

Simulation Study of QoS in IPv6-Based Military Networks with Limited Bandwidth Links

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ABSTRACT

Implementation of the IPv6 protocols stack has become a fact in military networks. Although the IPv6 protocol is very efficient but all problems with its adaptation to military networks have not been resolved yet. The assurance of necessary level of quality of services in changing condition of available bandwidth is an example of such a problem. Additionally, current military networks also use narrowband links, especially in their radio segments. The problem is serious, because the IP protocols suite (and thus IPv6) has been elaborated and developed for wideband networks and for carrying the non real-time traffic. Thus for instance, the voice or video can be provided using IPv6, but a minimum bandwidth is required or advanced QoS supporting mechanism have to be implemented. The researches carried out by the FKIE, the Computers Sciences Department of the University of Bonn and Polish MCI/MUT teams are related to developing of IPv6 QoS mechanisms that are well suited to the low bandwidth requirements in military networks. These mechanisms have been implemented both in real-life software and in simulation environment. The first experiments show their proper behaviour in IPv6-based military networks with limited bandwidth links. The paper covers a description of modelling and simulation of proposed QoS mechanism. The authors have concentrated on an analysis of the service multiplexing and priority handling procedures. The aim of this effort is to evaluate bandwidth utilisation that allows increasing the information flows. Achieved results give a possibility of using simulation model for testing of new developed or modified IPv6 mechanisms for wider tactical networks.

1.0 INTRODUCTION

Modern military networks often use the commercial of the shelf (COTS) solutions in the wide range of applications. It gives an opportunity to make use a wide class of new type of services, such as real-time (RT) services in a military environment. In our meaning, the real-time services represent the family of services in which the voice is transferred in packed mode with using the IPv6 protocol stack. Although the implementation of IPv6 protocol in the military area it is not essential problem, however the assurance of necessary level of QoS is still an open issue.

The well-know mechanisms that support a realisation of real-time services in the IP environment are IntServ and DiffServ. They have been standardised by IETF and are commonly used in the commercial IPv6 solutions. Unfortunately, they are not well suited for military networks. An essential restriction that has been noticed comes from the quality and bandwidth of transmission channels used (especially radio channels have been taken into consideration) as well from the changeability of channel characteristics. The researches conducted under NATO projects concentrates on improvements of IPv6 protocol mechanisms related to QoS for adapting them to the military networks characterised by limited bandwidth links. German FKIE, the Computer Sciences department of the University of Bonn have carried out the researches related to development of such mechanisms. They give an opportunity for realisation of RT and

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non real-time (NRT) services simultaneously over the narrowband links with 9.6 kb/s and lower bandwidth. These mechanisms have been implemented in the device (software) named a Network Adapter (NA).

The NA is a network element that is responsible for multiplexed transfer of RT (e.g. VoIP) and conventional (e.g. http, email, ftp) data across a narrowband link with special support of QoS and DiffServ. According to this, the header compression, priority handling and control traffic reduction mechanisms have been implemented. The NA uses the information from the *traffic class*- and the *flowlabel* - fields of the IPv6-header. The capability of connection of two segments of Ethernet network over the narrowband link and the RT and the NRT services availability is the main task of the NA (Figure 1).

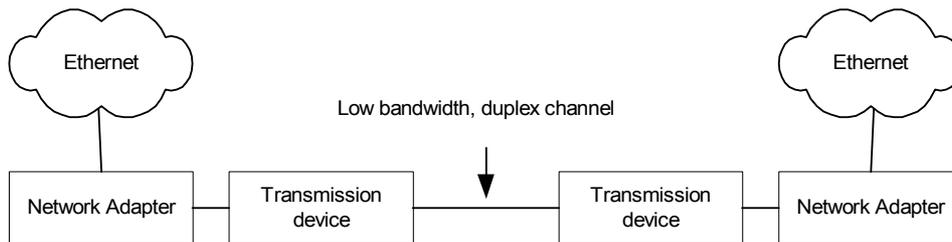


Figure 1. Position of the Network Adapter in telecommunication architecture

For each stream of data exchanged over the narrowband link, the NA recognizes the type of data. The NA also recognizes the priority of concurrent data flows what gives an opportunity to serve them with necessary QoS in the presence of limited bandwidth resources. Useful information, as source and destination addresses, priority, the RT or the NRT service, required bandwidth (if RT service), flow identification (if not 0), a corresponding connection index (ν_i) and the use of IPsec with separate enabled header compression (if the RT service) are stored in continuously updated connection table. In the next step, the header compression and data multiplexing processes are performed. The non-real time data are picked up from the queue related to a specific priority level, fragmented (in most cases but not generally) and filled into the gaps in between unfragmented packets belonging to real-time streams. The NA and related mechanisms as the header compression, priority handling and network control traffic reduction have been presented in [1, 5]. The results of first experiments presented in [3, 6] show the proper behaviour of the NA and its QoS mechanism in the presence of limited bandwidth links. They have been obtained for a real-life implementation of the NA in a simple configuration with 9.6 kb/s narrowband link. An estimation of the NA behaviour in more sophisticated network configuration and attributes enables the simulation model. Polish MUT/MCI team developed the model and it is based on the specifications given in [5] and functional algorithms presented in [2]. The results of this work are presented in the paper.

2.0 THE NETWORK ADAPTER SIMULATION MODEL

The basic objectives of the NA modelling and simulation is to establish an environment that is close to reality and it allows to investigate the wide tactical systems. The further objectives concentrates on:

- Verification of proposed compression and multiplexing mechanisms and procedures
- Estimation of adapter functionality in the diverse functioning conditions
- Evaluation of the limited bandwidth channel impact on the NA functionality and on the offered quality of services.

The researches carried out according to above objectives will make possible the estimation of usefulness of NA for using it in IPv6-based tactical military networks with limited bandwidth links.

2.1 Model assumptions and limitations

The NA model has been developed using the OPNETv.8.0 simulation package [4]. Considering the simulation objectives, the NA functional description and the abilities of simulation tool, the model requirements have been defined. It has been assumed that the simulation model of NA will be a separate network element, equipped with at least two interfaces: one interface for connection to narrowband link model or radio link model and one or more interfaces for connection to models of the Ethernet network. The characteristics of NA interfaces should be compatible with standard models of interfaces that are available in OPNET v8.0. Since the real-life implementation offers the functionality of VoIP QoS within IPv6 protocol as well as the possibility of data transfer with different level of priority, then the NA model is equipped with this functionalities as well. What is more, it gives an opportunity to collect the simulation statistics that will be useful for estimation of NA model behaviour according to the simulation objectives.

For this reason, the assumption and limitation that are described below have been adopted. The working environment is based on the Ethernet 10BaseT network model. The standard network models of OPNET use IPv4 protocol. It is inconsistent with real life implementation and for this reason the Network Adapter model will accept IPv4 packets and transform them into IPv6 packets. In this way, the functionality of IPv6 protocol is being achieved. The control packets such as ICMPv6, IPv6 routing packets and IPv6 packets with optional headers (i.e. Mobility Header, Destination Option Header, Router header) are generated using an additional application that transfers them through NAs. The others packets such as ICMPv4 or ARP packets are also handled as the control packets.

The NA handles the RT and NRT data. In the case of RT data with UDP protocol, the four lower layers of the model will be used. It means, the NA supports the Ethernet, Data Link, Network and Transport layers. This is because the header compression mechanism reaches the UDP header. The RT packets are handled with the higher priority, while lower priority and control packets are placed into the waiting queue. Next, they are segmented and transferred in the gaps between real time packets.

The narrowband link was modelled by full-duplex link model with throughput changeable within the range from 0 to 64 kb/s. The Ethernet interfaces were also configured to work in a full-duplex mode.

Along with model assumptions and limitations some variables and input parameters were defined to describe the NA model and the working environment. Some of them are presented in Table 1.

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Table 1: Input parameters of the NA simulation model

Variable name	Description
1	2
Low bandwidth link data rate	Data rate in low bandwidth link. This input is used for gaps calculation and buffer handling procedures. 9600 bist/sec is set as a default value.
Gaps prediction rule	The gaps between RT datagrams prediction rules. The gap prediction algorithm is based on the prediction of future gap between RT packets based on the gaps calculated between any numbers of RT packet in the past. The rule is based on the minimum gap or on the average gap from the history.
Gaps prediction history table range	Predicted gap length and the time of its validity depend on the number of history table entries. The history table has to hold at least two entries and no more then 100 entries.
Low bandwidth link buffer size	An overall output buffer size on the low bandwidth interface.
1	2
RT data buffer size	Buffer size for the RT datagrams.
NRT data buffer size	Buffer size for the NRT datagrams.
NRT data segmentation	A switch to disabling the NRT datagrams segmentation. It is used for testing of NA procedures efficiency.

Realization of established objectives requires definition of estimation measures that will be investigated during the simulation research. The definition of evaluation measures gives an opportunity of statistics selection. They are presented in Table 2.

Table 2: The evaluation measures

Evaluation measure	Description
Predicted gap length	Gaps length predicted in each NA using gaps prediction algorithm.
RT/NRT queue sizes	RT/NRT output queue characteristics.
RT/NRT queuing delay	RT/NRT output queue characteristics.
Segmentation buffer usage	Nr of bits in segmentation buffer in each NA.
Low bandwidth link utilisation	An utilization of the low bandwidth link connected to the NA in each direction.
Low bandwidth link throughput	A throughput of the low bandwidth link connected to the NA in each direction.
Ethernet link utilisation	An utilization of the Ethernet link connected to the NA in each direction.
Ethernet link throughput	A throughput of the Ethernet link connected to the NA in each direction.

The end-to-end measures used for QoS evaluation are as follow:

- Voice packets end-to-end delay
- Voice packets jitter

- Packets round-trip time
- Applications upload/download time.

2.2 General concept of modelling

The NA model was developed according to the specification given in [5]. The NA architecture presented in [5] enforced the general concept of NA modelling. It is shown in Figure 2. Detailed model specification can be found in [8]. The model allows receiving all Ethernet frames in promiscuous mode. IP packets are extracted from the frame and based on the information included in the ToS field, the type of data is recognised. If the RT packet is received, the Ethernet, IP and a part of UDP headers are processed and compressed, depending on the situation shown in Figure 3.

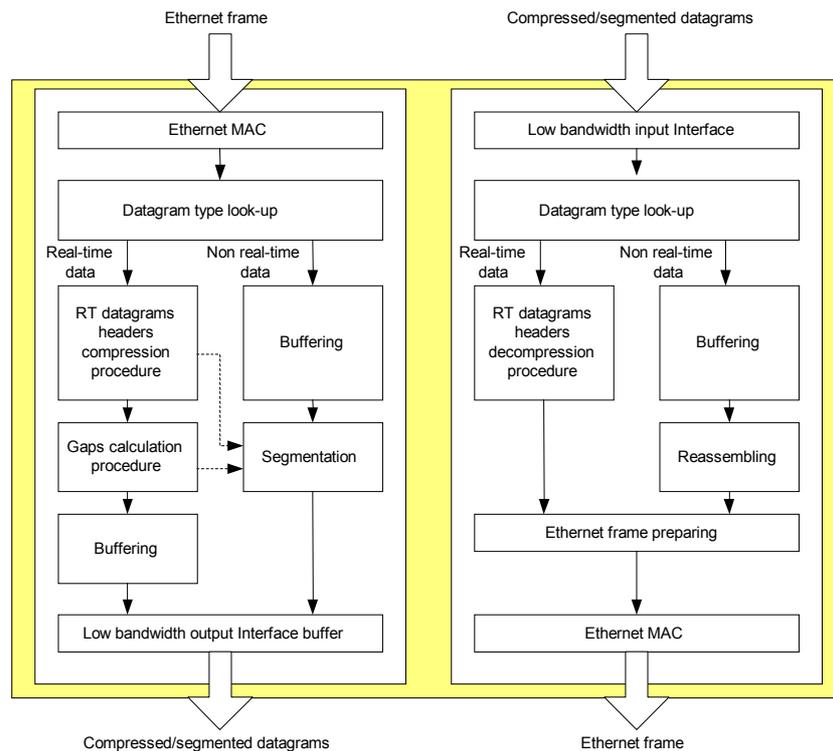


Figure 2. General concept of NA modelling

If there is the first IP packet from the RT stream received by the NA, then 0x8a flag is used and free “vi” index is assigned. The headers, together with *vi* index and current time are written to the memory. The flag, *vi* index and a checksum is included to the original Ethernet frame and such datagram is passed on to the gaps calculation procedure. If headers were written based on the previous packets, *vi* index is found in the memory and together with 0x8b flag is added to the new, compressed datagram, that does not include the Ethernet, IP and part of UDP headers. Such datagram is then passed on to the gaps calculation procedure. After buffering, the RT datagrams are placed into the low bandwidth interface buffer and sent to the receiving NA. The NRT packets are buffered and segmented. The segments calculated based on the information from gaps calculation procedure are placed into the output interface buffer based on the interrupts coming from the RT packet handling procedures. It allows sending them between the RT datagrams.

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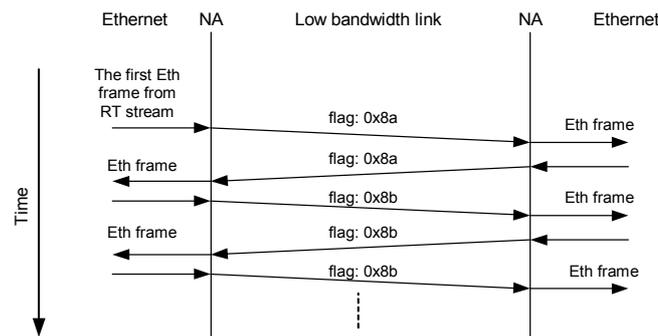


Figure 3. Real-time datagrams flow between NAs

The NA model allows receiving datagrams prepared by other NA models. Based on the flag included to the datagram, a type of data is recognised. If the RT data packet is received, vi index is read. If this is the first packet from the RT stream, the headers together with “ vi ” index and current time are written to the memory. If not, based on the vi , headers are found in the memory and full Ethernet frame is rebuilt. Such frame is passed on to the Ethernet MAC model and transmitted to the Ethernet network. The NRT segments are collected in reassembling buffer. If full Ethernet frame covering the NRT data is collected, it is passed on to the Ethernet MAC model and transmitted to the Ethernet network.

2.3 The NA model description

Based on the modelling concept presented in section 2.2, the NA model was created as a separate device in OPNET meaning. It allows connecting the NA model to a narrowband link model from one side and to the Ethernet network model from other side. The NA node model consists of following elements (Figure 4):

- Basic NA module (“NA”)
- Ethernet MAC module (“mac_0”)
- Transceiver in the interface to the Ethernet network models (“eth_rx_0”, “eth_tx_0”)
- Transceiver in the interface to the low bandwidth link (“RS232_t2”, “RS232_r2”)
- Streams connecting the modules in the node.

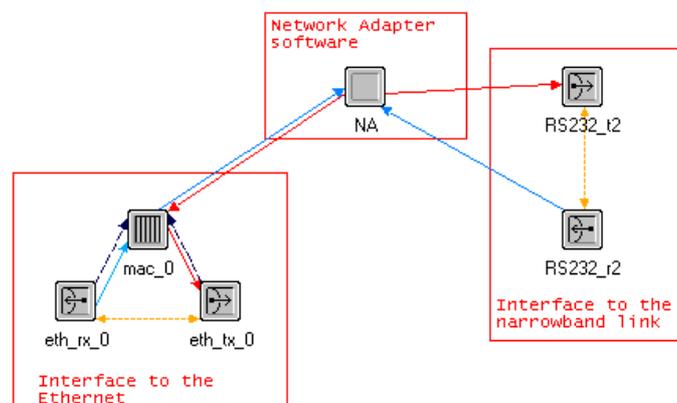


Figure 4. Modules included in the NA node model

The main process of the Network Adapter is located in NA module. This process is a reflection of algorithm presented in section 2.2. It contains the mechanisms of header compression and data segmentation and multiplexation. Among them the most interesting is the process of gaps calculation. It enables to calculate the time gaps between successive RT packets and fill them with NRT data segments. In contrast to real-life implementation two different algorithms of gaps calculation were implemented. They give an opportunity to estimate how the selected algorithm influences the behaviour of network adapter. Both of them are based on preparing the history table in which the information about predicted gaps is included. Depending on the range of the table, we can take into account some number of predicted gaps between successive RT packets, previously received by the NAs. The difference between algorithms used here is that the first one uses the minimum value from the history table in order to calculate the NRT segment size, and the second one uses the average value.

Interfaces between NA module and the Ethernet network models are ensured thanks to the process defined in MAC_0 module. This process is based on the Ethernet MAC process obtained from the standard OPNET models family. It represents the MAC layer of an Ethernet 10BaseT interface. The role of this process is to accept data packets from the NA module, encapsulate this data into Ethernet frames, and to send these frames in first-in-first-out order to the Ethernet network.

The model has been validated using both theoretical calculation and real-life test. The results collected during real-life testing can be found in [10].

3.0 THE EXPERIMENTS AND INTERPRETATIONS

Two basic network configurations have been assumed for performing simulation experiments. The first one (simpler) consists of two NAs connected together using 9.6 kb/s link and two Ethernet terminals connected directly to the NAs by the 10Mb/s Ethernet links (Figure 5).

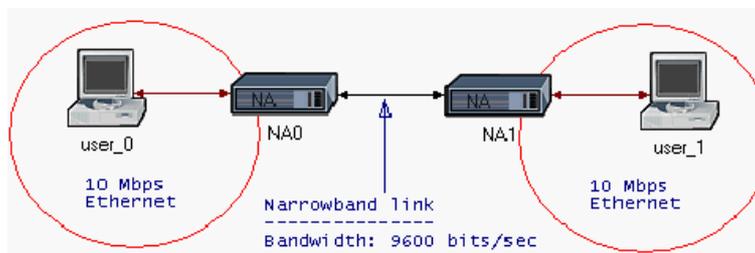


Figure 5. Simulation scenario – simple network

Let us assume that a terminal *user_0* initiates the VoIP call (the RT data service). Each terminal uses MELP 2.4kb/s vocoder. The call starts at 200 second of simulation time and it last for 1000 seconds. Figure 6 shows a traffic sent by voice terminal during the simulation time (a) and the Ethernet link throughput (b) caused by the call. We can see that the Ethernet link throughput is about 6.8 kb/s while from the source about 3.4 kb/s voice stream is generated. Of course, the Ethernet, IP and UDP headers cause the additional throughput in the Ethernet link. If such traffic appears directly in the 9.6kb/s link, it will take nearly whole bandwidth.

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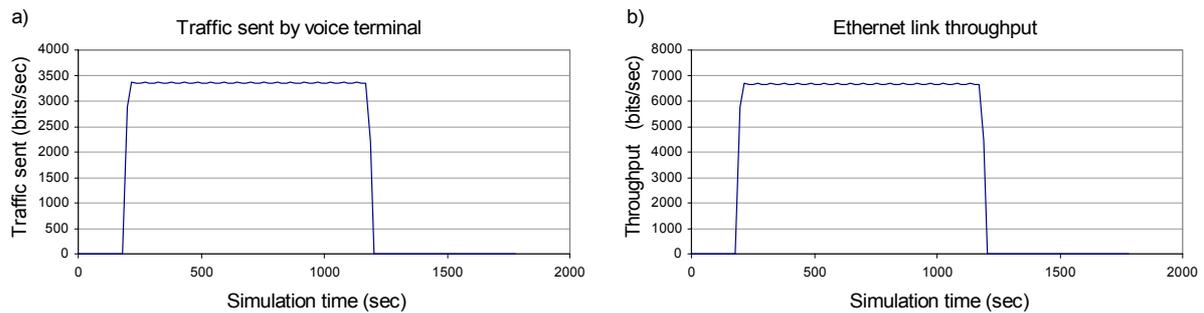


Figure 6. Traffic sent by voice terminal during the simulation (a) and the Ethernet link throughput (b) caused by voice

Additionally, the NRT data sent (for example FTP file or simple ping packets) during the call can cause that quality of voice will not be acceptable. Confirmation of this statement is shown in Figure 7. Five 1500 Bytes and two 10000 Bytes IP packets (NRT data) have been sent during the call over the 9.6 kb/s link. The NAs have been disabled. Three types of experiments have been performed: only voice was transmitted (Red), voice and NRT data were transmitted using simple FIFO scheme (Green) and using Priority Queuing (PQ) scheme (Blue). Figure 7a shows that voice packets end-to-end delay is not acceptable while NRT data are transmitted, even if standard priority queuing is used. The same situation is with voice packets jitter that is shown in Figure 7b. It reaches nearly 100 ms, even if PQ is used. Hence new mechanisms have to be used in order to ensure correct QoS. The next figures confirm that proposed mechanisms can be used for this purpose.

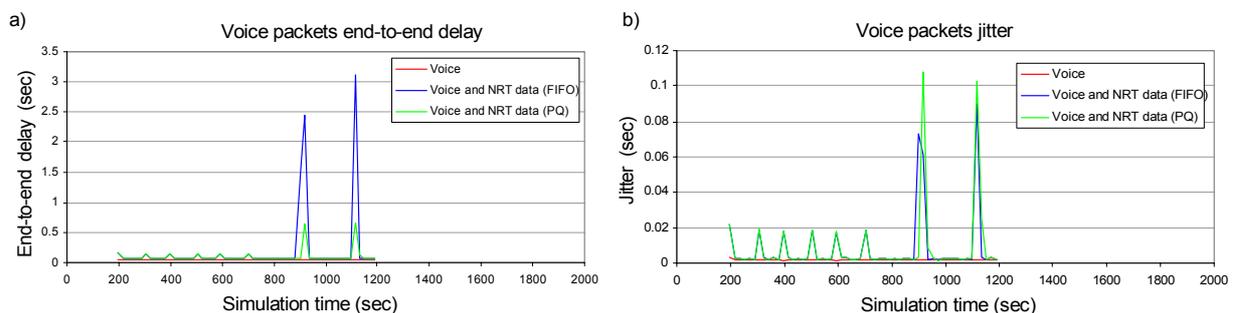


Figure 7. Impact of the NRT data packets onto the voice packets end-to-end delay (a) and jitter (b) while standard QoS schemes (FIFO, PQ) are used

Figure 8a shows that using header compression (the Ethernet, IPv6 and part of UDP headers) effective throughput in the narrowband link can be decreased nearly by half comparing to the Ethernet link throughput. In consequence, the narrowband link utilisation is just in about 40% (Figure 8b). Concluding, using only header compression, additional voice steam can be served over the 9.6 kb/s link.

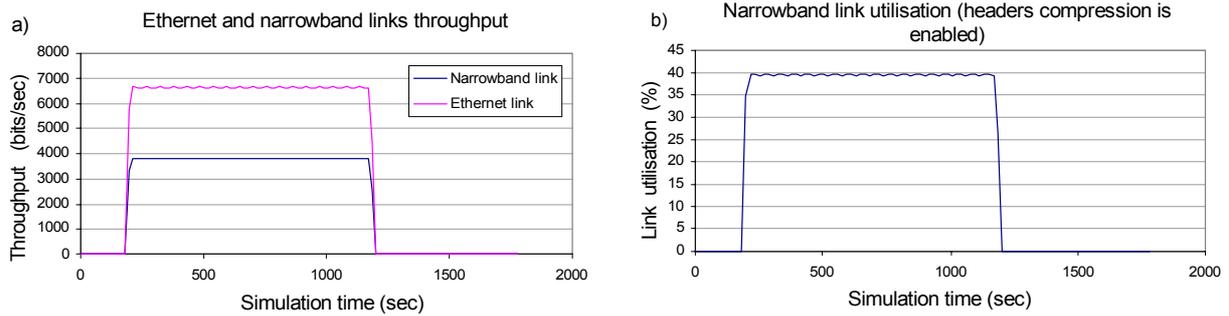


Figure 8. The Ethernet and narrowband link throughput (a) and utilisation (b)

Using the header compression without other mechanism we can handle additional calls, but the problem of simultaneous transmission of the RT and the NRT data still exists. The gaps between the RT packets prediction and the NRT datagrams segmentation are the solution for this problem. Enabling these mechanisms and performing simulation experiments depending on simultaneous voice and different NRT IPv6 packets transmission we can observe the NA and QoS behaviour. Figure 9 shows the end-to-end round-trip time of the NRT IPv6 packets having 148, 548, 948 and 1348 Bytes and sent periodically. From this figure we can learn that the NRT packets end-to-end transmission time is longer during the call transmission then while the link is fully accessible. The difference in time is nearly a half. Such behaviour can be simple confirmed by the calculation of remaining bandwidth (during the call) in narrowband link comparing to full accessible bandwidth. Analysing the impact of the NRT packet size onto its round-trip time we can see that the dependency is nearly linear.

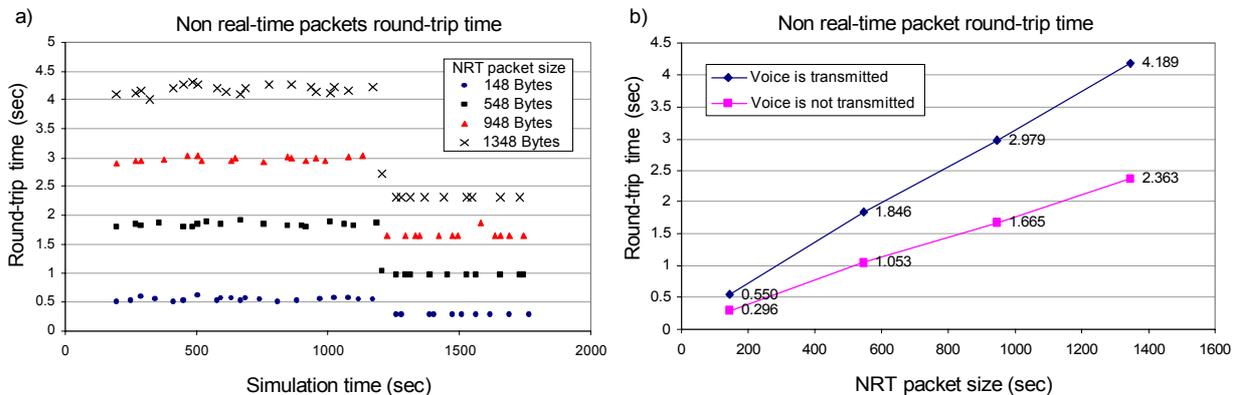


Figure 9. Non real-time packets round-trip time (versus simulation time (a) and versus NRT packets size (b))

More interesting is the impact of the NRT packets transmission on the quality of voice. The voice packets end-to-end delay during NRT data transition is shown in Figure 10. A small influence on the voice packets end-to-end delay is observed if the NRT packets change from 148 to 1348 Bytes. Nevertheless, the quality of voice is still on acceptable level – the RT packets end-to-end delay not exceeds the 58ms (this concerns the first packets from the stream that are not compressed). The higher reaction can be observed while transferring the bigger NRT packets or files (discussed later).

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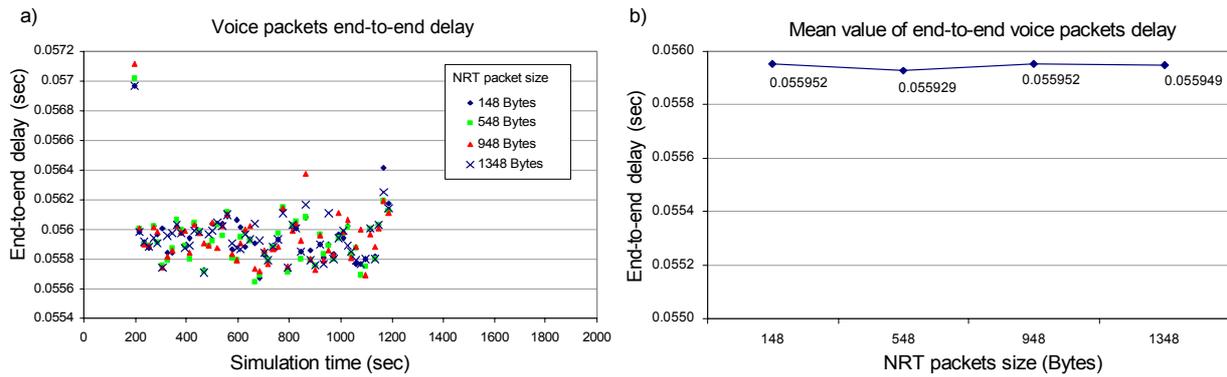


Figure 10. Voice packets end-to-end delay (NRT packets are sent during voice transmission)

Similarly, impact of the NRT packets on the voice packets jitter is small (Figure 11). The tendency for jitter to decrease is observed while increasing the NRT data packets. The jitter does not exceed 4ms. The jitter is calculated according to the rule proposed in [9] for a Transport Protocol for Real-Time Applications (RTP).

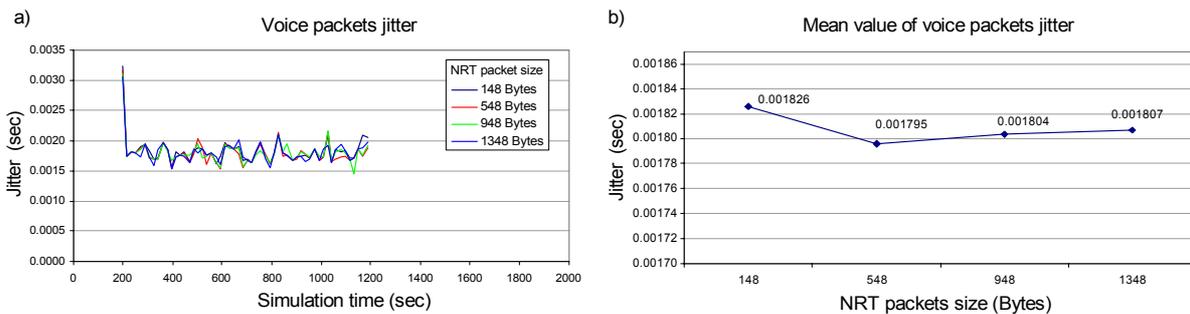


Figure 11. Impact of the NRT packets size onto the voice packets jitter

Above minimal influence on quality of voice, results from two factors: 1) the NRT packets are small enough in order to do not increase traffic in narrowband link significantly, what is observed in Figure 12, and 2) the QoS procedures implemented in the NAs are effective.

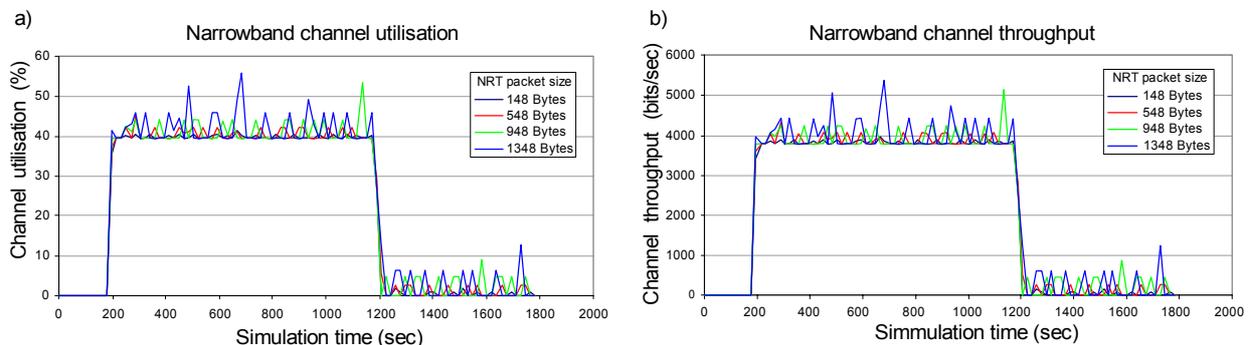


Figure 12. Impact of the NRT packets size onto the narrowband channel utilisation (a) and throughput (b)

More interesting results are shown in Figure 13. The *user_0* has sent 60000 Bytes packets to its correspondent and it receives the same answer. Of course, such big packets are fragmented onto the 1500 Bytes fragments resulting from the maximum Ethernet frame. Figure 13a shows the influence of these NTR packets onto the voice packets end-to-end delay while two types of gaps prediction rule have been selected.

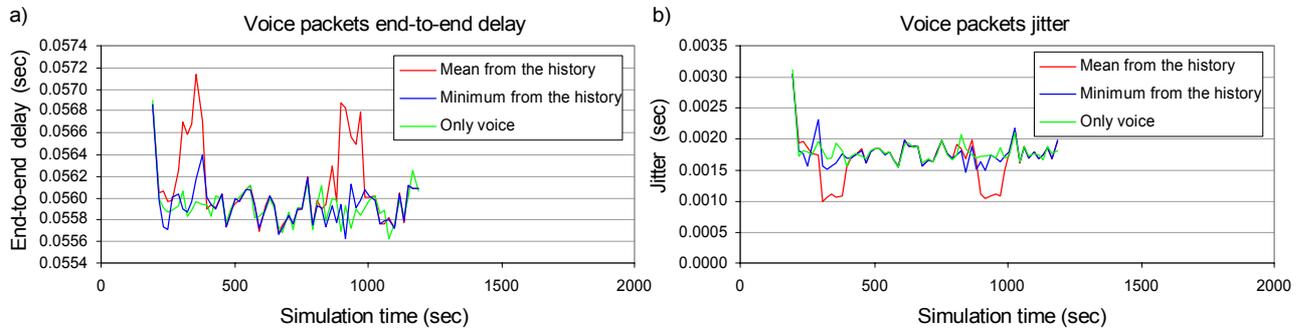


Figure 13. Impact of NRT data and the gaps calculation method onto the voice packets end-to-end delay (a) and jitter (b)

The highest end-to-end delay has been observed while gaps between RT packets are predicted based on the mean gaps written in the history table. It is because the rule “mean from the history table” can lead to the temporarily longer NRT segments transmitted over narrowband link, while the actual gap between RT packets is smaller. The reaction is, the voice packets have to wait in the interface queue. At the same time, the NRT packets cause decreasing the voice packets jitter while „mean from the history table” rule has been selected (Figure 13b). Enforced ordering of the RT packets causes such reaction because they have to wait a moment in the interface queue.

Gaps prediction rules are based on the history table entries. Figure 14 shows an influence of the number of entries in the history table on the quality of voice. General conclusion is that more than 10 table entries have not significant influence on the voice packets end-to-end delay (Figure 14a) in both cases of gaps prediction rule. Also more than 5 table entries have not significant influence on the voice packets jitter (Figure 14b) in both cases of gaps prediction rule.

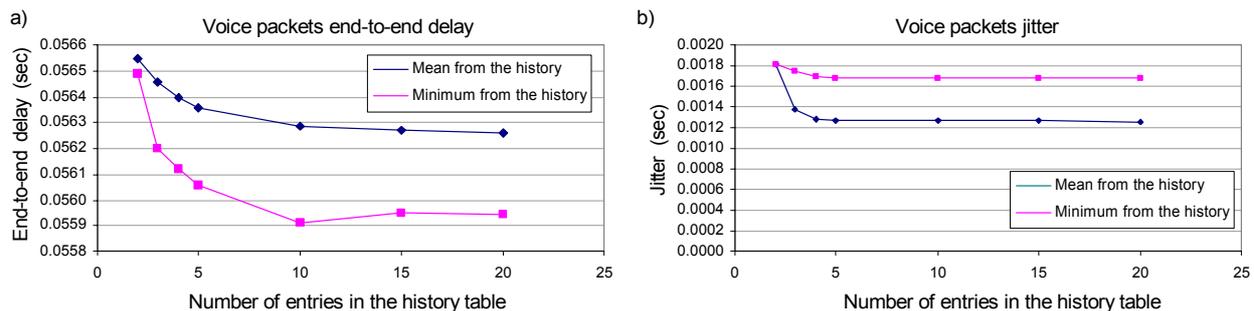


Figure 14. Impact of number of entries in the history table onto the voice packets end-to-end delay (a) and jitter (b)

From Figure 13 and Figure 14 we can conclude that the “minimum from the history table” gaps prediction rule have to be used taking into account the quality of voice. But after deeper analysis we can see that the increasing of QoS is not very high, but it can also cause increasing of end-to-end delay of NRT applications data by decreasing of the NRT segments size.

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The analysis of simple network configuration shows that based on the QoS-supporting mechanism proposed for IPv6 network parts covering narrowband links, the real-time and non real-time data can be transferred simultaneously in acceptable QoS level.

Let us also shortly analyse the extended network with a typical Internet applications sent over the narrowband link. Figure 15 shows the extended network configuration that has been used for simulation experiments. Two routers and a subnet model have been included in each Ethernet segment. The subnet is used to emulate additional IP packets latency that can appear in real networks. Packets latency has been drawn here using exponential distribution with 4 ms mean. The RIP protocol has been selected in the routers for routing. The *server* node has been also included in one of the Ethernet segment to simulate Internet services.

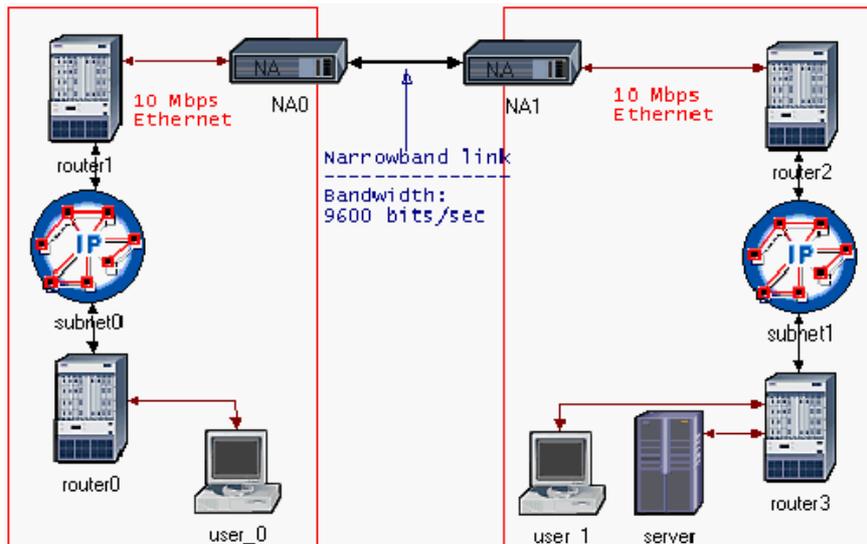


Figure 15. Simulation scenario – extended network

Lets also assume that just one voice call has been generated between the terminals. Additionally, in the same time, the File Transfer Protocol (FTP) has been used by the *user_0*, to periodically download files from the server. Every 200 seconds, starting from 210 second, 20000 Bytes files have been downloaded. Figure 16a shows the time of 7 files downloading. Four of them are downloaded during the voice call and the time of their downloading not exceeds 45 seconds. As we show in the first part of the paper, the influence on the quality of voice transmission is small. Voice packets end-to-end delay and jitter is shown in Figure 16b. Both characteristics do not exceed acceptable levels of QoS.

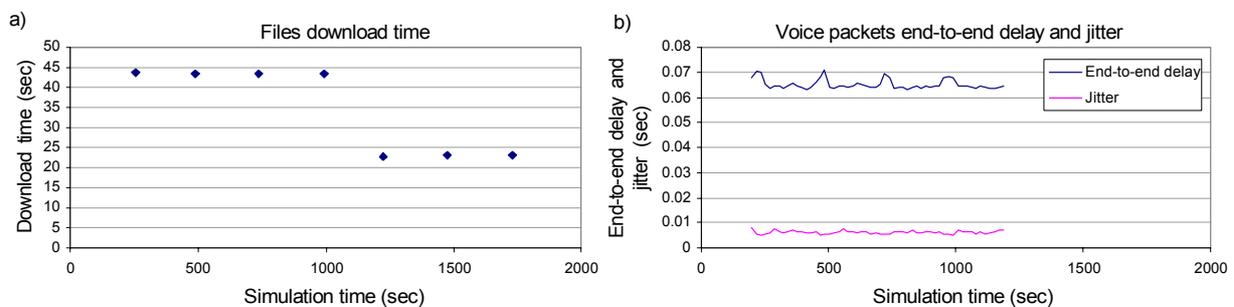


Figure 16. Files download time (a) and voice packets end-to-end delay and jitter (b)

The narrowband link throughput reaches 9.6 kb/s while voice and FTP are used simultaneously (Figure 17a). It means that the link is utilised nearly in 100% during this time (Figure 17b). In a typical situation (without NAs), the voice calls would be disconnected because of to high voice packets latency and jitter.

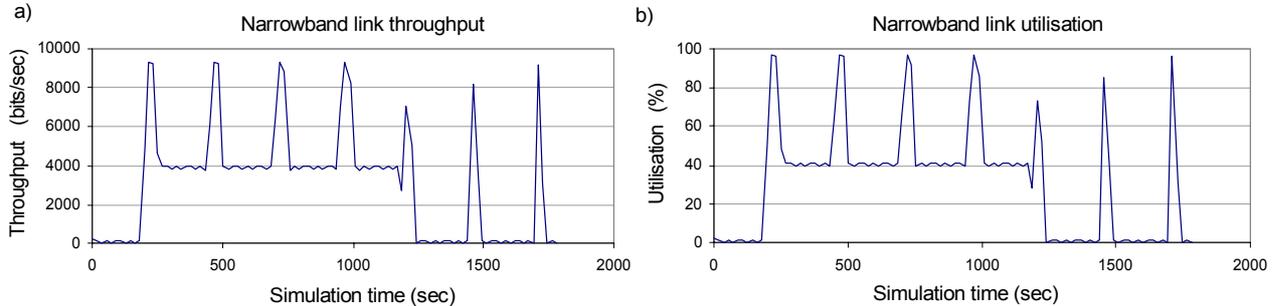


Figure 17. Narrowband link throughput (a) and utilisation (b)

The file downloading time depends on a prediction of gaps between the RT packets. Figure 18a shows the gaps times calculated during the simulation, based on the average from the history table including 5 entries. Using the NA attribute that defines the expected narrowband link bandwidth, the segments of NRT packets are calculated.

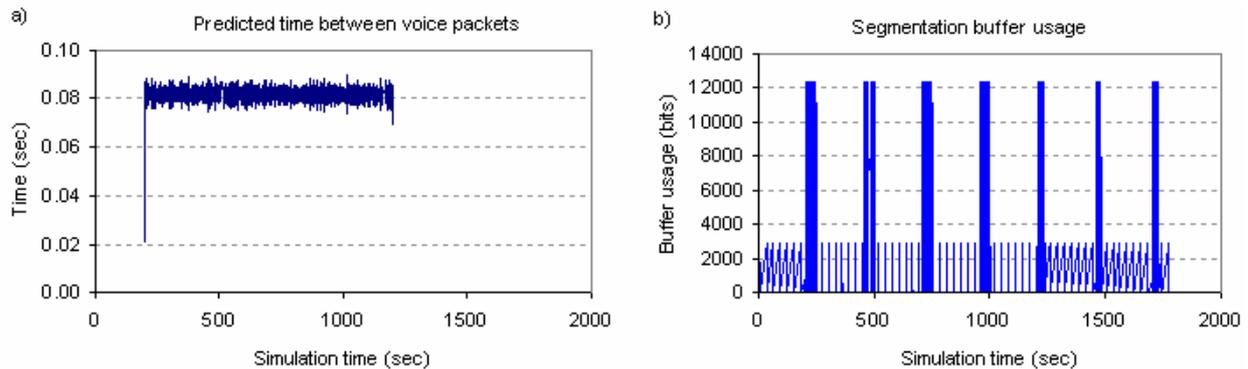


Figure 18. Impact of time between voice packets prediction onto the NRT packets segmentation

Figure 18b shows the segmentation buffer usage. The stripes visible in the figure are in reality the periodically changing steps that mean that the NRT segments are transmitted to the link interface. The first four stripes are wider than other. It means that files are downloaded longer. The small lines between the stripes means the RIP packets have been transmitted without segmentation, because they are smaller than 255 Bytes.

4.0 CONCLUSIONS

Narrowband links are still used in military networks. The fact is that the Internet services together with standard voice or video have to be accessible also over these links using IPv6 protocol. The simulation experiments show that the new mechanisms have to be used here in order to ensure adequate level of real-time data quality. What is more, proposed solutions allow simultaneous transmission of real-time and non real-time data keeping the real-time packets end-to-end delay and jitter on accessible level. It of course influences the application downloading (uploading) time, but this is a cost of higher priority of real-time applications. The model presented here will be used for the assessment of wide tactical network based on the IPv6 protocols stack.

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