A Practical Evaluation of QoS for Voice over IP

Daniel Zinca, Virgil Dobrota,

Mihai Vancea, Gabriel Lazar

Technical University of Cluj-Napoca, Romania

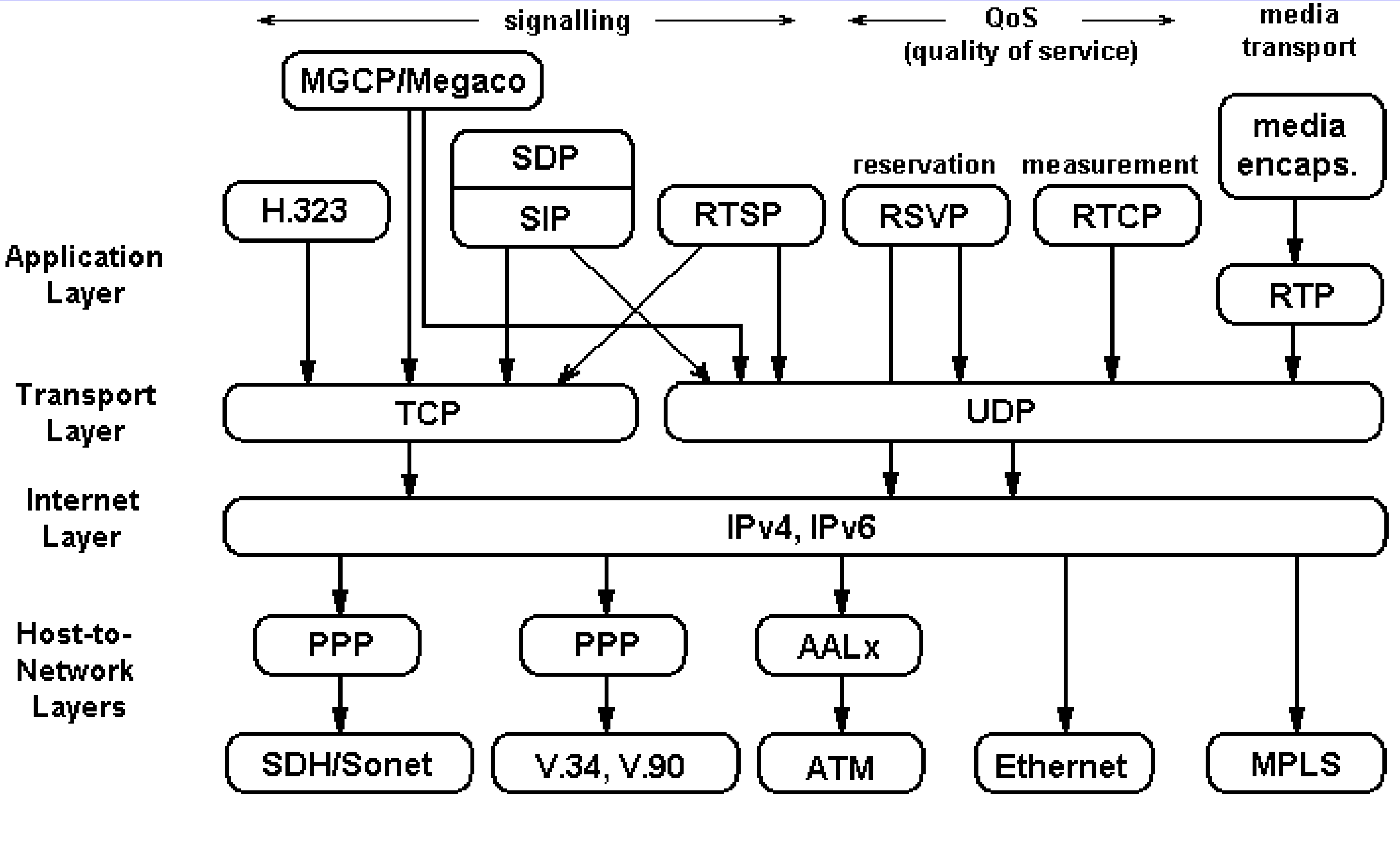
Department of Communications

{Daniel.Zinca, Virgil.Dobrota, Mihai.Vancea, Gabriel.Lazar}@com.utcluj.ro

Contents

- Internet Multimedia Protocol Stack
- R-factor and MOS (Mean Opinion Score)
- RTCP and VoIP parameters
- Testbed Demonstrator
- QoS Management Tool
    - ***Ethereal***
    - ***RTCP Analyzer***
    - ***NDIS Intermediate Driver***
    - ***Microsoft Generic QoS API***
    - ***realtool***
- Conclusions and further work

The Internet Multimedia Protocol Stack



R-factor Calculation

R-factor = simple measure of voice quality (0…100)

$$R = 94.2 - I_d - I_{ef}$$

Id = the impairment associated with the mouth-to-ear delay of the path

d = one-way delay (coding delay + network delay + de-jitter delay)

$$I_d = 0.024d + 0.11(d - 177.3)H(d - 177.3)$$

$$H(x) = 0 \text{ for } x < 0$$
$$H(x) = 1 \text{ for } x \geq 0$$

Ief = the equipment impairment factor

$$I_{ef}(G.711, concealment, bursty) \cong$$
$$\cong 30\ln(1 + 15e)H(0.04 - e) + 19\ln(1 + 70e)H(e - 0.04)$$

Mean Opinion Score (MOS)

R-factor	Quality of voice	MOS
90 < R < 100	Best	4.34 … 4.50
80 < R < 90	High	4.03 … 4.34
70 < R < 80	Medium	3.60 … 4.03
60 < R < 70	Low	3.10 … 3.60
50 < R < 60	Poor	2.58 … 3.10

RTCP (RTP Control Protocol)

- periodic transmission of control packets
- same paths & different UDP ports as data packets
- RTCP packets: SDES *(Source Description)*
  SR *(Sending Report)*
  RR *(Receiver Report)*
  BYE
  APP

Example of Sender Report

```
⊟ Real-time Transport Control Protocol
    Version: RFC 1889 Version (2)
    Padding: False
    Reception report count: 1
    Packet type: Sender Report (200)
    Length: 12
    Sender SSRC: 2129287575
    Timestamp, MSW: 3224588906
    Timestamp, LSW: 3198720832
    RTP timestamp: 144000
    Sender's packet count: 601
    Sender's octet count: 144240
  ⊟ Source 1
      Identifier: 793046859
    ⊟ SSRC contents
        Fraction lost: 0 / 256
        Cumulative number of packets lost: 0
    ⊟ Extended highest sequence number received: 0
        Sequence number cycles count: 0
        Highest sequence number received: 0
      Interarrival jitter: 49
      Last SR timestamp: 0
      Delay since last SR timestamp: 0
```

Interarrival jitter

$$J = J + (|D(i-1,i)| - J)/16$$

$$D(i-1,i) = [R(i) - R(i-1)] - [S(i) - S(i-1)]$$

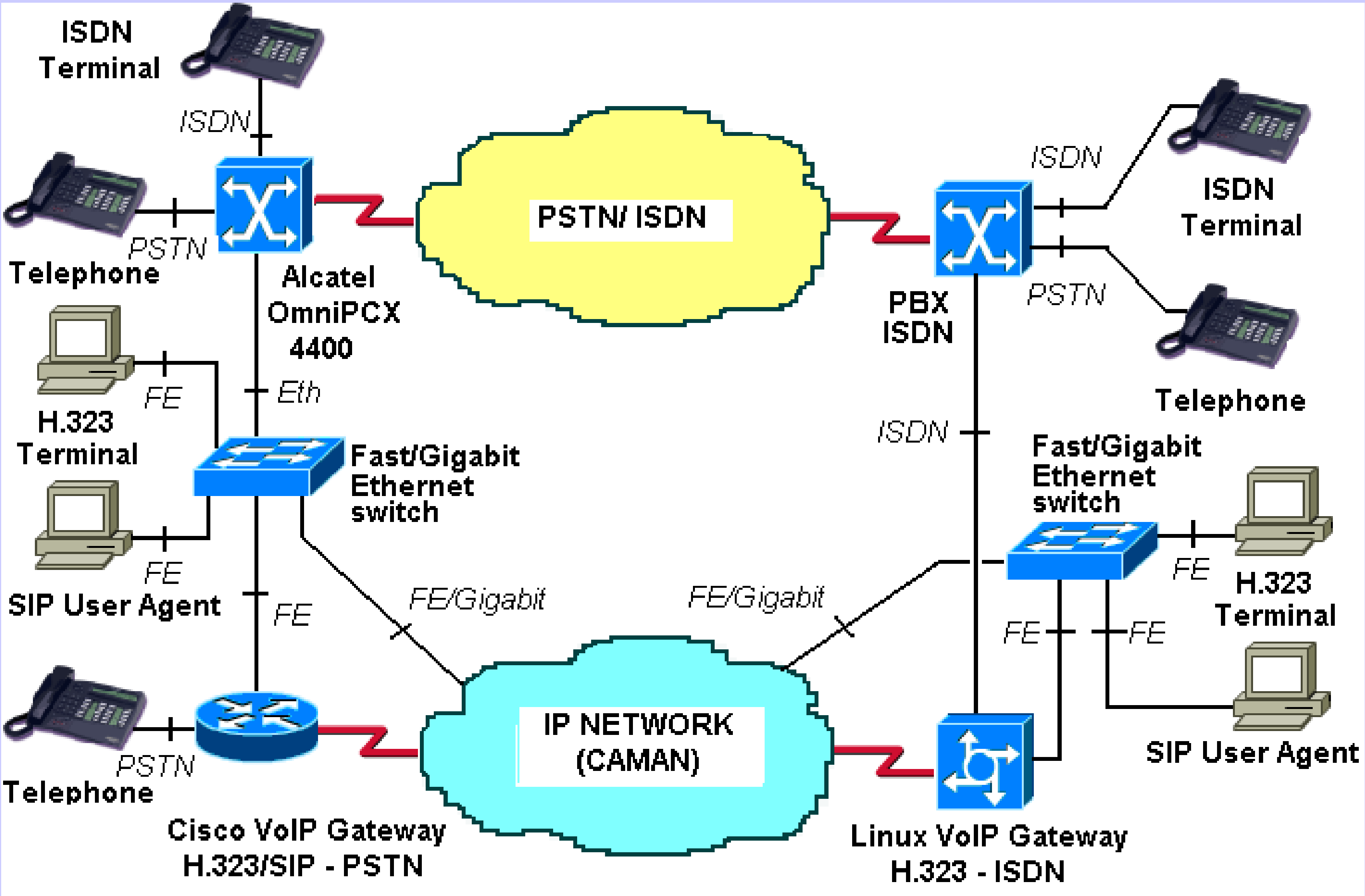
- D(i-1,i) = mean deviation of RTP timestamps for two consecutive packets
- R(i) = reception RTP timestamp
- S(i) = transmission RTP timestamp
- 1/16 = gain parameter

Return transfer delay

$$RTT = A - LSR - DLSR$$

- A = arrival time of the reception report
- LSR (Last SR Timestamp) = middle 32 bits out of 64 in the NTP timestamp within the most recent SR
- DLSR (Delay since Last SR)

VoIP Testbed Demonstrator



EXPERIMENTS

PC-to-Phone:

- **Experiment 1: H.323 - Cisco GW H.323 – PSTN terminal**
- **Experiment 2: H.323 - Linux GW H.323 - ISDN terminal**
- **Experiment 5: SIP User Agent - Cisco GW SIP - PSTN**

PC-to-PC:

- **Experiment 3: H.323 terminal – H.323 terminal**
- **Experiment 6: SIP User Agent - SIP User Agent**
- **Experiment 7: SIP User Agent - Cisco GW SIP - Cisco GW H.323 - H.323 terminal**

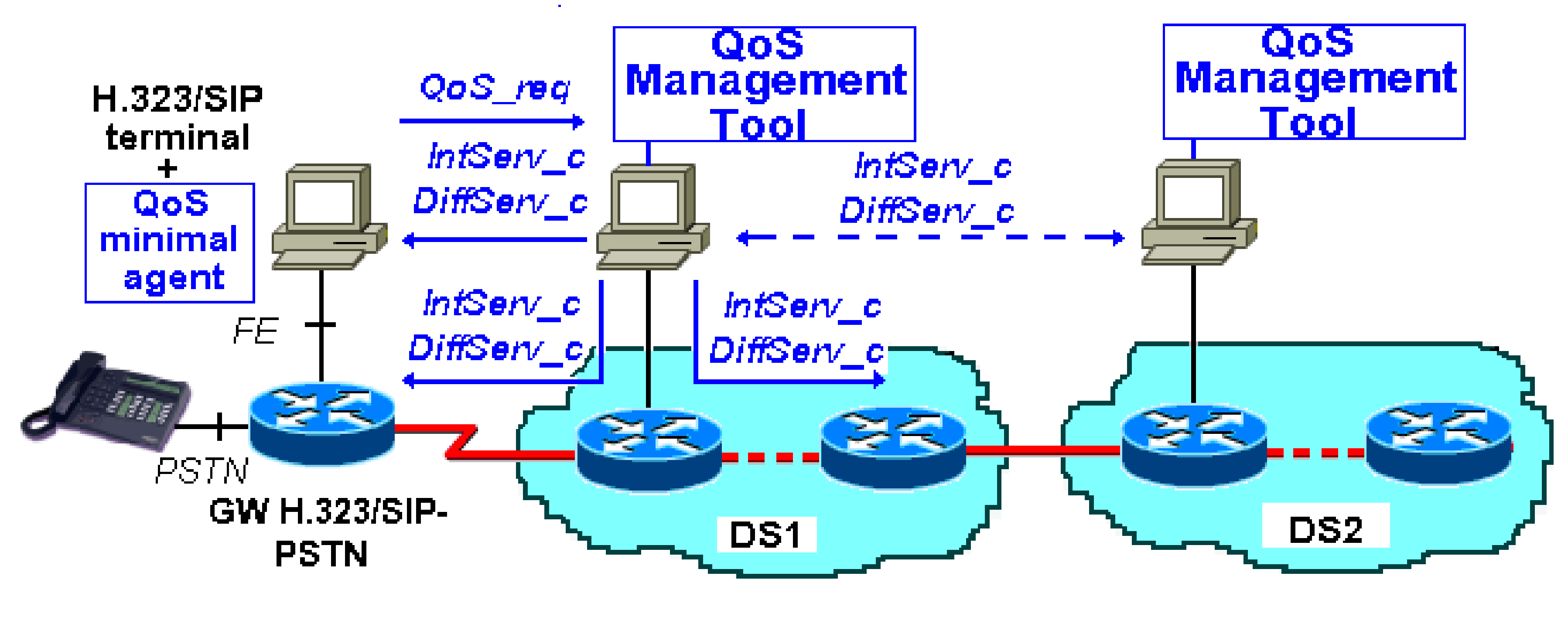
Phone-to-Phone:

- **Experiment 4: PSTN terminal - Cisco GW H.323 - Linux GW H.323 - ISDN terminal**

Experimental results

Exp.	Application	Type/ Operating System	Jitter	Packet lost	RTT	Int Serv	Diff Serv
1,2, 3,7	NetMeeting 3.0.1	H.323 terminal/ Windows	X	X	X	X	-
1,2, 3,7	OpenPhone	H.323 terminal/ Windows/Linux	X	X	X	-	X
1,4,7	Cisco 1750 Gateway	H.323-PSTN gateway/ Cisco IOS	X	X	X	X	X
2,4	RedHat 7.2 Gateway	H.323-ISDN gateway/ Linux	X	X	-	-	-
2,4	CAPI-based application	ISDN terminal/ Windows/Linux	X	X	-	-	-
5, 6,7	eStara SoftPhone	SIP User Agent/ Windows	-	-	-	-	-
5, 6,7	Helmsmann	SIP User Agent/ Windows	-	-	-	-	-
5,7	Cisco 1750 Gateway	SIP-PSTN gateway/ Cisco IOS	X	X	-	X	X
7	Cisco 1750 Gateway	PSTN-H.323-SIP-PSTN gateway/ Cisco IOS	X	X	X	X	X

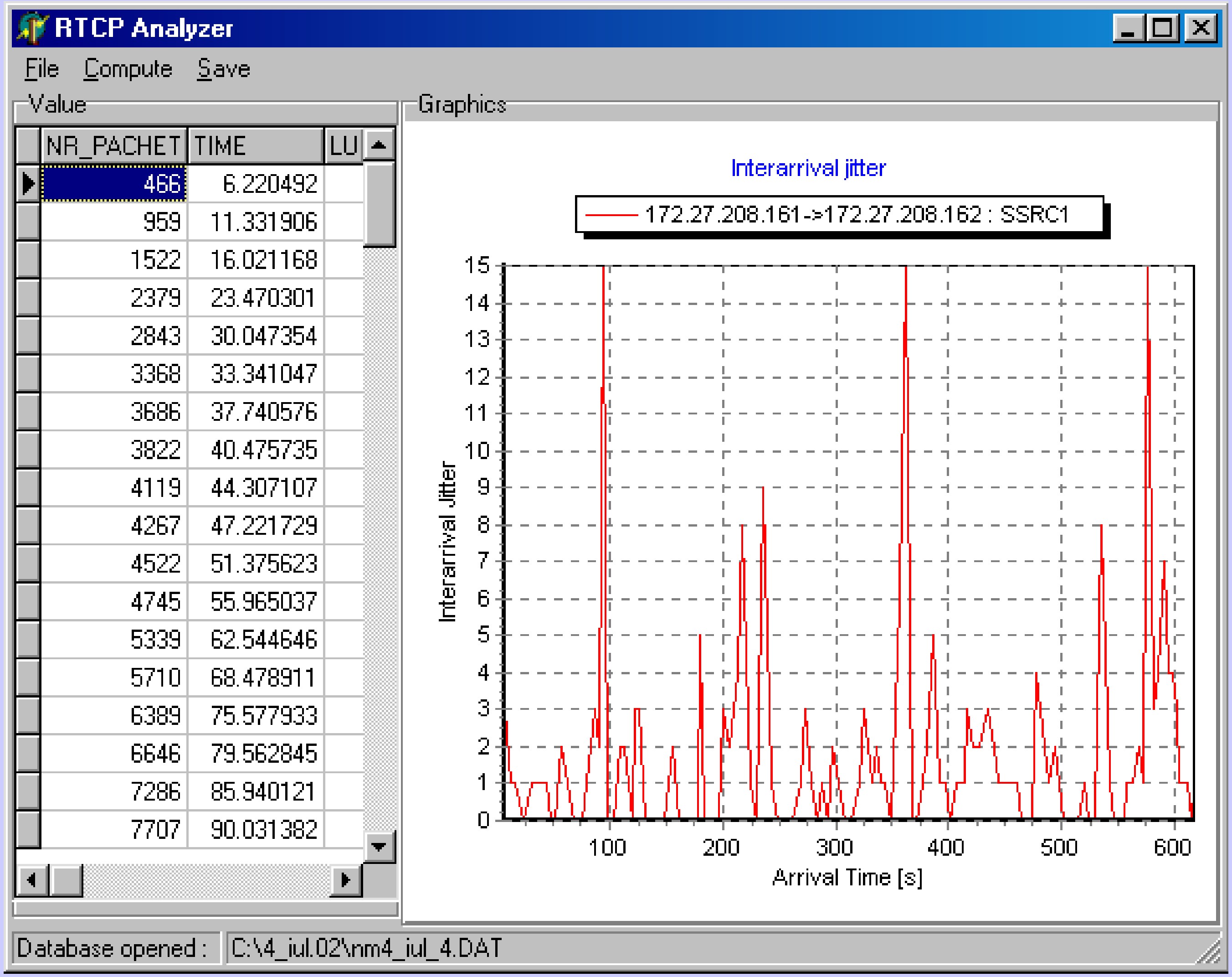
QoS Management Tool + QoS Minimal Agents



- **QoS_req:** QoS request for VoIP parameters + QoS scheme
- **IntServ_c:** IntServ configuration (RSVP)
- **DiffServ_c:** DiffServ configuration (DSCP)

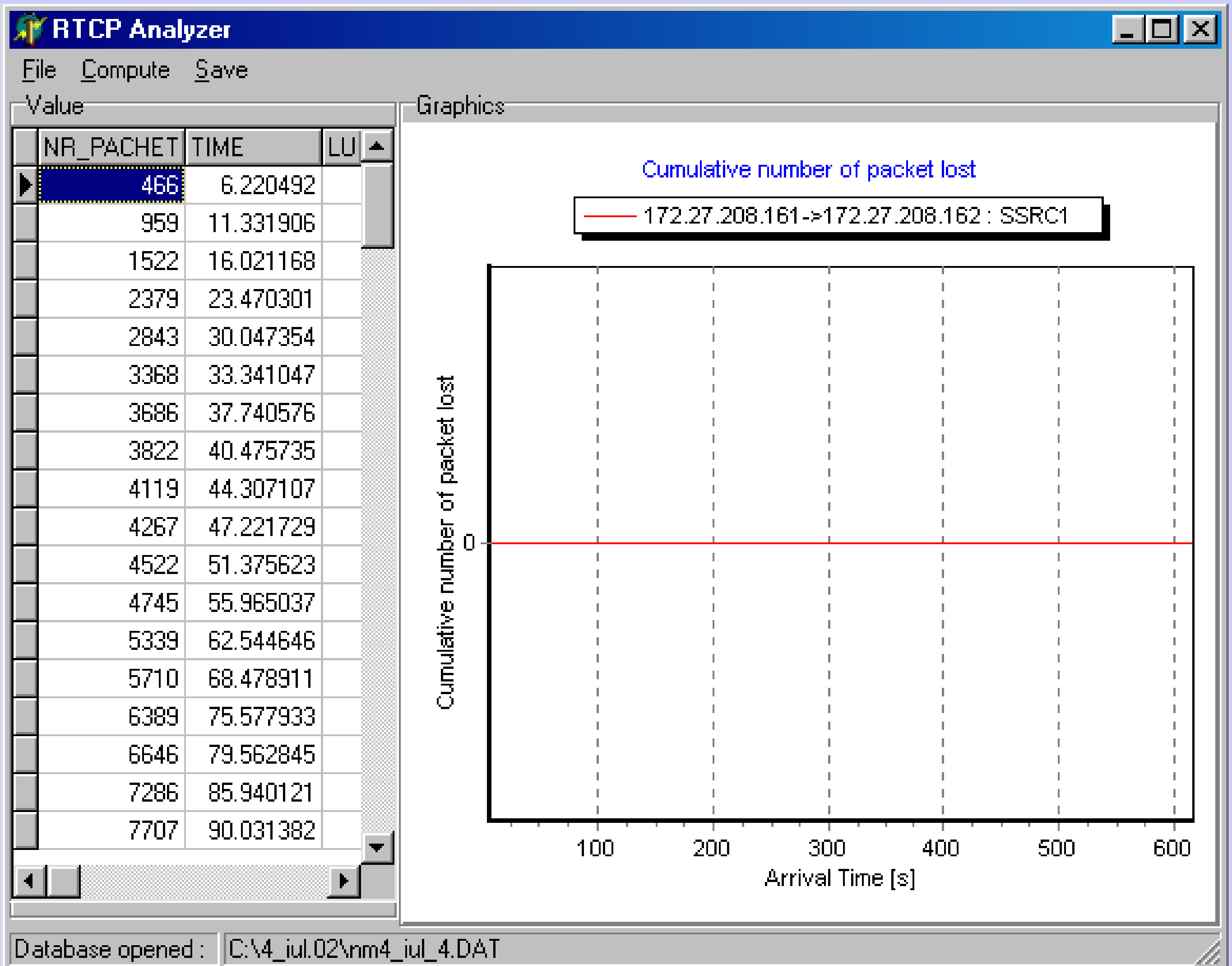
Experiment 3: H.323 <=> H.323 terminal

Interarrival jitter (4_iul, fig.11) - without QoS

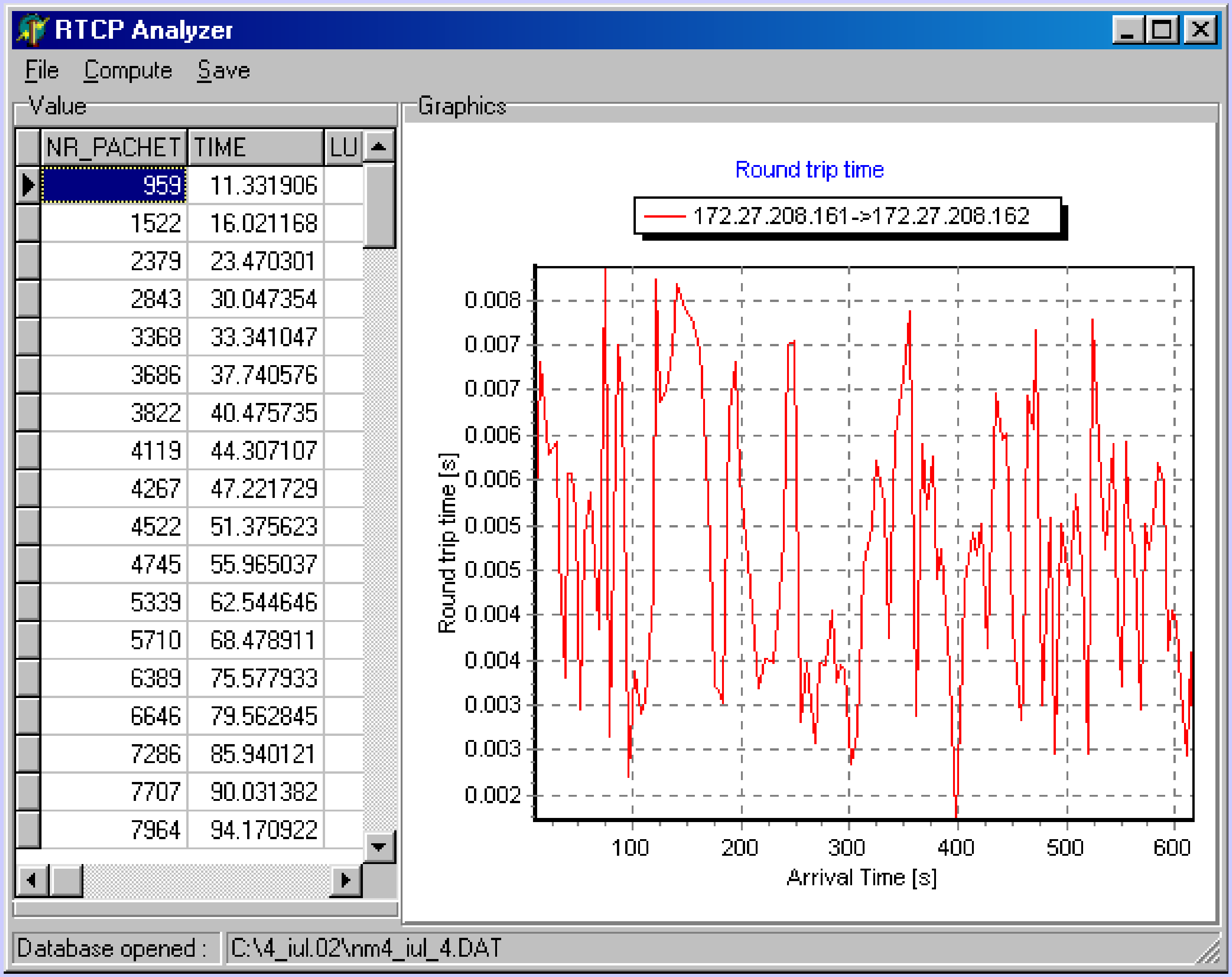


Experiment 3: H.323 <=> H.323 terminal

Packet lost (4_iul, fig.13) - without QoS



Experiment 3: H.323 <=> H.323 terminal
RTT evaluation (4_iul, fig.15) - without QoS



CONCLUSIONS

1. Qos Management Tool for IntServ and DiffServ
2. IntServ Approach is based on capturing, analyzing and generating of RTCP and RSVP packets
3. Some SIP User Agent implementations have difficulties to perform QoS measurements (RTCP) or QoS reservations (RSVP).

FURTHER WORK

1. Under progress: Phone-to-Phone experiments, H.323 to SIP gateways, QoS mechanisms for LAN–to-WAN interworking
2. DiffServ aproach
3. Security aspects may request firewalls.
4. Potential return of investment (ROI)? See Cisco´s financial modeling tool called CNIC (Converged Network Investment Calculator).

REFERENCES

[1] R.G. Cole and J.H. Rosenbluth, "Voice over IP Performance Monitoring", *Computer Communication Review*, ACM-SIGCOMM, Vol.31, No.2, April 2001, pp. 9-24.

[2] F. Kettler et al., "End-to-End Speech Quality Testing in a Complex Transmission Scenario", HEAD acoustics GmbH, 2002.

[3] Johnston, Alan B., *SIP: Understanding the Session Initiation Protocol*, Artech House, Boston, London, 2001.

[4] SIP WG et al., "SIP: Session Initiation Protocol", draft-ietf-sip-rfc2543bis-09.txt, IETF, February 2002

[5] V. Fineberg, "A Practical Architecture for Implementing End-to-End QoS in an IP Network", IEEE Communications Magazine, Vol.40, No.1, January 2002, pp.122-130

[6] Black, Uyless, *Voice over IP*, Second Edition, Prentice Hall, 2002