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A priority-based queuing model approach using destination parameters for real-time applications on IPv6 networks

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Abstract: In the early days of the Internet architecture, the most important aim is to transmit data over packet switched networks. The traditional Internet architecture used in these networks lacks quality of service. However, today, as real-time applications increase, it is needed. There are approaches to improving the quality of service using the flow label field in the Internet Protocol version 6 header. In this study, a novel algorithm that uses destination network parameters to reduce queuing and end-to-end delay is created. A round-robin-based time-aware priority queue new model is used within this algorithm. Data packets using this proposed queue are prioritized with metric values of the destination network. In order to provide end-to-end service quality, the prioritization value is used by placing it in the flow label field. For this purpose, a new approach to the use of this field is proposed. Delay, one of the most important factors affecting quality of service, is reduced with the proposed algorithm and flow label usage approach. As a result, the reduction in delay times between 22 and 39 ms resulted in various improvement rates between 16.79% and 35.13%.

Key words: Quality of service, end-to-end delay, flow label, priority queue, IPv6, voice over IP, video over IP

1. Introduction

In the first period of the use of the Internet, the data traffic was quite low compared to today. In this period, the most prominent approach in Internet architecture design is the transmission of data. Internet architecture is created with a network service called “best effort” and the logic of ”first come first served” on the packet switched networks. Therefore, there is no prioritization to improve service quality within the traditional Internet architecture. However, the parameters affecting this quality such as delay, latency, bandwidth, and jitter have no priority in the design of these networks [1]. Time-sensitive data packets must reach their destination within a certain time interval. The prioritization is that these packets are given precedence over other packets (nontime sensitive). It is important to increase the quality of service (QoS) that users and services need.

Nowadays, the development of Internet technology and the demand of end users to use value-added services over the Internet increase. Therefore, real-time applications (RTAs) such as voice over Internet Protocol (VoIP), Internet Protocol television (IPTV), and video conferencing are introduced [2]. The widespread use of these applications has led to time parameters such as delay and jitter which are not inherent in Internet architecture. However, it is found that delay parameter gives results for limit values or above these values

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in various networks [3]. It should be noted that these are delay values that are proposed by International Telecommunication Union (ITU) G.114 document. In order to eliminate such negative effects, the concept of QoS, which ensures the correct use of time parameters, has come to the fore. Thus, QoS offers intelligent mechanisms to prevent delay [4].

There are several technologies for providing QoS in RTAs. Various technologies, such as frame relay which can offer congestion control [5], asynchronous transfer mode (ATM) that can tender six different service classes [6], and multiprotocol label switching (MPLS) which can present labeling incoming packets [7], are integrated into network systems. Moreover, time-sensitive networking (TSN) used in industrial communications such as health and automotive together, support ultralow latency (ULL) [8]. However, since these technologies are defined in the data-link layer, they provide the necessary QoS only within their backbone. Therefore, these technologies do not have end-to-end support. Deterministic networking (DetNet) operates at the network layer but requires a single administrative control for networks. Therefore, it is not for large groups of domains such as the Internet [8]. Unlike these technologies, there are various architectures that provide QoS support. One of these architectures, integrated services (IntServ), reserves the resources required by Resource Reservation Protocol (RSVP) before the data transfer starts [9]. Differentiated services (DiffServ), another architecture, exhibits different behaviors to each class by creating various traffic classes [10].

In cases where the above technologies for RTAs are insufficient, Internet Protocol version 6 (IPv6) technology offers a new approach to solving the problem with its existing capabilities. IPv6 [11] is brought many innovations, despite it appears to be a technology that eliminates the inadequacy of IPv4 address space [12]. One of these innovations is the identification of the flow label (FL) field that provides QoS support by distinguishing data streams as shown in Figure 1. In addition, the type of service (ToS) field used for QoS support on IPv4 is replaced by traffic class (TC) on IPv6.

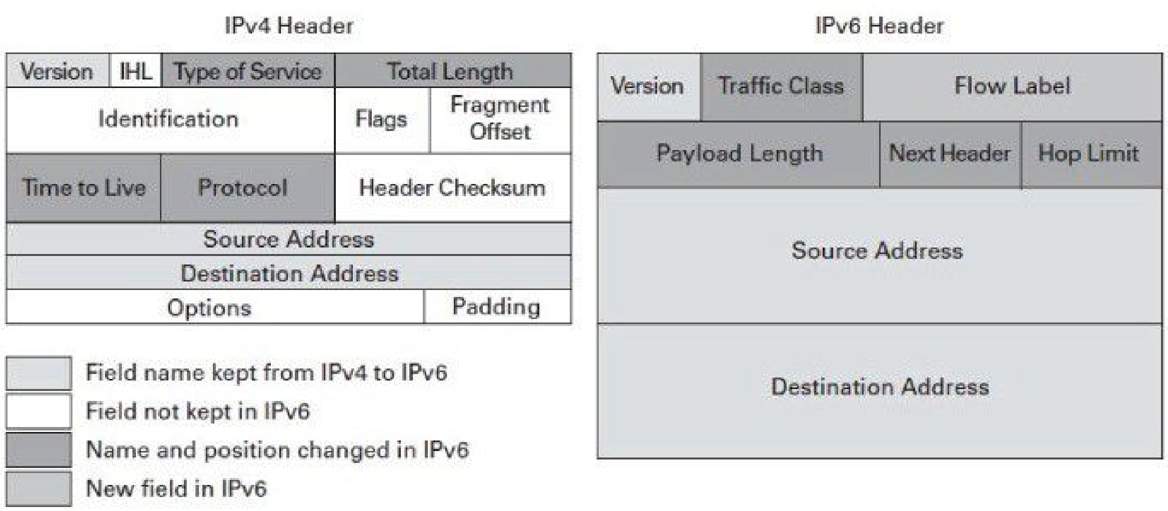


Figure 1. Comparative representation of IPv4 and IPv6 headers.

Traffic generated using the FL field provides advantages to routers that can recognize packets and behave special to them. Thanks to these advantages, IPv6 provides end-to-end support for QoS [13]. For the FL field, which is defined as a total of 20 bits, it is assumed that there is no labeling if all its bits are zero. However, various approaches are proposed in the literature, although how this field is used is not fully defined [14–17].

The network technologies used today detect the packets for RTAs by means of the Differentiated Services Code Point (DSCP) bits within the ToS and TC fields. Therefore, the QoS support needed by the service can be provided. However, these data packets are evaluated in the same behavior model regardless of the destination. In this case, there is no prioritization, including the destination parameters.

As it is known, fixed delays cannot be reduced without structural changes and their effect on total delay cannot be decreased [18]. In this study, a Round-Robin based Time-aware Priority Queue (TaPQ) model is proposed to reduce queuing delay, which is one of the variable delays. Data packets using the proposed queue are prioritized with the destination network parameters. This prioritization is carried out in two steps. In the first step, the destination networks of the voice/video packets are identified, and the metric value calculated by the routing protocol for this network is taken from the routing table. In the second step, these metric values are set in the FL field and is used as a criterion of prioritization. Therefore, a new approach for the FL field is proposed. The delay which is one of the important QoS parameters is reduced with the proposed model. For this purpose, packets going to the network with high metric values are prioritized over packets going to the network with low metric values. Thus, packets belonging to RTAs are not evaluated within the same behavioral model, and a prioritization structure is used in it-self.

This paper is organized as follows: Section 2 gives the related studies. Section 3 mentions the realization steps of the proposed model and the method to be applied. Section 4 presents the simulation environment and the results. Section 5 provides the evaluation of the results and future work.

2. Related works

One of the most important factors affecting QoS is delay. Improving the QoS by reducing this factor is one of the topics of interest for researchers [19–22]. Voice codecs used for VOIP applications also influence this factor [23, 24]. Since the codec delay is a fixed delay, the effect of this delay on the total delay cannot be reduced without structural change. This increased interest in the studies on queuing delay, which is one of the variable delays. Accordingly, different queue techniques for various RTAs are discussed [25, 26]. One of these techniques, PQ queuing, is found to give the best results for QoS parameters [27, 28].

Thanks to the FL field in the IPv6 header, the data stream can be classified by labeled. This case reveals the effect of IPv6 on QoS [29, 30]. Several approaches are proposed in the literature [14–17], despite the use of the FL field is not fully defined. Moreover, recommendations for these approaches are confirmed by some experimental studies [31]. FL, which can increase QoS, is used with different architectures such as DiffServ [13, 32].

It is previously mentioned that the PQ is a method that provides the best results for the delay. In this paper, the use of a new FL with a PQ-based TaPQ is proposed. In accordance with Diffserv architecture, unlike the literature, as shown in Figure 2, the packet classifier decides based on the information from the routing table.

3. Proposed model and method

This study is carried out on IPv6 networks. The FL field in IPv6 is used for end-to-end QoS support. The prioritization parameter is the destination network metric. These metric values are maintained by routing protocols on routers. The host running the RTAs uses the TOS or TC field in the packet processing process to indicate that these packets are a voice or video packet. As these packets move through the network, the

IP header is read by the first edge router to determine the destination network. Then, the metric value of the relevant network is taken from the routing table and set in the FL field. Along the way, other routers evaluate the FL field of the packets and place them in the corresponding priority queue and transmit accordingly.

The DiffServ model is widely used in FL applications [13, 32]. In this study, a different behavior model is not created for each metric value to be used in the prioritization process. Each voice/video packet is placed in one of the queues formed as TaPQ. Thus, instead of keeping different state information for each metric value on the routers, a limited number of state information is kept, and operation delay is minimized. The algorithm of the proposed prioritization model is simulated using the OPNET program [33]. The block diagram of the prioritization algorithm is given in Figure 2 and the flow diagram of the algorithm is presented in Figure 3.

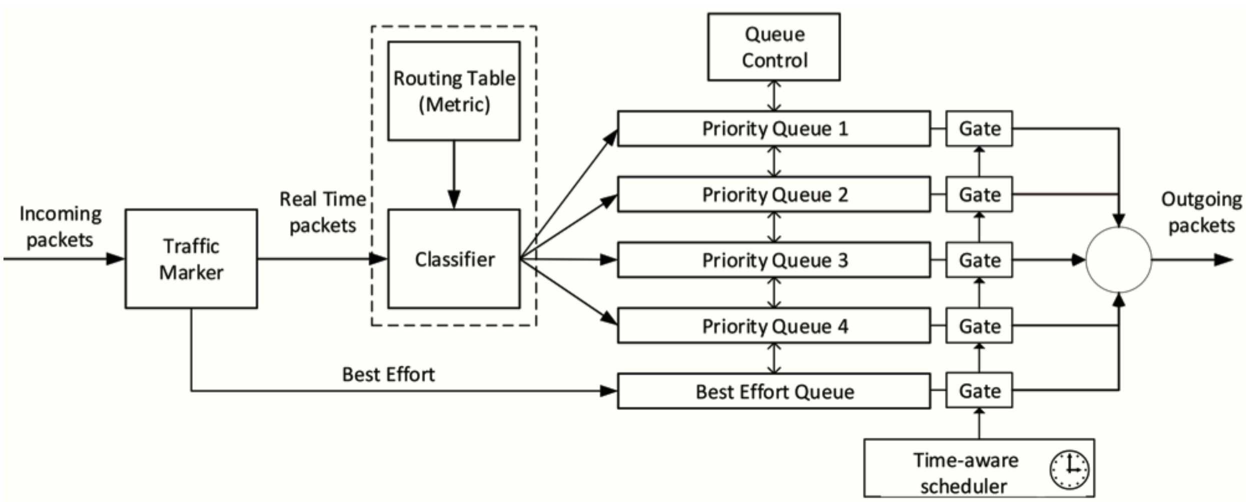


Figure 2. Block diagram of the prioritization algorithm.

3.1. Distinguishing of real-time packets

A host running RTA sends a packet produced to the router. In this case, the first step is to distinguish voice/video packets from the others to be prioritized. This step is shown with “A” in Figure 3. On the OPNET program, the ToS value of the voice packets is 6 (interactive voice), while the value of the video packets is 5 (interactive multimedia). Therefore, if the IP precedence bits in the TC field in IPv6 have a value of 5 or 6, these are the packets of the RTAs. Therefore, the system distinguishes RTA data packets, such as voice/video, and decides whether the incoming packet will be processed by the algorithm. If the incoming packet does not belong to the RTAs, it is sent directly to the “best-effort” queue.

3.2. Keeping the real-time packets under control

It is important to know the number of RTA data packets on the system in order to correctly plan queues and prevent queue overflow. This step is given with “B” in Figure 3. In the developed algorithm, a global variable counting each voice/video package is defined. This variable increases for each incoming voice/video packet and decreases for each outgoing. Thus, the number of RTA data packets on the system is controlled instantly.

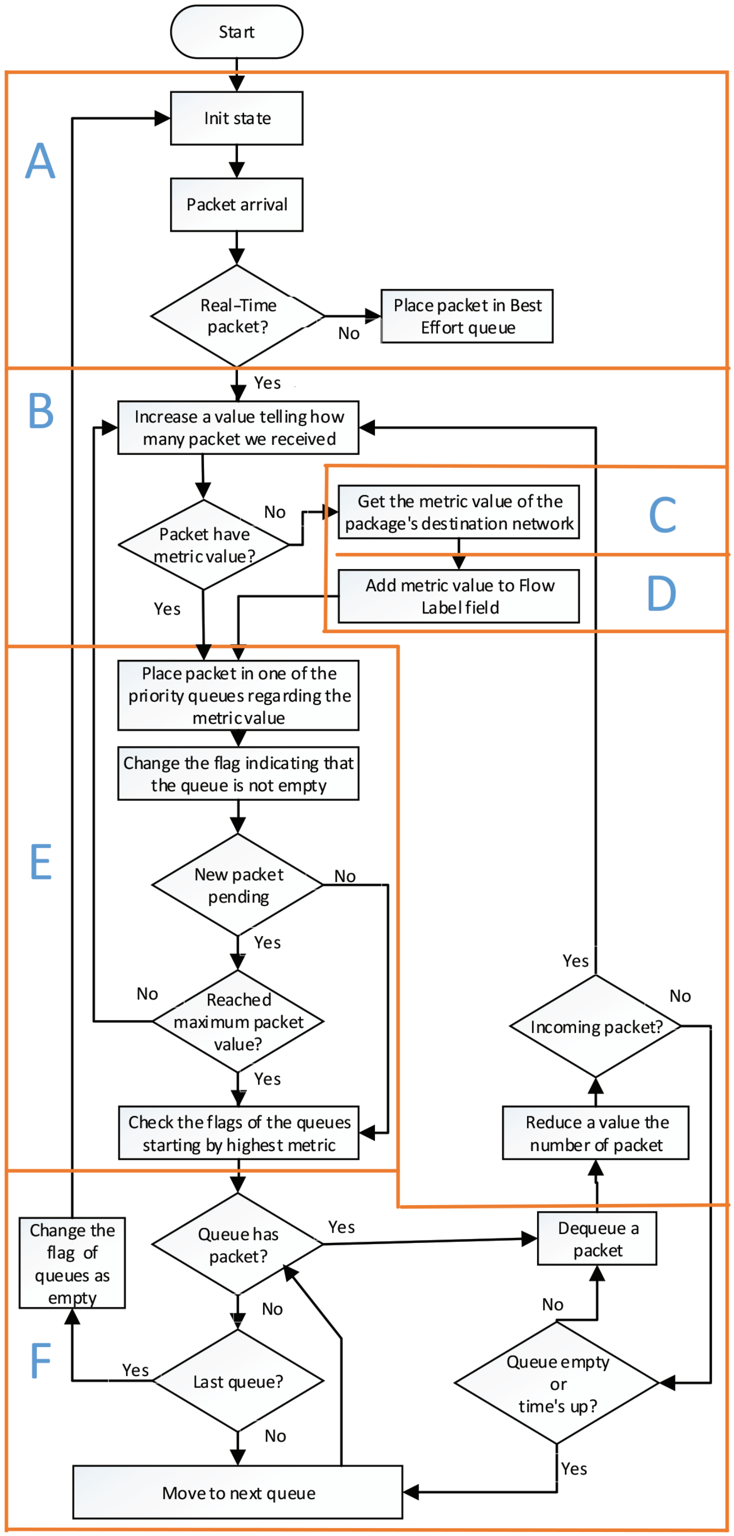


Figure 3. Flow diagram of the prioritization algorithm to be applied.

3.3. The obtainment of the metric values

The priority value is the metric value of the routing protocol. Therefore, the destination network of each RTA packet is determined, and the metric of that network is derived from the routing table. This step is shown in Figure 3 with "C". In this process, one of the most important points is access to the routing table and obtaining the metric value from this table. OPNET is a process-based simulator program. These processes communicate with each other through Interface Control Information (ICI) messages. The access to the messages that are from encapsulation process to the routing table process should be gained. The ICI message packet structure created for this access is given in Figure 4. The definitions of this package structure are given below.

- Message type (1 bit): 0 is defined for request and 1 is defined for reply.
- Address (128 bits): The IPv6 address of the network of which metric value is required.
- Metric (12 bits): Default value was defined as 0.

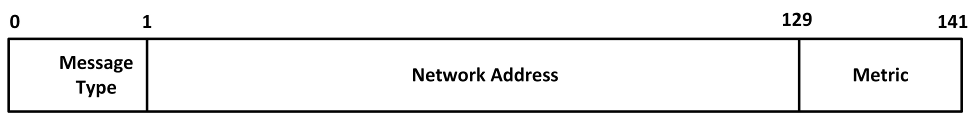


Figure 4. Packet structure of the ICI message.

When the system receives a voice/video packet, it sends an ICI message to the process to determine the metric value of the destination address and generates an interrupt to return the desired metric. Then, ip-process obtains the metric value of the destination address within the received ICI message from the routing table and generate a reply message containing the desired metric value. The metric value required for the system is derived from this reply message.

3.4. Addition of the priority value into FL field according to the metric value

The metric value calculated by the routing protocol is added to the FL field. Thus, the router receiving the data packet decides whether this data packet has priority. If the data packet has priority, it also determines how much priority it has. This step is indicated by "D" in Figure 3. The prioritization structure of the RTA packet that is created based on the destination metric value is applied to the FL field. The FL field is 20 bits long and the approach of using this field is given in Figure 5.

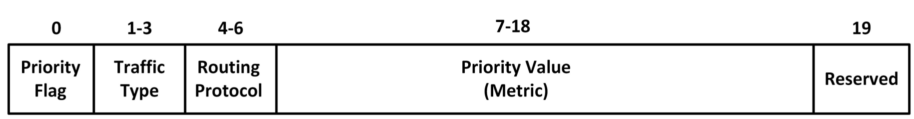


Figure 5. Approach to use the proposed FL field.

Per hop behavior identification code (DiffServ PHB-ID) [15] is quite a good approach in terms of performance compared to others [31]. The proposed FL structure in this paper is completely different in terms of usage of the field in this structure although it is comparable to the DiffServ PHB-ID approach. This proposed structure also conforms to the DiffServ architecture [13, 32]. In the use approach shown in Figure 5, the meanings of the bits given are presented.

- 0 (1 bit): This is the identifier of the prioritization algorithm. If 0, no algorithm is applied, and 1 is applied.
- 1-3 (3 bits): Used to mark voice/video packets. In total, 8 different types of traffic can be marked.
- 4-6 (3 bits): Routing protocol definition. A total of 8 different protocols can be defined.
- 7-18 (12 bits): The metric value of the destination network of the RTA packet is also defined as the prioritization value.
- 19 (1 bit): Reserved for future needs.

In the FL approach proposed, 12 bits are reserved as the metric value. Therefore, the proposed structure supports RIP and RIPng protocols directly. However, if the metric values of other routing protocols where the 12-bit field is insufficient, a conversion operation is required.

At this stage, if the process receives a packet stream, it checks the TC field to determine whether prioritization is required. If prioritization is required, it adds the metric value of the destination network from the ICI message into the FL field. In this case, the step of marking the RTA packets according to the metric value ends.

Adding the prioritization value to the RTA packets is done on the edge router where the packet first came by. Therefore, when packets are received by other routers, they have a prioritization value. These routers only read this value to perform the queuing process. When the packet arrives at other routers, it has a prioritization value. Therefore, steps “C” and “D” given in Figure 3 are omitted.

3.5. Implementation of the prioritization algorithm and queuing model on routers

Once the RTA data packets are marked with a priority value, they must be placed and forwarded to the respective queue. In this step, priority queue design and implementation phase are started. The queue structure designed, named TaPQ, developed from PQ, which gives the best results in RTA [27, 28]. A round-robin-based time-aware scheduler (TaS) is added to the PQ queue structure, like to 802.1Qbv. TaS defines a forwarding time for each queue for a certain period, and when this period expires, it assigns the forwarding right to the other queue. This structure prevents a queue being from forwarding continuously and blocking other queues. These processes are given in the field indicated by “E” and “F” in Figure 3.

The processes of planning techniques operate as a child process of the ip-process. These processes are responsible for placing and transmitting various data packets flowing on the router into different queues.

In order to implement our model, the prioritization structure is applied by defining the child of the algorithm on the process model. The state transition diagram of the new model consists of four states as idle, init, enqueue, and dequeue, as shown in Figure 6. The idle state is where the system operates normally without using the prioritization algorithm. The init state is the waiting state for packet arrival, and if the RTA packet arrives during the waiting period, the enqueue state is passed. The enqueue state is the field indicated by “E” in Figure 3. In the process of applying the algorithm and queue model, the voice/video package is placed in the corresponding queue. The functions of this process are given below.

1. When a packet is received, the value of the variable that holds the number of packets increases to control the number of packets in the queue.

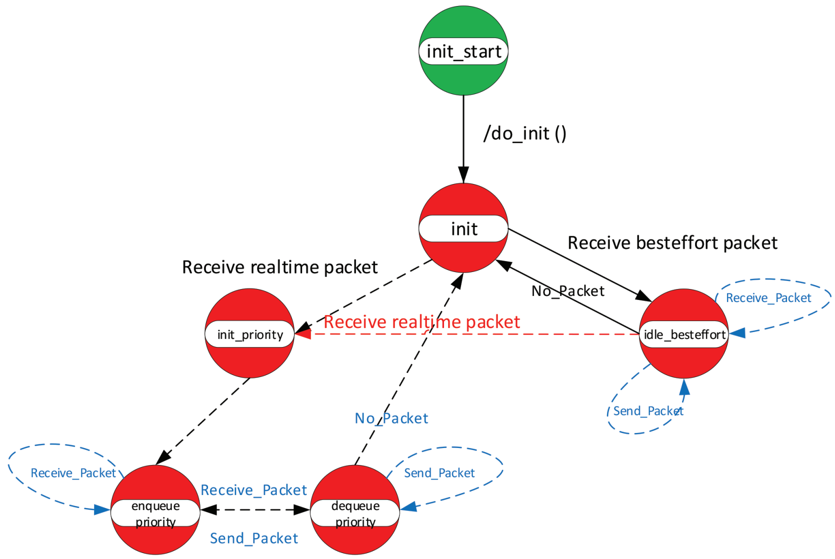


Figure 6. State transition diagram of the algorithm.

2. By checking the FL field in the incoming packet, the priority value is determined and placed in the corresponding prioritization queue. The lower metric value for routing protocols refers to the better path. Packets with a metric value of 12–15 are the group with the largest metric value and are placed in the highest priority queue. Similarly, packets having a metric value of 0–3 have the smallest metric and are placed in the lowest priority queue. In the simulation study with 8 and 16 queues, high levels of increase in jitter values are obtained due to algorithm complexity and processing delay. Since the optimum solution is obtained with 4 queues, the system has four queues.

3. If a packet is placed in a queue and the queue is empty until then, the status of the flag of that queue changes. The flag indicates whether there is one or more packages in the queue. If there is a packet in the queue, this queue is evaluated during the packet sending process.

4. In this case, it is checked whether a new package is arrived. However, if the packet arrives, it is determined whether the number of packets in the queue is maximum. If these packages are the maximum number, go to step 5. Otherwise, return to step 1.

5. Switch to the dequeue state to send packets waiting in the queue.

Dequeue State is the field indicated by “F’ in Figure 3. This is the step where the packets waiting in the queue are sent. Even if the packets continue to arrive, this is the case if the queues are full or there is no incoming packet. The functions of this case are given below.

1. Check all queues starting from the queue with the highest priority value. Mark the flag where all queues are empty if the last queue you have checked is empty.

2. If the flag of a queue is marked so that there is a packet in this queue, send a packet from this queue.

3. If there is an incoming packet, switch to the enqueue state, if there is not, continue sending the packet.

4. If the queue expires while sending the packet, the other queue is passed.

5. If there are no more packets to send in this queue, mark the flag of the queue as empty.

6. If there is no new incoming packet or all queues are empty, switch to the init state, otherwise, return to step 1.

This algorithm, which is defined on the routers, briefly determines whether the incoming packets are voice or video packets. Besides, if the packet belongs to the RTAs, it determines a prioritization value based on the destination network metric. It allows this value to be set into the FL field. According to the prioritization value within the FL field, it performs the process of placing and processing this packet in the relevant queue.

4. Simulation environment and results

Two reference topologies are generated to evaluate the proposed prioritization algorithm. In one of these topologies, only the RIPng protocol works and consists of a single autonomous system. The other reference topology consists of four autonomous systems in which the RINng and BGP protocols work together. The results obtained are compared before and after the application of the algorithm on this reference topologies.

4.1. Creation of reference topology with one autonomous system

The first reference topology formed consists of an autonomous system, as shown in Figure 7, and there are 8 routers within this autonomous system. To generate packet traffic for non-RTAs, HTTP and FTP servers are added to the voice/video traffic. In this way, packet traffic of RTAs and non-RTAs can be modeled in the environment.

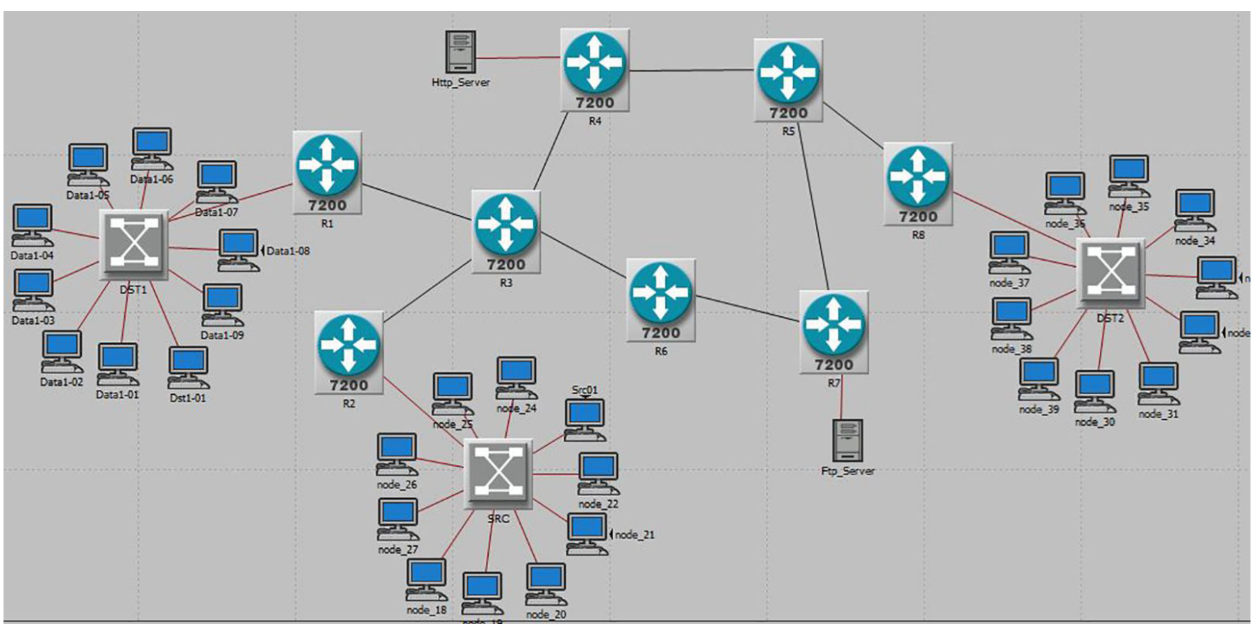


Figure 7. Reference topology consisting of one autonomous system [34].

Thirty users are added at three different locations on the generated topology as shown in Figure 7. Ten of these users are designated as source (SRC) and other 20 are designated as destinations (DST1 and DST2).

The IPv6 address is assigned to each interface of each hardware and RIPng is activated as the routing protocol. In the RIPng protocol, the update time is 30 s [35]. It is allowed 190 s for the routers to detect all route information in the autonomous system. The system starts sending the packet after this time period.

4.2. Creation of reference topology with four autonomous systems

The second reference topology is composed of four autonomous systems as shown in Figure 8. There are three routing domains in each autonomous system. Within each routing domain, there are eight routers as shown in Figure 7. In this topology, HTTP and FTP servers added for non-RTA traffic in addition to voice and video traffic.

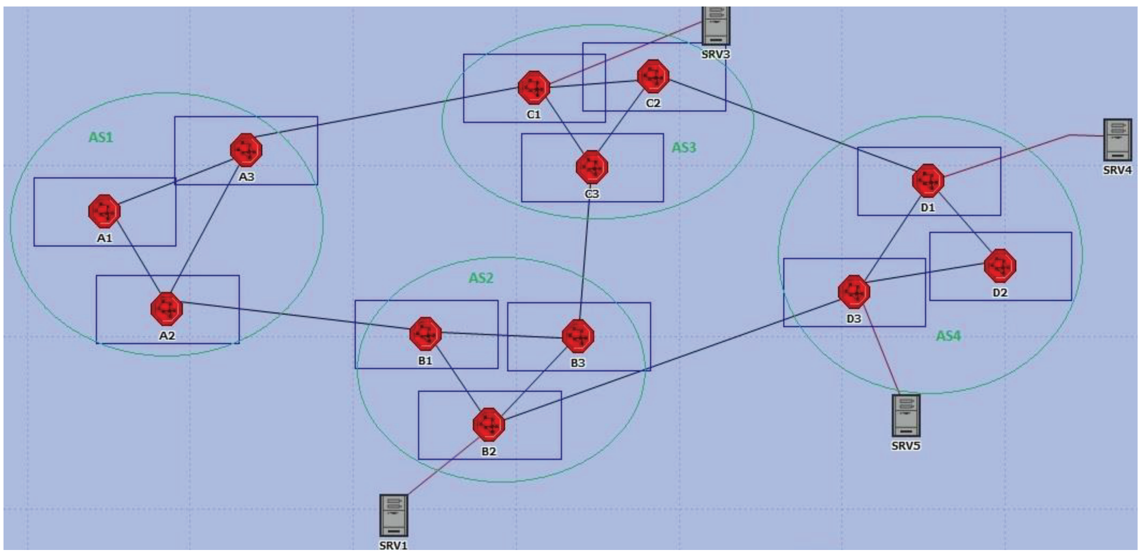


Figure 8. Reference topology consisting of four autonomous systems [36].

In this topology, RIPng runs as a routing protocol in the autonomous system and BGP-4 runs among the autonomous systems. The route information of the BGP protocol is redistributed into RIPng. Thus, RIPng protocol is provided to have routing information of BGP. The A1 domain is used as the source (SRC), while the C1 and D1 domains are designated as target (DST3 and DST4).

4.3. Evaluation of results

As mentioned before, reference topology consisting of one autonomous system is given in Figure 7. While LAN (SRC) connected to the R2 router is selected as the source, LANs connected to the R1 (DST1) and R8 (DST2) routers are selected as the destination. The results obtained are compared before and after the application of the prioritization algorithm on this reference topology.

As shown in Figure 9, the average delay value for the voice packets going from the SRC network to the DST1 network before the prioritization algorithm is 113 ms, while it is 111 ms for the video packets. After applying the prioritization algorithm, the average delay values for voice and video packets are 81 ms and 72 ms, respectively. While Figure 9 shows the delay values of the packets going from the SRC network to the DST1 network, Figure 10 shows the values of packets going to the DST2 network. Accordingly, before and after the prioritization algorithm is applied, the average delay values for voice packets are 116 ms and 94 ms, respectively. Similarly, for video packages, these values are 119 ms and 92 ms, respectively.

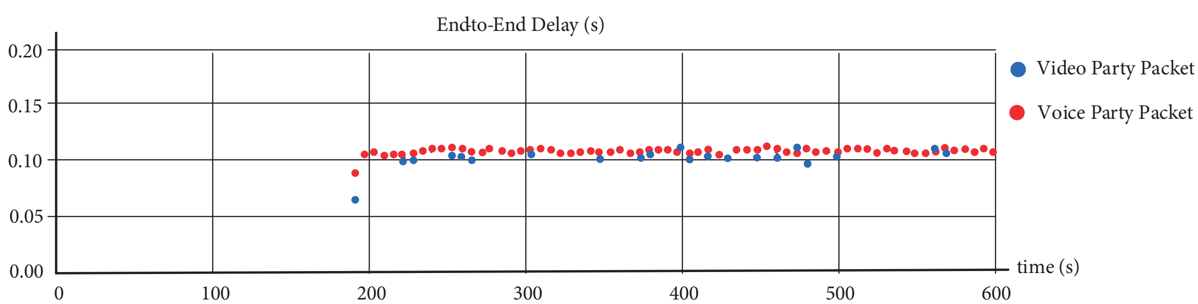
The reference topology consisting of four autonomous systems is given in Figure 8. A1 routing domain in AS1 is selected as the source, while C1 and D1 domains are determined as destination (DST3 and DST4). On this reference topology, the results obtained before and after applying the prioritization algorithm are given.

As shown in Figure 11, the average delay value for voice packets going from SRC network to DST3 network before the prioritization algorithm is 131 ms, while it is 134 ms for video packets. After applying the prioritization algorithm, the average delay values for voice and video packets are 109 ms and 105 ms, respectively. Figure 12 shows that the average delay value for voice packets going from the SRC network to the DST4 network is 139 ms, whereas for video packets this value is 141 ms. After applying the prioritization algorithm, the average delay values for voice and video packets are 111 ms and 107 ms, respectively.

All values of results are also given in Table.

Table . Delay values for audio and video packages before and after the prioritization algorithm is applied.

	DST1		DST2		DST3		DST4	
	Voice	Video	Voice	Video	Voice	Video	Voice	Video
Delay values before prioritization	113 ms	111 ms	116 ms	119 ms	131 ms	134 ms	139 ms	141 ms
Delay values after prioritization	81 ms	72 ms	94 ms	92 ms	109 ms	105 ms	111 ms	107 ms
Difference	32 ms	39 ms	22 ms	27 ms	22 ms	29 ms	28 ms	34 ms
Improvement rate	28.31%	35.13%	19.96%	22.68%	16.79%	21.64%	20.14%	24.11%



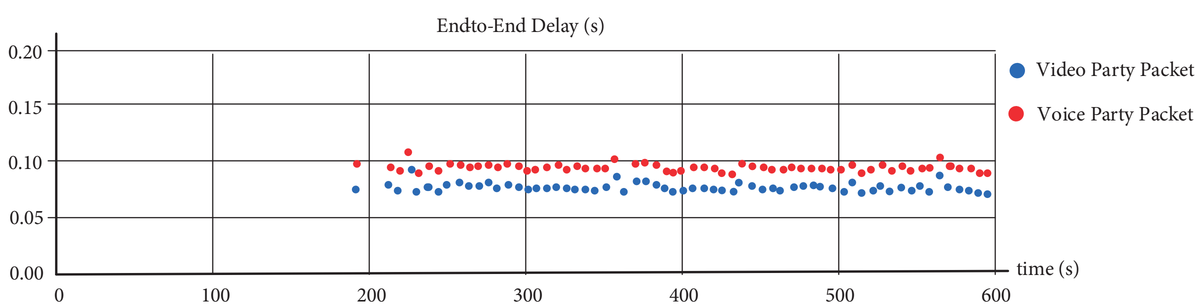


Figure 9. : Delay values for voice and video packets going from the SRC network to the DST1 network (a) before the prioritization algorithm is applied (b) after the prioritization algorithm is applied.

For RTA packets with the same metric value and placed in the same queue, a voice packet waits for the video packet that is being processed before it. Likewise, a video packet also waits for the voice packet being processed before it. A voice packet waits longer than a video packet since the video packet sizes (900–1500 bytes) are much larger than the voice packet sizes (75–250 bytes).

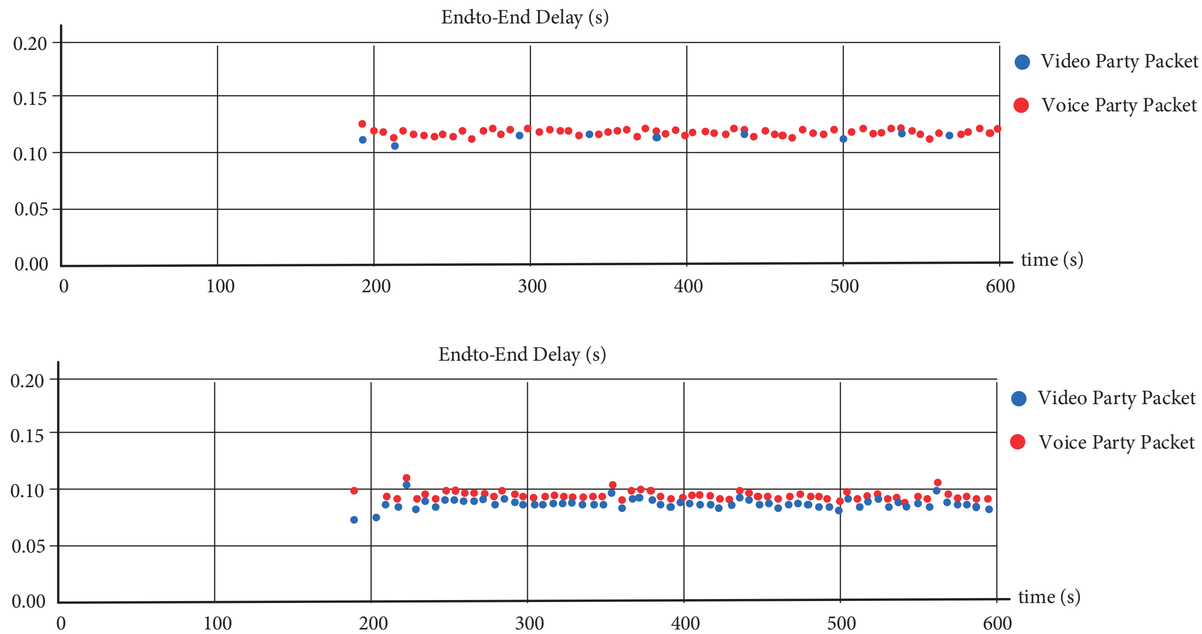


Figure 10. Delay values for voice and video packets going from the SRC network to the DST2 network (a) before the prioritization algorithm, (b) after the prioritization algorithm.

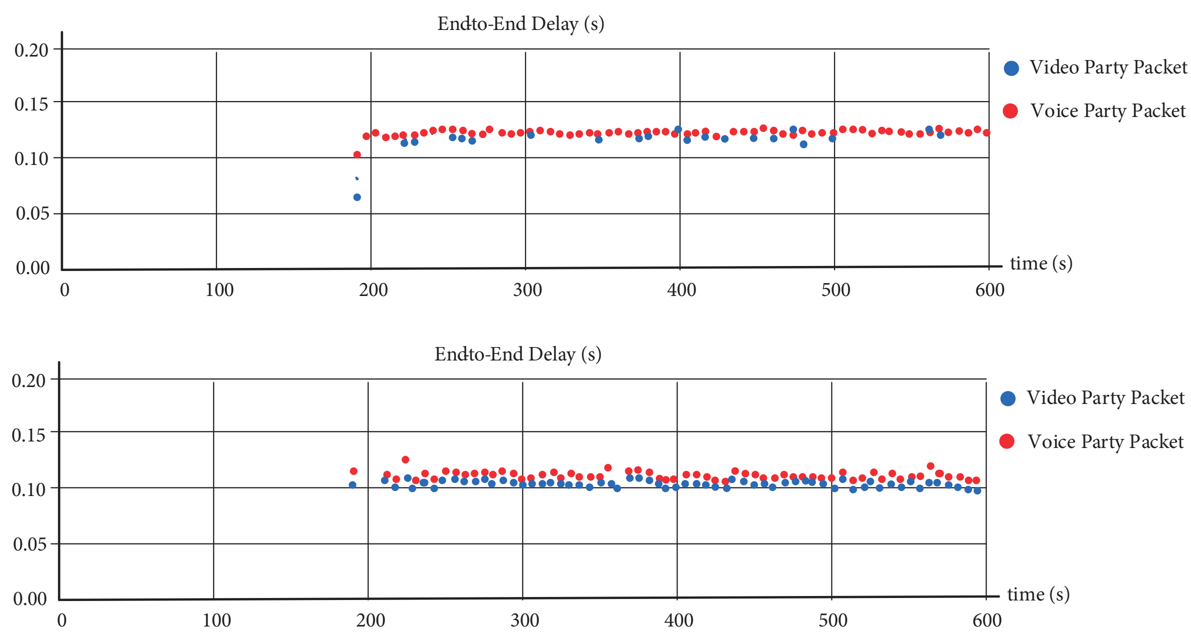


Figure 11. Delay values for voice and video packets going from the SRC network to the DST3 network (a) before the prioritization algorithm, (b) after the prioritization algorithm.

The end-to-end delay values obtained from the scenario where the prioritization algorithm was applied and not applied are compared. According to this comparison, it is found that the applied algorithm provides a significant improvement on the results. The improvement rate for voice packets going to the DST1 network

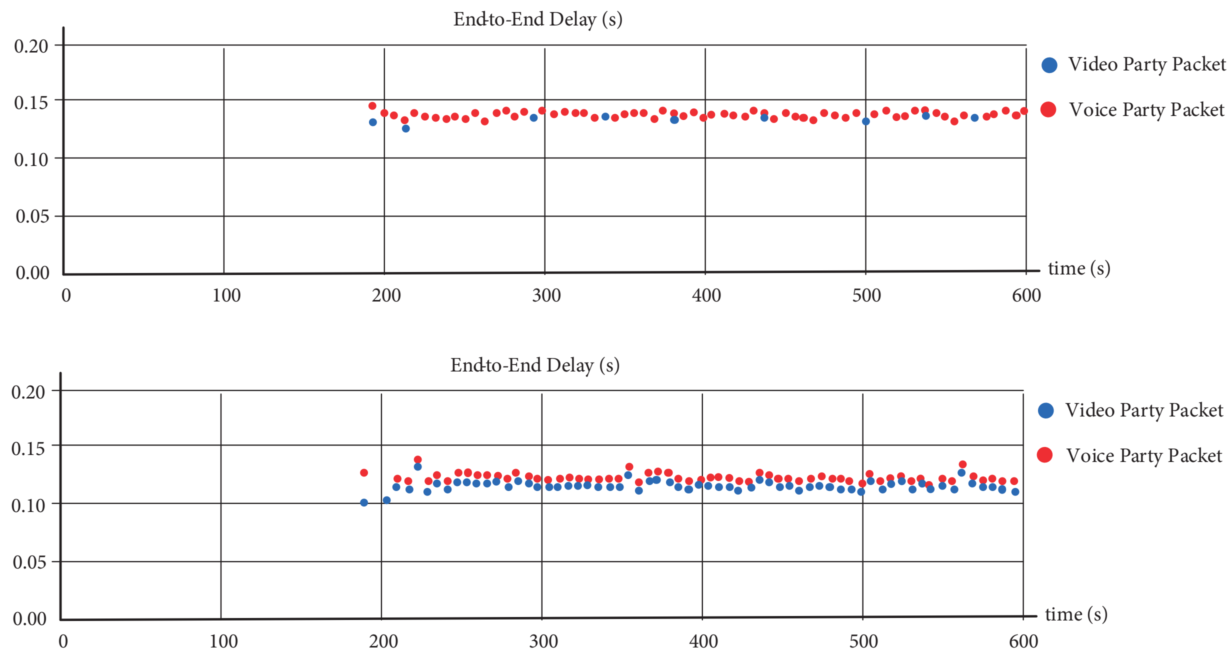


Figure 12. Delay values for voice and video packets going from the SRC network to the DST4 network (a) before the prioritization algorithm, (b) after the prioritization algorithm.

with low metric value is 28.31%, while for video packets this rate is 35.13%. Moreover, the improvement rates for the voice and video packets going to the high metric DST2 network were 19.96% and 22.68%, respectively. On topology consisting of four autonomous systems, the improvement rate for audio packets going to the DST3 network is 16.79%, while for video packages this rate is 21.64%. Likewise, the improvement rates for audio and video packets going to the DST4 network is found to be 20.14% and 24.11%, respectively. Similar results are obtained in some studies using a different technique than the method used in this study [37].

It should be noted that the results are obtained with multiple and different routing protocols and a new queuing model on small and large networks. However, a parameter such as the metric value of the routing protocol could be used to increase the QoS by prioritization. The use of this metric is used to reduce the delay as a novelty.

5. Conclusion

The metric values of the destination networks of the voice/video packets of the RTAs are inserted into the FL and intended to be used as a prioritization value. Thereby, in the present case, the data packets of the RTAs, which are prioritized according to best effort services, are also prioritized according to the destination parameter. For this purpose, the results are obtained by using the TaPQ classifier using the information from the routing table. The prioritization model prioritizes the RTA packets according to the metric value, thus providing a certain level of improvement in delay values. In addition, an approach to the use of the FL field within the IPv6 header is proposed. The novel prioritization model in this paper is expected to guide the study on supporting the Internet using different routing protocols and queuing techniques.

The proposed prioritization approach is suitable for public networks such as the Internet and does not require bordered networks under administrative control. Resource reservation is not required for prioritization and end-to-end support can be provided without interruption. The model in this article uses the metric value in routing protocols that use the distance vector algorithm as the priority value. Therefore, only the ”distance” parameter is used as the target network parameter value. However, parameters such as bandwidth found in link-state routing protocols are not included in this model. These parameters will be added to the model to further reduce the delay value in future studies.

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