Quality of Service Support in IPv6-based Military Networks with Limited Bandwidth Links[1]

Marek Amanowicz
Military Communication Institute
05-130 Zegrze
POLAND

m.amanowicz@wil.waw.pl

Peter Sevenich
Institute for Communications
Information Processing and Ergonomics (FKIE)
53343 Wachtberg
GERMANY

sevenich@fgan.de

Jacek Jarmakiewicz
Military University of Technology
Kaliskiego 2
00-908 Warsaw
POLAND

jjarmakiewicz@wel.wat.edu.pl

Markus Pilz
Computer Science
University of Bonn
Bonn
GERMANY

pilz@cs.uni-bonn.de

ABSTRACT

Adaptation of commercial Internet solutions for providing effective mechanisms supporting user's mobility, information security and quality of services is the subject of several research projects running both nationally and internationally. From the military perspective, the IPv6-based solutions must provide required level of performance for networks deployed in specific tactical environment. Such networks - by their nature - are relatively unreliable and varying, both in location and topology. In tactical networks, significant amount of data, both time-critical and non-real time, has to be exchanged over disadvantage (also in terms of bandwidth limitation) links. This calls for improving the standard IPv6 mechanisms and elaboration of new effective methods of QoS support. The NATO Information System Technology Panel ongoing Support Project (PL-IST-003) on "Implementation of IPv6 Protocol for Tactical Interoperable Communications Networks – TICNET" enters into these efforts. The paper gives an overview of the works on improving QoS support in IPv6-based military networks that are performed jointly by the teams from German and Polish research establishments. In particular, it presents the basic issues of providing QoS for both time-critical and non-real time services in tactical networks with limited bandwidth links and discusses the measurement methodology of prediction of available bandwidth. It also describes the results of technical experiments and simulations, aiming at evaluation the efficiency of QoS support mechanisms elaborated within the Project.

1.0   BACKGROUND

Military transformation process of the war fighting capabilities adjustment to new emerging strategic challenges leads to fundamental changes in military doctrine and operational concepts, military forces organisation, war fighters' equipment and training. Nowadays, the strategic thinking is mostly stimulated by the new challenging concept of operation, called Network Enabled Capabilities (NEC). It generates increased combat power by networking sensors, decision makers and shooters to achieve shared awareness, increased speed of command, higher tempo of operations, greater lethality, increased survivability, and a degree of self-synchronisation. This concept requires essential changes in military thinking as well as strong technological support.

Interoperable communications and information infrastructure is a core of net-centric operations. It enables creation of shared battlespace awareness and knowledge, which is leveraged by new adaptive command and control processes and self-synchronizing forces. The dynamics of military information flow demands the reliability and robustness of the information grid. However, military communication infrastructure is relatively unreliable and timely varying both in location and topology. This call for development of effective mechanisms and related procedures that allow deploying communication networks, which operate in a fashion that scales to the net-centric operations.

A significant part of these activities refers to adaptation of commercial Internet solutions for developing tactical communications systems. It is expected, that implementation of IPv6 protocol will significantly contribute to achieving this goal, providing the mechanisms supporting user mobility, information security and quality of services. However, IPv6 protocol requires detailed investigation of its efficiency and need to be seriously verified against performance in environments with unique link characteristics of tactical networks. That is a subject of several research investigations performed both nationally and internationally. The NATO Information System Technology Panel ongoing Support Project on "Implementation of IPv6 Protocol for Tactical Interoperable Communications Networks – TICNET" enters into these efforts. The works performed by research teams from Research Institute for Communications, Information Processing and Ergonomics (FKIE) from Germany, Military Communication Institute (MCI) and Military University of Technology (MUT) from Poland, are focused on identification the gaps and providing improvements to standard IPv6 mechanism, especially resulted in:

- elaboration effective procedures of QoS support providing both secure time-critical and non-real-time data over IPv6-based mobile networks with limited bandwidth links,

- creation the technical infrastructure and simulation environment for QoS mechanisms verification and validation, both at protocol and system level.

Additionally, the Project gives an opportunity for fostering collaboration between the Polish and German research establishments, improving their scientific base and effective usage of the research potential for national and NATO defence needs.

In this paper, we discuss the basic issues of providing QoS support in tactical networks with narrowband links (for instance ISDN or HF links) and shortly resume the mechanisms implemented in the Network Adapter (NA), such as header and data compression, traffic multiplexing and prioritisation. The methodology of prediction the end-to-end bandwidth that is available for handling of non-time critical traffic has been used for evaluation of the NA performance in the presence of both time critical and non-prioritised data. Some measurement data are discussed in section 3.

Then, we describe the technical experiments performed with the aim of evaluation the QoS metrics while multiplexing time-critical and non-real time data in the narrowband link. The experiments have been performed in testbedding environment with two Network Adapters located in FKIE (Germany) and MCI (Poland) and connected over the ISDN link. A general description of the testbed configuration, testing scenarios and obtained results is given in section 4.1.

In section 4.2 we describe the OPNET simulation model that have been used for evaluation the efficiency of the proposed mechanism of QoS support in complex environment. In the paper, we shortly describe the model architecture as well as scenarios used for the model validation. We also present the results of our simulations and the comparison with data obtained during technical experiments.

In conclusion, we present our comments on applicability of described solutions in tactical networks and specify the way ahead.

2.0   QoS SUPPORT IN TACTICAL ENVIRONMENT

2.1   Bandwidth limitation

The military users needs for providing the increasing amount of different types of information (including multimedia) with acceptable quality level from sensors to the engage units result in demands for deploying the networks with higher and higher bandwidth. Today, advances in information, communication and optoelectronic technologies allow achieving a balance between provision and consumption of the bandwidth. Additionally, many researches carried out under NATO and EU auspices are focused on developing effective methods of QoS support in Internet and other types of IP-based networks.

However, a direct implementation of these solutions in military networks is limited due to security, mobility or interoperability issues, but also because of the nature of tactical communication systems that widely use low bandwidth links (e.g. ISDN, HF or SATCOM). The bandwidth limitations cause that specific method of QoS support for simultaneous transmission of time-critical and conventional data has to be considered. It should be noted that quality of voice communications over the limited bandwidth link depends on coding and modulation methods, performance of the communication link (delay, jitter, loss ratio), traffic structure loading the link but also is sensitive to the network protocols (routing, security, etc.) that would require exchange additional information over the link.

Three basic areas of ensuring the QoS for voice communication in IPv6-based military networks with limited bandwidth links have been considered:

- use of robust application protocol that provides mechanisms for reliable communication establishing and disconnection as well as priority handling mechanism for voice streams;

- efficient use of limited bandwidth that includes application of spectrum-effective vocoders as well as compression of protocol information;

- management of the limited bandwidth link that results in reduction of exchanged protocol information and implementation of effective priority handling and multiplexing technique allowing simultaneous transmission of voice and data traffic.

The detailed information on the mechanisms that support QoS in military network with bandwidth limitation is presented for instance in [4]. Generic description of QoS mechanism implementation is given in the next section of the paper.

2.2   Network Adapter

The Network Adapter has been developed for providing the QoS mechanisms discussed above that allows managing a limited bandwidth link within the Ethernet network in an efficient way. For each IPv6-based user applications, the NA recognises both nature of the data (time-critical - RT or conventional data – NRT) and the transmission priority compared to those of concurrent data flows that use the same limited bandwidth resources. NA analysis all QoS sensitive information, like for instance: source-sink destination address, nature of data, priority, required bandwidth, the use of IPSec, etc. This continuously updated

information is processing in the scheduling, queuing and fragmentation algorithms. With simultaneous consideration of the data nature and priority, the IPv6 packets are multiplexed in the limited bandwidth link. Conventional data are picked up from the queue related to a specific priority level, then fragmented (in most of the cases) and filled into the gaps in between unfragmented packets belonging to the time-critical flows.

The NA architecture is asymmetric due to a significant bandwidth difference in link layer interfaces. Two types of interfaces are required – one fast (Ethernet) interface and at least one slow (e.g. V.24 or ISDN) for connection the NA to narrowband link.

The NA fulfils an additional task, i.e. it analysis the passing traffic like IPv6, ICMP or routing packets, stores these system requests and the corresponding answers from the network devices for a configurable time interval. During this period the NA answers the systems requests for which it has been stored the replies, without transmitting the request packets over the limited bandwidth link. This avoids bit errors in sensitive system traffic handled by this link.

For achieving the efficiency of the process of NA development and testing, the NA and related mechanisms have been implemented as a separate device on conventional PCs under Linux operating system. It was also assumed that two independent NA implementations would be produced and tested by FKIE and MCI/MUT for confirmation the efficiency of the proposed QoS mechanisms for military applications.

3.0   BANDWIDTH PREDICTION

3.1   Measurement Methodology

In this section, the focus is on available bandwidth estimation between two clients (end-to-end), i.e. across two Network Adapters. Although this estimation could be supported by the NA, e.g. with aid of additional signalling between the entities, this approach is based on measurements that are initiated and evaluated at the clients, i.e. the network edges. The motivation for this approach arises from the multitude of scenarios that are envisioned. The capacity estimation may support clients in situations when no NA is applied in the network, it should support the clients, when two NAs are on the network path between the clients and the solution should also work in larger scenarios, where multiple NAs (nested or in succession) are on a path between two clients. The approach is therefore uncoupled from the complexity of the network and omits protocol extensions to propagate state information between entities.

The conducted capacity estimations were based on packet pairs (ref. [2], [3]). Path capacity estimation techniques have been important subjects for research for a considerable amount of time. Prominent techniques are based on the dispersions a packet pair or packet train – sometimes called packet bunch – experiences on the way from the sender to the receiver. For a more detailed description about packet pairs and properties of packet pairs in the context of the NA, the reader is referred to [3].
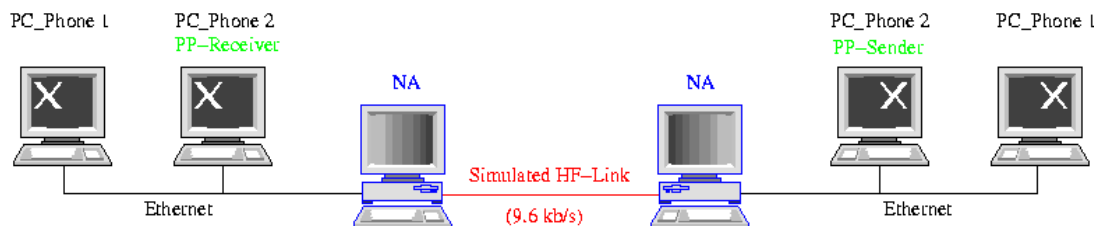

Figure 1: Testbed setup

Figure 1 shows the different system parts and applications used for measurements. The testbed comprised 4 PCs for sending and receiving VoIP data streams (PC_Phones), which are also used for the packet dispersion measurements. Additionally, two PCs (NAs) are equipped with the Network Adapter application (centre of Figure 1). The NAs are connected to each other via an HF-link simulator. The HF simulator provides serial line interfaces. The other boxes are connected via Ethernet.

During the measurements described in this paper only the 1.2 kbit/s MELP vocoder was used. In this case a voice packet, transmitted between the NAs, consists of 41 bytes including voice payload (three vocoder frames) and the compressed IPv6/UDP Header. In this case, every 20 ms a packet is sent from the application. That leads to a transmission rate of 1640 bit/s for one voice-connection.

Furthermore, it has to be taken into account that the start- and stop bit related to the asynchronous transmission mode of the serial interfaces connecting the network adapter have to be transmitted, therefore limiting the path throughput for IP data of a 9.6 kbit/s link to 7.68 kbit/s.

3.2   Measurement Data and Interpretation

The measurements comprise of samples of two packet pairs for every IP datagram size between 60 bytes and 1400 bytes. Each packet pair was sent from one client to another via the bottleneck link managed by two Network Adapters (NAs). Three different test series are covered in the following: a single packet pair stream, one voice and packet pair stream, and two voice streams and a packet pair stream. Each voice transmission had real-time characteristics and was prioritised. According to the behaviour of the Network Adapters, the packet pair packets are fragmented and, in presence of voice streams, small voice packets should interleave with the packet pair measurement packets. As a result, the signature of voice data should be visible in the packet pair measurements. As mentioned before, the 9.6 kbit/s serial link provides an effective link capacity of 7.68 kbit/s. In addition to the results presented in [3], this section concentrates on samples of a particular packet sizes, i.e. the figures are based on the packet sizes 60, 80, 100, 150, 200, 400, 600, 800, 1000, 1200, and 1400 bytes.

The first scenario depicted in Figure 2 represents the scenario with a single packet pair stream, i.e. no additional traffic is transported via the NA. On the left hand side the expected linear dependency between packet size and packet separation can be seen. Minor deviations are visible. Linear regression with all samples estimates the gradient of the resulting straight line to 0.0010978s/byte ± 0.0000003s/byte. From the gradient, the underlying link speed can be derived as it represents the time needed to transmit additional 1 byte of data. This would result in a link speed of approximately 7287 bit/s ± 2 bit/s. This estimated link speed is about 400 bit/s below nominal link throughput.
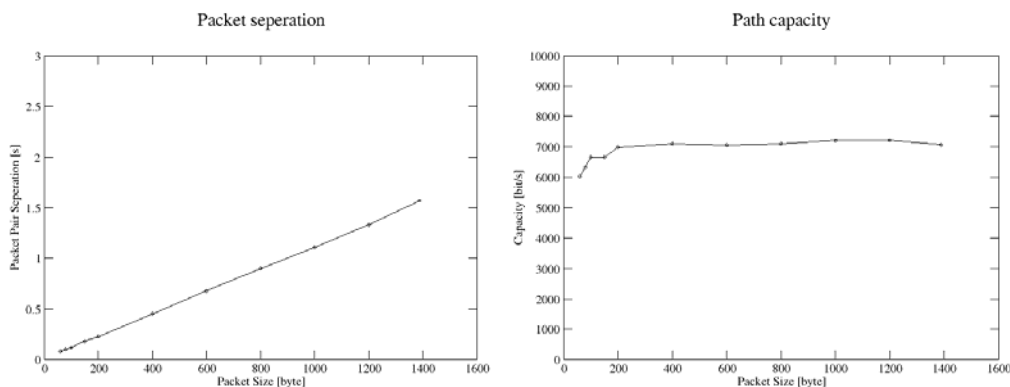

Figure 2: Packet pair separation and available bandwidth without voice data

The transformation to the bandwidth estimates is shown on the right hand side of Figure 2. Again, the measured bandwidth is below the maximum of 7680 kbit/s and minor deviations are visible. Both effects can be explanted by the behaviour of the NA, as an NA fragments arriving packet pair packets within certain time boundaries and needs additional protocol overhead to transmit the fragmented packets.

The next scenario contains a packet pair probing stream and a single 1.2 kbit/s voice stream. The linear dependency between packet size and packet pair separation is shown on the left hand side of Figure 3. Here, the deviation is slightly higher compared to the prior scenario. Linear regression estimates a gradient of 0.0013356 s/byte ±0.0000027s/byte which is about 5990 bit/s ± 12 bit/s. This value is between the theoretical maximum (7680 bit/s – 1640 bit/s = 6040 bit/s) and the estimation without voice traffic (7287 bit/s – 1640 bit/s = 5647 bit/s). This result suggests a slightly better performance of the NA in presence of prioritised voice data.
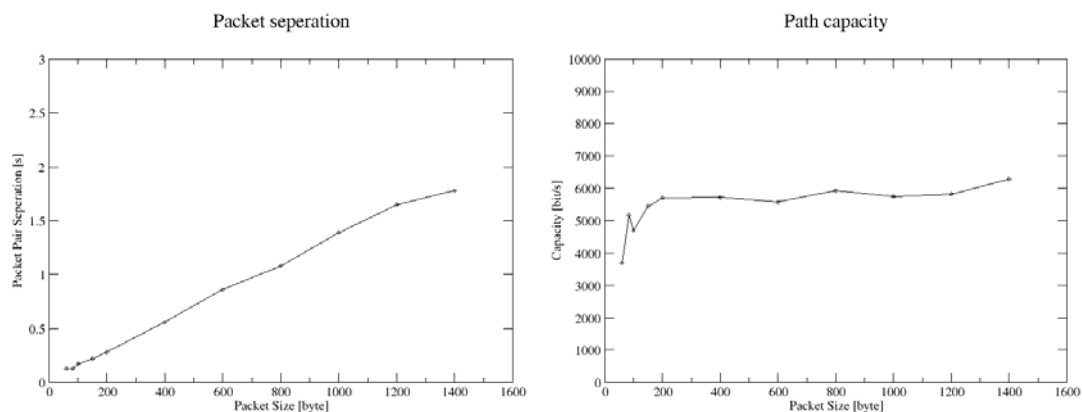

Figure 3: Packet pair separation and available bandwidth with one voice stream

In the third scenario, two voice streams and a packet pair probing stream are transmitted via the narrow HF link. Again, a linear dependency with minor deviants is present and can be seen on the left hand side of Figure 4. The overall gradient of the packet pair separation is 0.0017264 s/byte ± 0.0000027 s/byte corresponding to 4634 bit/s ± 8 bit/s. The corresponding theoretic value is 4400 bit/s (7680bit/s- 2*1640bit/s). In this case, the bandwidth estimation via packet pairs is slightly overestimating the path throughput.
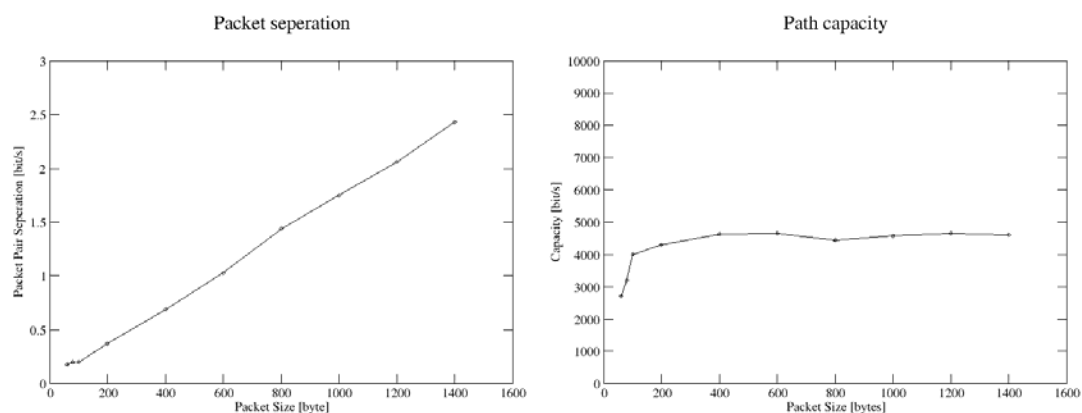

Figure 4: Packet pair separation and available bandwidth with two voice streams

The performed measurements exemplarily show the behaviour of the NA in the presence of voice data and non-prioritised data, i.e. packet pairs. The assumed fragmentation of the packet pair and prioritisation of voice data packets appear to be traceable by taking a close look at the packet separation and the resulting bandwidth. Furthermore, the estimated bandwidth is within a reasonable distance to the capacity available to non-prioritised traffic.

4.0   QoS MECHANISMS EVALUATION

In this section, we describe technical and simulation experiments performed with the aim of evaluation the efficiency of the proposed mechanisms of QoS support in IPv6-based network with limited bandwidth link. The FKIE implementation version of the Network Adapter has been used for execution the technical experiments. The experiments have been performed in testbedding environment with two Network Adapters located at FKIE (Germany) and MCI (Poland) that have been connected over the ISDN link. Simulation model gave an opportunity to evaluate the QoS metrics in a complex network environment, however it required prior validation. For this reason, the testing scenarios have been defined in a way that allows making comparisons between empirical and simulation results.

The technical experiments have been performed in the network configuration presented in Figure 5. The testbed is composed of two VoIP terminals, which are located on both sides of the network (in Wachtberg, Germany and in Zegrze, Poland) and one PC terminal (at MCI side) for generation of conventional data. PC Phone application has been used for time-critical data generation, while ping packets of predefined size have represented non-real time data. At the FKIE side of the network, a received packet has been retransmitted to the MCI terminal. We decided to do not use typical ftp application for the measurements, as they would be distorted by TCP mechanism. Additionally, two PCs equipped with the Network Adapter application (at both locations) have been connected to each other by ISDN link. A dedicated NA application allowed reducing the ISDN link bandwidth to 9.6 kbit/s. The other components (including IPSec Gateways) have been connected through Ethernet.
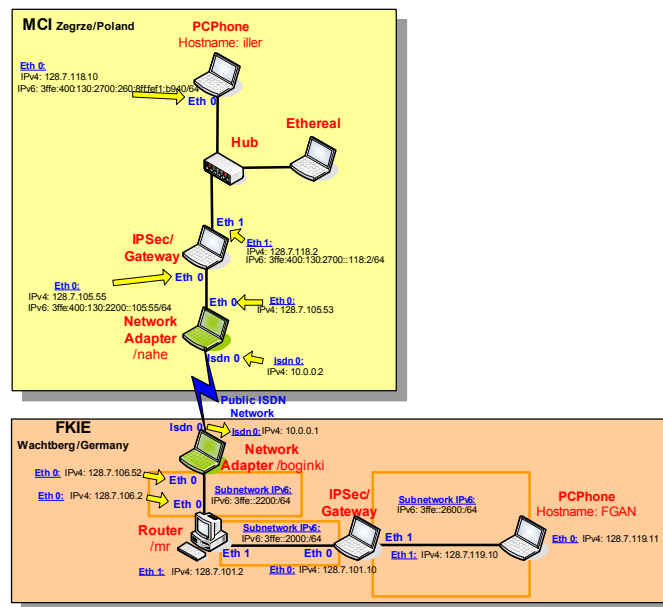

Figure 5: Testbedding configuration

The application performance statistics have been collected during technical experiments, including voice packets delay, voice packet jitter and non-real time (ping) packet delay.

Some empirical results are presented in Figure 6. Figure 6a) shows the measured values of voice packet jitter without presence of data traffic. The measured values of voice packet jitter if voice traffic was mixed with non-real time data are presented in Figures 6b), 6c) and 6d). The data packet size has been varied from 170B to 1512B, respectively. The measured values of end-to-end data packet delay are shown in Figure 6e). The theoretical values of packet delay if the network is loaded only by conventional data traffic (ping packets) are also drawn in this figure.
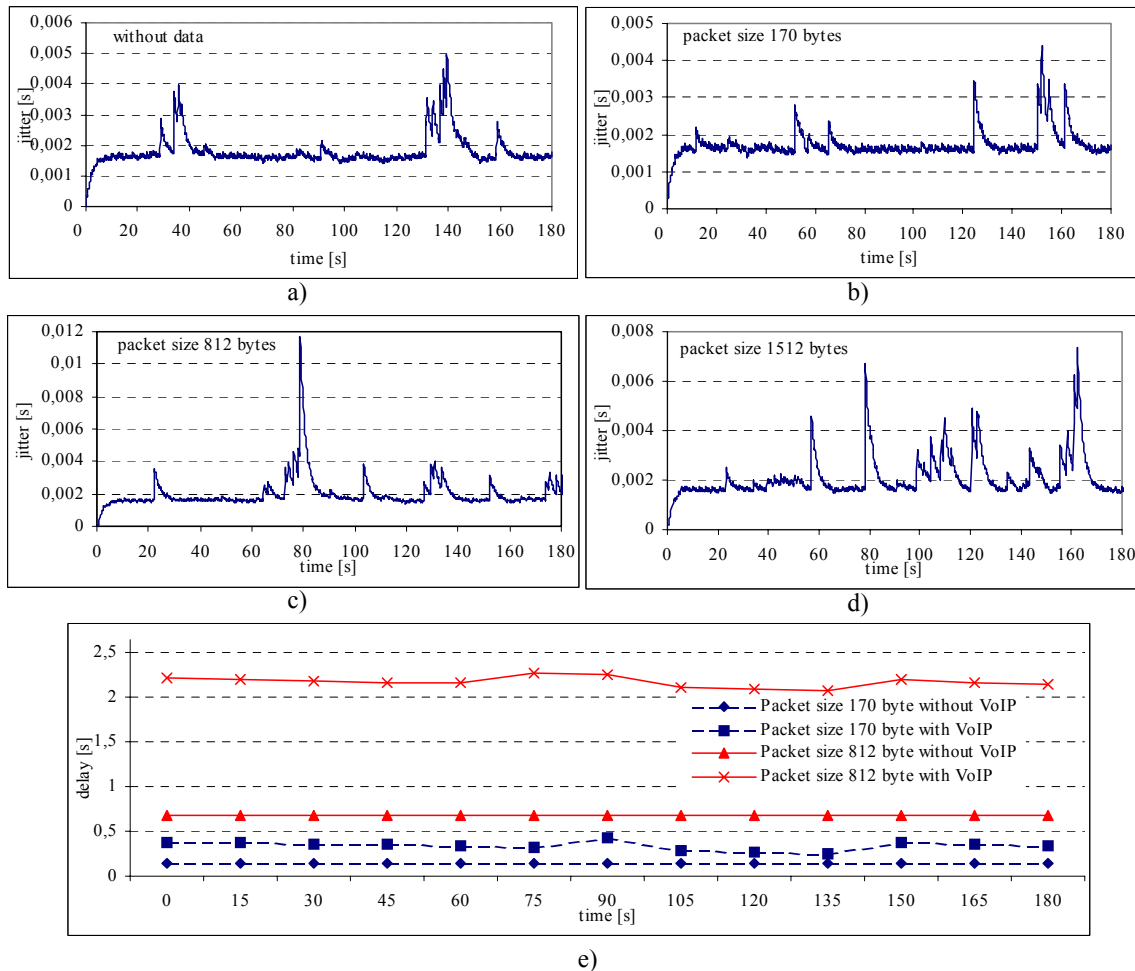

Figure 6: Application performance empirical results

As we expected, the conventional data packet flows do not influence the performance of voice application. We did not observe any essential difference in the voice performance if voice and data traffic is mixed (Figures 6a-d). Local changes of the voice packet jitter are caused by instability of PC Phone application. We can also observe the influence of mixed voice and data traffic on data application performance (Figure 6e). In this case, voice packets reduce the link bandwidth available for conventional data transmission, which leads to increasing the data packet delay. This effect is additionally intensified by data packet segmentation, which occurs if the packet size increases the predicted gap between two adjacent voice packets.

The Network Adapter simulation model have been produced in order to evaluate the mechanism of QoS support in IPv6-based networks as well as to examine the NA performance in a complex network environment. Additionally, the simulation model has been used for verification the correctness of source code components before their implementation in the NA.

The OPNET simulation software (version 8.0) with its standard library has been selected for developing the NA model. Due to this version limitation, IPv6 protocol has been emulated in the NA model (IPv4 to IPv6 conversion). It was considered that standard ARP and other IPv4 signalling packets are treated as ICMPv6 packets. The description of the Network Adapter OPNET model is also given in [7].

Two basic network configurations that have been used for examination of the NA performance are presented in Figure 7. In simple network configuration (Figure 7a), two terminals (Ethernet workstation) are connected to the limited bandwidth link through the Network Adapter. Extended configuration (Figure 7b) additionally includes 3 routers, IP subnet and Ethernet server.
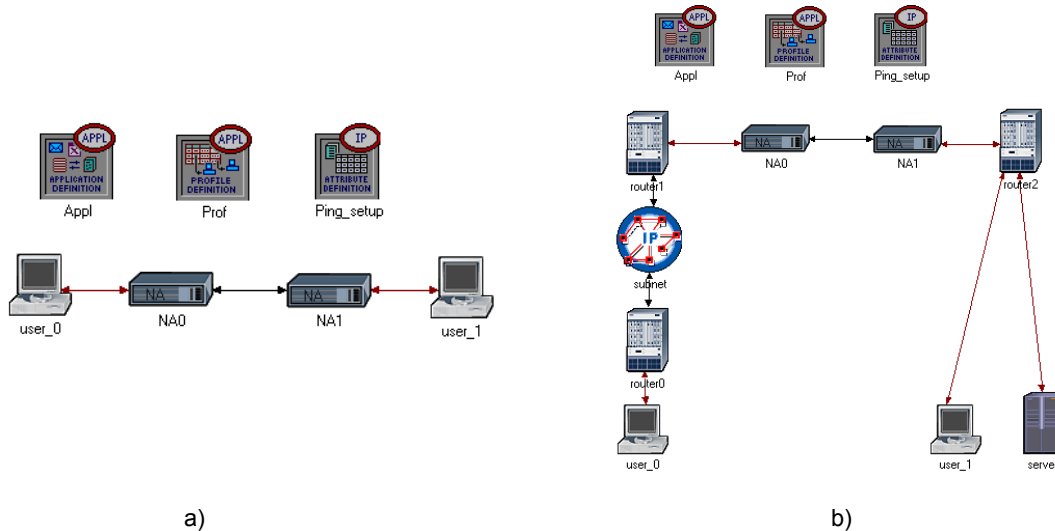

a)   b)

Figure 7: Network configurations

In both configurations, Ethernet workstation generated time-critical (VoIP) and ping applications. In extended network configuration, additional ftp traffic has been downloaded from the server.

The Network Adapter performance has been tested under the following traffic conditions: ping, VoIP, VoIP mixed with ping, two VoIP streams mixed with ping, one VoIP stream mixed with ping and VoIP mixed with ftp traffic (only in extended network scenario).

Ping application of different packet size has been selected in order to avoid the influence of TCP mechanism on the measured characteristics.

The following OPNET statistics have been collected and used for evaluation of the NA QoS mechanisms:

- at terminals: voice packet end-to-end delay, voice packet jitter, traffic sent, traffic received, ping response time, ftp response time

- at Network Adapter: RT datagram delay, RT forwarding memory queuing delay, predicted gap's length,

- at limited bandwidth link: link utilisation, link throughput.

Specification of the OPNET model parameters for considered scenarios and detailed discussion of the simulation results is given in [7] and [8]. In this paper, we present only selected results as well as summarise the basic conclusions.

a)   b)
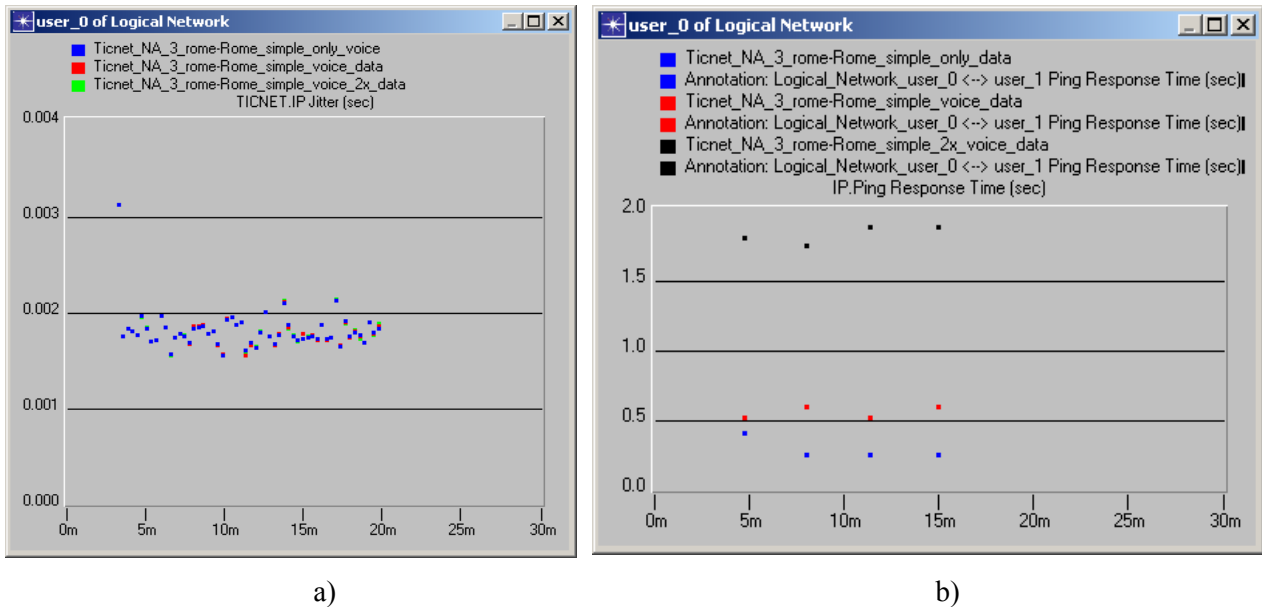
Figure 8: Application performance in simple network configuration

The simulation results for simple network configuration are presented in Figure 8. The voice packet jitter with single VoIP application (2.4 kb/s MELP vocoder) as well as one voice mixed with one or two ping applications (packet size of 64B) are shown in Figure 8a. Figure 8b shows ping application response time with single ping (packet size of 64B) and one ping (blue points) mixed with one (red points) or two (black points) voice applications (2.4 kb/s MELP vocoder). It is observed that increasing the number of data streams does not cause degradation of the VoIP application performance. Figure 8b) confirms significant influence of VoIP data stream on conventional data packet delay.
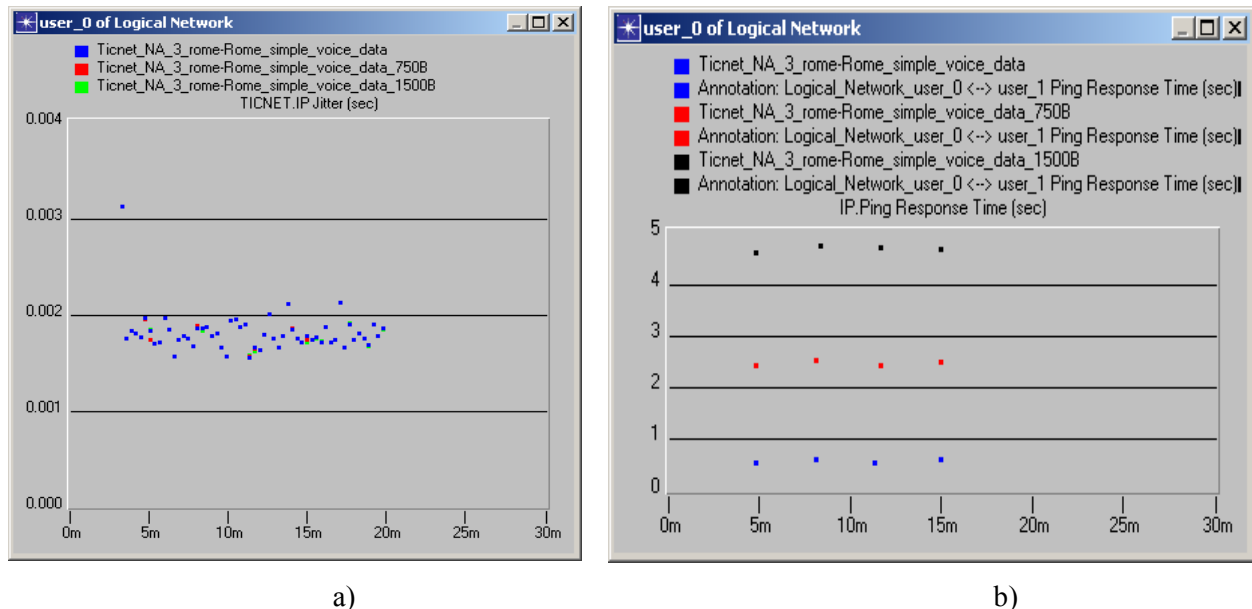

a)   b)

Figure 9: Application performance versus packet length in simple network configuration

Figure 9 shows the influence of non-real time packet size (ping) on the voice (a) and ping (b) application performance. As it was expected, an increase of the packet size results only in degradation of ping

application performance (Figure 9b). It also confirms that QoS mechanisms implemented in the NA allow maintaining the voice application performance in acceptable limits (~ 2 ms) for all considered scenarios (Figure 9a). Detailed presentation of simulation scenarios and discussion of the results is given in [7].

5.0   CONCLUSION

In the paper, the basic mechanisms of QoS support in IPv6-based military networks with limited bandwidth links have been discussed. We presented our approach to QoS support based on application of header compression, narrowband link bandwidth prediction and the link resources management, like traffic control, priority handling and traffic multiplexing. These mechanisms have been successfully implemented in the Network Adapter connected to the narrowband link edge devices (modem, HF transceiver, GSM terminal, ISDN switch, etc.). The empirical data as well as the results of our simulations obtained from several testing scenarios confirmed the efficiency of the proposed solution. For the time being, only FKIE version of the NA testing has been performed. Shortly, the tests will be re-run with two independent implementation of the Network Adapter software (FKIE and MCI/MUT) in order to proof the correctness of implementation and giving the opportunity for widely use of NA solution in different IPv6-based military networks. We are going to continue the efforts on investigation the methods of resource management that might be successfully implemented in IPv6-based military networks with limited bandwidth links.

6.0   REFERENCES

[1]   Maseng, Nissen: *Network-centric Military Communications*, IEEE Communications Magazine, Novemebr 2004

[2]   Prasad, Dovrolis, Murray, Claffy: *Bandwidth Estimation: Metrics, Measurement Techniques, and Tools*, IEEE Network, November/December 2003

[3]   Lies et al.: *The Effective QoS Mechanism for Real Time Services in IP Military Networks*, Proceedings of the RCMCIS Conference, Zegrze, October 2004

[4]   Beling, Sevenich: *Multiplexing Time-critical Data Over Heterogeneous Subnetworks of Low Bandwidth*, Proceedings of the RCMCIS Conference, Zegrze, October 1999

[5]   Göbel: *Beschreibung eines QoS-unterstützten Netzadapters (NA) für schmalbandige Subnetztypen*, FKIE-Bericht Nr.38, Januar 2002

[6]   Sevenich, Riechmann, Kiesow: *Multiplexing Time-Critical and conventional Data over Tactical IPv6 Networks of Low Bandwidth*, INSC Symposium, Den Haag, November 2003

[7]   Krygier, Lubkowski: *Simulation* IST-054/RSY-015 Symposium on Military Communications, Rome, April 2005

[8]   Krygier: "QoS support procedures implementation in the NA simulation model", MCI Report No. 170, March 2005.