Workshop on Computer Networks

Real-Time Voice Quality Guarantee On The IP Based Networks

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Abstract

In this paper, a new proposed VoIP (Voice over Internet Protocol) model has been implemented to guarantee the required voice quality based on the available network resources. Basically, the new proposed model depends on a simple Differentiated-Services (DS) blocks at the edge of the networks [1]. Although, it is so simple from the routers point of view, it is somehow complex from the host point of view due to the input characteristics analyzing process. This model has some determined values for both the required buffers through the path and the enough reserved bandwidth to guarantee the required quality. On the other hand, the QBone (Quality of service backbone) model [2] does not analyze the input traffic (simple at the host side) but it moves the complexity to the router side due to complex queuing disciplines implementation. In fact, the QBone model does not face any success till now from the practical implementation point of view because most of the users are not welling to pay much money to have voice conversations across the Internet. So, the new proposed model not only guarantees the voice session's quality but also saves the network resources (saves up to 74% of the consumed bandwidth more than the QBone model). In this paper, the G.723.1 (5.3)/(6.4), G.711, and G.726 voice encoders' characteristics representing the proposed model inputs have been analyzed by implementing some sessions capturing experiments. Then the model design and three simulation scenarios assumptions, descriptions, and analytical representation have been described. Also, the proposed model results, for two different configurations, have been obtained based on the node delay, routers

buffers consumptions, jitter, and packet loss. Finally, a detailed comparison between both the new proposed model and the QBone model has been presented.

1. Introduction

Currently, as shown in the figure 1, there is a great need to have voice sessions across the Internet [3]. Real-time voice traffic could be defined as datagrams that are delay sensitive, and packet loss sensitive (the voice packets could be considered to be lost because it is too late to be played-back).

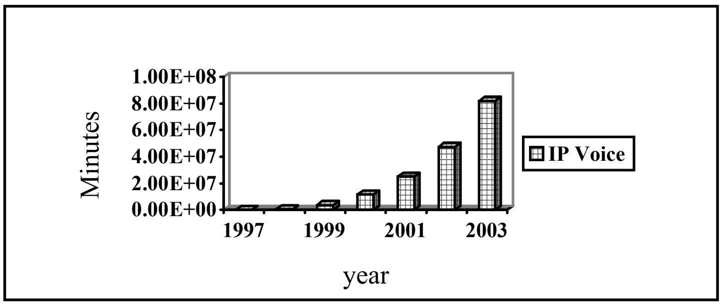

Figure 1: IP Voice Growth Forecast 1997-2003

IP makes no assumptions about the underlying protocol stacks and offers an unreliable, connectionless network-layer service that is subject to packet loss, packet duplication, and queuing delay. Because of the lack of any firm guarantees, the traditional IP delivery model is often referred to as Best-Effort (BE). Therefore, real-time voice traffic could not depend on the BE delivery of datagrams across the Internet. Also there is still a firm need to provide many applications with additional service classes offering enhanced quality of service (QoS) with regard to bandwidth, packet queuing delay, and loss. So, like a QBone model, there is a need to a model that gives a per-flow service assurance. The QBone model is scaleable and it guarantees the required QoS, but it does not have any quantified assumptions regarding both the minimal latency, and the jitter effects. Also, it reserves much more bandwidth than could be needed (equals to the input traffic peak rate) to have a very low packet loss. So, there is a need to have a new model that have quantified assumption regarding the latency, the jitter, and it should be network-resource saver to maintain an expected quality level of service for real-time voice streams across the Internet. The new proposed model, which achieves such services, has been described in details at section 3. The following section describes some voice over IP capturing experiments representing the proposed model inputs.

2. Encoded Voice Traffic Characterization

In this phase, the nature of the encoded voice traffic over the IP based networks has been studied carefully. Actually, the new proposed model architecture strongly relies on this phase. Normally, at the network-traffic modeling phase, the packet inter-arrivals are often assumed to be *poisson* processes for the analytical simplicity. But a number of studies have been shown that the distribution of packet inter-arrivals clearly differs from exponential especially for voice streams [4]. Many of them demonstrate the failure of *poisson* modeling to capture the burstness present in actual network traffics. It has been concluded that the Internet traffic is too complicated to be modeled using certain techniques. So in section 3.1, some encoded voice traffic on the IP based networks has been studied experimentally.

2.1. VoIP Experiments

In these experiments, two fully equipped computers at the same Ethernet Multimedia lab, Microsoft NetMeeting V.3.0 (as a videoconferencing tool) [5], and PacketBoy V.1.6 (as a packet analyzer software) [6] have been used to study and test some features of the encoded voice. Different types of encoders have been implemented during these experiments such as G.723.1 (5.3), G.723.1 (6.4), G.711 (a law), and G.726 (ADPCM), [7], [8], [9]. The PacketBoy analyzer has been used to capture the voice packets on assigned filter from the source to study the voice traffics characteristics of a certain type on encoder, in which the voice conversation is done between a source address (IP: 62.241.131.30, Ether: 00:b0:d0:65:b6:40) and destination address (IP:62.241.131.31, Ether:00: 03:47:07:53:61) by using protocol-filter of Ethernet/IP/UDP/data. The experiments have been implemented for 10 times each for 3.5 minutes with different encoder type, where the voice conversation consists of a combination of talk/pause time, high/low voice tone and fast/slow speech. Table 1 shows some of the obtained voice encoders' features. Also, figures 2 and 3, illustrates the most and the least repeated pattern for the studied voice encoders.

Encoder	**Max. Packet Size *(bytes)***	**Avg. Packet Size *(bytes)***	**Min. Inter-arrival time *(msec)***	**Avg. Inter-arrival time *(msec)***	**Avg. Rate (*kbps*)**	**Peak rate *(kbps)***
G.723.1 (6.4)	138	78	1.5	30	21	80
G.723.1 (5.3)	138	74	1.5	29.5	20	64
G.711	310	310	.5	30.5	81	128
G.726	310	310	1	61	40.5	88

Table 1: Voice Encoders results.

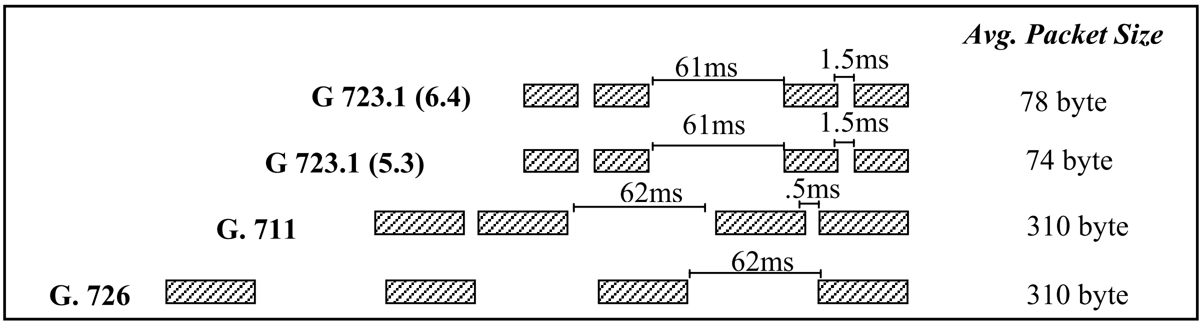

Figure 2: Voice Encoders most repeated pattern results.

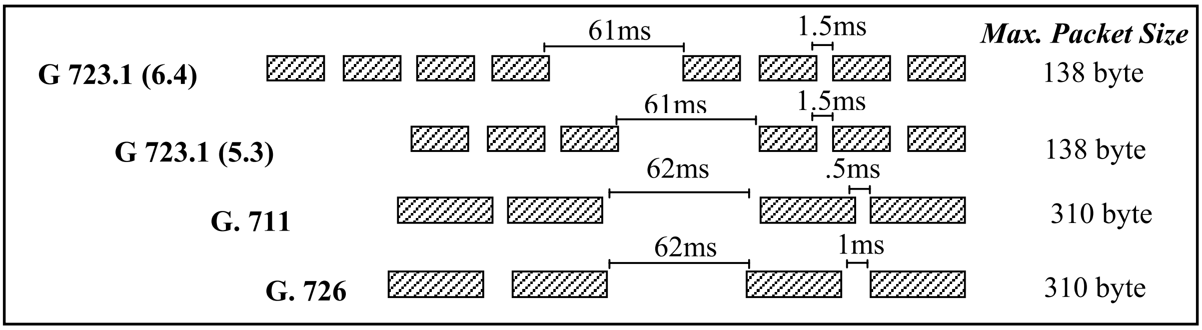

Figure 3: Voice Encoders least repeated pattern (burstness period) results.

3. The New Proposed Model Design

The new proposed model guarantees the required quality by using some of the Differentiated services simple blocks at the edge of the IP based networks [1]. This new model relies not only on marking the VoIP packets but also on dropping the out-of-profile VoIP marked packets. By implementing such simple blocks at the edge only, the overload on the core routers will be minimized. But the proposed model moves the design complexity to the host side by implementing ***N*** meters for ***N*** VoIP sessions. The new proposed model blocks description, and functionality will be discussed in the following sections.

3.1. The Proposed Model Architecture

In this section, as shown in figure 4, the overall structure of the new proposed model has been described. In this model, two types of traffics have been proposed VoIP, and Best-effort traffics. The following sections describe the main proposed model blocks.

3.1.1. Applications Classifier

It classifies the streams based on the used applications into two classes. The first one is Class-1, which has been used for the VoIP applications whereas the second is Class-2, which has been used for the BE applications. Then the application classifier, as shown in figure 4, passes the VoIP packets to VoIP marker and BE packets to the BE marker.

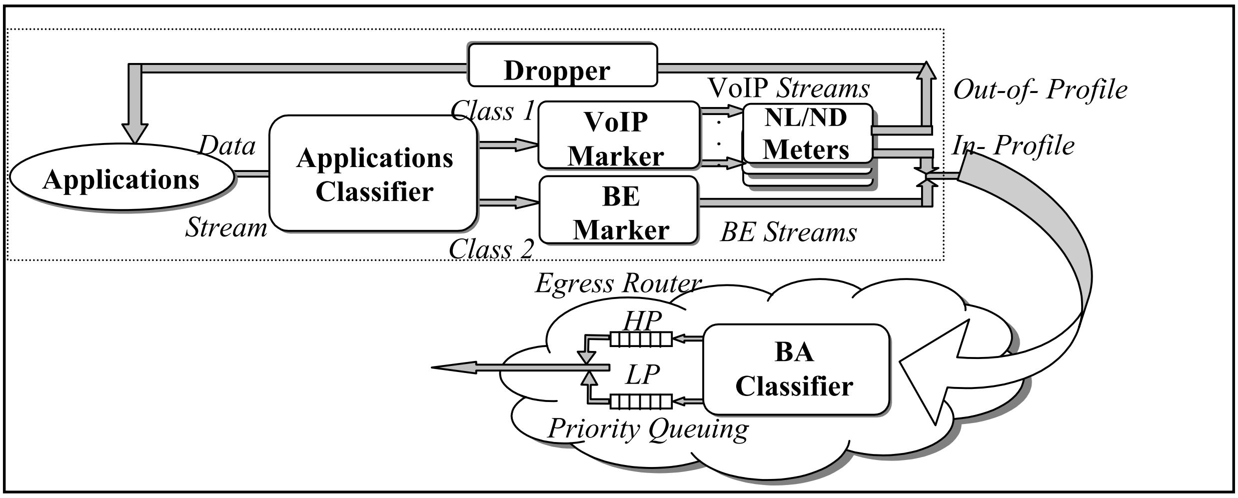

Figure 4: A New IP Quality of Service Model.

3.1.2. Marker

VoIP-marker marks the VoIP incoming packet with value equals to 6, and BE-marker marks the BE packet with value equals to zero in the precedence field located inside the TOS (Type Of Service) field in the IPv4 header [10]. The suggested values for precedence field could be described as shown in table 2. Then the VoIP-marker passes the marked VoIP packets to the NL/ND-meter(s), but the BE-marker passes the marked BE packets directly to the Egress router.

Precedence Value	**Description**
7	*Network Control*
6	*VoIP Applications*
5→1	*Unused*
0	*Best-Effort Applications*

Table 2: Proposed Precedence Values.

3.1.3. NL/ND (No-loss/No-delay) Meter(s)

The model has N meters for N VoIP sessions at the same time. Each meter design depends on both the voice encoder characteristics and the session required-quality. Actually, this is the complex part of the proposed model due to the possibility of having many voice sessions at the same time. This kind of meter is essential to study and analyze the input traffic characteristics without causing any delay or loss (No-loss/No-delay) for its *In-Profile* packets (packets that are conformed to a defined traffic profile). The proposed meter depends on *No queue Single Token Bucket*, in which it holds up to ***b*** (bucket depth) tokens that are generated at a rate of ***r*** tokens per seconds. When voice packets arrive at the bucket, if sufficient tokens are available then the traffic is said to be *In-Profile* and the corresponding number of tokens are removed from the bucket then it passes the *In-Profile* packets to the Egress router. On the other hand, if insufficient tokens are available then the traffic is said to be *Out-of-Profile* then it passes them to the dropper block. Finally, the analytical representation for the ***r*** bounds and the relation between both ***r*** & ***b*** that should be implemented to satisfy no-loss/no-delay for the input traffic could be represented in the following equations (derivations at [11]).

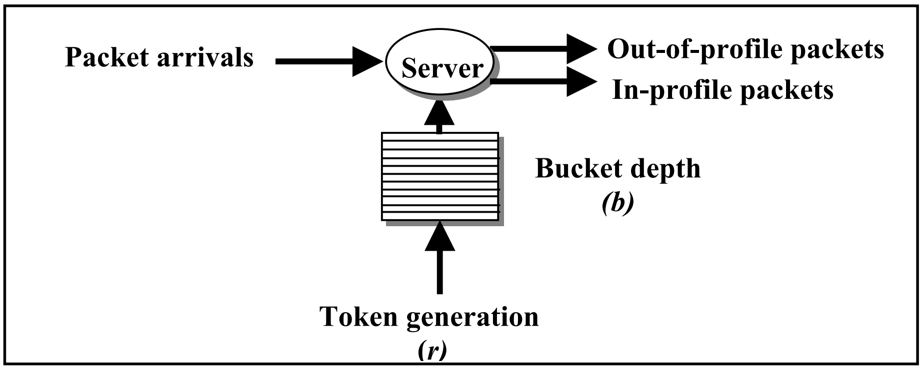

Figure 5: No Queue Single Token Bucket Mechanism.

$$\mathbf{r_{min} = Max\{[\frac{1}{\tau - s}(R_{avg} \times \Delta t_x + R_i \times \Delta t_i - P \times s)], R_{avg}\}} \quad (1)$$

$$\mathbf{r_{max} = P} \quad (2)$$

$$\mathbf{b = (P - r) \times s} \quad (3)$$

Where: **r** is the selected token-rate, $\mathbf{r_{min}}$ is the lower permitted bound of the token-rate, $\mathbf{r_{max}}$ is the upper permitted bound of token-rate, **b** is the selected token depth, **P** is the input traffic peak rate, **S** is the burstness time, $\mathbf{R_i}$ is the input traffic rate at any time i, $\mathbf{R_{avrg}}$ is input traffic average rate, $\Delta \mathbf{t_i}$ small time interval between $\mathbf{t_{i-1}}$ & $\mathbf{t_i}$, $\Delta \mathbf{t_x}$ time interval duration from t=0 to t=t_x, and $\boldsymbol{\tau}$ equals to $\Delta \mathbf{t_i} + \Delta \mathbf{t_x}$.

From equations 1,2 and experiments results, it could be concluded that the NL/ND-meter token-rate varies from ~$\boldsymbol{R_{avg}}$ to ***P***. This means that it is possible to design the required meter with token-rate equals to value much less than the input traffic peak rate (e.g. from ~21 kbps to 80 kbps for G.723.1 6.4 as shown at table 1). In the new proposed model, the token-rate equals to the session required bandwidth (should be in the range between ~$\boldsymbol{R_{avg}}$ to ***P)***. So, the bandwidth saving percentage (up to 74% at the pervious mentioned encoder) could reduce the overall session-costs with great values. Then from equation 3, the voice-session token-depth **b** could be obtained based on the token-rate that has been obtained from the session required bandwidth value.

3.1.4. Dropper

The Dropper block drops the *Out-of-Profile* packets (exceed) arriving from the NL/ND meter(s) and send an *Error_Drop* message to the used application to re-adjust the input traffic settings.

3.1.5. Egress Router

It is the boundary node, which handles the leaving traffic from the Domain. First, it classifies the incoming packets by using the BA (Behavior Aggregate) classifier, in which it search for the *precedence* field inside the *TOS* field in the IPv4 header. Then, it applies the two queues *priority queuing*, and sends the VoIP packets to the highest priority queue and sends the BE packets to the lowest priority queue.

4. Simulation Scenarios

In this section, three simulations scenarios, which suits the high, average, and low loaded networks respectively, have been implemented on the proposed model design, as illustrated at section 3, by using the voice encoders' characteristics that have been obtained at section 2. The target from such simulations is to prove that by implementing the new proposed model the voice session costs decreases (due to bandwidth savings) and the received voice quality is guaranteed. Also, by implementing the new proposed model, the required buffer size, the node delay, the jitter, and the packet loss for the voice sessions through the path are determined (i.e. the expected quality is defined). Actually, many configurations have been implemented on the three proposed scenarios during this work, but only two configurations (the voice maximum allowed bandwidth equals to 40% from the total link capacity 15Mbps at configuration-1 & 4.5Mbps at configuration-2) of them have been presented in this paper. The queuing system applied on the aggregated voice packets for the proposed scenarios has many items of concern and they have been summarized in the following section.

4.1. Scenarios' Queuing Model

In this system the source (voice session) is finite but very large, so it is customary to assume an infinite source system. This turns out to be a reasonable approximation, and it greatly simplifies the mathematics involved and it enables us from determining the required buffers at the routers. Also, the voice packets arrive at the queuing system randomly, one at a time, and there is never a collision in which two packets attempt to enter the system at the exact same time. The voice inter-arrival times have been assumed to be independent. The service time depends on the router service rate and the incoming voice packet size. Also, the arriving voice packets are allowed to enter the queuing system and wait, no matter how many waiting packets there are. So, queue capacity is infinite. Finally, a single server has been used in this system, so only single packet could be served at a time. The following sections represent the proposed scenarios' descriptions and assumptions that have been applied on this queuing model for the aggregated voice packets.

4.2. High Load Scenario

In this scenario many assumptions, but with very low probabilities, that may happen to the encoded voice traffics due to the traffic aggregations have been assumed. This scenario depends on aggregating the flows during its burstness period simultaneously. It is highly recommended to implement such scenario with the highly loaded links (more than 70%) because the scenario's assumptions guarantee that the packet loss equals to almost zero.

4.2.1. Scenario's Assumptions

During the flows' aggregations, many assumptions have been assumed and they are presented as follows:

- The BE packet arrives just before the voice packet by $\boldsymbol{\varepsilon}$ second (ε is a very small fraction of time ~ zero) and the BE packet has a maximum size (1514 bytes).
- The $\boldsymbol{f}$ flows have been aggregated simultaneously with the least repeated pattern (with probability less than 0.1%), as shown in figure 3, which consists of γ packets (4 packets for G.723.1 (6.4), 3 packets for G.723.1 (5.3) and 2 for both G.711 and G.726).
- All the voice packets have the maximum packet size and the minimum inter-arrival time.
- The router service rate (SR) has a very low value (2000 packet/sec.).

4.2.2. Scenario's Description

The worst event (highest burstness) is caused by the arrival of maximum BE packet size while the highest priority queue of the core router is empty, followed by the immediate arrival of the worst voice pattern. Where the worst pattern of a certain voice encoder happen when γ voice packets arrive with a maximum packet size and minimum interarrival time Δt_{min}. The worst aggregation interarrival time $\Delta t(i)\big|_{Wa}$ for ***f***-aggregated flows, for the same type of encoder, equation is as follows:

$$\Delta t(i)\big|_{Wa} = \frac{\Delta t_{min}}{f} \qquad (4)$$

In case of $\Delta t(i)\big|_{Wa}$ is greater than or equal to the processing delay (1/SR) for the voice packet, the worst queuing delay (WQD_{VoIP}) equation for ***f***-aggregated flows is as follows:

$$WQD_{VoIP} = \frac{1}{SR} \qquad (5)$$

Otherwise, $\Delta t(i)\big|_{Wa}$ is less than the processing delay of the voice packet, the worst queuing delay equation for ***f***-aggregated flows is as follows:

$$WQD_{VoIP} = (\gamma f - 1)\left[\frac{1}{SR} - \Delta t(i)_{Wa}\right] + \frac{1}{SR} \qquad (6)$$

Where γf represents the last aggregated packet in the worst pattern.

4.3. Average Load Scenario

In this scenario some of the possible assumptions, but with low probabilities, that may happen to the encoded voice traffics due to the traffic aggregations have been assumed. It is recommended to implement this scenario with the average loaded links (from 35% to 70%) because the scenario's assumptions guarantee that the packet loss will be very low (less than 10^{-6}).

4.3.1. Scenario's Assumptions

During the flows' aggregations, many assumptions have been assumed and they are presented as follows:

- The BE packet arrives just before the voice packet by *ε* second the BE packet has a maximum packet size.
- The ***f*** flows have been aggregated simultaneously with the most repeated pattern (with probability up to 99%), as shown in figure 2, and all the voice packets have the average packet size and the average inter-arrival time.
- The router has a very low service rate (2000 packet/sec.).

4.3.2. Scenario's Description

This scenario depends on the arrival of maximum BE packet size while the highest priority queue is empty, followed by the immediate and simultaneous arrival of the most repeated voice pattern with an average packet size. From the measurements the first ***N*** VoIP packets (from ***N*** = 1 to ***f***+1) have the following queuing delay equation.

$$\mathbf{QD_N} = \left(\mathbf{N \times SR}\right) - \left[\frac{\mathbf{t_{i1}} \times \left(\mathbf{N-1}\right)}{\mathbf{f}}\right] \qquad (7)$$

Then the queuing delay for the ***N*** VoIP packets (from ***N*** = ***f***+2 to 2***f***+1) have the following equations:

$$\mathbf{QD_N} = \begin{cases} \mathbf{0} & \left(\mathbf{t_{i1}} + \dfrac{\mathbf{t_{i2}}}{\mathbf{f}}\right) \quad \left(\mathbf{N \times SR}\right) \\ \\ \left(\mathbf{N \times SR}\right) - \left(\mathbf{t_{i1}} + \dfrac{\mathbf{t_{i2}}}{\mathbf{f}}\right) \times \left(\mathbf{N - f - 1}\right) & \end{cases} \qquad (8)$$

Where: QD_N is the queuing delay of packet number ***N***, $\boldsymbol{t_{i1}}$ is the first inter-arrival time between the voice packets, and $\boldsymbol{t_{i2}}$ is the second inter-arrival time between the voice packets.

The above equations have analyzed the VoIP packets from the first till the packet number 2***f***+1. Then the delay for the followed packets will be repeated due to the VoIP packets periodical behavior in this scenario.

4.4. Low Load Scenario

In this scenario some of the most expected assumptions, with a very high probability (up to 99%), that may happen to the encoded voice traffic due to the traffic aggregations have been assumed. It is recommended to implement this scenario with the low loaded links (less than 35%) because the scenario's assumptions do not guarantee the packet loss bounds.

4.4.1. Scenario's Assumptions

During the flows' aggregations, few assumptions have been assumed and they are presented as follows:

- The ***f*** flows have been aggregated simultaneously with the most repeated pattern, as shown in figure 2, and all the voice packets have the average packet size and the average inter-arrival time.
- The router has an average service rate (80,000 packet/sec.).

4.4.2. Scenario's Description

This scenario depends on the arrival of the most repeated voice pattern with the average packet size while the highest priority queue is empty. The first ***N*** VoIP packets (from ***N*** = 2 to ***f***+1) have the following queuing delay equation (the first packet queuing delay = zero):

$$\mathbf{QD_N = (N-1) \times \left[SR - \frac{t_{i1}}{f} \right]} \qquad (9)$$

Then the queuing delay for the ***N*** VoIP packets (from ***N*** = ***f***+2 to 2***f***+1) have the following equations:

$$\mathbf{QD_N = \begin{cases} 0 & \left(t_{i1} + \frac{t_{i2}}{f} \right) \geq (N-1) \times SR \\ (N-1) \times SR - \left(t_{i1} + \frac{t_{i2}}{f} \right) \times (N-f-1) & \end{cases}} \qquad (10)$$

5. Results

The following paragraphs describe some of the obtained results for the three-implemented scenarios applied on the two-selected configurations. These results cover the network requirements parameters such as bandwidth consumption and the buffer consumption. Also the results cover the applications requirements parameters such as the node delay (transmission + queuing + processing), the maximum jitter, and the packet loss probability.

5.1. High Load Scenario Results

From the scenario results, it has been concluded that the scenario probability is almost zero. And to lose a packet due to exceeding the voice delay bounds (150-400 msec.), this may happened only if the packet has faced worst scenario than the high load scenario. So, as a conclusion, the worst scenario probability should be lower than the high load scenario probability, which has almost zero value. As a result the packet loss percentage for such scenario is almost zero value.

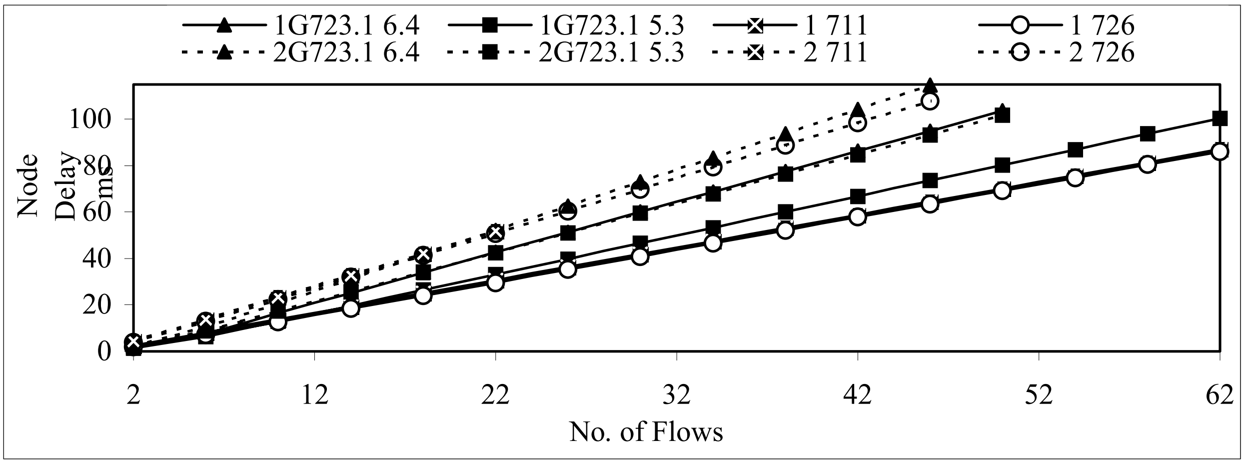

Figure 6: Node delay vs. number of aggregated flows.

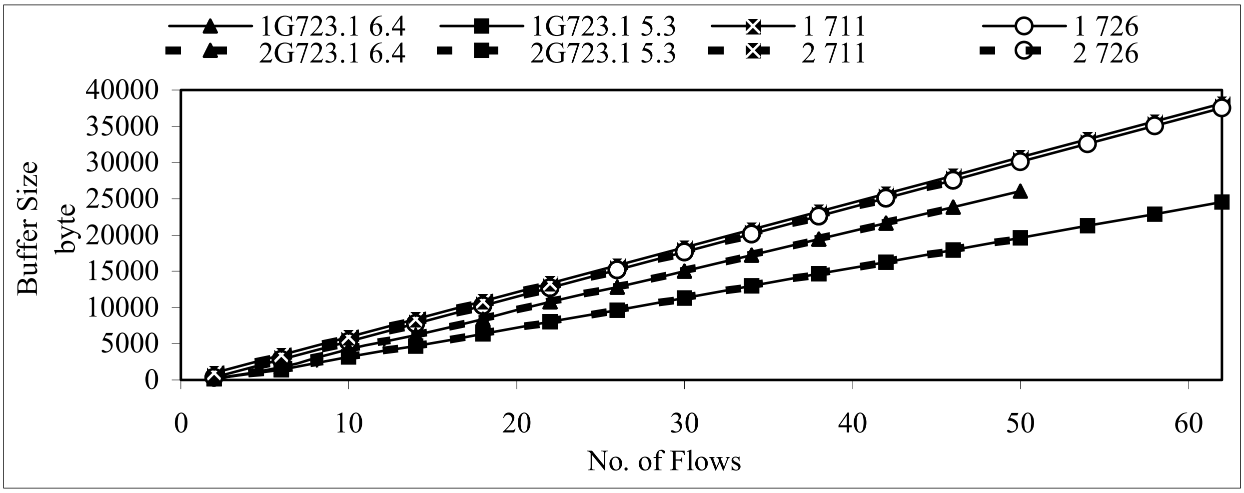

Figure 7: Buffer size vs. Number of aggregated flows.

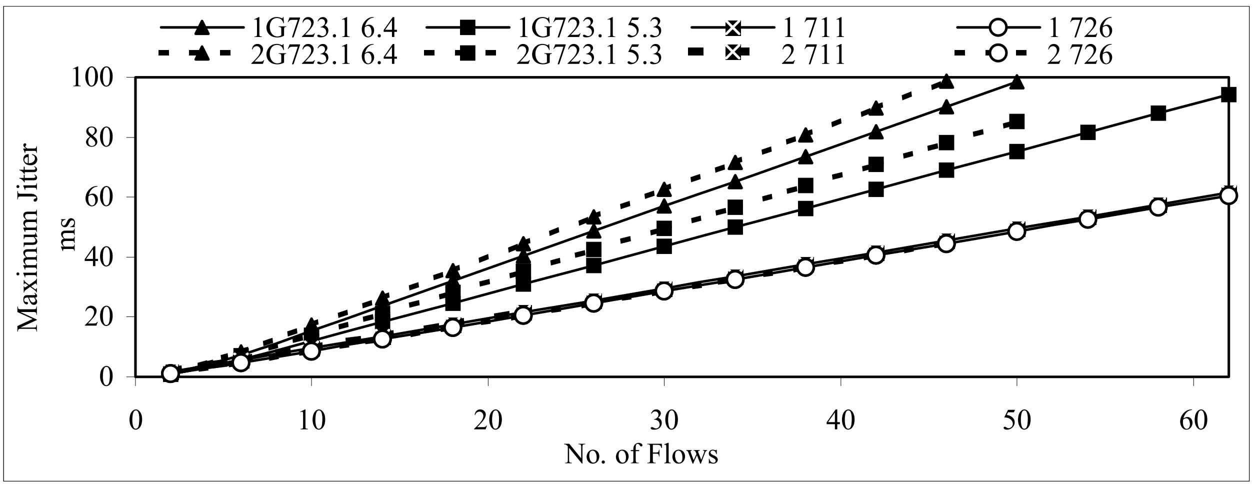

Figure 8: Maximum jitter vs. Number of aggregated flows.

From the previous figures the followings have been concluded on the average bases:

- For f G.723.1 (5.3) aggregated flows, the VoIP packet faces less than 1.8f msec. node delay, consumes less than 396f bytes from each node across the path, and faces less than 1.5f msec. jitter delay.
- For f G.723.1 (6.4) aggregated flows, the VoIP packet faces less than 2.2f msec. node delay, consumes less than 520f bytes from each node across the path, and faces less than 2f msec. jitter delay.
- For f-(G.711 or G.726) aggregated flows, the VoIP packet faces less than 1.9f msec. node delay, consumes less than 615f bytes from each node across the path, and faces less than 1f msec. jitter delay.

5.2. Average Load Scenario Results

From the scenario results, it has been concluded that the scenario probability is less than 10^{-6}. So, in general, the packet loss for such scenario represents an acceptable packet loss for the voice encoders.

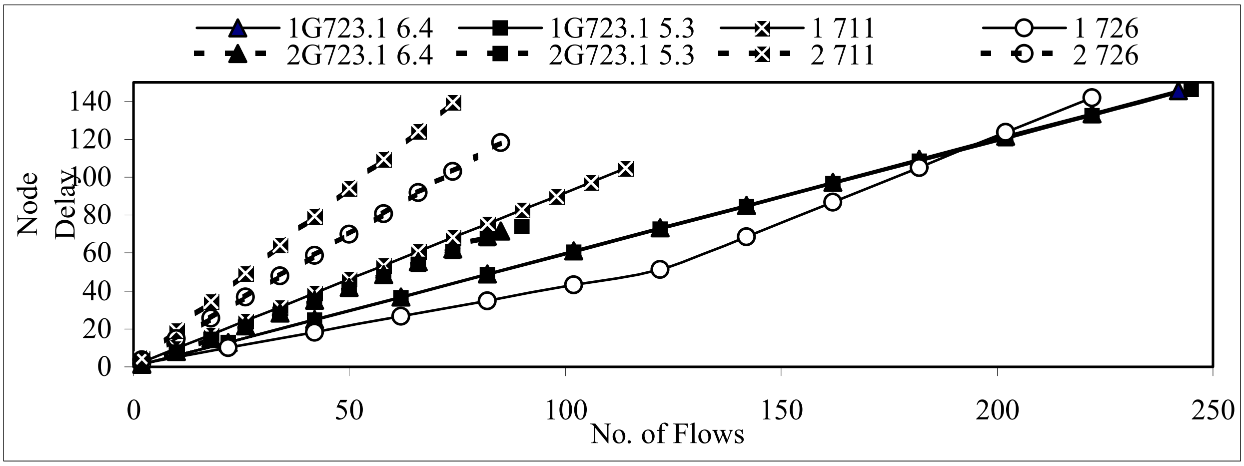

Figure 9: Node delay vs. Number of aggregated flows.

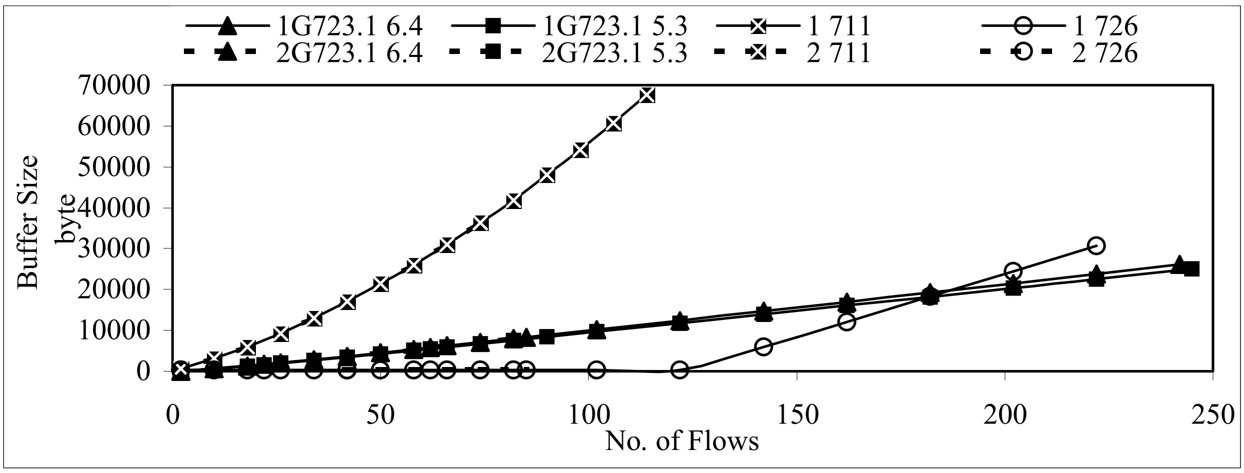

Figure 10: Buffer size vs. Number of aggregated flows.

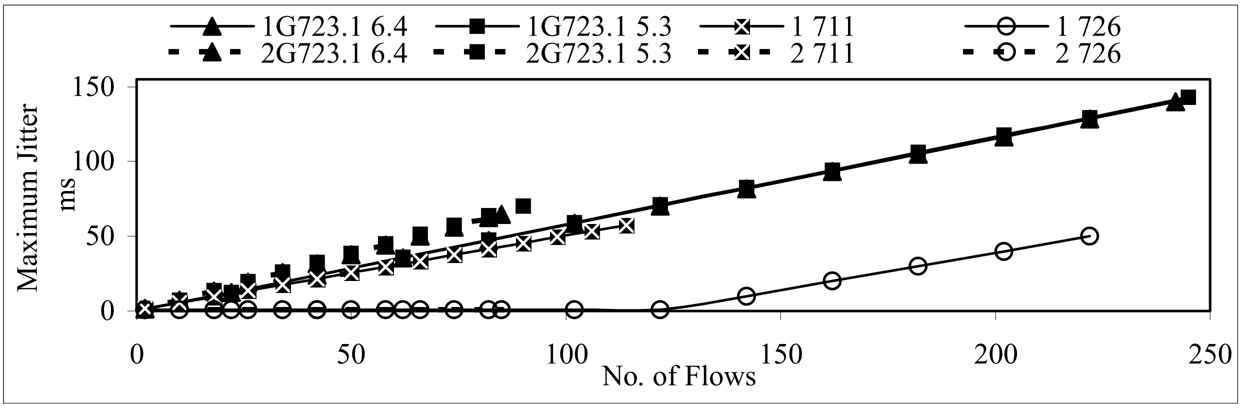

Figure 11: Maximum Jitter vs. Number of aggregated flows.

From the previous figures the followings have been concluded on the average bases:

- For ***f***-(G.723.1 family) aggregated flows, the VoIP packet faces less than 0.7***f*** msec. node delay, consumes less than 101***f*** bytes from each node across the path, and faces less than 0.5***f*** msec. jitter delay.
- For ***f*** G.726 aggregated flows, the VoIP packet faces less than 1***f*** msec. node delay, consumes less than 304***f*** bytes (after ***f*** =122) from each node across the path, and faces less than 0.5***f*** msec. jitter delay.
- For ***f*** G.711 aggregated flows, the VoIP packet faces less than 1.4***f*** msec. node delay, consumes less than 593***f*** bytes from each node across the path, and faces less than 0.5***f*** msec. (after ***f*** =130) jitter delay.

5.3. Low Load Scenario Results

From the scenario results, it has been concluded that the scenario probability value is up to 0.98. So, the packet loss for such scenario, from the theoretical point of view, could be very high and unacceptable. But, practically, this scenario deals only with lightly loaded networks (less than 35% loading) and this minimizes the probability of facing worst scenarios than the low load scenario.

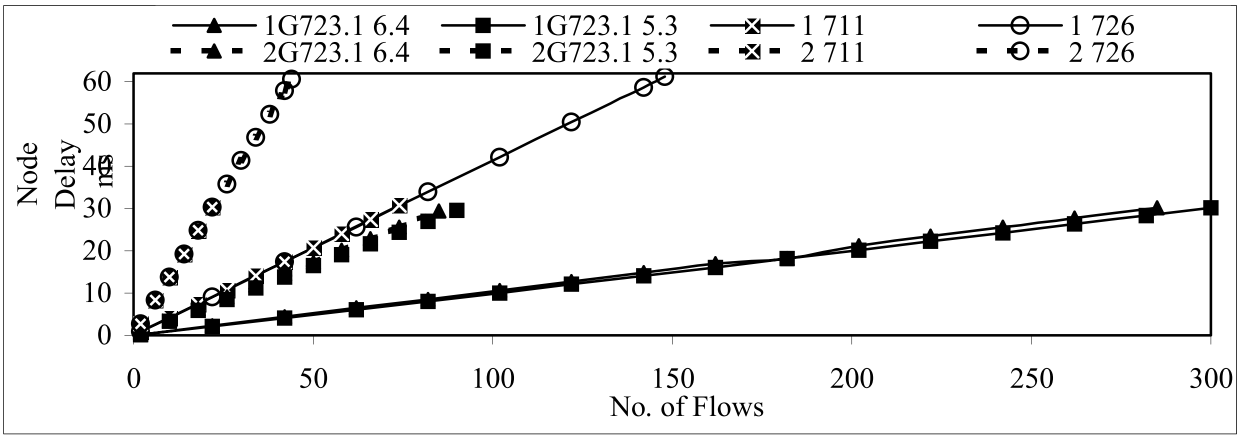

Figure 12: Node delay vs. Number of aggregated flows.

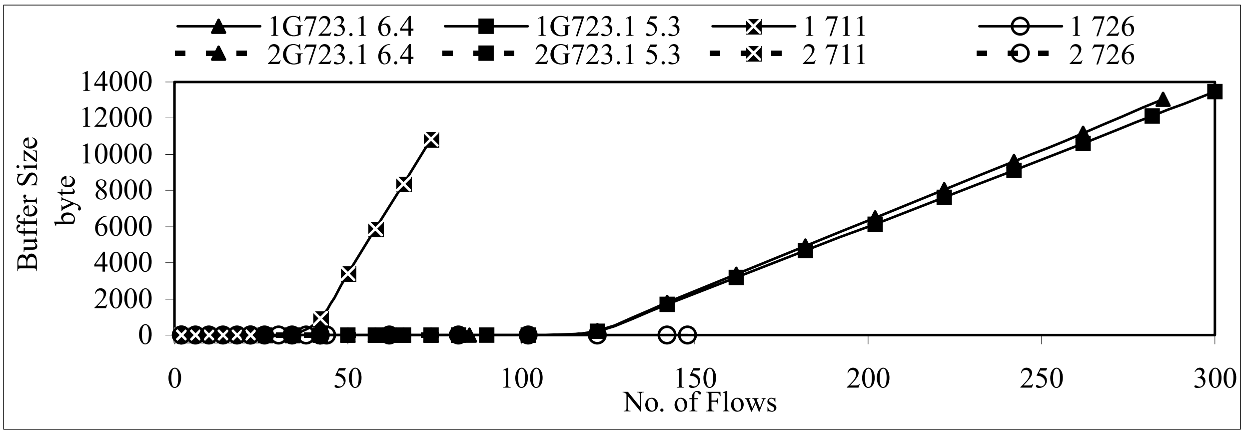

Figure 13: Buffer size vs. Number of aggregated flows.

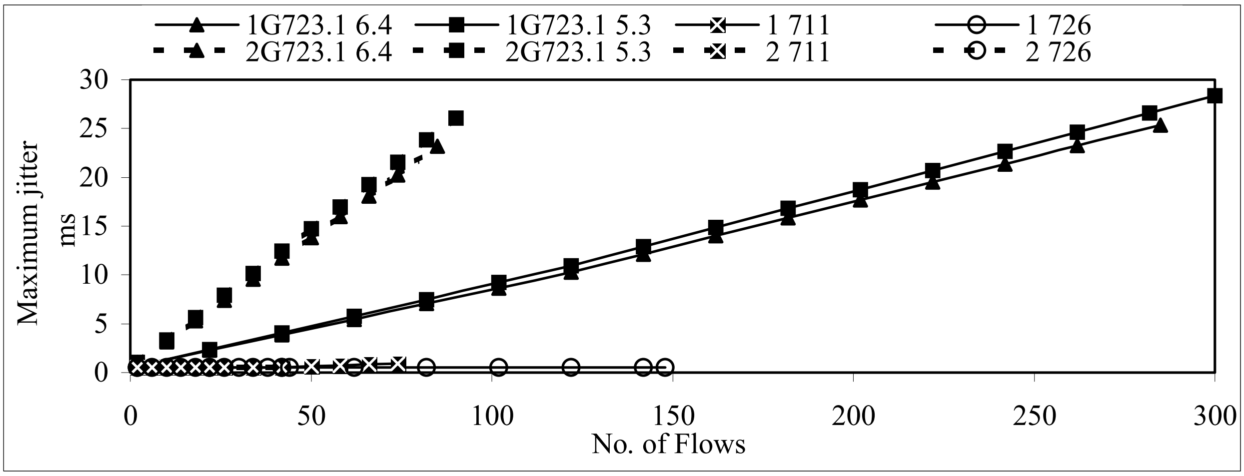

Figure 14: Maximum Jitter vs. Number of aggregated flows.

From the previous figures the followings have been concluded on the average bases:

- For f-(G.723.1 family) aggregated flows, the VoIP packet faces less than $0.2f$ msec. node delay, consumes less than $78f$ bytes (after $f = 122$) from each node across the path, and faces less than $0.2f$ msec. jitter delay.
- For f G.726 aggregated flows, the VoIP packet faces less than $0.9f$ msec. node delay, consumes almost zero bytes from each node across the path, and faces almost zero jitter delay.
- For f G.711 aggregated flows, the VoIP packet faces less than $0.9f$ msec. node delay, consumes less than $310f$ bytes (after $f=40$) from each node across the path, and faces almost zero jitter delay.

6. Comparative Analysis between the new proposed model & the QBone model

Sometimes the QBone model may have better quality than the new proposed model. However, it does not face much success, and this is because its huge consumption to the network resources. This leads to costs increase to the limit that the users prefer to use the free much-less-quality service across the Internet (based on BE service). But, the new proposed model gives not only an acceptable guaranteed quality of service (better than the BE service) but also a very low cost service, which should suit the Internet users. The following sections highlights on the main important differences between both the QBone model and the new proposed model.

6.1 The Models' Architectures

The new proposed model faces some complexities at the host part due to the possibility of have many voice sessions at the same time. But such complexities could be accepted due to two reasons. The first is the high processing capability of the host that enables it from handling large number of sessions at the same time. The second is the host capability to accept or reject any new voice sessions based on its loading percentage. On the other side, the QBone model suffers also from the complexities due to complex queuing disciplines implementations (such as weighted fair queuing or N priority queuing disciplines). The problem here, for the QBone model, is much more effective than the previous one due to two reasons. The first is due to the core-routers high loading percentage, and any extra complexities or loads due to QoS queuing issues may lead to the router congestion. The second is the routers disability to reject any extra QoS sessions that have been accepted by the host and they may lead to the router congestion, dropping the QoS session, or having unacceptable performance for the QoS session. Such serious problem is unacceptable at all for the user that pays for the QoS session.

6.2 Delay

In general, the QBone model has less end-to-end delay than the proposed model and this is clear because it reserves a bandwidth equals to the input traffic peak rate (QBone premium service). However, the proposed model reserves a bandwidth equals to the input traffic average rate or higher, based on the network resources availability. So, as a conclusion, both models will not exceed the voice session end-to-end delay bounds (150-400 msec.) but the QBone model has less delay that may not be an effective issue for the all applications. Finally, the new proposed model has estimated the values for both delay and jitter that may face the VoIP packets through the path for different voice encoders. But the QBone model does not have any assumptions regarding such delay, and jitter values.

6.3 Buffer Consumption

QBone model depends on that the routers through the path have enough buffers for the incoming packets. So, the QBone model does not have any assumptions regarding the required buffer size values from the routers through the path to guarantee the service required-quality. But, the proposed model has the expected sizes for the required buffers through the path as illustrated in the previous three simulation scenarios for different voice encoders. So, the QBone model performance could be seriously degraded specially at high-loaded links case, in which the value for the required buffer-size is undefined. This may happen when there are enough bandwidth to support a voice session between the source and the destination, but the required buffers size at one hop or more is insufficient. By implementing the QBone model, the session performance may be very poor at such case, but the new proposed model will not even start the session before making sure that the required network resources including both the required bandwidth and the expected value for the buffer size through the path are sufficient.

6.4 Packet Loss

In general, the QBone model may not suffer at all from packet loss. For the proposed model, the packet loss percentage is almost zero at the high loaded links, and acceptable for the average and low load scenarios. So, as a conclusion, the QBone model in general has less packet loss than the proposed model. But, two points should be noticed. First, the proposed model packet loss is still below the voice threshold packet loss (10^{-6}). Second, for the QBone model, at high loaded links, may lose packets with effective percentage because of the insufficient amount of buffers at some hops through the path.

6.5 Bandwidth Consumptions

In general, the new proposed model consumes bandwidth much less than the QBone model. In this paper, it has been concluded that the proposed model saves bandwidth up to 69% more than the QBone model for G.723.1 (5.3), and up to 74% for G.723.1 (6.4). Also, it saves up to 54%, and 37% for both G.726, and G.711 than the QBone model respectively. Finally, it is important to conclude that these savings may decrease the high costs of the QBone model down only from 26% to 73% of the original costs. These cost’s reduction should encourage the Internet users to have voice session which has not only very low cost but also a guaranteed quality.

6.6 An Illustrated Example

If it is required to support maximum number of voice sessions between a site at Philadelphia, USA, and another site at Paris, France. Also, it is required to have end-to-end delay less than 400 ms by using G.723.1-(5.3) voice encoder (this is represents the user's required quality). The static setup shows that there is a route that has 5 hops between the source and the destination and the permitted bandwidth between them is 4.5Mbps. The expected loading percentage for the link will be around the 60% value before having the voice sessions.

Under the previous conditions, the QBone model could consume almost the available bandwidth to support up to around 28 voice sessions with undefined end-to-end delay value, which *may be* better than the new proposed model if there are enough buffers at the core routers, and undefined required buffer size value. But the new proposed model could propose two solutions, the first consumes almost the available bandwidth to support up to around 42 voice sessions with 172 msec. end-to-end delay value, and it needs around 27.2 kB of buffer size value. The second solution consumes almost to support 90 voice sessions with 370 msec. end-to-end delay value, and it needs around 66 kB of buffer size value. It is clear that, both the proposed solutions not only have expected values for the end-to-end delay (172 msec. & 370 msec.) and the required buffer size (27.2 kB & 66 kB) but also they decrease the voice-sessions costs by great values (down to % 66.7 for the first solution, and % 31.1 for the second solution of the QBone costs).

7. Conclusion

Nowadays, the Internet users are used to obtain the voice sessions as a free service, and it is obvious that this service is sometimes unreliable. Till now the QBone model did not succeed to convince the Internet users to switch to QoS world and enjoy having a reliable service and this is because of the high costs. So, the new proposed model has been implemented to move forward toward having a reliable service, may be sometimes with less quality than the QBone model but much more than the Best-effort service, and having a low cost which should attract the Internet users to have such new service. It is important to know that the new proposed model has decreased the QBone model costs down from 26% to 73% (for the studied encoders) to have a guaranteed voice session quality across the Internet. Also, the new proposed model not only has a quantified values for the voice session end-to-end delay, maximum jitter, buffer size, and packet loss but also has many classes for the required received voice quality based on the amount of cost that the user is welling to pay. From the illustrated example at section 6.6, the user may select between two proposed solutions, one with better quality and acceptable cost (66.7% the QBone model costs) and the other with less quality (still acceptable and much better than BE) and very low cost (31.1% of the QBone model costs). Actually, the proposed model is more complicated to be implemented than the QBone model, especially at the input traffic analysis phase. But, today, it is important to realize that there are many types of traffics with different characteristics and different requirements crossing the Internet, and a general solution for all of them could not be obtained. So, in this paper, the new model proposes a guaranteed service for voice sessions across the Internet only by implementing G.723.1 5.3 & 6.4, G.711, and G.726 voice encoders. To use more voice encoders, the same methodology should be followed but after analyzing the new voice encoders characteristics carefully.

8. References

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