ANALYSIS AND OBSERVATION ON QUALITY OF SERVICE FOR DATA NETWORKS OF TURKISH TELECOMMUNICATION INFRASTRUCTURE

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ABSTRACT

ANALYSIS AND OBSERVATION ON QUALITY OF SERVICE
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INFRASTRUCTURE

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In this thesis, Jitter, Delay and Packet Loss values are measured for different points on TT IP/MPLS Backbone and compared with acceptable values. Measurements are made by using TT's Alcatel 7750-SR12 devices which run in Ankara, Konya, Kahramanmaraş and Van. During measurements the device running in Ankara is always used as resource.Each of the other points are used as targets.The transmission distances between the devices measurements are as Ankara-Dikmen distance 10 Km and Ankara-Konya 300 Km, Ankara-Kahramanmaraş 602 Km and finally Ankara-Van 1248 Km. In this way, the connection between Jitter, Packet Loss and Delay Values and transmission distance is determined

During tests every measurement is performed using 1500 and 4000 Bytes divided packets.The measurement values are taken using ping command on a software called SMART which can reach all devices on TT IP/MPLS Backbone.

Measurements made are calculated in a day time zone. The connection of the software by which we perform measurements to the backbone is 100 Mbps.

In the measurements taken, increase in Jitter and Delay value is observed when there is an increase in packet size. At the same time, increase in Jitter and Delay value is observed when there is an increase in transmisson distance. The packet loss value is evaluated as %0 in all of the tests made on TT IP/MPLS Backbone.

Keywords : Quality of Service, Service Level Agreement, Delay, Jitter, Packet Loss

ÖZ

TÜRK TELEKOM DATA ALT YAPILARINDA KALİTELİ SERVİSLER ÜZERİNE ANALİZ VE GÖZLEMLER

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Bu tezde, Türk Telekom IP/MPLS Omurgası üzerinde Jitter,Delay ve Packet Loss değerleri farklı noktalar için ölçülmüş ve kabul edilebilir değerler ile kıyaslanmıştır.Ölçümler Türk Telekom'un Ankara,Konya,Kahramanmaraş ve Van illerinde çalışmakta olan Alcatel 7750-SR12 cihazları kullanılarak yapılmıştır.Ölçümler esnasında Ankara da çalışmakta olan cihaz her zaman kaynak olarak kullanılmıştır.Diğer noktaların her biri hedef olarak kullanılmıştır.Ölçüm yapılan cihazlar arasındaki transmisyon mesafeleri,Ankara-Dikmen arası 10 Km, Ankara-Konya arası 300 Km, Ankara-Kahramanmaraş arası 602 Km ve son olarak Ankara-Van arası 1248 Km dir. Böylelikle Jitter, Packet Loss ve Delay değerleri ile transmisyon mesafesi ile arasında nasıl bir ilişkiye sahip olduğu tespit edilmiştir.

Testler sırasında her bir ölçüm 1500 ve 4000 Bytes lık parçalanmış paketler kullanılarak gerçekleştirilmiştir.TT IP/MPLS Omurgası üzerindeki tüm cihazlara erişebilen SMART adlı yazılım üzerinde ping komutu kullanılarak ölçüm değerleri alınmıştır.Yapılan ölçümler 1 günlük zaman dilimi içerisinde hesaplattırılmıştır.Ölçümleri gerçekleştirdiğimiz yazılımın çalıştığı sunucunun omurgaya bağlantısı 100 Mbps dır.

Yapılan ölçümlerde paket boyutu artıkça Jitter ve Delay değerinin arttığı gözlemlenmiştir. Aynı zamanda transmisyon mesafesi arttıkça Jitter ve Delay değerlerinde artış gözlemlenmiştir. TT IP/MPLS Omurgası üzerinde yapılan testlerin tamamında Paket Loss değeri %0 olarak tespit edilmiştir.

Anahtar Kelimeler : Hizmet Niteliği, Servis Düzey Anlaşması, Gecikme, Zaman Sapması, Paket Kaybı.

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LIST OF SYMBOLS

Bc	Committed Burst Size
Be	Excess Burst Size
D(i)	First Packet Delay
D(i-1)	Second Packet Delay
Kbps	Kilobits Per Second
Khz	KiloHertz
Km	Kilometer
Mbps	Megabits Per Second
Tc	Committed Rate Measurement Interval
Gbps	Gigabits Per Second
ms	Millisecond
be	Best Effort
af	Assured Farwarding
ef	Explicit Forwarding

LIST OF ABBREVATIONS

ABR	Available Bit Rate
ATM	Asynchronous Transfer Mode
BECN	Backward Explicit Congestion Notification
CAC	Connection Admission Control
CBR	Constant Bit Rate
CDVT	Cell Delay Variation Tolerance
CIR	Committed İnformation Rate
CLP	Cell Loss Priority
CLR	Cell Loss Ratio
CPE	Customer Premises Equipment
CRC	Cyclic Redundancy Check
CTD	Cell Transfer Delay
DCE	Data Communication Equipment
DE	Discard Eligibility
DLCI	Data Link Connection İdentifier
DSCP	Differentiated Services Code Point
DTE	Data Terminal Equipment
EA	Extended Address
EIA/TIA	Electronics Industries Association And Telecommunications Industries Association
EIR	Excess Information Rate
EXP	Experimental
FCS	Frame Check Sequence
FECN	Forward Explicit Congestion Notification
GFC	Generic Flow Control
GFI	General Format Idendifier
HDLC	High-Level Data Link Control

HEC	Header Error Check
IP	Internet Protocol
ISO	International Organization Standardization
ITU-T	International Telecommunication Union-Telecommunication
LABP	Link Access Balance Procedure
LAN	Local Area Network
LCI	Logical Channel Identifier
LDP	Label Distribution Protocol
LLC	Logical Link Control
LMI	Local Management Interface
LSP	Label Switched Path
LSR	Label Switched Router
MAC	Media Access Control
MBS	Maximum Burst Size
MCDT	Mean Cell Delay Time
MPLS	Multi Protocol Label Switching
NNI	Network to Network Interface
OSI	Open Systems Interconnection
OSPF	Open Shortest Path First
PCR	Peak Cell Rate
PLP	Packet Layer Protocol
PSTN	Public Switched Telephone Network
PT	Payload Type
PTI	Packet Type Identifier
RTT	Round Trip Time
QOS	Quality Of Service
S	Stack
SCR	Sustainable Cell Rate
SDLC	Synchronous Data Link Control
SLA	Service Level Agremeent
SNMP	Simple Management Network Protocol
TDM	Time Division Multiplexing
TT	Turkish Telecommunication

TTL	Time To Time Live
UBR	Unspecified Bit Rate
UNI	User To Network Interface
VBR	Variable Bit Rate
VBR-RT	Variable Bit Rate-Real Time
VBR-NRT	Variable Bit Rate-None Real Time
VCI	Virtual Channel İdentifier
VLAN	Virtual Local Area Network
VPI	Virtual Path İdentifier

CHAPTER 1

INTRODUCTION

When providing the voice transmission on Public Switched Telephone Network (PSTN), circuit switching method is used. By this method's concept, between the two users who will make call, a voice channel of 4 Khz bandwidth is used. Digital form of this bandwidth is 64 Kbps. When considering this substructure used for voice as to be used for data communication, upon the messaging speed between receiver and transmitter, we will need to dedicate one or more of these channels. As this line is assigned to a user, during the period not in use, it's impossible to be used by any other user.

In order to remove this disadvantage, it has been changed over from circuit switching to packet switching [1]. Thanks to this, each empty capacity wil be able to used for another user. One of the other advantages of packet switches which base the data networks, data to be sent are resolved into small packets; by passing of each packet through a different link and node, these packets are joined where they access, so the data transmission is implemented. In this way data and image are moved on the same structure [2].

It is a reality that the capacity of devices used in data networks and the links which interlink these devices are unlimited. It should be decided to give whom and how much of these sources which are limited in data networks, as in all. The mechanisms are used for this action are gathered under Quality of Service (Qos) title [3, 4, 5, 6]. Each data network infrastructure has its own Qos applications and its usage method.

Qos mechanisms have specific rules and policies. It has a comment and authority power forevery packet which passes through the network it works.

So, it has in its mechanisms that decides which packet will pass through the nodes point first, which will be dropped and which should be waited in buffer. As a result of these policies, it will be seen that some values important for the packet differentiation, while moving of packet from one point to another.

In this thesis, there have been some measurements about Qos mechanisms on TT IP/MPLS Backbone. The tests have been performed on TT's IP/MPLS Backbone, between Ulus-Dikmen, Ulus-Konya, Ulus-Kahramanmaraş, Ulus-Van devices. By measurements are made, it is observed how values like latecy, jitter, packet loss differ according to transmission length. Besides, by using packets fragmented of different sizes during tests, the change of these values according to packet size is observed. The values obtained from tests are compared with the acceptable values.

In chapter 2, brief information for data networks that have been used up to now and is stil used. Some information about Qos parameters like network delay, Jitter, Packet Loss, Throughput and Bandwidth in Chapter 3. In Chapter 4, some acquirements are given about applications for each data substructure which is used in Turkish Telecommunication. In Chapter 5, it is told how the tests are performed on TT IP/MPLS Backbone. At the same time, the results regarding to these tests are shown graphically. Finally in Chapter 6, all these results are evaluated.

CHAPTER 2

DATA NETWORKS

The devices used first, which generate data, had been designed to work as stand alone. Later on, because of the needs of both file and equipment sharing of these devices, the need of communication within each other became apparent. In order to provide this communication some specific levels have been established and each of these has been called as a protocol header. Each header is used for packet communications in its substructures by data devices. Open System Inteconnection (OSI) layered structure which is developed the data communication rules by International Standart Organization (ISO) [7,8].

On each layer of this structure which consists of 7 layers, different types of protocol definitions have been made Figure 2.1.

Application-Layer 7
Presantation-Layer 6
Session-Layer 5
Transport-Layer 4
Network-Layer 3
Data Link-Layer 2
Physical-Layer 1

Figure 2.1: OSI Model

On data communication, chronologically x25, Frame Relay, Asynchronous Transfer Mode (ATM) and finally Multi Protocol Label Switching (MPLS) have been used.

When we look these protocols, while x25 works at layer 3, ATM, MPLS and Frame Relay Layer Works at Layer 2.

2.1 X.25

X.25 is the first protocol which uses packet switching and is designed to transport data. The communication is performed on virtual circuit between one point and a second point, which is formed by connecting a Data Communication Equipment (DCE) and Data Terminal Equipment (DTE) item. While doing this, as in all communication technology, it take the advantage of OSI model. The structure of protocol is specified in Figure 2.2 [9].

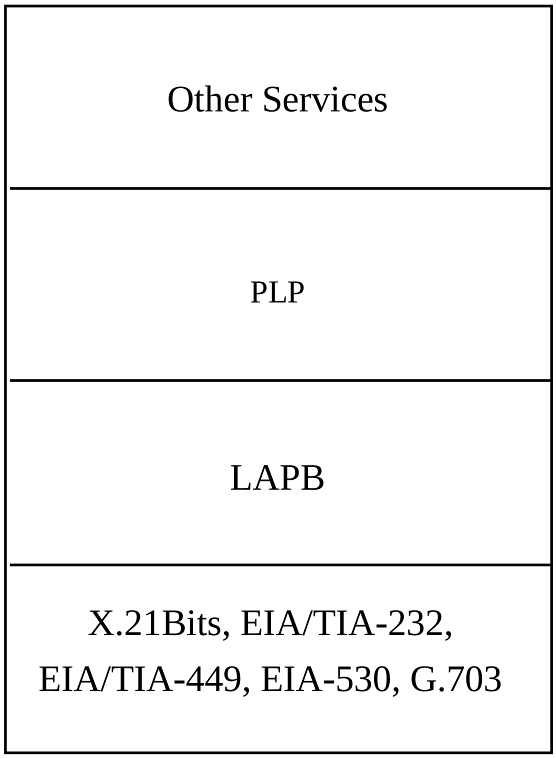

Figure 2.2: Header of Protocol of X.25

Packet Layer Protocol (PLP), works on the network layer, the third layer of OSI layered structure. It furnishes the administration of the virtual circuit between the two points meet [10,11].

PLP works as 5 different modes. The first of these is call setup, supplies the install of virtual circuit among DTE devices. On data transfer mode, yet, the communication between DTE devices is provided, for the errors may come into being bit padding, flow and error control are made. On idle mode however, virtual circuit install has been provided. However, they are the none-sending-data situations. In order to cut the virtual circuit between two DTE devices, it is run on call clearing mode. Restarting mode, then, is used to obtain sychronization for the DTE and DCE devices [10,,11,12].

There are 4 areas on the packet sent in PLP. Those, as shown in Figure 2.3, are General Format Identifier (GFI), Logical Channel Identifier (LCI) Packet Type Identifier (PTI) and User Data [10,11,12].

GFI (4 Bit)	LCI (12 Bit)	PTI (8 Bit)	USER DATA (8 Bit)

Figure 2.3: Structure of PLP Layer

But on Link Access Procedure Balance (LAPB) layer, the structure and areas of frame are as Figure 2.4.

Flag (1 Bit)	Address (12 Bit)	Control (1 Bit)	Data (Variable)	FCS (2 Bit)	Flag (1 Bit)

Figure 2.4: Structure of LAPB Layer

Flag is the area used to bound the packet. The address is an area showing whether it is frame transporter or not. Control, is the area which carries the frame sequence number and determines the frame type [10, 11, 12]. The datas from PLP is

includes header informations. The Frame Check Sequence (FCS) is the area necessary to make error control [10, 11, 12].
On physical layer, predominantly x.21 and x.121 are used [22,26].

X25 protocol is the protocol where Qos is used in the simplest way. For setting the queuing mechanism determines number of packets will be placed in the queue by the user [2,11].

A part from this, in order of priorities, it is possible to define 4 traffic types. It's possible to assign different priority to different traffic types [2,11].

2.2 Time Division Multiplexing (TDM)

On this structure, the same signal is used for all users. The differentiation based on user, yet, is obtained by time concept. On specific intervals, specific users can send and receive data. There are two types of Time Division Multiplexing (TDM) systems. Statisticial and SynchronousTime Division Multiplexing [1,13].

In this system, circuit switching technology is used. It means a circuit dedicated from one edge to the other. This dedicated circuit, it can only be used between these two points whether there is Exchange of information or not, as they are dedicated for those two users [1,13].

Existent link capacity in TDM are divided into time-slots [1,13]. Each time slot has a voice channel capacity. Two time-slots are used ,only for clock and synchronization. User data can be transported through channels except for these two channels [1,13].

As there is a consigned line from one edge to other for the user in TDM, there is no need of Qos and the applications. Every user sends his data from one edge to other at equal conditions. On Backbone, data of each user is forwarded instantly. Difference is the size of the data to be send. For instance, one has to use 4 time-slots in order to transmit 256000 bps data in a second.

There is a slice example of data transmission way in TDM networks on Figure 2.5.

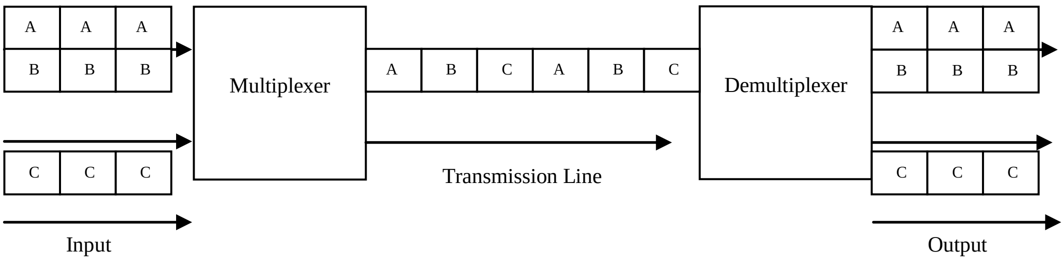

Figure 2.5: Working Mechanism of TDM

2.3 Frame Relay

It is one of the first protocols which uses the advantages given by packet switching. As seen in Figure 2.6, if two points are considered as linked on a frame delay on Backbone, switch working on Backbone is linked to devices by the help of Customer Premises Equipment (CPE) device User to Network Interface (UNI) which works on the side of the user. This interface which connects these switch devices to each other is called Network to Network Interface (NNI).

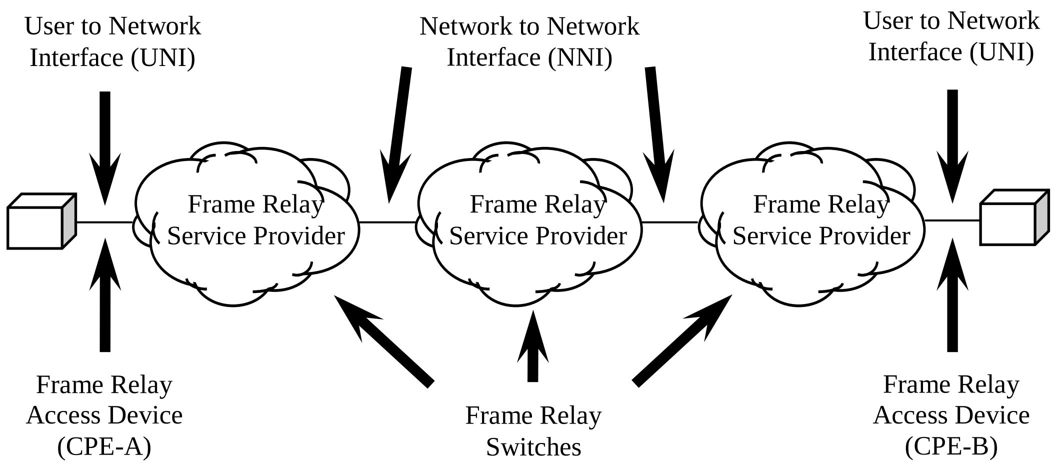

Figure 2.6: Working Mechanism of Frame Relay

Depend on this topology, between two users, the communication will be provided by the configurations made on Backbone. The definitions that should be made discrepancy.

Between the two points, through the logical connections are called Permanent Virtual Circuit (PVC) or Switches Virtual Circuit (SVC), the traffic on A is forwarded to B [9,.14,15]. In order this packet sent over Backbone to go ahead all the switching devices seamlessly, Frame Relay header should be added to the packets generated on users' devices. Packets are forwarded safely and properly to the target they should get, by the usage of areas in this header.

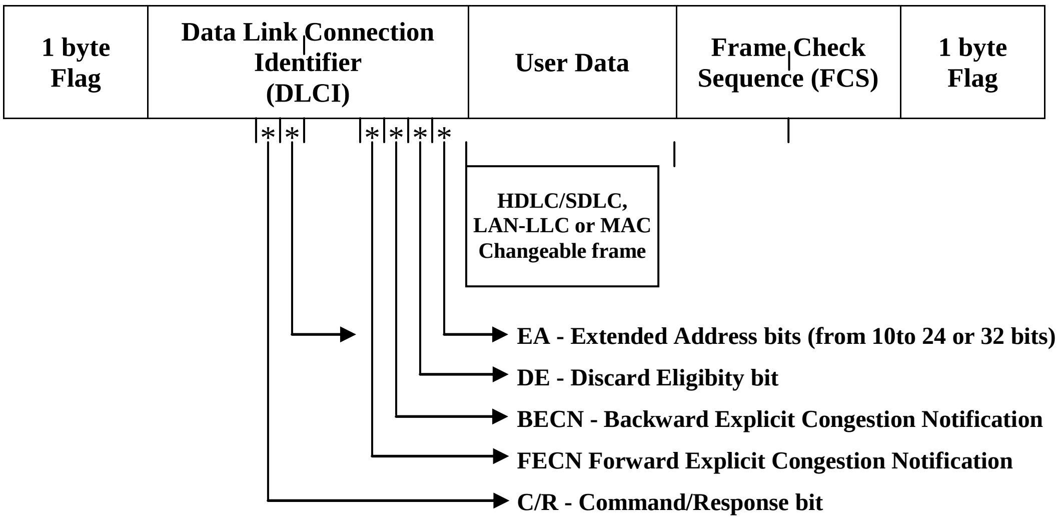

Figure 2.7: Header of Frame Relay

The Frame Relay header, which seems on Figure 2.7, has been standardized by International Telecommunication Union (ITU-T), authority on this topic, and called Q.922 (Annex-A). Should express the areas on this header, firstly Data Link Connection Identifier (DLCI) are told that, by this area, a physical port is divided into logic circuits. In this way, each DLCI composes a different edge of a separate circuit. While Frame Relay packets through on Backbone, it will be needed fixing congestions to be formed. The areas used to fix this congestion are

Backward Explicit Congestion Notification (BECN) and Forward Explicit Congestion Notification (FECN) [9,15].
According to the Frame Check Sequence (FCS) bits, the control whether the packet accesses its place is done, depend on checking the sent and received frame relay packets.

This header information, variable lengthed frames, can be send like High-Level Data Link Control (HDLC), Synchronous Data Link Control (SDLC), Local Area Network- Logical Link Control LAN-LLC or Media Access Control MAC on payload area [2].

On Frame Relay Backbones, thanks to two parameter, a band limit may be defined by allowing previously determined data which will pass in second. First of all is called Commited Information Rate (CIR). It is the speed info that is guaranteed totaly by the supplier. The second is Excess Information Rate (EIR). This speed is not guaranteed, and it will be used as much as the links in backbone allow. It is one of the packets which will be dropped first depend on a congestion of traffic.

On Frame Relay Backbones, the signalization called as Local Management Interface (LMI) is used in order to control edge devices [9, 14, 15].

Qos mechanisms which are performed on frame relay networks are much more developed as to x.25 protocols. The bandwidth is used much more effectively in Frame Relay networks. While doing this, two definitions of bandwidth are made. They are called as CIR and EIR. While CIR means guaranteed speed, EIR means unguaranteed speed [9,15]. The abbreviation of CIR is expressed as (bc) and EIR is expressed as (be). We may summarize this as: The speed expressed as EIR is fixed as first packed to be dropped in any case of jam [9, 14, 15]. This fixing is done by DE bit. The graphical summarize of this situation is shown in Figure 2.8.

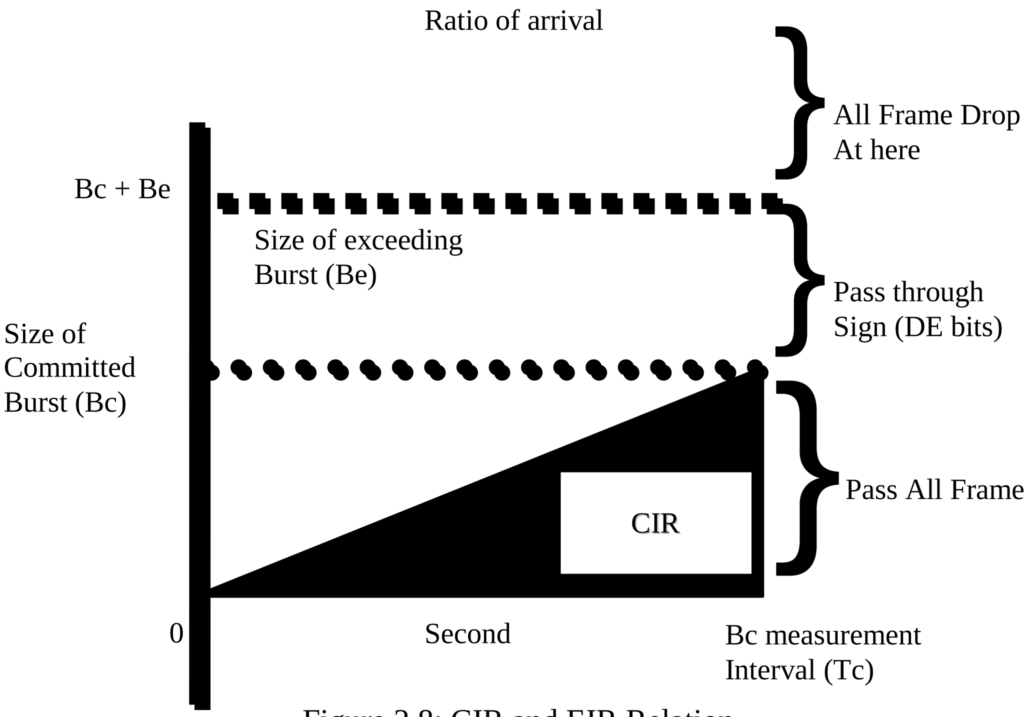

Figure 2.8: CIR and EIR Relation

Except this rule, the congestion within the network is detected towards forward or backward by setting FECN and BECN bits.

2.4 Asynchronous Transfer Mode (ATM)

It is a protocol using packet switching technology, which came into existence by making the usage of advantages and gripping disadvantages of data network structures that we have been trying to explain, designed to carry different traffic types, in data networks.

On ATM Networks, data generated by user devices are transported via cells. The data of user are placed into 48 bytes of cells; by adding 5 bytes of header informations, they are carried as 53 bytes of cell. You can see the structure of these headers in Figure 2.9 and Figure 2.10. There are two types of headers here. It changes depending on the connections either UNI or NNI [2].

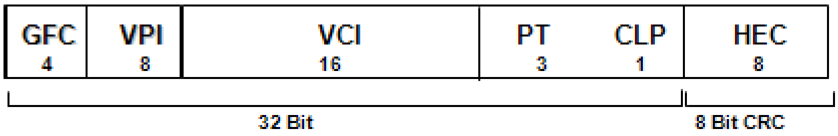

Figure 2.9: ATM UNI Header

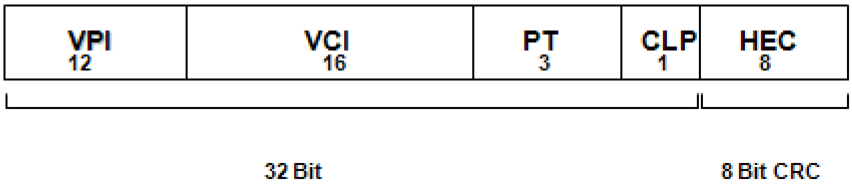

Figure 2.10: ATM NNI Header

On ATM substructure, the limit is 2 Mbps. At Frame relay, the parameter which corresponds DLCI is Virtual Path Information (VPI) for ATM and Virtual Circuit Information (VCI). Through, a physical port is used for logical circuit. At the same time the addressing is done [16,17].

Another field used in ATM headers is called Generic Flow Control (GFC). A control mechanism is run for different connections between Backbone and users [16,17].

By the field called Payload Type (PT), the type of the data in payload of packets are assigned. According to different variations, this area PT helps to determine whether the packet belongs to the user or it is used in the system communication of the Backbone [16,17].

Cell Loss Priority (CLP) is the first bit to be dropped of the cell which is included. In case of congestion, it assigns the priority of its cell to be dropped [16,17].

Header Error Control (HEC), is used to be able to control whether moved cell is forwarded to the target properly. On ATM, Cyclic Redundancy Check (CRC) mechanism is used for this area [18,12].

At the ATM Layer is where cell switching is done, GFC kontrols are realized, VPI/VCI transformation is performed [16,17].

It is the layer which transmits to an upper level according to the type and kind of the data to be forwarded on ATM adaptation layer. [16,17]

At the ATM protocol, Qos mechanisms are used quiet densely. The output purpose of this protocol is to provide transmitting totally different traffic types to where to reach, after melting them in a potash, by the help of specific rules and mechanism.

Different traffic types are developed as to the different values the parameters take, like Maximum cell delay (MCDT), Cell Delay Variation Tolerance (CDVT), Cell Transfer Delay (CTD) Cell Lose Rate (CLR), Maximum Cell Rate (PCR), Sustainable Cell Rate (SCR), Maximum Blast Size (MBS) [16,17].

They can be listed in order from the most quality service to the least quality service, as Constant Bit Rate (CBR), Variable Bit Rate-Real Time (VBR-RT), Variable Bit Rate-None Real Time (VBR-NRT), Available Bit Rate (ABR), Unspecified Bit Rate (UBR) [16,17].

According to the traffic type, before the data is forwarded between two points, resource allocation is made before the user data is sent, on every point in that cells will pass by the signalling the thanks to Call Admision Control (CAC) mechanism [16,17].

2.5 MPLS (Multi Protocol Label Switching)

It is the protocol, in order to transmit data, which provides connections from point to point, point to many points and many points to many points. As it has a feature of having all the data networks we’ve mentioned so far, it makes use of all the advantages of all data Networks [19,20,21].

At the present day, user devices are IP based. As the cost of communication decreases by Internet Protocol, it could be used for the online communication like video or voice [2].

In order to achieve this, A down-to-earth protocol like MPLS has been invented, which covers all the technologies so far, and which will be able to meet the needs of future.

According to this protocol, on its OSI stratified structure which is making the use of advantages of routing protocols on its 3. layer, also by making the use of the speed of switching mechanisms working on its 2. layer, it provides the Internet Protocol to be used in safer applications.

LABEL (20 Bits)	EXP (3 Bits)	S (1 Bits)	TTL (8 Bits)

Figure 2.11: Header of MPLS

In Figure 2.12, the areas of a MPLS header are depicted. By the help of this area of 20 bit, it is possible to generate 2^{20} labels. Of these labels, these labels from label 0 to label 3 are special labels used to signalization. Other labels are used to addressing [19, 20, 21].

The 3 bit area defined as experimental is the area used to define traffic type.8 different traffic type may be defined [19, 20, 21] by using this area.

Time to Time Live (TTL) area of 8 bit is the duration which determines when the generated data will be dropped. By using this area, the packets which cannot access the target they should be on Backbone, after the period to be defined in this area, it will be dropped via various mechanisms [19, 20, 21].

1 bit of area called stack, checks whether there is a label on previous area. In MPLS packet structures, more label may be used. The resolution among these labels are provided by this area.

Before transmitting data from another point on Backbone, signalization should be done in order to calculate and allocate resources on each Label Switch Router (LSR) data will pass. What starts signalization is the routing protocols. The most popularly used of these is Open Shortest Path Forwarding (OSPF) [18, 19, 20, 21]. At each node to be transited during the signalization, thanks to Label Distrubution Protocol (LDP), proper labels are prepared and reserved [19, 20, 21]. On the first place that the packets enter the network, pop operation is done by using suitable label. While packets move along on the network, swapping the labels on every point. Data is forwarded to the target desired at the last point by pushing label and dropping the label area. Figure 2.13. In the meantime, each LSR forms Label Forwarding Information Base (FIB) tables and the informations are kept in these tables [19, 20, 21].

MPLS Protocol where the Qos is used, is in the most effective way. At present, it still has this feature. By the help of MPLS, there is a chance of implementing end to end. By Using of this feature is not supported by other protocols, it is possible to make a Qos definition between the devices used by the last user.

The packets that is entered in MPLS Backbone, is categorized by means of some parameters like Differentiated Services Code Point (DSCP) bits and Virtual Local Area Network (VLAN) Id [19, 20, 21]. One of the Qos policy is applied according to the traffic type that comes after this classification. While the coming packet is being converted to MPLS packet, experimental bits are calibrated and is associated with one of the 8 traffic types [19, 20, 21]. By this association, packets are accessed to the target in the result of application Qos policies which contains functions like Queuing, Scheduling, Shaping, Marking at every point the packet passes. 8 traffic types are given on Table 2.1

Table 2.1: Types of Traffic in MPLS

Forwarding Class		Loss Priority	CoS Value
best-effort	be	Low	000
best-effort	be	High	001
expedited-forwarding	ef	Low	010
expedited-forwarding	ef	High	011
assured-forwarding	af	Low	100
assured-forwarding	af	High	101
network-control	nc	Low	110
network-control	nc	High	111

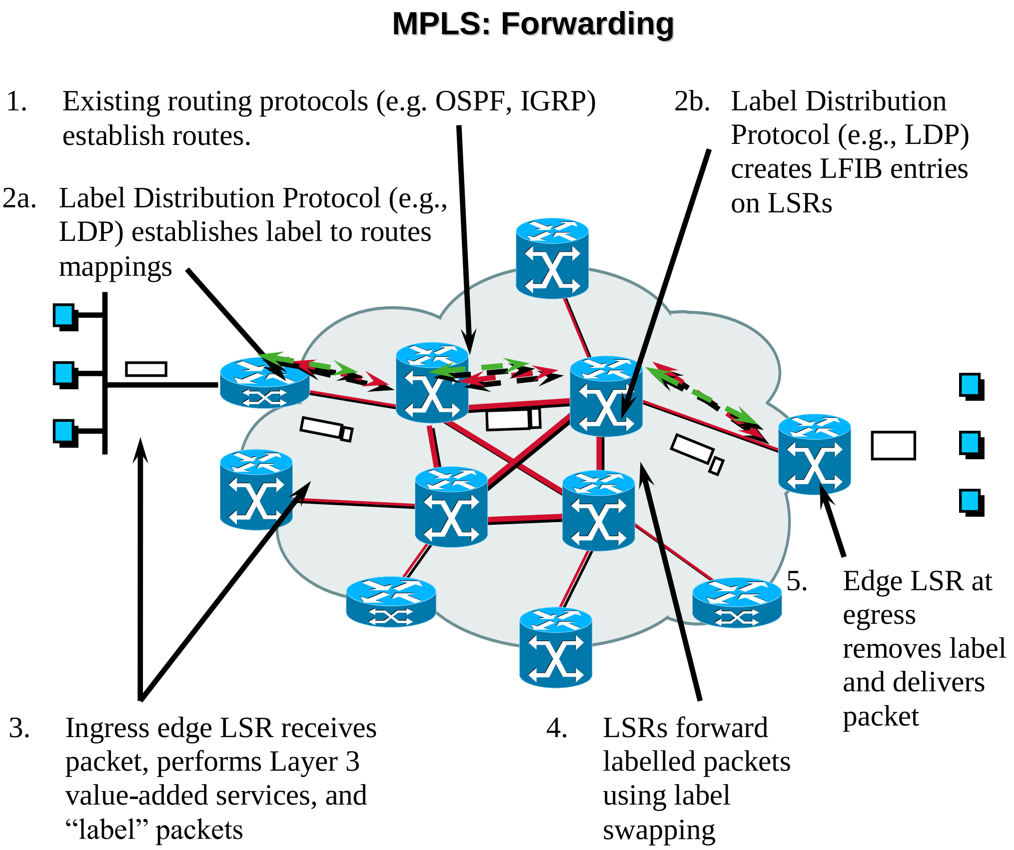

Figure 2.12: Working Mechanism of MPLS

CHAPTER 3

SERVICE LEVEL AGREEMENT (SLA)

In the services which are provided on data network, assured of the some services parameters with regard to service quality by the provider company is called Service Level Agreement (SLA) [22].These customers, who use high quality services, want some values be guaranteed for parameters such as network delay, bandwidth and throughput, jitter, packet loss according to the application type they use. In case of not providing the guaranteed values, the company is supposed to refund to the users. These parameters differ according to the type of data used [22].

3.1. Network Delay

A Delay is the total duration starting with the generation of the data between the source target and finishing in the delivered part. In online applications this parameter is the important because data is required to be transmitted as soon as possible in online applications.

There are some particular standards determined by ITU for the delays occured in network. According to G.114 these values are give in the Table 3.1 below.

Table 3.1: Delay Specifications

Range in ms	**Description**
0-150	Acceptable for most user applications.
150-400	Acceptable provided that administrators are aware of the transmission time and the impact it has on the transmission quality of user applications.
Above 400	Unacceptable for general network planning purposes. However, it is recognized that in some exceptional cases this limit is exceeded.

3.1.1. Propagation Delay

The delay which occurs on the communication links used for the transmission of data, that is generated in digital devices, from one point to another point is called propagation delay. Figure 3.1 [4,5,23]. The calculation method is found for dividing of propagation speed by the length of physical media.

Propagation delay =length of physical of link / Propagation speed in medium

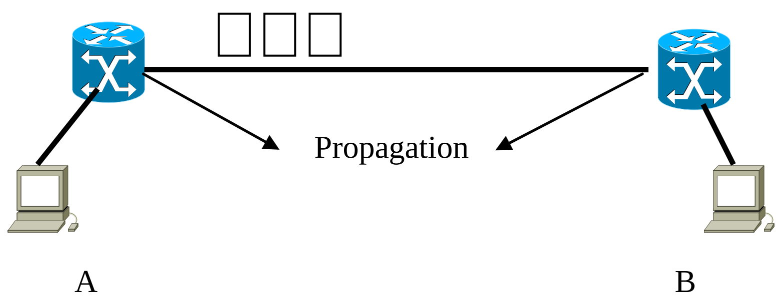

Figure 3.1: Propagation Delay

3.1.2. Switching Delay

The duration between a for packets input to the switching device and for output from the same device is called switching delay time.

The duration is in the level of micro-seconds. Although the switching delay is assumed theoretically it is very difficult to calculate. The calculating the switching delay is the total of delays occured in each switching device gives us the total switching delay.

Switching Delay=Number of Switches in Originating Network x Delay of Switch in each Network

3.1.3. Scheduling Delay

Scheduling delay is also called queuing. The packets are transferred out of the device in a particular way from the output port for the transfer to the next target. The capacity of the links are important during this transfer. The packets which can't be sent when link capacity is full. At that time it is kept in buffer for a while but the buffer capacity is also limited. The delay which occurs in this period is called scheduling delay [5,23].

Scheduling Delay = Size of Packet (bits) / Transmission Rate (bps)

3.2. Jitter

It is assumed that a transmission route cannot occur without a delay. A constant system delay is desired. Especially in online applications this value is very important [4,5].

In a network while packet A is arriving at the destination in 15 ms, and packet B arrives the same destination in 18 ms. The jitter in this network is 18-15=3 ms [4, 5, 23].

The Jitter Formula = [D(i) – D(i-1)]

3.3. Packet Loss

It is one of the important parameters of the quality of services. During the transmission of data on network, some packets sometimes is not arrived at the destination because of the some problems occur in network. In this case, the reason may be the packets are dropped. And this missing rate of the packet can be critically important according to type of data carried. Especially in online applications this value is important [24, 5].

Packet Loss rate is calculated such as.

Packet Loss=Number of Packet Loss / Number of Packet Expected

3.4. Bandwidth and Throughput

Another parameter is the bandwidth. This value forms the boundary of data sending on same transmission route in the same time. This is the amount of data transmitted in a second [9].

Throughput gives the amount of data that a device can run in a particular time interval [5, 24, 25].

CHAPTER 4

QOS

To understand how to use Qos applications in practice, we are going to analyse policies about Qos applications on data Networks which exist in TT Incorporated Business.

In data communication while the data is being transmitted from one point to another, as a result of measurement of some values are called Level Agreement, the quality of the used service is exposed. These values are one of the results of Qos policy that are fully applied [26,27].

4.1. Qos Applications of X.25 in Turkish Telecommunications (TT)

No Qos mechanism is applied for x.25 circuits by TT. The data network x.25 in which packet switching technology is used firstly. It is a protocol type that is preferred for lower speed. Each x.25 circuit, which uses this protocol that is mostly preferred for 9.6 Kbps links, is transported from the same situation, there is no superiority of one on any other.

4.2. Qos Applications of X.25 in Turkish Telecommunications (TT)

As seen in Figure 4.1,in TDM Backbone that belongs to TT as we told in teoric part, the common usage of the sources can’t be said because of making the capacity sharing as much as each user’s demand in Time Division Multiplexing (TDM) rule. So it doesn’t need Qos applications.

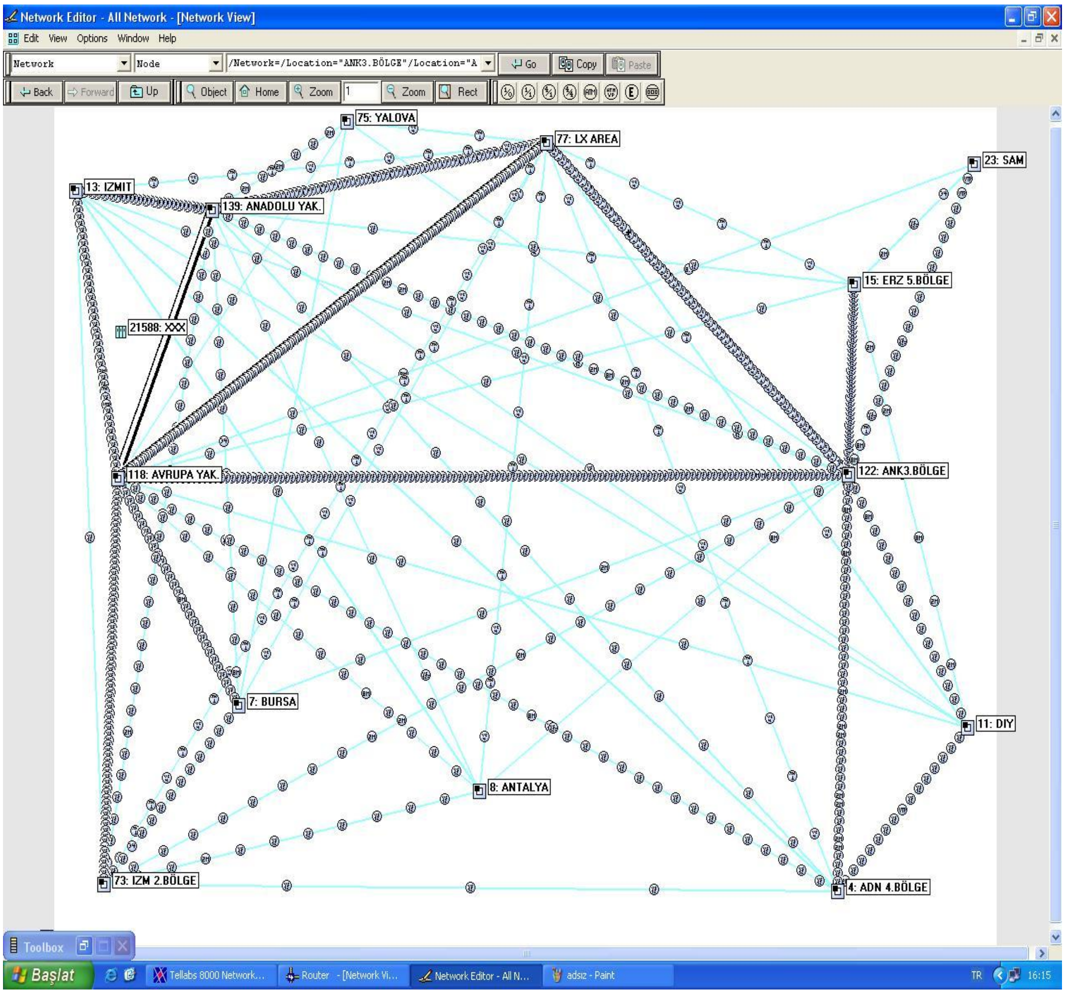

Figure 4.1: TT TDM Network Topology

4.3. Qos Applications of Frame Relay in Turkish Telecommunication (TT)

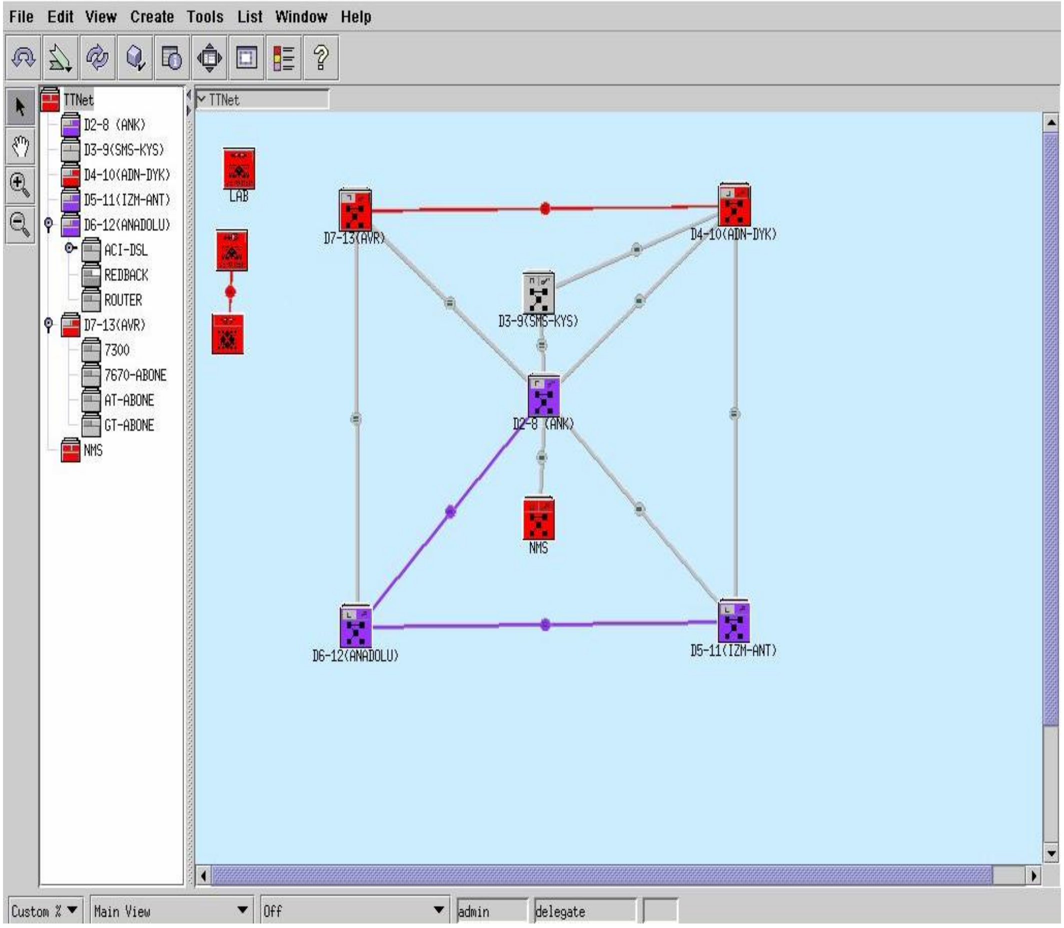

Figure 4.2: TT ATM and Frame Relay Topology

As seen on Figure 4.2 is an outlook of the Backbone of TTNET which performs Frame Relay and ATM services of TT. Each visible domain has seven pieces totally and under each of these there are frame relay and ATM switches. In order to manage properly specific areas are defined. One of these, Ankara domain is seen in Figure 4.3.

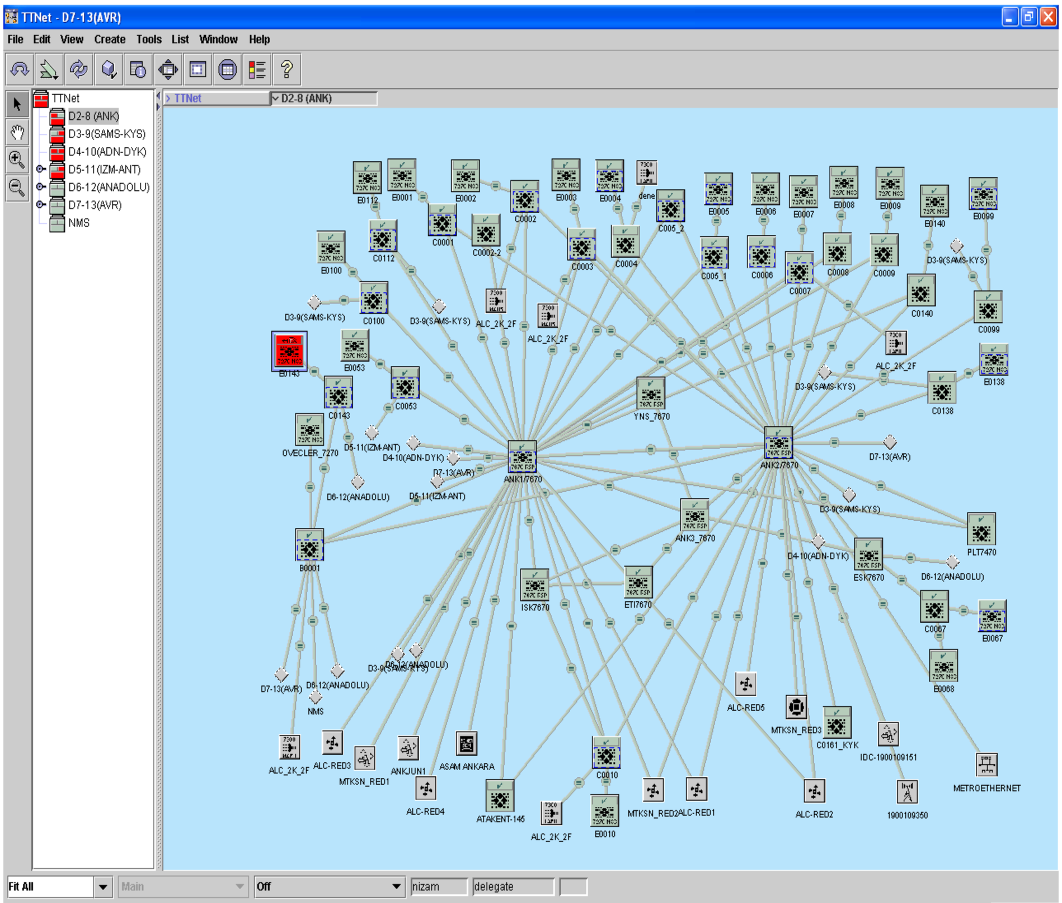

Figure 4.3: Ankara's Domain of TT ATM

While frame level services are provided from this Backbone, it is connected to the TT's Frame Relay after client's device. The whole traffic which reached the Frame Relay, being transformed into cells, is transported in ATM cells up to TT's Frame Relay device which is in the other point it reach. It is transported in ATM cells up to last switching device in the Backbone. The packets have been transformed again into Frame Relay here, are forwarded to customer's device.

As it is understood here, TT hasn't got an infrastructure which works Frame Relay from beginning to end. It is seen as a service in ATM infrastructure.

However Frame Relay Qos parameters are enabled. These parameters are defined in accordance with the customer's demand. To explain on a customer circuit defined by using these parameters:

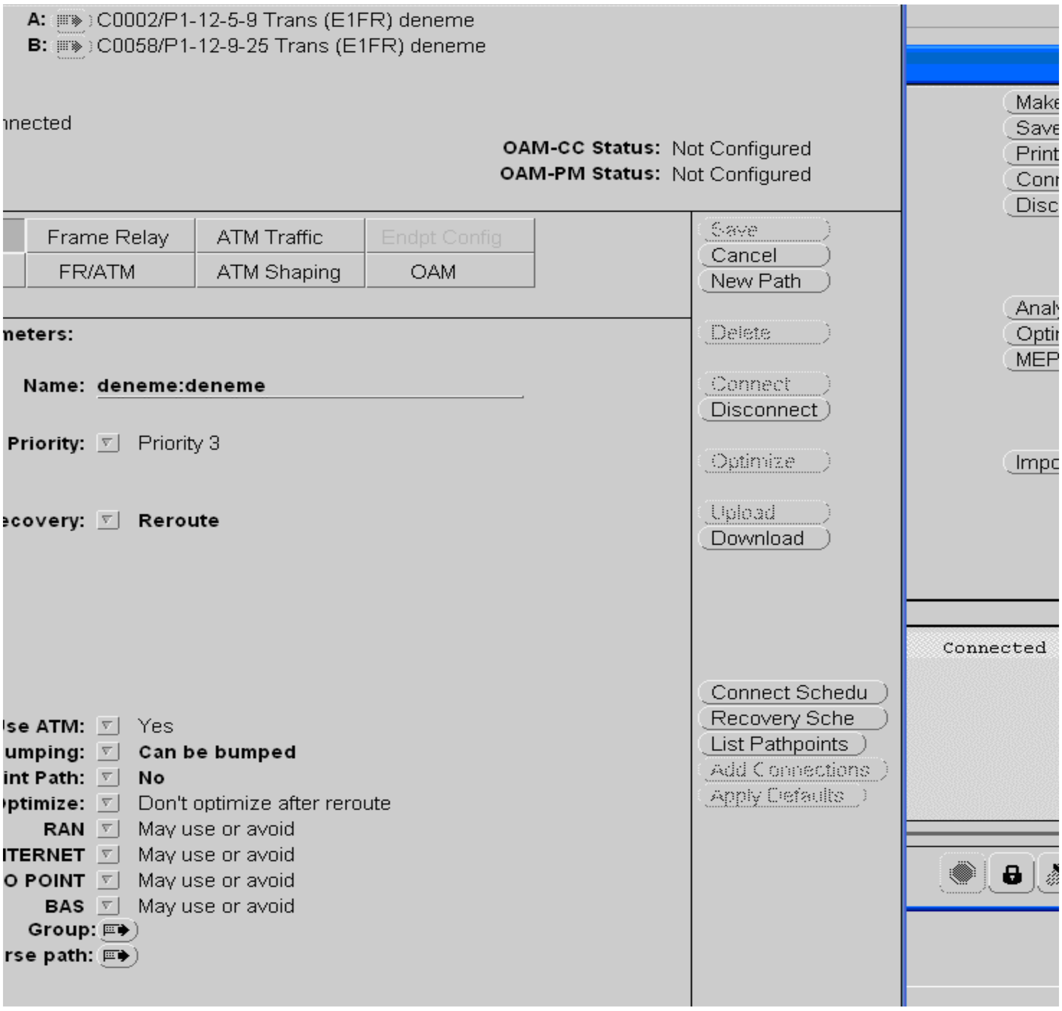

Figure 4.4: Example of Qos Configuration in Frame Relay Network in General

A Frame Relay path has been installed from A point to B point. It is seen as priority 3.For each packet which comes into Backbone independent from Frame Relay protocol’s characteristics, a priority from 1 to 16 can be given.

The priority of the whole Frame Relay circuits in TT is assigned 3. It is possible to implement different policies by providing of decomposition from the other trafic types in the Backbone. Being reroute of recovery parameter is exactly one of the characteristics of devices of TT. In the end of any transmission or being hard pressed for time, if the problem continues after a period of time, the circuit’s direction to the alternative links is provided by calculating the sources again.

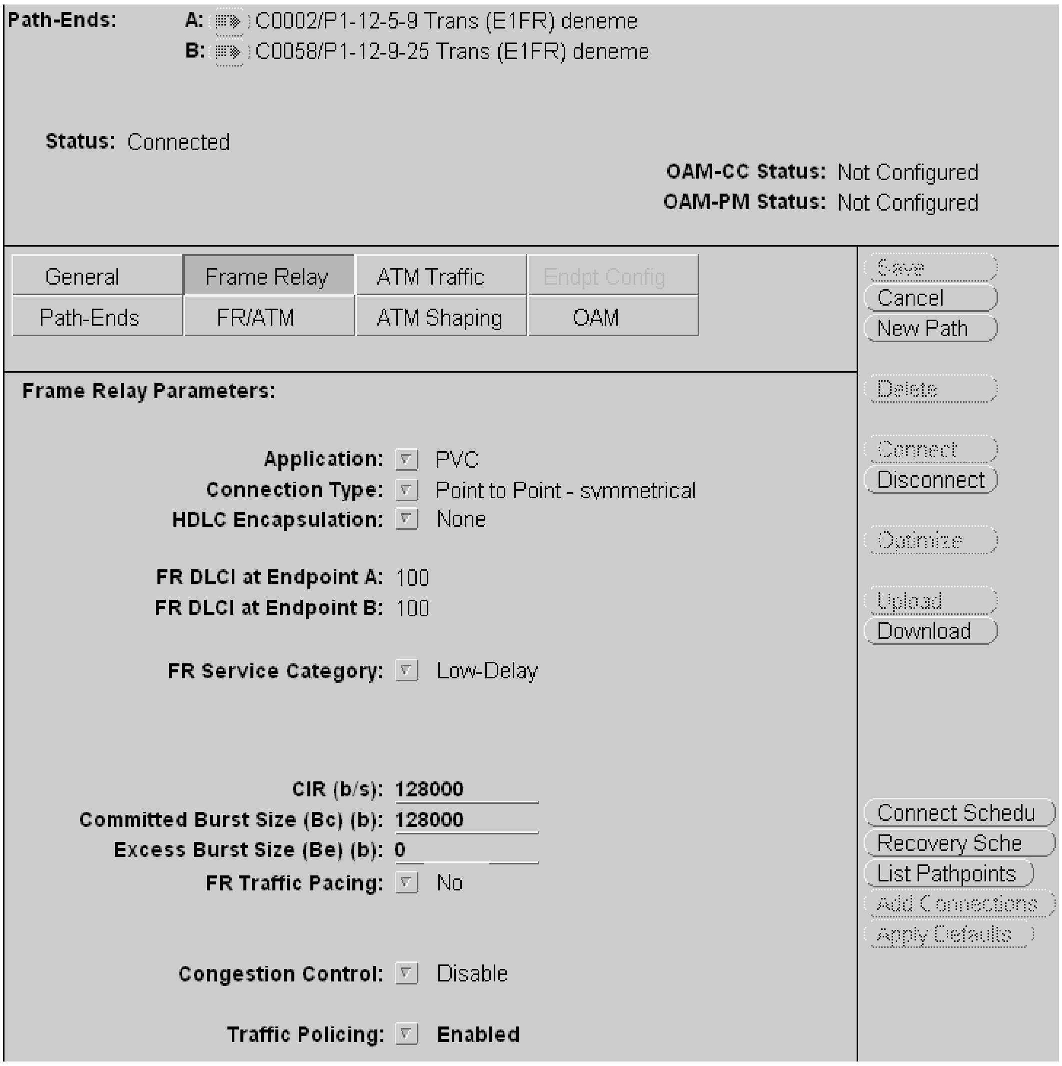

Figure 4.5: Example of Qos Configuration in Frame Relay Network in Frame Relay Tab

If we continue to analyse the same circuit, DLCI is defined as 100 for both ends. It is stated in the definition that there should be low delay as Service Category. The speed to be guaranteed is defined as CIR(Bc)=128 Kbps. No value has been defined for the unguaranteed speed EIR (Be)=0. And this shows that customer's need for bandwidth will be 128 Kbps total.

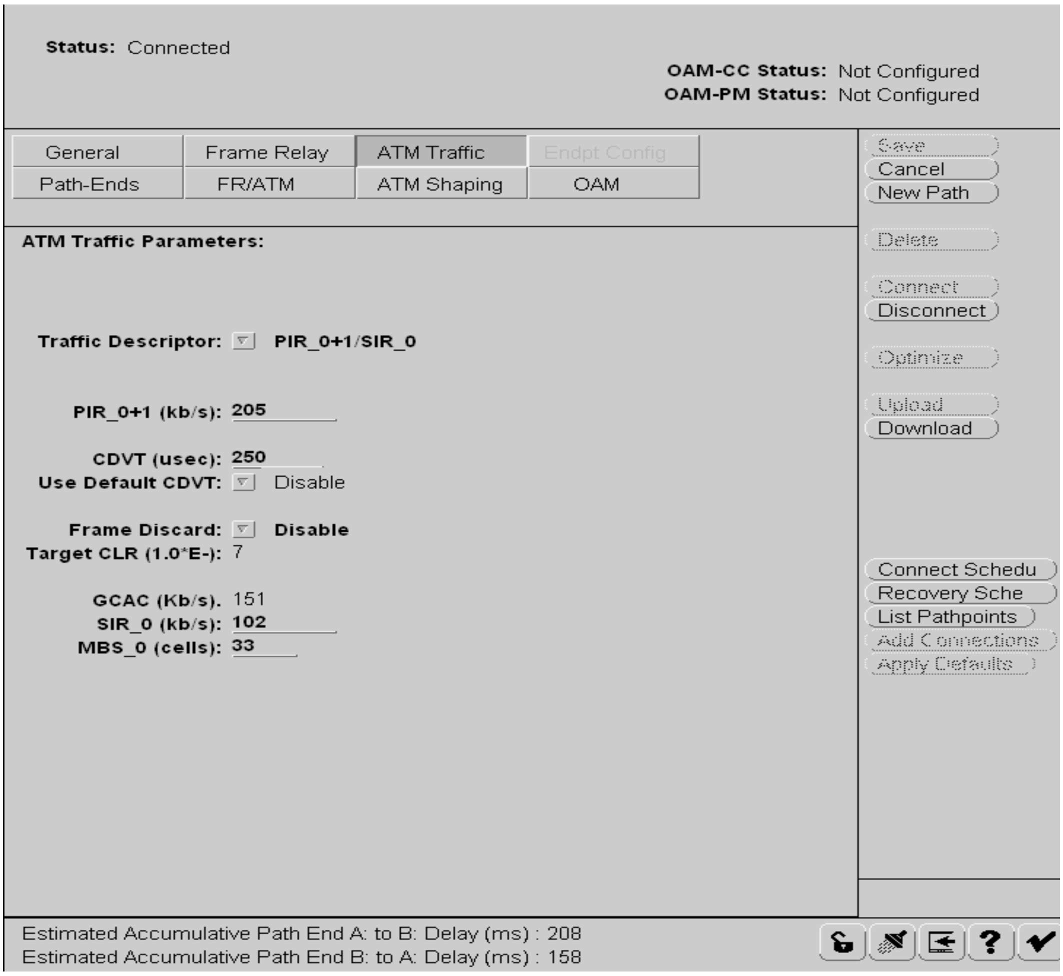

Figure 4.6: Example of Qos Configuration in Frame Relay Network in ATM Traffic Tab

As I mentioned before, packets formed for frame relay circuit, after some points it will have to pass through ATM devices. That's why, some ATM parameters should be determined for this circuit. The first of these is Cell Delay Variation Tolerance (CDVT). For this 250 ms is defined. The Frame Discard is disabled informs us that this circuit is more important than the others. At the very bottom of Figure 4.7, the delay of the virtual circuit from A end to B end and from B end to A end is computed according to the input values. At one side it is identified as 208 ms, while the other is 158 ms.

4.4. Qos Applications of ATM in Turkish Telecommunications (TT)

On TT ATM Backbone, in the order of priorities, CBR, NRT-VBR and UBR traffic types are used. UBR traffic type is usually preferred on the circuits used for connecting Digital Subscriber Line Acces Multiplexer (DSLAMs) bringing devices of Asymetric Digital Subscriber Line (ADSL) users together to the Backbone. Because of this reason, the priority of Internet traffic is always at the very end. For the point to point connections NRT-VBR is used, and for the circuits that have no strength CBR is used. If it is analysed the ATM circuit definition, it is seen on tha Figure 4.8 that A and B physical interfaces are connected to each other with an ATM pvc doing Here, A end is addressed as VPI=0 VCI=255, other end B is addressed as VPI=0 VCI=18. These values should be considered while setting the customer interfaces at end points.

The traffic type is setted as nrt-vbr shows that this circuit is setting for only data transportation not for synchronous communication.

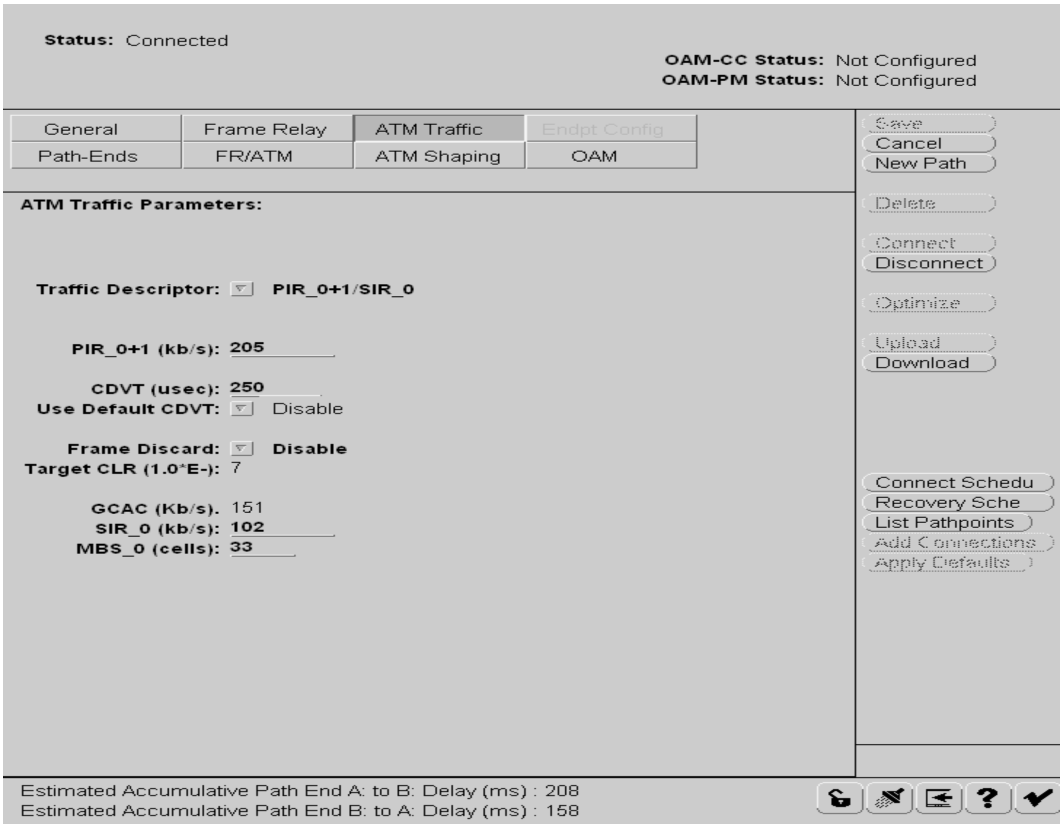

Figure 4.7: Example of Qos Configuration in ATM Network in ATM Traffic Tab

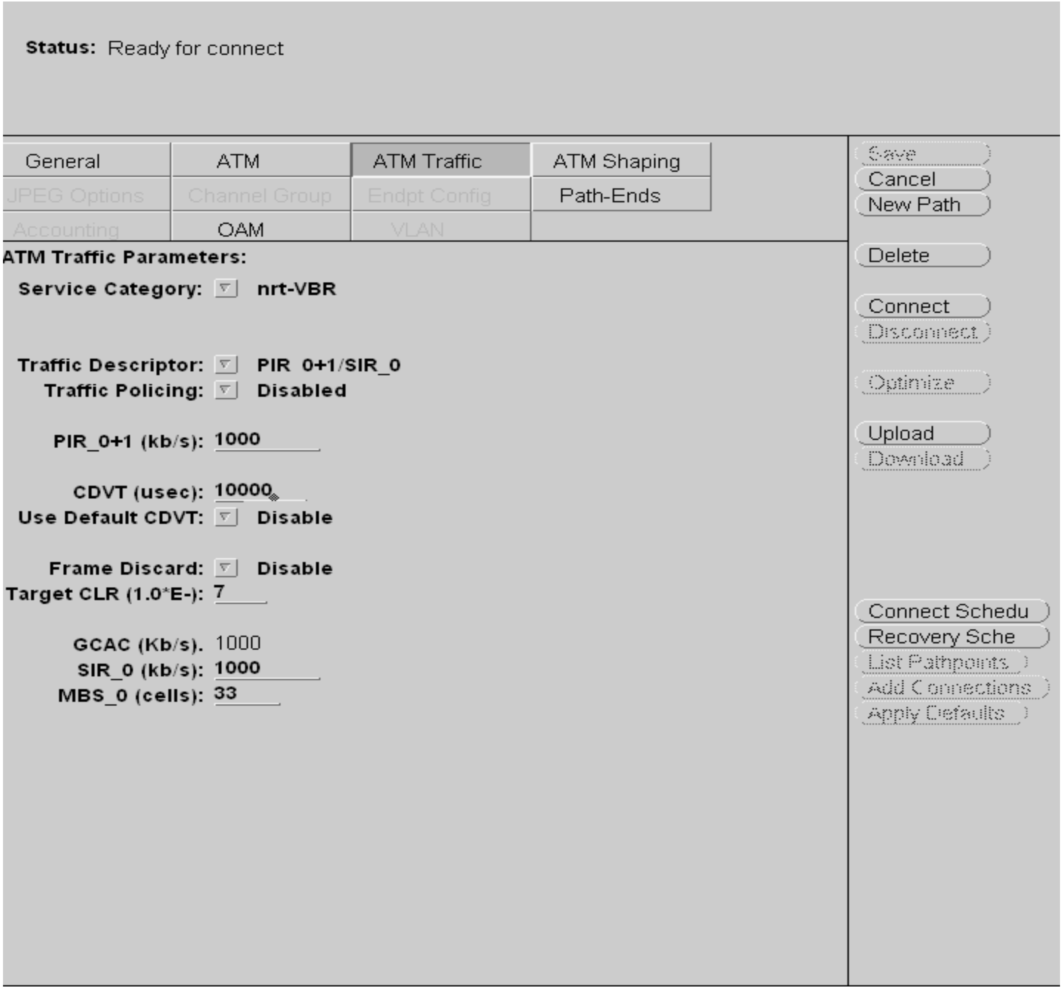

Figure 4.8: Example of Qos Configuration in ATM Network in ATM Traffic Tab

As it is understood from the definition in Figure 4.9, PIR and SIR parameters show that the bandwidth of the circuit is 1 Mbps. That the CDVT value is high like 10000. It shows that data lose is significant for the circuit. Setting these values as high, it will lengthen the period of the cells' waitings in buffer in case of a congestion.

The traffic type is nrt-VBR means that only data will be transported but not an online application on this circuit. CLR is set as 7, MBS value is calibrated as 33 Kbps. If bandwidth is desired to be over 1Mbps during the usage of circuit, by the help of these values, bandwidth may increase up to 1033 Kbps.

In accordance with the Qos policy implemented in TT ATM Backbone, as I have shown above, the priority order of the packets are setting as CBR, NRT-VBR and UBR in turn. The priority order in all switching devices on Backbone is activated like this. According to this priority order, the traffic types and their rates for the total capacity of links are used is obvious.

A circuit definition is made 10 times faster of a link capacity on TT ATM links. And it shows that TT has to use Qos parameters effectively.

4.5. Qos Applications of MPLS in Turkish Telecommunications (TT)

In the structure called TT's IP/MPLS Backbone, the topology in Figure 4.10 has been developed using the T-640 model router device of Juniper company and 7750 and 7450 service router device of Alcatel company.

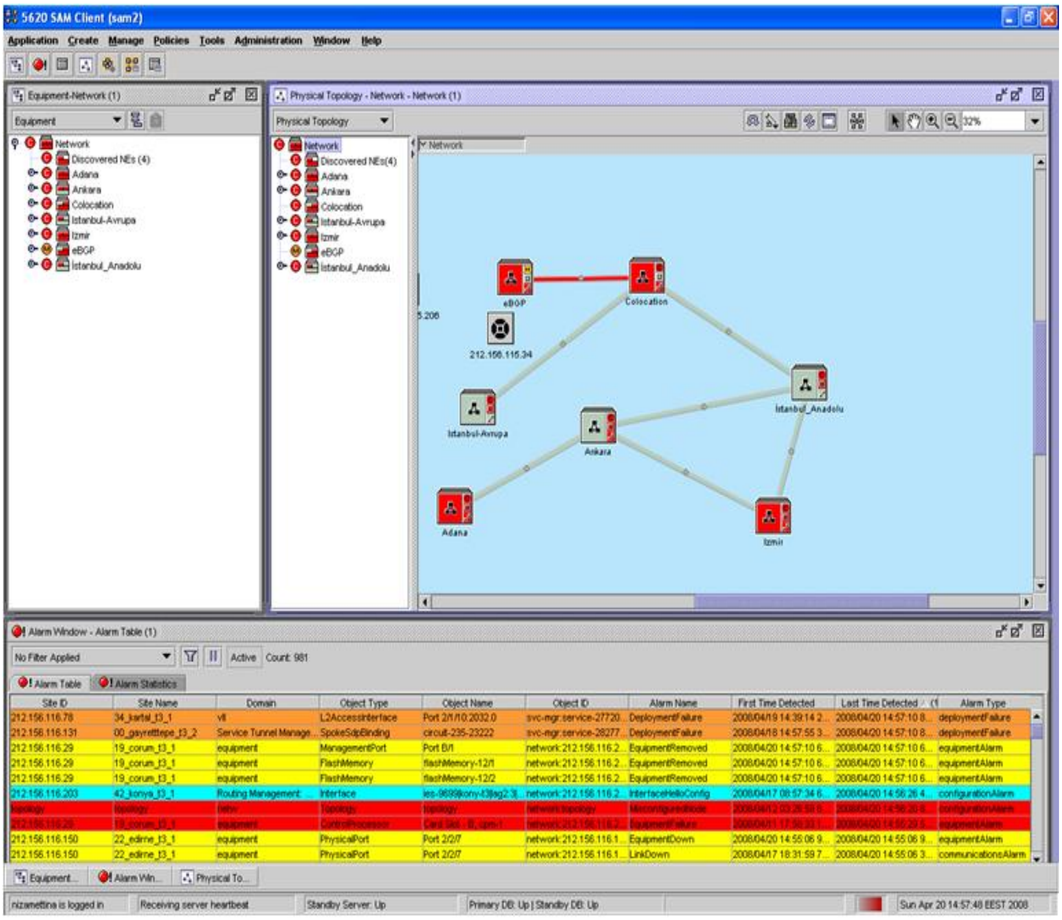

Figure 4.9: TT IP/MPLS Topology

Frame Relay and domains like ATM Backbone have been developed. There are devices in the zones an example of which is in Figure 4.11.

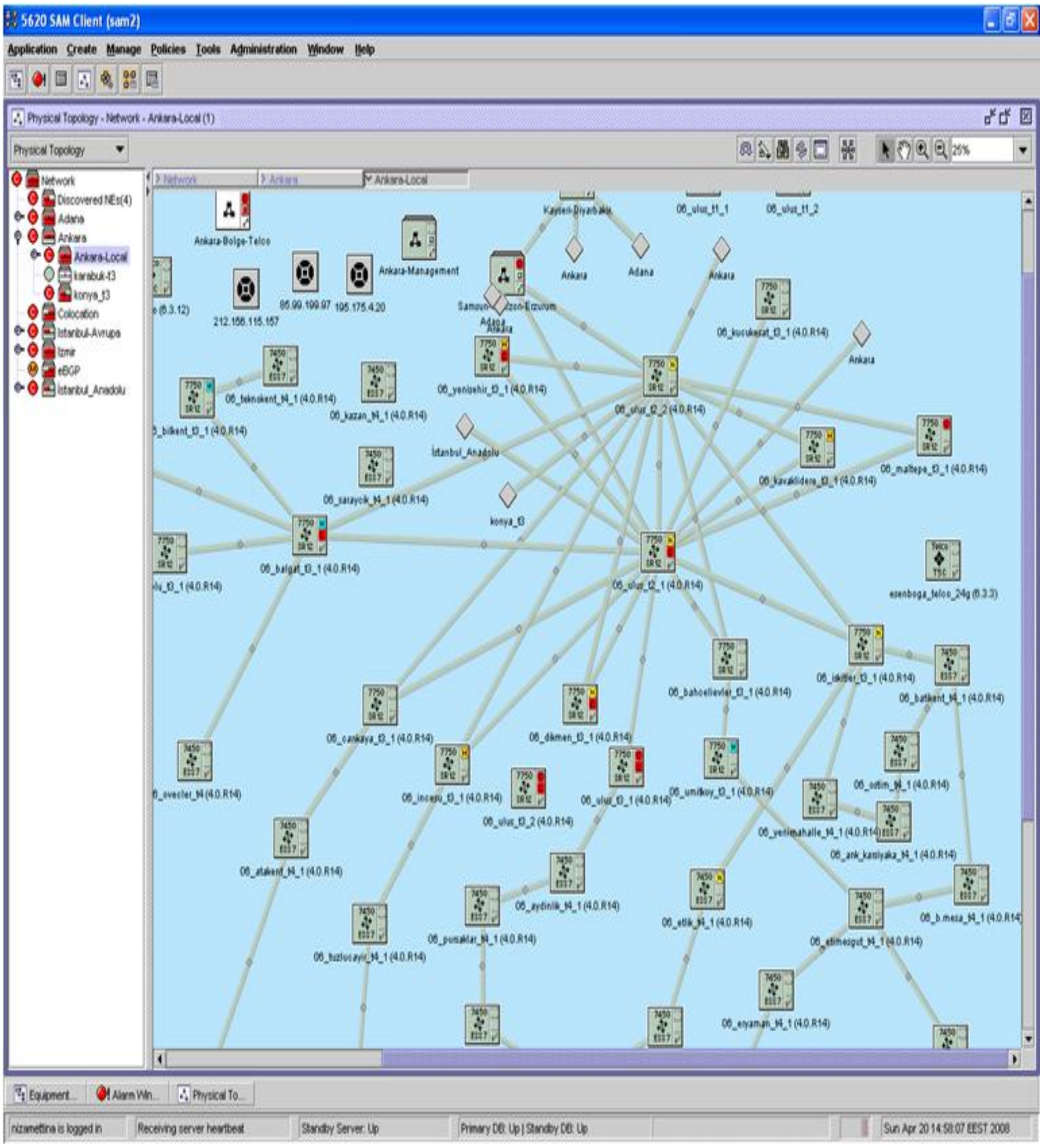

Figure 4.10: Ankara's Domain of IP/MPLS Backbone

On TT IP/MPLS Backbone, Qos policies are used according to the user. The customer has a chance of end-to-end Qos application between the two points to be connected, covering the user's local networks. To show more obviously, suppose that A server is used for voice and a server B for internet.

The traffic of server A can be prioritized at every point it passes. First of all, all the packets sent by the customer, starts with the marking of each packet by using specific parameters Figure 4.12.

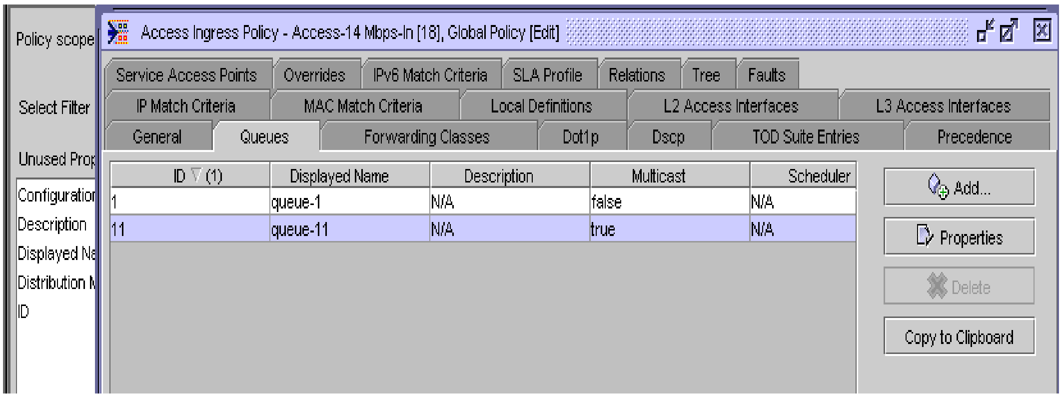

Figure 4.11: Classification of Traffic of Parameters

The traffic coming from server A is prioritized as to the traffic coming from server B, by using the DSCP bits on the device at the customer's side. As a result of this marking, packets reached to the Backbone are classified Figure 4.13. Here the prioritized traffic DSCP bits sent by the customer, has been marked as assured forwarding af11 and sent. Likewise, the traffic outgoing from Backbone towards server has been sent as af12.The traffic from server B and again to this server outgoing from Backbone is sent as best effort (be).

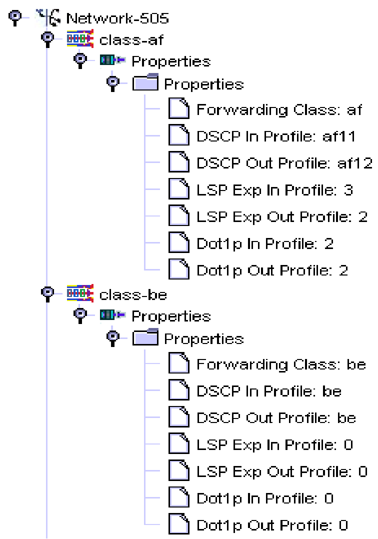

Figure 4.12: DSCP Matching

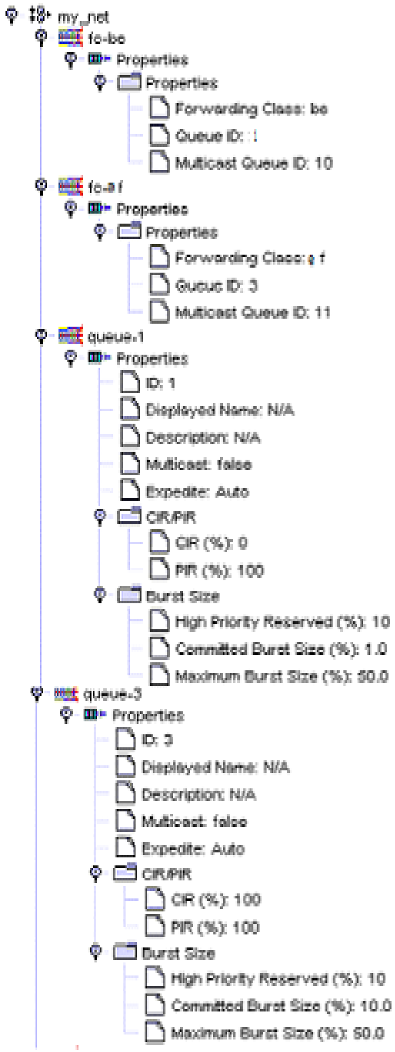

Figure 4.13: Classification of Traffic

The packets on the virtual route will be sent using queuing and scheduling mechanisms over links of specific capacity limit, for passing to the next hop point. As is understood from the configuration shown in Figure 4.14, while unprioritized internet best effort traffic uses queue 1, the other guaranteed assured forwarding traffic uses queue 3. Again if looked at the configuration in Figure 4.14, the prioritized traffic in queue 3 will be sent through the queue using a higher bandwidth than the traffic in queue 1.

The same Qos policy is implemented on the devices which is running over IP/MPLS Backbone.

The Qos definitions for the Backbone are global. These defined global values may be privatized according to the customer. An important feature of the Qos mechanism used in TT IP/MPLS Backbone is a seperate chip is used for the prioritized traffic while packets are processed even on the same device.

By the help of what mentioned above, It is possible to give Qos service from one end to other between two points by the help of TT IP/MPLS Backbone.

CHAPTER 5

MEASUREMENT OF QOS PARAMETERS IN TURKISH TELECOMMUNICATION IN DATA NETWORKS

In this chapter, an analysis will be done concerning the Service Level Agreement (SLA) values, measured dependent on Qos parameters, in the IP/MPLS Backbone of TT. The measurements are made in Backbone in order to calculate the Round Trip Time (RTT) packet lost values, by the help of a software jitter which works Simple Management Network Protocol (SNMP) based between Ankara-Van, Ankara-Dikmen, Ankara-Konya, Ankara-Kahramanmaraş devices. The edge-to-edge transmission distances measured between these locations chosen on Backbone, approximately, Ankara-Dikmen distance is 10km, Ankara-Konya distance is 300 km, Ankara-Kahramanmaraş distance is 602 km, Ankara-Van distance is 1248 km.

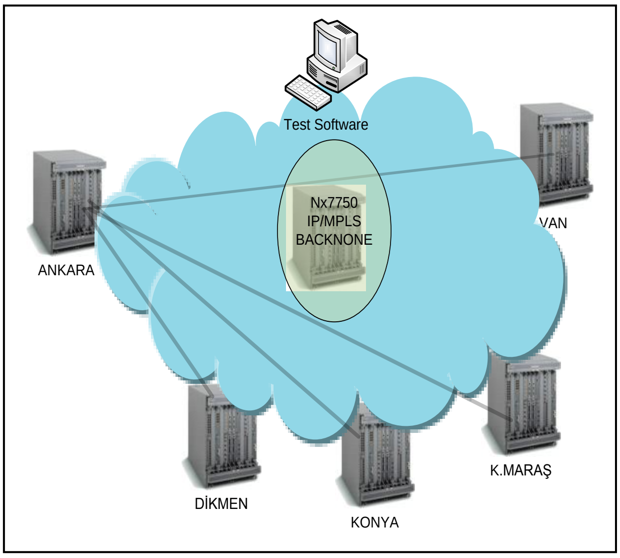

Figure 5.1: Topology for Measurement

In Figure 5.1 the topology is the software called SMART, produced by Alcatel Corporation for TT. By the help of this program, TT measures the SLA parameters that it supplies to its customers. By using this software, when calculating the values between the locations above, the measurement is made in the possible whole period of a day, in order to reduce the error rate. In all measurements, the orientation device in the Ulus location of TT Corporation is chosen as source, the other points are as targets. In each time of test, two ping packets are sent from source to target as 1500 bytes and 4000 bytes. By calculating the outgoing and incoming durations, data obtained has been transferred to graphic interface. The packets fragmented as 1500 bytes by the software called SMART, are sent to different targets through The connection of network server which smart software runs is 100 Mbps. After ping command is sent from source to target, an acknowledgement is sent by the receiver when packet reachs the target. The measurement value is taken, when this acknowledgement info is detected at first span by the source device packet sent. By this way, ping packets of 1000 items are sent in the whole periof of day. The average of the values are calculated for each packet group is taken. Likewise, same values are measured by using packets of fragmented as 4000 bytes.

At the time of measurement, all values are, for three different traffic types, defined on TT devices, and computed. These traffic types, best effort (be), is traffics of minimum quality, which can be called insignificant internet traffic. The traffic type defined as explicit forwarding (ef) is regulated for traffics of medium quality. Assured forwarding (af) traffic type is the one used for traffic transmission of most priority on Backbone.

In graphics, horizontal axis shows the time of measurement, vertical axis represents the values taken during the measurement in the mili-seconds. Apart from these, in the measurements concerning packet loss, while the packets lost is shown on vertical axis, measurement time is shown on horizontal axis.

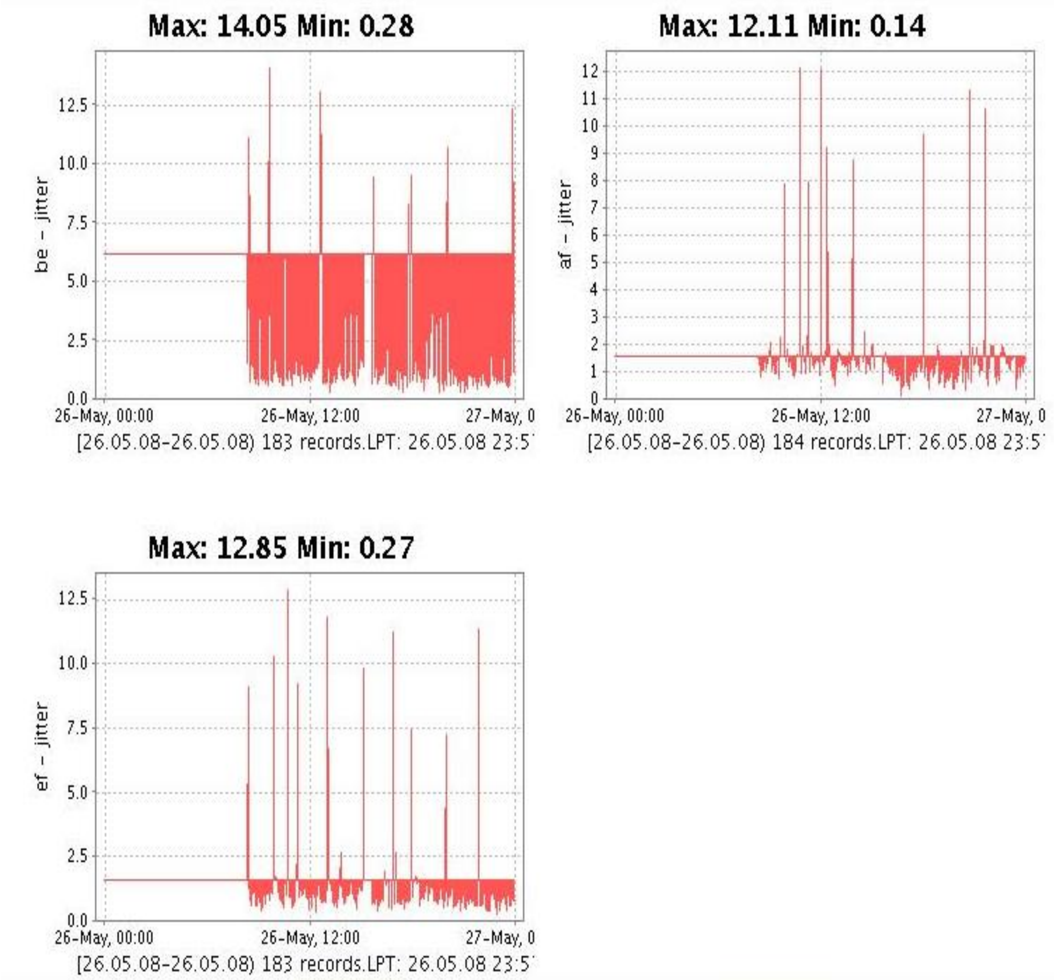

Figure 5.2: The jitter graphic measured between Dikmen and Ulus for 1500 Bytes

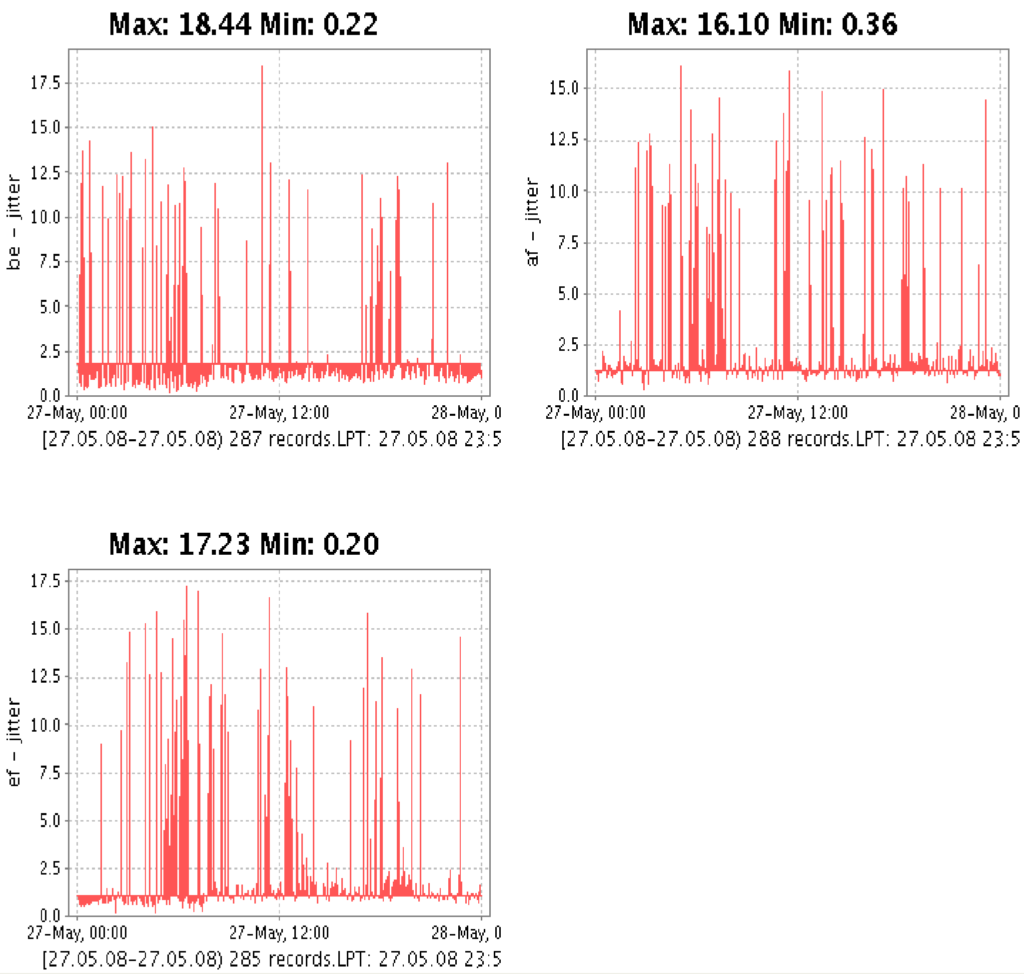

Figure 5.3: The jitter graphic measured between Dikmen and Ulus for 4000 bytes

In Figure 5.2, the graphic concerning the jitter value measured between Ulus-Dikmen distance, the shortest distance, is drawn by SMART software In this graphic, horizantal axis means the time interval where measurement is taken. As seen in horizantal axis, the beginning time of the test is at 06:00 pm on 26th May 2008. However it is not understood clearly from this graphic. Because of the not interfere to the current software, this negative view can not be fixed. But, it is understood from tha horizantal axis that the values are obtained after 12 hours of measurement. Also, another problem that can not be fixed in the graphic is the span graphic started to be drawn.

The drawing starts as soon as the graphic detects the first value. That's why, the contitions when cannot be are taken any measurements are expressed with straight line.

The measurements are made for best effort assured forwarding and explicit forwarding are shown separately. Vertical (y) axis represents the values measured in the type of mili-seconds. When they are compared the values on graphics in Figure 5.2 with ones on graphics in Figure 5.3, it is undersood that packet size is a positive function of Jitter. When compared the two graphics by traffic type, it is understood that values concerning best effort traffic type have less priority as to assured forwarding and explicit forwarding traffic types. The best jitter in these measurements belongs to assured forwarding traffic type.

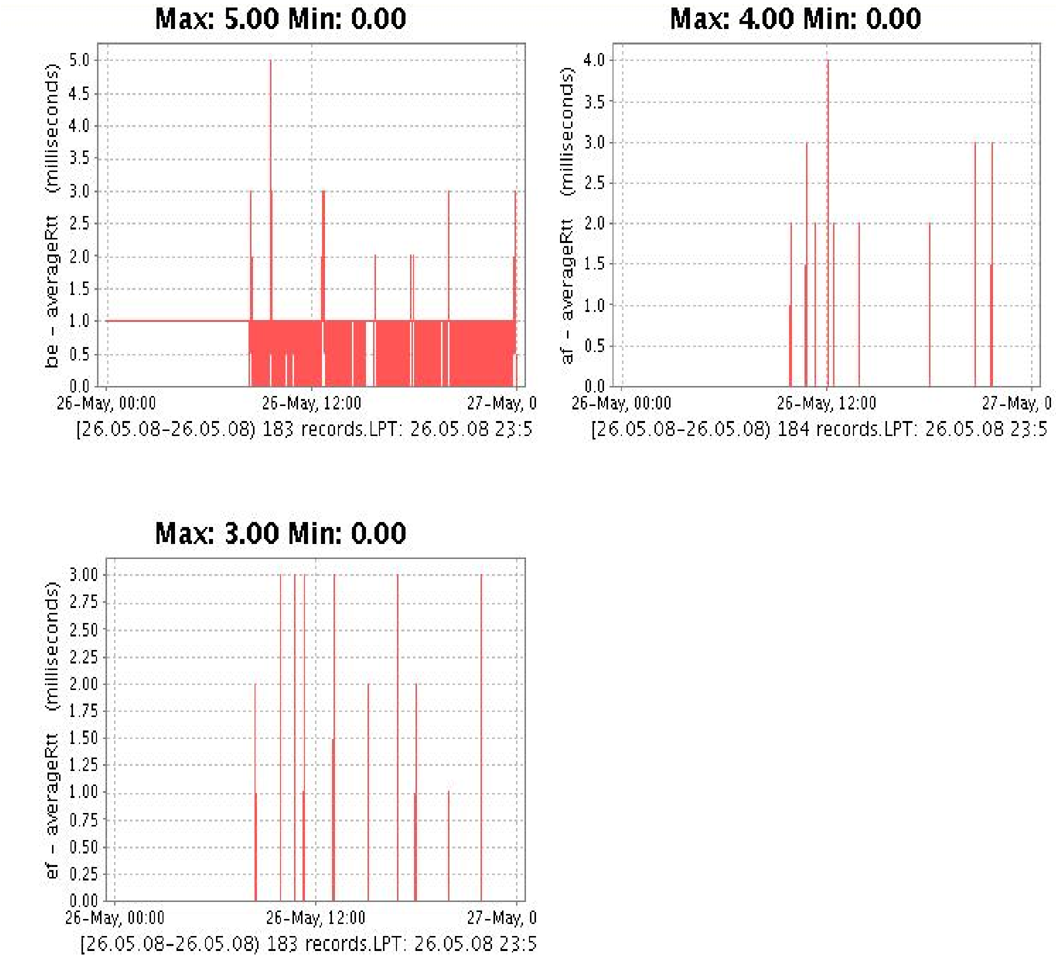

Figure 5.4: The Average RTT graphic measured between Dikmen and Ulus for 1500 Bytes

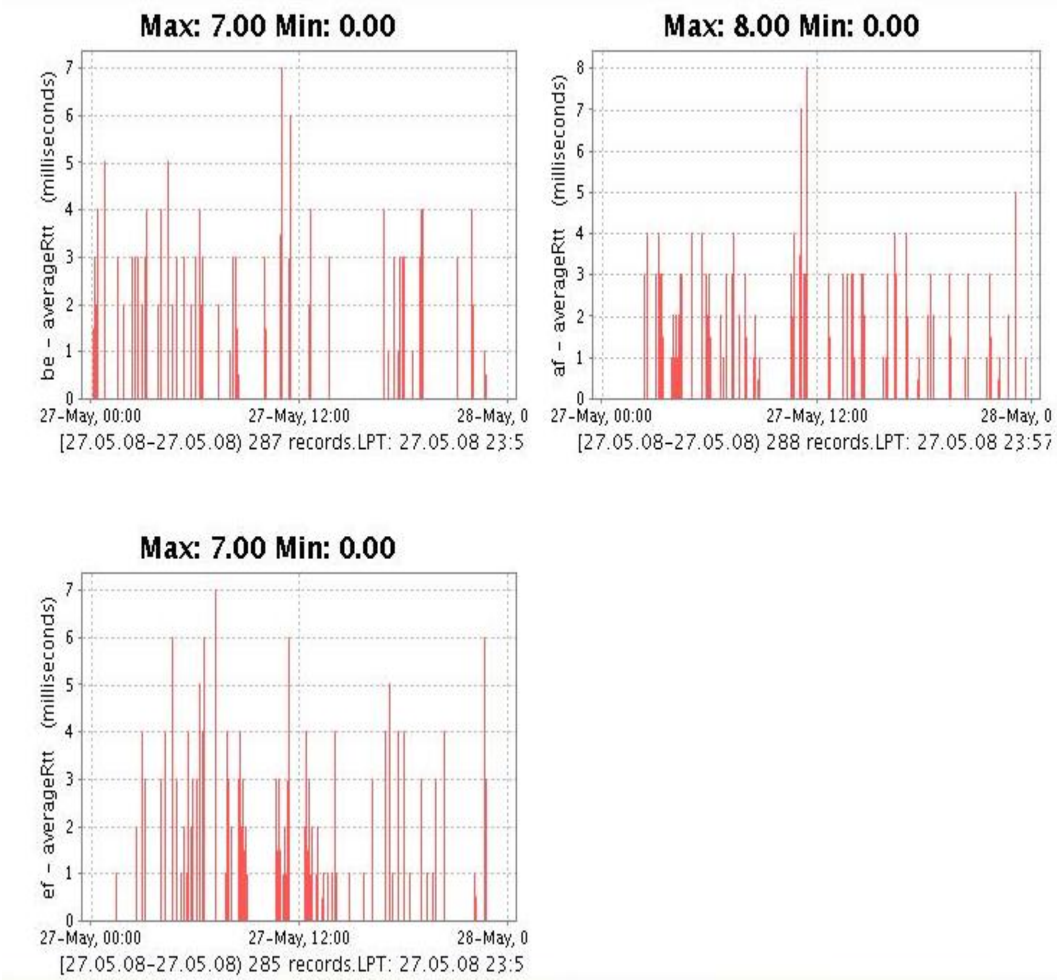

Figure 5.5: The Average RTT graphic measured between Dikmen and Ulus for 4000 Bytes

In Figure 5.4 and Figure 5.5, RTT, which means dual sided delay for two different packet size, is computed and graphics are drawn. As seen here, packet size and RTT value is direct proportion.

However, in this comparison, it seems that concerning best effort for traffic type are the same as values taken from measurements are made for explicit forwarding traffic type. And this shows that all sources are of the same quality for the two traffic types.

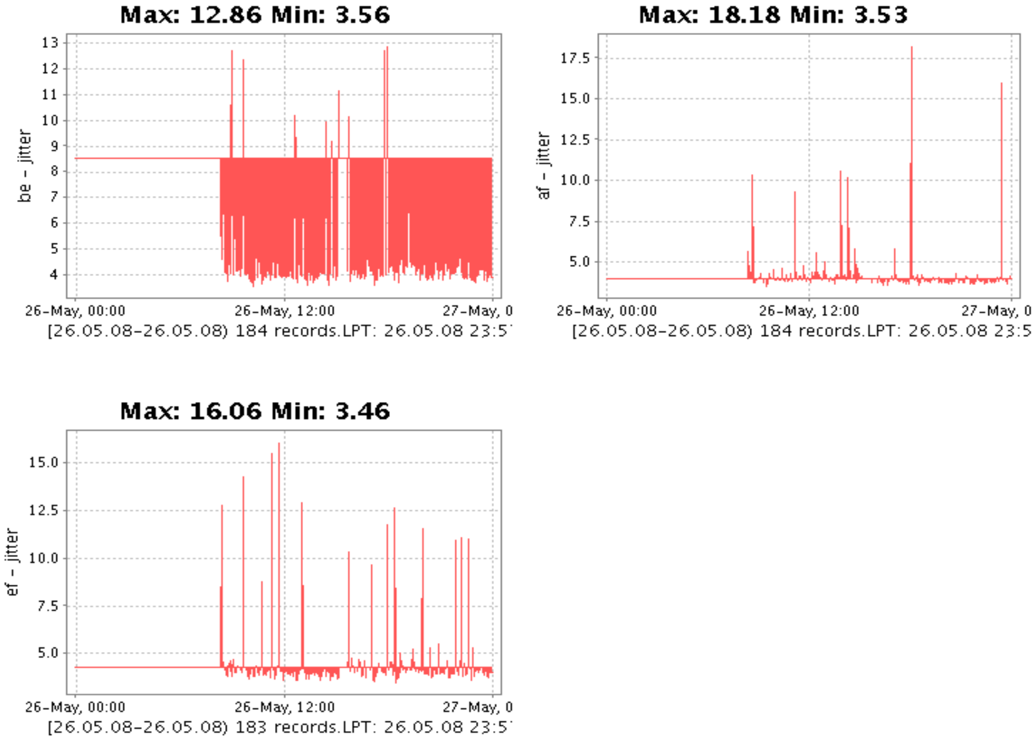

Figure 5.6: The jitter graphic measured Konya and Ulus for 1500 Bytes

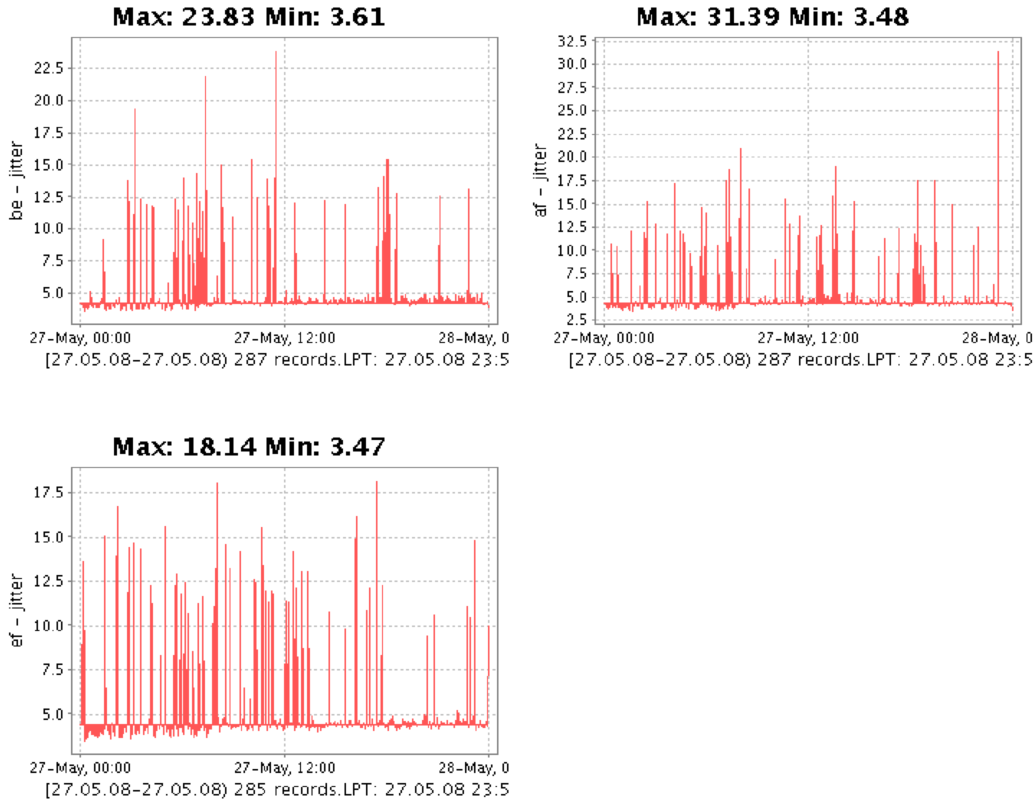

Figure 5.7: The jitter graphic measured between Konya and Ulus for 4000 Bytes

When graphics in Figure 5.6 and Figure 5.7 are compared with the measurements between Ulus-Dikmen, an increase in jitter value is observed because of the increase of transmission distance. Jitter is the positive function of transmission length.

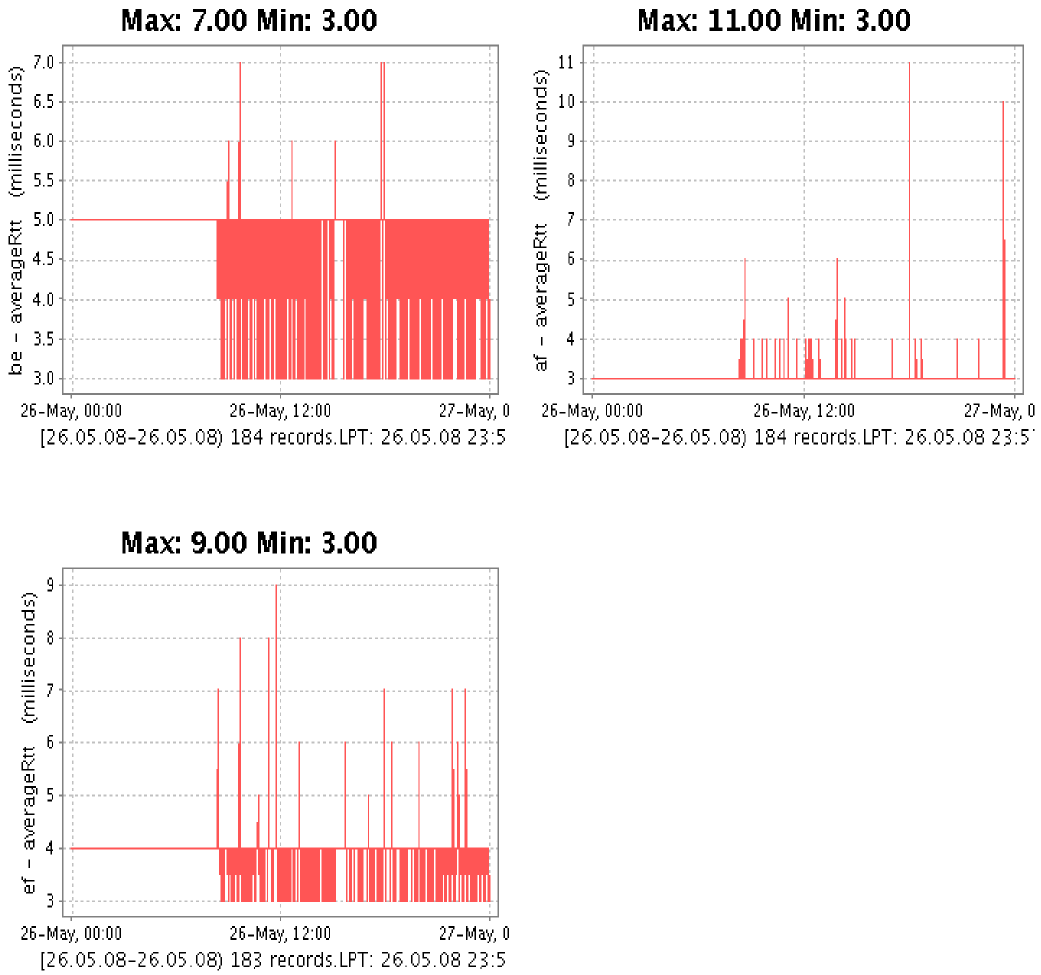

Figure 5.8: The Average RTT graphic measured between Konya and Ulus for 1500 Bytes

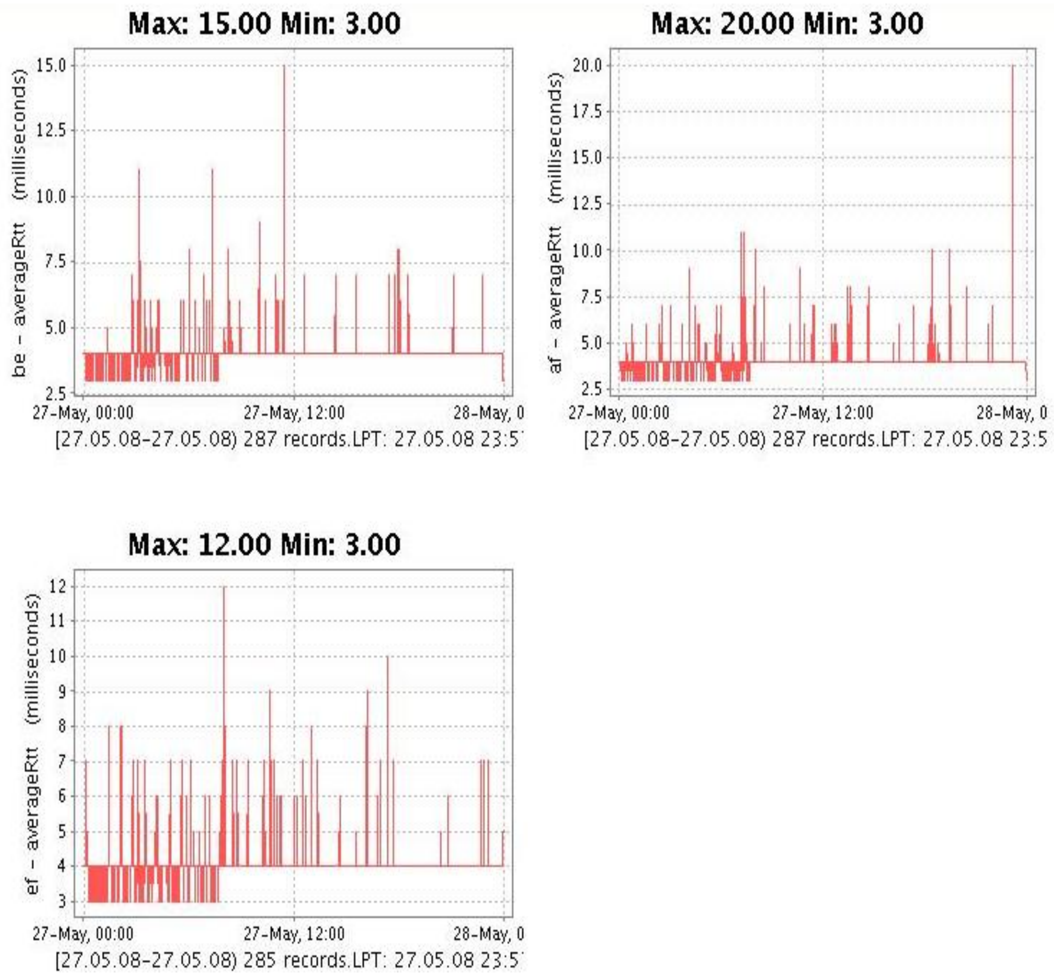

Figure 5.9: The Average RTT graphic measured between Konya and Ulus for 4000 Bytes

When the values of the graphics in Figure 5.8 and Figure 5.9 are compared with the measurements taken for Ulus-Dikmen, it is understood that transmission distance is the positive function of delay.

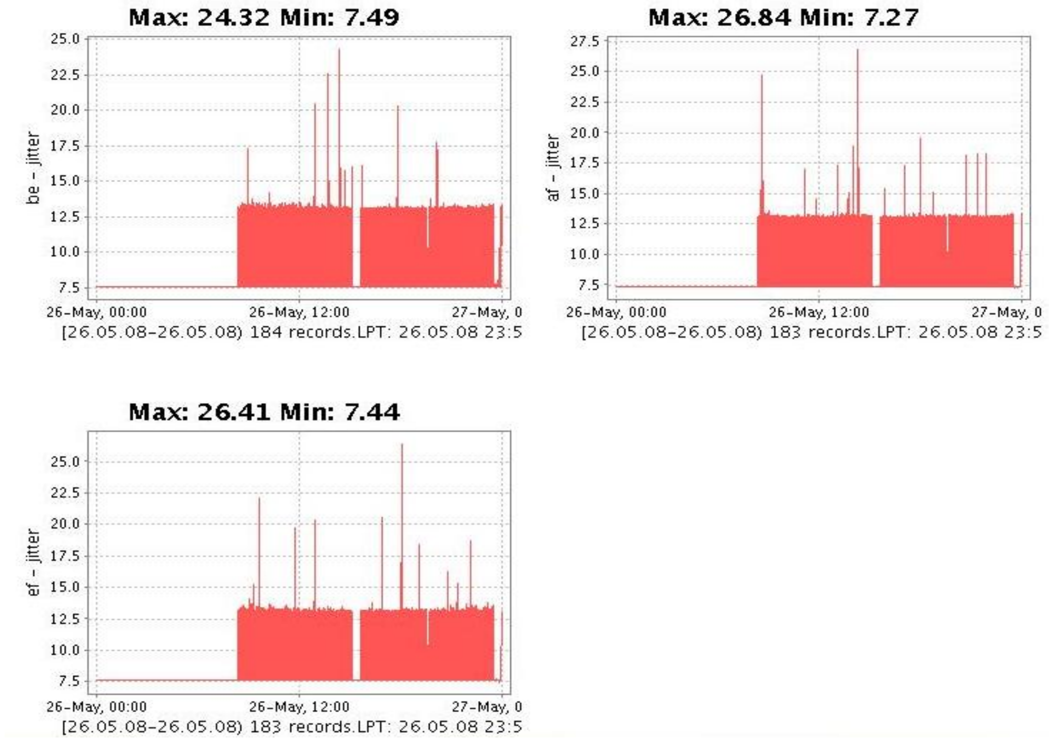

Figure 5.10: The jitter graphic measured between Kahramanmaraş and Ulus for 1500 Byte

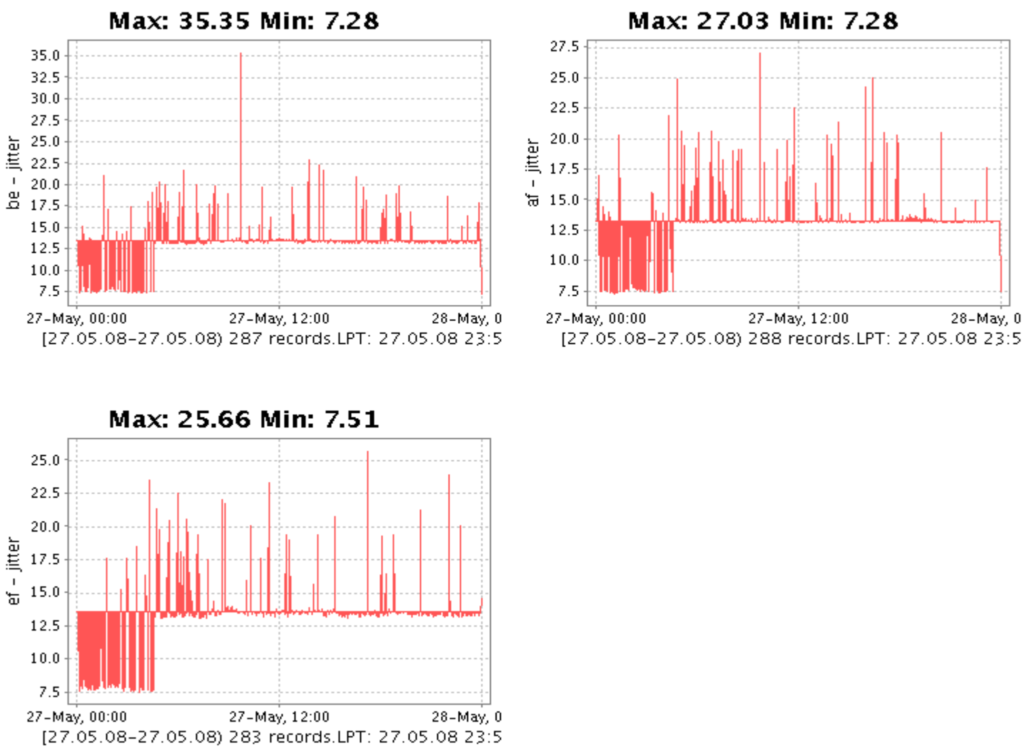

Figure 5.11: The jitter graphic measured between Kahramanmaraş and Ulus for 4000 Bytes

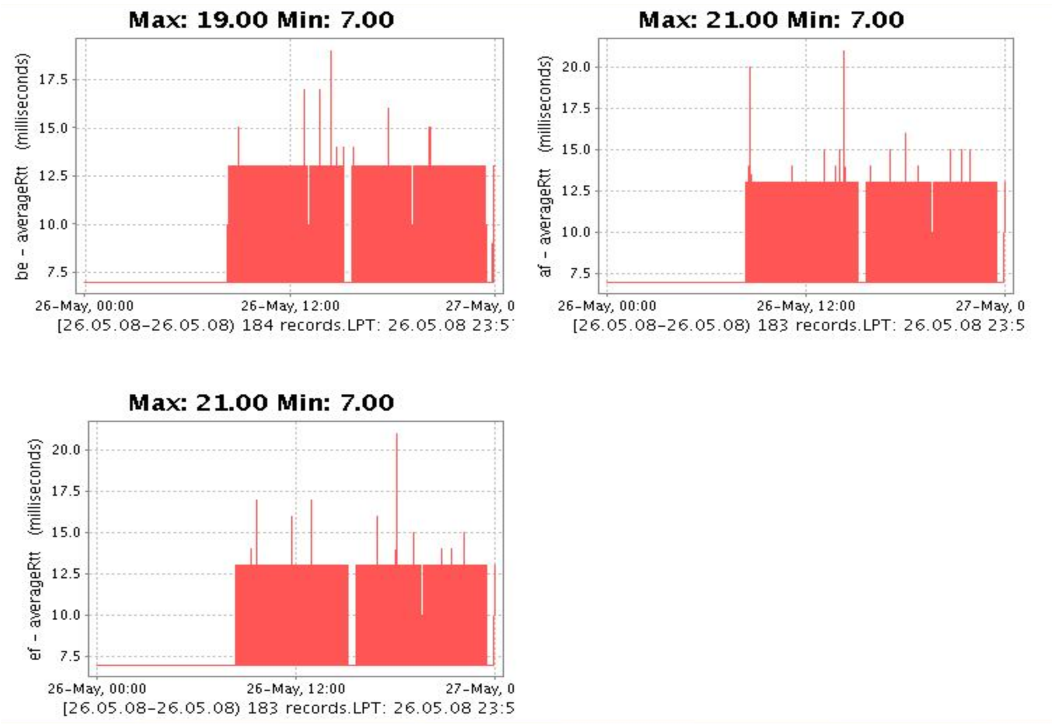

Figure 5.12: The Average RTT graphic measured between Kahramanmaraş and Ulus for 1500 Bytes

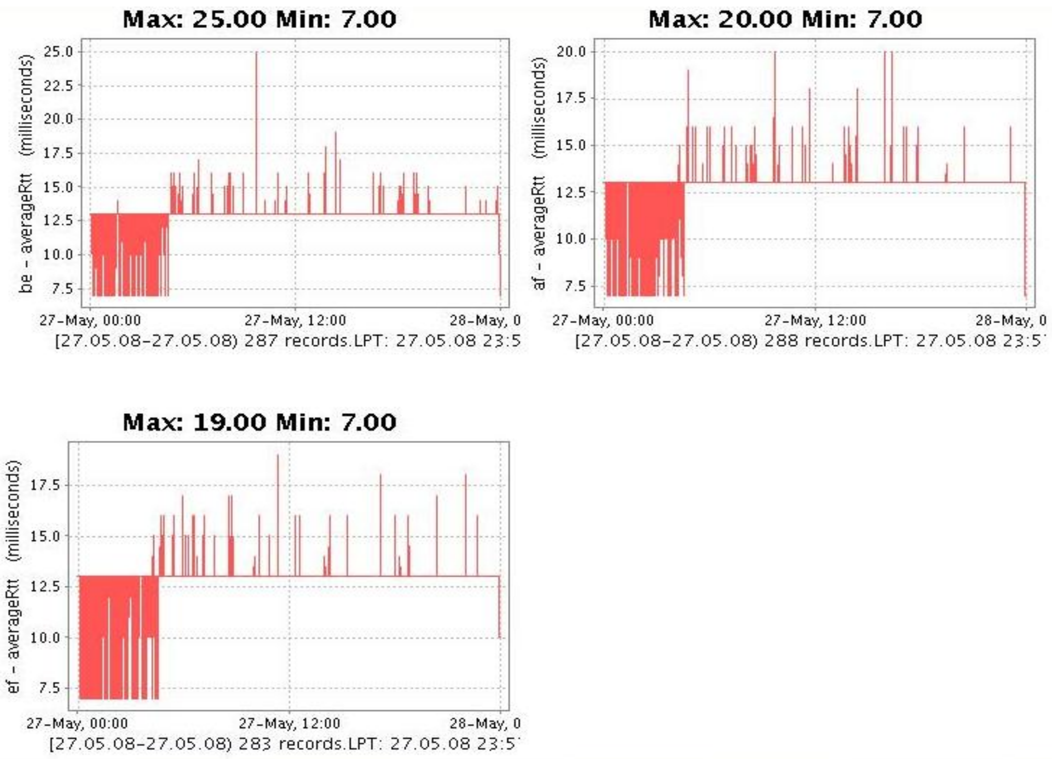

Figure 5.13: The Average RTT graphic measured between Kahramanmaraş and Ulus for 4000 Bytes

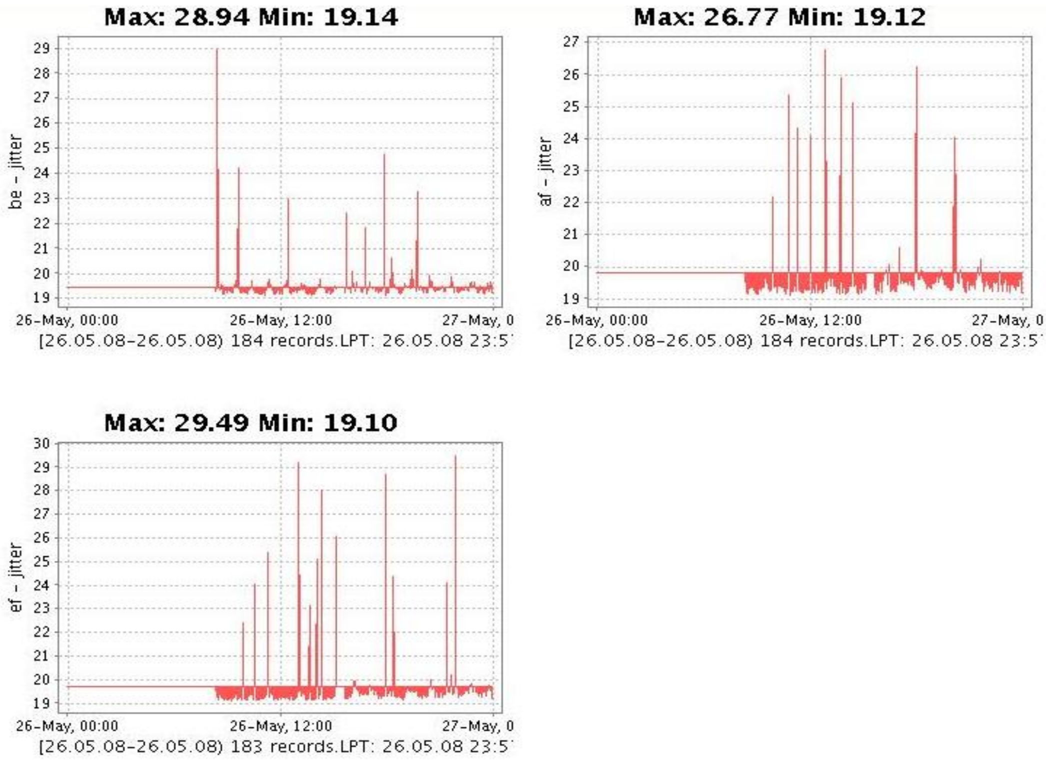

Figure 5.14: The jitter graphic measured between Van and Ulus for 1500 Bytes

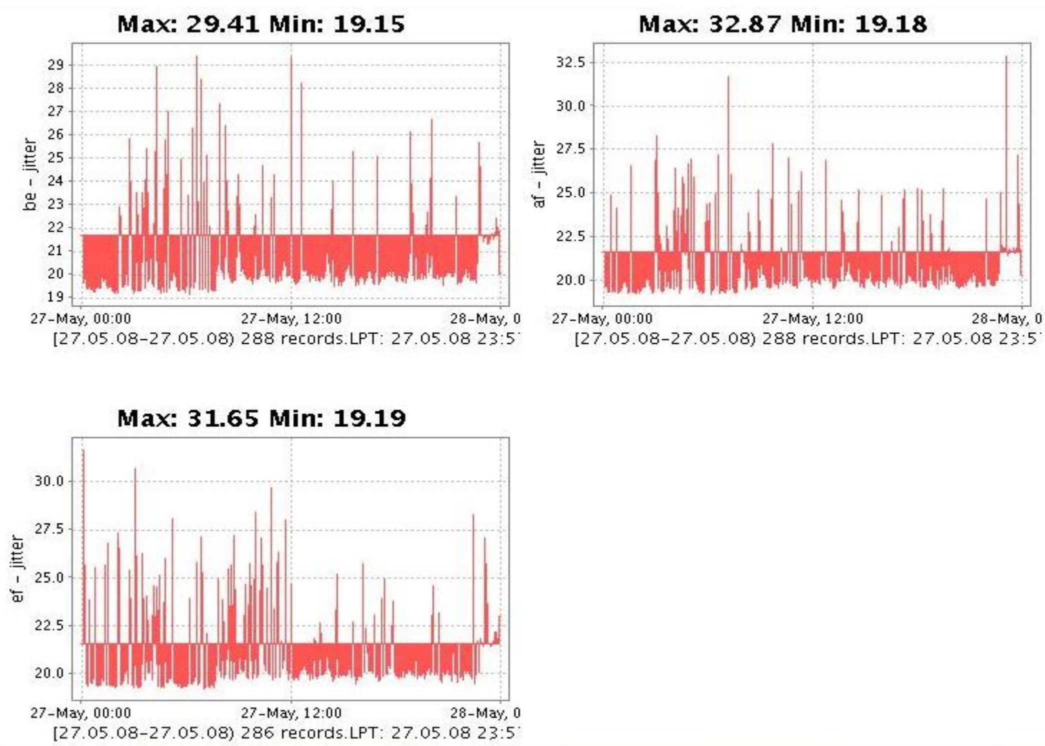

Figure 5.15: The jitter graphic measured between Van and Ulus for 4000 Bytes

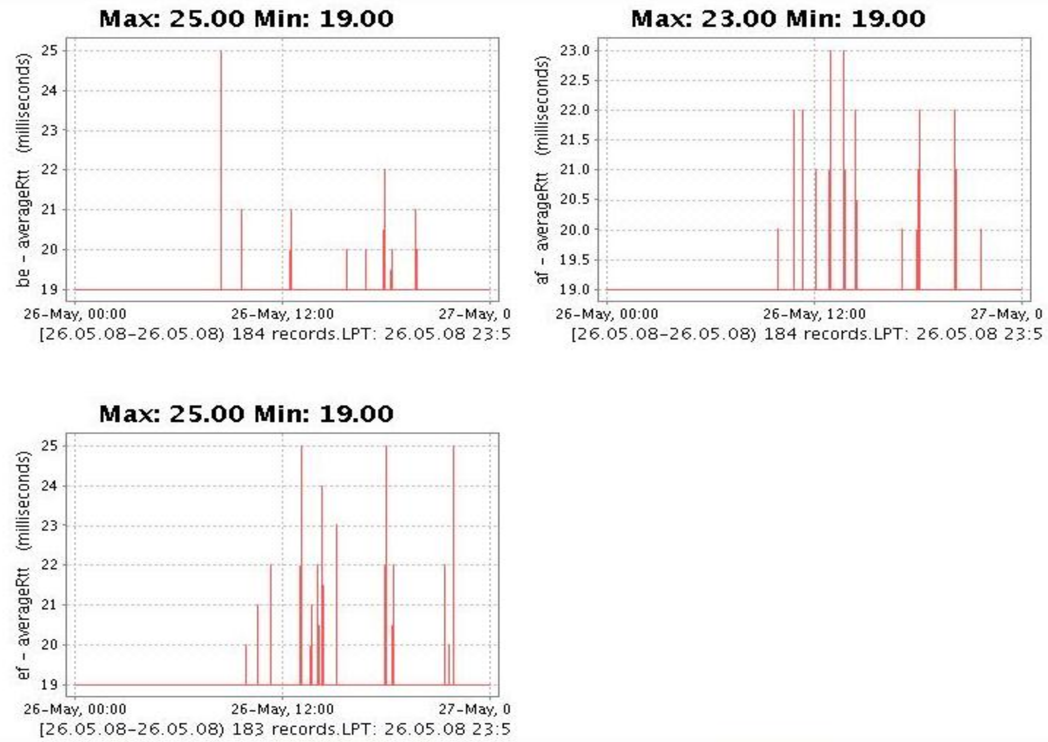

Figure 5.16: The Average RTT graphic measured between Van and Ulus for 1500 Bytes

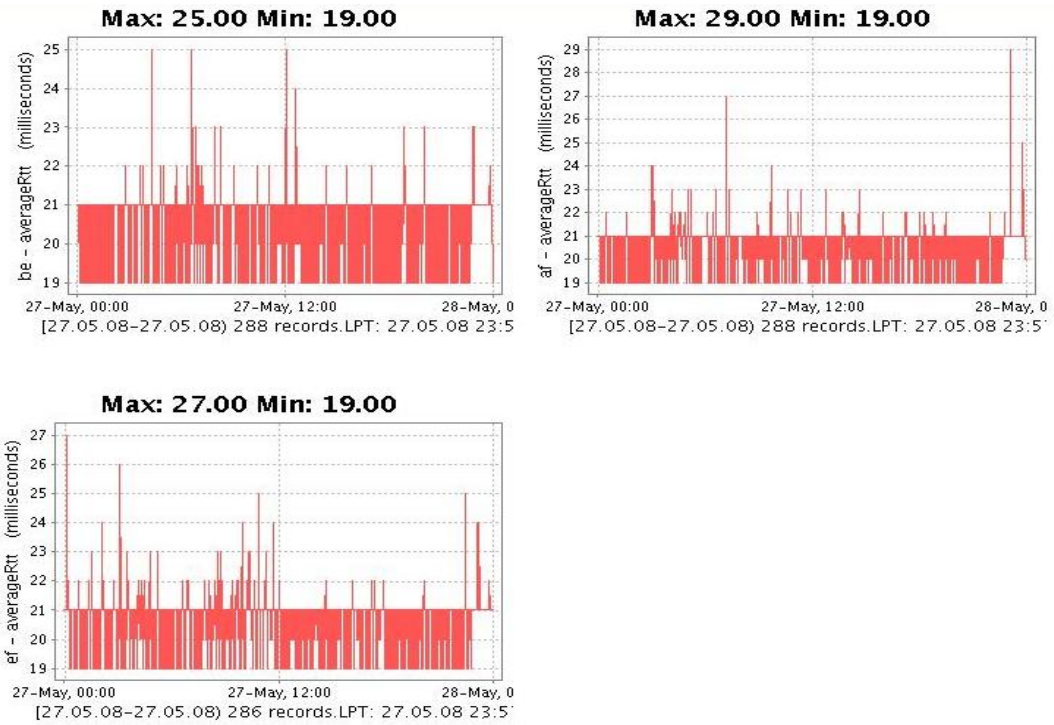

Figure 5.17: The Average RTT graphic measured between Van and Ulus for 4000 Bytes

When jitter values are shown in graphics in Figure 5.10 and Figure 5.11 are compared, it seems that there is a direct proportion between the packet size and jitter value. The same condition is detected in the measurement in Figure 5.14 and Figure 5.15.In Figure 5.14 and Figure 5.15, jitter values between Ulus-Van, the farthest distance in the tests done. The values have the highest jitter value of the tests done.

When RTT value in Figure 5.12 and Figure 5.13 is compared with the measurement between Ulus-Dikmen, the shortest distance, it seems that transmission distance is the positive function of delay. RTT values are obtained in Figure 5.16 and Figure 5.17 belong to the measurements between Ulus-Van. The tests are made, the biggest delay is measured in these tests.

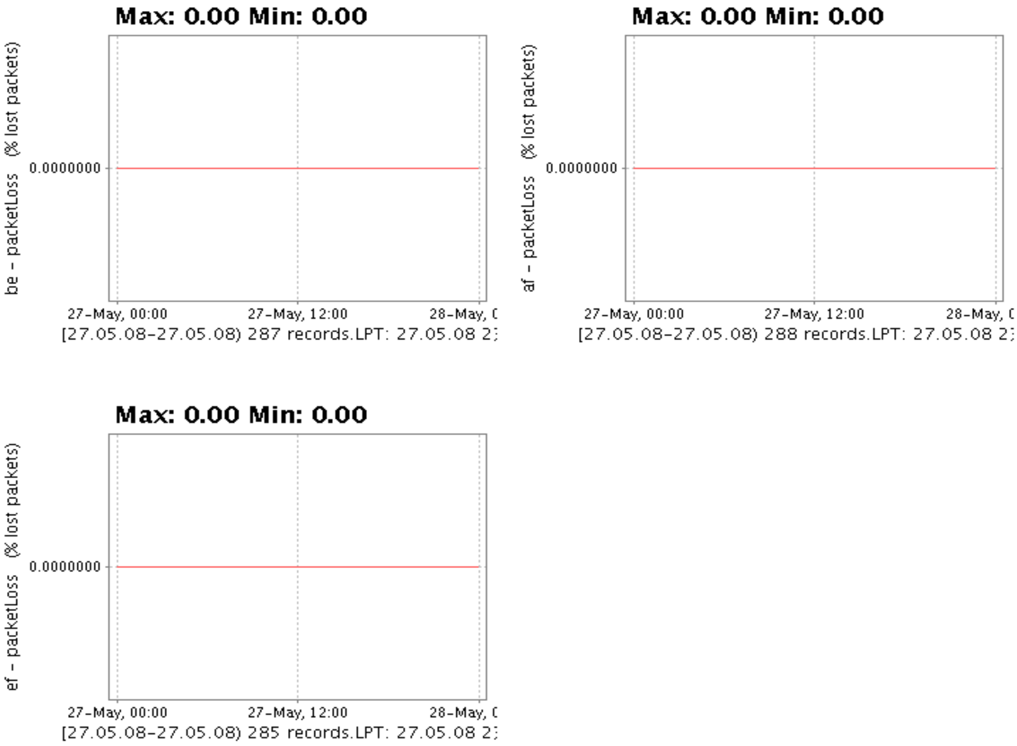

Figure 5.18: The packet loss measured between Dikmen and Ulus for 1500 and 4000 Bytes

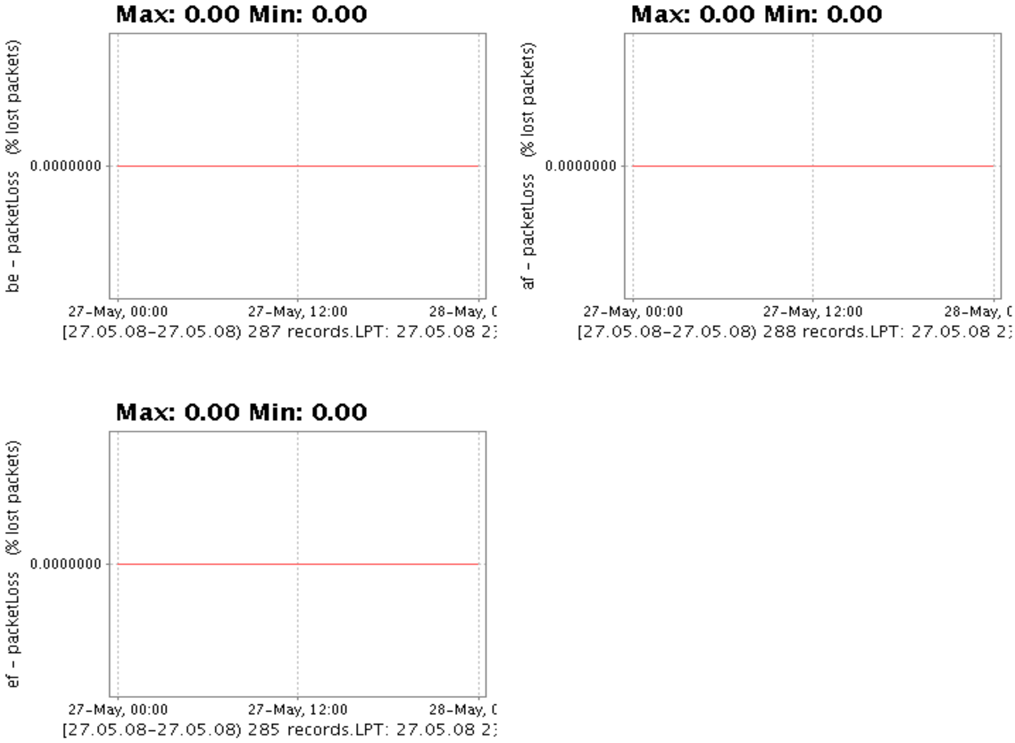

Figure 5.19: The packet loss measured between Van and Ulus for 1500 and 4000 Bytes

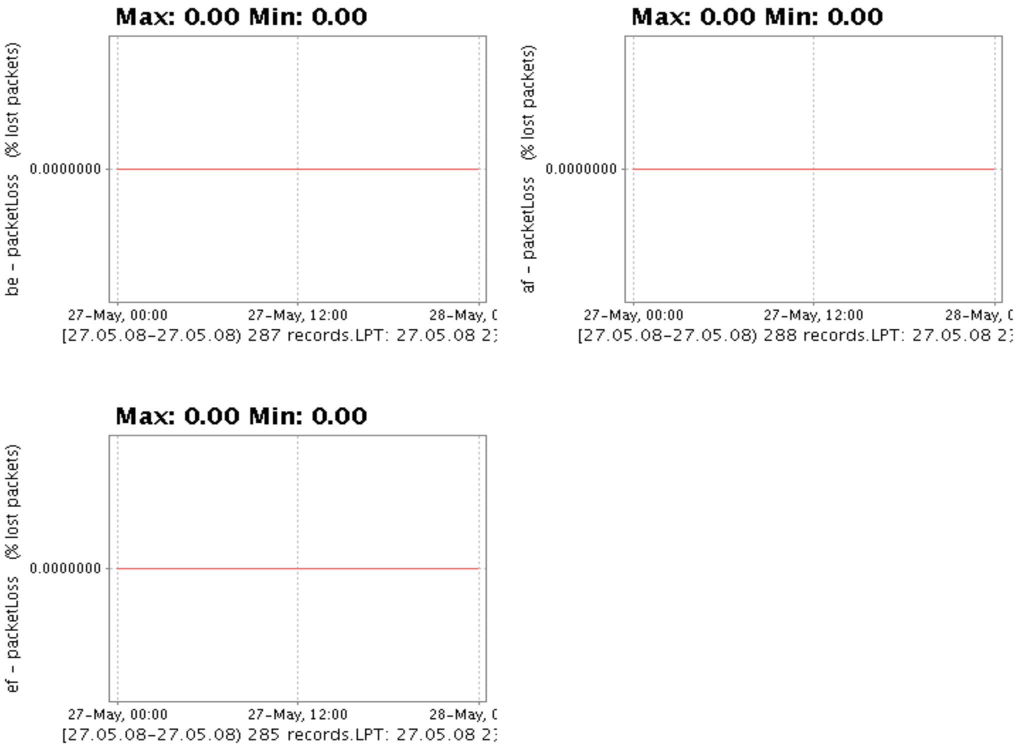

Figure 5.20: The packet loss measured between Konya and Ulus for 1500 and 4000 Bytes

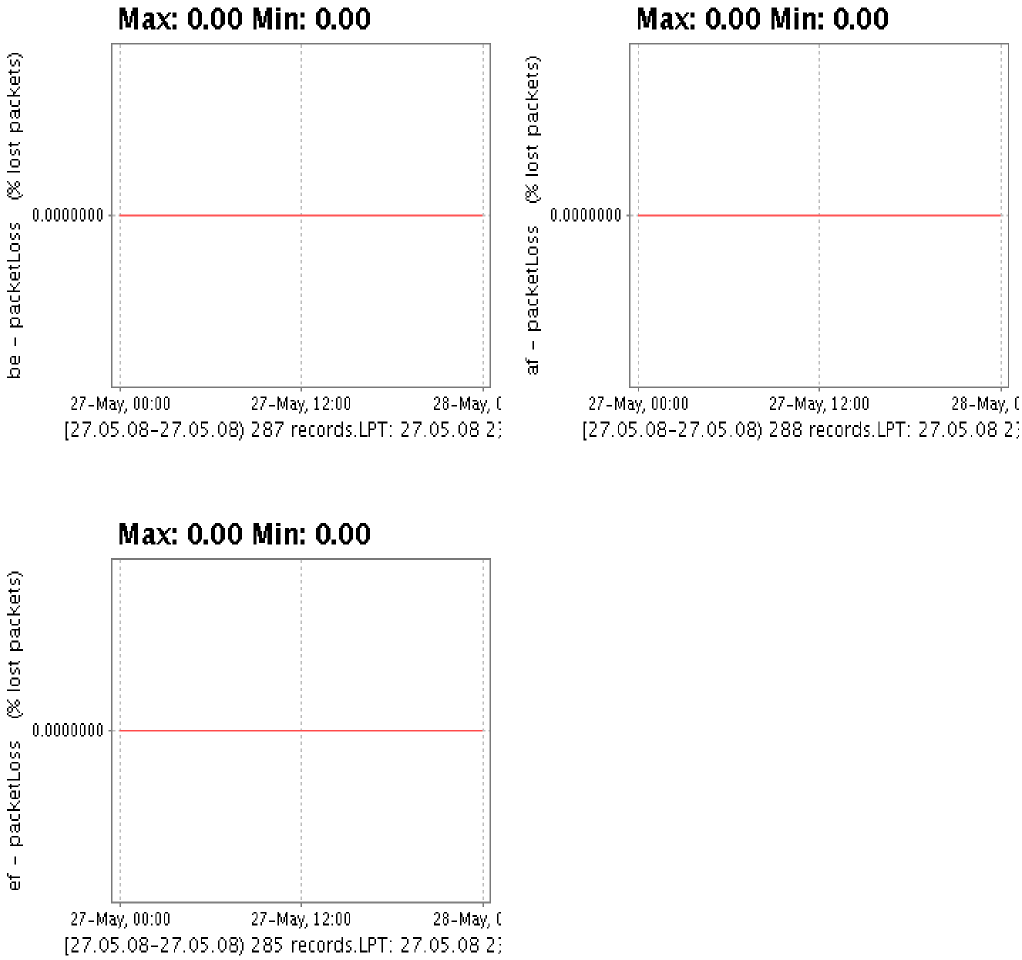

Figure 5.21: The packet loss measured between Kahramanmaraş and Ulus for 1500 and 4000 bytes

The graphics, belonging to packet loss value, is drawn by the program are shown as in Figure 5.18 Ulus-Dikmen, in Figure 5.19 Ulus-Konya, in Figure 5.20 Ulus-Kahramanmaraş and in Figure 5.21 Ulus-Van. As understood from these graphics, in the result of 1-whole day of measurement, there is no packet loss. The packet loss value on IP/MPLS Backbone of Turkish Telecommunications is zero percent.

CHAPTER 6

RESULTS AND SUGGESTIONS

The traffic types are transmitted on TT IP/MPLS in Figure 6.1 are as voice,image and data. For these traffic types, special Qos policies are introduced on Backbone. These traffic types, the one used for voice is assured forwarding, one developed for image is explicit forwarding and finally one used for data is best effort. For the quality measurements of Qos mechanisms, on Backbone jitter, RTT and Packet Loss values should be measured. In this thesis, the results obtained by the tests on Backbone are compared with acceptable values.

Measurements are performed by using different transmission distances and different packet sizes. By this way, the change in values are measured for different traffic types. During measurements, the shortest distanced transmission length is 10 km, the longest distanced transmission length is 1248 km. The reason of the choice of the size of the packet used as 1500 and 4000 Bytes are packet sizes which are generally used for voice and image transmission are near to these values. According to these parameters are chosen, it will be possible to detect the Qos quality for an application used between two farthest locations on TT IP/MPLS Backbone.

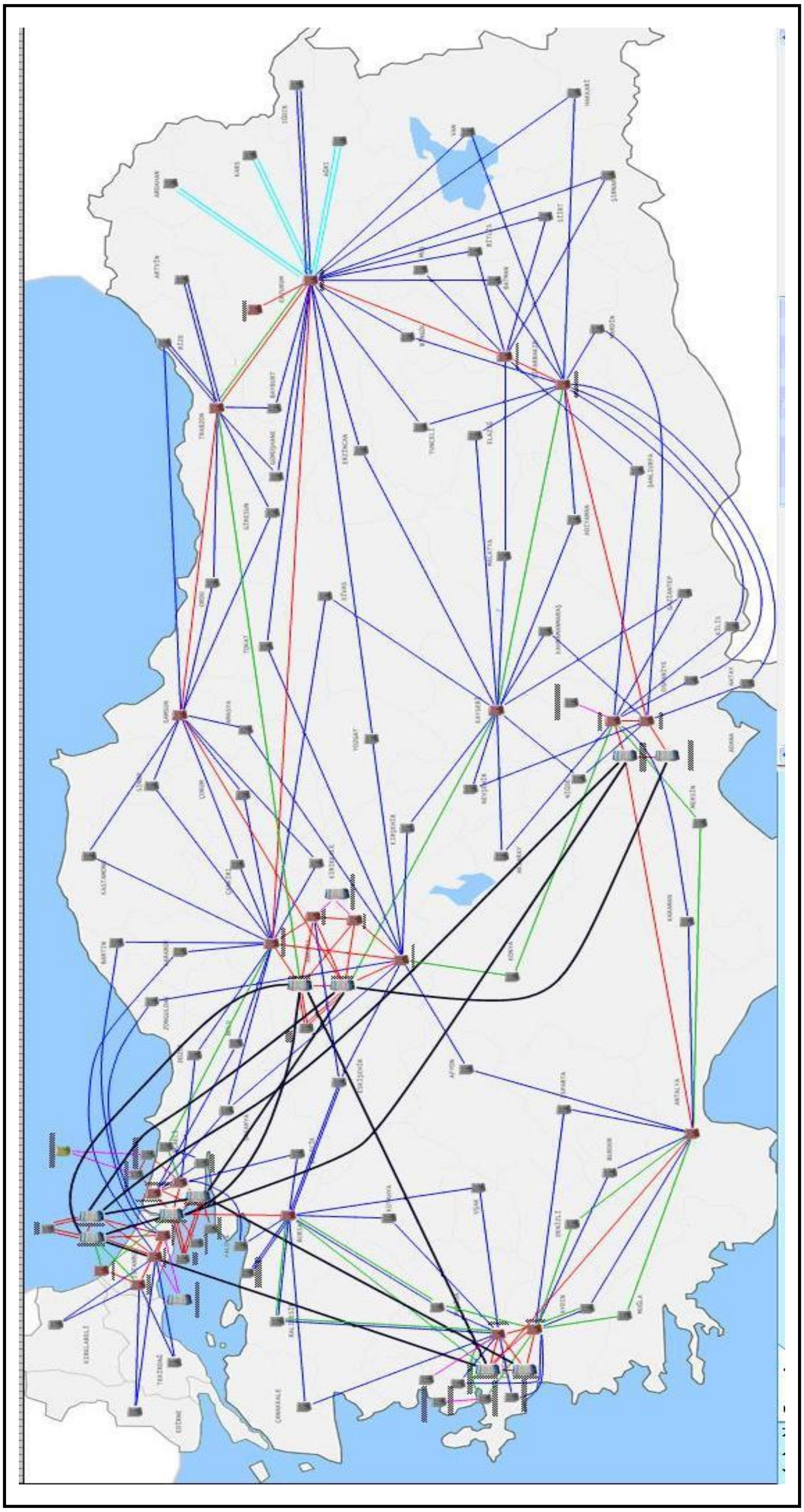

Figure 6.1: IP/MPLS Topology

Table 6.1: All Measurement Values

BEST-EFFORT						
	DISTANCE	ROUND TRIP TIME FOR 1500 BYTES	ROUND TRIP TIME FOR 4000 BYTES	JITTER FOR 1500 BYTES	JITTER FOR 4000 BYTES	PACKET LOSS FOR ALL
ULUS-DİKMEN	10 Km	5 ms	7 ms	14.05 ms	18.44 ms	0 ms
ULUS-KONYA	300 Km	7 ms	15 ms	12.86 ms	23.83 ms	0 ms
ULUS-K.MARAŞ	602 Km	19 ms	25 ms	24.32 ms	35.35 ms	0 ms
ULUS-VAN	1248 Km	25 ms	25 ms	28.94 ms	29.41 ms	0 ms
ASSURED FORWARDING						
	DISTANCE	ROUND TRIP TIME FOR 1500 BYTES	ROUND TRIP TIME FOR 4000 BYTES	JITTER FOR 1500 BYTES	JITTER FOR 4000 BYTES	PACKET LOSS FOR ALL
ULUS-DİKMEN	10 Km	4 ms	8 ms	12.11 ms	16.10 ms	0 ms
ULUS-KONYA	300 Km	11 ms	20 ms	18.18 ms	31.19 ms	0 ms
ULUS-K.MARAŞ	602 Km	21 ms	20 ms	26.84 ms	27.03 ms	0 ms
ULUS-VAN	1248 Km	23 ms	29 ms	26.77 ms	32.87 ms	0 ms
EXPLICIT FORWARDING						
	DISTANCE	ROUND TRIP TIME FOR 1500 BYTES	ROUND TRIP TIME FOR 4000 BYTES	JITTER FOR 1500 BYTES	JITTER FOR 4000 BYTES	PACKET LOSS FOR ALL
ULUS-DİKMEN	10 Km	3 ms	7 ms	12.85 ms	17.23 ms	0 ms
ULUS-KONYA	300 Km	9 ms	12 ms	16.06 ms	18.14 ms	0 ms
ULUS-K.MARAŞ	602 Km	21 ms	19 ms	26.41 ms	25.66 ms	0 ms
ULUS-VAN	1248 Km	25 ms	27 ms	29.49 ms	31.65 ms	0 ms

The first value measured is the Jitter value. According to ITU-T P.861 the acceptable jitter value for VOIP is 30 ms. The jitter values are taken for 10 km between Ulus-Dikmen devices on TT IP/MPLS Backbone, as seen in Table-6.1,14.05 ms is for best effort, 12.85 ms is for explicit forwarding, which is of more quality. It is 12.11 ms for the assured forwarding, which is the most quality. Considering all the jitter values are obtained in the result of the tests in Tablo-6.1, it seems that, 30 ms jitter value, the acceptable value, is not transcended for any of tests. During the measurements done for the 4000 Bytes of packets, an increase in the same jitter value is seen. The reason of this increase is thought that, the scheduling and queuing delay which occurs at every devices while the packets generated are being sent to source address. When packet size increases, delay values mentioned will increase and this increase will cause growth in jitter value. When these results are considered, during the measurements done between Ulus-Van, the jitter concerning best-effort traffic type is 28.94 ms, one for concerning assured forwarding traffic type is 26.77 ms, one for concerning explicit forwarding traffic type is 29.49 ms. When considered these results, though jitter value is expected to be high for the best effort as to Qos mechanism, it is the worst value when the result obtained for explicit forwarding is compared with the acceptable values. These results show that there is no superiority among the usage of 3 different traffic type defined on TT IP/MPLS Backbone. The reason of this is the low usage rate of sources like buffer, link and switching capacitiy, which are used sharingly on Backbone, which previously allocated as to service quality. As understood from here, on Backbone, both the internet user data, which uses the simplest traffic type, and the one which uses the critical traffic type are transmitted at equal conditions. During tests done using 4000 bytes of fragmented packets, the jitter value obtained for Ulus-Kahramanmaraş is 35.35 ms, the highest jitter value. This value is transcended by acceptable values. Although the acceptable value is adviced as 30 ms for VOIP, jitter value should be decreased up to 1 ms in order to make a conference of PSTN quality. In order to provide this, jitter buffer should be used on Backbone. In order to provide the usage of voice transmission technology more effectively and to reduce the cost, it is expected to replace the substructure used for PSTN with IP Backbone in the future. Because of this, in order to close the service quality up to the values obtained on PSTN network, there may be some betterments on IP/MPLS Backbone by TT.

According to ITU-T P.861, the single sided acceptable delay for Voice Over IP (VOIP) is 200 ms.

At the measurements are taken during this work, RTT, which is the two-sided delay value. Because of this, we should do the evaluations to be done at 400 ms acceptable values. The RTT values obtained from measurement are shown in Table 6.1. The lowest RTT value obtained during the measurements taken between Ulus-Dikmen, which has the shortest transmission distance, is 3 ms. The longest RTT value obtained during the measurements is taken between Ulus-Van devices, which has the longest transmission distance, is 29 ms. This value is obtained by the result of tests using fragmented packets of 4000 bytes. When they are compared with acceptable values for VOIP, it shows that the values obtained from tests have beter results than acceptable values.

According to ITU-T P.861, the packet loss value for VOIP is 1%. No packet loss is detected in the measurements. Also in terms of this value, It is understood that TT IP/MPLS Backbone has an ideal design for the online applications like VOIP. The transmission of the online applications like voice requires high costs on PSTN Network. For this reason, the usage of data network during voice transmitting is the most suitable solution on reducing expenses. Because of this, IP/MPLS Backbone is preferred in data Networks in order to serve of same quality as PSTN networks. Nowadays, the software and hardware used for online applications like voice and image are designed as to IP technology. As known, TCP/IP protocol is a routing based protocol. The general base running logic of Routing mechanisms is to calculate the shortest way and provide the sending packets through that way. By this way, transmitting traffic types of a specific quality like voice and video on a structure may have some difficulty. Because of this, the substructure should be chosen properly for the applications like voice and image. For this, a network has been established which works on MPLS protocol. When it is considered all these, we should not see it sufficient to keep the the values obtained after measurements in the acceptable values. On Backbone, by using solutions like jitter buffer, the transmitting need of online applications like voice and image in a more quality media should be met. If we compare the the values obtained in this thesis with the acceptable values and summarize the point, although it has a efficient result in terms of RTT and Packet Loss values in the work, because the results obtained from tests for Jitter are very close to threshold values, it is obvious that there is a need of improvement. For this, usage of the configurations that will use the existing devices in a more effective way by TT would be a solution. Because of this, without having a need of any investment, the jitter value may be reduced to the value desired.

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