Speech Quality Evaluation, Impact of Security on Quality

Miroslav Vozňák

miroslav.voznak@vsb.cz

Antonio Nappa , Universita degli Studi di Milano
Michal Halás , STU Bratislava
Eduardo Rocha , Universidade De Aveiro
František Hormek , CESNET

MOS Score and R-factor

MOS Mean Opinion Score

range 1-5

MOS – LQ, listening quality

MOS – CQ, conversational quality

R factor

range 0-95 narrowband codec

range 0-120 wideband codec

R-LQ, R-CQ

MOS can be obtained from R-factor (conversion formula)

Measuring MOS

subjective

- using a listening panel

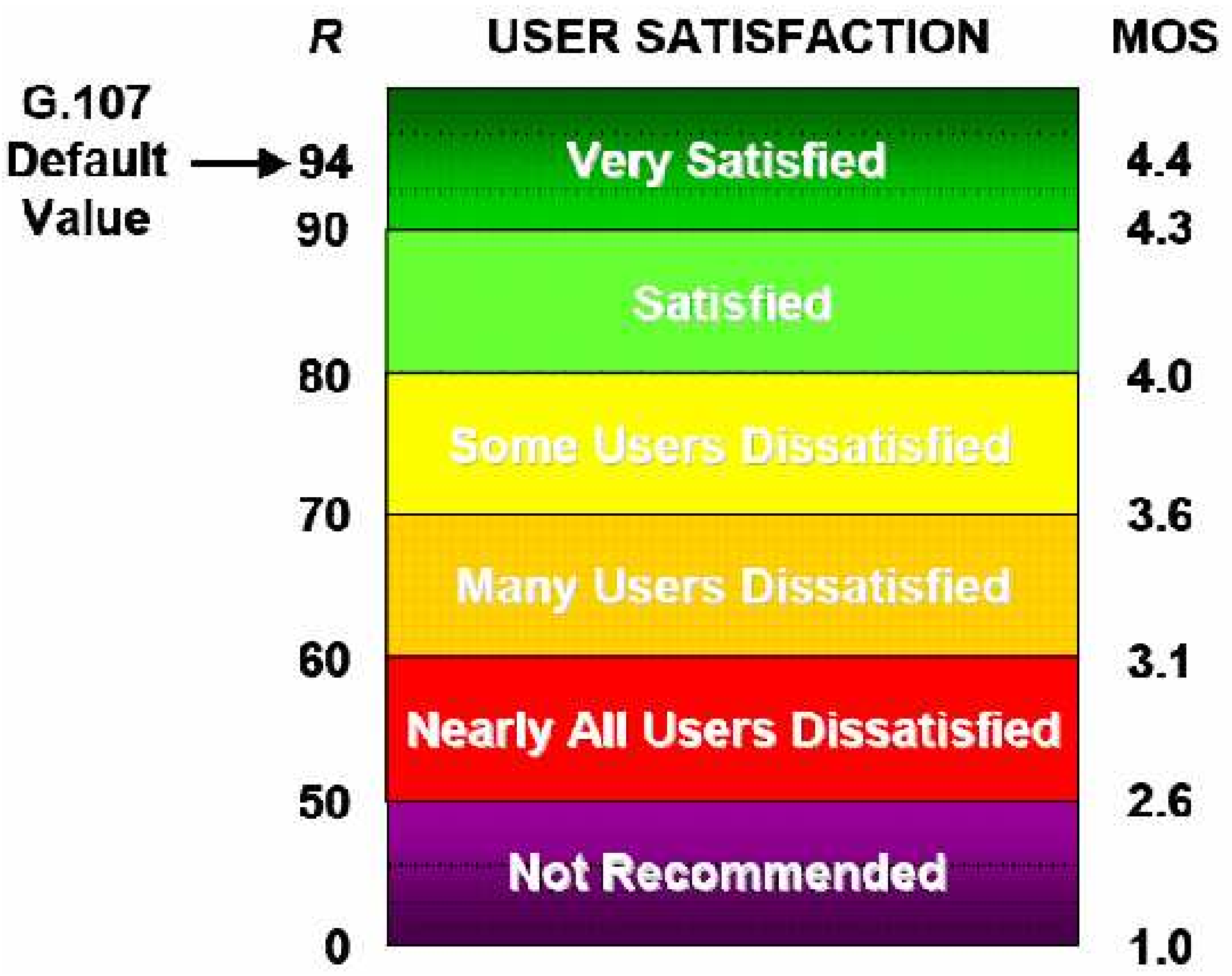
full reference approach

- comparing output with input
- PESQ (Perceptual Evaluation of Speech Quality), ITU-T P.862

no reference approach

- measurements at the receiving end, estimating MOS
- E-model, ITU-T G.107

E-model is computational model for use in transmission planning



$$R = Ro - Is - Id - Ie + A$$

Additive model

Ro – Signal to noice ratio

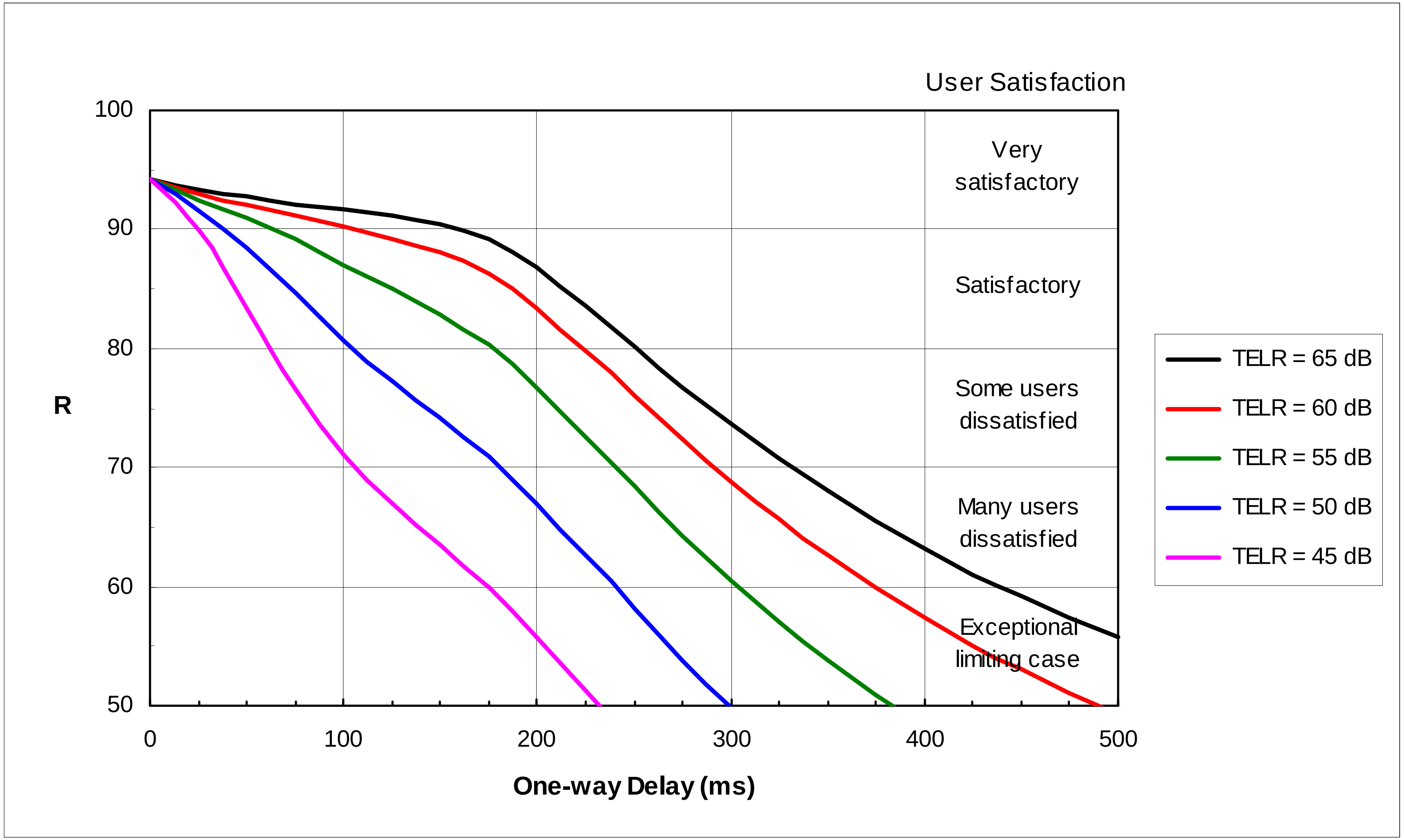
Is - Impairments simultaneous to voice signal transmission

Id – Impairments delayed after voice transmission

Ie – Effects of equipment (e.g. codecs)

A - Advantage factor

Delay


User Satisfaction
100
90
80
70
60
50
R
0
100
200
300
400
500
One-way Delay (ms)
Very satisfactory
Satisfactory
Some users dissatisfied
Many users dissatisfied
Exceptional limiting case
TELR = 65 dB
TELR = 60 dB
TELR = 55 dB
TELR = 50 dB
TELR = 45 dB

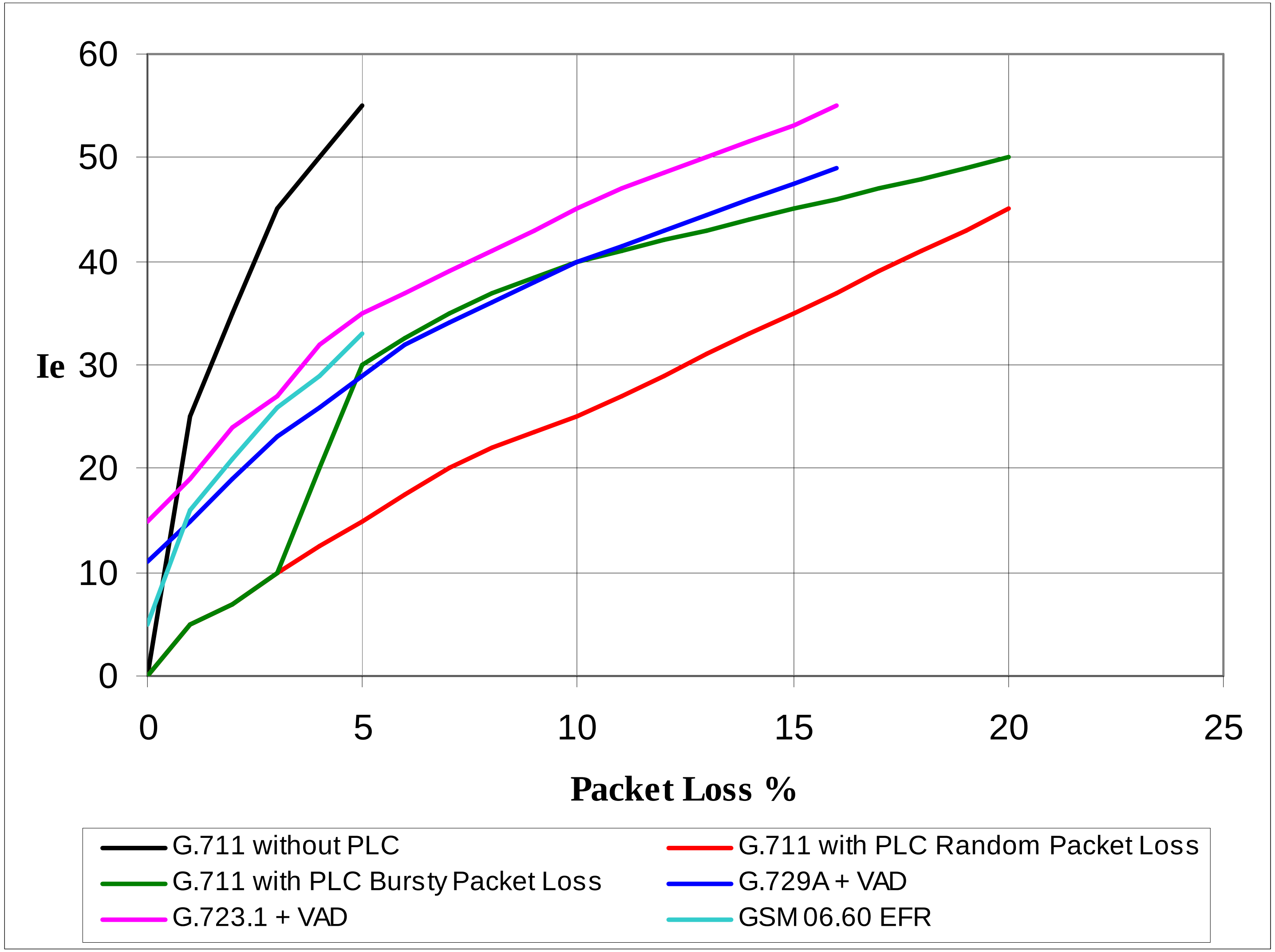
Codec

Codec Type	Reference	Operating Rate kbit/s	Ie Value
PCM	G.711	64	0
ADPCM	G.726, G.727	40	2
	G.721, G.726, G.727	32	7
	G.726, G.727	24	25
	G.726, G.727	16	50
LD-CELP	G.728	16	7
		12.8	20
CS-ACELP	G.729	8	10
	G.729-A + VAD	8	11
ACELP	GSM 06.60, EFR	12.2	5
ACELP	G.723.1	5.3	19
MP-MLQ	G.723.1	6.3	15

Packet Loss

Packet Loss (%)	G.711 without PLC (10 ms speech packet length)	G.711 + PLC Random Packet Loss (10 ms speech packet length)	G.711 + PLC Bursty Packet Loss (10 ms speech packet length)	G.729A + VAD 8 kbit/s (2 speech frames/ packet)	G.723.1 + VAD 6.3 kbit/s (1 speech frame/ packet)	GSM 06.60 EFR 12.2 kbit/s (1 speech frame/ packet
0	0	0	0	11	15	5
0.5	–	–	–	11	15	–
1	25	5	5	15	19	16
1.5	–	–	–	17	22	–
2	35	7	7	19	24	21
3	45	10	10	23	27	26
4	–	–	–	26	32	–
5	55	15	30	–	–	33
7	–	20	35	–	–	–
8	–	–	–	36	41	–
10	–	25	40	–	–	–
15	–	35	45	–	–	–
16	–	–	–	49	55	–
20	–	45	50	–	–	–

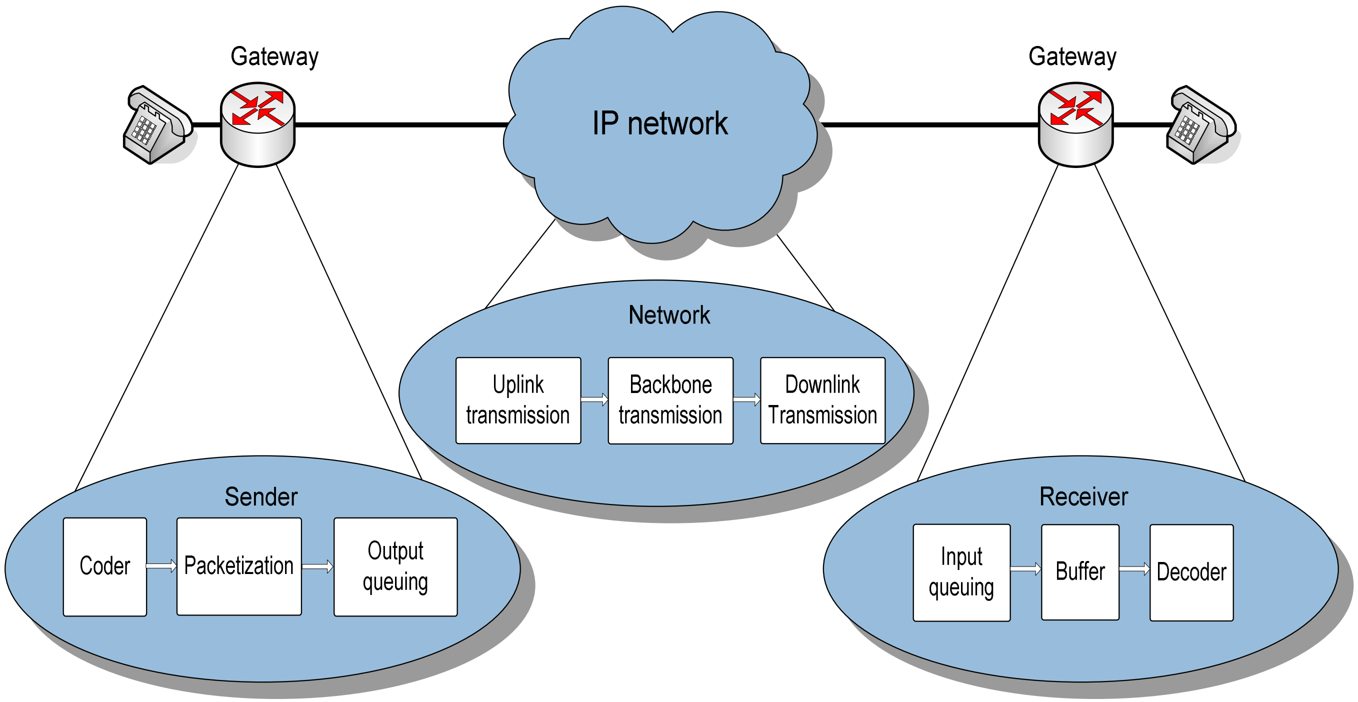
Packet Loss



60
50
40
Ie 30
20
10
0
0
5
10
15
20
25
Packet Loss %
G.711 without PLC
G.711 with PLC Random Packet Loss
G.711 with PLC Bursty Packet Loss
G.729A + VAD
G.723.1 + VAD
GSM 06.60 EFR

End to end delay and its prediction

- design of mathematical model
- this model describes all partial delay components



Coder delay

- frame size delay
- algorithmic delay

Coder	Type	Rate [kbps]	Packetization Period [ms]	Frame size [ms]	Algorithmic delay [ms]	Codec delay [ms]
G.711	PCM	64	20	0.125	0	0.125
G.723.1	MP-MLQ	5.33	30	30	7.5	37.5
G.723.1	ACELP	6.4	30	30	7.5	37.5
G.726	ADPCM	32	20	10	0	10
G.728	LD-CELP	16	30	0.625	0	0.625
G.729A	CS-ACELP	8	20	10	5	15

Packetization delay

$$T_{PD} = \frac{P_S}{C_{BW}} \qquad [ms]$$

Where :
T_{PD} – packetization delay [ms]
P_S – Payload size [b]
C_{BW} – Codec Bandwidth [kbit/s]

Serialization delay

$$T_{SER} = \frac{P_S + H_L}{L_S} \quad [ms]$$

Where :
T_{SER} – Serialization Delay [ms]
L_S – Line speed [kbit/s]
H_L – Header length [b]

Queuing delay

a designed model calculating queuing delay (jitter)

- PQ (Priority Queuing) optimalization
- servicing requirement technique in a priority queue responds to the model of queuing system *M/D/1/k*, where k is size of buffer
- this designed analytical model can ignore the buffer size hence *M/D/1/k* model can be replaced by ***M/D/1/∞*** model
- the voice traffic is modeled by source signal, which probabilistic random variable distribution matches Poisson's probability distribution
- $\lambda(t)$ is constant
- M sources using the same codec

Queuing Delay

system load:

$$\rho = \frac{\lambda}{\mu}$$

λ – arrival rate [s-*1]*
μ – service rate *[s-1]*
ρ – system load

for stability $0 \leq \rho < 1$

arrival rate is given by :

$$\lambda = \frac{C_{BW}}{P_S} \qquad [s^{-1}]$$

servise rate is given by:

$$\mu = \frac{1}{T_{SER} + T_S} \qquad [s^{-1}]$$

TSER – serialization delay [s]
TS – time of processing in router [s]

Queuing Delay

probability that k-attempts will be in system:

$$p_k = (1-\rho)$$ for k=0

$$p_k = (1-\rho)(e^{\rho} - 1)$$ for k=1

$$p_k = (1-\rho)\sum_{j=1}^{k} \frac{(-1)^{k-j}(j.\rho)^{n-j-1}(j.\rho + n - j)e^{j\rho}}{(n-j)!}$$ for k>2

Queuing Delay

$$p_k = \left(1 - M.C_{BW} \cdot \frac{P_S + H_S + L_S.T_S}{L_S.P_S}\right) \sum_{j=1}^{k} \left[(-1)^{(k-j)} \cdot \left(j.M.C_{BW} \cdot \frac{P_S + H_L + L_S.T_S}{P_S.L_S} \right)^{(n-j-1)} \cdot \cdot \left(j.M.C_{BW} \cdot \frac{P_S + H_L + L_S.T_S}{P_S.L_S} \right) \cdot \frac{e^{\left(j.M.C_{BW} \cdot \frac{P_S + H_L + L_S.T_S}{P_S.L_S} \right)}}{(n-j)!} \right]$$

for k≥2

$$p_k = \left(1 - M.C_{BW} \cdot \frac{P_S + H_S + L_S.T_S}{L_S.P_S}\right) \cdot \left(e^{\left(M.C_{BW} \cdot \frac{P_S + H_L + L_S.T_S}{P_S.L_S} \right)} - 1 \right)$$

for k=1

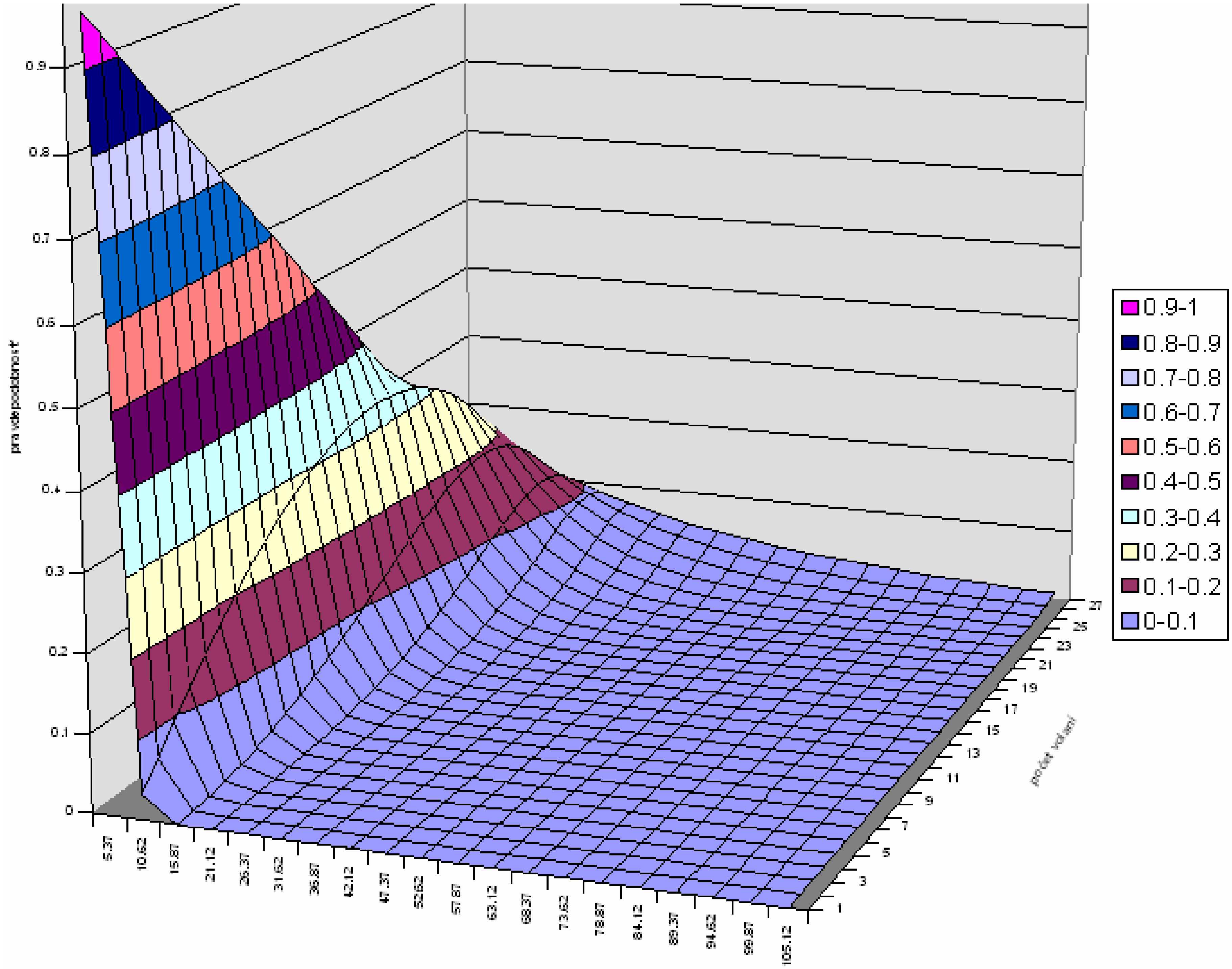
$$p_k = \left(1 - M.C_{BW} \cdot \frac{P_S + H_S + L_S.T_S}{L_S.P_S}\right)$$

for k=0

probability of delay is:

$$p_{Tk} = p_k \cdot \frac{P_S + H_L + L_S.T_S}{L_S}$$

for k = < 0 , ∞ >


pravdepodobnosť
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
5.37
10.62
15.87
21.12
26.37
31.62
36.87
42.12
47.37
52.62
57.87
63.12
68.37
73.62
78.87
84.12
89.37
94.62
99.87
105.12
1
3
5
7
9
11
13
15
17
19
21
23
25
27
0.9-1
0.8-0.9
0.7-0.8
0.6-0.7
0.5-0.6
0.4-0.5
0.3-0.4
0.2-0.3
0.1-0.2
0-0.1

Propagation delay

$$T_{Prop} = \frac{L}{v} \quad [ms]$$

Where:
T_{Prop} – propagation delay [ms]
L – line length [km]
v – light rate of spread in optical fiber = 2.07 . 10^{-8} [ms^{-1}]

De-jitter delay

- it is necessary to eliminate variance of these variable components with the help of supplementary buffer in receiver, **de-jitter** or **playout** bufer

Depacketization delay

- depacketization is done in opposite to the operation of packetization

Decompression delay

• decompression delay, likewise coder delay, is dependent on the compressing algorithm selection
• at average the decompression delay is approximately 10% of the compressing codec delay for each voice block in the packet

$$T_{DCD} = 0{,}1.N.T_{CD} \quad [ms]$$

TDCD – decomp. delay [ms]
N – num. of blocks in packet
TCD – coder delay [ms]

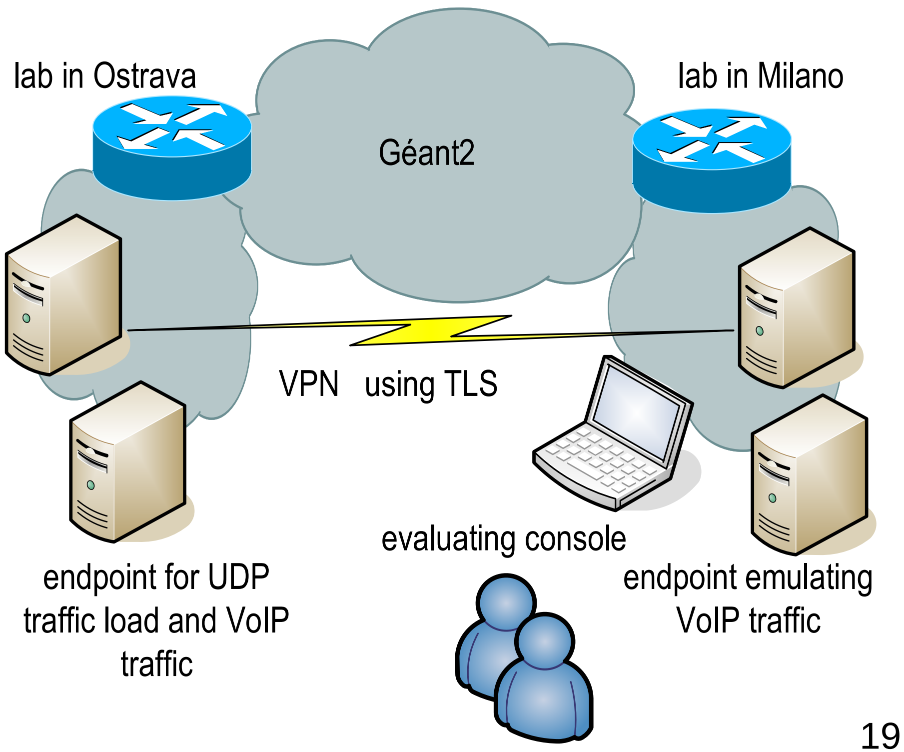
Verification

This described model is still under verification tests. We have gained and evaluated more than 5 thousands results in experimental measures and the aberration of our model is less than **6%** under condition that traffic load is less than **80%**.

Impact of Security on Quality

• We prepared a VoIP performance evaluation test between universities Ostrava and Milano and measured the throughput, transmission delay, packet loss, jitter under conditions of multiplexing VoIP and other data traffic.
• The whole traffic had been transmitted simultaneously with various combinations of the traffic situations and we were investigating an influence to the performance by using a background traffic and a security mechanisms.

• With regard to solved issue before we hade prepared an experimental infrastructure containing endpoints simulating VoIP traffic, a console evaluating the R-factor, endpoints generating a traffic for an investigation of the traffic load's effect and severs providing VPN by using TLS



Impact of Security on Quality

using components:

• Cisco 1751 in Ostrava (used for traffic shaping),
• Servers with Linux Debian (TLS client and server sides, Iperf client and server sides),
• Server with Linux Debian (endpoints of pairs emulating VoIP traffic),
• Notebook with WinXP and SW IxChariot (evaluating console).

conditions of experiment

• We prepared all required elements in Ostrava and we made decision to use only 6Mbps line so as not to cause a traffic congestion.

• Therefore we set the traffic shaping in Cisco router 1751 for the data traffic limitation.

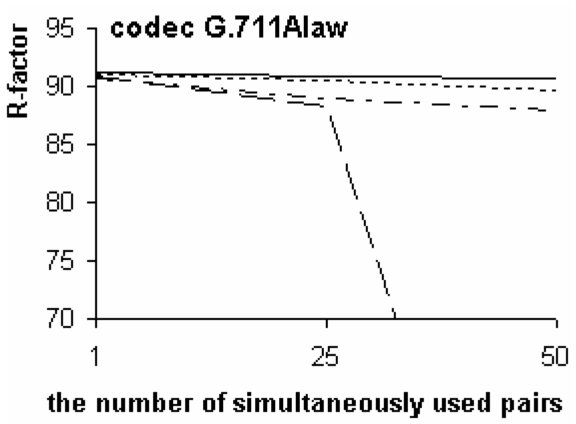
conditions of experiment

- traffic limitation (6Mbps, traffic shaping),
- used codecs G.711Alaw and G.729 (both sending 50 pps), without VAD
- dejitter buffer has been set to 60ms,
- traffic load has been set to 4Mbps,
- Security has been based on TLS (openSSL and openVPN).

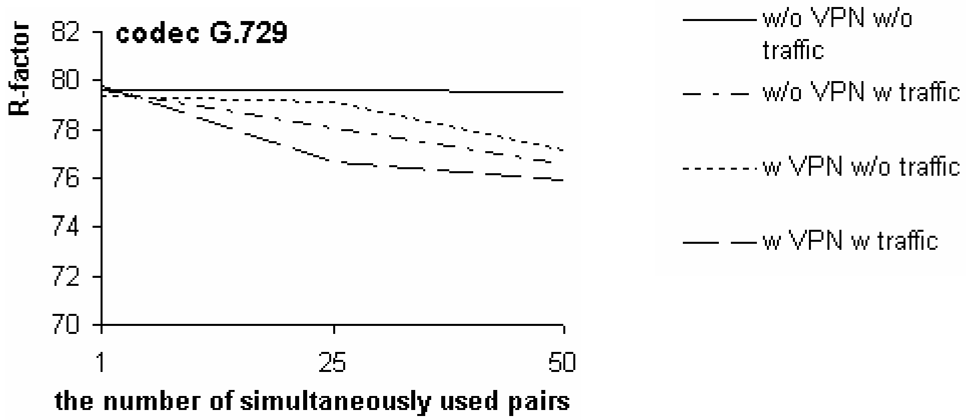
< 1 ms, cesare.laser.dico.unimi.it
2 ms,159.149.153.254
1 ms,ssr1-ssr7.bone.dsi.unimi.it
2 ms,159.149.251.21
2 ms,159.149.254.25
2 ms,ru-unimi-rt-mi3.mi3.garr.net
2 ms,rt-mi3-rt-mi1.mi1.garr.net
2 ms,garr.rt1.mil.it.geant2.net
14 ms,so-6-3-
0.rt1.vie.at.geant2.net
21 ms,so-7-0-
0.rt1.pra.cz.geant2.net
22ms,cesnet-
gw.rt1.pra.cz.geant2.net
29 ms,r96-r50.cesnet.cz
29 ms,iptel-21.osanet.cz

Results

We performed more than one hundred measurements, every measurement was repeated five times due to a suppression of the fault in measurement. In pictures 3 and 4 there are displayed the map curves for used codecs G.711Alaw and G.729.



R-factor for codec G.711.



R-factor for codec G.729.

Results

R-factor is arithmetic mean of all results obtained in one set of measurement.

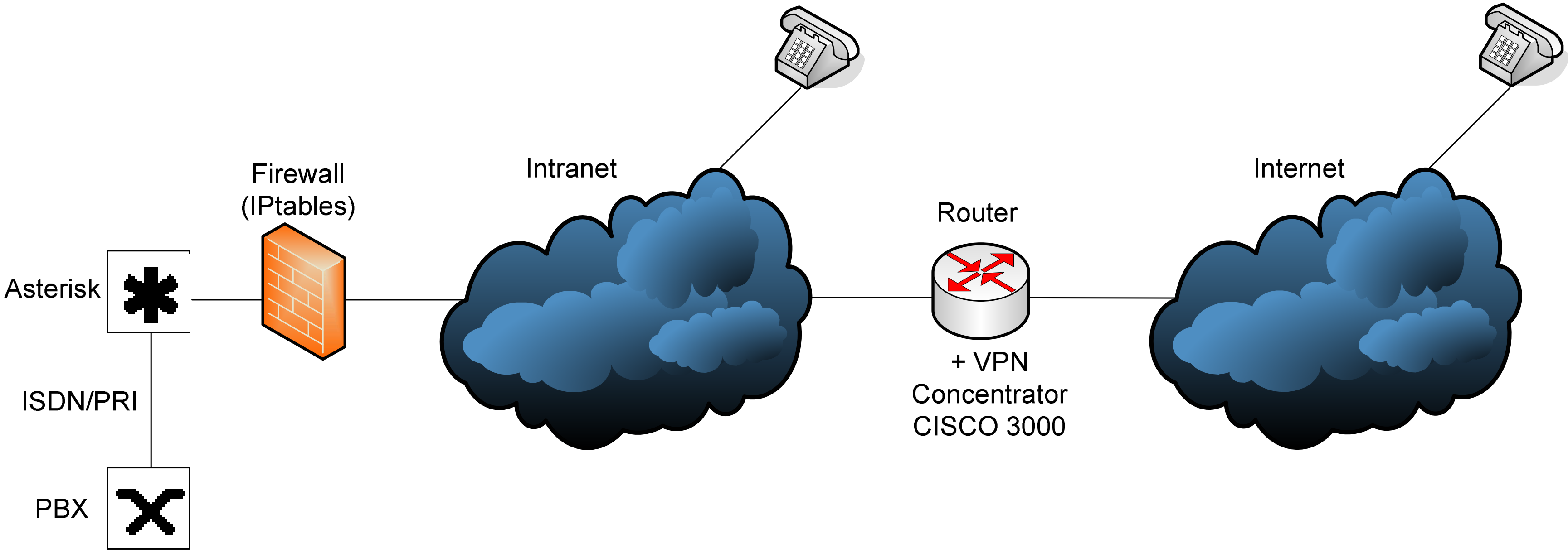
the absolute aberrance of measurement [%]		
	G.711Alaw	G.729
w/o VPN, w/o traffic	0,09	0,05
w/o VPN, w traffic	1,34	1,47
w VPN, w/o traffic	0,19	0,88
w WPN, w traffic	3,2	1,9
total 1,14%		

Table of the aberrances.

The total value of the absolute aberrance of measurement is 1,14 %.

If we compare the evaluated values so we can claim that used security mechanism TLS **is affecting** R-factor although this influence is not so significant how we expected. This influence is ranging from 1% to 5%.

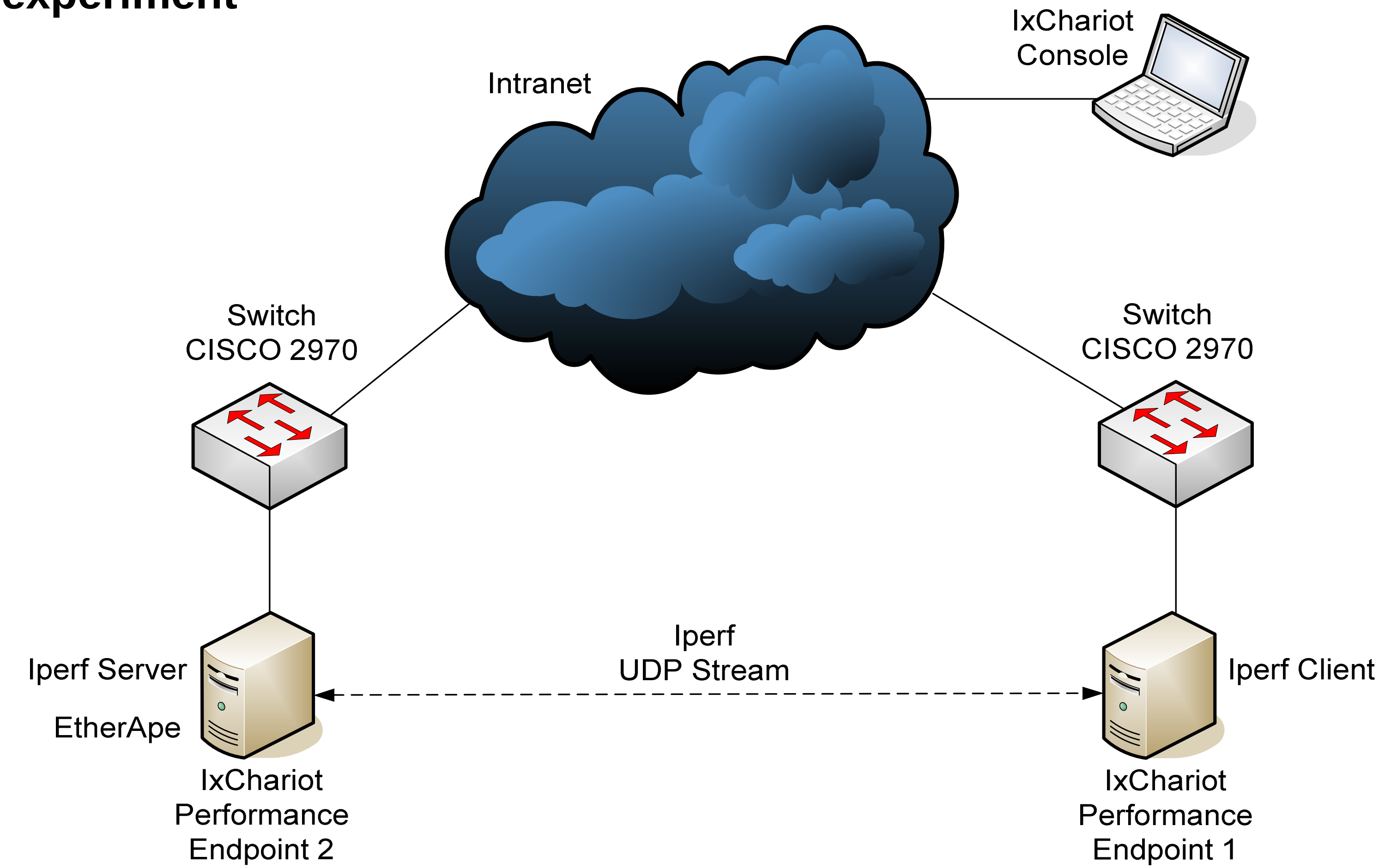
Impact of Security on Quality



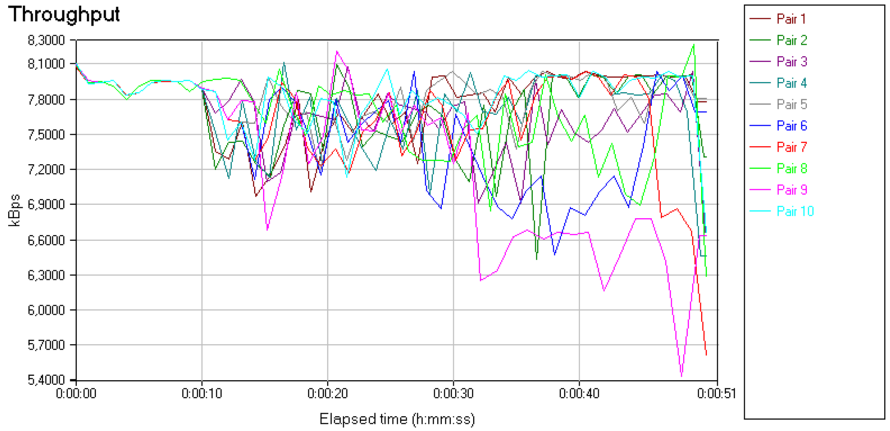
- delay
- jitter
- packet loss

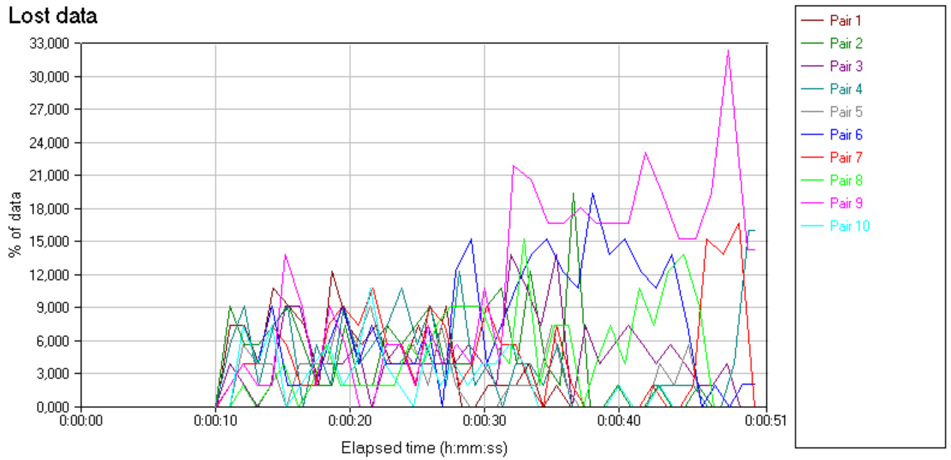
Impact of Security on Quality

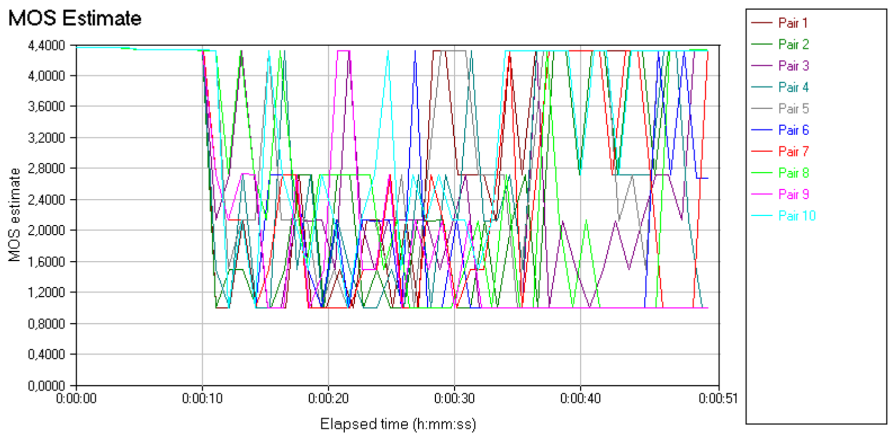
experiment

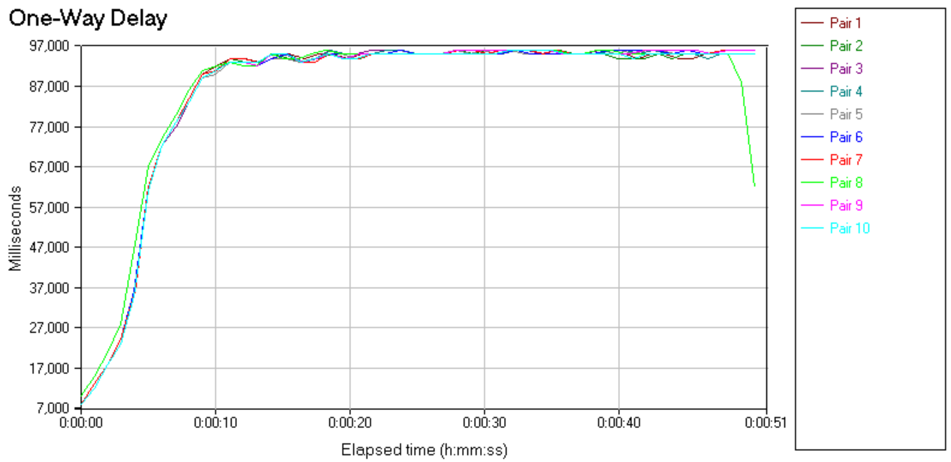


examples of results









Impact of IPtables

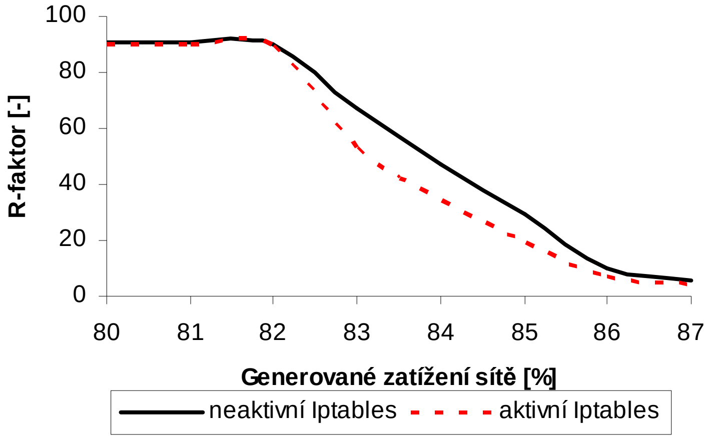
- without IPtables

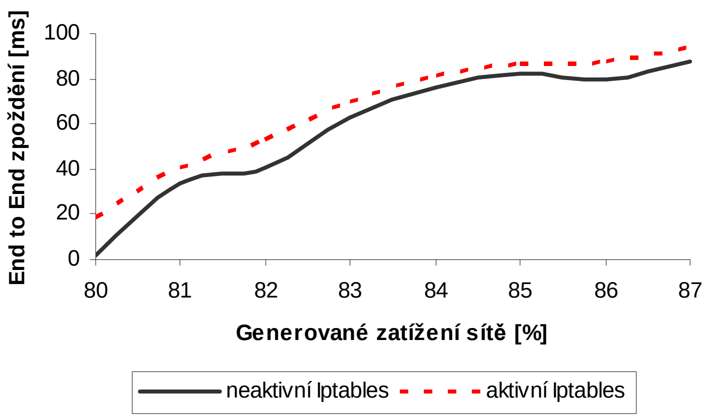
Zatížení sítě	Rfaktor	End-End Delay	Ztráty
[%]	[-]	[ms]	[%]
0	91	2	0
10	91	2	0
20	91	2	0
30	91	2	0
40	91	2	0
50	91	2	0
60	91	2	0
70	91	2	0
80	91	2	0
81	91	34	0
82	90	41	0
83	67	63	3
84	47	76	6
85	29	82	10
86	10	80	13
87	6	88	18

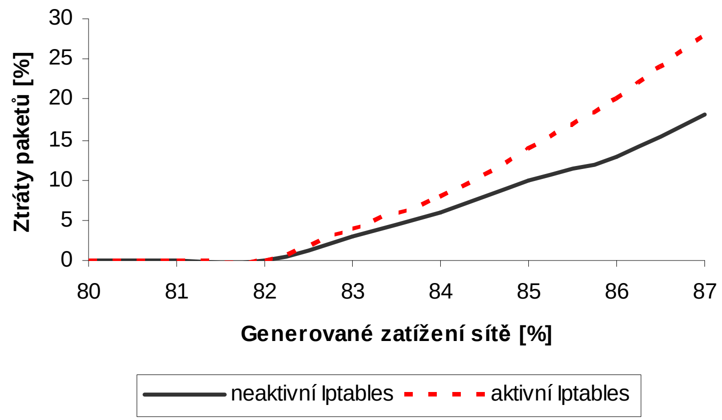
- with active IPtables

Zatížení sítě	Rfaktor	End-End Delay	Ztráty
[%]	[-]	[ms]	[%]
0	91	2	0
10	91	2	0
20	91	2	0
30	91	2	0
40	91	2	0
50	91	2	0
60	91	2	0
70	91	2	0
80	90	19	0
81	90	41	0
82	89	53	0
83	53	70	4
84	34	81	8
85	19	87	14
86	7	88	20
87	4	94	28

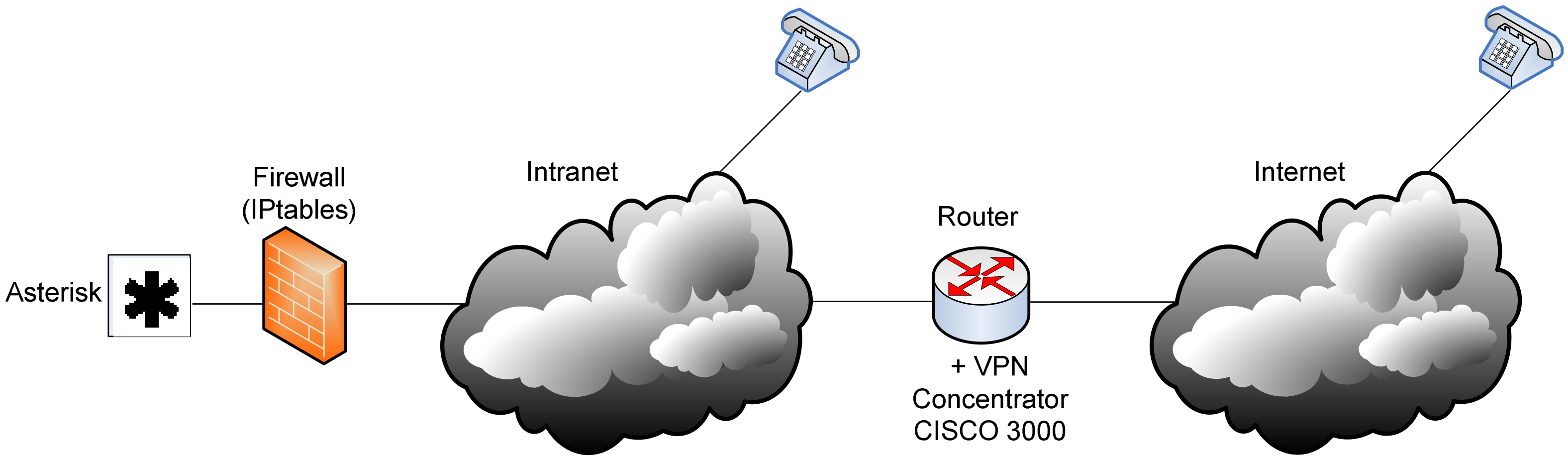
Impact of IPtables





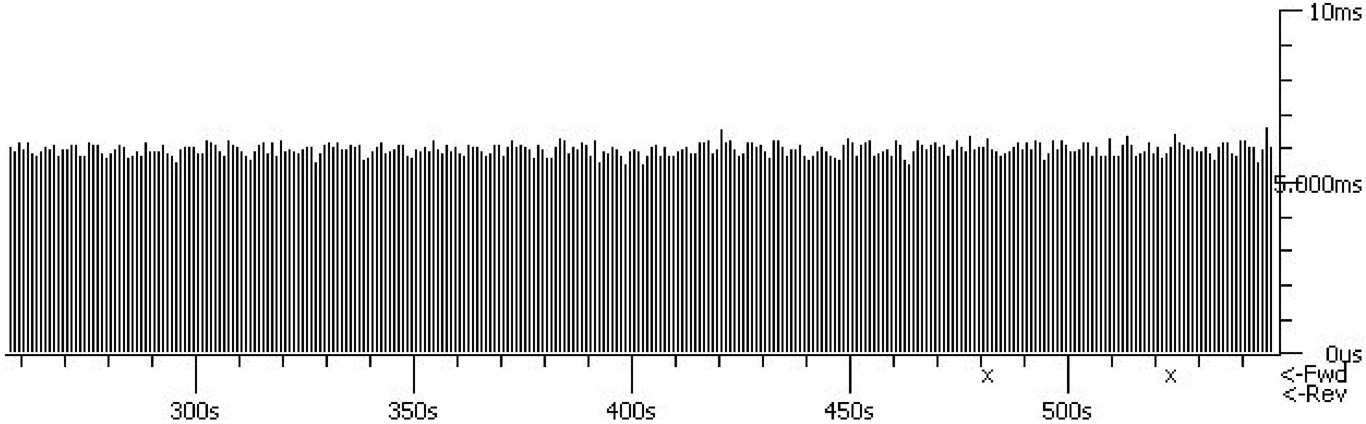


Impact of VPN

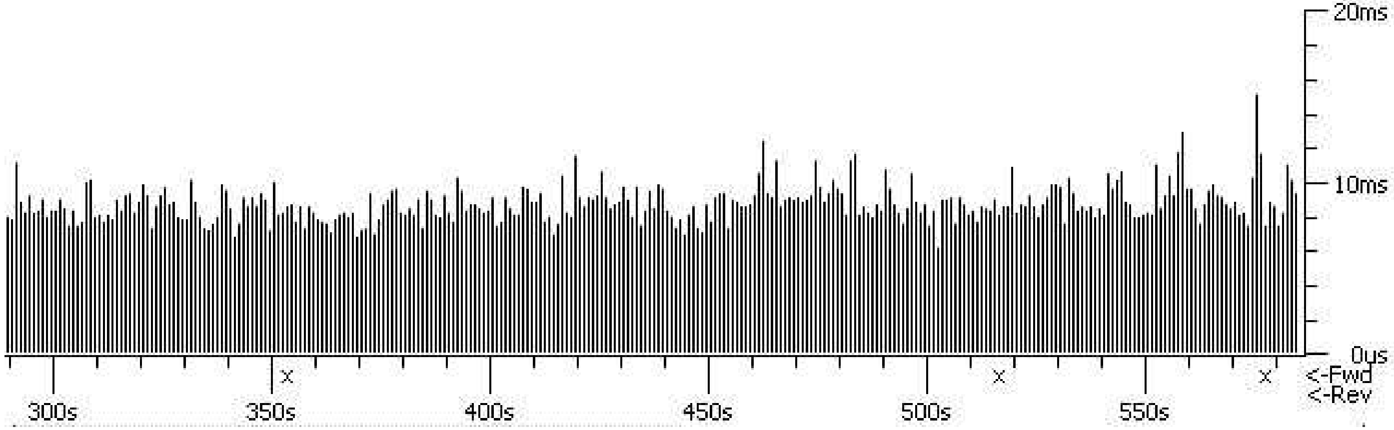


Jitter

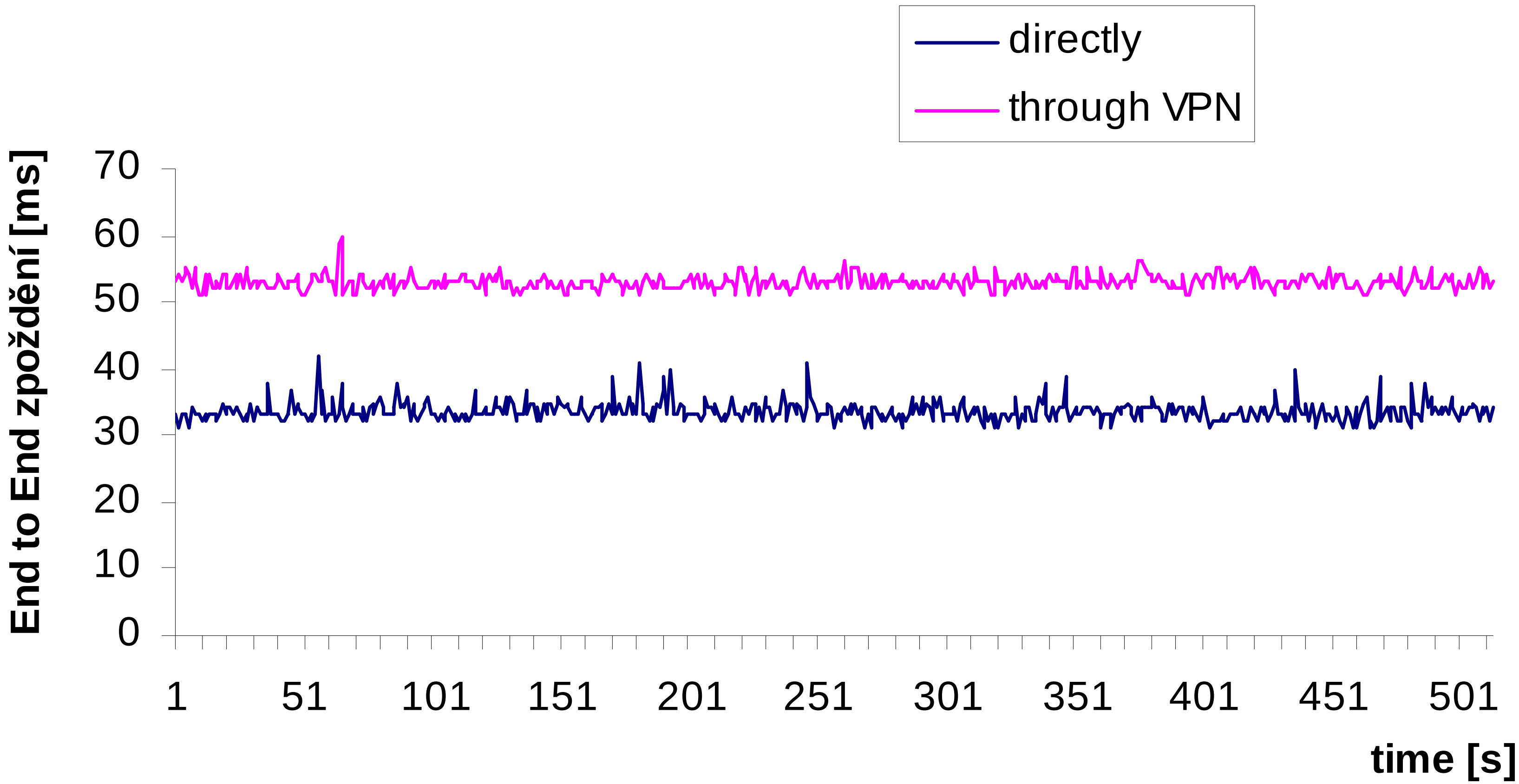
- without VPN



- with VPN



End to End delay



Thank you for your attention

miroslav.voznak@vsb.cz