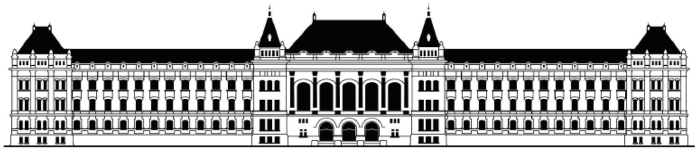

NEW METHODS FOR IMPROVING MULTIMEDIA QUALITY IN IP NETWORKS

Ph.D. Dissertation Summary

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1. Introduction

Multimedia applications are gaining prominence on the Internet in both wired and wireless environment. New wireless access technologies now provide higher bandwidth, making possible to use audio/video services, however, multimedia streaming applications still suffer from limited bandwidth, high delay and loss rate. These channel characteristics and the effects of handovers can distort the transmitted video stream.

The transport layer protocols that can be categorized as unreliable or reliable ones, depending on the retransmission mechanism deployed in it. Retransmitting all the lost packets and disabling retransmission in general are marginal solutions, therefore my idea was to found a middle way. If the network characteristics can not be changed, the streaming application must adapt itself the actual link conditions to improve the video quality. I have developed different adaptive retransmission decision algorithms, based on network load, delay and packet importance. Unreliable transport protocols used by multimedia applications do not retransmit the lost or damaged packets, although it could be possible in lot of cases. In my semi–reliable streaming solutions I have utilized the advantages of DCCP protocol [1], e.g. acknowledgements, information about the networks conditions provided by the congestion control mechanism deployed in DCCP. The retransmission method can be divided to content–based [2][3] and network characteristic–based [4][5] selective retransmission algorithms already studied in several papers.

Today's mobile devices are equipped with multiple interfaces to make the connection possible to different type of networks. In order to efficiently utilize the interface capabilities, multi–path multimedia streaming can be used. I have proposed a multi–path streaming method that chooses a set of paths maximizing the overall quality at the client. While the available paths have different bandwidth, delay and loss probability constrains, the packet distributor must take the video packet importance and the dependencies between packets into account. Using content–aware multi–path streaming, the available networks must be ordered. Numerous Multi Attribute Decision Making (MADM) methods [6][7][8] have been utilized for network selection purposes, but majority of these methods suffer from the *rank reversal* phenomenon. Rank reversal means that the relative rankings of two decision alternatives could be reversed when a decision alternative is added or deleted. I have found new normalization solutions that significantly reduce or even eliminate the unwanted phenomenon.

In mobile networks the effects of handovers on delay and administrative load are significant. Mobile IPv6 (MIPv6) protocol [9] was introduced in order to hide the roaming of the mobile clients and decrease the caused delay and additional load due to handovers. New extensions of MIPv6 appeared (HMIPv6 [10], RegReg6 [11]) to hide the handovers inside the domain. The standards do not address the realization considerations, however, the network load and delay caused by handovers can be reduced by locating the mobility agents in the domain structure efficiently. There are several potential mobility agent routers inside a RegReg6/HMIPv6 capable hierarchical domain and it is not obvious how to choose the best one. I have presented a method, showing how to configure network in order to reduce signaling traffic and delay. The proposed algorithm locates the mobile agents (GMA/MAP) in the hierarchical domain according to the mobile terminal's movement behavior. By reducing the number of GMA/MAP changes and reducing the hop number between the MN and GMA/MAP, the latency experienced by the users will be also lower. The reduced delay will make it possible to provide better quality of communication not only for multimedia applications.

2. Research Objectives

Multimedia applications are very sensitive to delay and losses, therefore these parameters must be kept on low levels, or adapting the transmission to the network characteristics. My aim was to develop new techniques and solutions to minimize the multimedia sensitive network parameters by modifying the network structure and localizing mobility management functionalities. By investigating new adaptive retransmission schemes and multi-path streaming methods I could also achieve lower packet loss ratio, delay and better video quality at the receiver.

Regarding to the previously presented research field I have grouped my research activity into three topic:

1. Transport protocols can be categorized as reliable and unreliable. Reliable ones provide errorless data transmission by retransmitting the lost or damaged packets, but increasing the delay due to extra data delivery. Reliable protocols are not suitable for multimedia applications due to the increased delay caused by retransmissions, but on the other hand the unreliable protocols can not increase the quality. One of my research objectives was to found a middle way between the two marginal solutions. For this purpose I have investigated a semi–reliable method for multimedia applications, especially for video streaming. In these selective retransmission schemes the most important question is how to select the packets for retransmission. For the decision process I have utilized the some network parameters like packet loss rate, network delay, etc. I have proved that using my adaptive retransmission scheme based on DCCP and the information provided by the protocol, the packet loss rate can be reduced and the delay can be kept under a given threshold without any additional administrative packets. The result is better video quality compared to other traditional methods.
2. New coding algorithms (MPEG–4, H.264, etc.) appeared to make video streaming available for mobile equipments, but it is common between them that all of them use different frame types with different importance. The effect of an error or loss will be different depending on the priority of the frame. My objective was to transmit the video data flow based on the frame types. Simultaneous connections to several networks through multiple interfaces are possible with today's mobile terminals, however, the delay, loss rate, capacity, etc. is different for each network. I have introduced a new multi–path streaming method that chooses a set of paths maximizing the overall quality at the client.
3. The third topic was the investigation of hierarchical domains using HMIPv6 and RegReg6 from administration load point of view. I have shown that the administrative load due to handovers depends on the location of the mobility agent in the hierarchical domain. My presented method selects the most appropriate mobility agent (MAP/GMA) for a mobile terminal based on the mobile's movement behavior and the hierarchical structure of the domain. My solution generates more balanced administrative load, while it also effectively hides the movement of the mobile terminal causing less mobility agent changes. With this method reduced traffic load can be achieved improving the overall performance of the network.

3. Research Methodology

The dissimilarity of the presented research objectives needs different tools and techniques to efficiently investigate them.

To develop and analyze new methods for different techniques for improved video streaming and mobile agent selection in hierarchical network topology, analytical considerations could not be ignored. For evaluating the effectiveness of the new methods, I used statistics, probability theory for the analysis of large sets of data.

The most of my developed algorithms assume large heterogeneous networks, congested links, unreliable channels, etc. or uses not implemented novel protocols. Some of my new methods were implemented and analyzed with measurements in a testbed, however, the detailed examinations were done by simulations. I have used the popular NS–2 simulator tool, but I have also created own simulation frameworks in C/C++.

The effectiveness of some of my methods could be measured by comparing video streams. To do this, I used the widely accepted *peak signal–to–noise ratio* (PSNR) video quality measurement algorithm.

4. New Results

4.1. Selective Retransmission Schemes

Retransmission–based error recovery is considered inappropriate for multimedia applications, because of its latency. However, this solution can be attractive because it requires minimal network bandwidth, processing cost and efficiently improves the quality. Despite its latency, retransmission can be used successfully in many cases, especially if playout buffering is employed.

Standardized transport protocols can be categorized as reliable (TCP, SCTP) and unreliable (UDP, UDPLite, DCCP) protocols. Retransmitting all the lost packets and disabling retransmission in general are marginal solutions, therefore my idea was to found a middle way using selective retransmissions. The proposal is simple: retransmit the lost packets if the conditions make it reasonable and disable retransmissions if it makes no sense to deliver the lost packets once more. Traditional error control mechanisms use retransmissions to provide reliability at the expense of latency, but late retransmissions in real time application are undesirable because the receiver side process already skipped the lost packets. Similarly, when the network is in congested state or near to this state the retransmissions should be also disabled. The unrequired retransmissions waste network bandwidth and CPU cycles, contribute to congestion and may delay new data. The question is, when is the retransmission reasonable and when not?

I have investigated source controlled retransmission techniques utilizing the benefits of the DCCP [1] protocol. DCCP may use TCP–Friendly Rate Control (TFRC) [14] congestion control algorithm that can be used to estimate the available bandwidth and the network delay.. The importance of the lost data can be also used as aspect, because the video quality improvement highly depends on recovered data importance.

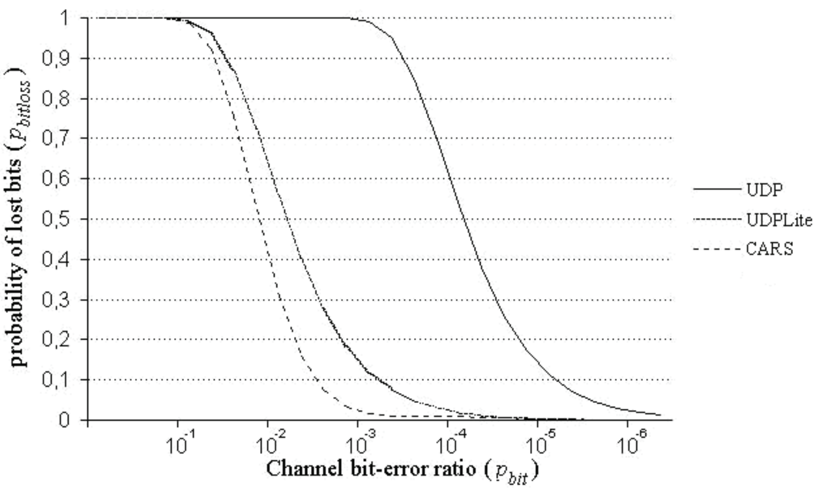
In the related works [2]–[5] the receiver controls the retransmission procedure by introducing retransmission request packets. In my proposals no administration messages are needed because the decision procedure is located at the transmitter. The other advantage of my transmitter side decision is that the input parameters of the decision algorithm (RTT, estimated link bandwidth, etc.) are available at the source side using the DCCP transport protocol.

I have investigated new semi–reliable retransmission methods that selectively retransmit lost or damaged user data packets based on different decision algorithms. The decision can be made according to the importance of a data packet and based on actual network conditions (delay, congestion) taking the multimedia stream features into account. I have also introduced an interdependence model between the congestion–based and delay–based retransmission schemes. To measure and compare the received video stream I have used generally used *peak signal–to–noise ratio* (PSNR).

I.1. Thesis [J6, C3] *I have developed a content–based retransmission scheme (CARS). I have shown that using the proposed CARS method the quality of the streamed MPEG video can be improved utilizing the DCCP protocol signaling process.*

The developed CARS (Content–Aware selective Retransmission Scheme) method is a DCCP/IP based selective retransmission scheme, which differentiates the lost packets based on its content. By successfully delivering the high priority packets, the quality of the streamed video can be significantly increased. The I–frames are more important then the other frames so these frames should be handled on a different way than other ones. If the damaged part of the I–frame is retransmitted, the quality of the streamed video should increase considerably because the MPEG's prediction method will be based on correct I–frame and the error will not spread to other frames.

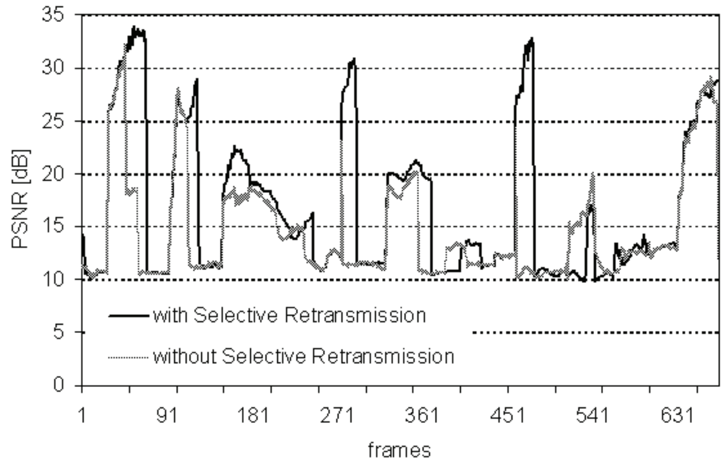
The sender side retransmission decision process can be solved by DCCP easily due to its sequence numbering and the partial checksum feature. The sender side must make the selection based on the packet importance. By retransmitting once the lost but differentiated packets, the probability of successful delivery of these packets will increase by $p(1-p)$, where p is the packet loss probability on the link.



1. Figure Probability of damaged or lost bits in the MPEG stream in case of UDP, UDPLite and the proposed CARS algorithm

The highest rate of bit corruption happens in case of UDP. A single bit error leads to the drop of the entire packet, while using UDPLite a single bit error leads to loss of one bit if it belongs to the application data.

Using the proposed CARS selective retransmission algorithm the bit loss probability is lower. The results show that the proposed selective retransmission algorithm is a very efficient method for increasing the quality of MPEG multimedia streams and even 50% less bit error probability can be achieved. The algorithm is especially effective in wireless networks with high bit–error ratio (10^{-2}–10^{-4} bit error ratio). I have implemented a DCCP/IP testbed to examine the behavior of the proposed CARS method. The improvement of the video quality is presented in the following figure.



2. Figure Measured video quality in the DCCP/IP testbed

The disadvantage of the method is that increased playout buffer delay is needed to make the retransmissions possible.

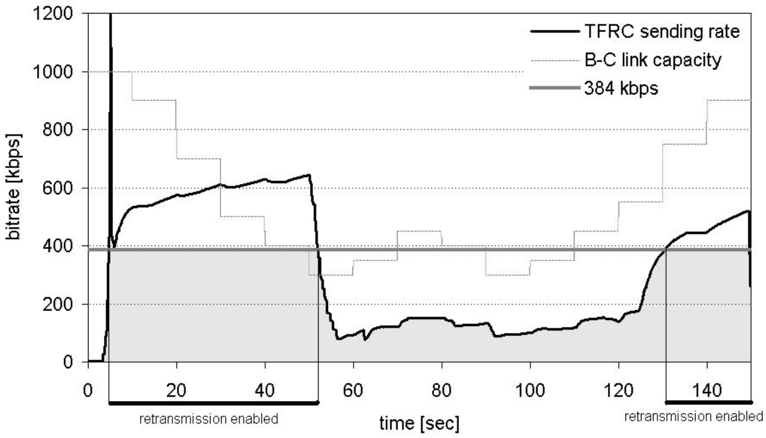
I.2. Thesis [J7, B2, C5, C11] *I have proposed a new method (TRS), which reduce the number of unsuccessfully retransmitted packets, based on the calculated TFRC sending rate and the video bitrate, while the successfully retransmitted packets improves the received video quality. I have also presented an independence model between the TFRC-based (TRS) and RTT-based (RRS) retransmission methods.*

When the network is in congested state or near to this state the retransmissions should be considered. In the case, when the buffers of the network routers are overloaded, the additional load will make the things worst. A retransmitted packet will be dropped at the routers or it will cause the loss of other packets. I have proposed the TFRC–based selective Retransmission Scheme (TRS), which disable or enable the retransmission of lost packets according to the current state of the network. The proposed TRS method uses the DCCP's TFRC (CCID3) [14] congestion avoidance algorithm to estimate the available bandwidth and decide whether the lost packet should be retransmitted.

Packet losses can occur in two different ways. The first one is due to router buffer overflow, the other one is due to wireless losses. In the first case no packets should be transmitted until the network is in congested state, because the packets will probably be lost again and the level of congestion will be increased. If the network routers have free capacity and the lost or damaged packets can be recovered. My proposal is to disable the retransmission when the calculated TFRC rate (X_{TFRC}) is under the video bitrate (X_{MPEG}).

$$X_{TFRC}(t) > X_{MPEG}(t), \quad (1)$$

If the proposed TRS method is used, the retransmitted packets will not overload the network. The TFRC congestion control algorithm can not adapt its sending rate immediately, however, this delay is not significant. When free capacity appears in the network, the calculated sending rate will also increase and make the retransmissions possible after an inertial delay. The TFRC–based retransmission method (TRS) does not need any additional traffic load to manage its functionality. All the needed information is provided by the DCCP protocol and the integrated congestion control algorithm. The transmitted packets should be stored in buffer on the server side to later retransmit the lost ones if possible.



3. Figure TFRC–based decision process

The NS–2 simulations showed that the video quality improvement was 15dB when the network was congested in 50% of the time. There was no retransmitted packet that was lost during the simulation.

The correlation of congestion level and RTT is obvious and already studied intensively. I have introduced a TFRC and RTT thresholds interdependence model, which utilizes the correlation of the TFRC sending rate and the RTT. The model estimates the correspondent RTT threshold using the TFRC–based sending rate threshold gaining similar retransmission behavior. Using the proposed model the number of retransmitted packets is similar in both cases, because corresponding RTT and TFRC throughput thresholds were used.

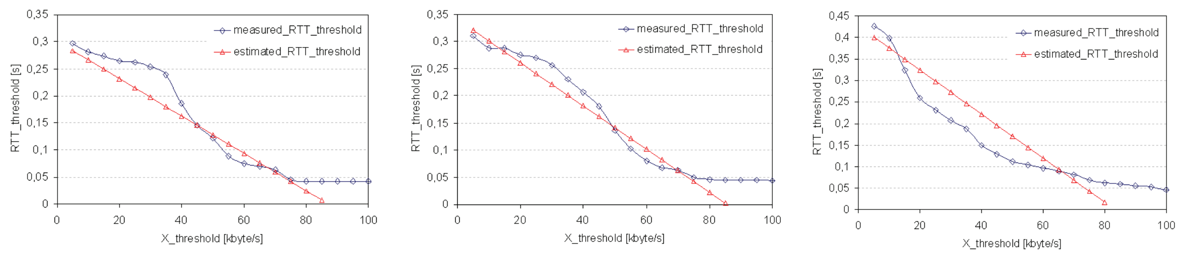
The TFRC–based method disables the retransmission when the TFRC rate is under a given threshold ($X_{Threshold}$), e.g. real–time video coding bitrate, so the lost packet can be retransmitted if $X_{TFRC} > X_{Threshold}$.

In the proposed estimation model I utilize the inverse relation of RTT and the offered sending rate of TFRC. The aim was to determinate $RTT_{Threshold}$ based on the number of retransmitted packets. The number of retransmitted packets should be equal using the TFRC–based threshold and the $RTT_{Threshold}$. In order to efficiently estimate the $RTT_{Threshold}$ from the $X_{Threshold}$, the relation of RTT and TFRC must be analyzed. The RTT and TFRC values are in inverse relation, as the TFRC equation it defines. The formal determination of the proposed linear estimation model is done in the following equation:

$$RTT_{Threshold} = \frac{(RTT_{min} - RTT_{max})}{2 \cdot X_{recv}} X_{Threshold} + RTT_{max} \tag{2}$$

Variable X_{recv} stands for the measured receiving rate, while the lowest/highest RTT values are considered as RTT_{min} and RTT_{max}. The RTT_{min} and RTT_{max} parameters are assumed as known values. RTT_{min} is the network delay when the network routers' buffers are empty, while RTT_{max} can be assumed as the round–trip–time when the network is in congested state.

The obtained results show that the model can be really effective when the network load is low. In case of higher congestion levels the estimation error increases to 20–30%. When the network load is below the congestion load the estimation error is only 1–2%. I have measured the corresponding thresholds, when the number of retransmissions is the same. The obtained results are illustrated in Figure 4.



a) 5 WWW users b) 10 WWW users c) 25 WWW users

4. Figure Linear estimation and measured values of $RTT_{Threshold}$

I.3. Thesis [C6] *I have developed a new retransmission decision algorithm (ARS) based on ARC (Analytical Rate Control), which is able to distinguish congestion and wireless losses.*

When used over wireless links, TFRC and TCP cannot distinguish between the wireless losses and the congestion losses. They both may suffer from the link underutilization if the connection traverses a wireless link. This is because they consider dropped packets as a sure sign of congestion and reduce the sending rate significantly. The inability to identify a wireless loss followed by unnecessary reduction in sending rate results in link underutilization. In order to avoid unnecessary retransmission prohibitions I have used ARC (Analytical Rate Control) [15] congestion control algorithm in the proposed ARS (ARC–based selective Retransmission Scheme) method. ARC is a rate–control scheme that uses the following equation:

$$X_{ARC} = \frac{1}{4RTT}\left(3 + \sqrt{25 + \frac{24}{p_c}}\right) \tag{3}$$

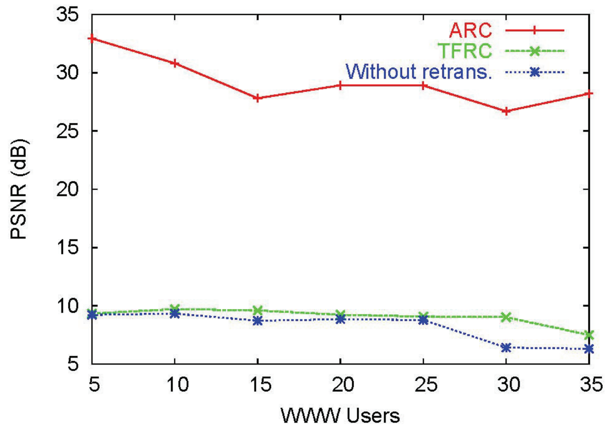
Parameter S is the sending rate in packets per second, RTT is the round–trip–time, and p_c is the congestion loss probability. The latter is related to the total packet loss probability π and the wireless loss probability ω through the expression:

$$p_c = \left(\frac{\pi - \omega}{1 - \omega}\right) \tag{4}$$

The proposed ARS (ARC–based selective Retransmission Scheme) is based on the DCCP transport protocol and the congestion control algorithms (ARC, WLED–ARC) integrated to DCCP. The decision algorithm disables or enables the retransmission of lost packets according to the current state of the network. To decide whether to retransmit a lost packet the ARC congestion control protocol is used. The congestion control algorithm calculates the actual sending rate to avoid congestion, while the video stream bitrate is absolutely independent from the calculated sending bitrate. When the network is in congested state or near to this state the calculated sending rate (X_{ARC}) should be lower than the video bitrate (X_{MPEG}).

$$X_{ARC}(t) \;>\; S_{MPEG}(t) \tag{5}$$

ARS performs significantly better than the other schemes when the wireless loss is high. This is because other schemes can not differentiate between congestion and wireless losses and the sending rate is very low. The video quality improvement of the proposed ARC–based selective Retransmission Scheme is illustrated in the following figure.

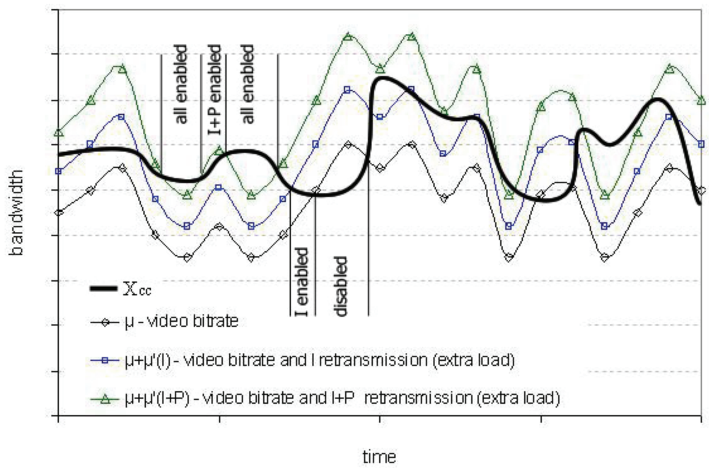


5. Figure Average PNSR without retransmission, with TFRC–based and ARC–based retransmission scheme (wireless loss is 10%)

I.4. Thesis [C7] *I have developed a new method (CNRS), which gains further video quality improvement when the network is near to congestion state. The CNRS controls the retransmissions by differentiating the MPEG (H.264) frame structure and the free network bandwidth.*

The aim of the congestion control algorithm is to reach the highest sending rate possible, but without causing congestion losses. By increasing the sending rate less and less free bandwidth remains that can be used for retransmissions.

I have proposed a novel Content–aware and Network–based selective Retransmission Scheme (CNRS), which allows the retransmission of all packets when the risk of congestion is low, but as it rises the retransmission is disabled step–by–step, but not all at once, in order of packet importance. The proposed retransmission scheme is a combination of content–aware and congestion control–based retransmission techniques. In this work the frame heterogenity of MPEG/H.264 video streams were utilized for the determination of packet importance. With disabling the retransmission of packets with specified content, we are able to control the overall stream bitrate between certain limits in order to avoid congestion. The following figure clearly illustrates the CNRS decision function.



6. Figure The decision algorithm of the proposed CNRS retransmission method

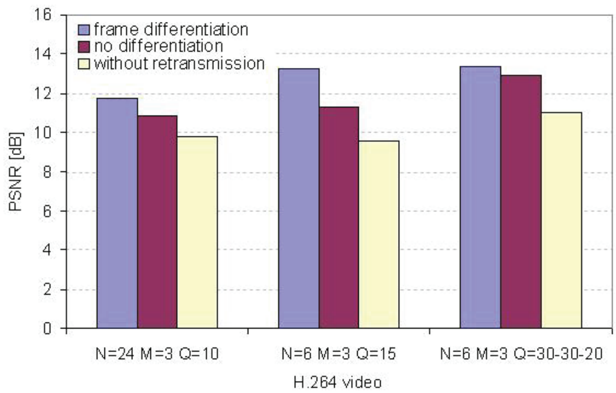
The stipulation of retransmitting I, I+P and all the frames can be determined in exact way as given in the following table.

1. Table CNRS: Frame type retransmission thresholds

	MPEG video structure (ρ_I, ρ_P, ρ_B)
I–frames	$\mu < X_{cc} < \mu + \mu'_I$
I+P frames	$\mu + \mu'_I < X_{cc} < \mu + \mu'_{I+P}$
all frames	$X_{cc} > \mu + \mu'_{all}$

Parameter X_{cc} stands for the calculated sending rate of the congestion control algorithm, μ is the video bitrate and μ' is the extra bandwidth caused by I, I+P and all frames retransmissions. The CNRS decision process will enable or disable different frame type retransmissions according to the additional load and the congestion control sending rate.

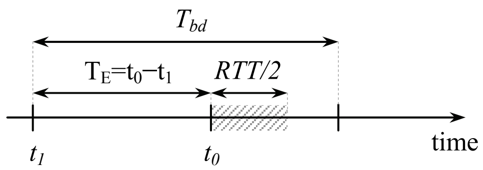
I have simulated the CNRS methods to analyze its performance in case of varying packet loss rate and congestion levels. I have compared my scheme with other techniques, which did not use retransmissions or frame differentiation (TRS). As the PSNR measurements show, the protection of I–frames effectively improves the quality of the analyzed three differently structured vide streams.



7. Figure Video quality measurements (PSNR)

I.5. Thesis [J1, C4, C8] *I have developed the delay sensitive selective retransmission scheme (RRS), utilizing the continuous measurements of TFRC algorithm. I have shown that the proposed RSS efficiently reduces the number of lately retransmitted packets, while the successfully retransmitted packets improve the video quality. I have also proposed a method (Flood method), which determinates how to upload the receiver buffer to make the retransmissions successful with probability higher then 95%.*

I have developed a source controlled selective retransmission method with a decision algorithm based on the actual round–trip–time (RTT). To minimize the probability of wastefully retransmitted packets, a playout buffer is set up at the receiver side to prefetch a certain amount of data before playback. The buffered data provides additional time to absorb the retransmission delay making the retransmission acceptable for one–way pre–recorded and one–way live media applications. I have proposed a DCCP/IP–based selective retransmission scheme, called RRS (RTT–based selective Retransmission Scheme), which disable or enable the retransmissions of lost packets according to the current network delay and the playout buffer delay. In order to determinate the remaining time before the playout, the source must know the playout buffer delay (T_{bd}). My aim was to control the retransmission decision from the source without administrative messages; therefore I have developed the Flood method to control the playout buffer uploading.

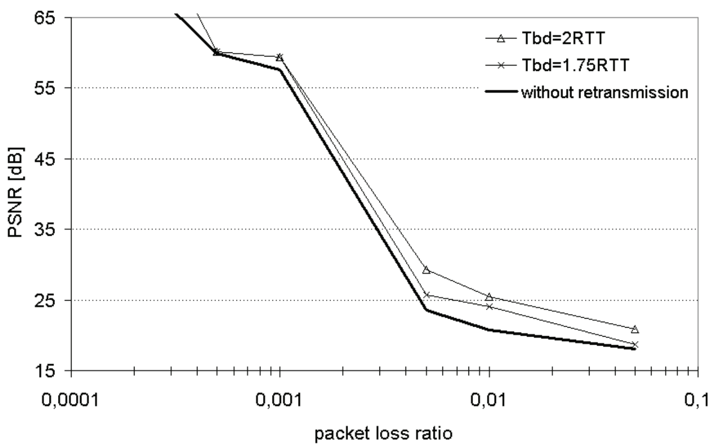


8. Figure Time sequence

After receiving an acknowledgement (t_0), that contains the sequence numbers of lost packet (sent at t_1), the transmitter should decide which packet is worth to retransmit. The decision algorithm calculates the remaining time that must be less then the one–way network delay for successful retransmission.

$$T_{bd} - (t_0 - t_1) > RTT / 2 \quad (6)$$

The main goal of my proposal is to improve the quality of video streams that was justified by simulations. The quality improvement depends on the receiver's playout buffer delay (T_{bd}), because if it is long enough, quite all the lost packets can be successfully retransmitted.

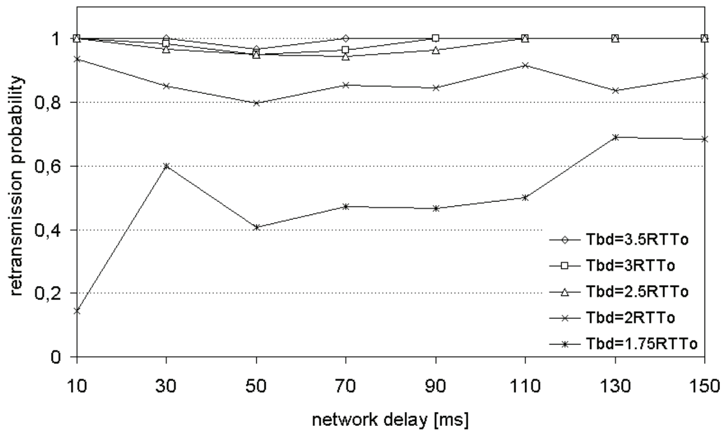


9. Figure Average PSNR of 150 seconds of 24 fps video as a function of packet loss

The limitations of the playout buffer delay (T_{bd}) are determined by the streaming application and the network delay. Too long playout buffer delay is not acceptable for real–time applications, while too short buffer delay will make the retransmissions impossible, because the retransmitted packet will not arrive in time. From the retransmission point of view the lower bound of the playout buffer delay is determined by the network delay. My goal was to find lowest delay possible to make the retransmission feasible. One of the input parameter of the decision algorithm is the actual RTT that is measured by the DCCP's TFRC congestion control algorithm.

To make the retransmission controlled by the source without administrative messages the transmitter must know the playout buffer level and its delay. My proposed solution is called Flood method. At the beginning of the video transmission the transmitter will not deliver the data packets immediately. It will heap up some data and transmit it all together after T_{bd} time. With this method the receiver will receive more data than it should process therefore certain amount of data will be heaped up in the receiver's playout buffer causing T_{bd} delay.

I have estimated the loss detection time using Normal and heavy–tailed Gamma distribution models of RTT. I have proved that setting the playout buffer delay to 3RTT with the Flood method, the retransmitted packets will be successfully received with higher then 95%. The analytical results were confirmed by simulations as well.



10. Figure The effect of playout buffer delay (T_{bd}) on the retransmission probability

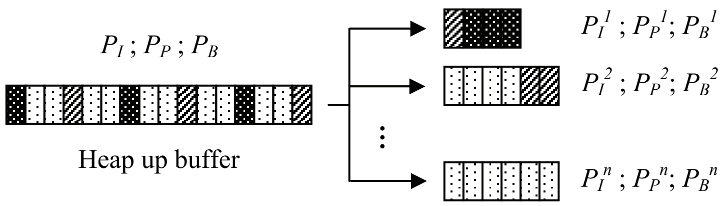
4.2. Multi–path Streaming and Network Ordering Abnormalities

In order to utilize the advantages of the heterogeneous network model, mobile devices are now being built as multihomed multi–functioning wireless terminals. Multi–path video streaming has recently been proposed as a solution to overcome the strict mobile network limitations. In order to increase the quality of video streams using multihomed devices that have multiple network interfaces independently connected to different networks (Ethernet, WLAN, UMTS, etc.), the available interfaces must be effectively utilized. The independent paths are characterized with different bandwidth, packet loss ratio and network delay parameters. I have developed the COMPAS scheme, which takes the video packet importance and the dependencies between packets into account and transmits the reference video frames on the most reliable links; therefore the error probability in the I–, P–, B–frame data will be different. Using the proposed packet distributor the loss probability of important data packets can be decreased, hereby increasing the measured video quality.

In order to efficiently distribute the packets, the link must be ordered based on the network attributes. Multi Attribute Decision Making (MADM) algorithms (e.g. TOPSIS, GRA, etc.) [16]–[18] are used for determining the ranking of alternatives in terms of their desirability with respect to multiple criteria that can influence the decision. GRA has been successfully applied in network selection process as well. However, similarly to other MADM techniques, GRA has been also criticized for its possible *rank reversal* phenomenon, which means that the relative rankings of two decision alternatives could be reversed when a decision alternative is added or deleted. Rank reversal is a disadvantageous behavior even for my proposed content–aware multi–path (COMPAS) video streaming method. Each appearance of a new network or a link disappearance may trigger a handover process increasing delay, causing gap in the transmission or even disconnecting the terminals. In order to avoid the effects of the rank reversal, I have analyzed the GRA–based network selection algorithm to found the reason of the phenomenon. I have found that modifying the normalization method, the probability of rank reversal can be significantly decreased or even eliminated. In order to justify my theoretical assumptions, I have examined the efficiency of the proposed solutions.

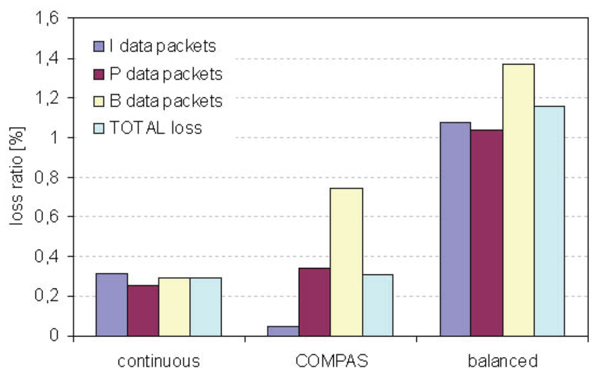
II.1. Thesis [J3, C10] *I have developed a packet distributor method (COMPAS) for multi–path MPEG video streaming. The proposed scheme decreases the loss probability of important data packets.*

Multi–path video streaming is a promising solution to overcome the strict mobile network limitations. Higher bandwidth and better quality can be provided to the users if joint links are used simultaneously. The independent but joint links can be successfully utilized by a streaming application in order to achieve high quality streaming. The available link behavior can be very different, therefore the data distribution must take the link parameters into account. I have developed a new technique, which distributes the video stream packets between the interfaces based on the packet content. The highest quality distortion is caused by I–frame losses; hence the reference frames of the MPEG video must be protected by transmitting them through the best channels. The main difference from the other multi–path streaming solutions is that in my COMPAS method a sender side buffer is used to heap up more data in order to utilize the available links more efficiently. The heap up buffer introduces extra delay, but it also increases the efficiency of packet distribution. If the heap up buffer is longer, the most reliable links can be used to the full.



11. Figure I–, P– és B–**f**rame distribution on the interfaces

Due to the proposed COMPAS interface selection algorithm the distribution of I–, P–, B–frame data bytes in the interface buffers will be different from the original video frame data distribution. The total number of packet losses was equal using the continuous packet distributor method and the selective (COMPAS) one. In spite of the similarity the number of I–frame data packet losses was significantly lower when the proposed COMPAS interface selection algorithm was used.



12. Figure Total and I–, P–, B–frame type data packet loss ratios

By changing the loss ratio probabilities of the different frame types, the COMPAS method was able to improve the received video quality. I have analyzed the proposed scheme in my own simulation framework. In the simulations the *mother_and_daughter* QCIF sequence was used as the reference video compressed with an MPEG–4 encoder. The obtained video quality improvement was about 1dB in the tests, but of course it can vary according to the link properties, heap up buffer size and the coded video sequence.

II.2. Thesis [J3] *I have proved that my proposed normalizing techniques reduce the rank reversal phenomenon to less then 10% or even eliminate it in case of GRA ordering method*

Grey Relational Analysis (GRA) [18] is a promising algorithmic approach that can realize dynamic interface ordering with multiple alternatives (interfaces) and attributes (network parameters), however, similarly to some other decision methods, GRA also suffers from rank reversal phenomenon. The GRA–based network selection method can be implemented following the steps bellow:

1. Classifying the j different network parameters (*lower–the–better*, *higher–the–better*)
2. Defining the upper (u_j) and lower (l_j) bounds of the parameters
3. Normalizing i^{th} link j^{th} parameter:

   In case of *smaller–the–better* attribute: $$s_i^*(j) = \frac{u_j - s_i(j)}{u_j - l_j} \quad (7)$$

   In case of *higher–the–better* attribute: $$s_i^*(j) = \frac{s_i(j) - l_j}{u_j - l_j} \quad (8)$$

4. Calculating the Grey Relational Coefficient (GRC) for link i:

$$GRC_i = \frac{1}{\sum_{j=1}^{k} w_j \left| s_i^*(j) - 1 \right| + 1} \quad (9)$$

5. Ranking the networks according to the GRC values

It can be realized that the GRC values will be changed only if the minimum or maximum of one of the parameters changes. This can happen if new network appears or disappears with a parameter, which is the highest or the lowest. It can be easily proved that the reason of this observable fact is the normalization method. My aim was to found other normalization techniques, which reduces the frequency of the rank reversal phenomenon, or even eliminates it. I have proposed three different methods to replace the actual normalization procedure described by equations (7) and (8).

I have proved that using predefined min–max ($E_{min\,j}$, $E_{max\,j}$) values, the normalized values will not change. By keeping the normalized values unchanged the calculated GRC will be the same even if a network is not reachable any more, therefore the rank reversal phenomenon is eliminated. The disadvantage of this solution (*norm_1*) that the difference between the GRC values can be very small.

The second modified normalization method (*norm_2*) is similar to the previous one, but it uses absolute bound only in the unwanted direction of an attribute. In case of *smaller–the–better* attribute an absolute maximum is determined, while for *larger–the–better* parameter an absolute minimum is used. The reason of this solution is that usually those networks are becoming unreachable which attributes are bad (e.g. low signal strength, delay, etc.).

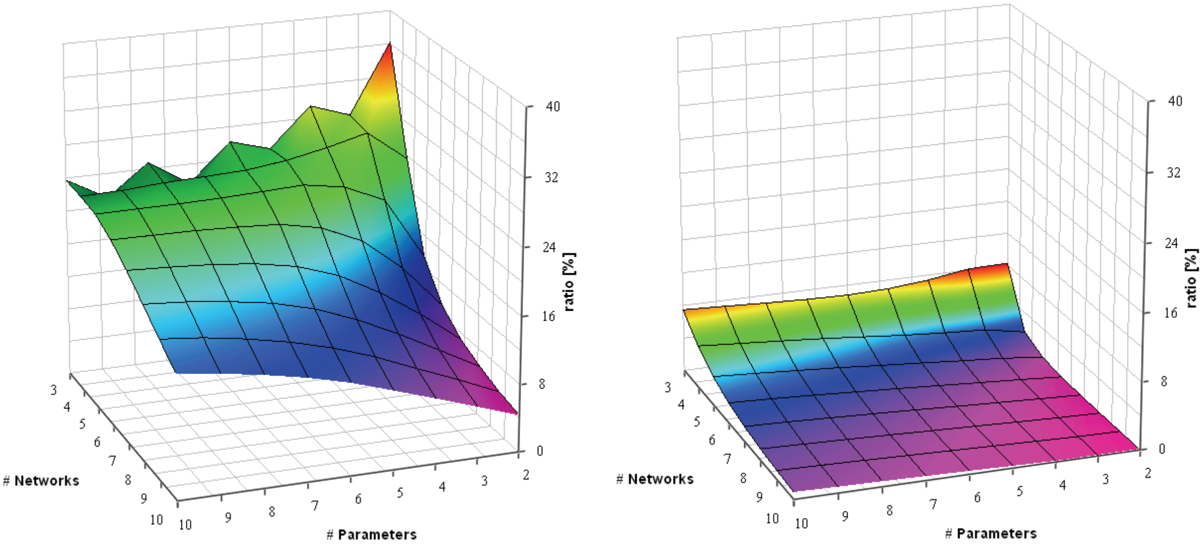
In the third normalization solution (*norm_3*) my aim was to avoid the usage of absolute min–max values. I have constructed the normalization function in such a way that the normalized value of the best parameter be equal to 1.

The normalization techniques of my three proposals are summarized in the following table.

2. Table Normalization Techniques for GRA–based Network Selection

	GRA	*norm_1*	*norm_2*	*norm_3*
smaller–the–better	$\frac{u_j - s_i(j)}{u_j - l_j}$	$\frac{E_{max\,j} - s_i(j)}{E_{max\,j} - E_{min\,j}}$	$\frac{E_{max\,j} - s_i(j)}{E_{max\,j} - l_j}$	$\frac{l_j}{s_i(j)}$
larger–the–better	$\frac{s_i(j) - l_j}{u_j - l_j}$	$\frac{s_i(j) - E_{min\,j}}{E_{max\,j} - E_{min\,j}}$	$\frac{s_i(j) - E_{min\,j}}{u_j - E_{min\,j}}$	$\frac{s_i(j)}{u_j}$

In order to analyze the efficiency of different normalization methods from the rank reversal phenomenon point of view, I have implemented a simulator tool. I have analyzed the frequency of rank order changes caused by network removals. The highest probability of rank reversal can be reached if the worst network is removed, likewise the disappearance of the worst network is the most realistic. Using normalization method with absolute min–max values (*norm_1*), the rank reversal phenomenon was eliminated. With *norm_2* and *norm_3*, the probability that the best network will change after the worst network removal is less then 10% compared the to the original GRA solution.

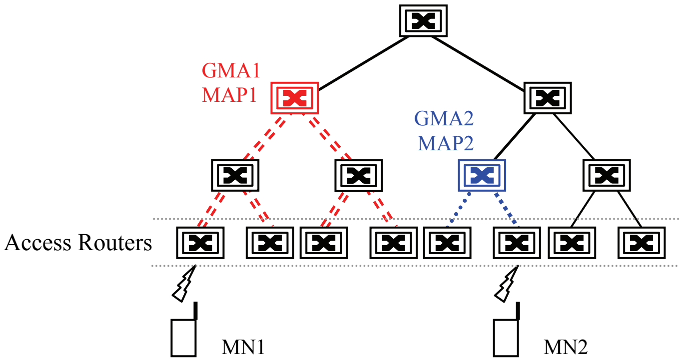


a) original GRA normalization   b) *norm_2* (*norm_3*) normalization

13. Figure Rank reversal probability in case of different normalization techniques in GRA

4.3. Mobility Agent Selection in Hierarchical Mobile Networks

In order to improve the efficiency of mobility management, new protocols (RegReg6, HMIPv6) were proposed based on Mobile IPv6 (MIPv6).These new protocols hide the mobile terminal's movement not only at home agent, but also in the visited domain by the mobility agent (GMA/MAP) implemented in a router. This router is the gateway through which traffic for the mobile node enters the local domain. The administrative load and therefore the hiding efficiency of the mobility agents highly depend on the number of served cells and its size. In a hierarchically structured domain the number of served cells by the GMA/MAP is determined by the hierarchical level of the mobility agent



14. Figure Mobility agents in the hierarchical domain

A GMA/MAP mobility agent on the top of the hierarchy can serve all cells in the domain, but in this case the hop number between the GMA/MAP and the mobile terminal, therefore the overall network load will be high. If the mobility agent is on a lower hierarchical level, the hop number of the incoming packet from the GMA/MAP to the MN (mobile node) will be low, but on the other hand the frequency of GMA/MAP changes will be high, if the mobile terminal is moving fast. To keep the number of GMA/MAP changes, as well the route length between the GMA/MAP router and the MN low, a new mobility agent selection algorithm (MASA) was proposed. Some similar solutions were presented in [12][13] after I have already published my method.

There are several potential mobility agent routers inside a RegReg6/HMIPv6 capable hierarchical domain. I introduced a new Mobility Agent Selection Algorithm (MASA) to select the most appropriate GMA/MAP for a mobile terminal with different movement behavior. The mobility agent is selected by the MN from a list of Regional CoAs attached to the Router Advertisement message, where each Regional CoA corresponds to a mobility agent. The proposed method was analyzed from the number of GMA/MAP changes and administrative load point of view.

III.1. Thesis [B1, C2] *I have developed a mobility agent (GMA/MAP) selection algorithm (MASA) for HMIPv6 and RegReg6 hierarchical networks. I have realized that if the GMA/MAP selection is based on the mobile node's handover frequency, the number of mobility agent changes can be reduced, while administrative load due to handovers is kept near constant.*

The handover frequency can be easily measured (e.g. based on past events) that can be used to select the mobility agent from the appropriate hierarchical level. The aim is to keep both the GMA/MAP change frequency and the path length between the GMA/MAP and the MN on minimum. If the agent changes are kept on a low level the necessary administrative load (Binding Update) will also remain low. To do this the proposed MASA algorithm uses the following equation:

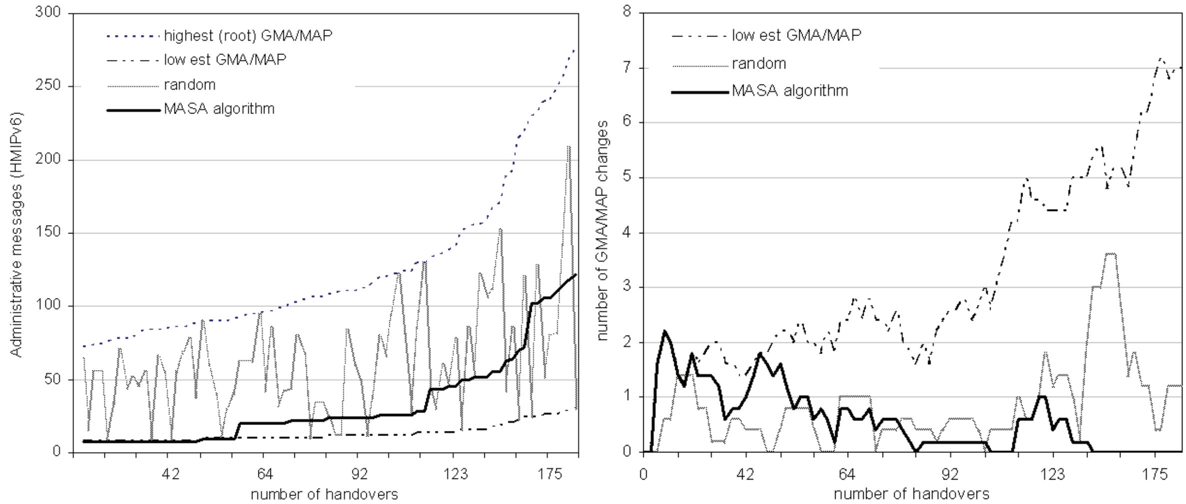
$$l = n - \left\lfloor \frac{\eta}{\eta_{max}} n \right\rfloor \tag{10}$$

Parameter l stands for the hierarchical level for the GMA/MAP in an n–depth hierarchical domain, where level l=1 is the root. The handover frequency is signed with η and η_{max} is the theoretical maximum of the handover frequency.

From the mobility agent changes point of view the HMIPv6 and the RegReg6 protocols are similar. The MASA agent selection algorithm has been examined both analytically and by simulations. The analytical examination was made for a special case, when the depth of the hierarchical domain was 3 and all the routers were considered as GMA/MAP capable router. In this case the number of agent changes can be reduced by 37% using the proposed MASA algorithm compared to random selection. Since analytical examination was too complex for large networks, I have developed a simulation tool. According to the simulation results the mobility agent changes were reduced by 54% in a 9–depth hierarchical domain and by 30% in 6–depth one.

III.2. Thesis [B1, C2] *I have shown that in case of HMIPv6 the administrative traffic load is reduced using the MASA algorithm, which selects the GMA/MAP based on the mobile terminal's handover frequency.*

To make the local roaming of the mobile terminal transparent HMIPv6 administrative messages (Regional Binding Update) are needed increasing the load on the link between the MN and the mobility agent (GMA/MAP). While the MASA algorithm tries to choose a mobile agent from the lowest hierarchical level, the hop number will be low between the MN and the GMA/MAP. If the mobile terminal is moving fast and changes its access point frequently, the path length also increases in case of MASA method. In the developed simulator tool the mobile terminals were modeled with one–dimensional random walk mobility model with different handover frequency values. The mobile terminal could choose the GMA/MAP from the lowest hierarchical level, from the highest level, randomly and according to the proposed MASA algorithm. The handover frequency determined the maximal number of handoffs during the fixed simulation time. In the simulations the fastest MN was able to change its access point 190 times. In the simulations the HMIPv6 administrative load was reduced by 12% in a 9–depth hierarchical domain and by 24% in 6–depth domain. Increasing the hierarchical levels of the domain, the efficiency of other agent selection strategies is decreasing, while the proposed MASA method adapts itself to the new environment.



15. Figure Administrative load of the HMIPv6 protocol and number of GMA/MAP changes in a 9–depth hierarchical domain

5. Application Possibilities of the New Results

Reducing end–to–end delay and loss ratio in IP–based networks is very important for all type of applications. In my dissertation I have presented new methods to improve the quality of communications. Some of the proposed solutions were investigated directly for video streaming applications, while others were studied from handover management point of view. Efficient handover management can decrease the number of necessary handover processes, resulting lower administration overload and latency, therefore improving the quality of delay–sensitive multimedia applications.

Multimedia applications can tolerate small amounts of data loss but with carefully thought out retransmissions, the loss ratio can be minimized improving the quality of the video stream. I have investigated new source controlled selective retransmission algorithms utilizing the advantages of the DCCP protocol. In the content–aware decision algorithms I have utilized that MPEG coding exploits temporal correlation between frames to achieve higher compression therefore errors in a reference frame will propagate to the dependent difference frames. By defending key frames of the video sequence I could achieve better video quality. Some of the presented retransmission scheme were also implemented and tested in real testbed.

I have shown that the temporal correlation between MPEG video frames can be also utilized, when the video is delivered over multiple links with different characteristics. In the proposed method a heap up buffer was deployed to increase the efficiency of packet (frame) distribution to the terminal's network interfaces. For the proposed multi–path video streaming scheme interface ordering is necessary. In order to determinate the rank order of the links, I have investigated a modified GRA network ordering technique if more then one network parameter is used for the ordering. My aim was to reduce and eliminate ranking inconsistency, also known as rank reversal phenomenon. I have proposed three different alternative normalization techniques that can minimize the probability of normalized value changes, causing significant reduction of rank reversal occurrence frequency.

The seamless mobility management in IP is an important task. It became obvious that Mobile IP (MIP) in itself is not capable of supporting real–time handovers, because lasting address registration processes cause intolerable interruption to user's data flow during handoff. Enhanced hierarchical solutions (e.g. HMIPv6 and RegReg6) appeared due to the demand for global QoS support. I have introduced novel mobility agent selection algorithm for HMIPv6 and RegReg6. By reducing the number of agent changes with the proposed technique, the number of address re–bindings was also decreased, gaining better QoS parameters for user applications.

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