to hide all access-specific peculiarities from the core network.

Access Stratum: Access layer or level

Admission Control: Procedure by which the network ensures that interference created after adding a new call will not exceed a pre-specified threshold. It is always performed when a mobile station initiates communication in a new cell either through a new call or a handover.

AMR (Adaptive Multi-Rate) speech codec: This is the speech codec originally standardised by ETSI for the GSM system and selected by 3GPP as mandatory speech codec for 3rd generation systems.

Bearer: An information transmission path of defined capacity, delay, and bit error rate.

- BSC:** A radio network element that is in charge of controlling the use and the integrity of the radio resources at the BTSs under its control in 3GPP2 IOS 4.0. In the 3GPP Specifications it is designated by RNC.
- BTS:** In 3GPP2 IOS 4.0, a logical fixed node responsible for communicating with mobile stations in one or more cells. It is connected to the BSC through the A-bis interface. Depending upon the context, the term base station may refer to a cell, a sector within a cell, an MSC, or other part of the wireless system. In 3GPP Specifications it is designated as Node B.
- Best effort service:** A service model that provides an unspecified QoS.
- Common channel :** a radio channel not dedicated to one particular user equipment.
- Core network (CN) :** A core network comprises the switching network (MSC) and the service network (for location management etc.). It includes all the functions related to call and bearer control for fixed transmission.
- Control Plane:** The control plane is a vertical layer in the ISDN protocol reference model. It consists of all functions in charge of transferring information for the control of user plane data
- Congestion:** Excessive fullness of network. In a heavily loaded network, congestion may be due to heavy traffic or bottlenecks.
- cTCRTP:** An optimised version of TCRTP for usage on the last mile. With cTRCTP the header of the outer IP packet carrying the tunnelled L2TP packet is IP -header compressed.
- Dedicated Channel:** A channel dedicated to a specific user.
- Delay :** Delay is a period in time from some start event to some end event in which further processing is suspended while the network waits for the arrival of data or some other event. One of the main interests in this report is the delay experienced for the transport of blocks of data over the Iub interface. This is the delay from the start of the transmission of a block of data from the RNC to the complete reception of that block of data in Node B.
- Diffserv:** Differentiated services define different classes of IP service in which QoS is determined by markings on each packet.
- Entity:** Network element comprising a set of functions and responsible for performing its allocated tasks.
- Fairness:** Fairness within a network implies that all endpoints, clients and servers, within a network are treated equally for purposes of data transmission.
- Flow Control:** Mechanism(s) used to prevent the network from becoming overloaded by regulating the input rate transmissions. It is a continuous process and if the load increases from the pre-defined value, the network takes appropriate action (bit rate reduction/transmission delay/dropping the low priority calls), as per service contract/traffic type/QoS requirements per user.
- Frame Selection:** See Macrodiversity Combining. Frame selection is the term used by 3GPP2.
- Guaranteed service:** A service model that provides highly reliable performance, with little or no variance in the measured performance criteria.
- Interface:** The common boundary between two associated systems (Source ITU-T I.113).

- Intserv:** Integrated services define different classes of service in which applications inform routers of the QoS treatment they require for particular flows and QoS is administered based on a per flow bases determined by the application and host.
- Iu:** Interconnection point between an RNC and a Core Network in 3GPP. It is also considered as a reference point. In 3GPP2 Specifications it is called A interface.
- Iub:** Interface between an RNC and Node B in 3GPP. It is designated by A-bis in 3GPP2 Specifications.
- Iur:** A logical interface between two RNC in 3GPP. While representing a point to point link between RNCs, the physical realisation may not be a point to point link. It is designated by A-3, A-7 in 3GPP2 Specifications.
- Jitter:** Variation in delay due to system imperfections in the network, either hardware or software, or due to traffic conditions within the network.
- Last mile:** The link between BTSs/NodeBs and their corresponding Edge Router(s).
- Node B:** A logical node responsible for radio transmission/reception to/from the User Equipment in one or more cells in 3GPP. It terminates the Iub interface towards the RNC. In 3GPP2 Specifications it is designated as BTS or BS.
- Macrodiversity Combining :** The process by which information received in the UE or UTRAN, via radio links to/from different radio cells, may be combined to improve performance. Such combining techniques might include selection of data blocks on which CRC checks do not indicate errors and/or from the radio link with the best signal to interference ratio or error rates. Macrodiversity combining is a 3GPP term, this term is often called frame selection in 3GPP2.
- Macrodiversity Combiner:** The functional entity in the 3GPP RAN that performs macrodiversity combining. The 3GPP2 term is SDU.
- Managed Network:** A network where admission, routing, and buffering decisions within the network are influenced or controlled by network management.
- MPLS:** Abbreviation for Multi protocol label switching, an IETF standard for IP service delivery.
- Packet:** An information unit identified by a label at layer 3 of the OSI reference model (Source: ITU-T I.113). A network protocol data unit (NPDU).
- Point-to-point (PTP):** A network configuration that involves only two network terminations with no routing or switching between them.
- Protocol:** A formal set of procedures that are adopted to ensure communication between two or more functions within the same layer of a hierarchy of functions (Source: ITU-T I.112).
- Quality of Service:** The collective effect of service performance that determines the degree of satisfaction of a user regarding a service. It is characterised by the combined aspects of performance factors applicable to all services, such as: service accessibility; service integrity; service operability and service retention.
- Radio access bearer (RAB):** The service that the access stratum provides to the non-access stratum for transfer of user data between User Equipment and Core Network. (Source: 3GPP).
- Radio Access Network (RAN):** The Radio Access Network provides a means of connection of mobile terminals, via a radio interface and Radio Access Network, to the Core

Network. The Radio Access Network hides radio related aspects of the user access connection from the core network.

Radio Bearer: The service provided by the RLC layer for transfer of user data between User Equipment and Serving RNC.

Radio frame: A radio frame is a basic time interval used for data transmission on the radio physical channel. In 3GPP a radio frame has 10 ms duration and it is divided into 15 time slots of 0.666 ms duration. The unit of data that is mapped to a radio frame (10 ms time interval) may also be referred to as radio frame (Source 3GPP). In the 3GPP2 system there are several kinds of radio frames. For the Sync Channel, a frame is 26.666 ms long. For the Access Channel, the Paging Channel, the Broadcast Channel, the Forward Supplemental Channel, the Forward Supplemental Code Channel, the Reverse Supplemental Channel, and the Reverse Supplemental Code Channel, a frame is 20 ms long. For the Enhanced Access Channel, the Forward Common Control Channel, and the Reverse Common Control Channel, a frame is 5, 10, or 20 ms long. For the Forward Fundamental Channel, Forward Dedicated Control Channel, Reverse Fundamental Channel, and Reverse Dedicated Control Channel, a frame is 5 or 20 ms long. For the Common Assignment Channel, a frame is 5 ms long (Source 3GPP2)

Radio interface: The tetherless interface between User Equipment and Node B or BTS (i.e., access point in radio access network). This term encompasses all the functionality required to maintain such interfaces.

Radio link: A logical association between single User Equipment and a single access point of radio access network. Its physical realisation comprises one or more radio bearer transmissions.

Radio Network Controller (RNC): The equipment in RNS, which is in charge of controlling the use and the integrity of the radio resources. In 3GPP2 it is designated by BSC (Base station controller).

RNS: The UTRAN consists of a set of Radio Network Subsystems (RNS) connected to the Core Network (CN) through Iu interfaces. An RNS consists of a Radio Network Controller (RNC) and one or more Node Bs controlled by that RNC. Each RNS is responsible for the resources of its set of radio cells and for handover decisions.

Real time service : This refers to a service where information must be delivered from the source to the destination within an agreed time delay. A shorter delay may be provided over the transport network but only if buffering can be provided in the destination. A longer delay may imply that the received information can not be processed by the destination. Even a non-real time service can change into a requirement for a real time service when information is sent from RNC to Node B to be transmitted into a predefined sequence of radio frames.

Release A: A particular version of standard produced by the 3GPP2. Release B and so on would be the following versions. In context with 3GPP Specifications, the versions are referred as release '99, release 00 and release 01 etc.

Release 99 (R99): A particular version of the UMTS standard produced by the 3GPP. Release 00, release 01, etc. are the following versions. In context with 3GPP2 Specifications, the versions are referred as release A, release B and so on.

SDU: Abbreviation for Selection and Distribution Unit. The SDU is a 3GPP2 term for the functional entity in the BSC that performs radio frame selection and distribution and soft handoff. The 3GPP term is Macrodiversity Combiner.

- Security:** The ability to prevent fraud and protect information availability, integrity and confidentiality.
- Service:** Set of functions offered to a user by an organisation.
- Services (of a mobile cellular system):** The set of functions that the mobile cellular system can make available to the user.
- Shared Channel:** A radio resource (transport channel or physical channel) that can be shared dynamically between several UEs.
- Signalling:** The exchange of information specifically concerned with the establishment, control and management of connections, in a telecommunications network (Source: ITU-T I.112).
- Soft Handover:** A procedures where the radio links are added and abandoned in such a way that the UE always keeps at least one radio link to the RAN.
- Speed:** A performance criterion that describes the time interval required to perform a function, or the rate at which certain function is performed. The function may or may not be performed with the desired accuracy (Source: ITU-T I.350).
- Transport channel:** The channels offered by the physical layer to Layer 2 for data transport between peer L1 entities. Different types of transport channels are defined depending how and with which characteristics data is transferred on the physical layer, e.g. whether using dedicated or common physical channels (Source 3GPP).
- Um:** The Radio interface between BTS and the User Equipment in 3GPP2. The identical term used in 3GPP Specifications is **Uu**.
- Universal Terrestrial Radio Access Network (UTRAN):** A conceptual term used in 3GPP Specifications for identifying that part of the network which consists of RNCs and Node Bs between Iu and Uu.
- User:** A logical identifiable entity that uses mobile telecommunication services.
- User Services Profile:** A collection of information identifying subscriber services, status and preferences.
- Uu:** The Radio interface between UTRAN and the User Equipment in 3GPP. The identical term used in 3GPP2 is Um.

4 OVERVIEW OF IP IN RAN AS TRANSPORT

The mobile systems to be considered are the UTRAN being developed by 3GPP for both FDD and TDD modes (3GPP-UTRAN) and also the RAN being developed by 3GPP2 for CDMA2000 (3GPP2-RAN). The application of IP may also be considered for other mobile systems particular where a sharing of a common transport network between 3rd generation mobile and other mobile systems might be considered to provide benefits to operators.

The main reasons for IP in the RAN, as a transport option, are cost reduction, deployment flexibility and scalability. IP in the RAN as transport option should use appropriate, existing/evolving Internet protocols to support compatibility with legacy networks and interoperability with future/next generation mobile networks. It is expected that this report should influence MWIF input to 3GPP, 3GPP2 and IETF to provide directions as to how IP can be applied within Radio Access Networks. This report should identify and resolve issues in relation to the application of IP based transport network to support Radio Access Networks for practical network implementations to verify and justify the benefits of IP as a transport option to operators and manufacturers. The result of this work should be liaison statements from MWIF to various groups within 3GPP, 3GPP2 and IETF.

The application of IP should be considered for all information transfers over all RAN internal interfaces and RAN interfaces to the core network or elsewhere. The main purpose is to consider interfaces being standardised within 3GPP and 3GPP2 although comments might be made in relation to other non-standardised interfaces.

The application of IP over RAN interfaces may be considered in relation to :

- Radio access bearers between the UTRAN and core network
- Radio bearers and radio links over the internal RAN interfaces
- RRC Signalling, between the UE and RAN, via radio bearers within the RAN
- In band signalling between network entities in the RAN (e.g. power control)
- Out of band signalling between network entities (e.g. signalling application layer protocols)
- Transfer of RAN related management information to/from management centres

Transport of IP based user services over the radio interface is a different topic from IP as a transport option. IP over the radio interface is not considered within this report, except where it might impact on the traffic flow over the RAN internal interfaces and interfaces to the core network.

The requirements of transport over RAN interfaces for various types of information transfers are considered from the work of 3GPP and 3GPP2. Then various IP based protocol architectures are considered to support transport network layer requirements over those interfaces, but without any changes to the architectures or Radio Network Layer for these systems. The target for 3GPP and 3GPP2 is to enable the transport network layer to be easily changed, or to include a range of Transport Network Layer options, without changes to Radio Network layer protocols.

To support this work, the current RAN architectures and protocols for the 3GPP-UTRAN and 3GPP2-RAN are provided. This indicates where IP is already enabled within these networks as well as the ability of those networks to easily replace the Transport Network Layer without any impact on the Radio Network Layer. The work of the IETF is considered to identify the IP developments that might be applied to RANs within 3rd generation mobile systems.

Possible practical implementation architectures for the transport network, based on IP transport, is an important part of this study to verify that IP networks can provide the performance requirements and

can provide benefits to operators. Performance may include benefits to operators in terms of implementation and operational costs of typical implementation scenarios.

Implementation architectures and traffic flow profiles include the range of RAN types, which might be of interest to operators (e.g. from small to large systems, range of user services supported, etc.). The sharing of the transport network between different mobile and non-mobile systems (e.g. 3rd generation, 2nd generation, other networks, etc.) is of interest to operators. Predictions of performance of various IP transport networks are provided to verify that IP transport networks can support the requirements of Radio Access Networks in 3rd generation mobile systems.

The changes should only be made to the Transport Network Layer (TNL) to include IP as a transport option, since the Radio Network Layer should be independent of the TNL. There could be some minor changes to the Radio Network Layer, e.g. addressing, where significant benefits can be provided to support IP as a Transport option.

Whenever possible, preference for already standardised protocols should be used, e.g. IETF protocols for the IP related parts, in order to provide wide spread acceptance and avoid the development of new protocols. Relevant 3GPP-UTRAN and 3GPP2-RAN recommendations, related to IP in the RAN as a transport option, may also be standardised in the IETF.

Where possible, common Transport Network Layer solutions should be identified for both 3GPP and 3GPP2 Radio Access Networks so the same (or similar) protocol stacks, based on IP, can be applied to both systems. It is possible that minor modification to the Radio Network Layer may be required to enable both systems to be supported on a common Transport Network Layer (or a set of Transport Network Layers) for both 3GPP and 3GPP2 systems.

Operators may wish to implement a common Transport Network Layer to support a range of mobile systems (e.g. co-located antennas and base stations for both 2nd and 3rd generation systems).

Any Transport Network Layer solution should be able to support the implementation of a range of RAN types from very small to very large systems. The Transport Network Layer should support the range of current and future services, which may need to be supported by the 3GPP-UTRAN and/or 3GPP2-RAN.

The Transport Network Layer should be able to support the range of traffic characteristics and profiles expected on different implementations of Radio Access Networks. Priority and congestion control mechanism possibly with discard during periods of overload will need to be supported.

5 RADIO ACCESS NETWORK ARCHITECTURES

This section describes the current architecture of the Radio Access Networks being developed within 3GPP and 3GPP2. It also includes an overview of the protocol models over the various RAN interfaces with an indication of where IP is already enabled as a transport option. Other mobile systems are also described, where it is also considered that IP as a transport option might also be applied to those other mobile systems.

5.1 3GPP-UTRAN Architecture and Protocol Models

5.1.1 UTRAN Architecture

The overall description and architecture for the UTRAN is described in 3GPP TS 25.401. An example of this architecture, in relation to the support of one User Equipment (UE), is illustrated in Figure 5.1.

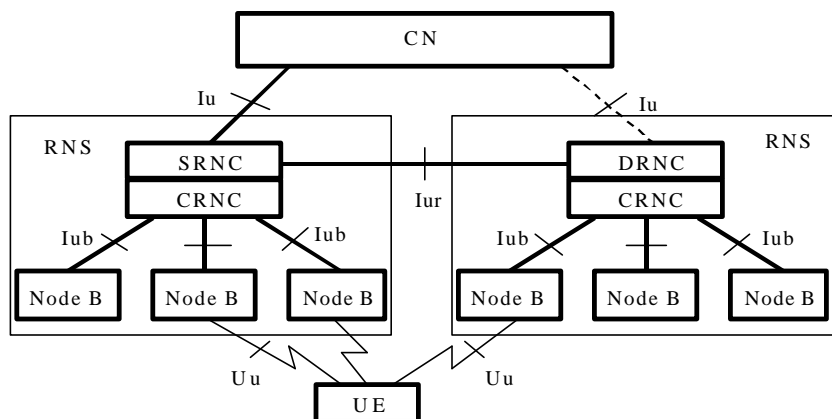


Figure 5.1 : UTRAN Architecture

The UTRAN consists of a set of Radio Network Subsystems (RNS) connected to the Core Network (CN) through the Iu interface. If the CN is split into separate domains for circuit and packet switched core networks, then there is one Iu interface (Iu-CS) to the circuit switched CN and one Iu interface (Iu-PS) to the packet switched CN for that RNS.

An RNS consists of a Radio Network Controller (RNC) and one or more Node Bs. A Node B is connected to the RNC through the Iub interface. Inside the UTRAN, the RNCs in the RNSs can be interconnected together through the Iur interface. The Iu and Iur are logical interfaces, which may be provided via any suitable transport network.

A Node B can support one or more radio cells, but the interface between a Node B and its radio cells is not being standardised within 3GPP. A Node B may support UEs based on FDD, TDD or dual-mode operation. During macro diversity (soft handover) a UE may be connected to a number of radio cells of different Node Bs and/or RNSs. Combining/splitting for soft handover may be supported within Node B, Drift RNC and/or Serving RNC. “Softer” handover provides better performance but is only possible within Node B, between radio cells connected to that Node B.

Each RNS is responsible for the resources of its set of radio cells and for handover decisions. The Controlling part of each RNC (CRNC) is responsible for the control of resources allocated within Node Bs connected to that RNC. For each connection between a UE and the UTRAN, one RNS is the Serving RNS. When required, Drift RNSs support the Serving RNS by providing radio resources, within radio cells connected to that Drift RNS.

Any RNC can take on the role Serving RNC or Drift RNC, on a per connection basis for a UE. This supports macro diversity (soft handover) when the UE roams into another RNS. Eventually a relocation process (separate to handover) may be used to reroute the Iu connection to that new RNS, after which that Drift RNS becomes the Serving RNS for that UE. Radio Access Bearers (RABs) are provided between the UE and Core Network, (via the Uu radio interface, UTRAN internal interfaces and Iu interface) for the transport of user data.. Control plane protocols provide the control of these Radio Access Bearers and the connection between the UE and the network. Control plane protocols over Uu would be carried between Radio Resource Control (RRC) entities in the UE and UTRAN, via an RRC connection over Uu, as described in TS 25.331.

5.1.2 General UTRAN Interface Protocol Models

From TS 25.401, Figure 5.2 shows the general protocol model for UTRAN Interfaces. The structure is based on the principle that the layers and planes are logically independent of each other, and if needed, protocol layers, or the whole protocol stack in a plane may be changed in the future by decisions within 3GPP standardisation groups.

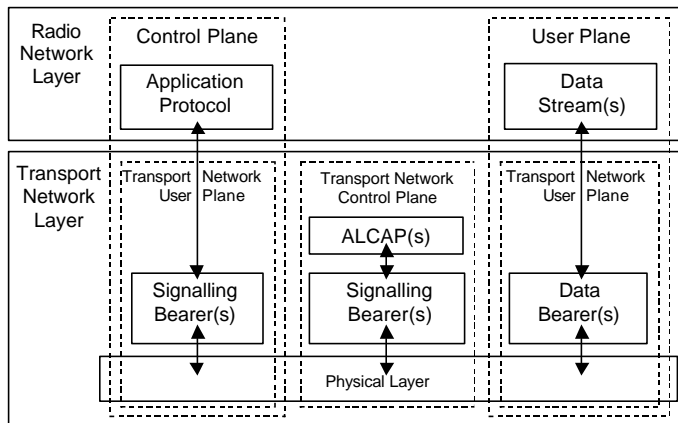


Figure 5.2 : General Protocol Model for UTRAN Interfaces

The Protocol Structure consists of two main layers, Radio Network Layer, and Transport Network Layer. All UTRAN related issues are visible only in the Radio Network Layer. The Transport Network Layer represents the standard transport technology (or technologies) which can be used over UTRAN interfaces (Iu, Iur and Iub).

The Control Plane Includes the UTRAN Application Protocols, i.e. RANAP (over Iu), RNSAP (over Iur) or NBAP (over Iub), and the Signalling Bearer for transporting these Application Protocol messages. General bearer parameters in these Application Protocols are provided that are not (directly) tied to any specific User Plane technology.

The Transport Network Control Plane includes the general ALCAP protocol(s) to set up the transport bearers for the User Plane and Signalling Bearers. The Transport Network Control Plane makes it

possible for the Application Protocol in the Radio Network Control Plane to be completely independent of the technology for Data Bearer in the User Plane.

5.1.3 Iu Protocol Models (UTRAN to CN)

The protocol models for the Iu are provided in TS 25.410 while the RANAP (over Iu) is described in TS 25.413. The Iu interface to the circuit switched CN (Iu-CS), is based on AAL2 in the user plane and SS7 on top of AAL5 for the control plane, Figure 5.3. AAL2 was chosen to minimise delay for real-time circuit switched services over the Iu interface.

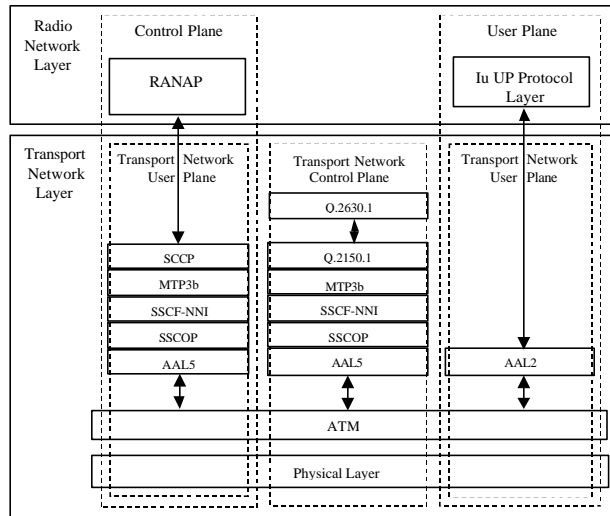


Figure 5.3 : Iu –Interface Protocol Structure towards CS Domain

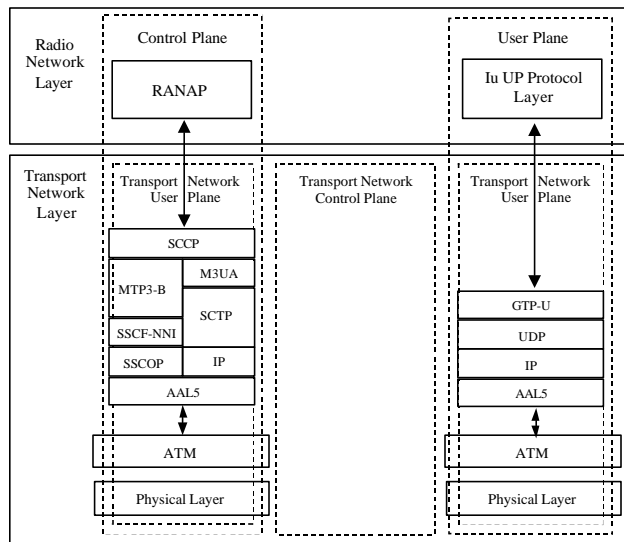


Figure 5.4 : Iu Interface Protocol Structure towards PS Domain

The Iu interface to the packet switched CN (Iu-PS) is already based on IP in the user plane with options for IP or SS7 on top of AAL5 for the control plane, Figure 5.4. Most of the Iu-PS user plane was for packet based non real time for best effort services where delay was not considered to be so important.

Transport over the Iu interface is closely related to the user traffic flow within Radio Access Bearers. Hence the traffic model over the Iu interface would be closely aligned with the traffic model for that user service.

For an AMR voice codec, the traffic flow over the Iu interface would be very similar to frames generated by the AMR codec. These AMR voice codec frames must be delivered in real time over the access stratum (e.g. between the Iu interface and the UE). These frames can not be delayed so some frames may be lost during severe radio congestion.

For (best effort) packet data over Iu, the traffic flow over Iu would closely relate to the transport of individual packets (e.g. user plane IP packets). During congestion these packets may be delayed in the UTRAN, e.g. waiting for radio resources). Hence their arrival times will be affected by congestion on the radio interface or elsewhere in the UTRAN.

5.1.4 Iur and Iub Protocol Models (Node B to SRNC)

The protocol model for the Iur interface, Figure 5.5, is described in TS 25.420 while the RNSAP protocol (over Iur) is described in TS 25.433. The Iur control plane includes options for SS7 or IP on top of AAL5. For the Iur user plane, AAL2 was chosen as many of the user plane protocol data units were small and had to be transported over Iur with minimum delay.

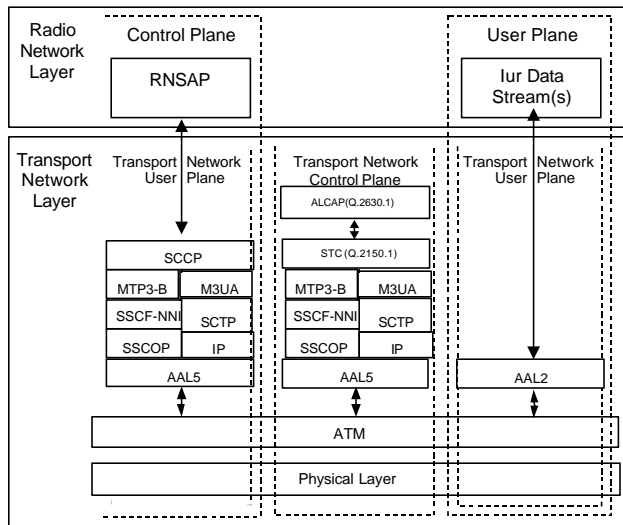


Figure 5.5 : Iur Interface Protocol Structure

The protocol model for the Iub interface, Figure 5.6, is described in TS 24.430 while the NBAP protocol (over Iub) is described in TS 25.433. The Iub control plane includes only SS7 on top of AAL5, and provides an UNI type of interface. For the Iub user plane, AAL2 was chosen as many of the user plane protocol data units were small and had to be transported over Iur with minimum delay. The Iub user plane includes various frame protocols options for the support of random access channels (RACH/FACH), dedicated channels (DCH) and shared channels (CPCH/DSCH).

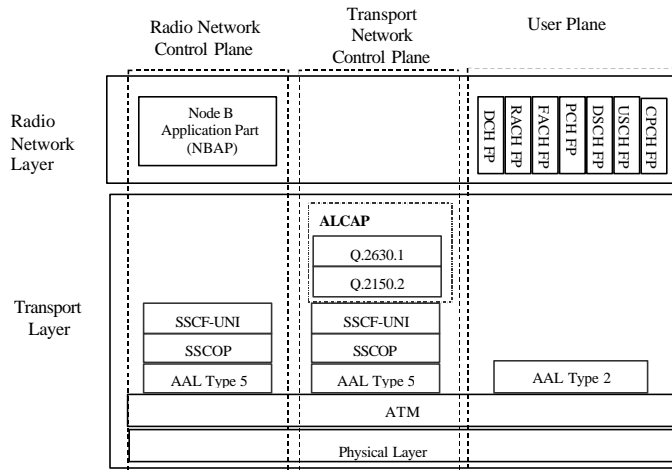


Figure 5.6 : Iub Interface Protocol Structure.

The user plane frame protocols over Iur and Iub transport data already processed by RLC and MAC layers within the RNC. This data has been prepared by the RLC and MAC in the RNC and is ready to be packed into a defined sequence of radio frames by layer 1 processing in the Node B. These frame

protocols include a reference to the radio frame number (with interleaving over 10, 20, 40 or 80 ms) in which that data must be transmitted over the radio interface.

The RLC (in the RNC) will fragment user data into these frame protocols over Iur and Iub, while the MAC layer (in the RNC) will enable a number of user services to be multiplexed. Only certain transport format combinations are allowed within these frame protocols, as defined during Radio Bearer establishment, and these formats may be restricted even more during congestion on the radio interface. The MAC layer will only accept data from the RLC layer that will comply with a defined transport format, and hence will correctly fit into a predefined sequence of radio frames.

For real time services, e.g. AMR codec, the transport formats over Iur and Iub will be designed to always carry whatever data arrives within a 20 ms AMR codec frame. For non real time services the RLC layer will fragment each user IP packet and only allow a block of data to be transferred to the MAC layer which can be carried within the next predefined sequence of radio frames (e.g. for interleaving over 10, 20, 40 or 80 ms).

During soft handover, RAB user data over Iu as well as RRC connections between the UE and UTRAN will be mapped into a number of separate Radio Bearers (via different radio cells) to the same UE. Hence the traffic flow over all these Iur and Iub interfaces during soft handover could be much greater than the RAB data traffic over the Iu for that UE.

5.1.5 Iu, Iur and Iub Transport Layer

The physical layer for the UTRAN interfaces is described in TS 25.411 (for Iu), TS 25.421 (for Iur) and TS 25.431 (for Iub). The physical media dependent layer includes options for physical layer bit rates from 1.5 Mbit/s to 622 Mbit/s.

O&M transport is already based on IP. From TS 25.442, IP datagrams containing O&M signalling are carried over the same bearer as Iub. A protocol stack for implementation specific O&M transport uses IP on top of AAL5.

Iu interface User Plane Protocols, from TS 25.415, provide data transport on top of User Data Bearer Protocols. Iub interface User Plane Protocols for Common Transport Channel Data Streams is described in TS 25.425 while Iub interface User Plane Protocols for Common Transport Channel Data Streams is described in TS 25.435.

The Iur and Iub interface user plane protocols for dedicated transport channels (DCHs) is described in TS 25.427 while the Iur and Iub Interface Data Transport & Transport Signalling for DCH Data Streams is described in TS25.426.

AAL2 was selected, as most of the protocol data units to be transported were (often) small and must be transported over Iur and Iub with the minimum delay. ATM switching provided low delay while AAL2 enables a number of small blocks of data to be packed into the same ATM cell with a low overhead.

5.1.6 Transport Channels over Iur and Iub

TS 25.401 provides an overall view of how the MAC layer is distributed over Uu, Iub and Iur for RACH, FACH and DCH. In this section the most general model is considered where both SRNC and DRNC are within separate entities, but there will be many occasions when a DRNC and Iur is not involved. Some terms used in the models are:

- **CCCH : Common Control Channel** : initial RRC signalling between UE and UTRAN (on RACH/FACH)
- **DCCH : Dedicated Control Channel** : RRC signalling between UE and UTRAN once an association has been formed between RRC in UE and UTRAN (on RACH/FACH or DCH)

- **DTCH : Dedicated Traffic Channel** : between UE and UTRAN for the transfer of user data.

For the RACH or FACH transport channel, Figure 5.7, dedicated MAC (MAC-d) provides the multiplexing of different traffic flows for the same UE, with queuing in the RLC layer as appropriate, e.g. for non real time services. The RACH or FACH is supported by a Common MAC (MAC-c) in the CRNC for the multiplexing of traffic flow for different UEs with queuing and priority as appropriate. The transport channels for the shared channels (CPCH/DSCH) have similar protocol models, which can be found in TS 25.401, but with MAC-c replaced by MAC-sh.

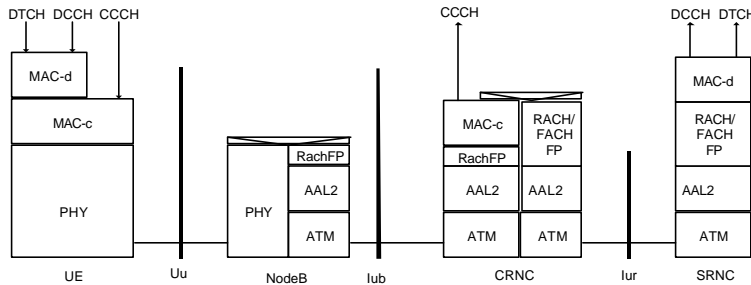


Figure 5.7 : RACH or FACH : Separate Controlling and Serving RNC

The DCH transport channel is dedicated to a specific UE, Figure 5.8. The Dedicated MAC (MAC-d) provides the multiplexing of different traffic flows for the same UE, with queuing in the RLC layer as appropriate, e.g. for non real time services. There is no MAC-c or MAC-sh as this channel is dedicated to one UE only. Some physical layer functions are included within the DRNC or SRNC, but only for soft handover combining/splitting.

The transport formats for these transport channels over Iur and Iub are constrained to a predefined set of transport format combinations for the number of bits, from each Radio Bearer for that UE, which can be accepted for transmission within a predefined sequence of radio frames. The MAC layer will ensure that only data formats that will fit within one of the acceptable transport format combinations is sent to the radio interface, i.e. over Iur and Iub. During periods of congestion (or high levels of radio interference) the allowed set of transport formats may be further reduced.

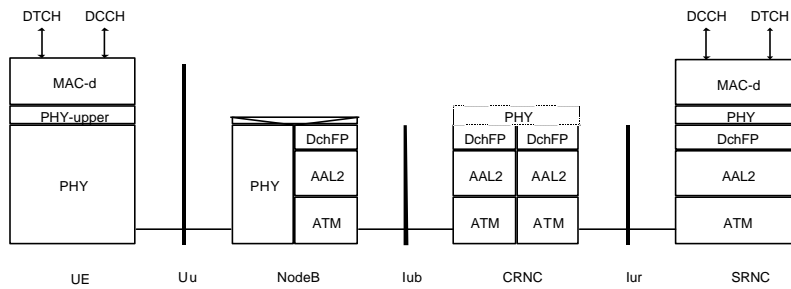


Figure 5.8 : DCH: Separate Controlling and Serving RNC

5.2 3GPP2-RAN Architecture and Protocol Models

5.2.1 General IOS Architecture

3GPP2 TSGA released the Inter Operability Specification (IOS) Version 4.0 in June 2000. This standard is based on TIA/EIA/634 and CDG IOS v3.1.0 and supports the IS-95 and cdma2000 systems. Figure 5.9 shows the interfaces specified by the IOS standards along with their protocol stacks:

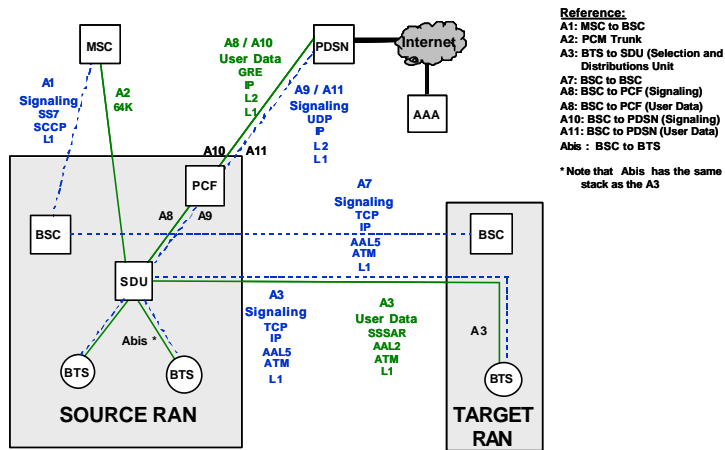


Figure 5.9 : 3GPP2 Inter Operability Specification (IOS) Functional Diagram

This standard describes the overall system functions, including services and features required for interfacing a Base Station (BS) with the Mobile Switching Centre, with other Base Stations, and with the Packet Control Function (PCF); and for interfacing the PCF with the Packet Data Service Node (PDSN). In addition, 3GPP2 is in the process of balloting a specification for the A-bis interface (PN-4604).

5.2.2 Protocol Models for RAN Interfaces to Core Network

The interfaces defined in this standard, between the RAN and core network, are described below.

- A1** The A1 interface carries signalling information between the Call Control (CC) and Mobility Management (MM) functions of the MSC and the call control component of the BS (BSC).
- A2** The A2 interface carries 64/56 kbit/s PCM information (voice/data) between the Switch component of the MSC and one of the following: Channel element component of the BS (in the case of an analog air interface), and Selection/Distribution Unit (SDU) function (in the case of a voice call over a digital air interface),
- A5** The A5 interface carries a full duplex stream of bytes between the Interworking Function (IWF) and the SDU function.
- A8** The A8 interface carries user traffic between the BS and the PCF.
- A9** The A9 interface carries signalling information between the BS and the PCF.

A10 The A10 interface carries user traffic between the PCF and the PDSN.

A11 The A11 interface carries signalling information between the PCF and the PDSN.

The following are the protocols stacks for IOS interfaces:

A1 Interface:

IOS
Application
SCCP
MTP3
MTP2
MTP1
Phys. Lyr.

A9 / A11 Interface (signalling connection):

IOS
Application
UDP
IP
L2
Phys. Lyr.

A2 Interface (user traffic):

56/64 kbits/sec PCM
DS0

A8 A10 Interfaces (user traffic):

GRE
IP
L2
DS0

5.2.3 Protocol Models for RAN Internal Interfaces and between RANs

The interfaces defined in this standard, within the RAN and between RANs, are described below.

A3 The A3 interface carries coded user information (voice/data) and signalling information between the SDU function and the channel element component of the BS (BTS). This is a logical description of the endpoints of the A3 interface. The physical endpoints are beyond the scope of this specification. The A3 interface is composed of two parts: signalling and user traffic. The signalling information is carried across a separate logical channel from the user traffic channel, and controls the allocation and use of channels for transporting user traffic.

A7 The A7 interface carries signalling information between a source BS and a target BS.

Abis Interface : The Abis interface connects the BSC to the BTS and is not standardized in IOS 4.0.

A3 Interface (signalling connection):

IOS
Application
TCP
IP
AAL5
ATM
Phys. Lyr.

A3 Interface (user traffic subchannel):

User Traffic Frame
AAL2
ATM
Phys. Lyr.

A7 Interface (signalling connection):

IOS
Application
TCP
IP
AAL5
ATM
Phys. Lyr.

5.3 Other Mobile Systems

The main focus of this technical report is to consider IP in the RAN as a transport option within 3rd generation CDMA mobile systems. At a later point in time the application of IP in the RAN as a transport option within 2nd generation mobile systems may be considered, in particular, where such systems share a common transport network with 3rd generation mobile systems.

Since the details of 2nd generation systems are different than 3rd generation systems, the simulation results in this document are not generally relevant to 2nd generation systems and new simulation models will be required. A particular barrier to using IP on 2nd generation systems is that the RAN protocols are proprietary, so any detailed analysis would have to be performed on a case by case basis. The analysis of IP in 2nd generation systems may be worthwhile because of the possible benefits of a common transport network for both 2nd and 3rd generation mobile systems.

The results in this document are not applicable to mobile wireless networks that do not require an extensive wired access network, such as 802.11 wireless LAN and Bluetooth wireless PAN. These wireless technologies include an extensive media access protocol at layer 2 that removes the need for a specialised radio access network. Consequently, IP is possible and deployed (for 802.11) up to the wireless access points, and even across the radio link.

6 TRANSPORT NETWORK REQUIREMENTS

This section describes the high level requirements for the IP in the RAN as a transport option for both 3GPP and 3GPP2. The objective is to consider the requirements of the Transport Network Layer to support the various interfaces within the 3GPP-UTRAN, 3GPP2-RAN and possibly other mobile systems. Many of these requirements might be independent of the specific type of Transport Network Layer while some may be specific to an IP based Transport Network Layer.

This section may contain other Transport Network requirements of interest to operators (e.g. cost benefits).

6.1 IP transport flexibility

By defining protocol stacks on the RAN interfaces, one may not make any restrictive assumption on IP transport network topology. They shall adapt to a wide range of networks (LAN to WAN) and no preference shall be expressed on routed vs. point to point networks. Optionally, UTRAN IP transport is to support transport over non-trusted networks and QoS mechanisms have to account for the presence of background traffic.

6.2 Layer 2 / Layer 1 independence

Higher layers should be independent from Layer2/Layer1. The IP transport network layer is defined for multiple layer 2s.

6.3 Coexistence of IP and ATM transport options

The RAN may have both ATM and IP transport networks. The following requirements with regards to ATM and IP transport network coexistence shall be met:

- The specifications shall ensure the co-existence of ATM and IP Transport options within RAN, i.e. parts of RAN using ATM and parts of RAN using IP transport.
- ATM and IP Transport Options shall rely on the same functional split between Network Elements for the scope of this document (MTR-006) as described in Chapter 5 for the 3GPP-UTRAN and 3GPP2-RAN.

6.4 Quality of Service

The mechanisms to secure the quality of service parameters, timing aspects, and packet loss shall be considered. Quality of service parameters includes service class definition and congestion control requirements. Timing aspects include delay and delay-variation requirements.

The UTRAN user plane has to support the QoS and delay requirements of the end user application. In turn, the IP transport has to support the QoS and delay requirements of the UTRAN user plane.

6.5 Reliability

In addition all the services with various traffic profiles should be supported with high reliability over RAN interfaces.

6.6 Efficient utilisation of transport resources

Efficient use of the bandwidth of the transport network shall be considered, e.g. by reducing the protocol overhead (via Header compression, multiplexing, etc.).

RAN protocols shall operate efficiently on low speed point to point links, which may be shared with other traffic (e.g. GSM/GPRS Abis, UMTS R99 compliant interfaces)

RAN shall support necessary compression and multiplexing protocols for bandwidth restricted links. Only some links along the path for a logical interface may be bandwidth limited. It is therefore not necessary to header-compress and to multiplex on all links of this interface.

The following requirements are also important for efficient utilisation of transport resources :

- Different schemes shall operate independent of each other
- Schemes for efficient bandwidth utilisation shall be optional (as they are not required in every deployment scenario)
- These schemes shall be applicable for both a single hop logical interface and end-to-end between UTRAN entities

6.7 Security

It must be possible to implement a RAN utilising both public and private IP networks without compromising the security/integrity of the user data, the radio network signalling, or the network elements.

The following requirements shall be reviewed, verified and enhanced by the MWIF security task force:

- Firewall functions shall be implemented on network entities that directly connect with external networks/network elements. All RAN interface with the external network(s) must be secured. Traffic through these interfaces must be protected (via Virtual Private Network services if necessary).
- All network elements must be able to prevent unauthorised access. The kinds of unauthorised access and the threats definition still need to be defined.
- Border routers (such as routers connecting different RANs) which exchanges routes with external routing entities should be able to set-up authentication mechanism between routing peers. This is a wise requirement for the transport network but it may imply constraints/functions on BTS and RNC. It is a matter for the operator deploying or subcontracting its transport network.

6.8 Addressing

6.8.1 Addressable Entities

Network elements, e.g. RNC, Node B, need to be identified by one or more IP addresses. In an IP based RAN, the transport network has to provide the means to uniquely address individual flows – both in the user as well as signalling planes

6.8.2 General IP Addressing Requirements

The following are general IP addressing requirements:

- IP addressing in the RAN shall be logical and should not have any dependency on network element or interface type.
- In case of IPv4, to ensure efficient usage of IPv4 addresses and routing efficiency, the IP based RAN shall adopt the classless IP addressing scheme, using Variable Length Subnet Masks (VLSM).
- The IP addressing in the RAN scheme must support hierarchical routing network design and work well with the chosen routing protocols to provide the best route convergence time in order to avoid network instability.
- Where applicable, IP addressing in the RAN must budget for multi-homing of network elements.
- IP addressing in the RAN must be scalable and take network element/interface growth and network expansion into consideration.
- The RAN IP Addressing scheme must be flexible and be suitable for different RAN sizes and topologies.
- IP addressing in the RAN must allocate addresses efficiently.

7 IP TRANSPORT

This section considers the standardisation of IP and related protocols within standards groups (e.g. IETF) and considers the options available for the support of the Transport Network Layer requirements within the 3GPP-UTRAN, 3GPP2-RAN and possible other forms of RANs (e.g. second generation mobile systems).

The intention of this section is to provide a background to the developments of IP and related protocols that may be applied as transport options within the RAN. The application of IP in the RAN as a Transport option is described in the next section. This section only relates to the specific capabilities of IP but should cover all the relevant features of various IP options and related protocols that might be applied within the RAN.

7.1 QoS Differentiation

There are three commonly used models for end to end quality of service. These are: best effort, integrated and differentiated services. A service model, also referred to as a level of service, describes a set of end-to-end QoS capabilities. A different technique for achieving end to end quality of service is MPLS. The three models and MPLS are described in subsequent sections.

When discussing QoS we need to introduce the concept of fairness. Fairness within a network implies that all endpoints, clients and servers, within a network are treated somewhat equally. For example, Clients with higher bandwidth connections achieve higher throughput without starving lower bandwidth clients. During periods of congestion, all endpoints experience reduced throughput while starvation of any single endpoint is avoided.

The concept of a Managed Network is introduced when discussing QoS within IP networks. A Managed Network is a network where admission, routing, and buffering decisions within the network are influenced or controlled by network management. IntServ and DiffServ IP networks are Managed Networks while a Best Effort IP Network is unmanaged.

7.1.1 Best effort service

Best effort is a single service model in which an application sends data whenever it must, in any quantity without providing prior information to the network. For best-effort service, the network delivers data if it can without any assurance of reliability, delay bounds or throughput.

A best effort IP network is an Unmanaged Network. The best of example of a large unmanaged IP network is the Internet.

7.1.2 IntServ service model for QoS differentiation

Integrated service is a service model that can accommodate multiple QoS requirements. In this model the application requests a specific kind of service from the network before sending data. The request is made through explicit signalling [RFC2205]. QoS is performed on a per flow basis.

The application informs the network of its traffic profile and requests a particular kind of service that can meet its bandwidth and delay requirements. Data that is sent before the confirmation, after the service expires, or above the service bandwidth may be sent best effort, may be dropped, or queued indefinitely. The network performs admission control, based on information from the application and available network resources. It also commits to meeting the QoS requirements of the application as long as the traffic remains within the profile specifications. The network fulfils its commitment by maintaining per-flow state as well as performing packet classification, policing and intelligent queuing based on that state. Many applications do not support the IntServ QoS Signalling protocol

RSVP. In this case the first hop router of the IP network can become an IntServ proxy for the application. That first router can query the port numbers of the IP packet to determine the application and to then initiate an RSVP request for the application. In this case, packets are delivered from the hosts to the first hop router with best effort service, but beyond the first hop router IntServ service is applied to the same packets. For example, applications running on hosts connected to an Ethernet may use the gateway router of the Ethernet to initiate RSVP requests as long as the Ethernet is properly designed to not inject bandwidth or delay limitations which are less than the RSVP request.

There are two types of Integrated services [RFC2210], namely guaranteed rate service [RFC2212] and controlled load service [RFC2211]. RSVP can be used to signal QoS requirements for both services to the router:

- **Guarantee Rate Service** : provides firm bounds on end-to-end datagram queuing delays. This service makes it possible to provide a service that guarantees both delay and bandwidth. Weighted Fair Queuing may be used to provide this kind of service
- **Controlled Load Service** : provides client data flow with a QoS closely approximating the QoS that same flow would receive from an unloaded network element, but uses admission control to assure that this service is received even when the network element is overloaded (definition taken directly from the RFC to be precise). A network element may employ any appropriate scheduling means to ensure that admitted flows receive appropriate service.

7.1.3 DiffServ service model for QoS differentiation

Differentiated service is a service model that can satisfy differing QoS requirements. However, unlike the integrated service model, an application using differentiated service does not explicitly signal the router before sending data. For differentiated service, the network tries to deliver a particular kind of service based on the QoS specified by each packet. This specification can occur in different ways, for example, using the IP Precedence bit settings in IP packets or source and destination address. The network uses the QoS specification to classify, shape and police traffic as well as to perform intelligent queuing. The differentiated service model is used for several mission-critical applications and for providing end-to-end QoS. Typically, this service model is appropriate for aggregate flows because it performs a relatively coarse level of traffic classification.

The IETF Diffserv WG presently has three standards track RFCs (RFC 2474, RFC 2597, and RFC 2598) and one informational RFC (RFC 2475). Conformance to RFC 2474 and implementation of its Code Selector Point Per Hop Behaviour Group is alone sufficient to claim Diffserv compliance. Expedited Forwarding ensures delivery of good quality voice. Assured Forwarding PHB group provides means for forwarding of IP packets in a number of independent classes. Within each of the classes, IP packets can be assigned to a given number of drop precedence.

7.1.4 Application of MPLS

Multi Protocol Label Switching (MPLS) uses a label to forward packets instead of an IP header. Label switching can be performed much faster than IP header forwarding. An MPLS label is much like an ATM virtual circuit identifier. At the edge of the MPLS network, a label is added to each IP packet containing information that alerts the next hop MPLS router to forward the packet in a pre-defined path. As a packet traverses from one router to another, it may be re-labelled to travel in a more efficient path. The final edge MPLS router strips the MPLS label thus leaving the original IP packet. As far as the original packet is concerned, the routers carrying it through the MPLS network appear as a single hop. Label switched paths (LSPs) can be set-up between a source and destination MPLS router using a label distribution protocol. The Label Distribution Protocol [LDP] is one of the better-known label distribution protocols. In addition, existing routing protocols, such as BGP, can be

extended to distribute MPLS labels. To set up traffic-engineered LSPs, the RSVP-TE and the CR-LDP protocols can be used.

The key advantage of MPLS is the QoS guarantee to traffic with network resource efficiency. It also provides many other features such as traffic protection, explicit and source routing, traffic engineering, and fast packet forwarding. These features allow a network operator to use their network resources in an efficient way while providing QoS guarantees to their customers.

Specifically, MPLS provides the following mechanisms for supporting the QoS requirements of IP flows:

1. **Traffic Engineered Paths.** MPLS uses the label prefixed to an IP packet to determine the path that the packet will take through the network, regardless of the IP addresses contained in the packet. Routes through the network can be engineered to meet the QoS requirements for each class of traffic supported by the network. The traffic at the edge of the MPLS domain can be segregated according to QoS class and the packets can be directed along the MPLS paths defined over the route that meets their QoS requirements. These kinds of guarantees are impossible to achieve in a pure IP routed network unless one massively over-engineers the capacity of the network
2. **Integration with Differentiated Services (DiffServ)** DiffServ provides a mechanism for defining the treatment that a packet will receive as it is forwarded through an IP network. Although there are no performance guarantees with DiffServ, it can be used to improve end-to-end performance over large scale, wide area networks. MPLS can support DiffServ in two ways:
 - By using the DiffServ marking in each packet to determine which path the packet should be sent over. Paths can then be engineered, as in (1), to provide more deterministic performance guarantees than are available with pure DiffServ in a routed network.
 - By using the DiffServ marking in each packet to determine the treatment that packets will receive over a specific path. In this model, closely resembling the basic DiffServ model, packets with different QoS requirements can be carried over the same MPLS path. Within that path, the DiffServ marking is used to prioritise and schedule packets to provide “better” treatment for some packets with respect to other packets carried over that same path.
3. **In-Sequence Packet Delivery.** Because the route that a packet will travel through the network is precisely defined by the LSP, it is guaranteed that packets are received in the same order that they were transmitted.

MPLS Virtual Private Networks (VPN) is another concept where VPN tunnels are created using MPLS and IPSec in a public IP network. This also favours a cellular operator to lease MPLS VPN tunnels similar to FR and ATM circuits thus reducing the network cost. Virtually all the interfaces in an IP based RAN could be implemented via a MPLS VPN tunnels leased from a public carrier. MPLS VPN services are already available through major telecom operator across the globe.

The current efforts in IETF are focused on defining a MPLS architecture and associated label distribution protocols. At the same time, efforts are underway to define the traffic engineering and voice over IP over MPLS (VoIPoMPLS) standards that will allow the use of MPLS for priority traffic. Multiplexing at MPLS layer is also being studied, which will allow small size packets to share a single label thus lowering header overhead.

7.1.5 QoS for ATM Transport

The most common method of mapping IP onto ATM and providing differentiated QoS is:

- Provision of separate ATM virtual circuits for various classes of traffic with varying guarantees of bandwidth – CBR, VBR or ABR.
- Provide a mapping of IP precedence/type of service bits to the virtual circuits.

However, care has to be taken that IP traffic of a lower priority does not “overtake” traffic of a higher priority. This might occur because the assigned ATM transport capacity of lower priority IP traffic is idle, while higher priority IP traffic needs to queue in order to get access to its assigned ATM transport capacity.

7.2 Link Efficiency Mechanisms

Link Efficiency mechanisms seek to optimise the throughput and/or delay over relatively slow links, e.g. T1/E1 or slower. When considering a link efficiency mechanisms, switchability is a concern. If a mechanism is not switchable, then every hop within the path of the packets must perform additional, and often CPU intensive, functions to route the packet.

7.2.1 Link Fragmentation and Interleaving (LFI)

Link fragmentation and interleaving describes a technique to ensure an upper bound on delay for one class of packet while occasionally sacrificing delay on another class of packets.

The IETF protocol that implements LFI is Multilink PPP, RFC1990 [RFC1990]. Multilink PPP enables two classes of packets by allowing packets over a link to be Multilink encapsulated or PPP encapsulated. Multilink encapsulated packets may be fragmented to achieve some maximum delay boundary for PPP packets which are interleaved between the fragments of a multilink packet. More recent work, in the Multiclass extension to Multilink PPP, RFC 2686 [RFC 2686], has extended the number of classes to allow more than two .

Multilink PPP is not routable. However, Multilink PPP within Layer 2 Tunnelling Protocol, RFC 2661 [RFC2661], is switchable. The Maximum Transmit Unit (MTU) defines the maximum packet size for packets that may be transmitted over a link. For example, Ethernet has an MTU of 1 500 bytes while X.25 has 128 bytes.

MTU discovery, RFC1191 [RFC1191], may be used as a weak delay guarantee mechanism by limiting the MTU value to be the upperbound of delay for a class of packets. A smaller MTU forces the IP layer to fragment larger IP packets into multiple small packets. In a similar technique to LFI, one class of packets may be interleaved between the multiple IP fragments of another packet class.

MTU discovery may work in some small networks but not in a large network with multiple service providers. MTU discovery may delay the initial packet reception by several seconds and relies on ICMP messages from hops within the network. Also, IP fragmentation adds significant overhead to the transmission of the fragmented packet because each fragment contains an IP header.

7.2.2 Multiplexing Schemes

Multiplexing schemes seek to reduce the overhead to payload ratio by including multiple payloads within the same packet. There is usually some minor additional overhead needed to define the boundaries of the multiple payloads. Multiplexing schemes are most useful with large headers such as an IP header, or when a packet is fixed length such as ATM.

The IETF is standardising a layer 2 multiplexing scheme using PPP multiplexing. In the proposed draft [PPPMUX], PPP multiplexing seeks to combine multiple payloads within the same PPP packet.

Multiplexing can be done above Layer 3 or at Layer 2. LIPE [LIPE] and CIP [TSG1718] are two proposals that apply multiplexing above Layer 3 while PPPMux is a Layer 2 multiplexing proposal.

The proposed draft [PPPMUX] on PPPMux allows multiple payloads to be multiplexed within the same PPP packet. The payloads from different users are identified by their source and destination IP addresses and UDP port numbers, as specified in the 3GPP RAN contribution [TSG1651]. The LIPE draft [LIPE] specifies how in-band signalling messages can be used to assign a user identifier.

PPP multiplexing alone is not switchable. If the PPP mux packet is encapsulated in L2TP according to draft-ietf-avt-tcrtp-00 [TCRTP] then the multiplexed packet is switchable. Also the PPP mux packet is switchable if encapsulated in MPLS [CMPLS].

7.2.3 Header Compression Techniques

Header compression in the TNL may be used to reduce the IP overhead per flow or per aggregated flow, typically over a bandwidth limited link like the last mile to the Node B. There is no need to consider header compression of the user plane. The latter is handled between the RNC and the UE in a manner transparent to the TNL.

7.3 Size for IP packets

7.3.1 Maximum Size for IP Packets

The following items justify the reasons why a maximum size shall exist for IP packets:

- A parameter (MTU: Maximum Transmission Unit) is defined for each network by the link layer, to which higher layers must adapt. Routers are required to forward IP packets up to 68 bytes without fragmenting them, as stated in RFC791. Hosts are not required to receive IP packets larger than 576 bytes. These values reflect only the minimum MTU, but in general it is constrained by layer 2 technologies. Typical values are 1500 bytes for Ethernet and 4470 bytes for FDDI.
- On the last mile link between a Node B and an Edge Router, one forwarded IP packet pre-empt the access to the medium for a duration proportional to the payload size. In order to guarantee some Quality of Service, a limit must be put on the packet size, so that low priority packets cannot block real time packets. On a 2 Mbps link, a 3 ms link blocking corresponds to 750 bytes.
- In an IP network, the deployment of QoS features is not sufficient to ensure guarantee of service. The network must be correctly dimensioned, so that the expected service can be provided. The provisioning of resource must be done with some over-dimensioning factor depending on the maximum packet size. The bigger the real-time packets, the more resource will be necessary.

For these reasons, a maximum size is expected for IP packets transporting user and control plane data. The actual size depends on the MTU and capacity of the link layer.

7.3.2 Size of Frame Protocol payloads to transport

User plane Frame Protocol payloads to be transported over the Iub and Iur interfaces may be of very different sizes. FP PDUs carrying voice (e.g., those generated by AMR codecs) are small by nature while PDUs carrying data may become very large.

Voice Frame Protocol payloads typically have a size between a minimum of 13 bytes when in the OFF state and a maximum of 50 bytes when in the ON state (see Section 9.6.2).

In order to estimate the maximum size of a FP PDU a 'worst case' is regarded. Assuming the maximum FDD net data rate of 384 kbps and a maximum TTI of 80ms this would result in a possible maximum RLC block size of 3840 bytes (384,000 bits * 80ms / 8bits) every 80ms.

It is clear that these Frame Protocol payloads will not fit into small IP packets. In order not to impact Radio Network Layer protocols, a segmentation/re-assembly function is needed in the Radio Transport Layer. On the other hand, a multiplexing functionality is also needed that aggregates small voice FP PDUs into one IP packet to amortise for header overhead.

7.4 Last Mile Quality of Service Issues

The last mile links are usually low bandwidth links, e.g. one or multiple T1/E1s, while the links between the IP network routers will have a much higher capacity. In order to assure QoS on the slow, last mile links, the transmission time for a packet (which is proportional to its length) has to be limited so that low priority packets cannot block higher prioritised real time packets. A mechanism is required that limits the packet size by segmenting long FP PDUs into smaller IP packets. There are three possibilities:

- Fragmentation on a layer below IP (data link layer) - Examples of link layers providing fragmentation are Multilink PPP and ATM. Fragmentation at the data link layer by using Multilink PPP is described in Section 7.2.1. It either works on a hop-by-hop basis with link layer fragments being reassembled at the end of each link or uses a layer 2 tunnel with the fragments being reassembled at the end of the tunnel.
- Segmentation on a layer above IP - Segmentation of data can also take place above the IP layer. This kind of segmentation works on an end-to-end basis with the FP PDUs being fragmented once in the source and being reassembled at the destination. The CIP and the LIPE approach described in Chapter 8 propose this kind of segmentation.
- Fragmentation on IP layer (IP fragmentation) - If segmentation is not supported at layer 2 or above layer 3 then IP layer fragmentation is required. This kind of fragmentation should be avoided because of the poor efficiency.

A short description and an investigation of usability for IPv4 and IPv6 fragmentation are presented in the next sections.

7.4.1 Fragmentation in IPv4

In IPv4, any intermediate IP layer on a path between two hosts can fragment an IP packet to adapt to layer 2 maximum size (MTU). The IP packet originator can optionally forbid that fragmentation. The fragmentation is indicated with a set of fields in the IP header [IP]. In IPv4, these fields are mandatory and are 4 bytes long. A fragmentation is indicated with the flags and the fragment offset field.

When a fragmentation is indicated, the IP header cannot be compressed. This fact is mentioned in [IPHC]. The text implies that, when an IP packet corresponds to a fragment, the full IP header must be sent in all circumstances and there is no gain from header compression. This is very restrictive since header compression has been foreseen in proposed solutions for user plane IP transport.

Another important concern is the implementation of this feature. It is clear that many applications or transport protocols are aware of MTU size and adapt their payload size to this value. Not so many packets are fragmented in IP networks, which implies that this feature is rarely implemented in hardware. That could be a difficult concern to both fragment packets and provide fast forwarding in intermediate routers.

Many applications or transport protocols propose means to cope with small IP packets for large payloads. For example, TCP and SCTP [RFC2960] propose solutions for this issue.

7.4.2 Fragmentation in IPv6

In IPv6, there is no support for fragmentation at the IP layer on routers [IPv6]. The header field identifying fragments is no longer supported in IPv6, and only end nodes can fragment packets. Instead, IPv6 provides a special fragmentation header that end nodes can use to identify fragment packets. The fragmentation header has a length of 8 bytes, which can be compressed to 6 bytes.

End nodes are given means to discover the MTU between them, using the P-MTU (Path MTU) discovery [RFC1981]. If an end node performs P-MTU and discovers that some link does not support an MTU length long enough for its packets, fragmentation can be performed by the end node. The end node uses the IPv6 fragmentation header to identify the fragment packets. The corresponding end node can use the fragmentation headers to identify and reassemble the original packet.

Additionally, IPv6 requires that every link support an MTU of 1280 bytes or greater. If a link cannot convey a 1280 byte packet in one frame, fragmentation and reassembly must be provided at L2.

7.5 Routing

7.5.1 Addressing

This study area is related to all addressing issues with regards to the introduction of an IP Transport Network. Also, addressing issues relating to inter-working with AAL2/ATM nodes should be considered.

Classless vs Classful addressing scheme

Classless IP addressing schemes with variable length subnet masks allow for efficient use of the available IP address space.

DHCP

DHCP enables dynamic allocation of IP addresses for entities in the radio access network as well as provisioning. Addresses could be allocated either permanently or for a fixed period of time. For network elements in the RAN, permanent addresses could be considered as the preferred option.

Inter working with ATM

When interworking with AAL2/ATM nodes, there is a need to map the ATM multiplexing identifiers (CIDs) to an IP address and possibly a UDP port address combination. This can be achieved in a static fashion or dynamically through the use of protocols like Megaco [RFC2885].

7.5.2 Routing aspects

The physical architecture of the UTRAN may require point-to-point communication for some interfaces (e.g. as per the current Iub specification) and routed for others (e.g. the Iur) while maintaining the same or similar protocol stacks. This encompasses both static and dynamic routing

The following routing protocols should be considered as options:

- RIP and RIPv2
- OSPF
- IS-IS

The main considerations for the choice of a routing protocol are convergence latency, scalability and ease of manageability.

7.5.3 Multicast routing

IP Multicast allows one packet to be sent to a group of registered IP hosts without having to replicate the packet for each individual destination. This saves bandwidth within the network. IP Multicast could be a useful technique to carry wireless applications such as paging etc.

The following multicast routing protocols should be considered as options:

- Multicast Extensions to OSPF (MOSPF – RFC 1584)
- Core Based Trees (CBT) Multicast Routing (RFC 2189)
- Distance Vector Multicast Routing Protocol (DVMRP). This scheme is captured in IETF draft [DVMRP]
- Protocol Independent Multicast– Dense Mode (PIM-DM)
- Protocol Independent Multicast– Sparse Mode (PIM-SM). This scheme is still in IETF draft [PIM]

Similar to routing protocol selection criteria, the main considerations for the choice of multicast routing protocol are convergence latency, scalability and ease of manageability. In addition, the choice of multicast routing protocol should also take the multicast network topology (such as sparse vs. dense) into consideration.

7.5.4 Tunnelling

Layer 3 tunnelling in general is an encapsulation mechanism that encapsulates IP datagram (carried as payload) within another datagram. It provides a means to alter the normal routing for IP datagrams, by delivering them to an intermediate destination that would otherwise not be selected based on the (network part of the) IP Destination Address field in the original IP header. Once the encapsulated datagram arrives at this intermediate destination node, it is de-capsulated, yielding the original IP datagram, which is then delivered to destination indicated by the original Destination Address field.

The following tunnelling protocols should be considered as options:

- IP Encapsulation within IP (RFC 2003): This protocol specifies a method by which an IP datagram may be encapsulated within an IP datagram. This technique may serve a variety of purposes, such as delivery of a datagram to a mobile node using Mobile IP.
- Minimal Encapsulation within IP (RFC 2004): This protocol specifies a method by which an IP datagram may be encapsulated within an IP datagram, but with less overhead than IP Encapsulation within IP (i.e. RFC 2003).
- Generic Routing Encapsulation (GRE – RFC 1701): This protocol provides a general purpose tunnelling mechanism for protocols such as IPX, IP etc.

There is also a layer 2 tunnelling mechanism called Layer 2 Tunnelling Protocol [RFC2661]. PPP [RFC1661] defines an encapsulation mechanism for transporting multi-protocol packets across layer 2 point-to-point links. L2TP extends the PPP model by allowing PPP endpoints to reside on different devices interconnected by a packet-switched network.

7.6 Security

Security considerations require identification of trusted and untrusted endpoints. For example, an IP BSC would be a trusted endpoint because it is under control of the service provider. The wireless terminal would likely be an untrusted endpoint because the terminal is under control of the end user.

Typically the service provider network elements are trusted and anything outside of the service provider control is untrusted. The devices at the boundary of trusted and untrusted elements require security features to prevent abuse of the service provider network and elements within the network.

Several IP security protocols are standardised or in progress. RADIUS [RADIUS] supports endpoint authentication using encrypted password. Diameter [AAA] is a backward compatible extension of RADIUS that allows definition of application profiles supporting different kinds of access control and authentication. IPSEC [IPSEC] supports secure tunnels between endpoints with and without encryption.

7.7 Availability

High availability is a function of all the elements in a network including the devices, the links, the protocols and the applications. While the mean time between failure of individual components is a factor, network availability is determined mostly by the network design. Adding redundancy enables a network to operate even though elements in the network fail. However, layer 2 and layer 3 services and protocols must be carefully employed to effectively utilise the redundant components. The following sections described some of services/protocols that can enhance availability in an IP based transport network.

7.7.1 IP Routing Protocol

When redundant routers are used, there will be multiple paths to a given destination. The routers can take advantage of these multiple paths, not only for load balancing, but also for using an alternate path in order to route around a failed router.

Convergence is the process of agreement, by all routers, on optimal routes. When a network event causes routes to either halt operation or become available, routers distribute routing update messages.

When network topology changes, network traffic must reroute quickly. The phrase “convergence time” describes the time it takes a router to start using a new route after a topology change. Routers must do three things after a topology changes:

- Detect the change
- Select a new route
- Propagate the changed route information

Convergence is also affected by the complexity of the algorithm used to calculate routes. The time required to run the algorithm depends on a combination of the size of the area and the number of routes in the database.

The convergence behavior of layer 3 routing protocols can be tuned as specified below:

- **Update interval:** Rate at which routing updates are sent. This is the fundamental timing parameter of the routing protocol.
- **Invalid interval:** Interval of time after which a route is declared invalid.
- **Hold-down interval:** Interval during which routing information regarding better paths is suppressed.

7.7.2 Virtual Router Redundancy Protocol (VRRP)

Defined in RFC 2338, VRRP is designed to eliminate the single point of failure inherent in the static default routed environment. It allows end hosts to use a single default gateway router address. One or more routers on a LAN are configured to use a virtual MAC address and virtual IP address – essentially a virtual router. The virtual router is a logical entity that represents a set of routers that are configured to provide backup to each other. VRRP specifies an election protocol that dynamically assigns responsibility for a virtual router to one of the VRRP routers on a LAN. The election process provides dynamic fail over in the forwarding responsibility should the Router in control become unavailable. This allows any of the virtual router on the LAN to be used as the default first hop router by end-hosts. The advantage gained from using VRRP is a higher availability default path without requiring configuration of dynamic routing or router discovery protocols on every end-host.

7.8 Comparison of IP version 4 and IP version 6

This section provides a generic comparison from a RAN transport point of view. No specific recommendations are implied. In general, IPv6 has been designed to overcome the address space limitations of IPv4.

A summary of the major changes is as follows:

1. Ethertype value is 86DD_{hex}
2. IPv6 address is 128 bits vs IPv4 32 bit address resulting in a very much larger addressing space
3. The following fields are eliminated: header length, ToS, identification, flags, fragment offset and header checksum.
4. The following fields are renamed: length, protocol type and time to live.
5. Options fields redone completely
6. Priority and flow label fields have been added.
7. Header is a fixed format
8. Hop to hop fragmentation is not permitted
9. Optional extension headers (e.g. destination options header)

The RAN transport is required to provide end point addresses to transport packets between different radio network elements in the RAN and core. If an assumption is made that an operator's network is a managed private network, the address space provided by IPv4 may be sufficient for the required number of end point addresses in the RAN (and possibly the core). If the core network as well as mobiles are IPv6 and the RAN transport is IPv4, appropriate inter working is required for routing protocols to work.

8 IP BASED RAN TRANSPORT NETWORK

This section describes how practical implementations of IP networks might be used to support the Transport Network Layer over various interfaces within the RAN.

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8.1 Hosts and Routers

Basically, the IP Transport Network is a set of nodes and links connecting Network Elements implementing UTRAN functions (Node B, RNC, and Management Platform). That network is responsible for transporting user, control plane, data and O&M data between the Network Elements implementing UTRAN functions with some requirements (addressing, security, QoS, etc.). Since standardisation of IP transport option is intended to be layer 2 independent, IP Transport architecture is limited to nodes implementing an IP layer.

In the IP Transport Network, one can distinguish between end nodes (hosts) and intermediate nodes responsible for forwarding IP packets [RFC1812]. Nodes implementing an IP layer are either hosts, or routers, or both. The forwarding capability is the only feature distinguishing routers from hosts.

IP Hosting is a necessary function for a network element supporting UTRAN functions (Node B, RNC) but these network elements may also be IP forwarding nodes. Like AAL2 switching for ATM transport, IP forwarding and routing are not part of UTRAN functions. Routers connect networks of IP hosts to build internets. Hosts are not allowed to route packets they did not originate.

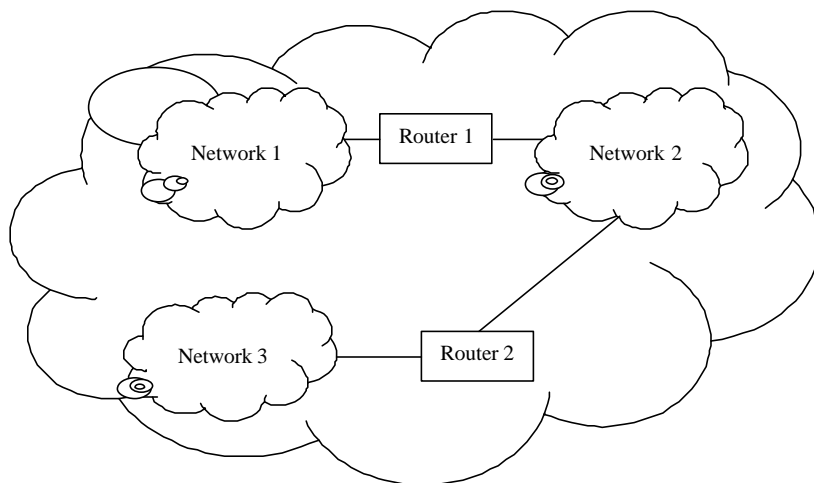


Figure 8.1: Routers interconnecting IP networks.

Routers forwarding IP packets in the transport network have the following characteristics:

- They can process user plane and control plane data at any layer lower or equal to IP.
- They may process higher layer information for Transport Network O&M or configuration purpose.

Other IP features encompass tunnelling mechanisms (e.g. GRE, MPLS, L2TP, IPSec) or mechanisms requiring storage of state information for every flow (e.g. RSVP). Such features can introduce excessive complexity into the Transport Network and should be limited wherever possible..

In IP architecture, a host sees only routers directly accessible without an intermediate router. If there are no multi-homed hosts, there is only one such router, named First Router in the Architecture. A node acting as a router may be a First Router for other Node Bs. If the First Router is part of the IP network of routers, it is typically named Edge Router.

In the special case when two UTRAN network elements are directly connected with a point-to-point link, taking no benefit of IP infrastructure, no intermediate router exists between the elements. However there are still benefits for IP (e.g. no QAAL2). This case constitutes one very specific topology solution.

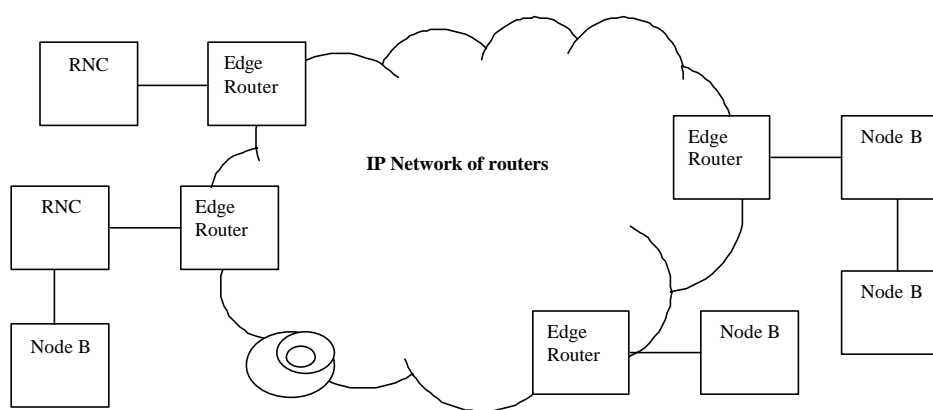


Figure 8.2 : Example Architecture for IP Transport Network

The physical medium between one Node B and the first router is expected to be often bandwidth limited. Even if a point-to-point link is the most likely alternative, the “Last Mile” connection can be any kind of network.

The same bandwidth issue is not expected between RNC and the first router.

The architecture in Figure 8.2 refers to transport and not to UTRAN RNL. Addressing is required for transport nodes level that is different from UTRAN bearer addressing. Any layer 2 protocol or any higher layer transporting IP through tunnelling can be used.

8.2 User Plane Transport Protocol Stack Descriptions

There are various IP based protocol stacks which may be considered for carrying User Plane traffic over the Iub or Iur interfaces. These are described below.

8.2.1 PPP Multiplexed Frame Option Over HDLC

PPP Multiplexing (PPPMux) [PPPMUX] provides a method to reduce the PPP framing [8.3.8.4] overhead used to transport small packets, e.g. voice frames, over slow links. PPPmux sends multiple PPP encapsulated packets in a single PPP frame. As a result, the PPP overhead per packet is reduced. When combined with a link layer protocol, such as HDLC, this offers an efficient transport for point-to-point links.

At a minimum, PPP encapsulating a packet adds several bytes of overhead, including an HDLC flag character (at least one to separate adjacent packets), the Address (0xFF) and Control (0x03) field bytes, a two byte PPP Protocol ID, and the two byte CRC field. Even if the Address and Control Fields are negotiated off and the PPP Protocol ID is compressed, each PPP encapsulated frame will include four bytes of overhead. This overhead can be reduced to one or two bytes.

The key idea is to concatenate multiple PPP encapsulated frames into a single PPP multiplexed frame by inserting a length field before the beginning of each frame. Each PPP encapsulated frame is called a PPP subframe (127 bytes maximum, including the PPP Protocol ID). Removing the PPP framing characters can save several bytes per packet, reducing overhead. The PPP Protocol ID field can also be removed for those subframes which have the same PPP Protocol ID as the preceding subframe.

During the LCP negotiation phase of PPP, a receiver can offer to receive multiplexed frames using an LCP Option. Once LCP has been negotiated, the transmitter may choose which PPP frames to multiplex. Frames should not be re-ordered by either the transmitter or receiver regardless of whether they arrive as part of the PPP multiplexed frame or by themselves.

The PPP Protocol ID field of a subframe can be removed if the PPP Protocol ID of that subframe is the same as that for the preceding subframe. A Protocol Field Flag (PFF) bit is a defined part of the length field (thus reducing the length field from an 8-bit to a 7-bit field). The PFF bit is set if the PPP Protocol ID is included in the subframe. The PFF bit is cleared if the PPP Protocol ID has been removed from the subframe. The PFF bit **MUST** be set for the first subframe in a PPP multiplexed Frame. The transmitter is not obligated to remove the PPP Protocol ID for any subframe.

The format of the complete PPP frame along with multiple subframes is shown in Figure 8.3. Note that regardless of the order in which individual bits are transmitted, i.e. least significant bit first or most significant bit first, the PFF bit is seen to be the most significant bit of a byte that contains both the PFF and the subframe length field.

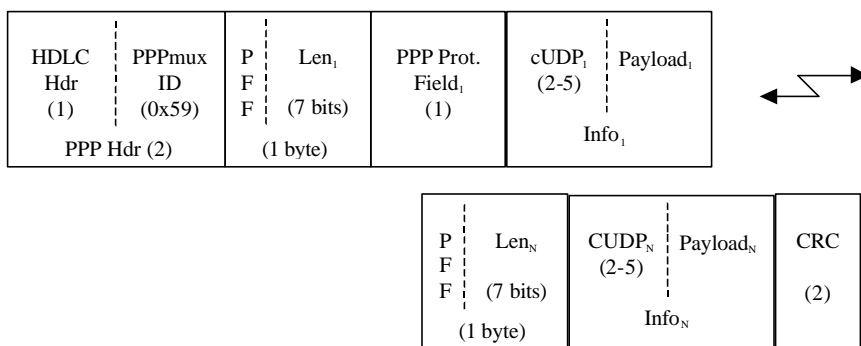


Figure 8.3 : PPPmux frame with multiple subframes

PPP Header The PPP header contains the HDLC header and the PPP Protocol Field for a PPP Multiplexed Frame (0x59). The PPP header compression options (ACFC and PFC) may be negotiated during LCP and could thus affect the format of this header.

Protocol Field Flag (PFF): This one bit field indicates whether the PPP Protocol ID of the subframe follows the subframe length field. PFF = 1 indicates that the protocol field is present for this subframe. PFF = 0 indicates that the protocol field is absent for this subframe. The first subframe of each PPP multiplexed frame **MUST** have PFF = 1. If PFF = 0 then the PPP Protocol ID is the same as that of the preceding subframe with PFF = 1.

Length Field: Each subframe has a seven bit subframe length field. This length does not include the byte containing the PFF and length field but does include the PPP Protocol ID if present (i.e. if PFF = 1). The maximum length of a subframe is 127 bytes. PPP packets larger than 127 bytes will need to be sent in their own PPP frame.

Protocol Field: This field contains the Protocol Field value for the subframe. This field is optional. If PFF = 1 for a subframe, the protocol field is present in the subframe, otherwise it is inferred at the receiver.

The receiver MUST support Protocol-Field-Compression (PFC) for PPP Protocol IDs in this field. Thus the field may be one or two bytes long. The transmitter SHOULD compress PPP Protocol IDs in this field that have an upper byte of zero (i.e. Protocol IDs from 0x21 thru 0xFD). This Protocol Field Compression is not related to the negotiation of PFC during LCP negotiation.

Information Field: This field contains the actual packet being encapsulated. The maximum length of this field is 127 bytes, if the Protocol Field is eliminated from the subframe. Any frame may be included here with the exception of LCP Configure Request, ACK, NAK and Reject frames and PPP multiplexed frames. If LCP is renegotiated then PPP Multiplexing MUST be disabled.

In the proposed protocol stack the Information Field is comprised of a compressed IP/UDP (cUDP) [8.4,8.5] header (with a minimum length of 2 bytes) and the payload of the packet. The cUDP compresses the IP/UDP headers (28 bytes long) to 2-5 bytes (2 bytes when using CIDs and no UDP checksum). For details about the cUDP header compression see RFC 2508 [VJCP]. For purposes of simulation, all simulations reported in Chapter 10 used 3 bytes for the outer cUDP header. The Alcatel simulation in Section 10.1 used 4 bytes for the inner cUDP header.

8.2.2 PPP Multiplexed Frame Option Over ATM/AAL5

This protocol stack uses the same PPPmux option as described above, but carries PPP over an ATM/AAL5 link layer [PPPC], Figure 8.4. Here the HDLC header and CRC trailer is replaced with an ATM header and AAL5 trailer.

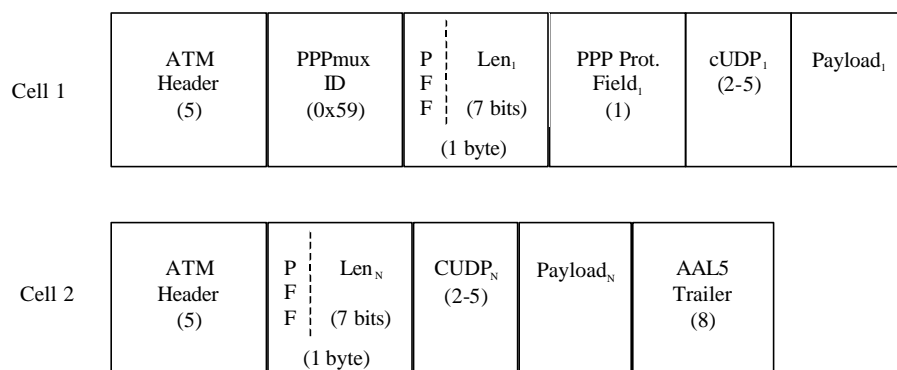


Figure 8.4 : PPPMux over an ATM/AAL5

8.2.3 PPP Multiplexed Frame Option Over L2TP Tunnel (TCRTP)

In cases where a routed WAN interface is required, one may still use PPPmux, but tunnel it via L2TP [RFC2661]. This protocol is called Tunnelled Compressed RTP (TCRTP) [TCRTP].

L2TP tunnels should be used to tunnel the cUDP payloads end to end. This is a natural choice since cUDP payloads are PPP payloads, and L2TP allows tunneled transport of PPP payloads. L2TP includes methods for tunneling messages used in PPP session establishment such as NCP. This allows the procedures of RFC 2509 to be used for negotiating the use of cUDP within a tunnel and to negotiate compression/decompression parameters to be used for the cUDP flow.

A companion draft [L2TPHC] describes a method of compressing L2TP tunnel headers from 36 bytes (including the IP/UDP/L2TP headers) to 20 bytes. L2TPHC packets include an IP header, using the L2TPHC IP protocol id. The UDP and L2TP headers are omitted. The added overhead is now the 20 bytes of the IP header.

Enhancements to CRTP [EHC] are not needed for cUDP header compression.

Figure 8.5 shows an IP packet containing an L2TP-encapsulated PPPMux packet, including L2TP header compression. The two colored boxes on either end of the packet indicate that the L2 data frame will additionally contain an L2 header and possibly a trailer. For example, on HDLC, the overhead will consist of a 1 byte HDLC header and a 2 byte CRC trailer. The exact number of header and trailer bytes will differ depending on the L2 through which the IP packet is traversing, which is why the figure does not contain a specific header and trailer count as is the case for the other figures in this section.

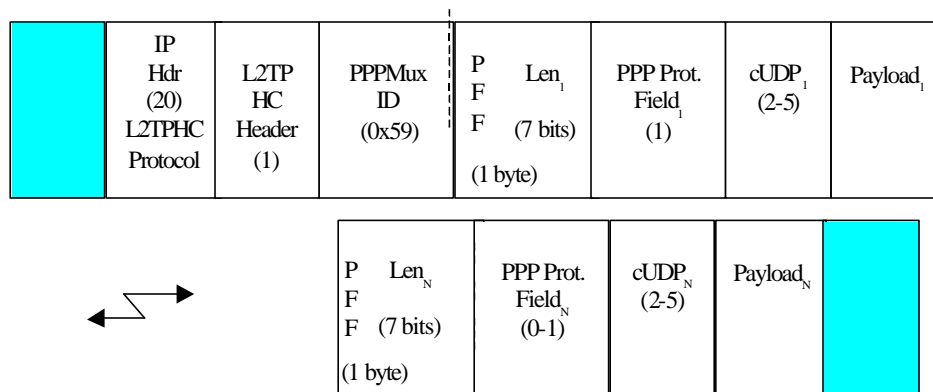


Figure 8.5 : PPPMux Tunnelled over Routed Network using L2TPHC

Compressed TCRTTP (cTCRTTP) is an optimised version of TCRTTP for usage on the last mile.

With TCRTTP, L2TP packets travel through a routed network as IP payloads. Passing the last mile point-to-point links between an Edge Router and a Node B, the IP headers of these IP packets can additionally be compressed in order to save bandwidth on the low bandwidth links. Within this context, these compressed packets are called cTCRTTP packets.

The table below shows the per container/per stream overhead for a TCRTTP packet with HDLC/PPP as a link layer:

HDLC	1 byte
PPP	1 byte
IP	4 byte, compressed IP

L2TP HC	1 byte
PPPMux ID	1 byte
PPP ID	1 byte
PFF, length	1 byte (per stream)
cUDP	3 byte, compressed UDP/IP (per stream)
Payload	
CRC	2 byte

Table 8.1 : Per Container/Per Stream Overhead for TCRTP/HDLC/PPP

8.2.4 Description of the Composite IP (CIP) Approach

8.2.4.1 Introduction to CIP

For an IP-based UTRAN, a user plane protocol stack named CIP (composite IP) is proposed which takes the following general requirements into account:

- **Bandwidth efficiency**
Efficient bandwidth usage in the RAN transport network, especially on the last mile links towards the Node Bs, is directly linked to transport costs. Means shall be provided in the protocol stack to reduce packet overheads.
- **Timing constraints**
In order to fulfil the timing requirements in the UTRAN, low transport delay and possibly a distinction between different service classes is required.
- **Channel addressing**
In order to distinguish between different user channels, a means to identify a particular channel is needed.
- **Independence of layer 1 & 2**
In order not to put any constraints on the underlying transmission technologies, the protocol stack shall be independent of physical and data link layer.

8.2.4.2 CIP Protocol Stack

The proposed user plane protocol stack is depicted in Figure 8.6.

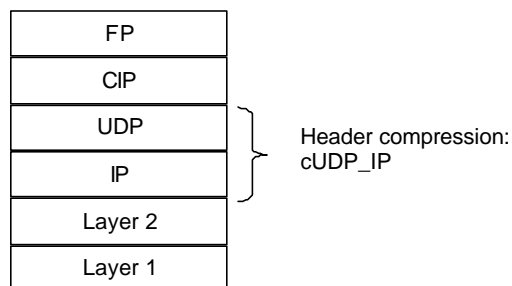


Figure 8.6 : CIP User Plane Protocol Stack

FP PDUs to be transported over the Iub or Iur interface may vary in size to a large extent depending on the data rate of a flow and the associated TTI. FP PDUs carrying voice are small by nature and have a low TTI (AMR codec). FP PDUs carrying data packets may be large and may have a high TTI.

As a consequence of the variable FP PDU sizes, a need to support the two following mechanisms has been identified:

- Aggregation of small FP PDUs into one IP packet in order to amortise for IP/UDP header overhead
- Segmentation of large FP PDUs into smaller chunks in order to keep transmission delays low and avoid blocking of small time-critical packets by large PDUs

8.2.4.3 CIP Container

The aggregation functionality allows the multiplexing of CIP packets having variable sizes into one CIP container, also of variable size. This supports efficient usage of the bandwidth of the links. It is achieved by amortising the IP/UDP overhead over several CIP packets. The resulting packet structure is depicted in Figure 8.7. The leading and trailing colored boxes indicate L2 frame overhead and the exact number of bytes in them will depend on the particular L2 being traversed by the IP packet.

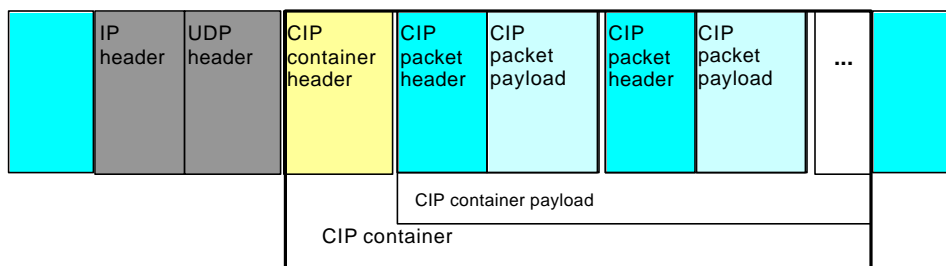


Figure 8.7 : Generic CIP Container format

In the current proposal, the CIP container header is omitted. The CIP container only consists of pairs of packet headers and packet payloads. Their format is described in the next section.

8.2.4.4 CIP Packet Segmentation and Re-assembly

A segmentation/re-assembly mechanism allows the splitting of large FP PDUs into smaller segments. There is a trade-off between efficiency (IP header / payload ratio) and transmission delay. Large data packets must be segmented in order to avoid unwanted IP fragmentation and to keep transmission delays low.

Figure 8.8 shows the segmentation process from a FP PDU to several CIP packet payloads.

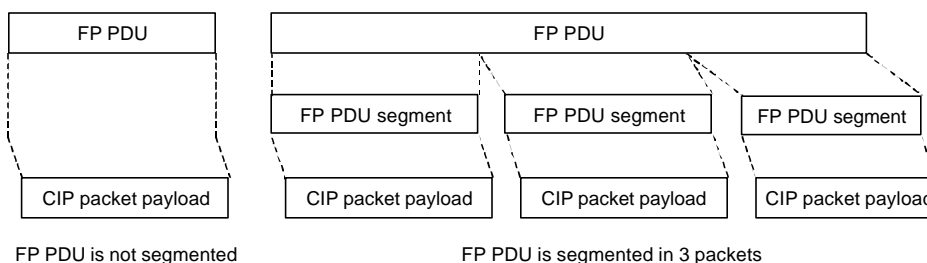


Figure 8.8 : CIP segmentation

8.2.4.5 CIP Packet Header Format

The proposed CIP packet header format is shown in Figure 8.9.

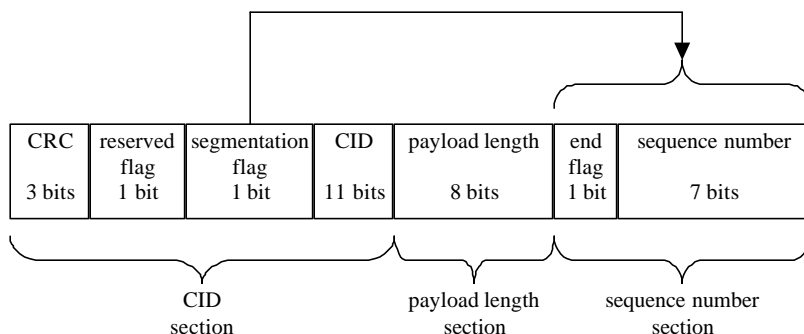


Figure 8.9 : CIP packet header format

8.2.4.6 CIP Packet Header Fields in Detail

The CIP packet header is composed of three sections:

1. **CID section**, also containing CRC and flags is used for multiplexing. This section is mandatory.
 - The **CRC** protects the reserved flag, the segmentation flag and the CID.
 - The **reserved flag** is for further extensions.
 - The **segmentation flag** indicates that the sequence number field and the end flag are present. These fields are only needed for segmented FP PDUs. In the case of aggregation of non-segmented PDUs, e.g. PDUs carrying voice, these fields are suppressed by means of the segmentation flag to save bandwidth.
 - The **CID** is the Channel ID. This is the identifier of the multiplex functionality, e.g. to distinguish the flows of different calls or users by the higher layers.
2. The **payload length section** is used for aggregation. This section is mandatory.
 - The **payload length** is the length of the CIP packet payload. So, CIP packets, containing e.g. FP-PDUs with voice or FP-PDU segments with data, can be between 1 and 256 octets in size.

3. The **sequence number section**, also containing the end-flag, is used for segmentation and reassembly. In case of lost and/or reordered packets, the sequence number assures the recognition of lost segments and provides a means to re-establish the original order of the reordered packets. The section is optional, it exists if the segmentation flag is set.
- The **end-flag** marks the last segment of a packet in a sequence of segments. This field is only present if the segmentation flag is set.
 - The **sequence number** is used to reassemble segmented packets. This field is only present if the segmentation flag is set. It is incremented for each segment (modulo) and is not reset if the segments of a new packet start. The sequence numbers are maintained for each CID individually.

8.2.4.7 CIP Overheads

This section presents the overall overheads associated with the CIP scheme for two different link layers. Our simulations are based on these overheads. The following table shows the per container and the per stream overhead for PPP/HDLC as layer 2 protocol:

CIP/cUDP/PPP/HDLC

HDLC	1 byte
PPP	1 byte
cUDP	4 byte, compressed UDP/IP
CIP	3 byte (per stream)
Payload	
CRC	2 byte

The following table shows the per container and the per stream overhead for AAL5/ATM as layer 2 protocol. The assumed format is 'VC-multiplexed PPP' (RFC 2364, "PPP over AAL5", Chapter 5).

CIP/cUDP/PPP/AAL5/ATM

PPP	1 byte
cUDP IP	4 byte, compressed UDP/IP
CIP	3 byte (per stream)
Payload	
AAL5 Trailer	8 byte

Additional overhead per cell: 5 Byte header, padding to 48 Byte payload in the last used cell. The following table gives a summary of the total overheads:

Layer 2	Overhead/stream	Overhead/container	ATM overhead
PPP/HDLC	3 byte	8 byte	N.A.
AAL5/ATM	3 byte	13 byte	5 byte + padding

8.2.5 Lightweight IP Encapsulation Scheme for UTRAN User Plane

8.2.5.1 Introduction to LIPE

RTP [TCRTP] is a protocol designed to provide various real time services to the application layer with no assumption on the underlying network providing timely delivery or quality-of-service commitments. To improve the transport efficiency, some multiplexing schemes have been proposed within the framework of RTP [8.11,8.14].

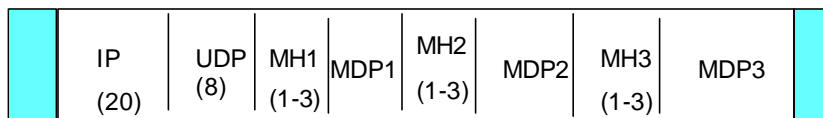
Many of the features of RTP are designed to provide media control information to cope with the unavailability of QoS guarantees from the underlying network at the application layer. As such guarantees become available in modern/future IP networks, some of these features become unnecessary. These features are also of limited value to non-RTP applications (e.g. most commercial wireless voice traffic) e.g. the FP PDUs carrying voice that need to be transported over the Iub or Iur interface.

This section describes a new LIPE protocol for multiplexing raw voice/video frames into a single IP packet to be transported across the Iub/Iur interface. LIPE is designed to carry multimedia traffic including both voice and data. The LIPE [LIPE] proposal was presented at August IETF meeting in Pittsburgh.

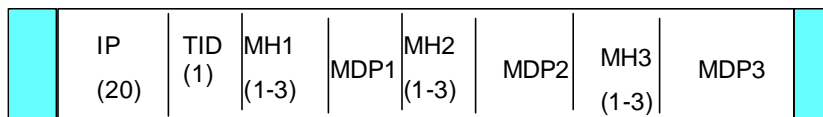
8.2.5.2 LIPE

The LIPE scheme uses either UDP/IP or IP as the transport layer. Each LIPE encapsulated payload consists of a variable number of multimedia data packet (MDP). For each MDP, there is a multiplexing header (MH) that conveys protocol and media specific information.

The format of an IP packet conveying multiple MDPs over UDP using a minimum size MH is shown in Figure 8.10. For the simple LIPE packet depicted in the top figure, the Multiplexed Header (MH) and Multiplexed Data payload (MD) are shown. In the bottom diagram, the packet is tunnelled and a Tunnel Identifier (TID) indicates the tunnel. The colored boxes on either end of the packet indicate L2 frame header and trailer overhead, if any, and their actual size will vary depending on the L2 being traversed by the packet. Details of the multiplexed header is described in the next section.



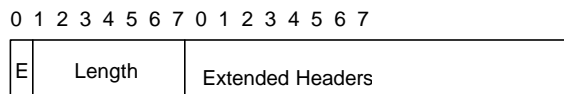
(a) Simple LIPE Packet



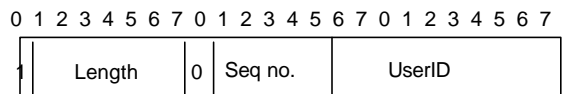
(b) Tunnelled LIPE Packet

Figure 8.10 : LIPE UDP/IP or IP Encapsulation Format

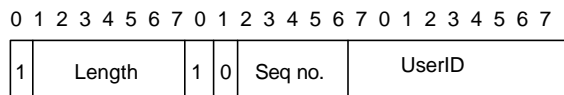
8.2.5.3 Details of Multiplexed Header



(a) Basic Multiplexed Header



(b) Basic Multiplexed Header with 3 bit header CRC and 12 bit UserID



(c) Extended Multiplexed Header with Sequence number and UserID

Figure 8.11 : Formats of Multiplexed Header

Basic Header

The Multiplexing Header (MH) consists of two components: The extension bit (the E bit) and the MDP length field. Optional Extension Headers can be supported via the E bit. The MH format is shown in Figure 8.11 (a). The E bit is the least significant bit of the first byte of the MH header. It is set to one/zero to indicate the presence/absence of an extension header. If the E-bit is set to one, the first header extension MUST be a Extended Header Identifier field. The Length field is 7 bit. This field indicates the size of the entire MDP packet in bytes, including the E bit, the length field and optional extension headers (if they exist).

Extension

Extension headers are used to convey user specific information. It also facilitates the customization of LIPE to provide additional control information, e.g. sequence number, voice/video quality estimator.

The 16-bit EHI is the first field in any Extension Header. It is used to identify MDPs belonging to specific user flows. The format of a LIPE encapsulated payload with a UserID extension header is shown in Figure 8.11 (b). The least significant bit of the 1st byte of EHI is the X-bit. When the X-bit is clear, it means there is a 3 bit header Sequence Number and a 12 bit UserID. When the X bit is set to one, it indicates that the EOF bit and the 3 bit Seq Number fields exist and that the UserID field is 11 bit. The second least significant bit is the end of fragment (EOF) indicator. When EOF is set to 0, it means this is the last fragment (for packets that are not fragmented, this bit is always 0). When EOF is set to 1, it means there are more fragments coming.

8.3 MPLS for IP Transport in the RAN

8.3.1 Introduction to MPLS

The Multi-Protocol Label Switching (MPLS) [8.16.8.17,8.18] protocol is an interstitial, layer 2.5 protocol which complements and enhances the IP protocol, in that it offers a complementary method of forwarding IP packets, while reusing the existing IP routing protocols (e.g., OSPF, BGP). MPLS forwards IP packets based on a 20-bit label, and offers a number of advantages, among them are improved traffic engineering and IP VPN support.

An ingress router at the edge of an MPLS domain, called a Label Edge Router (LER), decides which subset of incoming packets is to be mapped to which Label-Switched Path (LSP), and then adds the corresponding label to each packet as it arrives. This subset of packets that is forwarded in the same manner over the same LSP is called a Forwarding Equivalence Class (FEC). Packets are then forwarded through the MPLS domain by the Label Switched Routers (LSRs) based on the label. At the egress edge of the MPLS domain, the egress Label Edge Router (LER) removes the MPLS label from each IP packet, and subsequently the IP packets are forwarded by conventional IP forwarding.

Each pair of LSRs on the Label Switched Path (LSP) must agree on which label to use on that segment of the LSP. This agreement is achieved by using a set of procedures, called a Label Distribution Protocol, by which the upstream LSR (optionally) requests a label for a given FEC from the downstream LSR, and by which the downstream LSR informs the upstream LSR of the label binding it has made. The reference of upstream/downstream is with regards to the direction of the LSP – all LSPs are uni-directional.

In summary, a label distribution protocol associates a FEC with each LSP it creates. The FEC associated with an LSP specifies which packets are "mapped" to that LSP.

MPLS, as a complementary forwarding technique to IP forwarding, offers the following advantages:

- **Bandwidth-efficient tunnelling.** The MPLS header is only 4 bytes.
- **Coexistence with IP Hop-By-Hop Routing.** An LSR is capable of both forwarding IP packets and MPLS packets.
- **Flexibility due to label semantics.** The meaning of the labels can be tailored to what needs to be achieved in the network. For example, labels can be used to specify QoS, multiplexing, multicasting, micro-mobility, etc.
- **Flexibility due to label stacking.** MPLS supports the ability to stack more than one label in an IP packet. LSRs are capable of pushing, popping and swapping labels. This allows for:
 - Different addressing in different subnets.
 - Efficient inherent support for tunnels-in-tunnels. This can be used, for example, for IP VPN and mobility support.
 - Hierarchical routing domains.
- **Fast Rerouting.** MPLS protection switching mechanisms can be applied to achieve fast restoration from a node failure. Both local and end-end protection could be used to achieve fast tunnel restoration which is an essential requirement for a carrier grade network [HCMPLS]. Backup tunnels may also be combined with load sharing to allow a more even traffic distribution.

In addition, as discussed in more detail in Section 7.1.4, MPLS provides specific mechanisms that support QoS.

8.3.2 MPLS for IP-based transport in the UTRAN

8.3.2.1 Header Compression

An LSP can be created for any combination of IP address plus UDP port number. Once the path has been created, the IP address is no longer required to route the packet through the MPLS network – the MPLS label performs that function. In addition, since the information contained in the UDP header is static¹, it can be stored as part of the MPLS label context. Thus, both the IP and UDP headers can be stripped off and replaced by an MPLS label at the transmitter. At the receiver, the MPLS label context can be used to restore the static pieces of the IP and UDP headers while the dynamic pieces of the header can either be recalculated by the receiver (e.g. header checksum) or derived from information provided by MPLS label (e.g. time-to-live) or derived from information provided by layer 2 (e.g. packet length).

Note also that IP/UDP header suppression occurs at the end points of the MPLS path and the compressed packet passes transparently through the intermediate Label Switching Routers. This is in contrast to schemes based, for example, on PPP where either header (de-)compression must occur on a hop-by-hop basis or the compressed packets must be carried inside a second, uncompressed IP tunnel packet.

8.3.2.2 Elimination of Multiplexing

Multiplexing is usually introduced as a way to amortise (packet) overhead over a larger amount of user traffic. When IP header compression is used in a IP routed network, multiplexing becomes necessary to amortise the overhead (20 or more bytes) of the IP tunnel. Note, however, that the multiplexing mechanism itself often introduces protocols and associated overheads that must be balanced against the savings achieved by the original header compression.

MPLS provides a means to suppress IP and UDP headers, replacing them by an MPLS label that is bandwidth efficient (4 bytes). And, since a packet can be routed based on this label, there is no need to introduce additional tunnels and multiplexing schemes. The use of MPLS not only makes the transport layer more bandwidth efficient, it makes the network and the network nodes much simpler.

8.3.2.3 MPLS Header Compression “Session” negotiation

As with the other header compression techniques, a header compression session negotiation is required. This section gives two examples of how this can be done:

1. Use RSVP messages to negotiate the header compression [RMPLS]
2. Use the LDP to negotiate the header compression.

Using RSVP signalling for MPLS Header Compression session negotiation

The internet draft “MPLS Simple Header Compression” [RMPLS] describes a way of negotiating a MPLS Header Compression session using RSVP signalling. The compressor endpoint sends an RSVP PATH message to request an MPLS header compression session. The decompressor replies with an RSVP RESV message confirming that it will perform the decompression.

The compressor includes a SIMPLE_HEADER_COMPRESSION (SHC) RSVP object in the PATH message to communicate the header template and the set of operands. To allow multiplexing across an LSP the SHC objects also carry a one byte sub-context ID (SCID)

¹ The packet checksum is the only potentially dynamic information contained in the UDP header. However, the UDP checksum is redundant since error detection can be provided by the Layer 2 protocol. Hence the calculation of the UDP checksum, which is an optional function, can be turned off.

The decompressor includes a SIMPLE_HEADER_COMPRESSION_REPLY RSVP object in the RESV message to indicate which SCIDs it is agreeing to decompress.

The template in the SHC object consists of the first n bytes of a packet. All of the fixed fields are set to their appropriate values. The variable fields are set to zero. Fields are always delimited on byte boundaries. Each operand is simply an offset and a length. They serve to delimit the variable fields within the template.

Instructions on what to do with the variable fields (e.g., IP TTL, IP checksum, and IP length) is also signalled in the SHC object, using the T, C, and L flags, respectively.

The compressor removes the header from the packet. The term header is used loosely here. It refers to the first n bytes of the packet where n is the length of the header template. The compressor uses the operands to extract the variable fields from the header. These are concatenated together as a compressed header. The SCID is then prepended to the compressed header and the packet is sent.

The decompressor uses the incoming MPLS label and the SCID to locate the proper decompression context. The decompressor then uses the header template to reconstruct the original header. It uses the operands to populate the variable fields of the header with the contents of the compressed header.

Over the life of an RSVP session SCIDs may be added and deleted simply by refreshing the Path state with the updated set of SHC objects. The SHCR object provides synchronization between the sender and receiver as to which SCIDs may be used.

Using MPLS signalling for MPLS Header Compression session negotiation

A fundamental concept in MPLS is that two LSRs must agree on the meaning of the labels used to forward traffic between and through them. This common understanding is achieved by using a set of procedures, called a label distribution protocol, by which one LSR informs another of label bindings it has made.

The Label Distribution Protocol, LDP [LDP] describes one of the label distribution protocols, by which LSRs distribute labels to support MPLS forwarding along normally routed paths.

MPLS Header Compression session negotiation can be accomplished with the LDP protocol, by adding a new FEC TLV (Type-Length-Value) that includes a source IP address, source UDP port, destination host address and a destination UDP.

The compressor requests a label for a new 4-tuple combination {source IP address, source port, destination IP address, destination port} via the downstream on-demand method from the decompressor, which is its LDP peer in this case. The decompressor provides the MPLS label it wants to use for the IP address/UDP port back to the compressor. The decompressor also stores the mapping of MPLS label to FEC in a local table. The compressor also specifies how the IP TTL, IP checksum, and IP length fields are to be regenerated on the other end in the FEC TLV.

The compressor LSR can then suppress the UDP/IP header, and replace it with the appropriate MPLS label. When the decompressor LSR receives the MPLS frame, it looks up the MPLS label in the mapping table, and uses this information to restore the UDP/IP header.

9 TRAFFIC, NETWORK AND SYSTEM MODELS

This chapter provides the traffic, network, protocol and systems models to support the simulation of IP in the RAN as a Transport Option. These models could become complex, so it will be necessary to identify those aspects that are critical in terms of whether IP is acceptable as a transport option.

The initial simulation models were constructed for the 3GPP R99 UTRAN. This system was selected partially because the protocols on all RAN interfaces have been standardised and published. Therefore, the basis existed for constructing a simplified, generic model of the "last mile" in the UTRAN (RNC to Node B), which preliminary simulations indicated was the most crucial in controlling RAN performance. Another and perhaps more important reason was to limit the amount of work involved in doing the simulations, since simulation work is relatively time consuming. Provided resources are available to construct the models and perform the simulation, it is expected that future versions of this report may include results for the 3GPP2 IOS 4.0 RAN or a future version of the 3GPP2 RAN as well.

It should be noted that the simulation model was designed primarily to validate that IP was comparable to AAL2/ATM with regard to bandwidth, delay, and jitter characteristics for implementing the TNL within the UTRAN. Results were generated for a variety of IP protocol options, but the simulations were not designed to stress test the particular aspects of each protocol option.

It is assumed that all user services are carried within dedicated physical channels (DPDCH) over the radio interface. This is to provide an initial set of models for the simulation work. In the longer term it is expected that non-real time bursty data could be carried over shared physical channels over radio.

9.1 Transport over UTRAN Interfaces

This section outlines the main parameters which influence the models for the traffic flow and transport over the UTRAN interfaces (Iu, Iur and Iub). Only the Radio Network is considered in relation to transport over these interfaces, and models of the transport network are described elsewhere. The discussion in this section is quite general and is primarily concerned with a detailed examination of traffic, network, and system models. Of necessity, the actual simulation model constructed, which is described in the rest of the chapter, is simplified to capture the important aspects of the UTRAN for purposes of determining the bandwidth, delay and jitter while reducing the amount of time and effort put into the simulations themselves.

9.1.1 Overview of UTRAN Traffic Model

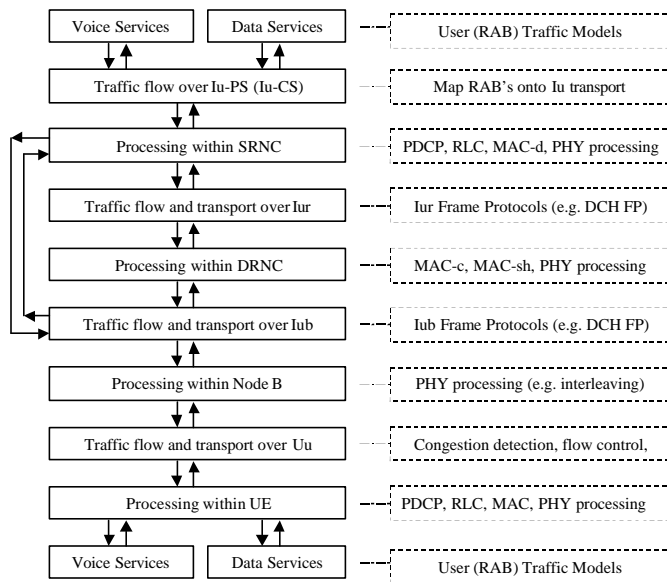


Figure 9.1 – Mapping of User Traffic Flow onto Transport over UTRAN Interfaces

Figure 9.1 shows the traffic model of the UTRAN for both uplink and downlink. Only transport over the UTRAN, between Iu and UE, is considered within this report. The DRNC and Iur are included but often a DRNC may not be involved for communications to many UEs.

9.1.2 UTRAN Configurations

The task is to transport service related user information between the Iu interface and the UE. The transport over Iu corresponds to the user data transported within Radio Access Bearers set-up by requests from the core network to support various user services (e.g. voice, data, etc).

9.1.3 User Service Traffic Models

Traffic flow over the UTRAN interfaces depends on the services demanded by the UEs and the roaming aspects of UEs. Service may include a mixture of many types of voice and data services. Roaming leads to the demands of handover and SRNC relocation which may not affect traffic flow but could place other demands on the transport network.

The description of the user traffic models is related to the user service as seen within a Radio Access Bearer including the impact of codec (e.g. AMR codec) on the traffic flow for that service.

To derive the traffic flow over the Iu interface it will be necessary to consider the protocol overheads and timing of information transfers over Iu.

To consider traffic flow over Iur and Iub, it is necessary to consider processing of user services in the RNC plus overheads and timing of information transfers over Iur and Iub.

9.1.4 Traffic Flow over Iu-PS and Iu-CS

Transport over Iu is based on transparent and support mode as described in TS 25.415 for Iu-CS. The impact of these modes on traffic flow of the Iu interface needs to be considered. Also there is a need to confirm whether such modes apply to Iu-PS. There is an interest in support mode for Iu-PS so RAB sub-flows and unequal error protection can be provided for real time services (e.g. voice over IP).

SRNC relocation must be supported but this relates more to the capabilities required of the transport network than to the traffic flow profiles over Iu. RANAP signalling must be supported over Iu but it is assumed that this is only a small traffic flow.

9.1.5 Processing within SRNC, DRNC and CRNC

The processing functions within the SRNC will change the nature of the traffic flow between the Iu and the Iur (and Iub). Traffic flow over Iu will be closely related to the support of the specific user service while traffic flow over Iur and Iub will be closely related to blocks of data prepared by the SRNC for transmission within a predefined sequence of radio frames.

The way in which that traffic flow changes is dependent on the user service being supported and the processing functions within the SRNC (i.e. PDCP, RLC and MAC layers). Traffic sent from the SRNC to the Node B (over Iur and Iub) will carry a reference to the specific radio frame(s) in which that data must be transmitted. Radio bearer traffic over Iur and Iub will be designed to arrive “just in time” at the Node B so that data can be transmitted within the predefined sequence of radio frames.

In the case of real time services, e.g. voice, it is not possible to delay the user service. There might be a small delay in the SRNC but only to wait for the next available radio frame or TTI interval (of 20 ms). Hence the traffic flow over Iur and Iub for a real time service should be similar to that over Iu plus overheads due to the frame protocols for that transport channel over Iur and Iub (i.e. DCH FP).

In the case of non-real time data, the RLC layer buffers arriving data. The MAC-d layer then only allows data to be sent from the SRNC to Node B which complies with one of the agreed transport formats for the transmission of that data over radio. For example data arriving over Iu might be bursty packet data at a wide range of bit rates (or bytes in each packet), but the formats over Iur and Iub might only allow one of (say) four block sizes to be sent over Iur and Iub.

For some services, PDCP might be used to provide compression of packet data to provide more efficient use of radio resources. Also some RLC modes may involve retransmission requirements.

Common (RACH/FACH) and shared (CPCH/DSCH) transport channels may support information transfers for many UEs. A MAC-C or MAC-sh within the CRNC is used to schedule transmission between UEs for these channels. Within the current work only transport over dedicated physical channels (DPDCH) is considered, i.e. the impact of common or shared channels is ignored.

During soft handover multiple Radio Bearers would be required to support that UE. This implies that the traffic flow for that UE would be duplicated over multiple connections over the Iur and Iub. How and where duplication must be supported depends on where soft handover (i.e. splitting and combining) is supported (e.g. in SRNC, DRNC and/or Node B).

9.1.6 Traffic Flow and Transport over Iur and Iub

All traffic related to Radio Bearers over Iur and Iub will be carried within frame protocols, defined for each of the types physical channel over radio. In this work it is assumed that all Radio Bearers are carried with dedicated physical channels (DPDCH) within a DCH frame protocol over Iur and Iub.

All DCH frame protocols carry the reference to the radio frame(s) in which that information must be transmitted. The SRNC will monitor the delay over Iur and Iub to decide on the radio frame number

in which each DCH frame protocol should be transmitted. If the DCH frame protocol arrives at the Node B too late then that information can not be transmitted. This applies for all user services.

There is also a need to consider whether or not all user services within DCH frame protocols are carried within the same transport connection over Iur and Iub. If separate connections are provided for real time and non-real time services then transport priorities can be provided. This can allow a longer time for the transmission of high bit rate non real time packet data while ensure a shorter delay over Iur and Iub for smaller blocks of real time data.

Transport over Iur and Iub also includes the need to transport RRC signalling messages between the SRNC and UE within RLC connections. Also RNSAP and NBAP signalling must be supported over Iur and Iub, but should only be a small traffic flow. The support of soft handover could require DCH frame protocols to be duplicated over multiple Iur and/or Iub interfaces.

The alignment of radio frames and TTIs needs to be considered in relation to traffic flow over Iur and Iub. It is understood that the alignment of radio frames and TTIs between UEs is distributed in some way so that traffic flow related to different UEs over the same Iur or Iub is distributed in time.

9.1.7 Processing within Node B

The Node B will receive frame protocols for a number of different Radio Bearers over Iub interface for the same UE. In this report only the transport of Radio Bearers within dedicated physical channels (DPDCH) over the radio interface is considered.

The Node B will interleave and multiplex all Radio Bearers for one UE into that one DPDCH. Interleaving for various Radio Bearers may be provided over 10, 20, 40 or 80 ms (1, 2, 4 or 8 radio frames) and the arrival time (TTI) of that data over Iub will be equal to that interleaving period.

The Node B will multiplex Radio Bearers for the same UE. Each of these Radio Bearers might require a different interleaving period (and arrival rate at Node B). Node B can provide different levels of error protection for each of these Radio Bearers before transport of the composite DPDCH over radio.

The combinations of the Radio Bearers that can be transmitted in each sequence of radio frames are limited to a predefined set of transport formats. These transport formats allow dynamic allocation of radio resources by the MAC-d layer (in the SRNC). These transport formats only allow a few pre-set number of byte options to be carried over the radio interface for each of those user services regardless of the traffic arrival rate over Iu interface.

9.1.8 Traffic flow and transport Uu

During congestion the UTRAN may limit the transport formats which can be used by MAC-d in the SRNC (for dedicated channels) and MAC-c/MAC-sh in the CRNC (for common or shared channels).

Congestion detection might be based on high levels of traffic flow detected in the CRNC. Also congestion detection might be based on high levels of radio interference detected within one or more Node Bs. This might be due to high traffic on that radio cell, adjacent radio cells or other reasons.

It is possible that the total traffic flow that can be supported over the radio interface might be more important than the bandwidth over Iub. For example if traffic flow over Iub is too high then congestion is detected and the transport formats which can be used in the MAC layer are restricted.

9.2 Voice Traffic Model

The user service traffic flow refers to the user service data carried within Radio Access Bearers. This is expected to be similar to the traffic flow over Iu but the impact of framing protocols and other

traffic flow over Iu (e.g. RANAP signalling) should be considered to assess traffic flow over the Iu interface.

Over the Iur and Iub interfaces the user service traffic flow for voice traffic will be similar to the traffic flow for the voice model over the Iu interface. User traffic flow might be delayed slightly within the RNC but only to provide alignment with radio frames. The framing protocols over Iur and Iub will add some overhead but available transport formats over Iur and Iub should be defined to align with all possible AMR codec rates for voice services. Signalling within RNSAP over Iur, NBAP over Iub and RRC signalling within RRC connections will also add additional traffic over Iur and Iub.

The voice traffic is modelled as individual AMR generator per call. Each generates 3 RAB sub-flows for the different classes of bits from the AMR codec and 1 RAB sub-flow for signalling information. The AMR generator consists of the two states as “ON” and “OFF” due to DTX. The duration a user remains in one state is given by a distribution function. Signalling is modelled by messages, which are included in the traffic stream according to the given distribution functions. The number of active calls is assumed to be constant. Call duration is modelled because the time offset inside the 20 ms period will change with each call according to a uniform distribution.

The voice traffic model, Table 9.1, was derived from TR 101.102 v3.2.0 while the AMR codec model, Table 9.2, was derived from TS 26.101 v3.1.0. Both of these models were adjusted to suit the requirements for the simulation work described in this report. The number of concurrent users is adjusted during the simulation to achieve the required load on the network.

Class	Parameter	Values	Remark
Call Statistics	Call duration distribution	Exponential	
	Call duration mean	120 sec	
ON/OFF Statistics (during a call)	Distribution	Exponential	
	Mean	3.0 sec	
Voice Frames (over Iu)	Interpacket arrival time	20 msec	
	Inter arrival time distribution	Constant	
	Distribution of time offset between UEs	See RNC model	
Voice Frames (over Iur/Iub)	Transmission Time Interval (TTI)	20 ms	
	Inter arrival time distribution	Constant	
	Distribution of time offset between UEs	See RNC model	
Signalling (over Iur/Iub)	Signalling time distribution	Constant	
	Signalling interarrival time	300 ms	
	Signalling message size distribution	Constant	
	Signalling message mean size	10 bytes	
Frame	ON state (for Without signalling	40 bytes	

Protocol (over Iur/Iub)	12.2 kbit/s AMR Codec)	With signalling	50 bytes	
	OFF state (for comfort noise)	Without signalling	13 bytes	
		With signalling	23 bytes	

Table 9.1 - Voice Traffic Model Assumptions for Simulation Scenarios

The simulation work in this report only considered the OFF state of the AMR codec (comfort noise) and the ON state of the AMR codec (for 12.2 kbit/s). The interarrival time of packets from the AMR codec is 20 ms (as described in Table 9.1).

AMR State	AMR mode (kbit/s)	Total (bits)	Class A (bits)	Class B (bits)	Class C (bits)	Remarks
ON state	4.75	95	42	53	0	
	5.15	103	49	54	0	
	5.90	118	55	63	0	
	6.70	134	58	76	0	
	7.40	148	61	87	0	
	7.95	159	75	84	0	
	10.20	204	65	99	40	
	12.20	244	81	103	60	
OFF state	Comfort Noise	39	39	0	0	

Table 9.2 - AMR Codec Payload Size for each 20 ms time interval

9.3 Data Traffic Model

The user service traffic flow refers to the user service data carried within Radio Access Bearers. The data traffic might be modelled as the arrival of packets of various sizes with various interarrival rate distributions. The traffic flow over Iu is expected to be similar to the user service traffic flow but the impact of framing protocols and other traffic flow (e.g. RANAP signalling) should also be considered.

Over the Iur and Iub interfaces the user service traffic flow for data traffic will be very different to the traffic flow over the Iu interface for non real time packet data. The RNC may include header compression for user packet data (within the PDCP layer) after which that data must wait in the RNC (within the RLC layer) until resources become available over the radio interface. The data, which can be transported over the radio interface for each TTI is constrained by the set of transport formats allocated for that data service. The framing protocols over Iur and Iub adds some overhead. Also padding may be added to align waiting data to available transport formats. Signalling within RNSAP over Iur, NBAP over Iub and RRC signalling within RRC connections will add additional traffic over Iur and Iub.

Data traffic can be categorised in three basic classes. The characteristic of their traffic requires individual modelling.

- **Background traffic** : Background traffic is typically generated by simple messaging services like SMS and email. After arrival of a packet of this class it is transmitted over a channel with

a constant bit rate. For a short term simulation, it is reasonable to assume a constant number of active message transmissions. Individual packets are modelled to consider the packet size dependent overhead.

- **Interactive data traffic** : Interactive data traffic is mainly generated by WWW serving. As for the background traffic, the number of active users is assumed to be constant. The parameters are listed in Table 9.3.
- **Multimedia data traffic** : Multimedia data traffic is used to model traffic which is typically generated by video streaming.

The data traffic model for the simulation work in this report, Table 9.3, was derived from TR 101.102 v3.2.0 for World Wide Web browsing. This model has been adjusted to suit the requirements for the simulation work described in this report. The number of concurrent users is adjusted during the simulation to achieve the required load on the network. In Table 9.3 a packet call refers to the arrival of a number of packets within a session. It is assumed that the Radio Access Bearer is maintained for the duration of the session.

Each data user represents a high-speed best-effort data transfer, such as file downloads during web browsing. There are no explicit delay bounds applied to data traffic, however a finite queue is applied. The required simulation queue size is indicated by delay and packet-loss parameters.

Each data user sends data packets during ON-periods (packet call ON state), and does not send data packets during OFF periods (packet call OFF state). The transmission during the packet call ON state is derived from empirical studies of web traffic indicating a mean transfer size of 12 kbytes. The OFF period mean is derived from empirical studies of web traffic indicating a mean idle/think time of 12 seconds.

Class	Parameter	Values	Remark
Packet call arrivals within a session	Packet call inter arrival distribution	Exponential distribution	
	Packet call mean inter arrival time	$D(pc) = 12 \text{ sec}$	
Number of packets in a packet call	Number of packets distribution	Geometric distribution	
	Mean number of packets	$N(d)=25$	
Packet sizes within a packet call	Packet size distribution	Pareto distribution	See Pareto equations (see Annex D) Value of α derived from these equations
	Mean packet size (unlimited Pareto)	$\mu = 896.5 \text{ bytes (for Pareto)}$	
	Minimum packet size	$k = 81.5 \text{ bytes}$	
	Alpha value for Pareto distribution	$\alpha = 1.1$	
	Maximum packet size (limit when Pareto suggests a larger value)	66666 bytes	Value of μ_m derived from equations following this table
	Mean packet size (after size limit)	$\mu_m = 480 \text{ bytes}$	

Transmission over Iu within a packet call	Packet inter arrival distribution	Exponential distribution	Dependent on packet size and bit rate over Iu
	Packet mean inter arrival time	8.3 ms (for 480 kbit/s over Iu)	Allows an arrival rate greater than peak rate over the radio interface
Transmission over Iur / Iub	Peak bit rate	64, 144 or 384 kbit/s	
	Transmission Time Interval (TTI)	40 ms	
	Transport format set	See RNC model	

Table 9.3 - Data Traffic Model Assumptions for Simulation Scenarios

9.4 Processing in SRNC, DRNC and CRNC

9.4.1 Processing within RNC

Some priority and queuing models need to be provided in relation to the processing in the SRNC, DRNC and CRNC. In the SRNC this would relate to the processing within PDCP, RLC and MAC layers.

PDCP provides header compression, which will impact on the traffic flow over Iur and Iub, but might be ignored when comparing transport network protocol options.

RLC provides formatting of user data into frame protocols for transport over Iur and Iub. Real time (voice services) should be sent over Iur and Iub in the next available TTI (of 20 ms) but non real time (data services) must wait for radio resources to become available (i.e. queuing within RLC).

MAC will ensure that any data sent to Node B complies with one of the allowed transport formats for the frame protocols over Iur and Iub. Also MAC must add, to that data, the radio frame number in which that information (voice or data) must be transmitted over the radio interface.

Within the SRNC, DRNC and CRNC the support of soft handover will impact on traffic flow over Iur and Iub (i.e. duplication of traffic flows), but this might be ignored when comparing transport network protocol options.

The RNC user plane model has the following characteristics:

- A mapping from (source type, bit rate) to {channel type, TTI, transport block (TB)}
- Time offset at the Iub : The arrival of frames at the Iub can be modelled by assuming that frames can arrive at some fixed subdivision interval of the minimum TTI, i.e. a fixed division of 20 ms.

Two independent queues are assumed in the RNC; one for voice traffic and one for data traffic. A queue scheduling algorithm is applied whereby packets in the voice queue have priority over packets in the data queue. Furthermore, voice packets cannot pre-empt data packets. The voice queue will be serviced until empty, at which time the data queue will be serviced until empty or the voice queue has become non-empty.

9.4.2 RNC Model for Voice Traffic

The agreed model for simulation of the RNC for the support of voice traffic is as follows :

- Voice traffic model : as described in Table 9.1
- RNSAP or NBAP signalling : this traffic flow is ignored

- RRC signalling : this signalling traffic flow is included (see signalling messages in Table 9.1)
- Direction of traffic : the main interest is in delay and traffic flow over Iur and Iub. The analysis is for downlink only but should also apply to the uplink
- Buffering in RNC : no buffer of voice data except to align with the next available radio frame
- Transmission time interval (TTI) : 20 ms equal to the arrival rate over Iu from the AMR codec.
- Transport formats over Iub : defined to support exactly the various AMR modes (Table 9.2). Only the ON state at 12.2 kbps and OFF state (comfort noise) is included in the simulations.
- Time offset for frames for each UE : using either discrete or continuous offset groups within a TTI but maintained constant for each UE voice call
- Access to resources : voice traffic is independent of data traffic (e.g. on different UEs)

Protocol Layer	Protocol Information	AMR Codec State	
		ON state at 12.2 kbps	OFF state comfort noise
Higher layer	RAB or Signalling RB	4 RAB sub-flows	4 RAB sub-flows
	Direction	Down link	Down link
RLC (in RNC)	Logical channel type	DTCH	DTCH
	RLC mode	TR	TR
	DCH0: AMR class A + padding	11 bytes	5 bytes
	DCH1: AMR class B + padding	13 bytes	0 bytes
	DCH2: AMR class A + padding	8 bytes	0 bytes
	DCH3: AMR signalling	0 or 10 bytes	0 or 10 bytes
	RLC header size	0	0
MAC (in RNC)	MAC header size	0	0
	MAC multiplexing	N/A	N/A
FP over Iur / Iub	Header CRC and CFN	2 bytes	2 bytes
	TFI indication (for 4 RABs)	4 bytes	4 bytes
	TB size	As for DCH 0 to 3	As for DCH 0 to 3
	Payload CRC	2 bytes	2 bytes
	Total FP size	40 or 50 bytes	13 or 23 bytes
Layer 1 (Node B)	TrCH type	DCH	DCH
	TTI	20 ms	20 ms
	Coding type	TC ??	TC ??
	CRC	2 bytes	2 bytes

Table 9.4 : Protocol Layers and Formats over Iur / Iub for Voice Traffic

9.4.3 RNC Model for Data Traffic

The agreed model for simulation of the RNC for the support of data traffic is as follows :

- Data traffic model : as described in Table 9.3
- RNSAP or NBAP signalling : this traffic flow is ignored
- RRC signalling : this signalling traffic flow is ignored
- Direction of traffic : the main interest is in delay and traffic flow over Iur and Iub. The analysis is for downlink only but should also apply to the uplink
- Header compression in PDCP layer : ignored as it does not impact on the transport network

- Buffering in RNC : buffering in RLC layer until resources available over the radio interface as defined set of transport formats allocated for that user service over Iur and Iub with padding added if the waiting data can not fill the most suitable transport format.
- Size of buffer in RLC layer (in RNC) : 256 kbyte buffer size per connection should be large enough so there is no need to worry about dropping strategies in the simulations.
- Transmission time interval (TTI) : 40 ms (not related to the arrival rate over Iu)
- Transport formats over Iur and Iub : defined for various simulation scenarios (Table 9.4)
- Time offset for frames for each UE : using either discrete or continuous offset groups within a TTI but maintained constant for each UE data session.
- Access to resources : data traffic is independent of voice traffic (e.g. on different UEs)
- RLC protocol data units : 320 bits giving a transport block (TB) size of 336 bits over Iur/Iub

Protocol Layer	Protocol Information	Channel bit rate		
		64 kbps	144 kbps	384 kbps
Higher layer	RAB or Signalling RB	1 RAB	1 RAB	1 RAB
	Direction	Down link	Down link	Down link
RLC (in RNC)	Logical channel type	DTCH	DTCH	DTCH
	RLC mode	AM	AM	AM
	Payload (block) Size	40 bytes	40 bytes	40 bytes
	Maximum data rate	64 kbps	144 kbps	384 kbps
	RLC header size	2 bytes	2 bytes	2 bytes
MAC (in RNC) FP over Iur / Iub	MAC header size	0	0	0
	MAC multiplexing	N/A	N/A	N/A
	Header CRC and CFN	2 bytes	2 bytes	2 bytes
	TFI indication	1 byte	1 byte	1 byte
	Data Payload	See Table 9.6	See Table 9.6	See Table 9.6
	Payload CRC	2 bytes	2 bytes	2 bytes
	Total FP size	Payload+5 bytes	Payload+5 bytes	Payload+5 bytes
Layer 1 (Node B)	TrCH type	DCH	DCH	DCH
	TB size	336 bits	336 bits	336 bits
	TBs in transport format set	See Table 9.6	See Table 9.6	See Table 9.6
	TTI (see Table 9.6)	40 ms or 20 ms	40 ms or 20 ms	40 ms or 20 ms
	Coding type	TC	TC	TC
	CRC	16 bits	16 bits	bits

Table 9.5 : Protocol Layers and Formats over Iur / Iub for Data Traffic

Peak RLC payload data rate over Iur / Iub	Number of transport blocks included within a transport format set.	
	TTI = 40 ms (mandatory)	TTI = 20 ms (optional)
64 kbps	{0,1,2,3,4,6,8}	{0,1,2,3,4}
144 kbps	{0,1,2,4,8,12,16,18}	{0,1,2,4,8,9}
384 kbps	{0,1,2,4,8,12,16,20,24,32,40,48}	{0,1,2,4,8,12,16,20,24}

Table 9.6 : Transport Formats Sets over Iur / Iub for Data Traffic

9.5 Simulation Scenarios

9.5.1 Traffic Models

Various mixtures of traffic flow should be simulated as described in Table 9.7. A split between voice and data traffic is indicated but the number of users to be supported should be adjusted to provide various loading levels on the UTRAN interfaces while maintaining this traffic mix.

Traffic Scenarios	Voice Traffic (as % of total traffic)	Data Traffic (as % of total traffic)
Voice traffic only	100 %	0 %
Data traffic only	0 %	100 %
Mainly voice traffic	80 %	20 %
Mainly data traffic	20 %	80 %

Table 9.7 : Mixtures of traffic types for Simulation Scenarios

Quality of Service (QoS)

- Voice has priority over data
- Non preemptive on Iub, but voice frames may be interleaved with data on Iub if a voice packet arrives while a large fragmented data packet is being sent (but this would require data packet buffering in Node B).

AAL2/ATM Simulation

- Packet size : 45 byte mandatory, 64 bytes optional based on AAL2 specifications
- Use one ATM virtual channel for data and a separate ATM virtual channel for voice
- AAL2 timers : all CU timers set to 3 ms

9.5.2 Performance Metrics

The reference point is the entry to Node B. The delay to be evaluated is the 99.9 percentile (Table 9.8).

The most important performance criteria are delay and link utilisation. The delay figures contain the packetisation delay, the queuing delay at the transport level and the transmission delay from the RNC to Node B. Confidence intervals were calculated based on the results of several independent simulation runs.

The transmission delay over the Iur and Iub refers to the time from the start of the transmission of the frame protocol from the RLC or MAC layer in the RNC to the time when that frame protocol has been fully received by the Node B.

For voice traffic this would refer to the data within one DCH AMR codec payload. For data traffic this would refer to data inserted within one DCH frame protocol (not a complete user packet) for transmission over Iub.

Performance for voice and data traffic over Iub were derived for a range of traffic loads and mixtures of traffic flow (Table 9.7) for various TNL protocol options over Iub. The results of this work were compared to AAL2 transport as a reference over Iub.

Parameter	Statistic to be measured
Voice Traffic over Iub	Delay for 99.9% of transmissions (ms)
Data Traffic over Iub	Delay for 99.9% of transmissions (ms)

Table 9.8 : Performance Metrics for Simulation Scenarios

9.6 Traffic Flow and Transport over Iur and Iub Interfaces

9.6.1 Data Traffic over Iur and Iub

The amounts of overheads introduced by each protocol are under the following assumptions.

- Since both the data-payload sizes are large, we *do not* multiplex data packets in this study (without PPPmux schemes on data packets).
- We *do not* assume compression on user plane TCP/IP headers for data packets.

The RNC adds the required control field to data bits to form a data payload. Thus a data payload of the ON-state is estimated as in the Figure 9.3.

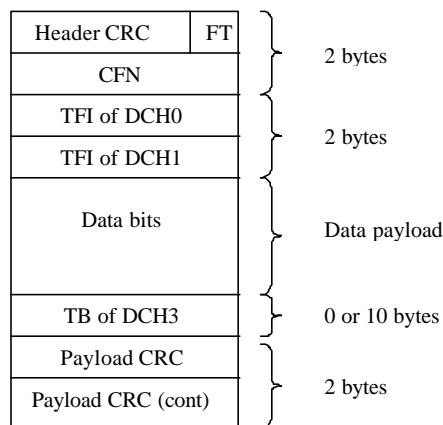


Figure 9.3 - Data Payloads for the ON-state

We assume that signalling messages are sent to a UE every 300 ms and consider the following two cases of the ON-state.

- Without signalling: $2(\text{Header CRC}) + 2(\text{TFI}) + \text{Data Payload (data bits)} + 2(\text{CRC}) = (\text{Data Payload} + 6)$ (bytes); and
- With signalling: $2(\text{Header CRC}) + 2(\text{TFI}) + \text{Data Payload (data bits)} + 10(\text{signalling}) + 2(\text{CRC}) = (\text{Data Payload} + 16)$ (bytes).

The average size of a data payload on the ON-state is thus (Data Payload +6.67) bytes. Each data packet consists of (Data Payload + 6.67) bytes and a data overhead over different protocols.

9.7 Protocol Stack Models for an IP Network

9.7.1 Summary of IP protocol stacks being studied

The companies involved in the simulation work (Alcatel, Lucent, Motorola) considered the performance of the protocol stacks in Table 9.9 for the Transport Network Layer over Iub in their simulation programmes. Note that AAL2/ATM is used as a reference for performance comparisons.

Protocol Stack	Companies Simulating this protocol stack		
cUDP/PPPMux/HDLC	Alcatel	Lucent	Motorola
cUDP/PPP/HDLC		Lucent	Motorola
TCRTP/HDLC	Alcatel		
cTCRTP/HDLC	Alcatel		
CIP/cUDP/PPP/HDLC	Alcatel		
LIPE/PPP/HDLC		Lucent	
cUDP/PPPMux/AAL5/ATM			Motorola
cUDP/AAL2/ATM		Lucent	Motorola
Reference Protocol Stack			
AAL2/ATM	Alcatel	Lucent	Motorola

Table 9.9 : Simulation work by Companies on Protocol Stack Options

9.7.2 Protocol Header and Payload Formats

When using IP as transport, payloads of various user flows are encapsulated with headers appropriate for the network they will traverse. It is beneficial to multiplex a few short payloads together in order to amortise the headers overhead.

A multiplexed packet is simply a concatenation of several encapsulated user payloads. In this section, we focus on the comparison of encapsulation of one payload via different proposed schemes in order to evaluate their efficiency as well as how they fit in the overall IP based RAN architecture.

PPPMux / cUDP :

1

2-3

len	cUDP header	payload
-----	-------------	---------

In this method, the payload is encapsulated in an IP packet. The IP packet is compressed using cUDP header compression. During compression of the RTP/UDP/IP header, a small sequence number called a context identifier (CID) is used to maintain synchronization and detect packet loss between the compressor and decompressor [RFC 2509]. The CID field can additionally be used to identify user flows, because it will be unique to each user flow.

The cUDP header includes the CRTP packet type and is usually 3 bytes long. When multiplexing with PPPMux, if the cUDP packet type of a multiplex is the same as in the previous multiplex, the cUDP packet type byte can be omitted and the header will be 2 bytes long (for cUDP, only two packet types are used. I.e. FULL_HEADER packet type is used once per user stream, the rest are COMPRESSED_UDP packet type.)

LIPE1:

1 2

len	UserID	payload
------------	---------------	----------------

In this method each user flow is assigned a UserID. The UserID field in each multiplex identifies the payload. Payloads of different streams are multiplexed together. The multiplexed packet is encapsulated into an IP packet.

LIPE2 / (crtp/cudp):

1 3

len	crtp/cudp header	payload
------------	-------------------------	----------------

Similar to PPPmux proposal, LIPE2 carries payloads encapsulated in IP packets. The IP packets are compressed using crtp/cudp header compression. The crtp/cudp header includes a CID field to identify each payload.

CIP:

2 1

CID	len	payload
------------	------------	----------------

Similar to LIPE1, the CID field identifies the payload. The multiplexed packet is encapsulated into an IP packet.

All schemes include a length field that indicates the length of the multiplex. There is not much difference in the length of one multiplex between the various schemes.

9.7.3 Overhead Introduced by Protocol Stack Options

In this section, we consider the amount of overhead introduced by each protocol stack. Listed below are assumptions made in order to estimate the amount of overhead added to each voice payload for different protocol stacks in our simulations.

- HDLC Address and Control fields can be signalled as unused, and are not required to be transmitted, therefore do not contribute additional overhead.
- For PPPmux, the size of PPPmux frame was limited to less than 300 bytes. This number is arbitrarily chosen so that some packets that have higher priority than voice packets, such as real-time, out-of-band, call control packets, are delayed by at most 1.25 msec behind a voice PPPmux frame. This corresponds to multiplexing about 10 voice payloads in a PPPmux frame on average.

- The padding bytes for partially-filled ATM cells when AAL2 is used were not included. The reason is that the padding bytes are amortized over a large number of ATM cells. Therefore, the overhead due to padding bytes per each voice frame becomes negligible.
- For cUDP/PPPMux/AAL5/ATM, the overhead due to PPPmux for each voice frame with this protocol stack is 1.2 bytes (i.e., $1 \text{ (length)} + [1 \text{ (PPPMux ID)} + 1 \text{ (PPP PID)}] / 10$). The AAL5 Padding and Trailer must be included.

The average total number of bytes (voice payload + protocol overhead) generated by a voice frame when different protocol stacks are used can be computed as follows:

cUDP/PPP-HDLC: Voice payload + 3 (cUDP/IP) + 4 (PPP-HDLC).
cUDP/PPPMux-HDLC: Voice payload + 3 (cUDP/IP) + 1.5 (PPPMux-HDLC).
cUDP/AAL2/ATM: [Voice payload + 3 (cUDP/IP) + 3 (AAL2 header)]*53.0/47.0.
cUDP/PPPMux/AAL5/ATM: [Voice payload + 3 (cUDP/IP) + 1.2 (PPPMux) + 23.5/10 (avg. AAL5 padding) + 8/10 (AAL5 trailer)]*53.0/48.0
LIPE: Voice payload + 3 (MH) + [1 (HDLC) + 3 (cUDP) + 1 (PPPMux) + 2 (CRC)] / 10
CIP: Voice payload + 3 (MH) + [1 (HDLC) + 4 (cUDP) + 1 (PPPMux) + 2 (CRC)] / 10

Bytes per voice frame Statistics		
Protocol stack	Average (bytes)	Variance (bytes)**2
AAL2/ATM	34.02	231.75
cUDP/PPP-HDLC	34.17	182.25
cUDP/PPPMux-HDLC	31.67	182.25
cUDP/AAL2/ATM	37.40	231.75
cUDP/PPPMux/AAL5/ATM	37.87	222.20
LIPE	30.87	182.25
CIP	30.97	182.25

Table 9.10- Average and variance of the total number of bytes generated by a voice frame

Table 9.10 shows a summary of the average and variance of the total number of bytes generated by a voice frame with all five protocol stacks under study.

10 PERFORMANCE OF IP IN THE RAN TRANSPORT

The focus of this section is on the performance of the Transport Network Layer over the Iub interface within 3GPP-UTRAN. The Iub is generally a lower bit rate interface than other interfaces within the RAN (e.g. Iur and Iu) leading to a greater impact on transport delays.

Options for the Transport Network Layer were described in Chapter 8 while general simulation models and traffic characteristics were discussed in Chapter 9. Within Chapter 10, the simulation results from the four companies (Alcatel, Lucent, Motorola and Siemens) are presented (within separate sub-sections) with discussions and conclusions by those companies.

The results from each of the companies are split into a general structure as follows.

- General Comments and introduction
- Protocol Stacks Simulated (but only deviations from the discussion in Chapter 8)
- Simulation Models and Traffic Models (but only deviations from the discussion in Chapter 9)
- Voice Traffic Only Simulation Results (100% voice) for all protocol options
- Data Traffic Only Simulation Results (100% data) for all protocol options
- Mainly Voice Traffic Simulation Results (80% voice plus 20% data) for all protocol options
- Mainly Data Traffic Simulation Results (80% data plus 20% voice) for all protocol options
- Discussion and Comparison of Results
- Conclusions by that Company (general consensus and conclusions are provided in Chapter 11)

10.1 Alcatel Simulation Results

Performance simulations were performed by applying the simulation framework as explained in Chapters 8 and 9 to compare different protocol stacks for UTRAN user plane. Pure voice traffic, pure data traffic and mixed voice/data are considered in the simulations as a traffic source. The simulation work is still ongoing, especially on the mixed traffic scenarios. Therefore, the document will be extended by additional simulation results.

Different protocol stacks are compared in terms of their efficiency on the Iub interface. The results compare the number of users that can be supported by these stacks. The limiting factors for all stacks are the total bandwidth of the link and the maximum delay permitted.

10.1.1 Protocol Stacks Simulated

The following protocol stacks have been simulated:

Voice only:

- AAL2/ATM (as reference)
- TCRTP via PPP/HDLC
- compressed TCRTP (cTCRTP) via PPP/HDLC and AAL5/ATM
- PPPMux via PPP/HDLC
- CIP via PPP/HDLC and AAL5/ATM

Data only:

- AAL2/ATM (as reference)
- CIP via PPP/HDLC

80% voice, 20% data:

- AAL2/ATM (as reference)
- CIP via PPP/HDLC

10.1.2 Simulation and traffic Models

The simulation model used for the Alcatel simulations were described in Chapter 9.

Voice and data traffic are transported over an E1 line at 1.92 Mbit/sec. Three traffic scenarios are presented : 100% voice traffic, 100% data traffic and a traffic mix with approximately 80% voice and 20% data traffic. The data traffic uses 64 kbps channels.

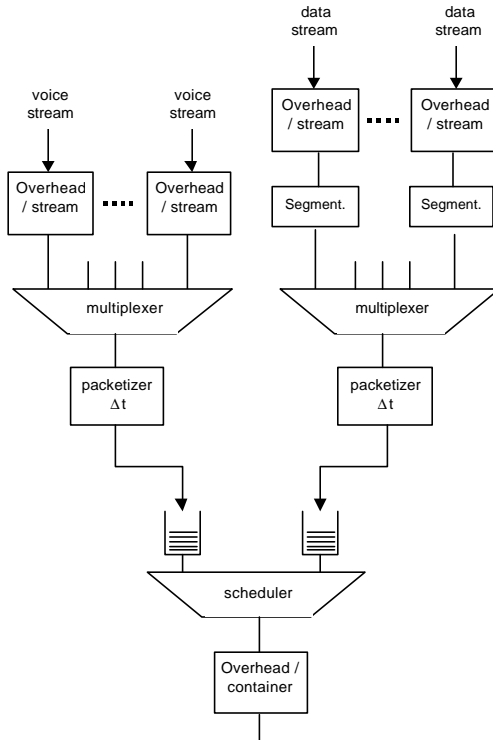


Figure 10.A1 : Implementation structure

The implementation structure in Figure 10.A1 shows one voice queue and one data queue for mixed voice/data simulations. In that case the voice queue is serviced until empty, at which time the data queue is serviced until the voice queue has become non-empty or the data queue is also empty. Voice packets cannot pre-empt data packets.

For pure voice traffic simulation, the data queue of the scheduler remains empty all the time, and vice versa. In fact this means the scheduler has no impact on the simulation result, but the queue is used to adapt the sources to the link rate. The same implementation structure is applied for all investigated protocol stacks.

10.1.3 Voice Traffic Only Simulation Results (100% voice)

The voice delay as depicted in Figure 10.A2 is measured for individual voice packets travelling from the RNC to the NodeB. As shown by the 99.9 percentile there are significant differences between the various protocol stacks. The maximum number of users supported by a predefined link is mainly determined by the efficiency of the protocol. There are two reasons for the delay of the packets: Packetisation and queuing/transmission. In a heavily loaded system packetisation delay is not significant because container packets are generated with a high rate. Therefore, the voice delay is predominantly determined by queuing and transmission delay.

Parameter	Value	Remark
Link Bit Rate	1,920,000 bit/sec	30*64 kbit/sec (E1)
Maximum Container Size	300 Byte	maximum size of a packet transported over the link
AMR Codec	12.2 kbit/s	model is described in detail in Chapter 9
Simulation duration	20,000 sec divided in 10 part tests	

Table 10.A1 : Simulation parameters

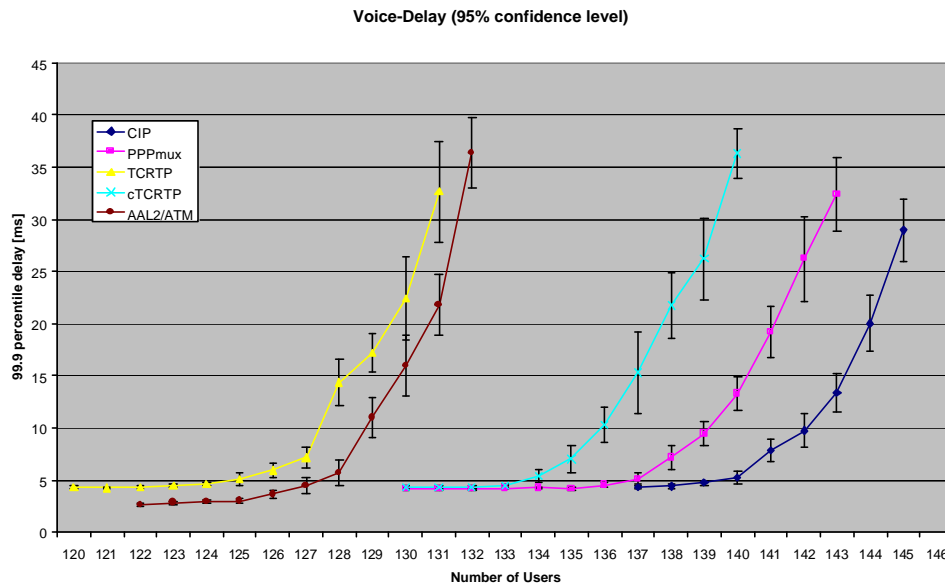


Figure 10.A2 :Voice packet delay versus number of concurrent voice users for a 1.92 Mbit/s link using PPP/HDLC transport

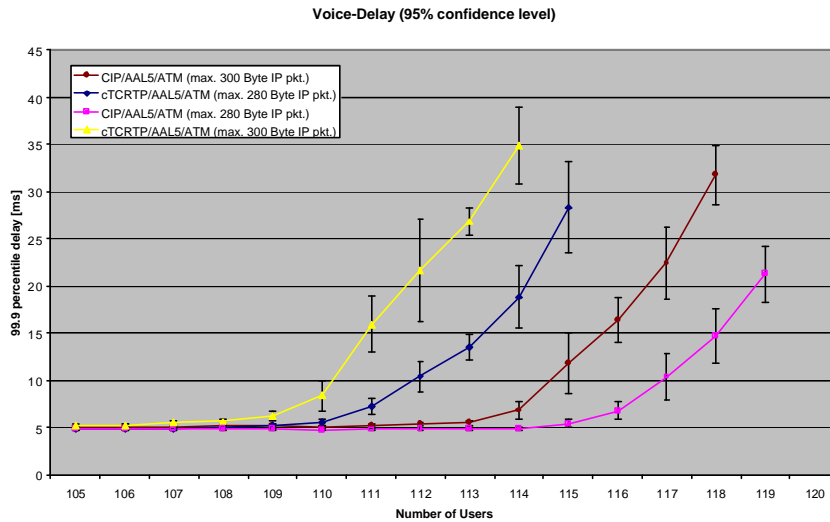


Figure 10.A3 : Voice packet delay versus number of concurrent voice users for a 1.92 Mbit/s link using AAL5/ATM transport

Utilisation

The link utilisation for the various protocols is depicted in Figure 10.A4. The link utilisation is almost linear within the simulated range of concurrent users because the system is heavily loaded and the additional overhead resulting from container packets that have not the maximum size is negligible.

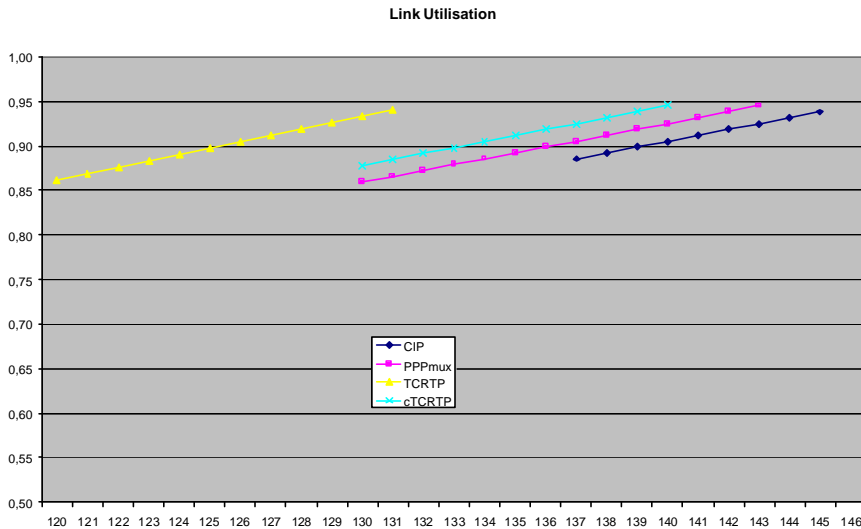


Figure 10.A4 : Link utilisation versus number of concurrent users for a 1.92 Mbit/s link

10.1.4 Data Traffic Only Simulation Results (100% data)

Please note that the results presented including data services are preliminary. The validation process is still ongoing. It is intended to update this document for the coming meetings.

FP-PDU delay

The FP-PDU delay as show in Figure 10.A5 is measured between the creation of a PDU in the FP layer in the RNC and the complete reception of the PDU in the NodeB.

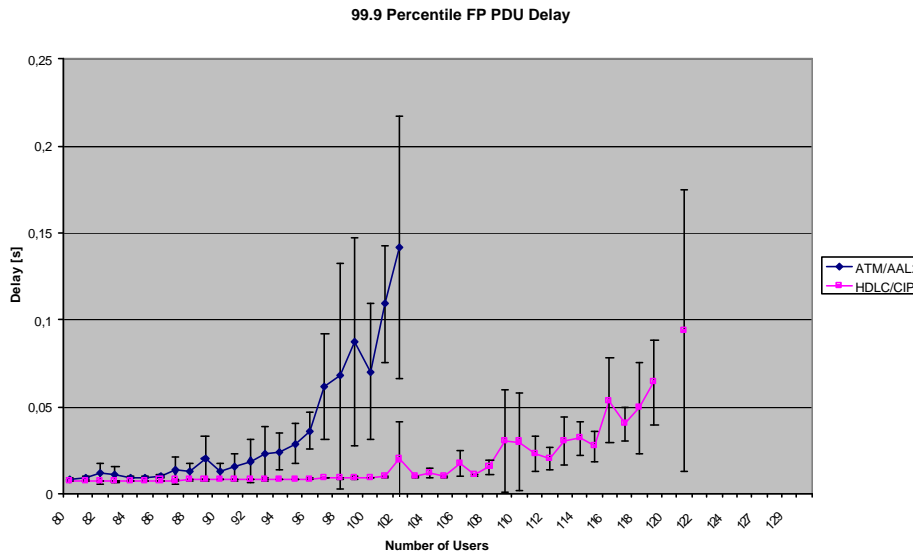


Figure 10.A5 : 99.9 Percentile FP-PDU Delay

It can be seen, that the HDLC/CIP protocol stack performs significantly better than ATM/AAL2

Utilisation

The link utilisation for the two protocol stacks ATM/AAL2 and HDLC/CIP is shown in Figure 10.A6. It can be derived from Figure 10.A5 and 10.A6, that the significant delay increase corresponds with a link utilisation of about 56 percent for both protocols.

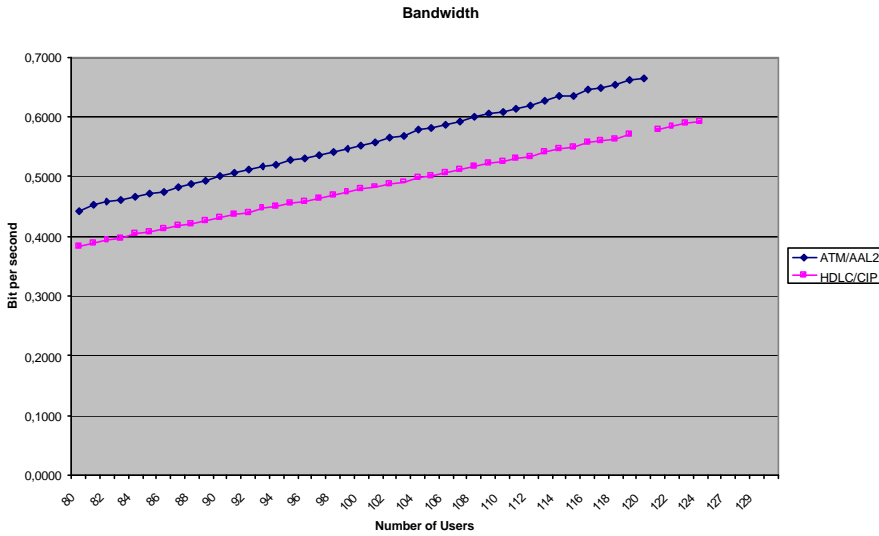


Figure 10.A6 : Link utilisation for 100% data traffic

10.1.5 Mainly Voice Traffic Simulation Results (80% voice : 20% data)

Because only whole-numbered voice and data users are realistic, a traffic mix is generated with 79% voice user traffic and 21% data user traffic.

FP-PDU delay

Figure 10.A7 depicts the 99.9 percentile of the FP-PDU delay for ATM/AAL2 and for HDLC/CIP.

Both protocol stacks perform well in respect the given QoS requirement to prioritise voice traffic. The 99.9 percentile of the CIP voice delay is marginal higher than the AAL2 delay. This is caused by the multiplexing container while having a moderate load. The lower overhead of the HDLC/CIP protocol stack results in a significant higher performance for data traffic.

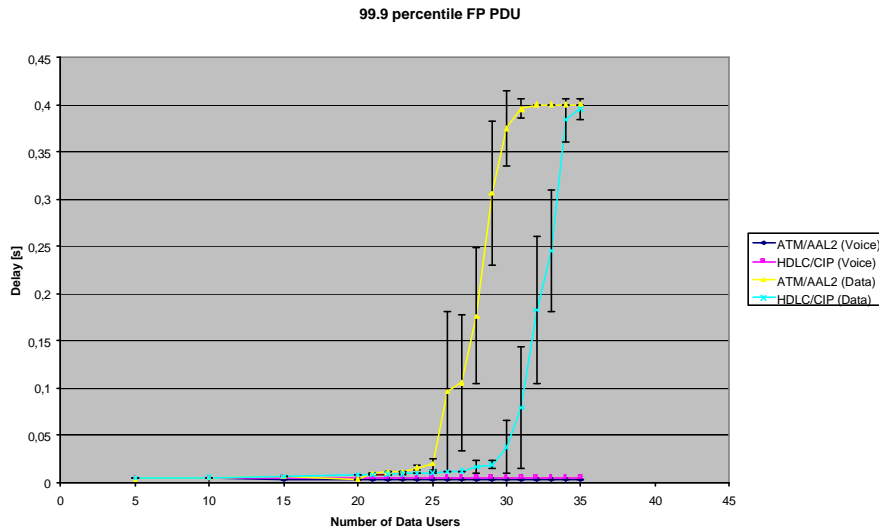


Figure 10.A7 : 99.9 percentile of the FP-PDU delay for 80:20 traffic mix

Utilisation

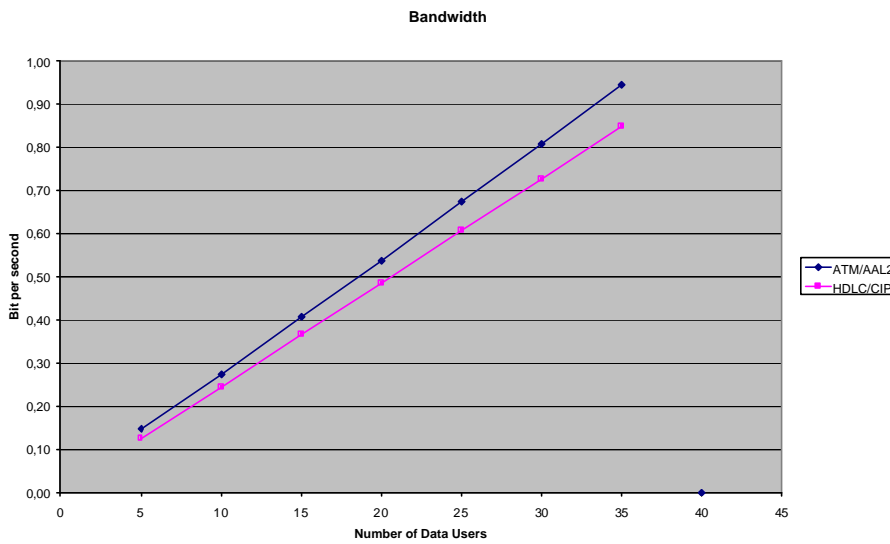


Figure 10.A8 : Bandwidth for 80:20 traffic mix

10.1.6 Mainly Data Traffic Simulation Results (80% data : 20% voice)

No results provided for this section.

10.1.7 Discussion and Comparison of Results

Voice Traffic Only

Assuming a maximum delay of 8ms, Tables 10.A2 to 10.A4 show the maximum number of concurrently active users which can be supported over a 1.92 Mbit/s link via PPP/HDLC and AAL5/ATM. The AAL2/ATM protocol is used as a reference. For all protocols a delay of 8 ms is equivalent to an utilisation of about 91%.

Protocol	Maximum Number of Users	
AAL2/ATM	128	100 %

Table 10.A2 : R99 Reference: Maximum number of users supported over a 1.92 Mbit/s

Protocol	Maximum Number of Users	
TCRTP	127	99 %
cTCRTP	135	105 %
PPPMux	138	108 %
CIP	141	110 %

Table 10.A3 : Maximum number of users supported over a 1.92 Mbit/s link (PPP/HDLC)

Protocol	Maximum Number of Users	
cTCRTP (max. 300 Byte IP pkt.)	110	86 %
cTCRTP (max. 280 Byte IP pkt.)	111	87 %
CIP (max. 300 Byte IP pkt.)	114	89 %
CIP (max. 280 Byte IP pkt.)	116	91 %

Table 10.A4 : Maximum number of users supported over a 1.92 Mbit/s link (AAL5/ATM)

Data Traffic Only

The HDLC/CIP protocol stack has a significant better performance than the AAL2/ATM stack. This is mainly caused by its higher efficiency. Assuming a maximum 99.9 percentile delay of 100 msec with a user data rate of 64 Kbit/s, ATM/AAL2 will support about 102 users while CIP will support 121 users.

Mainly Voice Traffic (80% voice, 20% data)

The HDLC/CIP protocol stack performs better than ATM/AAL2. The marginal higher 99.9 percentile for the voice FP-PDU delay in a moderate loaded system is uncritical. Due to the lower overhead CIP will support more concurrent users than ATM/AAL2 in a heavy loaded system.

10.1.8 Conclusions by Alcatel

Simulation results describing the bandwidth efficiency and delay properties of different proposed transport protocol stacks for use in an IP-based RAN have been presented.

The results show that within the environment described in Chapter 9 most protocols perform better than the reference protocol stack (AAL2/ATM) when used with PPP/HDLC link layer. One exception is TCRTP which has a degraded performance due to its tunneling overhead.

During the simulations it has been noticed that the maximum size of the transported packets is an important issue. The maximum packet size was limited to 300 bytes. FP PDUs larger than this were segmented.

The CIP protocol with its low overhead and segmentation capability shows a very good performance in all investigated scenarios.

10.2 Lucent Simulation Results

In this section Lucent describes two IP-based transport layer alternatives to ATM/AAL2 for transport over the Iub interface. The current studies assume a point to point physical link between RNC and Node B. This section compares the voice call carrying capacity of the RNC–Node B link with an IP based protocol and ATM/AAL2 protocol. The capacity for non-real time services such as web browsing and mixed voice and data scenarios are also described to assess the impact of interaction between voice and data traffic with respect to call carrying capacity.

The use of IP in the UTRAN can influence all aspects of Iub such as signalling, O&M message communication and control plane messaging. However, in this document we are concerned with the impact of IP on call carrying capacity.

10.2.1 Protocol Stacks Simulated

AAL2 - ATM - L1 stack

Container Overhead	5 bytes + 1 byte AAL2 prefix
Stream overhead	3 bytes
Maximum size of AAL2 PDU	45 bytes
Container size with overhead	53 bytes fixed

In ATM /AAL2 transport, the packets from layer 3 (framing protocol PDU (FP PDU) in the Iub) are fragmented into blocks with maximum size 45 bytes, if needed. An ATM/AAL2 header of 3 bytes is added to each fragment. The AAL2 header contains a connection identifier and a length field. The fragments are packed into ATM cells. If a fragment does not completely fit into a cell, part of it is sent in the given cell and the remainder portion is sent in the subsequent cell.

LIPE – IP – L2/L1 stack

Container Overhead	HDLC HDR	1 byte
	PPP ID	1 byte
	c_UDP	3 bytes
	CRC	2 bytes
	Total	7 bytes
Stream overhead	LIPE CID + CRC + Length	3 bytes
Maximum size of PDU	Not specified, however fragmentation is done to keep the overall container size limited. We limit the container size to 300 bytes.	
Container size with overhead	307	

The LIPE scheme uses either UDP/IP or IP as the transport layer. Each LIPE encapsulated payload consists of a variable number of multimedia data packets (MDP). For each MDP, there is a

multiplexing header (MH) that conveys protocol and media specific information. The frame format for a LIPE frame over a PPP connection is given in Figure 10.L1.

HDLC PPP 2 bytes	c_UDP/IP 3 bytes	MH 3 bytes	PAYLOAD	MH 3 bytes	PAYLOAD	CRC 2 bytes
------------------------	---------------------	---------------	---------	---------------	---------	----------------

Figure 10.L1 : Structure of a LIPE frame

10.2.2 Simulation and Traffic Models

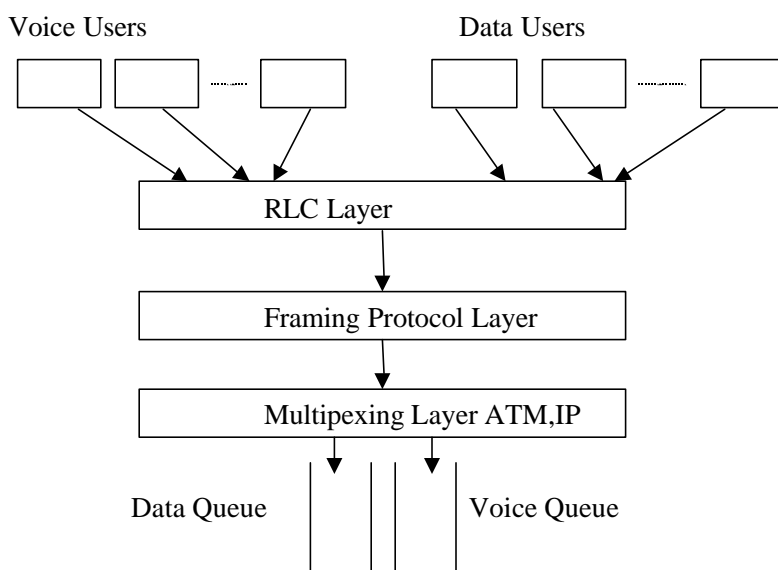


Figure 10.L2 : Schematic of the Simulation Model

A schematic of the simulation model is presented Figure 10.L2. The RLC layer is transparent to the voice users. At the multiplexing layer, voice and data packets are treated separately. Voice and data frames are never multiplexed into the same container. The voice queue is given highest priority. Packets from the data queue are scheduled only if the voice queue is empty.

The TBS (transport block set) format used for data is for peak rate of 64 kbps with a TTI of 40 ms. Packets are fragmented according to the TBS at the RLC layer. A TBS frame is sent every TTI .

Voice Traffic Model

Silence and talk spurts are modelled as a Markov modulated ON/OFF process with the following parameters:

- ON time – exponentially distributed with mean 3s.
- OFF time – exponentially distributed with mean 3s.
- Packet size in the ON state with FP overheads = 40 bytes.

- Packet size in the OFF state with FP overheads = 13 bytes.
- A signalling overhead of 10 bytes is added every 300ms to the voice payload.
- The TTI (transfer time interval) for voice service is assumed to be 20ms.

Data traffic

We use a web data model where the file size is Pareto distributed with mean 12000 bytes. The file is split into IP packets with a maximum size of 1500 bytes and sent back to back to the RNC. The duration between the receipt of a file and transmission of the next file (also known as “think time”) is exponentially distributed with a mean of 12s. The transfer time interval for data is assumed to be 40ms. The TBS (transport block set) format used for data has a peak rate of 64kbps with a TTI of 40ms. Packets are fragmented according to the TBS at the RLC layer.

10.2.3 Voice Traffic Only Simulation Results (100% voice)

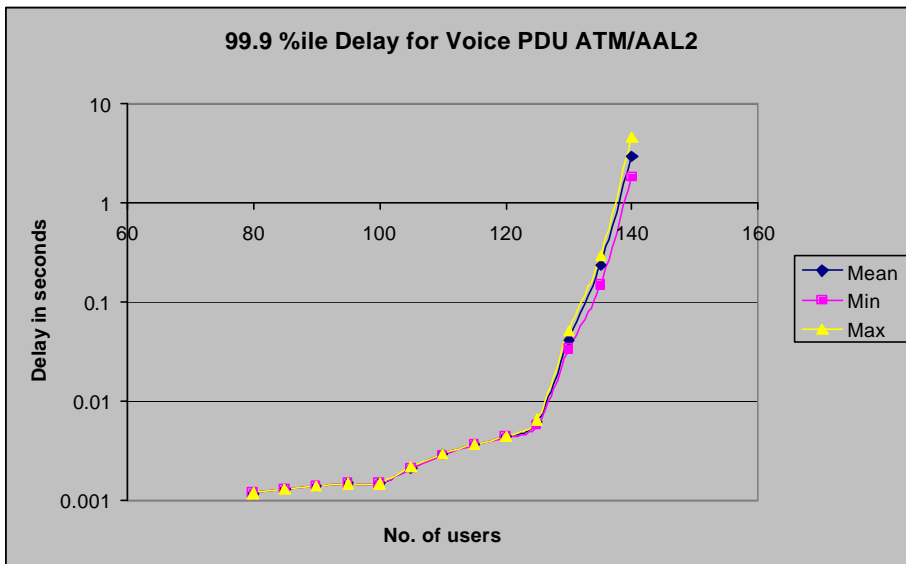


Figure 10.L3 : 99.9%ile Delay for Voice PDU for AAL2/ATM

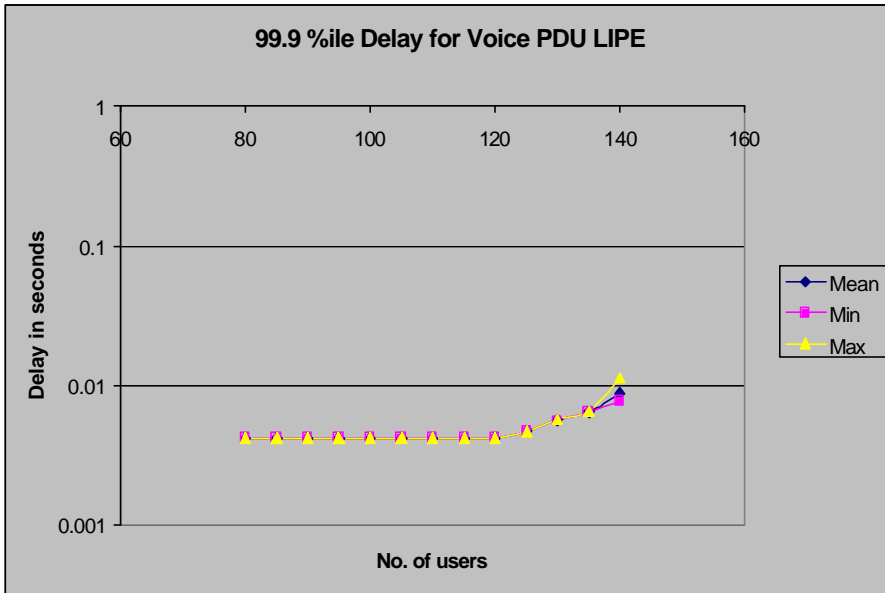


Figure 10.L4 : 99.9%ile Delay for Voice Traffic and LIPE

10.2.4 Data Traffic Only Simulation Results (100% data)

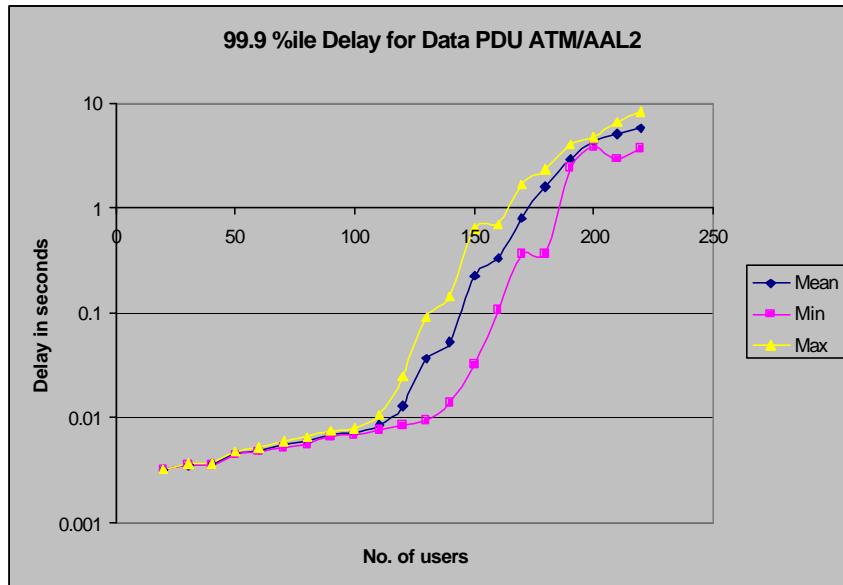


Figure 10.L5 : 99.9%ile Delay for Data Traffic and AAL2/ATM

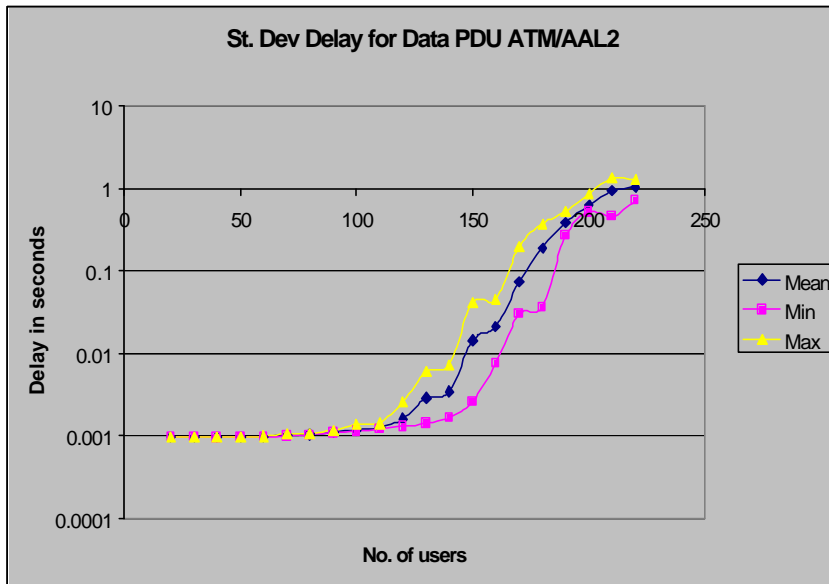


Figure 10.L6 : Standard Deviation of Delay for Data Traffic and AAL2/ATM

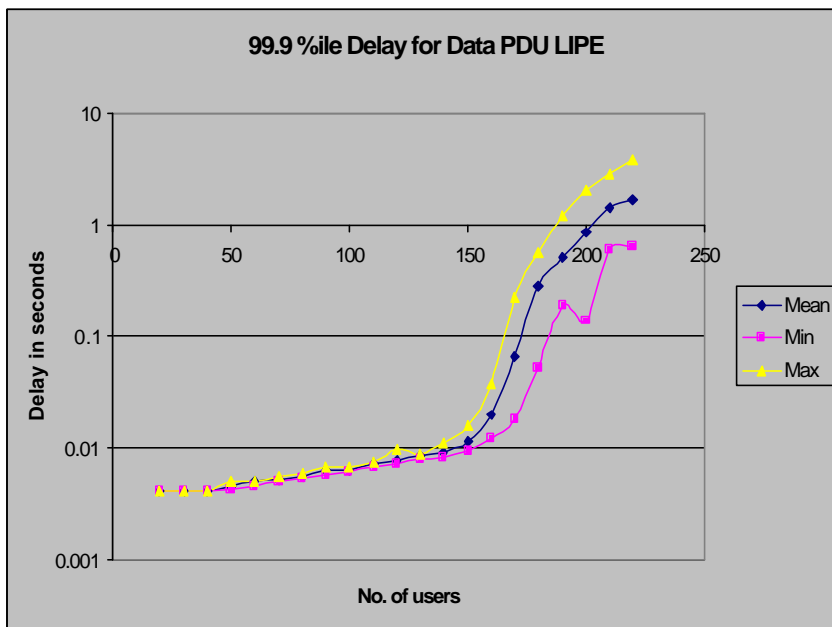


Figure 10.L7 : 99.9%ile Delay for Data Traffic and LIPE

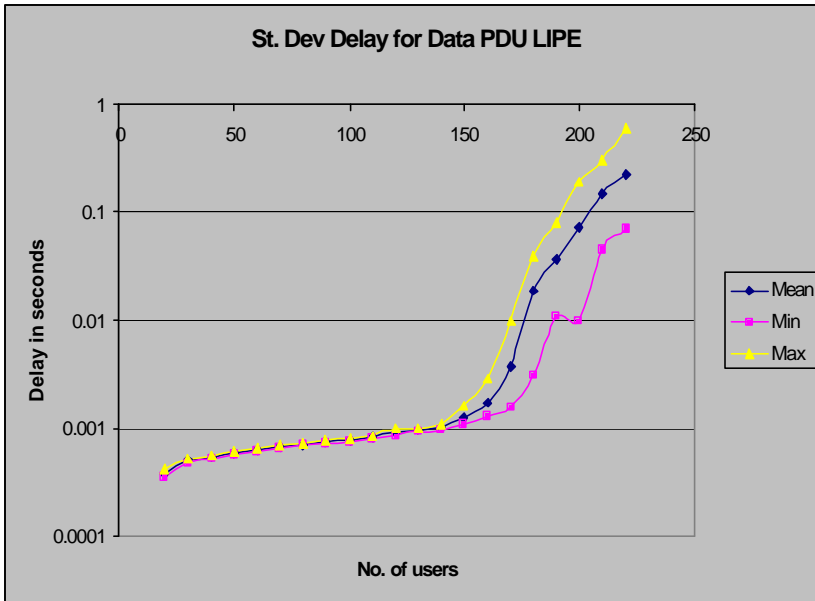


Figure 10.L8 : Standard Deviation of Delay for Data Traffic and LIPE

10.2.5 Mainly Voice Traffic Simulation Results (80% voice : 20% data)

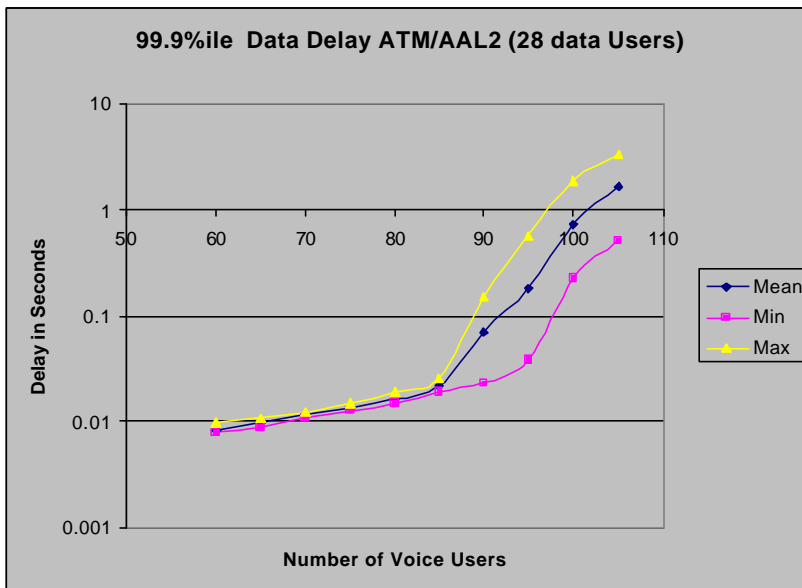


Figure 10.L9 : 99.9%ile Delay for Data Traffic and ATM/AAL2 (28 data users)

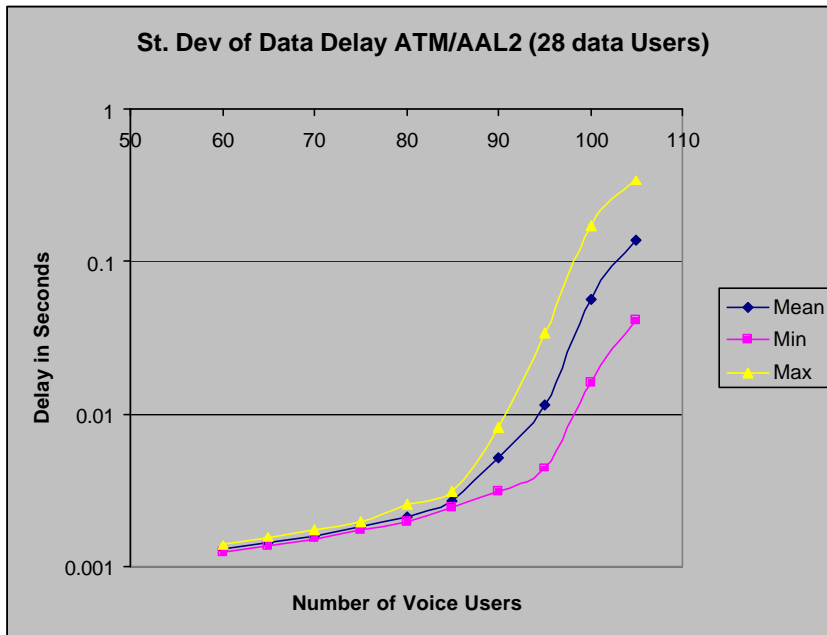


Figure 10.L10 : Standard Deviation of Delay for Data Traffic and ATM/AAL2 (28 data users)

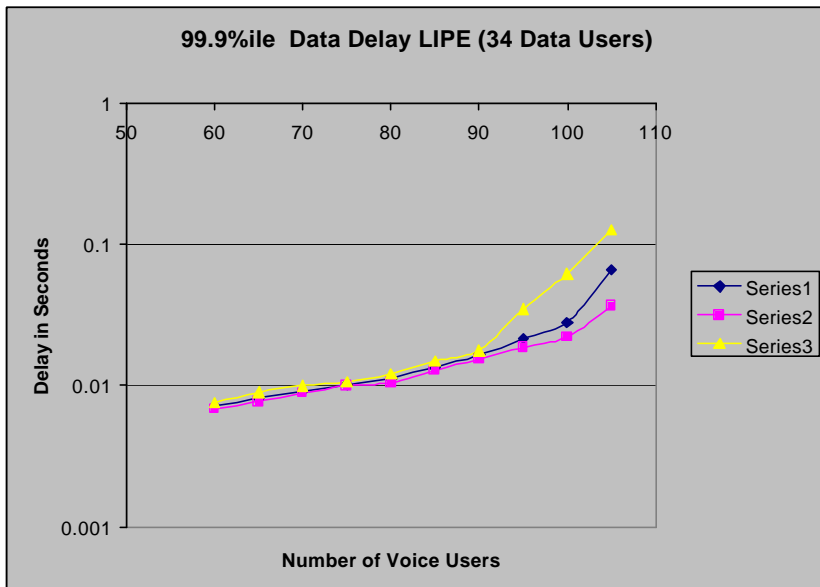


Figure 10.L11 : 99.9%ile Delay for Data Traffic and LIPE (34 data users)

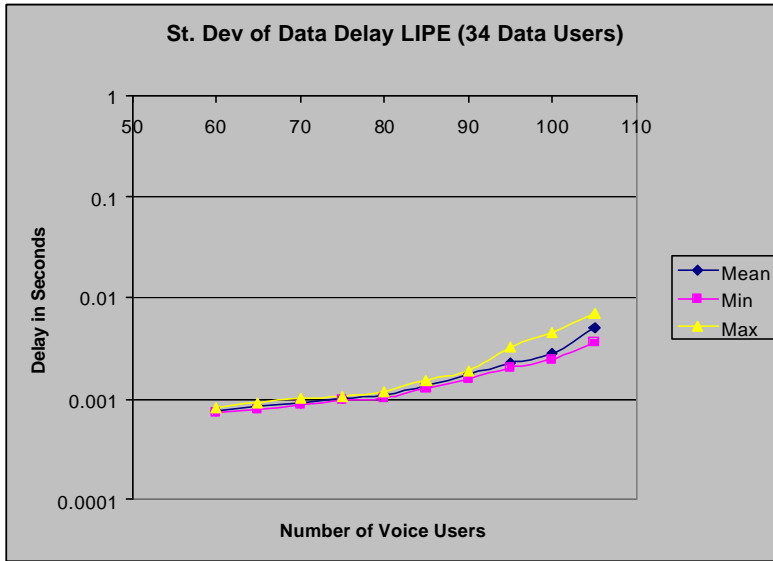


Figure 10.L12 : Standard Deviation of Delay for Data Traffic and LIPE (34 data users)

10.2.6 Mainly Data Traffic Simulation Results (80% data : 20% voice)

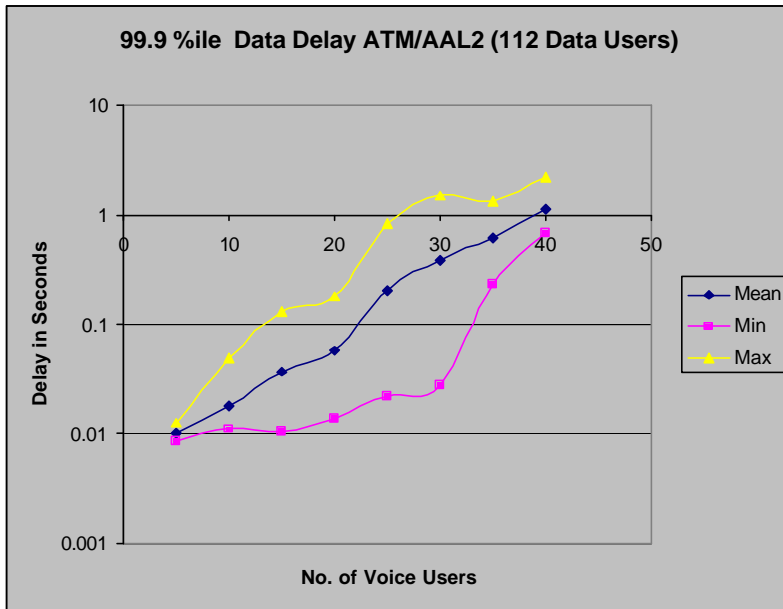


Figure 10.L13 : 99.9%ile Delay for Data Traffic and ATM/AAL2 (112 data users)

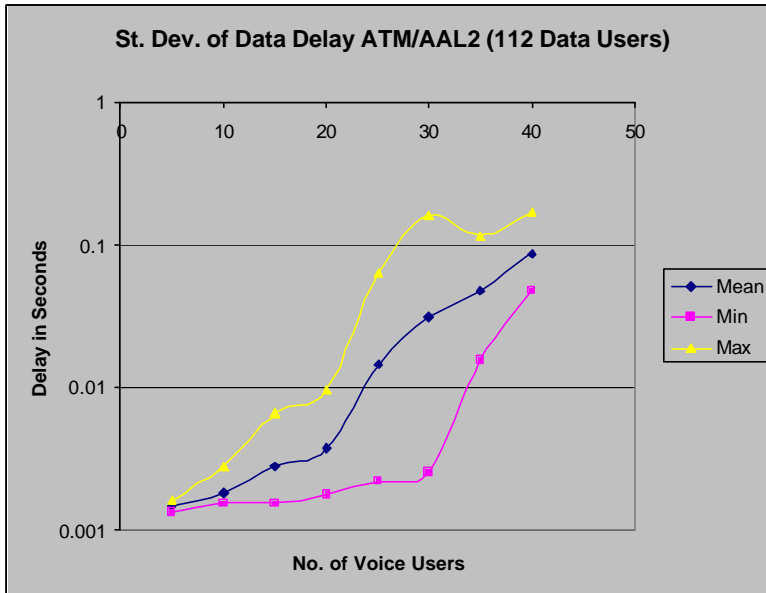


Figure 10.L14 : Standard Deviation of Delay for Data Traffic and ATM/AAL2 (112 data users)

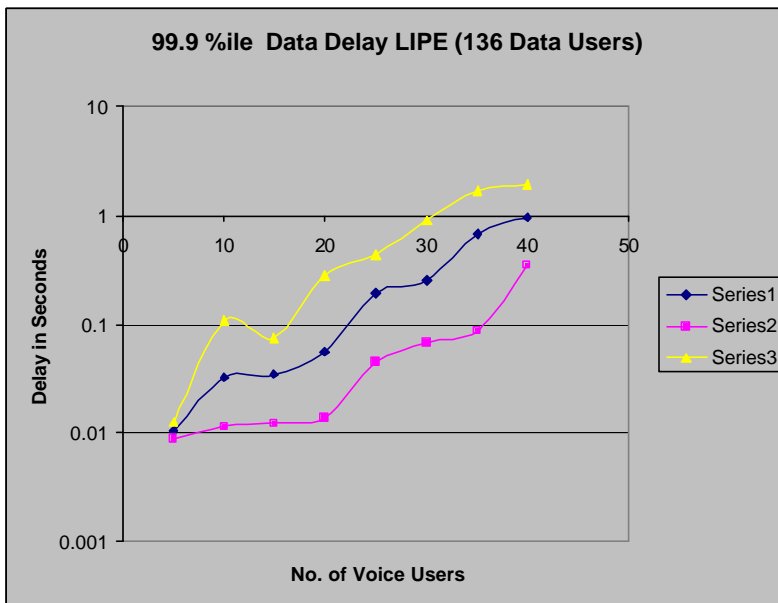


Figure 10.L15 : 99.9%ile Delay for Data Traffic and LIPE (136 data users)

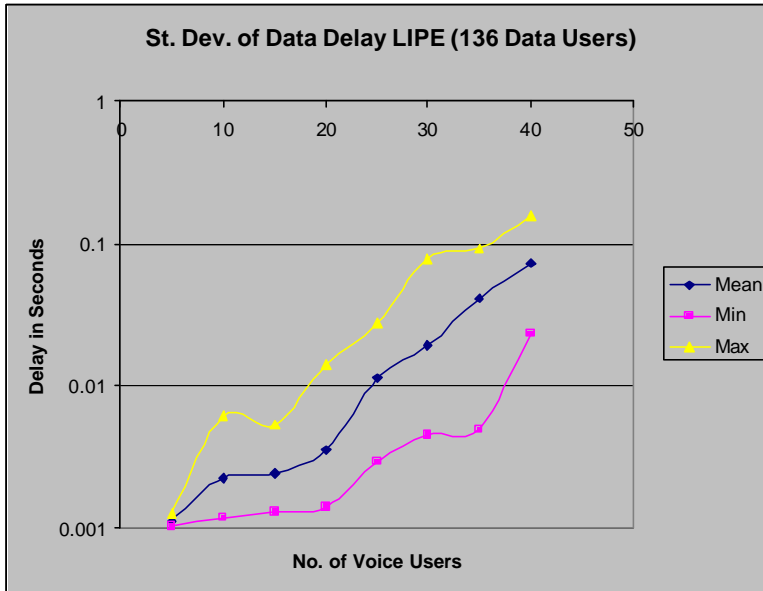


Figure 10.L16 : Standard Deviation of Delay for Data Traffic and LIPE (136 data users)

10.2.7 Discussion and Comparison of Results

Voice only results

The call carrying capacity of voice is presented in Table 10.L1. The capacity is defined to be the number of voice calls that can be supported keeping the 99.9%ile delay below 6ms.

Transport type	Criterion	Measured Capacity (voice users)	Utilisation	99.9%ile voice delay (ms)	St. deviation of voice delay (ms)
ATM AAL2	99.9%ile delay <= 6ms	120-125	85.6%-89.1%	4.4-6.1	0.6-1.0
IP LIPE		130-135	82.5%-85.6%	5.64-6.04	0.74-0.86

Table 10.L1 : Voice call carrying capacity for the ATM/AAL2 and LIPE

Data only results

The capacity of the three protocols for data sessions is presented in Table 10.L2. Note that the delay budgets for data traffic are higher, however due to synchronous nature of the air link, it is essential for the inter arrival times of the FP PDUs to be within the TTI. Hence the jitter in the delay of data FP PDUs is critical to call carrying capacity. We used the standard deviation of the delay as a metric for the jitter. Our capacity is criterion is to have the standard deviation of the delay to be 20 % of the TTI (8ms).

Transport type	Criterion	Measured Capacity (data users)	Data Throughput (kb/s)	99.9%ile data delay (ms)	St. dev. of data delay (ms)
ATM AAL2	Std. Dev. Of Delay averaged over experiments < 8ms	140	930	51	7.3
IP LIPE		170	1128	65.3ms	3.75

Table 10.L2 : Data session carrying capacity for the three transport protocols

Data+Voice results for 64 kbps case 20%data

In this scenario, the number of data users is set to 20% of the data carrying capacity, and the number of voice users is varied. We obtain the maximum number of voice users that can be supported subject to keeping the standard deviation of the data delay variance of data below 8ms. The 99.9%ile delay for voice was observed to be less than 6ms. Table 10.L3 contains the results.

Transport type	Criterion	Number of Data users (fixed)	Number of voice users	99.9%ile data delay (ms)	St. dev. of data delay (ms)	Utilisation %
ATM AAL2	Std. Dev. Of Delay averaged over experiments < 8ms	28	90	69.8	5.2	79
IP LIPE		34	105	66.17	5.0	81.62

Table 10.L3 : Voice carrying capacity with a 20% data load.

Data+Voice results for 64 kbps case : 80%data

In this scenario, the number of data users is set to 80% of the data carrying capacity, and the number of voice users is varied. We obtain the maximum number of voice users that can be supported subject to keeping the standard deviation of the data delay variance of data below 8ms. The 99.9%ile delay for voice was observed to be less than 6ms. Table 10.L4 contains the results.

Transport type	Criterion	No. of Data users (fixed)	Number of voice users	99.9%ile data delay (ms)	St. deviation of data delay (ms)	Utilisation %
ATM AAL2	Std. Dev. Of Delay averaged over experiments < 8ms	112	20	57	3.7	69.7
IP LIPE		136	20	54.8	3.53	69.43

Table 10.L4 : Voice carrying capacity with a 80% data load.

10.2.8 Conclusions by Lucent

In this contribution, we have presented preliminary results on the capacity of AAL2/ATM and LIPE (an IP-based solution) at the Iub interface for voice only, data only and mixed voice and data scenarios. Our results show that the proposed IP-based solution (with PPP/HDLC link layer) gives a higher transport efficiency when compared to the AAL2/ATM solution.

10.3 Motorola Simulation Results

The simulations were designed to measure efficiency of bandwidth utilisation by e.g. multiplexing/header compression resource management, and the use of segmentation. Lower speed

links, such as E1, or shared higher speed links may require different techniques (e.g. header compression and multiplexing) than dedicated higher speed links.

The section presents the simulation results and comparison of the capacity (defined by the number of voice users and that of data users a system can support simultaneously) of three protocol stacks on the Iub. This includes AAL2/ATM (as a reference) plus two other low-overhead protocol stacks, i.e. PPPmux/HDLC and LIPE.

10.3.1 Protocol Stack Simulated

The cUDP header includes the cUDP packet type and is usually 3 byte long. However when multiplexing with PPPmux, if the cUDP packet type of a following multiplex is the same as in the previous multiplex, the cUDP packet type byte can be omitted, resulting in 2 byte long cUDP overhead. The simulation considers the worst case, using 3 bytes for cUDP header. The average overhead for one payload while multiplexing 10 voice frame payloads is described in the following.

PPPMux for voice: FP PDU + 4 (cUDP + Len) + [1(HDLC)+ 1 (PPPMux) + 1 (PPP ID) + 2(CRC)]/10 (bytes)

PPP for data: FP PDU + 3 (cUDP) + 1 (PPP ID) + 1 + (HDLC) + 2 (CRC) (bytes)

(1) **LIPE** (see [TSG2146]):

LIPE for voice: FP PDU + 3 (MH) + [1(HDLC)+ 3 (cUDP) + 1 (PPPMux) + 2(CRC)]/10 (bytes)

PPP for data: FP PDU + 3 (cUDP) + 1 (PPP ID) + 1 + (HDLC) + 2 (CRC) (bytes)

(2) **CIP** (see [TSG1716]):

Without segmentation, the average size of one payload is as follows.

CIP for voice: FP PDU + 3 (MH) + [1(HDLC)+ 4 (cUDP) + 1 (PPP) + 2(CRC)]/10 (bytes)

PPP for data: FP PDU + 3 (cUDP) + 1 (PPP ID) + 1 + (HDLC) + 2 (CRC) (bytes)

The average overhead of a voice FP PDU for the above protocols satisfies the following expression:

PPPMux (4.5bytes for worst case) < CIP (3.8 bytes without segmentation) < LIPE (3.7 bytes)

The simulation was only performed on PPPmux and LIPE.

10.3.2 Simulation and Traffic Models

The simulation and traffic models used by Motorola were described in Chapter 9.

10.3.3 Voice Traffic Only Simulation Results (100% voice) (AAL2; PPPmux; LIPE)

Figure 1. 99.9% voice frame delay for protocol stacks

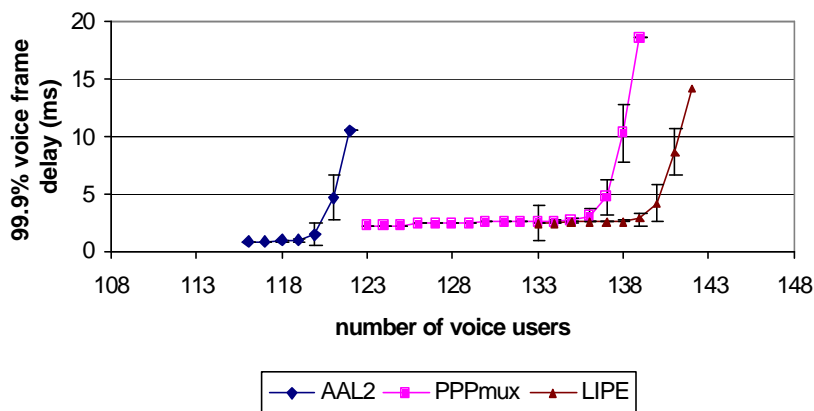


Figure 10.M1 : 99.9% Voice Frame Delay for Protocol Stacks

Figure 10.M1 shows that the knee is at 120 voice users for AAL2/ATM, at 136 voice users for PPPmux/HDLC, and at 140 voice users for LIPE. Table 10.M1, 10.M2 and 10.M3 present the statistics of interest.

# of voice users	118	119	120	121	122
AAL2 utilisation (%)	87.36	88.1	89.03	89.58	90.31
99.9% voice delay (ms)	0.9	0.98	1.39	4.65	10.49

Table 10.M1 : The statistics of voice only case for AAL2/ATM

# of voice users	134	135	136	137	138	139
PPPmux utilisation (%)	86.16	86.83	87.46	88.11	88.76	89.38
99.9% voice delay (ms)	2.6	2.66	3.04	4.7	10.28	18.65

Table 10.M2 : The statistics of voice only case for PPPmux/HDL-C

# of voice users	137	138	139	140	141	142
LIPE utilisation (%)	88.11	88.76	89.38	90.06	90.08	91.33
99.9% voice delay (ms)	2.57	2.61	2.8	4.14	8.64	14.05

Table 10.M3 : The statistics of voice only case for LIPE

10.3.4 Data Traffic Only Simulation Results (100% data) (AAL2; PPP)

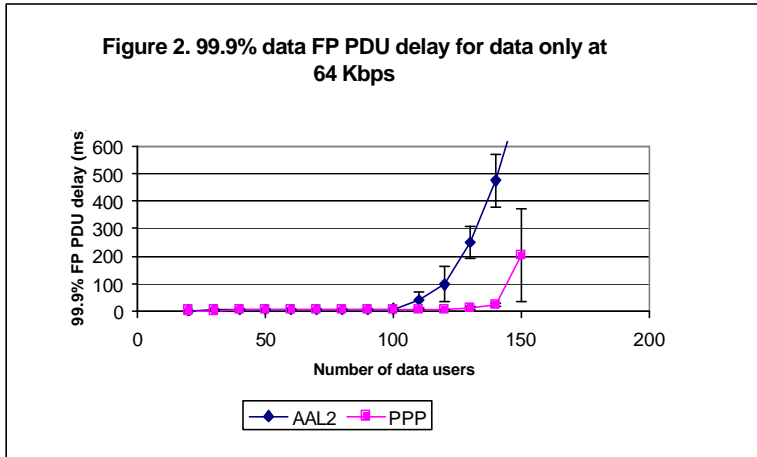


Figure 10.M2 : 99.9% data FP PDU delay for data only at 64 kbps

Figure 10.M2 shows the knee is at 120 data users for AAL2/ATM, and at 140 data users for PPP/HDLC.

# of data users	80	90	100	110	120	130	140
AAL2 utilisation (%)	37.74	42.5	46.67	52.46	56.66	61.29	65.79
99.9% delay (ms)	6.16	7.04	7.9	41.63	95.84	248.49	474.64

Table 10.M4 : The statistics of data only case at 64 kbps for AAL2/ATM

# of Data users	90	100	110	120	130	140	150
PPP utilisation (%)	36.54	40.13	45.1	48.72	52.69	56.56	61.02
99.9% delay (ms)	5.19	5.7	6.54	7.19	11.62	21.08	204.93

Table 10.M5 : The statistics of data only case at 64 kbps for PPP/HCLC

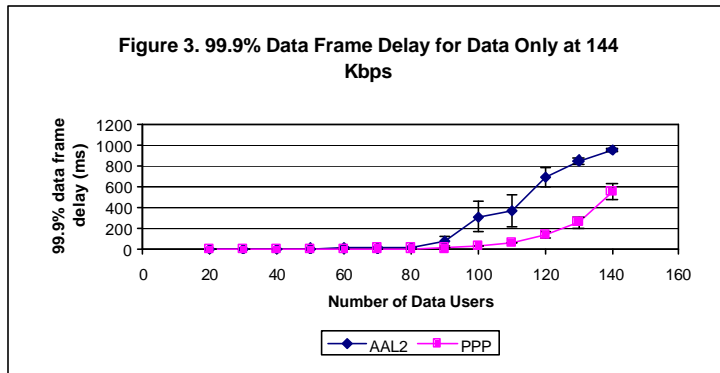


Figure 10.M3 : 99.9% data FP PDU delay for data only at 144 kbps

Figure 10.M3 shows the knee for AAL2/ATM is at 90 data users for AAL2, and at 110 data users for PPP/HDLC.

# of data users	60	70	80	90	100	110	120
AAL2 utilization (%)	29.4	34.38	39.46	44.26	49.35	54.49	59.59
99.9% delay (ms)	12.74	14.52	18.5	81.18	314.12	275.16	697.55

Table 10.M6 : The statistics of data only case at 144 kbps for AAL2/ATM

# of data users	80	90	100	110	120	130	140
PPP utilization (%)	33.76	37.87	42.27	46.62	50.98	54.73	59.13
99.9% delay (ms)	11.95	13.63	37.71	67.4	144.31	259.12	553.35

Table 10.M7 : The statistics of data only case at 144 kbps for PPP/HDLC

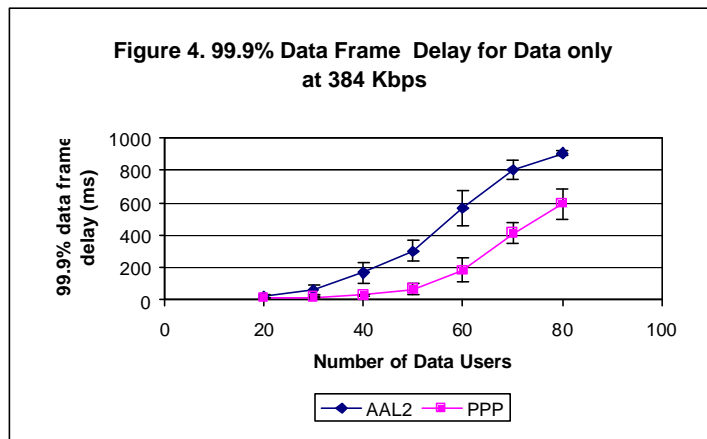


Figure 10.M4 : 99.9% data FP PDU delay for data only at 384 kbps

Figure 10.M4 shows the knee is at 30 data users for AAL2/ATM, and at 50 data users for PPP/HDLC.

# of data users	20	30	40	50
AAL2 utilisation (%)	10.87	16.38	21.76	27.17
99.9% delay (ms)	25.52	62.54	166.78	302.85

Table 10.M8 : The statistics of data only case at 384 kbps for AAL2/ATM

# of data users	40	50	60	70
PPP utilisation (%)	18.53	23.13	27.68	32.38
99.9% delay (ms)	29.14	67.89	185.31	413.46

Table 10.M9 : The statistics of data only case at 384 kbps for PPP/HDLC

10.3.5 Mainly Voice Traffic Simulation Results (80% voice : 20% data)

Table 10.M10 and 10.M11 contain the utilisation measures for data and voice. In these tables, each pair of the number of voice users and that of data users yields approximately 80% voice utilisation and 20% utilisation.

Pair	1	2	3	4	5	6	7
# of voice users	24	29	35	40	46	51	57
# of data users	15	17	19	21	23	25	27

Table 10.M10 : The combination of 80% voice and 20% data (1)

Pair (Cnt.)	8	9	10	11	12	13	14
# of voice users	62	68	73	80	88	93	98
# of data users	29	31	33	35	37	39	41

Table 10.M11 The combination of 80% voice and 20% data (2)

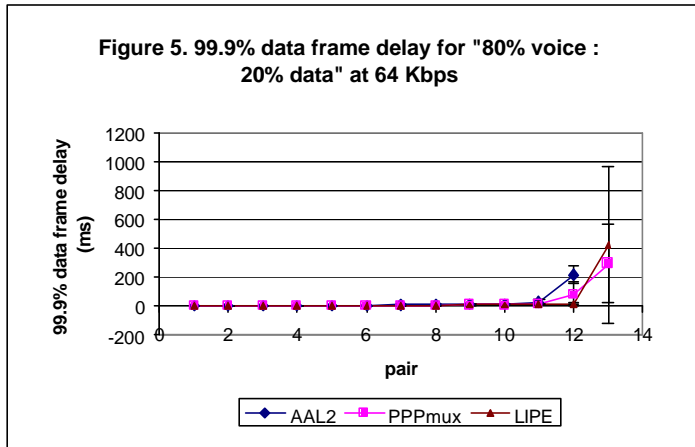


Figure 10.M5 : 99.9% data FP PDU delay for 80% voice and 20% data at 64 kbps

The 99.9% delay of voice frames is consistent for all pairs within each protocol stack: 1.9 ms for AAL2/ATM, from 3.4 ms to 3.5 ms for PPPmux/HDLC, and from 3.38 ms to 3.48 ms for LIPE. Figure 10.M5 shows the knee of 99.9% data frame delay is at 73 voice users/33 data users for AAL2/ATM, at 88 voice users/37 data users for both PPPmux/HDLC and LIPE. Table 10.M12 summarises the utilisation when the system reaches capacity.

Protocol stacks	AAL2/ATM	PPPmux/HDLC	LIPE
# of voice users	80	88	88
# of data users	35	37	37
Voice utilisation (%)	54.03	58.06	56.59
Data utilisation (%)	15.54	14.95	15.06
Total utilisation (%)	69.57	73.01	71.65

Table 10.M12 : The statistics of 80% voice v. s. 20 % data at 64 kbps

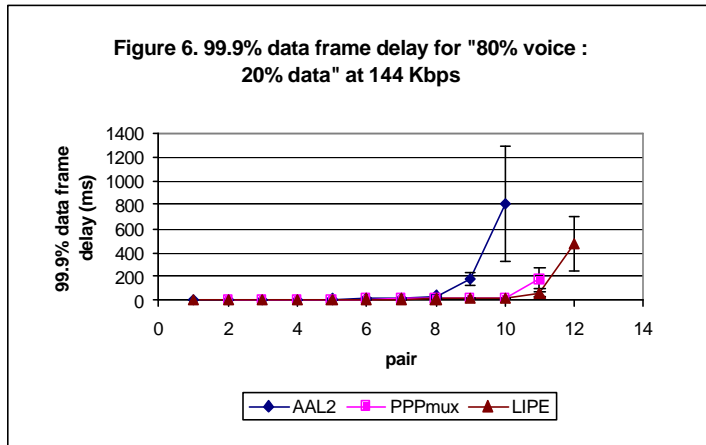


Figure 10.M6 : 99.9% data FP PDU delay for 80% voice and 20% data at 144 kbps

The 99.9% delay of voice frames is consistent for all pairs within each protocol stack: from 3.9 ms to 4.0 ms for AAL2/ATM, from 5.1 ms to 5.2 ms for PPPmux/HDLC, and from 4.98 ms to 5.08 ms for LIPE. Figure 10.M6 shows the knee of 99.9% data frame delay is at 62 voice users/29data users for AAL2/ATM, at 73 voice users/33 data users for PPPmux/HDLC, and at 80 voice users/35 data users for LIPE. Table 10.M13 summarises the utilisation when the system reaches capacity.

Protocol stacks	AAL2/ATM	PPPmux/HDLC	LIPE
# of voice users	62	73	80
# of data users	29	33	35
Voice utilisation (%)	45.9	48.17	51.45
Data utilisation (%)	14.39	13.68	14.72
Total utilisation (%)	60.29	61.85	66.17

Table 10.M13 : The statistics of 80% voice v. s. 20 % data at 144 kbps

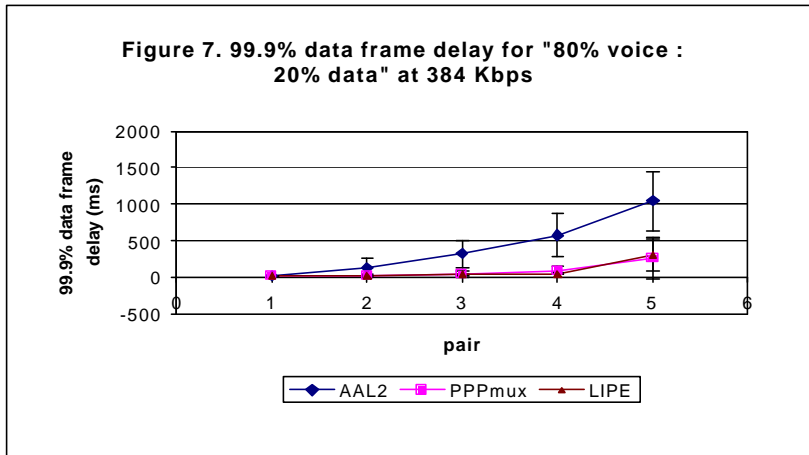


Figure 10.M7 : 99.9% data FP PDU delay for 80% voice and 20% data at 384 kbps

The 99.9% delay of voice frames is consistent for all pairs within each protocol stack: from 8.4 ms to 8.5 ms for AAL2/ATM, from 8.9 ms to 9.0 ms for PPPmux/HDLC, and from 8.88 ms to 8.98 ms for LIPE. Figure 10.M7 shows the knee of 99.9% data frame delay is at 24 voice users/15data users for AAL2/ATM, at 40 voice users/21 data users for PPPmux/HDLC, and at 40 voice users/21 data users for LIPE. Table 10.M14 summarises the utilisation when the system reaches capacity.

Protocol stacks	AAL2/ATM	PPPmux/HDLC	LIPE
# of voice users	24	40	40
# of data users	15	21	21
Voice utilisation (%)	17.78	26.4	25.72
Data utilisation (%)	8.19	9.53	9.62
Total utilisation (%)	25.97	35.93	35.34

Table 10.M14 : The statistics of 80% voice v. s. 20 % data at 384 kbps

10.3.6 Mainly Data Traffic Simulation Results (80% data : 20% voice)

Table 10.M15 contains the utilisation measures for voice and data. In these tables, each pair of the number of voice users and that of data users yields approximately 20% voice utilisation and 80% data utilisation.

Pair	1	2	3	4	5	6	7	8	9	10
# of voice users	10	11	12	13	14	15	16	17	18	19
# of data users	80	85	90	95	100	105	110	115	120	125

Table 10.M15 : The combination of 20% voice and 80% data (1)

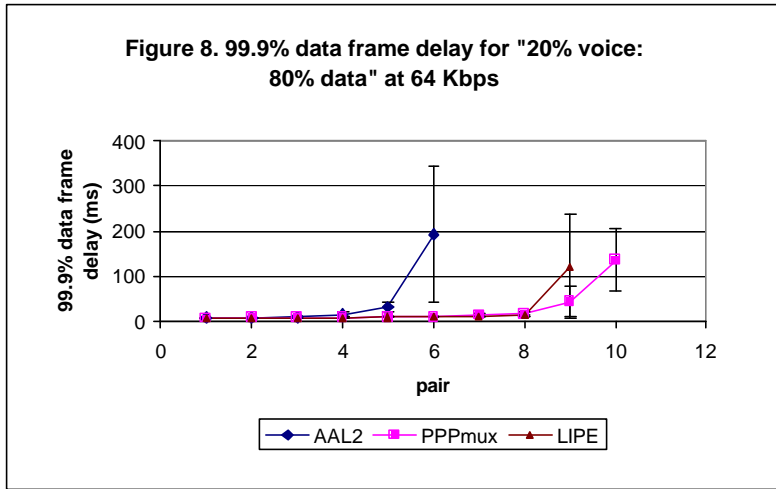


Figure 10.M8 : 99.9% data FP PDU delay for 20% voice and 80% data at 64 kbps

The 99.9% delay of voice frames is consistent for all pairs within each protocol stack: at 1.9 ms for AAL2/ATM, at 3.5 ms for PPPmux/HDLC, and from 3.38 ms to 3.48 ms for LIPE. Figure 10.M8 shows the knee of 99.9% data frame delay is at 24 voice users/15data users for AAL2/ATM, at 40 voice users/21 data users for PPPmux/HDLC, and at 40 voice users/21 data users for LIPE. Table 10.M16 summarises the utilisation when the system reaches capacity.

Protocol stacks	AAL2/ATM	PPPmux/HDLC	LIPE
# of voice users	14	18	17
# of data users	100	120	115
Voice utilisation (%)	10.36	11.87	10.93
Data utilisation (%)	46.83	48.32	46.27
Total utilisation (%)	57.19	60.19	57.10

Table 10.M16 : The statistics of 20% voice v. s. 80 % data at 64 kbps

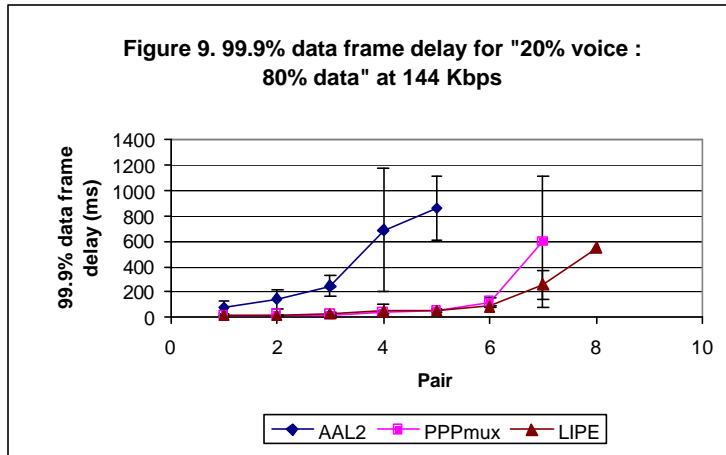


Figure 10.M9 : 99.9% data FP PDU delay for 20% voice and 80% data at 144 kbps

The 99.9% delay of voice frames is consistent for all pairs within each protocol stack: from 3.9 ms to 4.0 ms for AAL2/ATM, from 5.1 ms to 5.2 ms for PPPmux/HDLC, and from 5.08 ms to 5.18 ms for LIPE. Figure 10.M9 shows the knee of 99.9% data frame delay is at 10 voice users/80 data users for AAL2/ATM, at 14 voice users/100 data users for PPPmux/HDLC, and at 15 voice users/105 data users for LIPE. Table 10.M17 summarises the utilisation when the system reaches capacity.

Protocol stacks	AAL2/ATM	PPPmux/HDL-C	LIPE
# of voice users	10	14	15
# of data users	80	100	105
Voice utilisation (%)	7.41	9.24	9.64
Data utilisation (%)	39.62	41.62	44.04
Total utilisation (%)	46.67	50.86	53.68

Table 10.M17 : The statistics of 20% voice v. s. 80 % data at 144 kbps

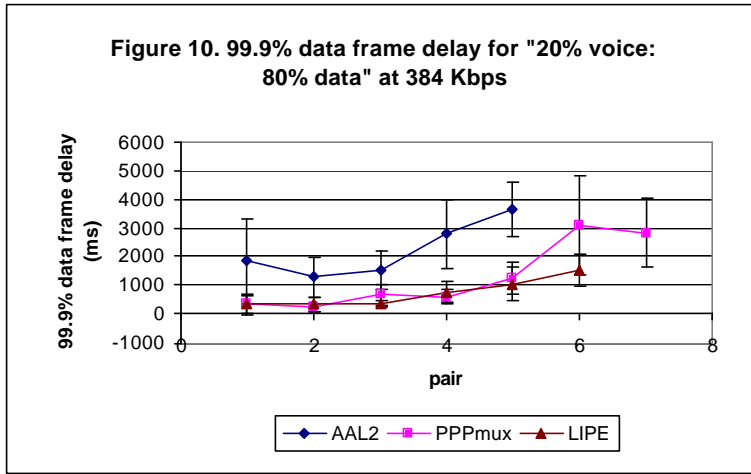


Figure 10.M10 : 99.9% data FP PDU delay for 20% voice and 80% data at 384 kbps

Pair	1	2	3	4	5	6	7	8	9
# of voice users	4	5	6	7	8	9	10	11	12
# of data users	50	55	60	65	70	75	80	85	90

Table 10.M18 : The combination of 20% voice and 80% data (2)

Table 10.M18 shows the number of voice users and that of data users corresponding Figure 10.M10. the figure does not show obvious knees for the 99.9% data delay curves.

10.3.7 Discussion and Comparison of Results

As a baseline case for comparison, simulation cases are run with voice and data traffic transported on the AAL2 protocol stack over an E1 line (30 DS0). Four scenarios are studied: 100% Voice Traffic, 100% Data Traffic, 80% Voice: 20% Data Traffic, and 20% Voice: 80% Data Traffic. Comparison of three radio-link data rates is studied: 64, 144, 384 kbps. Each statistic is the average of ten simulation runs. Each simulation run is 20,000 sec. The 95% confidence intervals of 99.9% voice frame delay and of 99.9% data frame delay are also measured.

The capacity is decided according to the “knee” of the curve of 99.9% voice frame delay, and of the curve of 99.9% data frame delay. However, in real systems other criteria should also be taken into account, for example, the delay requirement from RNC to Node B. We propose for voice frames the 99.9% delay from RNC to Node B to be less than 5 ms.

The 80% Voice: 20% Data scenario, and 20% Voice: 80% Data scenario are roughly according to the ratio of throughput of voice traffic and that of data traffic. The summary of simulation results is included in this section.

	Capacity (# of voice users)	99.9% delay for voice frames (ms)	Utilisation of E1 (%)
AAL2/ATM	120	<= 1.39	89.03
PPPMux/HDLC	136	<= 3.04	87.46
LIPE	140	<= 4.14	90.06

Table 10.M19 : Voice only (AAL2/ATM; PPPmux/HDLC; LIPE)

	Capacity (# of data users)	99.9% delay for data frames (ms)	Utilisation of E1 (%)
64 kbps			
AAL2/ATM	120	<= 95.84	56.66
PPP/HDLC	140	<= 21.08	56.56
144 kbps			
AAL2/ATM	90	<= 81.18	44.26
PPP/HDLC	110	<= 67.4	46.64
384 kbps			
AAL2/ATM	30	<= 62.54	16.38
PPP/HDLC	50	<= 67.89	23.13

Table 10.M20 : Data only for 64, 144, 384 kbps (AAL2/ATM; PPP/HDLC)

	Capacity		99.9% delay (ms)		Utilisation (%)	
	# voice users	# data users	Voice	Data	Voice	Data
64 kbps						
AAL2/ATM	80	35	<= 1.9	<= 28.15	54.03	15.54
PPPMux/HDLC	88	37	<= 3.5	<= 81.63	58.06	14.95
LIPE	88	37	<= 3.48	<= 17.88	56.59	15.06
144 kbps						
AAL2/ATM	62	29	<= 4.0	<= 37.38	45.9	14.39
PPPMux/HDLC	73	33	<= 5.2	<= 22.85	48.17	13.68
LIPE	80	35	<= 5.08	<= 64.35	51.45	14.72
384 kbps						
AAL2/ATM	24	15	<= 8.5	<= 25.57	17.78	8.19
PPPMux/HDLC	40	21	<= 9.0	<= 86.95	26.4	9.53
LIPE	40	21	<= 8.98	<= 48.76	25.72	9.62

Table 10.M21 : 80% voice : 20% data for 64, 144, 384 kbps (AAL2; PPPmux/HDLC; LIPE)

	Capacity		99.9% delay (ms)		Utilisation (%)	
	# voice users	# data users	Voice	Data	Voice	Data
64 kbps						
AAL2/ATM	14	100	<= 1.9	<= 31.13	10.36	46.83
PPPmux/HDLC	18	120	<= 3.5	<= 43.67	11.87	48.32
LIPE	17	115	<= 3.48	<= 12.98	10.93	46.27
144 kbps						
AAL2/ATM	10	80	<= 4.0	<= 73.87	7.41	39.62
PPPmux/HDLC	14	100	<= 5.2	<= 53.65	9.24	41.62
LIPE	15	105	<= 5.18	<= 84.62	9.64	44.04
384 kbps						
AAL2/ATM	*	*	*	*	*	*
PPPmux/HDLC	*	*	*	*	*	*
LIPE	*	*	*	*	*	*

PS. The curve of 99.9% data frame delay for “384 kbps” does not show knees.

Table 10.M22 : 20% voice : 80% data for 64, 144, 384 kbps (AAL2; PPPmux/HDLC; LIPE)

10.3.8 Conclusions by Motorola

1. Section 3 tentatively concludes the capacity for each scenario, based on the knee of the curves of 99.9% delay of voice traffic and of data traffic. Other criteria should be taken into account, for example, the delay requirements for voice traffic and data traffic from RNC to Node B. We propose 5 ms for voice traffic and 100 ms for data traffic.
2. Subject to the criterion in this paper, PPPmux/HDLC for the voice-only scenario yields 13% improvement over AAL2/ATM in capacity, while LIPE yields 16% over AAL2/ATM in capacity.
3. Other criteria should be used to evaluate the improvements of PPPmux/HDLC and LIPE over AAL2/ATM for the 80% voice : 20% data and 20% voice : 80% data scenarios.

11 RECOMMENDATIONS

Since, on the basis of the simulation results, the IP protocol options appear to be comparable with AAL2/ATM with respect to bandwidth, delay and jitter, IP appears to be a viable option for implementing the TNL in the UTRAN. We recommend that the 3G SDOs undertake further study of IP for this purpose.

A : ANNEX A: 3GPP SPECIFICATIONS

The TS and TR references in this section are those approved at the RAN Plenary #08 held in Dusseldorf, Germany, from 21st to 23rd June 2000. All of these specification documents can be found on <http://www.3gpp.org/ftp/Specs/2000-06/> or <ftp://ftp.3gpp.org/Specs/2000-06/>

TS = Technical Specification

TR = Technical Report

A.1 Terminology in 3GPP and UTRAN

TR 21.905 v3.3.0, R-99, SA WG1, 3G Vocabulary

TR 25.990 v3.0.0, R-99, RAN WG4, Vocabulary for UTRAN

A.2 Radio Layer 1 Specifications (RAN WG1)

TS 25.201 v3.1.0, R-99, RAN WG1, Physical layer General Description

TS 25.211 v3.3.0, R-99, RAN WG1, Physical channels and mapping of transport channels (FDD)

TS 25.212 v3.3.0, R-99, RAN WG1, Multiplexing and channel coding (FDD)

TS 25.213 v3.3.0, R-99, RAN WG1, Spreading and modulation (FDD)

TS 25.214 v3.3.0, R-99, RAN WG1, Physical layer procedures (FDD)

TS 25.215 v3.3.0, R-99, RAN WG1, Physical layer; Measurements (FDD)

TS 25.221 v3.3.0, R-99, RAN WG1, Physical channels and mapping of transport channels (TDD)

TS 25.222 v3.3.0, R-99, RAN WG1, Multiplexing and channel coding (TDD)

TS 25.223 v3.3.0, R-99, RAN WG1, Spreading and modulation (TDD)

TS 25.224 v3.3.0, R-99, RAN WG1, Physical layer procedures (TDD)

TS 25.225 v3.3.0, R-99, RAN WG1, Physical layer; Measurements (TDD)

A.3 Radio Layer 2 and Layer 3 Radio Resource (RAN WG2)

TS 25.301 v3.5.0, R-99, RAN WG2, Radio Interface Protocol Architecture

TS 25.302 v3.5.0, R-99, RAN WG2, Services provided by the physical layer

TS 25.303 v3.4.0, R-99, RAN WG2, UE functions and inter-layer procedures in connected mode

TS 25.304 v3.3.0, R-99, RAN WG2, UE Procedures Idle Mode and Cell Reselection Connected Mode

TS 25.305 v3.2.0, R-99, RAN WG2, Stage 2 Functional Specification of Location Services (LCS)

TS 25.321 v3.4.0, R-99, RAN WG2, Medium Access Control (MAC) Protocol Specification

TS 25.322 v3.3.0, R-99, RAN WG2, Radio Link Control (RLC) Protocol Specification

TS 25.323 v3.2.0, R-99, RAN WG2, Packet Data Convergence Protocol (PDCP) protocol

TS 25.324 v3.1.0, R-99, RAN WG2, Radio Interface for Broadcast/Multicast Services

TS 25.331 v3.3.0, R-99, RAN WG2, Radio Resource Control (RRC) Protocol Specification

A.4 UTRAN Architecture (RAN WG3)

TS 25.401 v3.3.0, R-99, RAN WG3, UTRAN Overall Description

TS 25.402 v3.2.0, R-99, RAN WG3, Synchronisation in UTRAN Stage 2

A.5 Iu Interface and Transport Specifications (RAN WG3)

TS 25.410 v3.2.0, R-99, RAN WG3, UTRAN Iu Interface General Aspects and Principles

TS 25.411 v3.2.0, R-99, RAN WG3, UTRAN Iu interface Layer 1

TS 25.412 v3.4.0, R-99, RAN WG3, UTRAN Iu interface signalling transport

TS 25.413 v3.2.0, R-99, RAN WG3, UTRAN Iu interface RANAP signalling

TS 25.414 v3.4.0, R-99, RAN WG3, UTRAN Iu interface data transport & transport signalling

TS 25.415 v3.3.0, R-99, RAN WG3, UTRAN Iu interface user plane protocols

TS 25.419 v3.1.0, R-99, RAN WG3, UTRAN Iu Cell broadcast between SMS-CBC and RNC

A.6 Iur Interface and Transport Specifications (RAN WG3)

TS 25.420 v3.1.0, R-99, RAN WG3, UTRAN Iur Interface: General Aspects and Principles

TS 25.421 v3.0.0, R-99, RAN WG3, UTRAN Iur interface Layer 1

TS 25.422 v3.4.0, R-99, RAN WG3, UTRAN Iur interface signalling transport

TS 25.423 v3.2.0, R-99, RAN WG3, UTRAN Iur interface RNSAP signalling

TS 25.424 v3.3.0, R-99, RAN WG3, UTRAN Iur data transport & transport signalling for CCH data

TS 25.425 v3.2.0, R-99, RAN WG3, UTRAN Iur interface user plane protocols for CCH data streams

A.7 Iub Interface and Transport Specifications (RAN WG3)

TS 25.430 v3.2.0, R-99, RAN WG3, UTRAN Iub Interface: General Aspects and Principles

TS 25.431 v3.0.0, R-99, RAN WG3, UTRAN Iub interface Layer 1

TS 25.432 v3.1.0, R-99, RAN WG3, UTRAN Iub interface signalling transport

TS 25.433 v3.2.0, R-99, RAN WG3, UTRAN Iub interface NBAP signalling

TS 25.434 v3.2.0, R-99, RAN WG3, UTRAN Iub data transport & transport signalling for CCH data

TS 25.435 v3.3.0, R-99, RAN WG3, UTRAN Iub interface user plane protocols for CCH data streams

A.8 Iur and Iub Transport Specifications (RAN WG3)

TS 25.426 v3.3.0, R-99, RAN WG3, UTRAN Iur / Iub data transport & transport signalling DCH data

TS 25.427 v3.3.0, R-99, RAN WG3, UTRAN Iur / Iub interface user plane protocols for DCH data

A.9 RAN Related Technical Reports (RAN WG1, WG2, WG3 and WG4)

TR 25.832 v3.0.0, R-99, RAN WG3, Manifestations of Handover and SRNS relocation

TR 25.922 v3.2.0, R-99, RAN WG2, Radio Resource Management Strategies

TR 25.924 v1.0.0, R-00, RAN WG2, Opportunity Driven Multiple Access (ODMA)
TR 25.925 v3.1.0, R-99, RAN WG2, Radio Interface for Broadcast/Multicast Services
TR 25.926 v3.1.0, R-99, RAN WG2, UE Radio Access capabilities definition
TR 25.928 v1.0.0, R-00, RAN WG1, 1,28Mcps UTRA TDD Physical Layer
TR 25.931 v2.0.0, R-99, RAN WG3, UTRAN Functions, examples on signalling procedures
TR 25.932 v1.0.0, R-00, RAN WG3, Delay budget within the access stratum
TR 25.933 v0.0.0, R-00, RAN WG3, IP transport in the UTRAN
TR 25.934 v0.0.0, R-00, RAN WG3, AAL2 QoS optimisation
TR 25.950 v0.0.0, R-00, RAN WG2, UTRA high speed downlink packet access

A.10 Other Related 3GPP Specifications

TS 23.107 v3.3.0, R-99, SA WG2, Quality of Service Concept and Architecture
TS 26.701 v3.0.1, R-99, SA WG4, AMR speech Codec Generation Description
TS 26.101 v3.1.0, R-99, SA WG4, AMR speech Codec Frame Structure
TS 26.102 v3.1.0, R-99, SA WG4, AMR speech Codec interface to Iu and Uu

B : ANNEX B: 3GPP2 SPECIFICATIONS

Only the top three most important references to 3GPP2 specifications are listed here. These deal with the network reference architecture and the IS-634 MSC to BSC A Interface specification. For more information about specifications and a detailed introduction to the A Interface, see "CDMA Internetworking: Deploying the Open A-Interface," by Su-Lin Low and Ron Schneider.

CDG-IOS Version 3.1.1, CDMA Development Group MSC to BS Interface Inter-Operability Specification. June 1999.

TIA/EIA/IS-634-A, MSC-BS Interface (A-Interface) for Public 800 MHz. July, 1998.

TIA/PN-2716, TR-45 Wireless Network Reference Model, October, 1997.

C : ANNEX C: PARETO DISTRIBUTION

Pareto pdf

The Pareto probability density function is as follows with parameters set as defined in Table 9.3.

Probability density function, $f(x) = \frac{a \cdot k^a}{x^{a+1}}$, for $x \geq k$ while $f(x) = 0$, for $x \leq k$

Cumulative distribution function, $F(x) = 1 - \left(\frac{k}{x}\right)^a$, for $x \geq k$ while $F(x) = 0$, for $x \leq k$

Mean value of the distribution, $m = \frac{k \cdot a}{a - 1}$; If the mean value is given then, $a = \frac{m}{m - k}$

Hence k defines the minimum value of x (e.g. minimum packet size) while μ defines the mean value of the distribution (e.g. mean packet size).

If the distribution is limited to a maximum value of 'm', i.e. $x = \text{minimum}(\text{Pareto}, m)$ then the mean

value of this new (limited) distribution is reduced and is given by $m_m = \left(1 - \frac{1}{a} \left(\frac{k}{m}\right)^{a-1}\right) \cdot m$

Motorola/Lucent Data Model

- For the Motorola / Lucent Data model each web browsing download has Pareto distributed file size with a parameter $\alpha = 1.1$, mean 12 kbytes, minimal file size 1858 bytes, maximal file size 5 Mbytes. The p. d. f. (probability density function) is

$$f(x) = \begin{cases} \frac{a \cdot k^a}{x^{a+1}}, & k \leq x \leq m \\ \frac{k^a}{m^a}, & x < m \end{cases}, \text{ where } a=1.1, k=1858, \text{ and } m=5,000,000$$

- Chop the file into IP packets with size of 1500 bytes (and one less than 1500 bytes if the size is not a multiple of 1500).
- Inter-arrival time of IP packets is exponential distributed with mean of 8.3 ms. This yields about 1445.78 kbps IP packet arrival rate (less than 64, 144, 384 kbps data transmission rates). Therefore, the inter-arrival time has no significant impact on simulation results.
- Reading time is defined by the time that the last bit of a file leaves from the RNC to the time that the first bit of the next file arrives to the RLC data buffer. The distribution of reading time is exponential with mean 12 sec.

D : ANNEX D: STYLE LIST

Abbreviation	used for abbreviations (Chapter 3)
Document title	reserved for document title on front page
editor's note	used for editor's notes (in red italics)
Figure caption	for figure captions – underneath figure
Figure	for figures
First page header	header style used on first page only
Footer	footer to include date, MWIF CONFIDENTIAL and page number
Footnote reference	used for footnotes
Footnote text	
Header	header to include MWIF technical report reference number
Heading 1 – Heading 9	no more than four levels of header recommended
Hyperlink	
List 2	free format list - second level
List bullet 2	bulleted list – second level
List bullet	bulleted list - first level
List continue 2	list continuation paragraph – second level
List continue	list continuation paragraph – first level
List numbered 2	numbered list– second level
List numbered	numbered list– first level
List	free format list - first level
MWIF	reserved for MWIF name on front page
Normal	style for standard paragraphs
Notice	reserved for IPR notice on front page
Page number	
Reference	used for references (Chapter 2)
Table caption	for table captions – underneath table
Table of Figures	
TOC base, TOC1 – TOC9	Table of contents use only

DOCUMENT HISTORY

Date	Version	Comment
16 May 2000	Vx.x.x Draft	First draft proposal for discussion at IPinRAN group in MWIF
17 May 2000	Vx.x.x Draft	Updated contents list, companies names added to provide contributions
18 May 2000	V0.0.0 Draft	Updated contents list with draft scenarios for evaluation added to Chapter 8
12 July 2000	V0.0.0 Draft	Editorial update only to align with the Technical Report format suggested by MWIF (in mwif2000.165)
21 July 2000	V0.0.1 Draft	Update to most of the chapters in this report. No input provided for Chapters 6 or 8
03 August 2000	V0.0.2 Draft	Inclusion of changes discussed at MWIF meeting in Toronto (July 2000) including voice and data models.
11 August 2000	V0.1.0	An attempt to confirm all the changes discussed in the MWIF meeting in Toronto (July 2000).
12 September 2000	V0.1.1	Chapter 2 new references added. Chapter 3 abbreviations and glossary of terms updated. Chapter 7 sections added on IP fragmentation updated, multicast routing, tunnelling, and availability. Chapter 8 sections added on User Plane Protocol Stack Descriptions, Composite IP and Lightweight IP Encapsulation. Chapter 9 simulation models updated hopefully as agreed in audio conferences. General editorial changes throughout report.
4 th October 2000	V0.1.2	Editorial correction throughout document. References renumbered. Changes to 7.2.5 (Size of Frame Protocol Payloads to transport), 7.3 (Last Mile QoS Issues), 6.8 (Security), 6.9 (Addressing), 7.1 (QoS Differentiation), 7.2 (Link Efficiency), 9.3 (Motorola/Lucent data model added), 7.2.3 (KWHC header compression added), 10 (Performance new section added), 10.1 (Alcatel Simulation Results), 10.2 (Lucent Simulation Results), 10.3 (Motorola Simulation Results).
5 th October 2000	V0.1.3	Some editorial changes. Added 7.9.2 (Protocol Header and Payload Formats), updated 10.2 (Lucent Simulation Results), updated 10.3 (Motorola Simulation Results).
14 th October 2000	V0.2.0	Implementation of changes as discussed in MWIF#07 in Berlin. This is a temporary document as other changes still need to be included in this version 0.2.0.
23 rd October 2000	V0.2.1	Implementation of changes and corrections from Alcatel, Toshiba, Cisco, Motorola, Siemens, Lucent and Nortel.
1 st November 2000	V0.2.2	Implementation of changes and corrections from Cisco, Alcatel, Motorola, Sun and Siemens.

10 th November 2000	V0.3.0	Editorial changes, text added to Conclusions and Recommendations chapters. Submitted to working group for review prior to TC balloting.
8 th December 2000	V0.3.1	Insertion of comments from working group review. Submitted to TC for balloting.
12 th March 2001	V2.0.0	Added comments received after cutoff date for Version 1, submitted to TC for Review. Added Section 10.4 with Nortel simulation results. Removed Chapter 12, since this is the last version of the document.
4 th June 2001	V2.0.0	Document ballot passes with quorum by MWIF Membership.
18 th June 2001	V2.0.0	Document formally ratified by the MWIF Board of Directors