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RIP VS. EIGRP

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Key words: Computer networks, routing, RIP, EIGRP, convergence time, simulation.

Summary. RIP and EIGRP are two basic protocols which are used in small networks. Evaluation of performances is necessary for the selection of protocol which will best suit to the specific requirements of computer network as well as for the detection of errors. This paper shows comparison of two performance indicators: time of convergence and message sending time. Time of convergence is evaluated using mathematical means, measurements and simulation method. On the other hand, message sending time is evaluated using mathematical and simulation methods.

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1 INTRODUCTION

Routing tables can consist of static and dynamic routes. Unlike the static route, which are configured by the administrator and do not have much overhead, dynamic route information are updated automatically and are suitable in situations where the network topology is constantly changing or when its unstable.

Characteristics of dynamic routing protocols depend on the algorithm applied. However, all routing protocol algorithms are responsible for the implementation of the following processes: sending and receiving routing information, computing the best paths, updating of routing tables and detecting topology changes.

Routing protocols can be classified based on the type of algorithm they use to determine the best path to the destination network: distance vector protocols, link state protocols and path vector. Distance vector protocols transmit routing information as vectors of distance and direction. Distance is transmitted using metrics, and the direction is the next router or exit interface.

Distance vector protocols are RIPv1, RIPv2, IGRP and EIGRP. In this paper we will compare performances (time of convergence and message sending time) of two distances vector protocols RIP and EIGRP.

2 RIP

RIP is an internal routing protocol, which using Bellman-Ford algorithm updates the routing tables within the same autonomous system. Since the RIP metric is represented as the hop count, it belongs to the group of distance vector protocols.

There are two versions of RIP. RIPv1 is specified in STD 34/RFC 1058, while RIPv2 is described in RFC 2453. This paper is focused on RIPv1.

2.1 RIPv1 message format

Based on the RIP message format [12], we concluded that we have to add the following to each RIP message: RIP message header (32 bits), UDP header (2*32 bits), IP header (6*32) and the data link header (56 bits if the Serial link is used or 27 bytes if the Fast Ethernet link is used). If we consider that each route needs 5 rows and the header is represented by an additional row and that the size of each row is 32 bits, we conclude that the total amount of information transferred during one update is:

$$N_k = \begin{cases} brp*160+344, \text{if routers are connected using Serial links} \\ brp*160+504, \text{if routers are connected using Fast Ethernet links} \end{cases}. \tag{1}$$

In the above formula *brp* stands for the number of transmitted routes (this number will be calculated by the simulation model).

2.2 RIP packet exchange

Until the network becomes convergent, routers repeat the same process - send their routing table to all their neighbors and process received routes. Therefore we can say that the process of convergence consists of *n* cycles. Each cycle includes the time router needs to send its routing table to its neighbors, but also the time that the neighboring routers need to process the received routes.

However, due to different distances, different interfaces delays, wide bandwidths, etc. it is clear that the time needed for exchange of routing tables and route processing may vary on each link. Therefore, we concluded that duration of each cycle is equal to longest period of time which is needed for routes to be exchanged (between two adjacent routers) and routing tables to be updated.

Based on the above, we concluded that the total time of convergence is given by the formula:

$$t_u = \sum_{i=1}^{n} (\max(t_x) + t_k)\,, \tag{2}$$

where t_x represents the time neighboring routers need to exchange updates, t_k represents commutation time (i.e. processing time) and *n* represents the number of cycles needed to be executed before the network converge (this number will be calculated by the simulation model).

3 EIGRP

EIGRP also belongs to the class of distance vector protocols. It uses a composite metric for finding the best path to a destination. Routing information is transmitted by a vector which includes delay, load, and reliability. EIGRP use Diffusing Update Algorithm (DUAL) to calculate the best path.

Unlike the RIP protocol, which is standardized by the RFC, EIGRP is Cisco proprietary protocol. Cisco has recently decided to publish EIGRP specification, but its standardization is still in progress.

3.1 EIGRP message format

There are three types of packets that can be transmitted by EIGRP messages: updates, queries and Hello packets. Since the aim of our study is to estimate the time of convergence in a situation when all routers start at the same time, there was no need to process query packets.

Beside that, EIGRP messages can transmit information about: internal IP routes, external IP routes or EIGRP parameters. Since we have considered only the EIGRP updates within one autonomous system, our study includes only internal IP routes.

Based on the EIGRP message format [12], we concluded that we have to add the following to each EIGRP message: EIGRP message header (4*32 bits), IP header (6*32) and the data link header (56 bits if the Serial link is used or 27 bytes if Fast Ethernet link is used). If the update message is transmitted, than each route needs 7 rows and the size of each row is 32 bits. Thus, the total amount of information transferred during one update is:

$$N_k = \begin{cases} brp * 224 + 376, \text{if routers are connected using Serial links} \\ brp * 224 + 536, \text{if routers are connected using } Fast\ \ Ethernet\ \ links \end{cases}. \tag{3}$$

However, if the Hello message is transmitted, than 3 rows are transmitted and the size of each row is also 32 bits. Thus, the amount of information transmitted by a Hello packet is:

$$N_H = \begin{cases} 472, \text{if routers are connected using Serial links} \\ 632, \ \ \text{if routers are connected using } Fast\ \ Ethernet\ \ links \end{cases}. \tag{4}$$

3.2 EIGRP packet exchange

In the same way as with RIP protocol, process of convergence can be divided into cycles. Duration of each cycle equals to the longest exchange of routing information between the two neighbors. Therefore, time of convergence for EIGRP protocol can be calculated using the formula (2).

However, before any exchange of routing information starts within the EIGRP routing domain, EIGRP have to find all its neighbors.

For finding neighbors and establishing neighborhood EIGRP uses Hello packets. In most networks EIGRP Hello packets are sent every 5 seconds. However, in networks whose speed is 1.544 Mbps or less, Hello packets are sent every 60 seconds. Routers which receive Hello packets within the stipulated period of time believe that their relationship with their neighbor is valid.

So, if EIGRP protocol is configured, in order to calculate time of convergence, we need to add the time needed for Hello packets to be sent (t_h) to the value of the formula (2). We will calculate t_h later in this paper.

4 TIME OF CONVERGENCE

The time routers need to exchange routing information with their neighbors, so that all routers keep the same information about the network topology, is called time of convergence. The network is operable only when the process of convergence is completed. Time of convergence is one of the most important indicators of system performance, because whenever any topological change happens, the process of convergence is re-executed (e.g. time of convergence is significantly increased if networks which contain one or more unstable interfaces). In addition, if routing protocols are not correctly configured time of convergence will increase or decrease. Therefore, it is important to calculate the time of convergence, because if we know the value of time of convergence when there are no errors in the system, we can easily detect errors.

4.1. The results of measurements on real system

In this chapter, results of measurements on Cisco Catalyst 1700 routers will be explained. All routers had three additional modules - module with two Serial interfaces and two modules with four Fast Ethernet interfaces. We had six routers available. All measurements can be found in [14].

4.1.1. Preparation for measurements

Before we began the process of measurement, we had to solve the following problems: how to synchronize routers, which debug commands should be used, and whether we have to reload all network routers before measurement.

System clock on the router displays the time elapsed since the last boot. Therefore, in order to perform synchronization of all routers, we had to reload all routers at the same time. Interpretation of results was difficult because of the fact that the system clock on the router doesn’t show milliseconds.

There is a number of debug commands which can be used in routers. We choose commands which show in detail the process of router convergence. In case of RIP protocol, debug commands used show all RIP messages routers send to each other. But, in the case of EIGRP protocol, we used debug commands which show all EIGRP messages that routers exchange with each other, along with debug commands that show activity of DUAL algorithm.

In theory, router reload is not necessary between two measurements, if after the cancellation of the first measurement, period known as holdown time expires. However, due to the fact that it is necessary to go to each router, reset protocol, and then to wait for the timers to expire, before each new measurement we reloaded all routers.

4.1.2. Analysis of measurement results

After we analyzed all the messages which routers exchanged between each other, we divided the measurements into three groups: measurements which will be rejected (group I), measurements consistent with the theory (group II) and measurements that contain a certain delay (group III).

The first group contains those measurements in which the FastEthernet0/0 interface was unstable (went up/down although we did not change anything). There are many reasons which can lead to the interface instability (damaged cables, damaged interfaces, operating system errors, etc…). Since we are not interested in this case of network convergence, all measurements which belong to this type are discarded. Despite the fact that we rejected this type of measurements in this study, we do not rule out that the time of convergence of networks which have unstable interface won't be the topic of one of our next researches.

The second group contains theoretical ideal measurements, i.e. those measurements in which the process of convergence coincides with the theoretical explanation of the process of convergence.

The third group contains all measurements which include certain delay compared to the theoretical ideal measurements.

Very often synchronization in RIP configured networks was not good enough. All measurements which had synchronization problems lasted 2 seconds longer then the theoretically ideal measurements. In networks which were configured with EIGRP protocol, two types of measurements which do not coincide with the measurements consistent with theory were observed. The first type contains measurements which had synchronization problems (same as with RIP protocol). These measurements lasted less than 5 seconds. Second type of measurements contains measurements during which routers weren't able to create neighborhood with one of the neighbors. This is possible due to the fact that routers can't simultaneously send updates through two serial interfaces. In order to create neighborhood, routers had to wait for Hello packets to be resend. Hello packets are sent every 5 seconds, so the second type of measurement always lasted 5 seconds.

During our analysis, we also noticed that some measurements lasted one second longer than the measurements consistent with theory. We ranked this type of measurements also in the first group. We did it because the time of convergence is never an integer, and if the process of convergence starts at the end of a second, then the time of convergence is extended for one more second. This logic also applies to the measurements that contain a certain delay.

4.1.3 Evaluated forms

Based on the measured results we can conclude that if the network is configured as bus topology the following stands:

- Convergence time in network configured with RIP protocol is calculated by formula (number_of_routers-1)*2. We must note that if the synchronization is not good enough, convergence time is increased by 2 seconds.
- Convergence time in networks which consist of more than 2 routers and which are configured with EIGRP protocol is 5 seconds. If the network consists of only 2 routers and is configured with EIGRP protocol, convergence time can be [1 – 5] seconds.

Also, based on the obtained results, we concluded that the time of convergence does not depend on the number of networks which are connected to routers. If we consider the speed of today's CPUs, this result was expected.

In addition, after the analysis of the update packages, we concluded that:

- The commutation time on routers which are configured with RIP protocol is two seconds and does not depend on network topology.
- The commutation time on routers which are configured with EIGRP protocol is five seconds if the network contains three or more routers, and one second if the network contains only two routers. The commutation time is independent of network topology.

4.2 Mathematical description of time of convergence

We already said that t_x represents the time neighboring routers need to exchange updates, and that t_h represents time of sending Hello packets, but we did not say how we calculate these values.

The time neighboring routers need to exchange updates (t_x) [1] [2] includes the following: interface delay (t_z), propagation time (t_p) and time needed for routing information to be sent (t_r). i.e.:

$$t_x = t_z + t_p + t_r . \tag{5}$$

After the expiry of the delay interval (d) [6], the process of route sending starts. The default delay value depends on the type of interface, but can be changed by the administrator. Interface delay is expressed in micro-seconds. Thus, the delay time is calculated by the following formula:

$$t_z = d * 10^{-6} . \tag{6}$$

To establish connection between the two neighboring routers propagation time has to end (t_p). Propagation time is calculated by the following formula:

$$t_p = \frac{L}{C} , \tag{7}$$

where L represents the distance between two routers, and C represents the sped of signal trough the wire.

Time needed for routing information to be sent (t_r) is calculated by the following formula:

$$t_r = \frac{k}{b*10^3}, \tag{8}$$

where k represents amount of transferred data, and $b*10^3$ represents bandwidth (expressed in Kbps).

Based on (1)-(8) we concluded that the total convergence time (t_u) is calculated by the following formula:

$$t_u = \sum_{i=1}^{n} (\max(d_i * 10^{-6} + \frac{N_{Ki}}{b_i * 10^3} + \frac{L_i}{C_i}) + t_{k_i}) + t_h, \tag{1}$$

where $t_h = 0$ if RIP protocol is configured, but if EIGRP protocol is configured t_h represents the time needed for Hello packets to be sent. We conceded two cases: the minimum time (t_{h_1}) and the maximum time (t_{h_2}).

It may happen that all routers send routing update information and Hello packets at the same time. In this case the time for which Hello packets are sent is equal to the longest exchange of Hello packets, since the process of convergence started. Analogously to the formula (9), it can be concluded that the minimum time needed for Hello packets to be sent is:

$$t_{h_1} = \max(d_i * 10^{-6} + \frac{N_{Hi}}{b_i * 10^3} + \frac{L_i}{C_i}) * 2 \cdot \tag{2}$$

However, it is possible that the sending of Hello packets never coincides with EGRP routing updates. In this case, we have to find the number of sent Hello packets.

If bandwidths of all links are less than 1.544Mbps, the number of Hello packets sent (r) is:

$$r = \left\lfloor \frac{t_u + t_{h_1}}{60} \right\rfloor. \tag{3}$$

However, if there is a link with a bandwidth greater than 1.544Mbps, the number of Hello packets sent (r) is:

$$r = \left\lfloor \frac{t_u + t_{h_1}}{5} \right\rfloor. \tag{4}$$

In this case, based on (10) - (12), we concluded that the maximum time needed for Hello packets to be sent is:

$$t_{h_2} = \sum_{i}^{r} \max(d_i * 10^{-6} + \frac{N_{H_i}}{b_i * 10^3} + \frac{L_i}{C_i}) + t_{h_1} \cdot \tag{5}$$

4.3. Description of the simulation

Simulation allows input of arbitrary network (i.e. it is possible to enter any network topology), selection of one of the protocols (RIP or EIGRP) and display of the total network convergence time. Simulation stores all information about the routers connections to their neighbors (the length of links, bandwidths of links, delays of output interfaces and delays trough channels). Based on this information the simulator computes metrics in the same way that the metric is calculated on a real system. It is also possible, on each simulated router, to specify a list of routes that will be distributed to their neighbors. Unlike the real systems, in which EIGRP allows entry of the wild-card mask and the area to which it belongs, the simulation does not allow these values to be changed.

The initial routing tables should contain only directly connected, static and default routes. The simulation supports first level and second levels routes, but the routing tables should not contain more than 100 of first level routes and no more than 50 of second level routes. Distribution of default static routes is possible, but the distribution of non static default routes is not possible. The autosummarisation of the EIGRP protocols is off. The simulation does not work with supernet routes.

If the network is configured using the RIP protocol, format of routing table is the same as the format of routing table on Cisco router, i.e. the initial routing tables can be obtained by copying the routing tables on real routers.

However, if the network is configured with EIGRP protocol, at the end of each row bandwidth and delay of the output interface should be added. Bandwidth should be expressed in Kbps and the delay in μs. Based on inputted values, EIGRP is configured with default metric values and all default EIGRP routes have exterior route metrics.

When the process of convergence ends, all routing tables are changed. In the case of RIP protocol routing tables are nearly identical to routing tables which can be obtained on real routers. The only difference between the routing tables on real routers and routing tables which are created by the simulator is that the simulator does not sort the routes. If the EIGRP protocol is used, routing tables (compared to the routing tables on the real routers) differing only at the end of each line, because they contain the minimum bandwidth and sum of interface delays. The simulation does not sort routes nether if EIGRP protocol is configured.

4.3.1 Time of convergence obtained by simulation

Table 1 shows the time of convergence in networks which are organized using bus topology (end routers have their own LANs), which were obtained by the simulation model. During the testing we assumed the following: the distance between two neighboring routers is 2m, the speed of signal over the cable is $3 \cdot 10^8 m/s$, all links between routers have 64 Kbps of bandwidth, all LAN links are Fast Ethernet, and commutation time depends on the configured protocol (as input values we took the values which we obtained by the measurement method).

Number of routers	RIP	EIGRP
2	2.027875	1.084125
3	4.058250	5.113500
4	6.091125	5.142875
5	8.126500	5.172250
6	10.164380	5.201625
7	12.204750	5.231000
8	14.247630	5.260375
9	16.313000	5.314250
10	18.340880	5.319125
11	20.391250	5.348500
12	22.444130	5.377875
13	24.499500	5.407250
14	26.557380	5.436625
15	28.617750	5.466000
16	30.680630	5.495375

Table 1. The time of convergence in networks organized as bus topology obtained by the simulation

The results showed in Table 1 are also shown in Figure 1. Referred to Figure 1, we can conclude that in contrast to the time of convergence of RIP protocol, which is an increasing function, the time of convergence of EIGRP protocol is almost a constant. This stands if the networks are organized using bus topology.

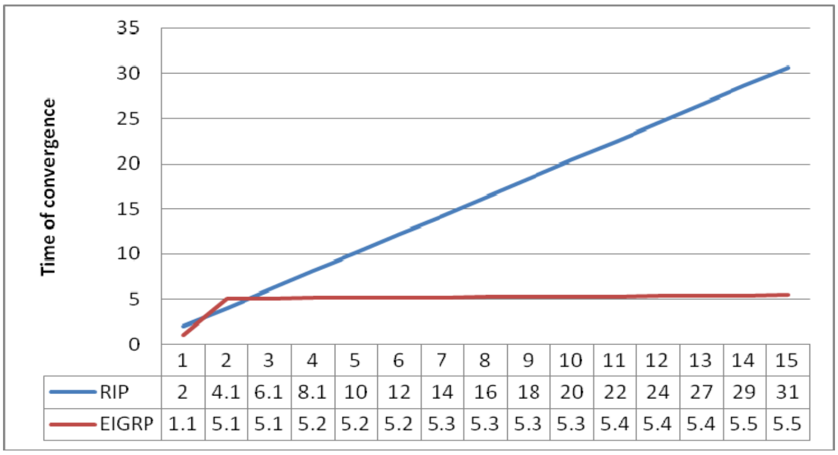

Figure 1: Comparison of time of convergence if networks are organized using bus topology

As you can notice, the simulated results are consistent with the results of measurements. In addition, the simulation shows the time of convergence with greater accuracy than the measurement results does, because the Cisco Catalyst router system clock shows the time in the form of HH:MM:SS. In addition, the simulation does not depend on the number of routers which are available in the moment nor the manufacturer of network equipment, because the simulation allows to the user to enter the commutation time which user obtained by measuring.

However, if instead of bus topology, routers are configured using a hierarchical structure (star topology), the difference between the time of convergence of RIP and EIGRP protocols is significantly lower (Table 2). In practice, hierarchical topology is used more often than the bus topology. The reasons for this are numerous: better scalability, the ability to create redundant links, better network performances, better network security, easier management, and so on.

The results shown in Table 2 assume that the tested networks are organized in the form of a complete full tree, and that each leaf is connected to a LAN. All graph nodes have the same degree.

Number of routers	Degree of node	RIP	EIGRP
3	2	4.05825	5.11350
7	2	8.15150	5.18625
15	2	12.32225	5.28700
4	3	4.06575	5.12050
13	3	8.21150	5.23175

Table 2. The results of the tree (hierarchical) topology testing

All parameters used in networks configuration shown in Table 1. are same as the parameters which we used during testing the networks which are shown in Table 2. So, the only difference is the network topology.

5 MESSAGE SENDING TIME

After the network converged, i.e. became operational, users can exchange messages. Time needed for messages to be send inside the network is a very important performance indicator. Same as with the time of convergence, only if we know how long is needed for messages to be sent inside the network when there are no errors in the system, we will be able to detect errors when they occur, and thus to fix them.

Today, there are many attacks which can violate network performances more or less. Based on the message sending time, it is possible to detect some of them, such as: DOS attacks, attacks which intercept messages, redirection attacks and so on.

Also, based on the message sending time we can detect congestions inside the network. If we notice that the congestions are constant, to improve the network performance, we have to add additional network devices, or in worst-case scenario we have to change the network topology.

As we have already said, except estimation of the time of convergence, we will also estimate message sending time (from source to destination) in networks which converged using RIP or EIGRP routing protocol. Before we show concrete results, we will briefly describe the basic features of a part of model which is used for sending messages.

5.1 Description of the simulation

Simulation, presented in this paper, assumes the following:

- Packages are (between routers) exchanged using the ``store and forward`` method with commutation of packages
- Probability of error is negligible
- Data flow control through the network is done in accordance with the Idle-RQ protocol
- Verification is done hop by hop
- Load balancing is off.

Simulation can work with any file size, as well as with any packet size. It is possible to enter any route as the source of the file. Simulator will calculate the time needed for file to be sent and it will display the number of the destination router. If the file cannot be sent, simulator will display the appropriate message, as well as the time which elapsed before the message was rejected.

Simulation requires that a packet size is inputted, because when a packet is sent from source to destination the sender and receiver must agree on the maximum packet size which can be transferred between them. Since there is no standard which determines what should be the packet size, it has to be entered manually. File size can be expressed in KB, bytes and bits, and packet size can be expressed in bytes and bits. Simulation assumes that the packet size is less than the file size.

5.3 Mathematical description of message sending time

As we have already said, the messages are transferred (from the source to the destination) using “store and forward” method with commutation of packages. Therefore, the number of packets (b_p) is calculated by the following formula:

$$b_p = \left\lceil \frac{vf}{vp} \right\rceil, \tag{14}$$

where v_f represents the file size and v_p represents the packet size.

For the first packet to be transmitted from the source to the destination, the propagation time must elapse. In this case, the propagation time is calculated using the formula (7).

Except the propagation time, it is required that the time needed for sending useful information is calculated, as well as the time needed for acknowledgements to be transferred. If we consider that each packet, except the first, must wait until the previous packet is acknowledged (i.e. $t_{p_i} + t_{ACK_i}$), time needed for messages and their acknowledgements to be sent between the source and the destination can be calculated using the following formula:

$$t_r = \sum_{i=1}^{b_p} t_{r_i} + b_p * \max(2 * t_{p_i} + t_{ACK_i}) - t_{p_1} . \tag{15}$$

When we calculated time needed for useful information and their acknowledgements to be sent we calculated $\max(2 * t_{p_i} + t_{ACK_i})$, because time needed for information between neighboring nodes to be sent can vary, and therefore we need to find the bottleneck. Since we already calculated time needed for establishing a connection between the source and the first router (when we calculated the propagation time), we subtracted t_{p_1} from the total time needed for useful information and their acknowledgements to be sent.

Each packet which has to be sent is packed into the TCP/IP format (each TCP/IP packet contains a header of 20 bytes). Therefore, time needed for data to be sent between adjacent nodes (t_{r_i}) is calculated using the following formula:

$$t_{r_i} = \frac{vp + 20 * 8}{b * 10^3}. \tag{16}$$

Analogously, it can be concluded that time needed for acknowledgements (ACK) to be sent between adjacent routers is:

$$t_{ACK_i} = \frac{N_{ACK}}{b * 10^3}, \tag{17}$$

where N_{ACK} represents ACK packet size.

Of course, we must not forget the time of needed for the last ACK packet to be sent:

$$t_{ACK} = \sum_{i=1}^{k} (t_k + t_{p_i} + t_{ACK_i}), \tag{18}$$

and the commutation time:

$$t_z = \sum_{i=1}^{k} (t_k + t_{r_i}). \tag{19}$$

Based on the above, we concluded that the total time needed for file to be sent (using "store and forward" method with commutation of packages) from the source to the destination can be calculated according to the following formula:

$$t = \sum_{i=1}^{k+1} \frac{L_i}{C_i} + \sum_{i=1}^{b_p} t_{r_i} + b_p * \max(2 * t_{p_i} + t_{ACK_i}) - t_{p_1} + \sum_{i=1}^{k} (t_k + t_{p_i} + t_{ACK_i}) + \sum_{i=1}^{k} (t_k + t_{r_i}). \tag{20}$$

5.4 Message sending time obtained by the simulation

If network is configured as bus, point-to-point, ring or hierarchy (star) topology message sending time will be same in networks configured with RIP protocol, as within networks configured with EIGRP protocol. This is true because in all these situations, there is only one path from the source to the destination, so both protocols will choose to send messages using that path.

Also, in networks configured as bus, point-to-point, ring or hierarchy (star) topology, message sending time will not depend on the number of packages sent form the source to the destination, nor on the bandwidth of links (for the same reason).

Message sending time will differ for RIP and EIGRP configured networks only if the following stands:

- There are redundant links
- RIP protocol calculates path (from the source to the destination) which is shorter, but minimum bandwidth on that path is less than minimum bandwidth on the path which EIGRP protocol chose.

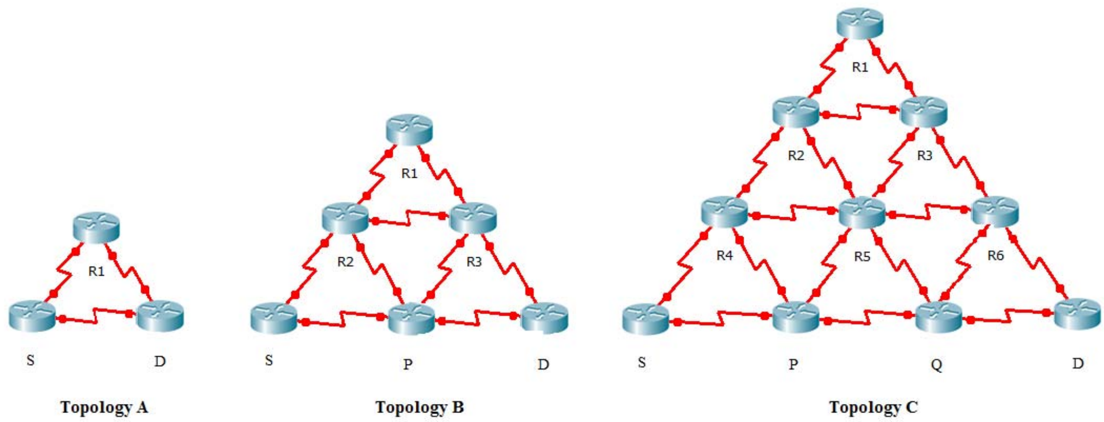

Figure 2. Tree topologies used for testing message sending time

There are a number of networks which satisfy listed conditions, but we will show only message sending times measured on topologies which are shown in Figure 2.

In order to create networks in which RIP and EIGRP protocols will chose different paths, we assumed that the following paths are low bandwidth paths:

- S-D in Topology A
- S-P-D in Topology B
- S-P-Q-D in Topology C,

and that all other links have the same bandwidth. Table 3 shows paths through which RIP configured and EIGRP configured networks will send all their packets form the source (S) to the destination (D).

	RIP	EIGRP
Topology A	S-D	S-R1-D
Topology B	S-P-D	S-R2-R1-R3-D
Topology C	S-P-Q-D	S-R4-R2-R1-R3-R6-D

Table 3. Paths through which packets will be send in RIP/EIGRP configured networks

We wanted to know how message sending time depends on two parameters: number of packets which are sent from the source to the destination and bandwidth of links.

Table 4 show results (obtained by the simulator) in case when there is a big difference between bandwidths of paths which RIP and EIGRP chose. Namely, we assumed that bandwidths of slow links are 56Kbps (dial-up) and bandwidths of all other links are 10Mbps (Ethernet).

Table 5 shows results (obtained by the simulator) in case when there is a small difference between bandwidths of paths which RIP and EIGRP chose. Namely, we assumed that bandwidths of slow links are 10Mbps (Ethernet) and bandwidths of all other links are 100Mbps (Fast Ethernet).

Results in both tables depend on the number of packages which have to be sent. We considered two situations: when 100 packages are sent between the source and the destination and when 8388608 packages are sent between the source and the destination.

	RIP		EIGRP	
v_f / v_p	100MB/1MB	1MB/1bit	100KB/1MB	1MB/1bit
Topology A	14979.95714	25315.6765	82.74285772	138.5027406
Topology B	15129.75871	25315.68151	84.38529056	138.5067736
Topology C	15279.56029	25315.68653	86.0277234	138.5108066

Table 4. Message sending times when bandwidth of RIP path is 56Kbps and bandwidth of EIGRP path is 10Mbps

	RIP		EIGRP	
v_f / v_p	100MB/1MB	1MB/1bit	100MB/1KB	1MB/1bit
Topology A	81.9216413	138.5007241	8.27608639	13.90240572
Topology B	82.74285772	138.5027406	8.443929698	13.90640905
Topology C	83.56407414	138.5047571	8.611773006	13.91041238

Table 5. Message sending times when bandwidth of RIP path is 10Mbps and bandwidth of EIGRP path is 100Mbps.

During the testing we assumed the following: the distance between two neighboring routers is 2m, the speed of signal over the cable is $3 \cdot 10^8 m/s$, the commutation time is 1ms and all packets have 8 bit acknowledgement. File sizes and packet sizes are already shown in tables.

Based on the results which are shown in Table 4 and Table 5, we can conclude that message sending time is in these situations fewer for EIGRP protocol. This result was expected (although EIGRP sends packages trough longer path, because of greater bandwidth trough which information is sent, message sending time is much fewer for the EIGRP protocol).

6 CONCLUSIONS

This paper presents a simulation model which not only imitates the routing process on real routers, it estimates the time of convergence and time which is needed for messages to be sent from the source to the destination. The flexibility of the model is reflected in the fact that it can work with an arbitrary network topology, but also in the fact that it can be easily upgraded.

An important feature of the model presented in this paper is that it is calibrated. Routing tables, obtained at the end of the routing process, are identical to the routing tables on the real systems, because the simulation model truly reflects the process of sending messages between neighboring routers, as well as the processing of those routes. However, in order to correctly estimate the convergence time, we had to estimate the time of commutation. The only way to obtain these values was to measure the time of convergence of real routers and to use gained results for calibration of our simulation model.

If we are considering the time of convergence, the results obtained using the simulation model showed that the EIGRP is prefered protocol in networks which have more than three routers, because with the increase of the number of routers through which the information has to be shared, the difference between the measured times of convergence also increases. If we take into account the results which we obtained using measurements, these results were expected. Of course, the difference between the time of convergence in RIP and EIGRP configured networks is significantly lower if instead of bus topology, the network is created as hierarchy topology.

Also, in this paper we presented the relation between the message sending time from the source to the destination and bandwidth of links, but also the relation with the number of packages which have to be sent. We concluded that in most cases (when network is configured as bus, point-to-point, ring or hierarchy topology) message sending time is equal for the RIP and for the EIGRP protocols and does not depend on the bandwidth of links or on the number of sent packages. But in cases when because of the greater bandwidth of links, EIGRP chooses faster but longer path, message sending time trough those paths is much fewer then trough paths which RIP chose.

Based on all results which we obtained in this study, it can be concluded that in most cases EIGRP is preferable protocol, especially when great amount of data has to be sent trough the network.

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