Ethernet Transport Performance and Triple-Play QoS in Large Ship Environment

Performanse Ethernet prijenosa i Triple-Play kvaliteta usluge na velikim brodovima

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Summary

Long lasting Ethernet standards have grown beyond traditional LAN networks through MAN to WAN environment, where it has become the technology which many providers use for developing convergent networks that transport new generation services, such as triple-play widely deployed everywhere – from airports and hotels to large cruise ships. The question of network suitability for providing various communication services has to be considered in practice, especially in case of large ships, as the content of traffic is very diverse with exceptionally variable intensity. Under such specific circumstances for ships theoretical models cannot be directly applied. Rather, network design and implementation depend on practical preliminary test results and analysis. To this end, the out-of-service simulation of either network or user equipment was done according to the RFC 2544 recommendation by means of a smart test device that generates the appropriate test traffic over the Ethernet link from a 3G base station (NodeB) towards the Radio Network Controller (RNC). Network troughput, latency, jitter and frame loss were measured, thus providing the picture of soundness of Ethernet transport layers. However, even the best results in this respect do not necesarily imply a high quality QoS at the application layer. Therefore, the end-to-end QoS was tested specifically for concurrent data and for VoIP applications.

KEY WORDS
Large ship network
QoS
Ethernet
VoIP

Sažetak

Dugotrajni Ethernet standardi nadilaze tradicionalne LAN mreže i primjenjuju se, ne samo u MAN mrežama, nego i u WAN okolini, gdje Carrier Ethernet postaje tehnologija koju mnogi pružatelji usluga koriste za razvoj konvergentnih mreža koje prenose servise nove generacije, kakav je triple-play, široko rasprostranjen – od zračnih luka i hotela, do velikih brodova za kružna putovanja. Tako je pitanje podobnosti mreže za prijenos različitih komunikacijskih usluga, osobito kada su veliki brodovi u pitanju, važno razmotriti u praksi, pošto je sadržajno promet veoma raznolik, a vremenski izuzetno promjenjivog intenziteta. U ovim posebnim brodskim uvjetima, teorijski modeli ne mogu se izravno primijeniti, već projektiranje mreže i implementacija zavise od rezultata prethodnih praktičnih ispitivanja i analize. S tim ciljem izvršena je simulacija izvan radnog režima mrežne ili korisničke opreme, a u skladu s preporukom RFC-a 2544, uz pomoć inteligentnog ispitnog uređaja kojim je generiran pogodni ispitni promet putem oglednog Ethernet linka između 3G bazne postaje (Čvor B) i radijske upravljačke jedinice (RNC). Mrežna propusnost, kašnjenje, varijabilnost kašnjenja i učestalost gubitka okvira su mjereni, dajući sliku zdravlja Ethernet transportnih slojeva. Međutim, kako ni najbolji rezultati u ovome primjeru ne jamče nužno i visoku kvalitetu usluge na aplikacijskom sloju, ona je ispitana u kontekstu konkurentnih podatkovnih i VoIP aplikacija.

KLJUČNE RIJEČI
mreža na velikim brodovima
kvaliteta usluge
Ethernet
VoIP

INTRODUCTION / *Uvod*

The question of network suitability for providing various communication services must be considered in practice. With large ships especially, the traffic content is very diverse, ranging from voice and video to data transmission (which are commonly referred to as triple-play services). In addition, traffic intensity could be exceptionally variable, tracking not only quite diverse passenger activites throughout the daytime, but also depending on whether the ship is anchored in a port and many

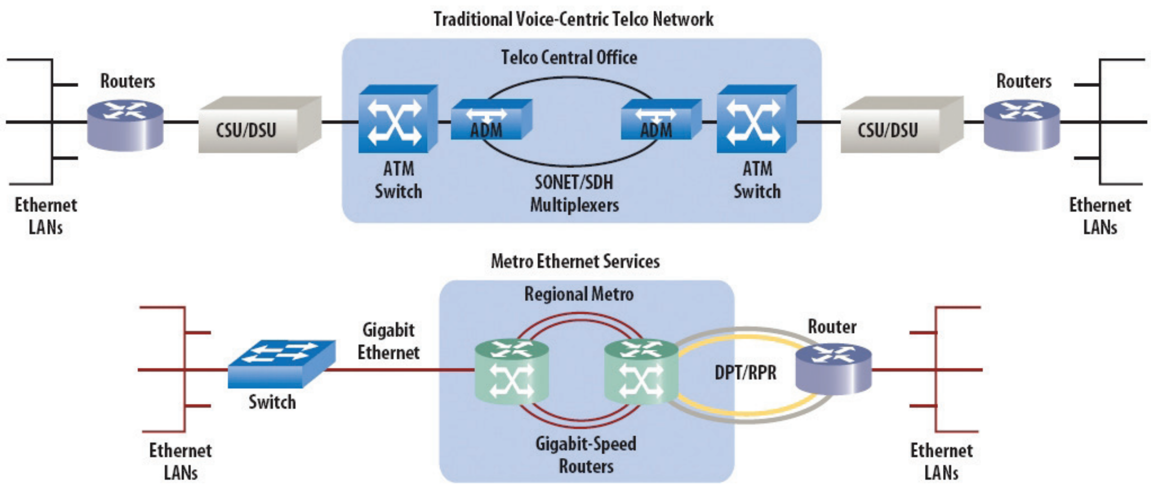


Figure1 Traditional and Ethernet network architecture
Slika 1. Arhitektura tradicionalne i Ethernet mreže

passengers are not onboard, or the ship is in cruising phase, when the passenger demand for network services usually increases significantly. Under such special circumstances of large ship environment, theoretical models do not apply in straightforward manner and practical tests or at least modeling with industry standard tools must be conducted before the actual network services are made available to passengers. With this regard, the quality of service (QoS) of such specific networks in large ship environment should be tested whether it can accomodate not only sustainable traffic utilization, but also expected traffic bursts.

In this respect, let us consider the long lasting Ethernet and related network standards that have grown beyond traditional LAN networks continuing their evolution through MAN and even WAN environment. Ethernet has become the technology which many providers use for developing convergent networks that transport new generation services [1-2], such as triple-play. What definitively contributes to this trend is a high standardization, scalability and IP-friendliness of Ethernet, which is not found with legacy voice-centric Synchronous Transport Hierarchy (SDH) networks, Fig. 1.

For many years Ethernet has been widely accepted as it provides accessable bandwidth on demand and secure communication of private networks. However, in some instances, telecom service providers hesitate to fully implement Ethernet in their transport networks that offer triple-play services as it brings more complex QoS management resulting in numerious technical and operational challenges.

Specifically, for delivery of triple-play services, Ethernet can serve as aggregation network enabling convergence of voice, video and data - all with adequate transmission performance. On the network management side, however, this enables simple scalability, reliability, recognition of Service Level Agreement (SLA) key performance indicators (such as connectivity, bandwidth usage, packet loss and jitter), and finally convergence of various network services. No doubdt that these new challenges on Ethernet aggregation networks increase demands on network quality-of-service (QoS), specifically with regard to video and Voice-over-IP transmission, where e.g. lost packets or slow internet connection can produce dissatisfaction of users, Table 1.

Table 1 Recommended limits on delay, delay variation, availability and accuracy of synchronization for Carrier Ethernet, SDH and mobile telephony networks
Tablica 1. Preporučena ograničenja za kašnjenje, varijaciju kašnjenja, dostupnost i točnost sinkronizacije za Carrier Ethernet, SDH i mreže mobilne telefonije

Parameter	Ethernet	SDH	Mobile telephony
Delay	< 25 ms	< 100 ms	< 5 ms
Jitter	< 10 ms	< 3,2 μs	< 1 ms
Availability	99,95 % < 263 min/year	99,999 % < 5,3 min/year	99,999 % < 5,3 min/year
Network synchronization accuracy	< 50 ppb *	< 50 ppb *	<50 ppb
	* achieved by means of high-quality oscillators		

KEY ETHERNET PERFORMANCE INDICATORS ACCORDING TO RFC 2544 RECOMMENDATION / *Ključni pokazatelji performanse Etherneta sukladno preporuci RFC 2544*

Now let us review the key practical performance indicators that need to be tested with practical Ethernet networks: throughput, latency and frame loss.

Definition of data throughput is not straightforward as it is determined by prior specification of an acceptable QoS. For example, if 10% traffic is either errored or with lost frames, then throughput would be accordingly tested (tolerating 10 % such transmission errors). Furthermore, the absolute throughput maximum that is equal to nominal data rate e.g. 100 Mbit/s or 1 Gbit/s, cannot be reached beacause of the so called frame lenght effect, as transmitting shorter frames implies more redundancy (control overhead of non-data interframe spacing), Table 2 and Table 3. The RFC 2544 recommendation requires testing for all standard frame lengths (64, 128, 256, 512, 1024, 1280 and 1518 bytes).

Table 2 100BaseT Ethernet throughput [3]
Tablica 2. Propusnost Etherneta 100BaseT [3]

Frame Size	Data Throughput	Preamble & IGP	Frames per sec
64 byte	76.19 Mbit/s	23.81 Mbit/s	148,809
128 byte	86.49 Mbit/s	13.51 Mbit/s	84,459
256 byte	92.75 Mbit/s	7.25 Mbit/s	45,289
512 byte	96.24 Mbit/s	3.76 Mbit/s	23,496
1024 byte	98.08 Mbit/s	1.92 Mbit/s	11,973
1280 byte	98.46 Mbit/s	1.54 Mbit/s	9,615
1518 byte	98.69 Mbit/s	1.30 Mbit/s	8,127
1522 byte (includes VLAN)	98.70 Mbit/s	1.30 Mbit/s	8,106

Table 3 1000BaseT Ethernet throughput [3]
Tablica 3. Propusnost Etherneta 1000BaseT [3]

Frame Size	Data Throughput	Preamble & IGP	Frames per sec
64 byte	761.90 Mbit/s	238.10 Mbit/s	1,488,095
128 byte	864.86 Mbit/s	135.14 Mbit/s	844,594
256 byte	927.54 Mbit/s	72.46 Mbit/s	452,898
512 byte	962.40 Mbit/s	37.59 Mbit/s	234,962
1024 byte	980.84 Mbit/s	19.16 Mbit/s	119,731
1280 byte	984.61 Mbit/s	15.38 Mbit/s	96,153
1518 byte	986.99 Mbit/s	13.00 Mbit/s	81,274
1522 byte (includes VLAN)	987.02 Mbit/s	12.97 Mbit/s	81,063

On the other hand, end-to-end delay includes overall time it takes the frame to travel its source to destination, so it is effectively the sum of processing times in network elements and propagation time along the transmission medium. To measure it, a test frame is transmitted that contains time stamp, which is checked once the frame arrives at the receiver.

Furthermore, frame loss is the ratio of frames successfully transmitted but never received at a destination. This can occur due to various reasons, among them transmission errors, over-subsription and excessive transmission delay.

Moreover, frame errors occur when data link layer devices (such as e.g. switches and bridges) discard frames with incorrect frame-check-sequence (FCS), which can be produced by even a single errored bit, or oversubscription of available (insufficient) bandwidth leading to discards of frames of some subscribers. So, instead of elsewhere (SDH) relevant bit-error-ratio (BER), frame-error-ratio (FER) is relevant in Ethernet environment.

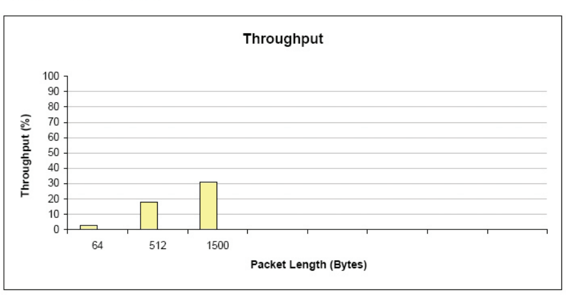
Excessive delays are not compatible with native Ethernet access procedure (CSMA-CD) and can produce frame discards. This has to be taken care of during the testing, as the receiving test device must wait for a certain time period for all sent frames to be received and taken into account. However, at certain predefined time instant, the test device must have a threshold that determines whether a sent frame will not be received and so accounted as a lost one. Most frequently used waiting time is 2 seconds.

Finally, the recommendation RFC 2544 [4] also strongly suggests that tests include back-to-back frame loss ratio, which is sending bursts of frames with minimal interframe spacing towards the device under test (DUT) and counting the ones returned by the DUT. If the count of transmitted frames equals the count of returned frames, the burst length is increased and the test restarted. Analogously, if the count of received (i.e. forwarded) frames is lesser than the count of the returned ones, the burst length is reduced and the test started over. The outcome of this procedure is the count of frames of the longest burst that the DUT can handle without losing any frame.

TESTING ETHERNET TRANSPORT PERFORMANCE / *Ispitivanje performanse Ethernet prijenosa*

The RFC 2544 testing could be done manually, but this is quite tedious, time consuming and prone to errors. So it is easier and more reliable to conduct fully automatic tests, which enable users of network services to simply input details of desired test scenarios and then start the tests and quickly obtain needed results.

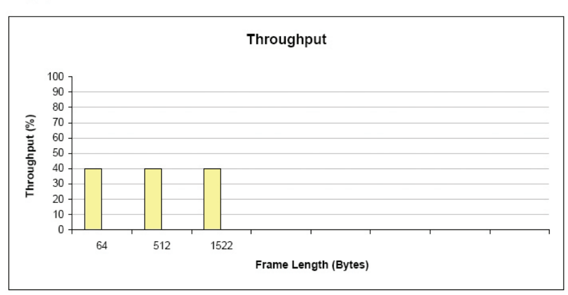
Throughput Test Results:



Packet Length (Bytes)	Cfg Rate (Mbps)	Measured Rate (Mbps)	Measured Rate (%)	Measured Rate (frms/sec)	Pause Detected
64	3,00	3,00	3,01	3538,00	No
512	18,00	18,00	18,01	4062,00	No
1500	31,00	31,00	31,01	2513,00	No

Fig. 2 Testing throughput of Ethernet link for standard frame lengths (test: fail)
Slika 2. Ispitivanje propusnosti Ethernet veze za standardne duljine okvira (rezultat: ne zadovoljava)

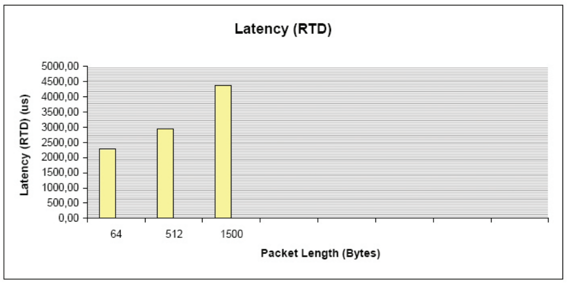
Throughput Test Results:



Frame Length (Bytes)	Cfg Rate (Mbps)	Measured Rate (Mbps)	Measured Rate (%)	Measured Rate (frms/sec)	Pause Detected
64	40,00	40,00	40	59524,00	No
512	40,00	40,00	40	9399,00	No
1522	40,00	40,00	40,01	3243,00	No

Fig. 3 Testing throughput of Ethernet link for standard frame lengths (test: pass)
Slika 3. Ispitivanje propusnosti Ethernet veze za standardne duljine okvira (rezultat: zadovoljava)

Latency (RTD) Results

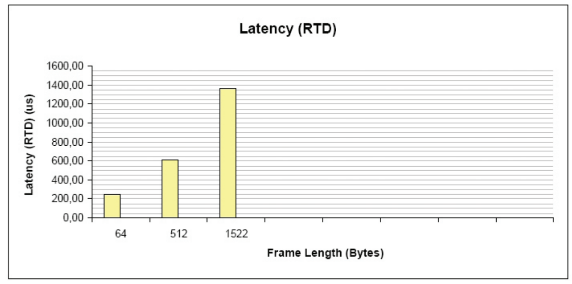


Packet Length (Bytes)	Latency (RTD) (us)	Measured Rate (Mbps)	Measured Rate (%)	Measured Rate (frms/sec)	Pause Detected
64	2285,10	3,00	3,01	3538,00	No
512	2951,20	18,00	18,01	4062,00	No
1500	4368,50	31,00	31,01	2513,00	No

Fig. 4 Testing latency of Ethernet link for standard frame lengths (test: fail)

Slika 4. Ispitivanje kašnjenja Ethernet veze za standardne duljine okvira (rezultat: ne zadovoljava)

Latency (RTD) Results



Frame Length (Bytes)	Latency (RTD) (us)	Measured Rate (Mbps)	Measured Rate (%)	Measured Rate (frms/sec)	Pause Detected
64	249,50	40,00	40	59524,00	No
512	607,20	40,00	40	9399,00	No
1522	1364,60	40,00	40,01	3243,00	No

Fig. 5 Testing latency of Ethernet link for standard frame lengths (test: pass)

Slika 5. Ispitivanje kašnjenja Ethernet veze za standardne duljine okvira (rezultat: zadovoljava)

Avg. and Max. Avg. Pkt. Jitter Test Results:

Packet Length (Bytes)	Latency (RTD) (us)	Measured Rate (Mbps)	Measured Rate (%)	Measured Rate (frms/sec)	Pause Detected	
64	Avg 0 Max Avg 0	3,00	3,01	3538,00	No	N/A
512	Avg 0 Max Avg 0	18,00	18,01	4062,00	No	N/A
1500	Avg 0 Max Avg 0	31,00	31,01	2513,00	No	N/A

Fig. 6 Testing average and maximum frame jitter of Ethernet link for standard frame lengths (test: fail)

Slika 6. Ispitivanje prosječne i maksimalne varijabilnosti kašnjenja Ethernet veze za standardne duljine okvira (rezultat: ne zadovoljava)

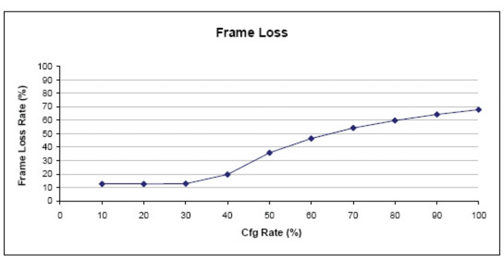
Avg. and Max. Avg. Pkt. Jitter Test Results:

Frame Length (Bytes)	Latency (RTD) (us)	Measured Rate (Mbps)	Measured Rate (%)	Measured Rate (frms/sec)	Pause Detected
64	Avg 0 Max Avg 0	40,00	40,00	59524,00	No
512	Avg 0 Max Avg 0	40,00	40,00	9399,00	No
1522	Avg 0 Max Avg 0	40,00	40,01	3243,00	No

Fig. 7 Testing average and maximum frame jitter of Ethernet link for standard frame lengths (test: pass)

Slika 7. Ispitivanje prosječne i maksimalne varijabilnosti kašnjenja Ethernet veze za standardne duljine okvira (rezultat: zadovoljava)

Frame Loss Test Results: **64 byte packets:**

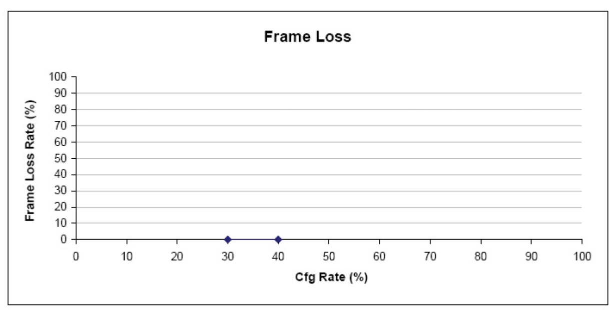


Cfg Rate (%)	Throughput Rate (%)	Frame Loss Rate (%)	Frames Lost	Pause Detected
100	32,416	67,8	4974093,00	No
90	32,467	64,25	4250148,00	No
80	32,461	59,76	3504380,00	No
70	32,289	54,1	2786192,00	No
60	32,454	46,42	2045141,00	No
50	32,354	35,84	1315273,00	No
40	32,344	19,64	577006,00	No
30	26,28	12,93	286249,00	No
20	17,642	12,52	184188,00	No
10	8,975	12,58	92290,00	No

Fig. 8 Testing frame loss ratio of Ethernet link with 64 bytes frame length (test: fail)

Slika 8. Ispitivanje učestalosti gubitka okvira Ethernet veze s duljinom okvira od 64 bajta (rezultat: ne zadovoljava)

Frame Loss Test Results: **64 byte frames:**



Cfg Rate (%)	Throughput Rate (%)	Frame Loss Rate (%)	Frames Lost	Pause Detected
40	40,001	0	0,00	No
30	30,001	0	0,00	No

Fig. 9 Testing frame loss ratio of Ethernet link with 64 bytes frame length (test: pass)

Slika 9. Ispitivanje učestalosti gubitka okvira Ethernet veze s duljinom okvira od 64 bajta (rezultat: zadovoljava)

Several key areas for Carrier Ethernet involve various protocol stack layers and include: signaling (call connection), conformance of actual traffic parameter values with the SLA defined target values, interoperability, traffic ststistics, as well as meeting performance goals through protocol stack layers all the way upwards to the application itself. This is in accordance with the so-called bottom-up testing model, which usually presumes starting tests at the lowest (physical) OSI layer, followed by link layer and then network layer as well as transport layer related performance and QoS tests, specifically tracking those whose (negative) effects propagate upwards on the stack.With this respect, both in-service and out-of-service tests are conducted, where the latter is often the only reliable option.

With this regard, we performed out-of-service simulation of either network or user equipment by generating appropriate test traffic (complying to RFC 2544) over the Ethernet link from a 3G base station (NodeB) towards the Radio Network Controller (RNC), all over the following hops: NodeB – MUX SDH – CISCO MAN 3400 – RNC, using the JDSU Smart Class Ethernet v. 3.0.0. test device.

The measurement results are presented on Figs. 2 – 15. The overall link characterization result could finally be FAIL or PASS.

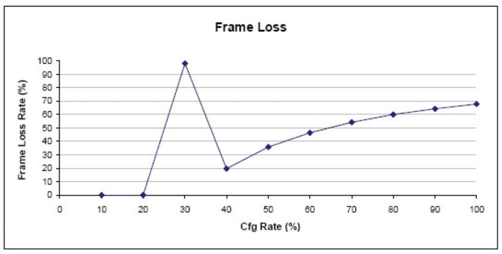
Based on the Figs. 2 - 3, we identify throughput variations with Ethernet frame length (standard values of 64, 512 and 1500 bytes). However, after eliminating the cause of network performance degradation, the repeated testing revealed constant throughput for various frame lengths.

Similar observations apply to latency and jitter testing, as well as frame loss ratio (back-to-back, too), Figs. 4 - 15.

TESTING TRIPLE-PLAY APPLICATION LAYER QOS / *Ispitivanje Triple-Play kvalitete usluge aplikacijskog sloja*

Once QoS of Ethernet layers (the two lowest) is tested, if the achieved results are good, it does not necessarily imply that application layer end-to-end QoS is also good. This is especially

Frame Loss Test Results: **512 byte packets:**

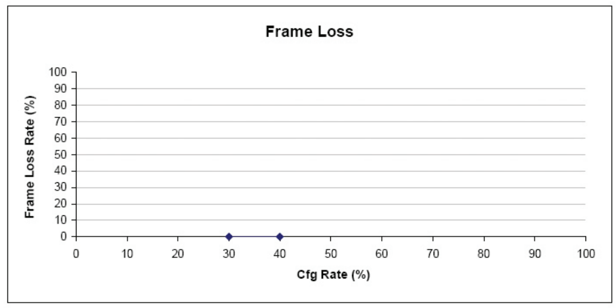


Cfg Rate (%)	Throughput Rate (%)	Frame Loss Rate (%)	Frames Lost	Pause Detected
100	32,327	67,85	959722,00	No
90	32,292	64,26	810798,00	No
80	32,274	59,85	671400,00	No
70	32,307	54,1	533185,00	No
60	32,318	46,45	391387,00	No
50	32,3	35,7	250753,00	No
40	32,317	19,59	110028,00	No
30	0,697	98,08	413269,00	No
20	20,199	0	2,00	No
10	10,009	0	0,00	No

Fig. 10 Testing frame loss ratio of Ethernet link with 512 bytes frame length (test: fail)

Slika 10. Ispitivanje učestalosti gubitka okvira Ethernet veze s duljinom okvira od 512 bajta (rezultat: ne zadovoljava)

Frame Loss Test Results: **512 byte frames:**

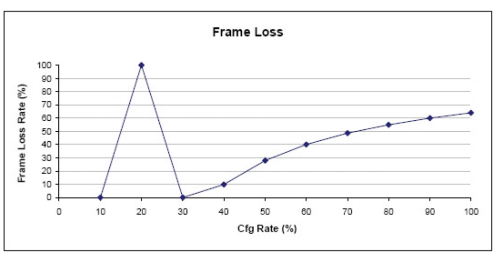


Cfg Rate (%)	Throughput Rate (%)	Frame Loss Rate (%)	Frames Lost	Pause Detected
40	40,003	0	0,00	No
30	30,002	0	0,00	No

Fig. 11 Testing frame loss ratio of Ethernet link with 512 bytes frame length (test: pass)

Slika 11. Ispitivanje učestalosti gubitka okvira Ethernet veze s duljinom okvira od 512 Bajta (rezultat: zadovoljava)

Frame Loss Test Results: **1500 byte packets:**

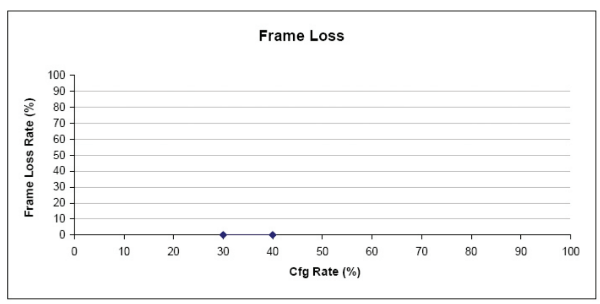


Cfg Rate (%)	Throughput Rate (%)	Frame Loss Rate (%)	Frames Lost	Pause Detected
100	36,017	63,95	322754,00	No
90	36,017	59,95	273985,00	No
80	36,015	54,94	221907,00	No
70	36,182	48,51	171394,00	No
60	36,062	39,93	121006,00	No
50	36,16	27,91	70305,00	No
40	36,017	9,91	19994,00	No
30	30,261	0	0,00	No
20	0,037	100	100633,00	No
10	10,012	0	0,00	No

Fig. 12 Testing frame loss ratio of Ethernet link with 1500 bytes frame length (test: fail)

Slika 12. Ispitivanje učestalosti gubitka okvira Ethernet veze s duljinom okvira od 1500 bajta (rezultat: ne zadovoljava)

Frame Loss Test Results: **1522 byte frames:**



Cfg Rate (%)	Throughput Rate (%)	Frame Loss Rate (%)	Frames Lost	Pause Detected
40	40,006	0	0,00	No
30	30,002	0	0,00	No

Fig. 13 Testing frame loss ratio of Ethernet link with 1522 bytes frame length (test: pass)

Slika 13. Ispitivanje učestalosti gubitka okvira Ethernet veze s duljinom okvira od 1522 bajta (rezultat: zadovoljava)

Back to Back Test Results:

Packet Length (Bytes)	Average Burst (frms)	Average Burst (secs)	Pause Detected
64	80	0,001	No
512	10	< 0.001	No
1500	50	0,006	No

Fig. 14 Testing Ethernet link parameters for standard frame lengths (test: fail)

Slika 14. Ispitivanje parametara Ethernet veze za standardne duljine okvira (rezultat: ne zadovoljava)

Back to Back Test Results:

Frame Length (Bytes)	Average Burst (frms)	Average Burst (secs)	Pause Detected
64	193	0,001	No
512	159	0,007	No
1522	53	0,007	No

Fig. 15 Testing Ethernet link parameters for standard frame lengths (test: pass)

Slika 15 Ispitivanje parametara Ethernet veze za standardne duljine okvira (rezultat: zadovoljava) questionable for diverse applications such as the triple-play ones. Therefore, in addition to Ethernet transport tests that have been reviewed so far, we conducted the appropriate tests of mutual impact of voice and data application performance, specifically the impact of introducing Voice-over-Internet-Protocol (VoIP) on the QoS of concurrent business applications being transported by convergent network platform of a major service provider [5-8].

It is common practice that prior to implementation of new network solutions either in physical or logical topology, it is necessary to conduct detailed and overall analysis of existing network environment. This includes not only quantitative analysis providing percentages of use of particular network resources (devices and links), it is desirable to get insight into types of services and applications that use the network in subject. Only then the planning of new network services can be done based on solid prediction on how justifiable investment would be.

In the following part we present the test results of link utilization at the central switch during specific time intervals.

The reports are grouped according to device type. So e.g. application servers with installed Oracle data base and client-server applications provide users with possibility to access data inquiry/modification/reading via network environment. With such centralized application system, the code must be optimized to enable parallel work of as many users as possible, as well as that server hardware resources support sufficient number of concurrent data processing tasks. The goal of these tests was to find out whether or not it is necessary to invest in corporate Ethernet LAN links capacity extension in order to accommodate the potentially demanding new service – VoIP.

The network traffic analysis is done by means of a dedicated software tool [9].

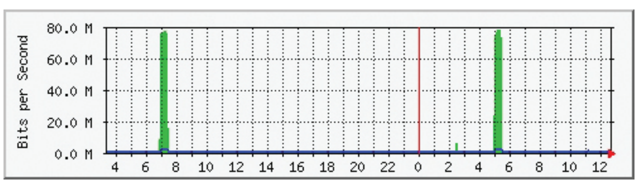
ETHERNET TRAFFIC ANALYSIS BEFORE INTRODUCING VOIP SERVICE / *Analiza Ethernet prometa prije uvođenja VoIP usluge*

The statistics of network utilization for the particular application servers is presented in Figs. 16 – 17, and for authentication and collaboration servers on Figs. 17 – 21, respectively.

Based on the above diagrams for network links of data servers, the average utilization during work time is measured to be equal to 1.89 %. Particularly, the utilization of the outgoing link was 0.27 %. Maximum utilization has been identified as occurring within the period of 2 to 6 pm, specifically during network back up of all servers in Ethernet LAN network. However, during other work time periods of daytime, network links toward servers have a relatively low usage.

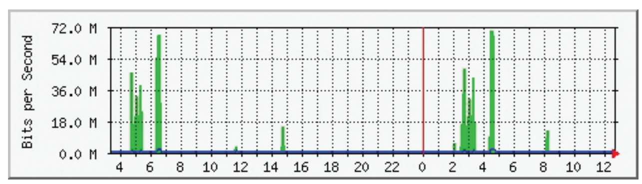
On the other hand, the analysis of network links of servers dedicated for authentication of users in Windows domains, reveals that the average link utilization during work time was 0.3 %, while the average utilization of the outgoing link was 0.028 %.

Overall, top utilization occurred from of 2 to 6 pm, specifically during network back up of all servers, with low



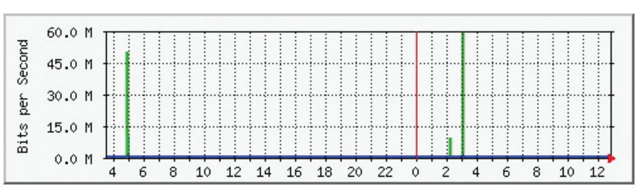
	Maximal	Average	Instantaneous
Outgoing	78.3 Mb/s (78.3%)	2128.8 Kb/s (2.1%)	192.5 Kb/s (0.2%)
Incoming	1781.5 Kb/s (1.8%)	159.4 Kb/s (0.2%)	305.4 Kb/s (0.3%)

Fig. 16 Statistics of network utilization of the network management application server
Slika 16. Statistika mrežne uporabe poslužitelja aplikacije upravljanja mrežom



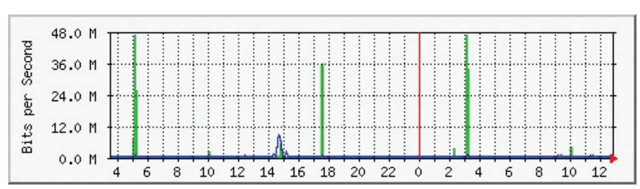
	Maximal	Average	Instantaneous
Outgoing	69.8 Mb/s (69.8%)	2461.8 Kb/s (2.5%)	77.2 Kb/s (0.1%)
Incoming	1577.0 Kb/s (1.6%)	95.2 Kb/s (0.1%)	112.2 Kb/s (0.1%)

Fig. 17 Statistics of network utilization of the business -financial applications server
Slika 17. Statistika mrežne uporabe poslužitelja poslovno financijskih aplikacija



	Maximal	Average	Instantaneous
Outgoing	59.5 Mb/s (6.0%)	351.0 Kb/s (0.0%)	43.0 Kb/s (0.0%)
Incoming	1115.9 Kb/s (0.1%)	43.9 Kb/s (0.0%)	51.4 Kb/s (0.0%)

Fig. 18 Statistics of network utilization of the authentication server for Windows users
Slika 18. Statistika mrežne uporabe autentifikacijskog poslužitelja za korisnike Windowsa



	Maximal	Average	Instantaneous
Outgoing	47.0 Mb/s (4.7%)	573.8 Kb/s (0.1%)	25.2 Kb/s (0.0%)
Incoming	8182.4 Kb/s (0.8%)	129.9 Kb/s (0.0%)	26.6 Kb/s (0.0%)

Fig. 19 Statistics of network utilization of the Exchange server
Slika 19. Statistika mrežne uporabe Exchange poslužitelja

values during other intervals.

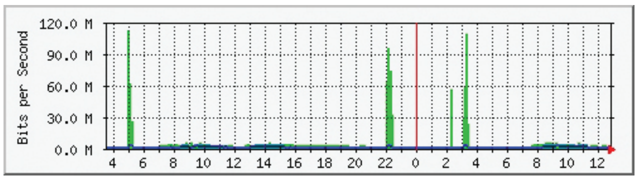
When network links towards link-layer switches are in question, the average utilization during work time was found to be 0.046 %, while the average utilization of incoming links was 0.066 %. Again, maximum utilization was found within 2 to 6 pm period due to servers backup, while showing low values out of this time interval.

Sporadic congestions were identified to occur during rare short periods when particular computers generated excessive number of broadcasts, thus causing peak traffic loads. However, these problems were not the consequence of poor network design or implementation, but exclusively due to inadequate antivirus protection on particular computers.

Finally, based on the above test results showing that the existing network capacities can satisfactorily bear the existing traffic load with considerable redundancy, it is realistic to expect that, from the point of view of throughput alone, adding new services, such as e.g. VoIP might not lead to significant degradation of network QoS.

ETHERNET TRAFFIC ANALYSIS AFTER INTRODUCING VOIP SERVICE / *Analiza Ethernet prometa nakon uvođenja VoIP usluge*

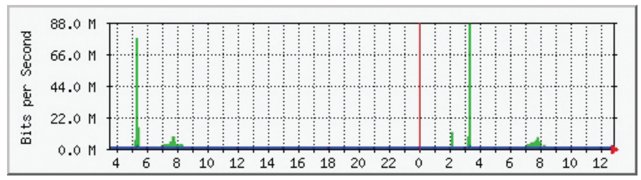
In spite of the ad hoc considered realistic expectation that



	Maximal	Average	Instantaneous
Outgoing	111.7 Mb/s (11.2%)	4041.4 Kb/s (0.4%)	3495.0 Kb/s (0.3%)
Incoming	3298.2 Kb/s (0.3%)	681.8 Kb/s (0.1%)	1284.9 Kb/s (0.1%)

Fig. 20 Statistics of network utilization of the web proxy / firewall

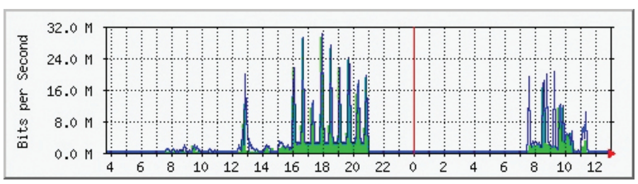
Slika 20. Statistika mrežne uporabe proxy web/ vatrozid poslužitelja



	Maximal	Average	Instantaneous
Outgoing	86.5 Mb/s (8.6%)	929.2 Kb/s (0.1%)	1288.0 Kb/s (0.0%)
Incoming	1624.9 Kb/s (0.2%)	50.4 Kb/s (0.0%)	10.7 Kb/s (0.0%)

Fig. 21 Statistics of network utilization of the antivirus software server

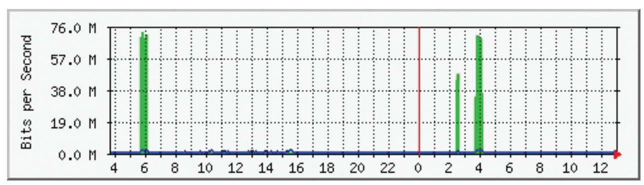
Slika 21. Statistika mrežne uporabe poslužitelja protuvirusnog programa



	Maximal	Average	Instantaneous
Outgoing	29.4 Mb/s (2.9%)	1512.7 Kb/s (0.2%)	31.0 Kb/s (0.0%)
Incoming	29.8 Mb/s (3.0%)	1893.9 Kb/s (0.2%)	112.4 Kb/s (0.0%)

Fig. 22 Statistics of network utilization of the second layer switch

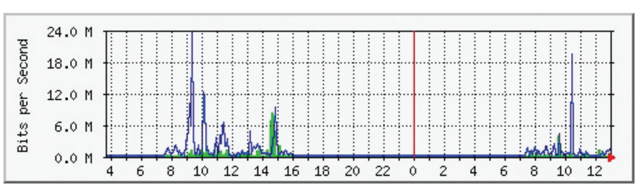
Slika 22. Statistika mrežne uporabe preklopnika drugog sloja



	Maximal	Average	Instantaneous
Outgoing	72.5 Mb/s (7.3%)	2030.4 Kb/s (0.2%)	214.8 Kb/s (0.0%)
Incoming	2033.3 Kb/s (0.2%)	231.1 Kb/s (0.0%)	343.1 Kb/s (0.0%)

Fig. 23 Statistics of network utilization of the second layer switch

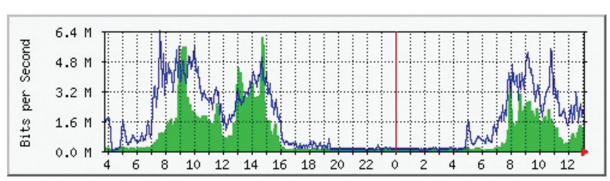
Slika 23. Statistika mrežne uporabe preklopnika drugog sloja



	Maximal	Average	Instantaneous
Outgoing	8310.4 Kb/s (0.8%)	203.2 Kb/s (0.0%)	73.1 Kb/s (0.0%)
Incoming	23.3 Mb/s (2.3%)	644.3 Kb/s (0.1%)	1153.9 Kb/s (0.1%)

Fig. 24 Statistics of network utilization of the second layer switch

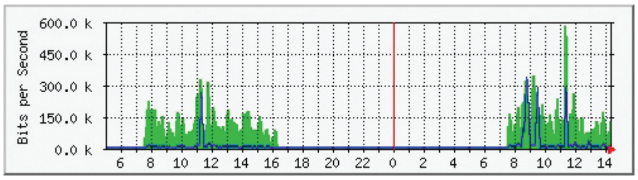
Slika 24. Statistika mrežne uporabe preklopnika drugog sloja



	Maximal	Average	Instantaneous
Outgoing	6076.6 Kb/s (0.6%)	864.6 Kb/s (0.1%)	1156.4 Kb/s (0.1%)
Incoming	6388.0 Kb/s (0.6%)	1594.6 Kb/s (0.2%)	1981.7 Kb/s (0.2%)

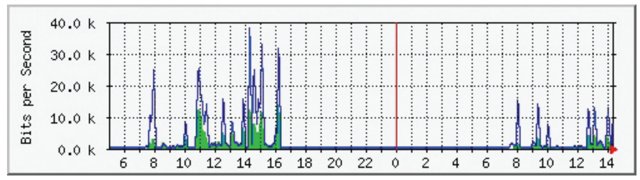
Fig. 25 Statistics of network utilization of the second layer switch

Slika 25. Statistika mrežne uporabe preklopnika drugog sloja



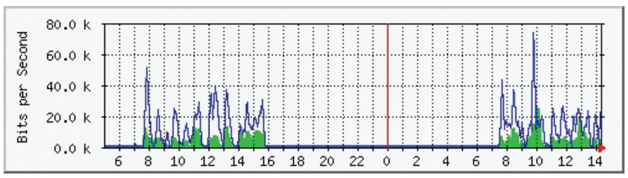
	Maximal	Average	Instantaneous
Outgoing	580.8 Kb/s (0.6%)	61.3 Kb/s (0.1%)	128.3 Kb/s (0.1%)
Incoming	332.4 Kb/s (0.3%)	10.7 Kb/s (0.0%)	8056.0 b/s (0.0%)

Fig. 26 Statistics of network utilization of the switch port after implementing VoIP
Slika 26. Statistika mrežne uporabe preklopnika nakon primjene VoIP-a



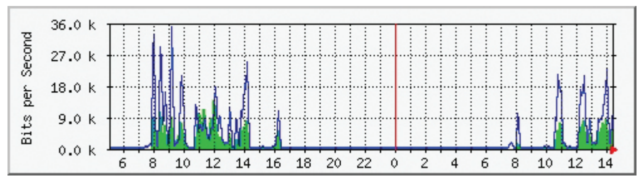
	Maximal	Average	Instantaneous
Outgoing	13.9 Kb/s (0.0%)	848.0 b/s (0.0%)	4360.0 b/s (0.0%)
Incoming	37.7 Kb/s (0.0%)	2208.0 b/s (0.0%)	14.1 Kb/s (0.0%)

Fig. 27 Statistics of network utilization of the switch port after implementing VoIP
Slika 27. Statistika mrežne uporabe preklopnika nakon primjene VoIP-a



	Maximal	Average	Instantaneous
Outgoing	27.0 Kb/s (0.0%)	2376.0 b/s (0.0%)	5032.0 b/s (0.0%)
Incoming	73.4 Kb/s (0.0%)	6312.0 b/s (0.0%)	20.3 Kb/s (0.0%)

Fig. 28 Statistics of network utilization of the switch port after implementing VoIP
Slika 28. Statistika mrežne uporabe preklopnika nakon primjene VoIP-a



	Maximal	Average	Instantaneous
Outgoing	13.7 Kb/s (0.0%)	1296.0 b/s (0.0%)	4896.0 b/s (0.0%)
Incoming	34.6 Kb/s (0.0%)	2696.0 b/s (0.0%)	7776.0 b/s (0.0%)

Fig. 29 Statistics of network utilization of the switch port after implementing VoIP
Slika 29. Statistika mrežne uporabe preklopnika nakon primjene VoIP-a

links utilization prior to VoIP deployment most likely will not become a bottleneck for network QoS, we checked how realistic was the expectation by continuing measurements of network links utilization after introducing VoIP service. The results of tests analogous to the previously presented ones for the switches' network links are as it follows on Figs. 26 – 29.

From the above diagrams, we can calculate that the average daily utilization of incoming links was very low – just 0.2 %, while the analogous value for the outgoing links was exceptionally low – close to 0.0 %.

In addition, the maximum daily utilizations for incoming and outgoing links were found to be equal to 1.1 % and 0.8 %, respectively, meaning that traffic volumes in both directions are very close. Peak network links utilization was registered within the period from 7.30 am to 4 pm.

Overall, it was found that VoIP traffic did not impose any significant increase in overall utilization and that reliability, response time and accessibility of the business system network applications were not degraded by introducing VoIP service.

CONCLUSION / *Zaključak*

Deploying evolving Ethernet standards enable telecom network service providers to develop convergent networks that transport new generation services, such as triple-play. With this regard, important considertions deal with practical suitability of Ethernet transport for concurrent triple play services, specifically data and VoIP. The tests were made at both Ethernet and application layers. The results reflect the fact that Ethernet bandwidth and scalability preserve enough redundancy to accomodate inclusion of VoIP into network services list.

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