

Transmission Phase in 3G, using ATM

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**KTH Information and
Communication Technology**

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Abstract

Nowadays a very important aspect of telecommunication is quality of services. 3G networks offer all of the customers' best quality for each type of information including voice/video and data transmission. One of the vast discussion in this area is increasing the throughput and prevent the congestion in rush traffic hours in the network. Congestion occurs when transfer rate in the network is lower than requested rate by application. Congestion leads to cell loss and dropped cells should be retransferred to recover the data which is double job and affect the throughput and even can affect the quality of the services. Time sensitive information (voice/video) requires no data loss and they employ Forward Error Correction (FEC) codes to recover the data. The number of the FEC codes should be kept small to prevent the overload in the network. In this paper I will show that just simple FEC is not enough when network interworks with Asynchronous Transfer Mode (ATM). A powerful buffer management gives higher throughput and in that condition block loss rates reduces. In addition, effective utilization of IuB interfaces that link the Radio Network Controller (RNC) and Base Station (BS) has another effect on throughput. Different service categories are used to transform different type of information. In this paper I have also introduced all kind of information types offered by 3G networks and further analyzed the weakness of the existing transmission phase in 3G networks.

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1. Introduction

Third generation mobile networks offer users a wide range of services and has a greater network capacity in compare with second generation. 3Gs offered services include voice, video and data communication in mobile environment. One of the significant changes in 3G mobile networks in compare with the 2G is entering the asynchronous transfer mode (ATM) which is a packet switching and multiplexing protocol that encodes data in small packet size. This small packet size transports all information types (Voice, Video, data). Traffic descriptor in 3G network is a set of transmission parameters between user and network that carries information in ATM cells. To enhance the best capacity utilization in traffic descriptor, various services have been allocated for each information type.

The master thesis was conducted at the department of 3G planning and optimization Nokia Siemens network (NSN), Riyadh in Saudi Arabia from 26th October 2008 to March 2009. This master thesis was given by the British company Advance wireless technology group (AWTG) which has a 5 years contract with NSN.

AWTG is a leading company in wireless operations which has a close contact with Kings College located in London. AWTG is able to offer its customers a wide range of network services from business planning through to network management.

Transmission phase in 3G networks is the part of the planning team in AWTG and it includes the traffic descriptors behavior from base station to radio network controller and further, design and optimize the network capacity based on the traffic. This thesis project is about the architecture of 3G transmission phases in terms of increasing the throughput and analyzing the weaknesses of the existing 3G traffic management.

1.1. Background

AWTG engineers team plan and optimize the 3G Networks from radio frequency (RF) to transmission phase. Transmission phase includes design of the base station and their load based on the number of users and their allocated services. 3G networks invite the users to voice/video and internet connection at the same time. It means that the networks load and capacity should be able to maintain these services with the optimal quality of service (QoS). The number of users, the peak rate of the load in rush hours and the geometry of the area are important parameters for engineers to design the network. Based on these information transmission team designs the traffic descriptors and allocate the different services to each base station and their links to Radio network controller (RNC). We can measure the traffic load in each base station

and eventually expand the capacity of the links in the case of higher traffic load. But these kinds of expansions are costly and may need more equipment. In the other hand if we can optimize the traffic descriptor and decrease the congestion in carriers

by new traffic management methods then we can get the better throughput and increase the QoS in the network.

In the 3G network congestion is very likely to happen especially in high traffic hours. Congestion leads to buffer overflow in ATM switch which reduces the throughput. When ATM cell loss starts to happen more end-to-end packets become useless. Loss data are retransmitted which is not effective because of the round trip time especially for real time transport protocols. When retransmission is not cost effective, forward error correction (FEC) takes into use. In this protocol number of cell groups into packets and number of packets groups into blocks and an extra FEC packet would be added to each block. In this scenario each lost data packet can be recovered by FEC codes. The numbers of FEC codes kept small to prevent the overload in the network.

In this thesis I will show that even with FEC recovery schemes, random dropping of the ATM cell, due to the congestion can decrease the throughput and I propose a method to trace the cell-loss in packets and with help of buffer manager in ATM switch can increase the throughput.

1.2. Aims

Users should always be able to have access to all available network bandwidth when they need it, while being guaranteed that the chance of losing data in congestion is negligible. At the same time QoS should be considered as well. This thesis project discusses these issues and describes congestion control for giving the better throughput in the network. Carriers in transmission network play a very important role in controlling the congestion and increasing the throughput. Link between Radio Network controller and Base station (Iub) is defined to split the bandwidth to several services and transmission optimizing in this relation means how to allocate the services in Iub to get the highest network utilization. In addition, the weakness of the traffic management in transmission phase affects the throughput. Combination of different control policies must be used to enhance the QoS and increase the throughput. Analysing the mechanisms to control end-to-end packet-loss rates of real time applications over ATM is the focus of this project. This could lead to throughput increasing in different networks using ATM.

1.3. An overview of this project

The report has started with an introduction to present the underlying reason to why this master thesis was launched. Chapter 2 will presents the transmission architecture of 3G networks including different equipments used in 3G transmission

phase. This chapter is essential in the way that it gives the reader the theoretical basics needed further in this report. Chapter 3 presents digital data transmission

technology (ATM), a method within 3G transmission, which is the biggest distinction between 2G and 3G in transmission relation. Chapter 3 explains the definition and capabilities of ATM, and further on, where information is encapsulated into ATM cells. Chapter 4 presents the traffic descriptor which is the traffic parameters that deal between the user and network. In this chapter, the reader will find out the main link between base station and radio network controller and different services assigned in this link, which is the focus of this thesis. Subchapter 4.4 analyze the weakness of the traffic descriptor in 3G and point out the drawbacks of the services in existing 3G network and respectively traffic descriptors' architecture. Subchapter 4.4.3 presents the analysis of the solution, based on research and practical experiences. After this, chapter 5 will introduce the configuration in traffic descriptor based on the data base that planning engineers use in Nokia Siemens networks. Further, the functionality of the ATM switch relating buffer management and flow control are introduced. Subchapter 5.4 presents the main focus of this thesis project, which is the weakness analysis of the traffic control in existing 3G networks. It contains the drawbacks of the buffer management techniques that are used in transmission phase in 3G networks. After this, chapter 6 presents the analysis of the technique that can give higher throughput and ATM loss rate reduction with help of new policies in ATM buffer management. Finally, chapter 7 summarizes the total work, studies and conclusions, and presents suggestions to future works. The contribution of this thesis project is introduced in subchapter 4.4.3 and chapter 6.

1.4. List of acronyms

Title	Notation
3G	Third generation
ATM	Asynchronous transfer mode
NSN	Nokia Siemens network
RF	Radio frequency
QoS	Quality of service
RNC	Radio network controller
ARQ	Automatic repeat request
FEC	Forward error correction
WBTS	Wide base transceiver station
CN	Core network
UE	User equipment
BS	Base station
RAN	Radio access network
NMS	Network management system
GSM	Global system for mobile
TX/RX	Transmitter/receiver
O&M	Operation and maintenance

IFU	Interface unit
PDH	Plesiochronous digital hierarchy
SDH	Synchronous digital hierarchy
RRM	Remote releasing unit
CBR	Constant bit rate
VBR	Variable bit rate
ISDN	Integrated services digital network
TCP	Transmission control protocol
IP	Internet protocol
VP	Virtual path
VC	Virtual circuit
VBR-NRT	Variable bit rate-non real time
VBR-RT	Variable bit rate-Real time
PCR	Peak cell rate
SCR	Sustained cell rate
MBS	Maximum burst size
ABR	Available bit rate
MCR	Minimum cell rate
PDU	Protocol data unit
IMA	Inverse
LCR	Link cell rate
CEPT	Conference of postal and telecommunication
TDM	Time-division multiplexing
STM	Synchronous transport module
SAP	Service access point
CDVT	Cell delay variation tolerance
MDCR	Minimum defined cell rate
CPS	Cell per second
CoCo	Connection and configuration
CAC	Cell admission control
UPC	Usage parameter control
UNI	User network interface
NNI	Network to network interface
DCN	Dynamic circuit network
MOC	Measure of congestion
IWU	Interworking unit
RM	Resource management
CLP	Cell loss priority
BLF	Block loss flag
PLF	Packet loss flag
C-NBAP	Common Node B Application Part
D-NBAP	Dedicated Node B Application Part

2. Transmission architecture in 3G

Transmission network enclose the interfaces between wide base transceiver station (WBTS) and radio network controller (RNC), the RNC and core network (CN), and between RNCs, Iub, Iu, and Iur respectively. The simplified block diagram of 3G network architecture is illustrated in figure 1. The architecture consists of user equipments (UE), a set of base stations (BS), radio access network (RAN), CN and network management system (NMS).

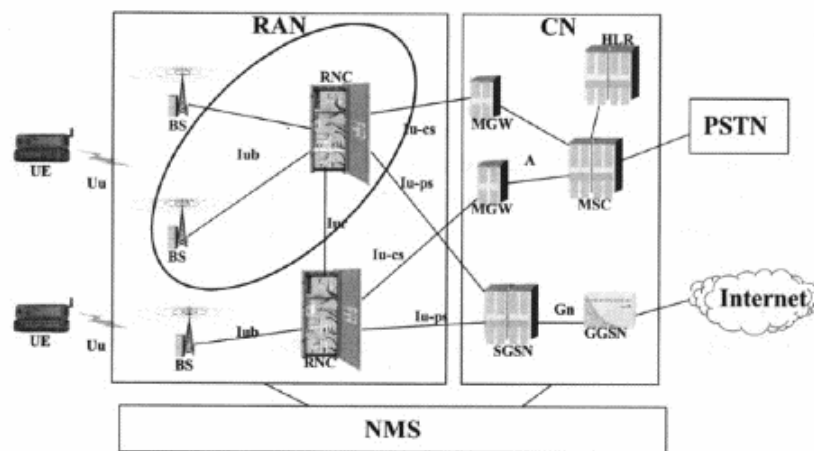


Figure1: 3G Network architecture [1]

Fundamentally, the transmission planning in 3G is mostly the same as for GSM networks, but there is one important addition: The inclusion of asynchronous transfer mode (ATM) technology that occurs in the BS and Iub interface. The dimensioning of the 3G transmission network is different compare to GSM and in detailed planning there is ATM parameter setting to consider.

2.1. Base station (Node B)

Node B is the equipment that facilitates wireless communication between user equipment (UE) and a network. Its function is handover channel management, base-band conversion, channel encoding and decoding and interfacing to other networks. Figure 2 illustrates a three-sector 3G base station. The block diagram consists of a set of antennas, transceivers and signal processing modules. In more details we have:

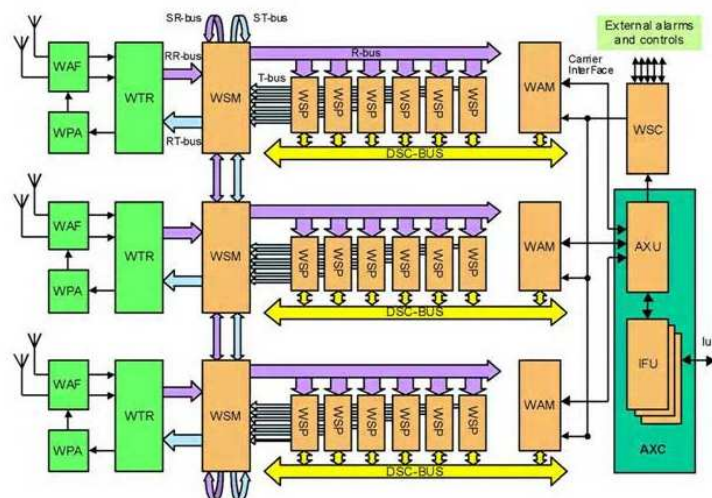


Figure2: Three sector (1+1+1) 3G-BTS [2]

WAF: Antenna filter. Combines and isolates TX/RX signals and amplifies received signals

WPA: Power Amplifier. A multi carrier amplifier with an operating bandwidth of any 20MHz of whole 60 MHz WCDMA allocation.

WTR: WCDMA transmitter and receiver unit consists of a transmitter and two receivers.

WSM: summing and multiplexing unit. Sums TX signals from signal processing units or other WSMs.

WSP: signal processing unit. Performs RX and TX code channel processing, coding and decoding functions.

WAM: application manager. Performs O&M functions and carrier control.

IFU: Transmission interface unit to connect the BS to the network.

AXU: ATM cross connect unit. Connecting element between WAM and IFUs. It can handle ATM cross connection at a transmission network layer.

WSC: System clock. It performs synchronization functions and reference clock generation to the WCDMA BS.

2.2. Wireless access manager (WAM) dimensioning:

There are no logical relationship between a wireless access manager (WAM) and a TRX or between a WAM and a cell but there are general rules that should be noticed.

First is to allocate one WAM per TRX and/or to allocate one WAM per 3WSPs. In practice if the capacity of the base station is low it should be kept in mind that there is no logical limit how many TRXs one WAM can handle.

2.3. Radio network controller (RNC):

Radio network controller is the heart of the whole 3G network and in other words all of the decisions in the network are done by RNC. In general, the whole network area is divided into regions each handled by a single RNC. Each RNC is connected to one or more Node Bs. The RNC's responsibility is to control the entire node B connected to it and management of the radio channel (Uu) [3]. A simplified block diagram of the RNC is illustrated in figure 3.

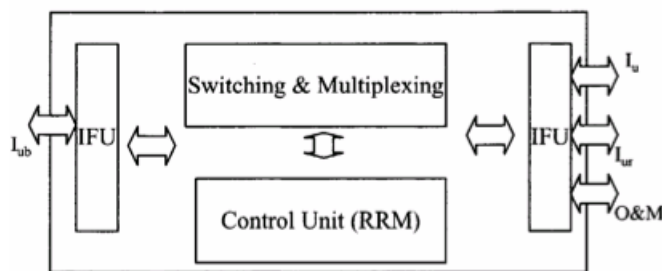


Figure 3: Block diagram of radio network controller (RNC)

As we can see in Figure 3 different interfaces are connected to the RNC. We should notice that the RNC interfaces can accept only pure ATM cells so all of the carriers connected to an RNC should be de-multiplexed to ATM cells before they enter to RNC. There are mainly two kinds of interfaces unit (IFU), plesiochronous digital hierarchy (PDH) and synchronous digital hierarchy (SDH) [3].

Control management includes among others load control, admission and handover control which is done by the remote releasing unit (RRM) Control Unit. Switching unit provides the required support for the ATM traffic, AAL2 switching and multiplexing of traffic. In addition, we should notice that the number of RNCs in the network depends on the number of users or based upon the I_{ub} bandwidth requirements [3].

3. Asynchronous Transfer Mode in WCDMA

ATM is a switching and multiplexing mechanism operating over a fiber based physical network such as SONET. It uses a cell (53 byte packet with 5 byte header) as its basic switching element and all information types (Voice, Data, Video) are transported inside the cell. The biggest advantage of ATM is in its ability to do multiplexing and thus effectively can handle bursty, variable bit rate (VBR) and CBR (constant bit rate) traffic types. It is primarily a connection-oriented technology using a combination of virtual circuits and virtual paths to establish an end-to-end connection. An ATM cell is illustrated in Figure 4 which consists of a header and payload.



Figure 4: ATM cell

3.1. Definition of ATM

Asynchronous transfer mode (ATM) is a technology that has its history in the development of broadband ISDN in the 1970 and 1980. From the technical point of view it can be seen as an evolution of packet switching. Similar to packet switching for transmission control protocol[TCP]/internet protocol[IP], ATM integrates the multiplexing and switching functions, is well suited for bursty traffic (in contrast to circuit switching), and allows communications between different devices which operate with different speeds. Unlike packet switching, ATM is designed for high-performance multimedia networking. ATM is also a capability that can be offered as an end-user service by service providers or as networking infrastructure for these and other services. The most basic service is ATM virtual circuit, which is an end-to-end connection that has defined end points and routes but has no bandwidth dedicated to it. Bandwidth is allocated on demand by the network as users have traffic to transmit. ATM also defines various classes of services to meet a broad range of application needs.

3.2. ATM capabilities

Developing end user services- voice, data or multimedia- require flexible transmission in terms of variable bit rates, real time transmission, and efficient usage of transmission capacity. ATM has been chosen by the international standardisation bodies as the transmission technology for the 3G networks because it can fulfil demands as well as the related requirements on quality of services.

3.2.1. ATM cross connect unit (AXU)

The ATM cross-connect unit (AXU) is the master unit which controls the AXC nodes. AXU performs the main functionality for communication within the base station, as well as for the connection to other network elements. The ATM switch fabric of the AXU enables flexible cross-connections simultaneously on both VP and VC level [4].

3.2.2. Virtual channel connections between BTS and RNC

AXC connects at least five virtual channels (VC) in the Iub interface between the base station and radio network controller. Four VCs are required for user plane which is the function that deal with the user-to-user information transfer and associated controls such as flow control and error control mechanism and control plane connections between the BTS (WAM) and the RNC. The fifth VC connection is required for operation and maintenance traffic (O&M). [Figure5]

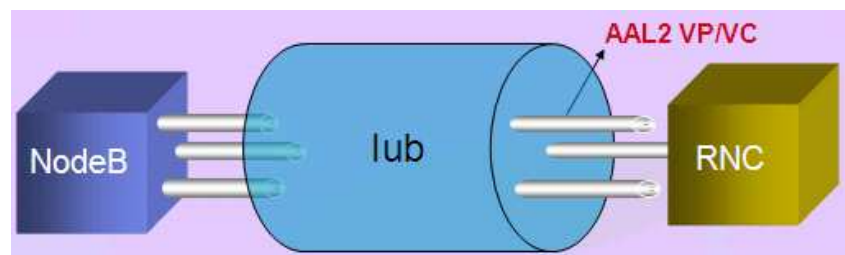


Figure 5: Virtual channels in the Iub interface [18]

3.3. ATM service categories

- **Constant bit rate (CBR)**

Constant bit rate services category has been defined for connections that continuously require a constant bit rate. The bandwidth is further determined by the peak cell rate (PCR). For CBR services, there is a defined quality of service guaranteed; in the condition that cell rate is not exceeded. CBR connections can transport data at peak cell rate without having an impact on the quality of service. CBR supports real time applications that have strong requirements for cell transfer delay and cell delay variation. If maximum cell delay variation is exceeded then ATM cells are discarded. Typical applications are voice traffic and Video-Conference [5].

- **Unspecified Bit Rate (UBR)**

UBR has been chosen for services that have no real time requirements. Because quality of service is not guaranteed, the connection has to be secured on higher layers. A typical application is data transfer or management data [5].

- **Variable bit rate-None real time (VBR-NRT)**

This class allows users to send traffic at the rate that varies with time depending on the availability of user information, statistical multiplexing is provided to make optimum use of network resources. The delivery guarantee is required by this service. As example of applications using this service we can name banking and credit card processing, airline reservation [5].

- **Variable bit rate-Real time (VBR-RT)**

This class is similar to VBR-NRT but is designed for applications that are sensitive to cell-delay variation. This service applies for the services that cannot tolerate lengthy delay. For instance, compressed video or voice applications and multimedia. When you have this type of traffic service you have peak cell rate (PCR), Sustained cell rate (SCR) and Maximum burst size (MBS) [5].

- **Available bit rate (ABR)**

This class of ATM services provides rate-based flow control and is aimed at data traffic such a file transfer and e-mail. Although the standard does not require the cell transfer delay and cell-loss ratio to be guaranteed or minimized, it is desirable for switches to minimize delay and loss as much as possible. ABR uses minimum cell rate (MCR) to control the traffic. The network continuously allocates the MCR to hold the connection alive and at the same time all other traffic runs across the network in a priority over ABR [5].

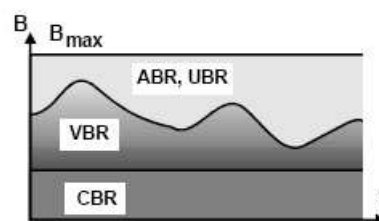


Figure 6: ATM categories of services [6]

3.4. ATM adaption layer (AAL)

The job of AAL is to map the higher level data packets into ATM cells. As you can see in Figure 7, the highest layer is AAL which performs segmentation and reassembly of packets.

For UMTS, the main ATM adaption layers are:

- AAL5: used for control plane signalling.

- AAL2: used for user data transfer [5][6].

ATM layer

- Common flow control.
- Cell switching (based on VCI/VPI).
- Cell multiplexing, De-multiplexing.[5][6]

Physical layer

- TC: Cell rate de-coupling; generate/verify transmission frame adaption.
- PMD: Bit timing; physical medium.[5][6]

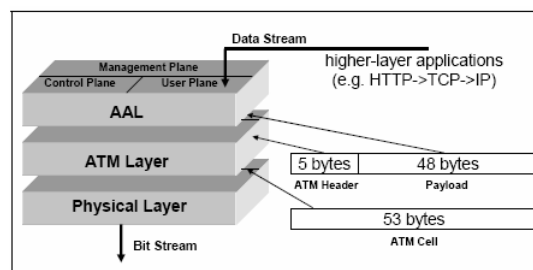


Figure7: The reference model used for ATM [6]

ATM adaption layer segment data stream into protocol data unit (PDU), encapsulate the PDU (header or/and trailer), segment the encapsulated PDU into service data unit and at the end encapsulate the segments into ATM cells or 48 data payload as illustrated in Figure 8 [5][6].

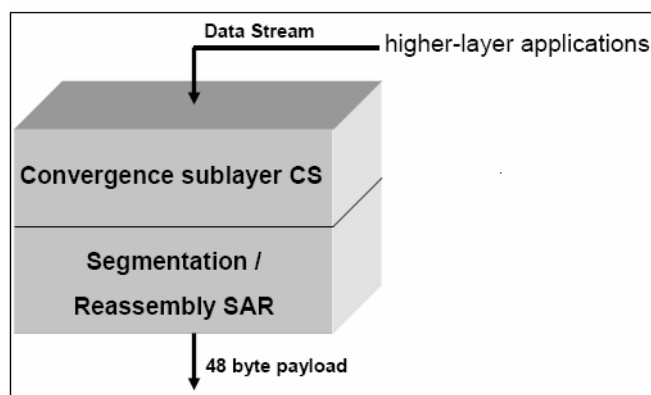


Figure8: ATM adaption layer (AAL) [6]

3.5. Inverse multiplexing for ATM (IMA)

Inverse multiplexing for ATM enables efficient transport for broadband in existing transmission network. This technique allows combining several physical links into one logical link (IMA group) and saves capacity by enabling the division of high bandwidth ATM data stream into several lower bit rate transmission links. In IMA terminology, the individual physical circuits are called links and resulting aggregated cluster of circuits is called an IMA group. An IMA group can contain up to 32 links, however in practice it is limited to eight or fewer links. All of the ATM cells are transmitted in a round-robin fashion over all of the physical links in the IMA group. We can see in Figure 9 the first cell of each frame will always be transported over the first physical link in the group. The second cell sent via second physical link and so on the transmitting end of an IMA link aligns the transmission of IMA frames on all physical links. This allows the receiving end to adjust to differential link delays for each of the links. This ensures that the receiving end of the IMA link can recreate the original ATM cell stream and pass it back to the ATM layer. The maximum differential link delay that IMA engine can tolerate is 25 micro second. The majority of IMA links are based on E1 and T1 circuits. All of the links in IMA group required to operate at the same link cell rate (LCR), otherwise the cell may arrive at their destination out of order [4] [7].

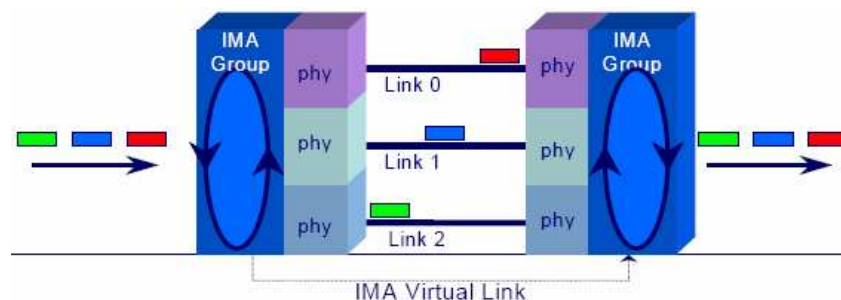


Figure 9: Inverse multiplexing/De-multiplexing ATM cells in an IMA group [7]

Without IMA feature no VPC can have a bandwidth larger than what an interface in corresponding IFU (ATM switch interface) card can support.

VPC size < 2Mbps for IFU [4]

The sum of VPCs cannot exceed what an interface in corresponding IFU card can support. For example without IMA 3 VPCs- each 1.2 Mbps-has to be put into their own 2 Mbps (E1) lines. With IMA we can put those 3 VPCs to two Mbps lines.
($3 \times 1.2 \text{ Mbps} / 2 \text{ Mbps} = 1.8$)

3.6. ATM over different carriers

In digital communication, a single wire pair can be used to transfer many applications simultaneously. Depending on the needed bandwidth hierarchy level of the carrier could be different. E0 is the first carrier level and all of the higher levels are based on E0 and use multiplexing to reach the other levels. Rate of the E0 is 64kbit/s and with multiplexing we can transfer higher level E1, E2 and STM-1. The PDH hierarchy levels are shown in Figure 10 [8].

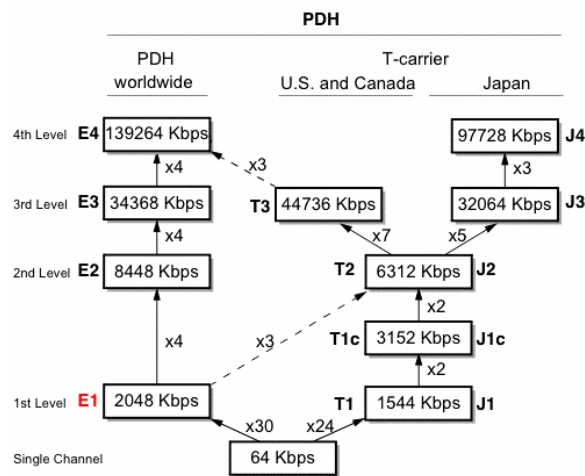


Figure 10: PDH hierarchy levels [8]

3.6.1. ATM over E1 Carrier/PDH

In Europe E1 carrier was developed and standardized by European Conference of postal and telecommunication Administration (CEPT). E1 operates at 2 Mbps over coaxial cable. E1 transmission link contains 32 channel slots (0-31), each has a rate of 64kbps or in another word 32 E0s. With help of TDM multiplexing we can have up to 30 telephone channels of 64kbps into one 2Mbps signal with the format called E1. E1frame has 32 time slots of 8bits. Each slot has sampling rate of 8000 bit/s or $1/8000=0.000125$ second for transmitting one bit [9] [10].

$$32 \times 8000 \times 8 = 2048000 \text{ bit/s} = 2.48 \text{ Mbps}$$

Time slot 0 (TS0) is reserved for framing purposes. It allows the receiver to lock on to the start of each frame and match up each slot in turn. Time slot 16 (TS16) is often reserved for signalling purposes. The rest of the 30 slots are assigned for payload and carry voice /video or data. The 32 ATM time slots for the ATM cell mapping are shown in Figure 11. The receiver on the other hand receives the slots and with help of reserved time slots indicates when the first interval of each frame begins [9] [10].

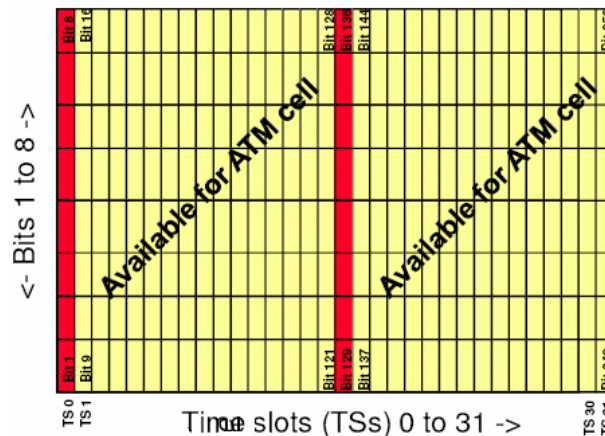


Figure11: ATM cell mapping into PDH 2Mbps [11]

Plesiochronous Digital Hierarchy (PDH) is a technique showing how to map different signals into one signal. All of the ATM cells from different users maps into E1 with help of PDH technology. As we can see in Figure 11 with this technique we allow transmission of data streams that are nominally running at the same rate [9][10][11].

In order to move multiple 2Mb/s data stream from one place to another place, data streams multiplexed (TDM) in different groups. Each group receives the data from specific 2Mb/s data stream and then the other one receives the second 2Mb/s stream and so on. The transmission multiplexer adds additional bits to help the decoder from receiving side to identify which stream belongs to which group. These additional bits are called stuffing or justification bits [9] [10].

3.6.2. ATM over STM-1/SDH

Synchronized Digital Hierarchy (SDH) standard is a replacement for PDH for transmitting larger amount of data traffic and defines the reliable architecture for transporting telecommunication services on a worldwide scale. This standard allows different data streams with different rates to be multiplexed and transferred over optical transport. AS we can see in Figure 12 different VPs can multiplexed over SDH frame one by one and instead of transmitting the header and then payload (Ethernet), part of the overhead being transmitted with part of the payload, then the next part of the overhead and so on [11][12].

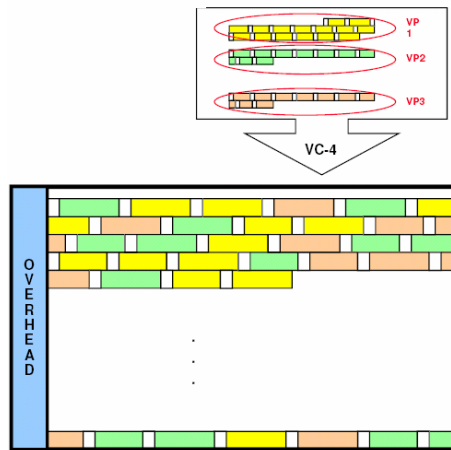


Figure 12 SDH frames [11]

STM-1 is based on SDH and transmits the data stream with the bit rate of 155.52 Mbits/s or almost 77 E1s. Figure 13 shows the STM-1 frame and it contains 9 bytes column and 270 bytes row [11].

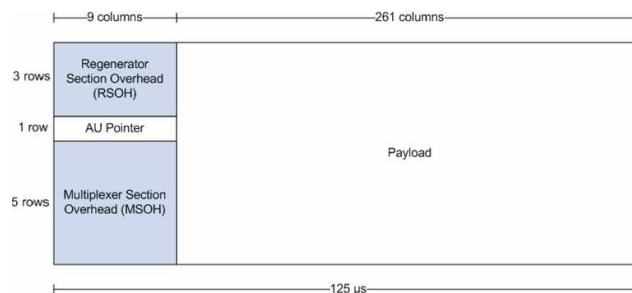


Figure 13: STM-1 Frame [13]

Total content: 9×270 bytes = 2430 bytes

Overhead: $9 \text{ rows} \times 9 \text{ bytes}$

Payload: $9 \text{ rows} \times 261 \text{ bytes}$

Period: 125 micro second

Bitrate: 155.520 Mbits/s ($2430 \times 8 \text{ bits} \times 8000 \text{ frames/s}$)

Payload capacity: 150.336 Mbit/s ($2349 \times 8 \text{ bits} \times 8000 \text{ frame/S}$)

STM-1 framing consists of two parts: the transport overhead and the payload. The payload is the actual data that are multiplexed over whole frame and overhead.

4. Traffic descriptor in 3G

Traffic descriptor is a set of traffic parameters that deal between the user and network, at connection establishment. These descriptions clarify the worst possible value of the parameters related to the requirements of the users. Traffic descriptor parameters are divided into two categories. First one is the traffic parameters that show the specifications of the transfer of traffic in the network. It mostly includes the speed and delay of the traffic. The second one is related to what user defines for connection. The QoS in this relation is the level that user can tolerate for the transfer [15].

4.1. Traffic parameters

Traffic parameters control the transmission in speed and delay relations and describe the traffic characteristics of an ATM connection. The important thing about the ATM traffic parameters is the ability to test that if a connection obeys the value of these parameters [14] [15].

Peak cell rate (PCR): PCR is the inverse of time interval between cells. For instance, if one user define the interval between cells as 10 ns then PCR would be 100 000 000 cells per seconds. This connection parameter is defined at the physical layer service access point (SAP) [14] [15].

Sustained cell rate (CSR): The calculation of the average of the cell rate in long term in one specific connection called CSR. When there is no variation in cell rate (CBR) then CSR and PCR would be the same [14] [15].

Cell delay variation tolerance (CDVT): Due to the multiplexing of connections in ATM layer functions, the cell stream may experience variable delay before entering the network, therefore UPC (usage parameter control) cannot purely trust the PCR and some tolerance considering CDV should be taken in to account [14] [15].

Maximum burst size (MBS): MBS defines the maximum cell rate that could be sent at peak cell rate. It gives an upper bound on the length of the burst transmitted at peak cell rate [14] [15].

4.2. IuB description/design

The physical connection between WBTS and RNC is IuB. Each IuB link is configured as a single virtual path (VP) and several virtual circuits (VC) inside the Virtual path.

4.2.1. IuB virtual circuits and virtual paths

ATM in 3G networks uses virtual circuits and virtual paths.

- A virtual path (VP) is a bundle of virtual circuits (VC)
- Virtual paths are identified by the virtual path Identifier (VPI)
- Virtual circuits are identified by the VPI and Virtual Circuit Identifier (VCI)

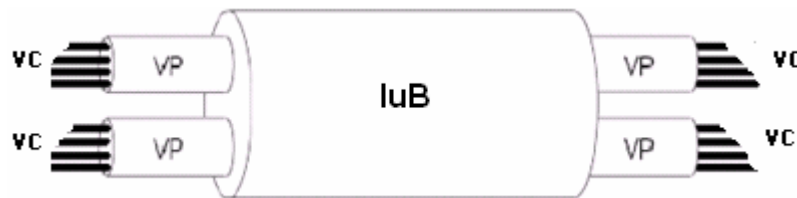


Figure 16: Virtual path & virtual circuits

4.3. Virtual circuit types in each VP

In the BTS, AXC connects five VCs in the IuB interfaces between Radio network controller and base station. The following values are predefined for VPI/VCI.

VPI=0(preconfigured, default) [4].

VCI=30 preconfigured for IP over ATM connection (DCN) [4].

VCI=33 predefined for C-NBAP connection [4].

VCI=34 predefined for D-NBAP connection [4].

VCI=35 predefined for AAL2 signaling [4].

VCI =36 and 37 predefined for AAL2 user data [4].

The following figure shows an example of different VCs in IuB interface [4].

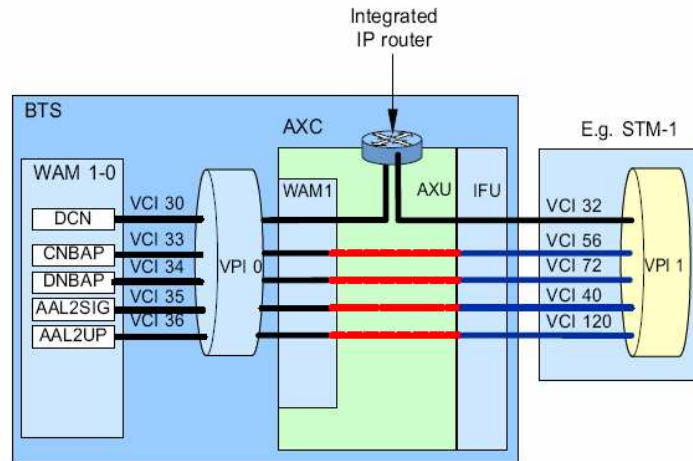


Figure 17: Virtual channel in the IuB interfaces [4]

Within the VP there are a number of Virtual circuits (VCs), which can be divided into 5 main types:

AAL2 user plane-CBR, which carries the actual user data.

AAL2 signaling-CBR, which carries control signaling related to setting up AAL2 connections within the AAL2 user plane VC.

DNBAP-CBR, dedicated Node B application part. This carries messages related to the setting up and releasing of radio links.

CNBAP-CBR, which is common NodeB application part. This carries messages relating to setting up first radio links (SRBs) and RRI messages.

O&M-UBR, operations and maintenance, which is controlling the alarms in RNC and downloading the software between BTS and RNC.

4.4. Analysis of cons and pros of IuB architecture and services in 3G traffic.

IuB services architecture has important effect on the networks behavior and respectively throughput of the network. Transmission optimization in 3G network means how to utilize the IuB capacity in such a way that each of the information type could be transferred with best quality. In section 4.4.1 I point out the weaknesses of the services that are used in existing 3G networks. Further on, in section 4.4.2, the drawbacks of the IuB architecture relating VP and VC allocation are introduced. In section 4.4.3 the solution to the mentioned weaknesses will be analyzed based on research and practical experiences.

4.4.1. Weakness of the allocated services within IuB

As per our discussion about the different services defined within each VC, all of the VCs are running with Constant bit rate (CBR) except O&M which is running with UBR service. UBR doesn't offer any minimum cell rate guarantee and generally is used for the applications that are very tolerant of delay or cell loss, meaning that O&M link could be completely starved if the other links are full of traffic. This would be unacceptable for O&M since it carries alarms which can't be missed or delayed. We can artificially create a minimum cell rate for O&M by making sure the combined cell rates of the other links leave some headroom and don't completely fill the available bandwidth.

4.4.2. Weakness of the IuB architecture relating VP and VC allocation

A very important issue in transmission architecture is how IuB is introduced in AAL upper layer. Basically, in network architecture each IuB is configured as a single VP with couple of VCs (usually five VCs). In this condition, all of the VCs are within one single VP. O&M is one kind of information that carries alarms and is configured in one VC. O&M link should always be connected, due to its important role. Engineers control the communication between RNC and BS with help of O&M link. What happened if VP connection fails and BS lose connection with RNC?

4.4.3. Analysis of the solution

UBR+ is an alternate to UBR and is defined by minimum defined cell rate (MDCR) and peak cell rate (PCR). User traffic can go as high as the peak cell rate (which could be the physical link capacity), if free capacity is available, for a continuous time period. MDCR is guaranteed to support a minimum throughput in case of high IuB load. In addition, if you have multiple UBR+ connections (VC's) in a VP, each UBR+ VC can utilize the bandwidth up to a specified PCR. This PCR's of the UBR+ VC's can be all the line interface level, in other word UBR+ offers the whole ATM capacity continuously. This wouldn't be possible with VBR connection [17].

As it is mentioned in section 4.4.2, assigning all of the VCs in one single VP has an important drawback. To keep track of the eventual fail or disconnection between RNC and BS, O&M link should be assigned in the independent VP with the UBR service. In this case of VP fail, the other VP is still on run.

4.5. IuB capacity

IuB VC capacities are usually quoted in units of ATM cells per second (CPS).

$1\text{CPS} = (53 \times 8) / 1000 \text{ Kbps}$

IuB capacity = user plane + AAL2 signaling + CNBAP + DNBAP + O&M

5. Traffic descriptor configuration/Connection & Configuration (CoCo)

Nokia Siemens network has specific traffic descriptor data base called CoCo, relating each RNC and all of the sites connected to it. CoCo details contain all of the sites in each RNC and their interfaces. With the help of this database planning engineers can easily allocate each interface for specific site and trace the network up to RNC. In the case of adding another Node B you can easily identify which interface in ATM switch is free and what is the related port to RNC in regard to the other sites.

				KLM												
Site Name	CoCo ID	SAXC	IMAG	E1 #1	E1#2	E1#3	E1#4	E1#5	E1#6	E1#7	E1#8	Interface ID	SET	WAM	DCN	VP
151	1510	T100	3-45	2.6.1	2.6.2	2.6.3	2.4.1	2.4.2	2.4.3			300	1	4	24	
1032	10320	T500	3-38	2.1.1	2.1.2	2.1.3	2.4.1	2.4.2	2.4.3			17	16	4	10	
1043	10430	T300	4-35	1.3.1	1.3.2	1.3.3	1.5.1	1.5.2	1.5.3			9	8	4	36	
1047	10470	T200	3-40	2.1.1	2.1.2	2.1.3	3.1.1	3.1.2	3.1.3			4	3	4	14	
1052	10520	T100	3-44	2.3.1	2.3.2	2.3.3	3.2.1	3.2.2	3.2.3			300	1	4	22	
1111	11110	T400	4-35	1.3.1	1.3.2	1.3.3	1.5.1	1.5.2	1.5.3			11	10	1	36	
1116	11160	T100	3-33	1.1.1	1.1.2	1.1.3	1.2.1	1.2.2	1.2.3			300	1	1	0	

Figure 18: traffic description details related to one RNC, Part one

	CBR	CBR	CBR	CBR	CBR	CBR	CBR	CBR	CBR	CBR	CBR
		CNBAP	DNBAP	DNBAP	DNBAP	DNBAP	DNBAP	AAL2SIG	USERPLANE	DCN	DCN
COCO VP	VP1	VC 42	VC52	VC53	VC54	VC55	VC 62	VC 72	VP0	VC32	
25	26792	404	808	808	808	808	404	22752	151	150	
11	26792	404	808	808	808	808	404	22752	151	150	
37	26792	404	808	808	808	808	404	22752	151	150	
15	26792	404	808	808	808	808	404	22752	151	150	
23	26792	404	808	808	808	808	404	22752	151	150	
37	26792	404	808	0	0	0	404	25176	151	150	
1	26792	404	808	0	0	0	404	25176	151	150	

Figure 19: Traffic description details related to one RNC, part two

Site name: Is the name of the NodeB which should be unique.

CoCo ID: which is the identification number related to each Node B.

SAXC: Is the name of the ATM switch with respect to different cards related to each switch. In Figure 18 we can see T100, T200, T300 and T400 which are the ATM switch cards.

IMAG: IMA group identify the ID of the IMA related to each card. IMA ID should be unique in each card.

E1: number of E1s in each Node B and their ID. E1s ID begins with 1.1.1 to 3.3.3 and should be unique in each RNC.

Set: which is the port of the RNC connected to each site. It is important to notify that each ATM Card should enter the same port. For instance, all of the sites connected via T100 are connected to port number 1.

WAM: number of WAMs related to each node B. Each site can have up to 4 WAMs and 12 WSP (3 WSP per one WAM). In the case of expanding the sites due to the high traffic, WAM card should be added.

VP1: This is the Virtual path capacity and all of the VCs inside related VP. You can easily justify each VC and its capacity.

Site ID 151: $VP1 = VC42 + VC52 + VC53 + VC54 + VC55 + VC62 + VC72$

$$26792(\text{CPS}) = 808 + 808 + 808 + 808 + 404 + 22752$$

VP0: Is used for O&M and it is recommended to be 151 CPS (~64Kbps) per Node B.

5.1. Traffic management in ATM switch

ATM network traffic control is very challenging because of the diverse nature of QoS to be guaranteed for voice, data, and video traffic. It is widely held in the research community that congestion control will ultimately decide feasibility of ATM technology.

A common cause of congestion is when traffic of the network is higher than the capacity that buffer can handle. The consequence of congestion is loss of data, due to the buffer overflow. The universal solution is to increase the memory of the buffer which is costly and is not the proper solution.

Due to the wide diversity of traffic classes and high speed of the network, no single mechanism can achieve full control in such networks. Thus a combination of control policies must be used. These policies called traffic managements. The popular techniques that have been used include call admission control (CAC), Traffic shaping, usage parameter control (UPC), Queuing, Priority control (using the CLP bit) and finally selecting discarding.

5.2. Traffic management functions

- **Connection Admission Control (CAC)**

CAC is one of the traffic management concepts that applies only to real time media traffic. In the 3G networks AXC always checks the consistency of VPI/VC value to determine whether a VP/VC connection request may be accepted or not. AXC also checks if new VPCs or VCCs can be accepted considering the total capacity requirements of all connections and the available capacity of the physical interface. As long as IMA groups are concerned, CAC accepts the requests based on the number of available links in the IMA group.

- **Usage parameter control (UPC)**

The network monitors and controls that the traffic contract is respected in terms of the traffic offered and the validity of the ATM connections. UPC is performed at the UNI (user network interface) and NNI (network node interface) and this function can be enabled or disabled for all connections at an interface.

AXC provides a method to detect the harmful connections. These traffics are either tagged (CLP=0 toggles to CLP=1) or discarded. The cell delay variation tolerance value is used by the usage parameter control algorithm that checks the submission to the declared cell rates of an observed cell stream.

- **Traffic shaping for CBR connections**

Traffic shaping alters the traffic characteristics of a VPC/VCC cell stream in order to achieve better network efficiency. All of the CBR connections must be associated with one traffic descriptor that is already provided by RNC and BTS. Traffic shaping is performed with respect to the peak cell rate that is already specified in the traffic descriptor of the CBR connection.

- **Traffic shaping for UBR**

Unspecified bit rate (UBR) traffic is normally used for DCN (dynamic circuit network) connections that are terminated in integrated IP router of AXC.

- **Packet discard and partial packet discard**

AXC features both packet discard and partial packet discard for VC connections carrying AAL5 traffic. Packet discard occurs in case the ATM buffer is congested and exceeds the

defined limit. Partial packet discard applies for cells that are to be discarded due to policing violation, threshold violation, or because no free buffer space is available.

- **MOC (Measure of congestion)**

Measure of congestion works in combination with UPC. UPC has two primary functions: cell marking and cell dropping. Each marked cell is fed to a selective threshold discarder module. Inside the module, the decision is made whether to transmit this marked cell or drop it. This decision is made based on a measure of congestion value.

5.3. Congestion and flow control in ATM switch

Congestion control in different networks has a direct effect on throughput. Powerful traffic management techniques can result in higher throughput and reducing block loss rates. If the speed provided by the network is lower than that requested by the application, there would be congestion at the entrance of ATM switches and the error and flow control systems in the end nodes must arrange the differences. Most applications use retransmission technique to recover lost data.

Two techniques are used in ATM networks. Automatic retransmission request (ARQ) and forward error correction (FEC). ARQ is based on retransmission when receiver for some reasons doesn't receive the data. FEC is an alternative for ARQ and send redundant information in form of codes that helps to recover the data without retransmission. ARQ incurs overhead in supporting status information and each retransmission causes round trip delay which is not suitable for time sensitive applications such as multimedia applications (voice, video). ABR and CBR services of ATM need information delivery for quick consumption.

FEC is an appropriate method for operation in high bandwidth delay product network. In this technique, redundant information are sent with original data so that if some of the original data is lost, it can be recovered using the redundant information.

Upon entering the interworking unit (IWU) of an ATM backbone in Ethernet network or transmitting CES interworking function (IWF) inside the BTSs, AXC takes the E1 signals and segments that stream into ATM cells. We assume that data packets are sent out from the ATM switch (CES) in sequence with numbers (required in many protocols). Since packets are sent out in ascending order, on reception side the missing packet can be identified by the gap in sequence. The missing packet can be considered as a sequence of bit-erasures, whose exact bit location is known. [Figure20]

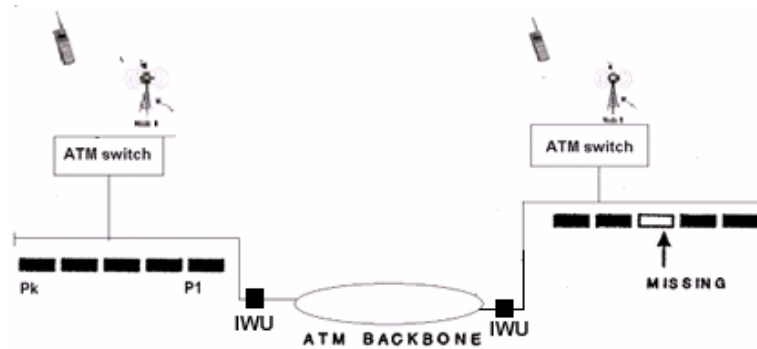


Figure20: packet streams through ATM network

Flow control means adjusting the cell rate of the source in response to congestion conditions and requires the implementation of closed loop congestion mechanism. This doesn't apply to CBR traffic. With ABR traffic, resource management (RM) cells are defined, which allow signalling of the explicit rate to be used by traffic sources. ABR is targeted at those applications that do not have fixed bandwidth requirement and require access to any reserve bandwidth as quickly as possible. This allows network operators to maximize the bandwidth utilization of their network and cell reserve capacity to users while still providing QoS guarantees [19].

5.4. Weakness of the traffic control in ATM network

As per our discussion about the ATM small cell size that provides many advantages like easy switching (because of the fixed and small cell size), reduced delay and arranged delay bound for real-time traffic, we can still see many problems of this small cell in 3G network. Even if segmentation the data into ATM cells (done in AAL5) is not a problem, but during the congestion a loss of one cell may submit the whole block of packets not functional. Loss packets are usually retransmitted. But retransmission causes low throughput and delay due to the wasted bandwidth and round trip time [19].

Real time transport protocols use FEC codes when retransmission is not effective, real time transport protocols use FEC codes. Every lost data packet can be recovered at the destination site using FEC coded packets. For the efficiency of FEC the number of redundant information kept small. Due to the parity check mechanism in FEC, this mechanism could be effective only and if and only if packet deletion is dispersed over several blocks, since multiple erasures over the same block cannot be recovered. In other words, we should control all of the cells in each packet and all of the packets in each block to optimize the throughput. Otherwise, for recovery of more than one packet in a block, more FEC coded packet must be designed and sent from the sender's side but the complexity of the decoder and encoder increases considerably which should be prevented [19].

6. Analysis of the solution

As per our discussion, FEC can recover loss data as long as dropped cells are dispersed over different packets and blocks. In my scheme FEC is applied at two levels—at the packet level and the cell level. If we assume that source sends blocks of K packets with an FEC coded parity check (figure 21), then two cases need to be considered. First one is the cell loss in the packet and second is the number of packets loss in each block. Ethernet frames have typically MTU size of 1500 byte which could contains about 32 ATM cells and in E1 (used in 3G as a carrier) up to 4830 CPS(cell per second) [1].

As soon as cell enters ATM switch, due to the usage parameter control, some cells would be marked and dropped at random depending on the MOC at the switch. The cell marking is done by the marker module and would be achieved by setting the cell loss priority bit of the cell (CLP=1). CLP bit marking usually is done at the user-network interface (UNI) of an ATM switch. Inside this module by the help of MOC value decision will be made whether to reserve the cell or to drop it. MOC value updated from the ATM switch's buffer content at any requested time. Buffer looks at the outgoing links that depending of the VCs services has two kinds of queues, one for real time traffic (qrt) and the other one for non real time traffic (nrt) times some arbitrary functions α and β . Threshold value for congestion control in the buffer can be chosen as follows:

$$\text{MOC} = \alpha \text{qrtl} + \beta \text{qnrl} \quad [19]$$

The cells with Red Cross (X) in figure 21 represent marked cell that may get dropped. Losing one cell in a packet could be recovered if we use FEC recovery scheme. But if more than one cell is lost, then the whole packet becomes useless.

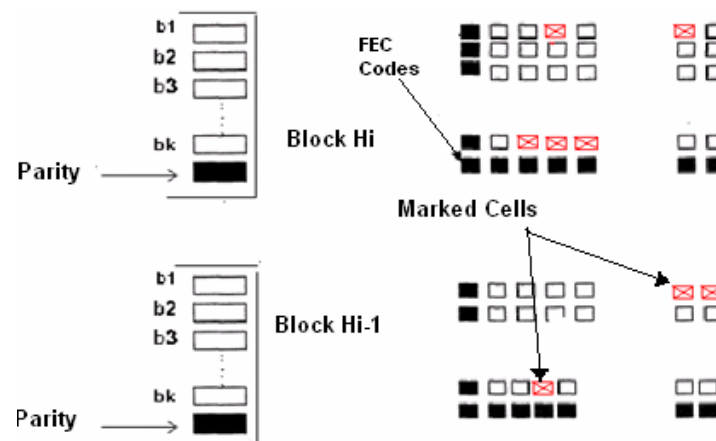


Figure21: Conversion of packets to ATM cell.

The idea is to first distribute the losses over different packets in different blocks. This could be done by localizing cell dropping to one packet as much as possible so that other packets could remain uncorrupted. In figure 21, if more than one cell in

packet b_1 is dropped then the whole packet become useless. Now if cells corresponding to packet b_k are lost then the whole packet becomes useless too. None of those two packets could be recovered in this case resulting in the loss of whole block [19] [20].

We must search for the suitable candidates that can be dropped without affecting the throughput. Existing cells in the buffer from a packet that has suffered too many losses or cells from uncorrupted packets are good candidates. Also blocks that have lost more than one packet are good candidates [19] [20].

To accomplish this aim I suggest three modules in traffic management as tools, MOC, UPC and finally selective discarding. With help of these tools buffer intelligence could trace the losses within a frame and the block. With help of UPC buffer can controls and monitors the traffic in the networks entrance and dropping decision is based on the chosen MOCs value. For keeping track of the cells I need to introduce gatekeepers and porter in ATM output buffer [19] [20].

In my scheme two flags will be introduced to track the losses both in the packets and blocks. These flags are called packet loss flags (PLF) and block loss flags (BLF). PLF denotes whether a packet lost any cell and respectively BLF denotes if any block has lost any packet. BLF and PLF are designed to have two values, zero and one. Value one identifies that packet or block has room for losing one cell or one packet. Value zero shows that packet or blocks has already lost one cell or packet and has no room for further losses. Last packet bit cell in each packet is set so when this cell passes by the ATM switch the value of the PLF reinitialized to one. Parity packet end also showing that the next block is coming and the value of BLF reinitialized to one after each block [19] [20].

To accomplish this we should add intelligence to the output buffer. This intelligence operates in two states, Gate keeper and porter. Threshold value activates the gatekeeper whenever load values exceed the upper threshold. In that case, gatekeeper filters all of the cells that could be dropped based on the CLP value [19] [20].

We should notice that all of these values are defined per VC. When one cell arrives to the buffers interface its CLP value would be checked. If CLP is not set it is admitted into the buffer, otherwise MOC will be checked to sense the actual traffic in comparison with the upper threshold. If the load is greater than upper threshold, gatekeeper becomes active and drops the cell by checking the PLF value. If $PLF > 0$ then cell will be dropped and PLF value decrements to zero, If not the gatekeeper checks the value of PLF and BLF at the same time. If $PLF = 0$ and $BLF = 1$, this means that one packet could be dropped and gatekeeper filter the whole packet and update the value of BLF to zero meaning that block has already dropped one packet and should preserve the other packets. If both BLF and PLF are equal to zero, this mean that cell dropping results in the whole block to be useless. Porter in the other hand does the rescue job and cleans up the buffer and removes all of the eligible

candidates from the buffer. We should notice that porter needs memory and processing time. All of these

mentioned procedures are performed only once until MOC goes under upper threshold. The value of the MOC is based on the range of the lower and upper threshold and is updated at a specific time according to the new value of the network load [19] [20].

7. Conclusions and future works

1- In this paper I describe the transmission phase in 3G network including Node B, RNC and interconnection between them. Node B is a transceiver which has a different part. Among others, node B receives the radio signals from user equipment and maps them into ATM cells and transmits them through the IuB interface to the RNC. The transmitted information through the IuB can be in different type depending on the user's request. The type of service category that is allocated to each information type is also different. For instance Voice can not tolerate the delay and is categorized as a real-time transmission and should be able to use a sufficient bandwidth constant. That's why we allocate the constant bit rate for voice and other delay sensitive information type to guarantee the QoS. Each IuB is defined as one VP and inside each VP five VCs are introduced. Each VC is allocated for specific kind of information to communicate with RNC. One of the bottlenecks of the existing 3G network is that VC related to O&M is introduced as a UBR. It means that O&M link can be totally starved if the other links need more bandwidth. In the future, another type of service will be used which is a new version of UBR service and is called UBR+. In this condition, the minimum cell rate and peak cell rate for the O&M is defined and as a result we can guarantee that O&M can always be on the run.

Each IuB is configured as a single virtual path and within this VP there are a number of virtual circuits, which can be divided into 5 main types. Operation and maintenance is allocated in one of the VCs and it has a very important role in communications between RNC and Node B. In practice we can consider that one VP for some reasons can fail and loose connection and as a result the whole connection between the node B and RNC is lost. As I mentioned before O&M is used to show the alarms and download the software in RNC and engineers with help of this link can keep track of the connectivity and easily find the solution in case of interruption in the link. In future O&M will be configured in a separate VP and as a result if the whole VP fails there would be another VP on the run and we can still communicate with the RNC and repair the connection.

2- IuB transfer the information in ATM cells and it has a fixed bandwidth. To enhance the best QoS in the network each user should be able to use the different services without any limitation. In rush hours there would be congestion in the links and it could lead to information loss. ATM switches has buffer to prevent the losses but the buffer capacity is limited. The lost data are usually retransmitted but it is costly and affects the throughput. FEC protocol is negotiated to be a reasonable solution to recover the lost data. FEC codes are transmitted with the original data and in case of data loss; these codes can recover the data. Any lost data can be recovered at the

destination site using FEC coded packet. The amount of extra information (FEC codes) kept small, so that FEC is efficient. So the FEC packet helps to recover part of

the losses, while the additional data due to FEC increase the overall load, which makes the loss-rate worse. A necessary condition for FEC to be effective is that losses scatter over several blocks. In my scheme, I have suggested a method to mark the cells and packets and keep track of the losses in different blocks. With the help of this method we can prioritise cell losses and at the same time save the other cells in each packet. As a result, ATM switch's buffer manager works as a gate and in case of congestion and threshold in the traffic drop some cells and send the other cells through the gate and prevent them to become useless. In the future, I can propose a scheme that defines the efficient size of ATM switch buffers, which guarantees no-loss transport of real-time information.

8. Appendix

[1] The E1 transmission link consists of 32 transmission channels (0-31), each of which is 64 Kbits/sec. The overall transmission rate is 2.048 Mbits/sec. Channels 0 and 16 are reserved for transmission management, while all other channels are used for payload. The payload bandwidth is thus 1.920 Mbits/sec. Since ATM uses 48 out of the possible 53 bytes for payload transmission, the net transmission rate becomes 1.738 Mbits/sec.

One ATM cell has 53 byte (including header)/53*8=424bits. Then E1 can handle $2.048 \times 10^6 / 424 = 4830$ CPS

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