Communication Network Evaluation - Exercise 7

Written Report

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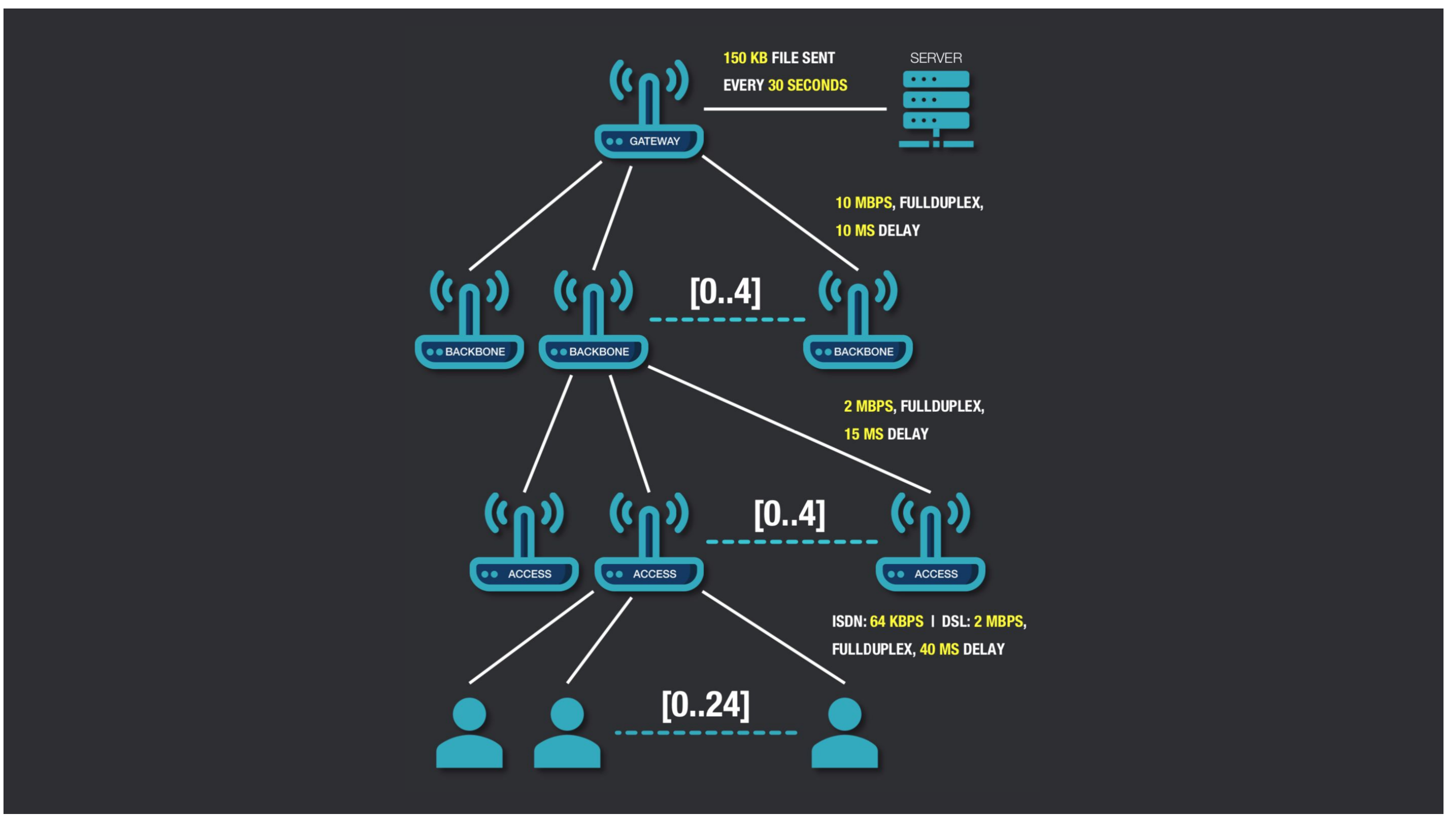
Objective

The objective in this exercise is to analyze the given network and determine the bit rate as well as the number of supported VoIP calls with the provided settings. Complete analysis consists of the following scenarios:

- To determine the average bit rate that a customer experiences when he is surfing the internet via ISDN.
- To determine the worst-case packet loss rate of Voice over IP service between any two users in the current network.
- To determine the average bit rate that a customer uses when he is online via the links upgraded to DSL.
- To determine the worst-case packet loss of a VoIP call between any two users over the upgraded links.

Network Description

The network consists of a gateway which is connected to the internet and has five directly connected backbone routers via full-duplex links. Each backbone router is connected via full-duplex links to five access routers. Moreover, twenty-five users can connect to each access router via an ISDN connection. Furthermore, all the links have fixed propagation delays and average bit rates.



Network Details

The given network can support 625 users and has the following properties:

- Full-duplex links between the gateways and backbone routers have propagation delay of 10ms and a bit rate of 10Mbps.
- Full-duplex links which connect backbone routers and access routers have propagation delay of 15ms and 2Mbps bit rate.
- ISDN connection has a propagation delay of 40ms and provides a bit rate of 64Kbps.
- Output buffer ports of all the routers have a capacity to hold at most 40 IP packets.
- The routers support only IP packets containing TCP segments with a Maximum Segment Size (MSS) of 1000 Bytes.

OMNeT++ Implementation

In order to implement the described network topology in the OMNeT++, we had to opt for arrays to accommodate such large network topology. First, we connected the gateway to the internet via an ideal full-duplex link to ensure that no bottleneck is created. Next, we used a “for loop” to connect the backbone routers one by one to the gateway router. In the next step, each of these backbone routers are connected to five subsequent access routers with the use of nested “for loops”. Finally, we needed to connect 25 users to each access router. We achieved this with the use of nested “for loops” where we looped through the access routers and used another loop to connect 25 users to each access router, hence, all 625 users got connected to the internet. The following snippet shows the implementation of the above-mentioned process of the network formation in OMNeT++.

```
connections:
        server.pppg++ <--> GatewayServer <--> gatewayRouter.pppg++; // server to gateway connection
// backbone to gateway connection
        for i=0..numBBrouterPerGateway-1 {
            backboneRouter[i].pppg++ <--> BackboneGateway <--> gatewayRouter.pppg++;
        }
// backbone to access connection
        for i=0..numBBrouterPerGateway-1, for j=0..numAroutersPerBBRouter-1 {
            backboneRouter[i].pppg++ <--> BackboneAccess <--> accessRouter[i*numAroutersPerBBRouter + j].pppg++;
        }
// access to user connection
        for i=0..(numBBrouterPerGateway*numAroutersPerBBRouter)-1, for j=0..numUsersPerAccRouter-1 {
            accessRouter[i].pppg++ <--> AccessUser <--> user[i*numUsersPerAccRouter + j].pppg++;
        }
}
```

In the first scenario, all the users are connected to the internet via ISDN connections and they only surf the internet. The following snippet shows the essential properties of the connection established between the users when they are surfing the internet. As mentioned in the exercise sheet, the upper bound on the congestion window is 100 packets. In order to implement this in the IDE, we used the receiver's advertised window because the sender chooses the minimum from the congestion window and the advertised window.

Allowed sender's window=min(receiver advertisement,congestion window)

```
# TCP Settings
**.tcp.typename = "Tcp"
**.tcpAlgorithmClass = "TcpNewReno"
**.tcp.mss = 1000 # MTU - IP header - TCP header = 1040B - 20B - 20B = 1000B
**.tcp.advertisedWindow = 100 * this.mss  #upper bounded at max CWND for 100 packets
**.tcp.windowScalingSupport = true # enabled since advertisedWindow > DEFINE_MAX

# General Server Settings
**.server.numApps = 1 # number of applications on the server
**.server.app[0].typename="TcpGenericServerApp" # Server application type
**.server.app[0].localPort = 10021 # TCP server listen port, default is 1000

# Router Settings
**.gatewayRouter.ppp[*].ppp.queue.typename = "DropTailQueue"
**.accessRouter[*].ppp[*].ppp.queue.typename = "DropTailQueue"
**.backboneRouter[*].ppp[*].ppp.queue.typename = "DropTailQueue"
**.gatewayRouter.ppp[*].ppp.queue.packetCapacity = 40 # Output port buffer at gateway router
**.accessRouter[*].ppp[*].ppp.queue.packetCapacity = 40 # Output port buffer at access router
**.backboneRouter[*].ppp[*].ppp.queue.packetCapacity = 40 # Output port buffer at backbone router

# TCP Configurations
**.user[*].numApps = 1 # number of applications on the user
**.user[*].app[0].typename = "TcpBasicClientApp" # Internet acces application type

**.user[*].app[0].connectAddress = "server" # destination address
**.user[*].app[0].connectPort = 10021 # destination port, default is 1000
**.user[*].app[0].replyLength = 1B * int(exponential(150000))
**.user[*].app[0].idleInterval = exponential(30s)
**.user[*].app[0].thinkTime = 0s
**.user[*].app[0].startTime = uniform(1s,30s)
```

For the stated reason, we have set the receiver's window to 100*MSS which is effectively equal to 100K Bytes. During simulations, we also noticed that if all the users start creating a session with the server at the same time, it produces a huge traffic burst at the server. In order to get rid of this burst, we have set the "startTime" for all the users to uniform distribution which ensures that not all of them begin creating their sessions at the same time. This randomness not only mimics the real behavior but also ensures that the network is not flooded with requests in the initial stage.

For the VoIP application, the maximum “end-to-end” delay of the packets should not be more than 200ms, and any packet with a delay larger than this threshold should be considered lost. Since UdpBasicApp does not provide such parameters to implement this requirement, we had to override some of the member functions. The following code snippet depicts the insertion of the intended behavior.

```
void VoIPApp::processPacket(Packet *pk) {
    simtime_t actualDelay = simTime() - pk->getCreationTime();

    EV << "Time between packet generation and time in application of receiver: " << actualDelay << endl;

    if (actualDelay.dbl() > acceptableDelay) {
        numLatePackets++;
        EV << "Packet is late. Its 'end-to-end' delay is " << actualDelay << endl;
        EV << "Total Number of packets that arrived past max acceptable delay is " << numLatePackets << endl;
        delete pk;
    }
    else {
        UdpBasicApp::processPacket(pk);
    }
}

void VoIPApp::finish() {
    totalLostPackets = this->numSent - this->numReceived;
    packetLossRate = static_cast<double>(totalLostPackets) / static_cast<double>(this->numSent);

    recordScalar("[VoIP] Total SENT Packets", this->numSent);
    recordScalar("[VoIP] Total RCVD Packets", this->numReceived);
    recordScalar("[VoIP] Total LOST Packets", totalLostPackets);
    recordScalar("[VoIP] Total LATE Packets", numLatePackets);
    recordScalar("[VoIP] AVG PK LOSS Rate", packetLossRate);

    UdpBasicApp::finish();
}
```

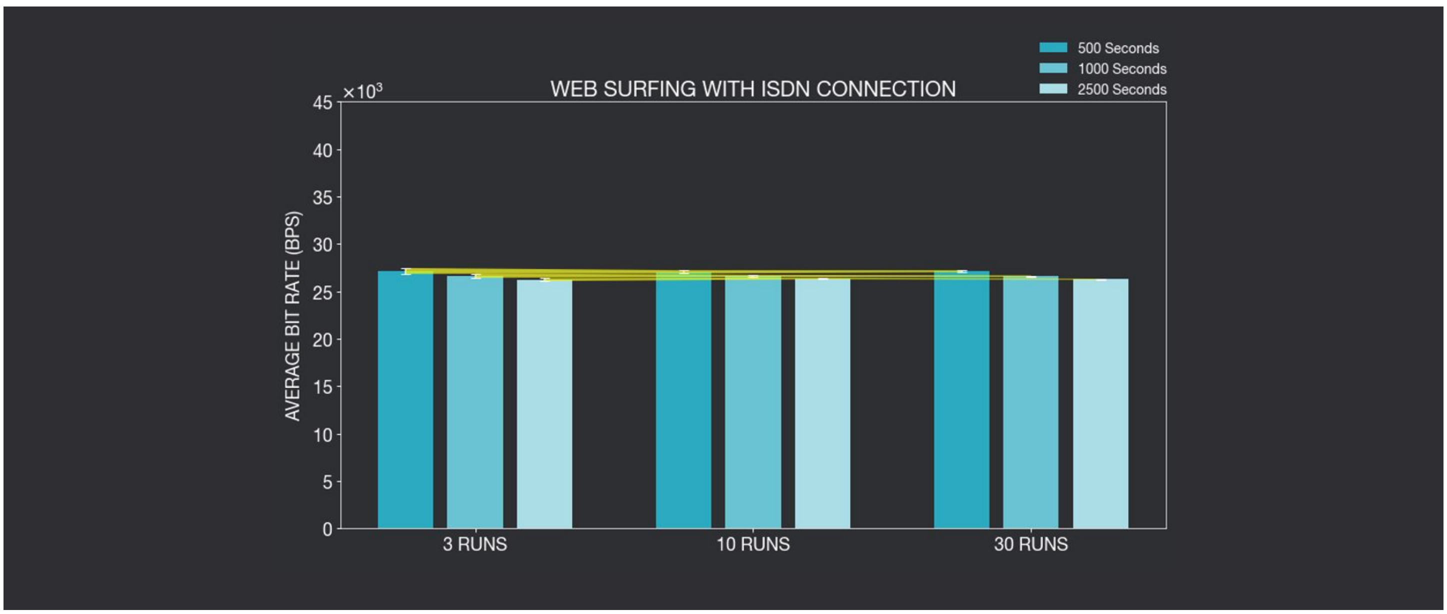
Exercise 7 Questions

QUESTION 1

What is the average bit rate that a customer uses when he is online via ISDN and surfing the internet as described in the user model?

We analyzed the throughput of the network when all the users are connected via ISDN connection to surf the internet. Users experience varying bit rates with an average around 26700 bps. We drew confidence intervals to determine a good estimate of the variation and sample mean for the observed throughput values.

The following plots depict the different scenarios depending upon the number of repetitions or sample size. It is evident that as we increase the sample size, we approach the true mean or population mean. For the first two bar graphs, we used Student's t-test scores as our sample size is less than 30 which is considered as the estimated boundary between t-test and z-test. For sample size 30, we had the option to either use t-test or z-test as both give almost the same results but we opted for the z-test.

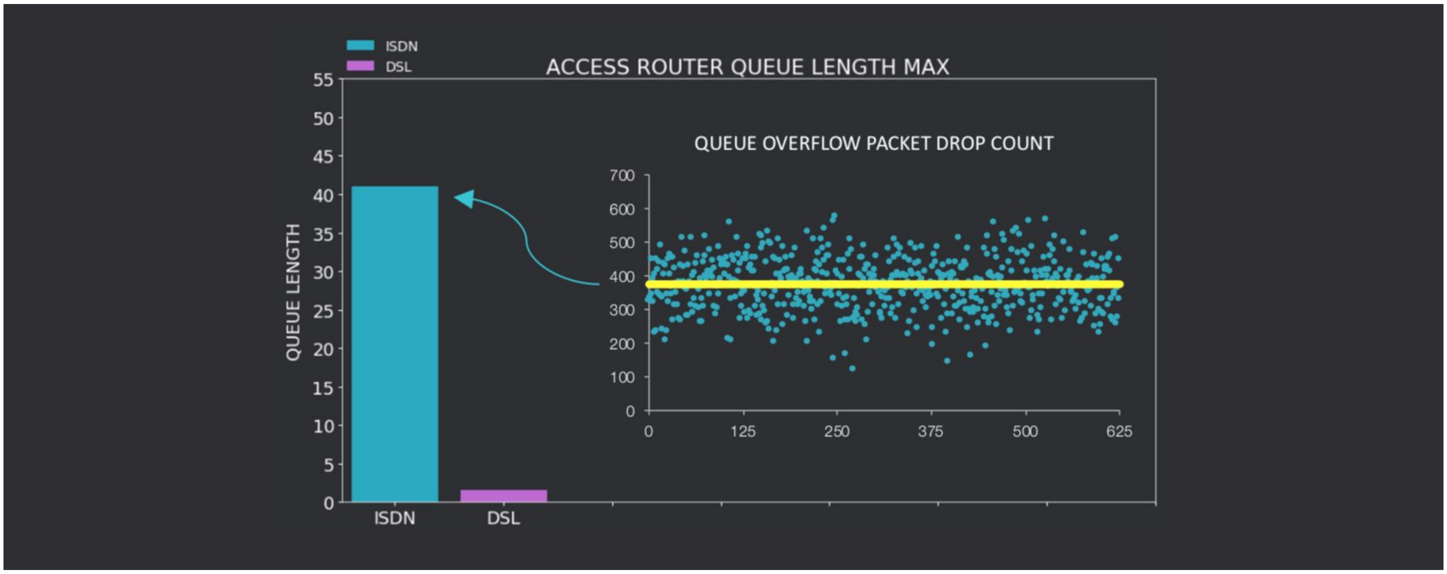


Confidence interval size shrinks as we move from left to right with the increasing number of samples which brings the sample mean closer to the population mean. Additionally, increasing the simulation time also assisted in decreasing the confidence interval size as it takes out the transient effects appearing at the start of the simulation. Furthermore, to perform steady-state analysis, we need to run the simulation for a longer period, which helps in eradicating the transient effects appearing at the initial stages.

All the customers are accessing the internet via ISDN connections having a bit rate of 64Kbps so we expected the users to experience a better average bit rate but surprisingly the actual throughput was a lot lower. Therefore, we decided to dig deeper to find out the actual reason for the overall lower bit rate.

We started from the gateway and looked at the output buffer for queues and found no evidence of queue formation or overflow. In the next step, we moved to the backbone routers and carefully examined the different parameters including the queue length and queue overflow count but everything appeared normal here as well. Finally, we went into a detailed analysis of the access routers and the findings were very surprising to the least.

We observed that the output buffers were experiencing regular overflow and queues remained full almost all the time which resulted in numerous loss packets. Since the congestion window is dynamic, it is halved every time a packet is lost in the network. On the contrary, with a correctly received packet, it grows at a rapid pace as well which results in more packets being pushed through the high data rate links between backbone and access router into a slow 64Kbps ISDN connection. Consequently, packets are buffered at the output ports of the routers and any packet arriving when the queues are full is dropped from the tail. Following plots show that in case of ISDN connection, access router queues remained full which resulted in regular drops of packets.



In conclusion, we observed that a lower bit rate experienced by the users when surfing the internet is dictated by the slow ISDN connection which resulted in queue formation at the access router and consequently packets were dropped from the tail.

QUESTION 2

What would be the worst-case packet loss rate of a Voice over IP service between any two users in the current network? Would it be acceptable?

With the current network setup the users are making VoIP calls and surfing the internet at the same time via ISDN connection. We have set up different scenarios to check the possibility if any VoIP calls are possible while all the users are downloading files from the internet.

We created various scenarios where:

- Customers communicate with each other within the same access router
- Customers communicate with user from other access router but within the same backbone router
- Customers try to communicate via different access routers in the different backbone router
- Some customers communicate with others via gateway which is the longest route in the network

Calculating the bandwidth requirements for VoIP calls:

Bandwidth calculations for VoIP calls take into account many different factors. These include, individual packet size and protocol headers thus can influence bandwidth consumption on the network. Following factors are included in the calculations of bandwidth consumption:

Total packet size = headers + checksum + voice payload size
Headers = (UDP + IP + RTP)
Packet rate = 50 PPS
Bandwidth = Total packet size * Packets per second (PPS)

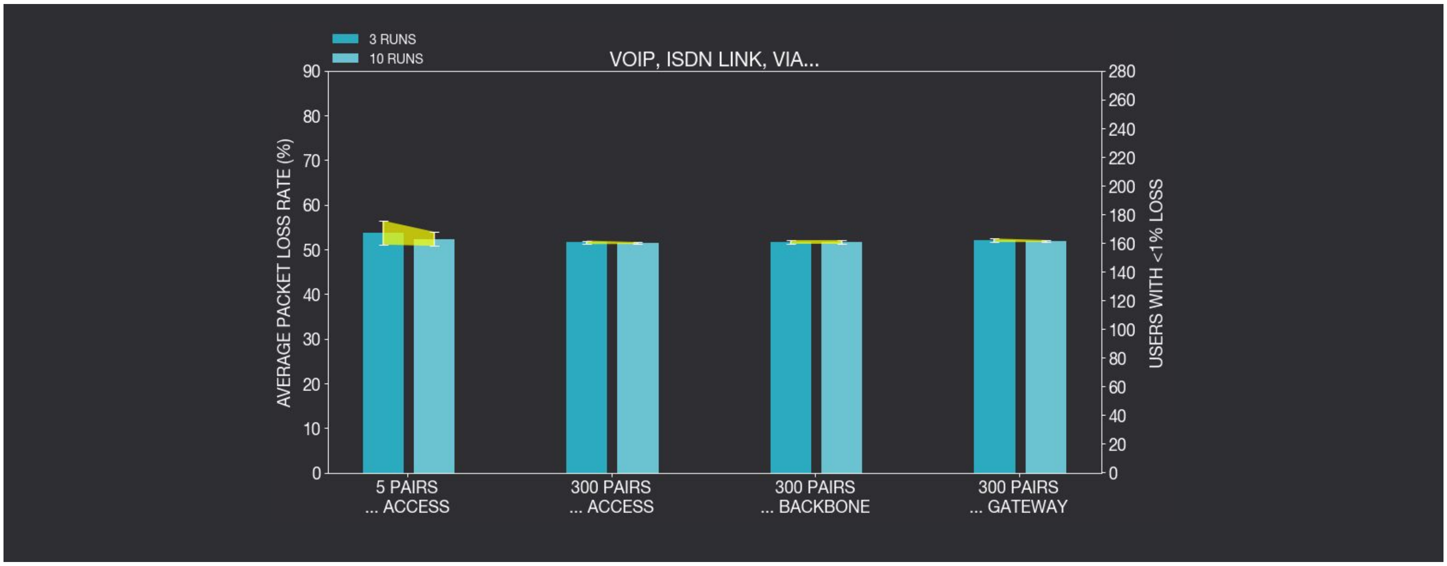
We can calculate the packet size as follows:

Total packet size = (8 + 20 + 12) bytes (headers) + 32 bytes (voice payload size) = 72 bytes
Total packet size in bits = 72 * 8 bits per byte = 576 bits
Bandwidth per call = 576 bits * 50 PPS = 28.8 Kbps

Since UDP does not have any congestion control algorithm thus VoIP services continue to generate UDP packets at a mentioned rate regardless of the situation of the network. As calculated above, a single VoIP call generates roughly 28.8Kbps traffic which results in filling the router queues very quickly as the link capacity in case of ISDN is merely 64Kbps and users are surfing the internet as well.

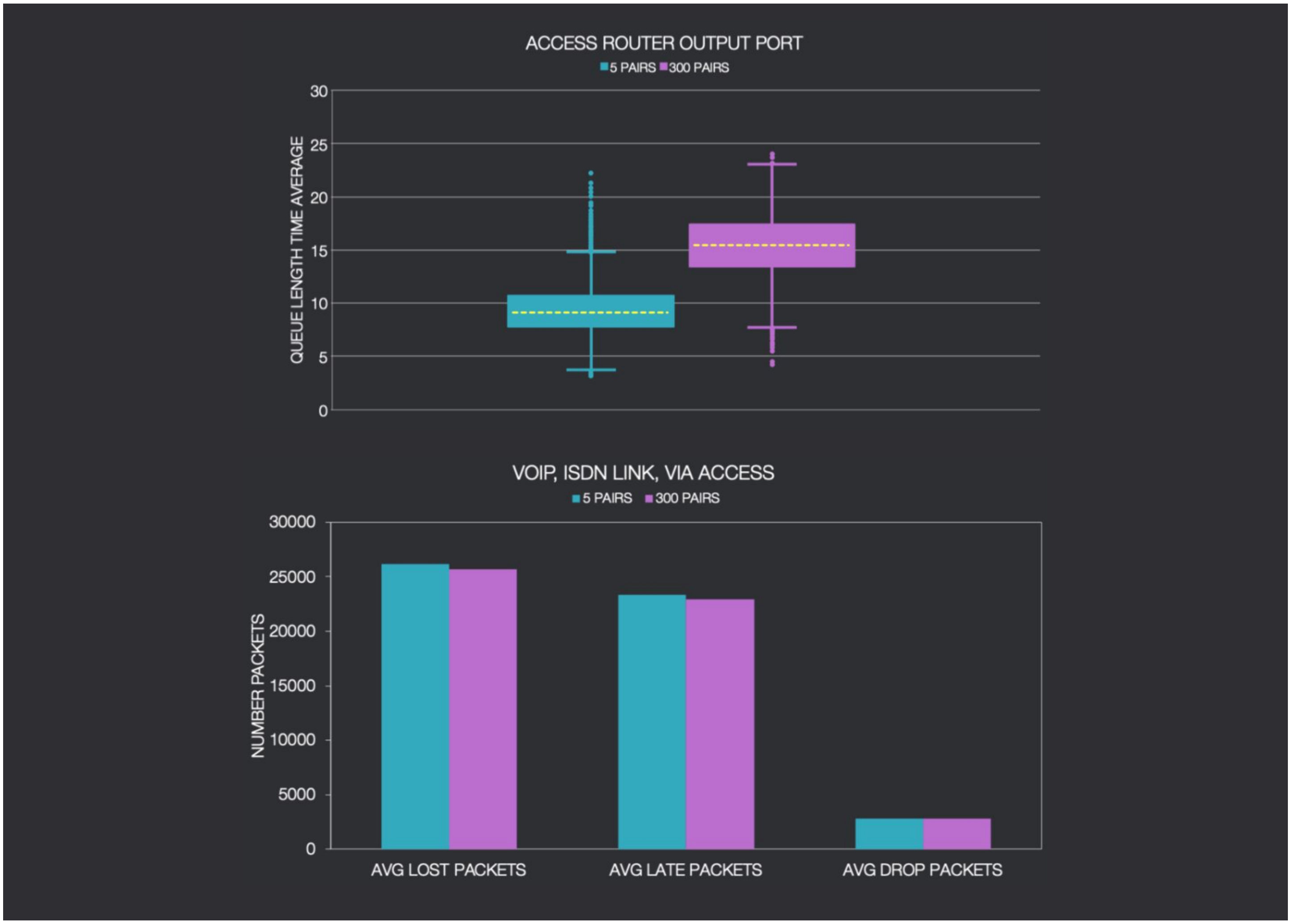
We created different scenarios for the users to make VoIP calls from different locations in the network topology when they are using the internet at the same time. We ran these scenarios with different simulation time settings and the following plot depicts the results obtained from settings mentioned above.

In this first scenario, all the users are communicating within the same access routers and it is evident from the plot that average loss rate is roughly 51.4% which means all of them experience the degraded VoIP service. The confidence interval for the loss rate is very small which is an indication that loss variation among the users is modest.



In the second scenario, users from different access routers but within the same backbone router are communicating via VoIP service and they face slightly higher loss rate in comparison to the first mentioned scenario because there is an additional delay caused by an extra hop between the users. The interesting observation was that the total number of lost packets had a slightly higher proportion of delayed packets in comparison to when the users were connecting via the same access router.

In order to investigate the reasons behind such a higher loss rate even within the same access router, we observed the average time a packet has to wait in the output queues before it is served in the access router. It is clearly evident from the plot that a packet has to wait for 9.3 seconds on average when 10 users are using VoIP services within the same access router and this rises further to 15.3 seconds on average when 300 VoIP calls are being made. This adds to the total delay a packet has to bear which violates the strict condition of the maximum allowed delay of 200ms. Further observations revealed that delay caused by the queuing and propagation delay added the higher proportion of lost packets to the total number of lost packets.



In the next scenario, when users are communicating via the backbone router, we observed a minor increase in the total loss percentage which was found to be 51.4%. It indicated that when users from different access routers but the same backbone router try to make VoIP calls, the extra hop does not add much to the overall loss percentage. We observed that an additional tiny fraction of the packets experience a delay greater than 200ms which is added by the backbone router. However, still the main contributors in the delay and thus the cause of the degraded VoIP service, were the access routers.

In the last scenario users connect via the gateway router, the packet loss percentage rose a bit further to 51.8%. As observed in the last case, some extra delay is added by the links between the additional hops as well as queues at the gateway and backbone routers which resulted in a small increase in the overall packet loss rate.

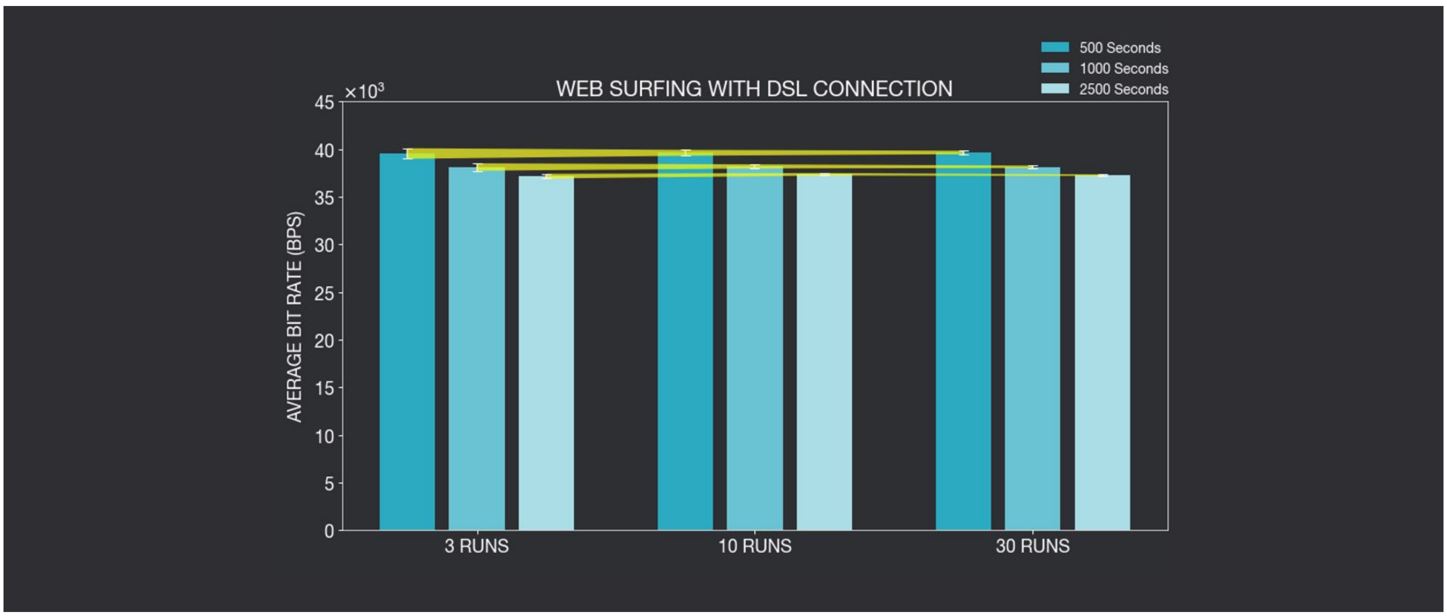
In conclusion, for the ISDN connection, no VoIP call is possible from anywhere in the network but out of all the positions of the users in the network, the worst case loss would be when they try to connect via the gateway router. This situation would be deemed unacceptable since the loss rate is very high and virtually none of the users can make VoIP calls.

QUESTION 3

What is the average bit rate that a customer experiences while surfing the internet if the access links have been upgraded to DSL?

We analyzed the throughput of the network when all the users are connected via the upgraded DSL links to surf the internet. Similar to the previous case with ISDN connection, users experienced different bit rates with an average around 38400 bit/sec. The variation gap in the bit rates observed was high with some users surfing at as low as 6762 bit/sec. On the other hand, throughput for some users went as high as 96440 bit/sec in the best case. To get a clear picture of the scenario, we decided to analyze the behavior using confidence intervals to get a good estimate of the mean value.

Confidence interval plots for average bitrate via upgraded DSL links are shown in the following plots.



Different scenarios were involved depending upon the number of repetitions or sample size and simulation time. Since we wanted to analyze the steady state behavior to eliminate the transient effects which appear at the beginning of the simulations so we involved different simulation time settings. The plots clearly show that throughput gradually settles as the simulation time is increased from 500 seconds to 2500 seconds and we eventually approach the true mean. It is also noticeable that as the number of repetitions or sample size is increased, the confidence interval gets smaller which indicates that the sample mean is approaching the true mean. Similar to the first scenario, Student's t-test scores were used in the first two bar graphs where sample size is less than 30 and for the sample size 30, we opted for z-test. Confidence interval size continues to shrink with increasing sample size as we move from left to right which indicates how close the sample mean is to the population mean.

Next, during throughput average analysis we expected the users to experience a better bitrate as all the links have been upgraded to DSL with bitrates of 2Mbps for the users to connect to the internet. The results showed that there was an overall increase in the average throughput value in comparison to the ISDN connection but the improvements were a lot lower than the expected values. In fact, the average bit rate improved by only 50% over DSL links with 2Mbps data rate as compared to ISDN connection which merely had 64Kbps data rate.

In order to investigate and search for the limiting factors, we once again started by analyzing the output buffers of the routers. In this case, access router output buffers were fine and had no overflow problem. Moving to the upper levels, we looked at the gateway router queues and found that they were filled to their capacity and found continuous queue overflow issues. Gateway router produces bottleneck situations so when output buffers are full, newly arriving packets are dropped resulting in the losses and thus the congestion window continues to fluctuate.

In the next step, we moved to the backbone routers and carefully examined the different parameters including the queue length and queue overflow count and found the same issue here as well. The output port buffers were experiencing regular overflow and similar to the gateway, queues remained full almost all the time. These two levels posed a bottleneck situation in case of DSL and thus limited the overall bit rate.

QUESTION 4

What would be the worst-case packet loss rate of a Voice over IP service between any two users when all the links have been upgraded to DSL? Would it be acceptable?

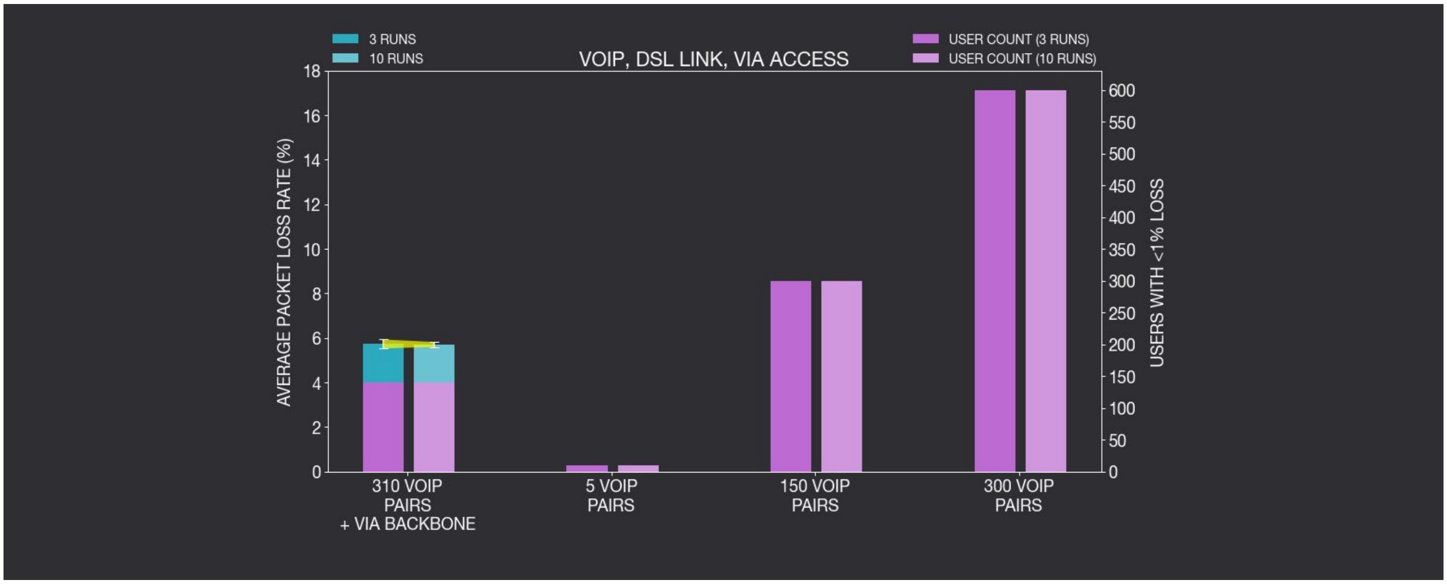
The ISDN connection is replaced by the upgraded DSL links with an increased bitrate of 2Mbps and the users can make VoIP calls and surf the internet simultaneously. In the same manner as with the ISDN connection, we investigated different scenarios to evaluate the possibility that the VoIP calls and internet surfing function at the same time. These scenarios include:

- Customers communicating via the same access router
- Customers communicating via other access router but within the same backbone router
- Customers communicating via different access and backbone routers
- Customers making VoIP calls via the gateway router

As mentioned earlier, UDP produces traffic at a consistent rate regardless of the network situation and does not have any congestion control algorithm thus it quickly floods the network with UDP packets. However, upgrading to DSL prevents the queue formation at the access routers thus almost all the packets are served immediately. To further observe the behavior, we evaluated the performance with the differing user positions in the network topology.

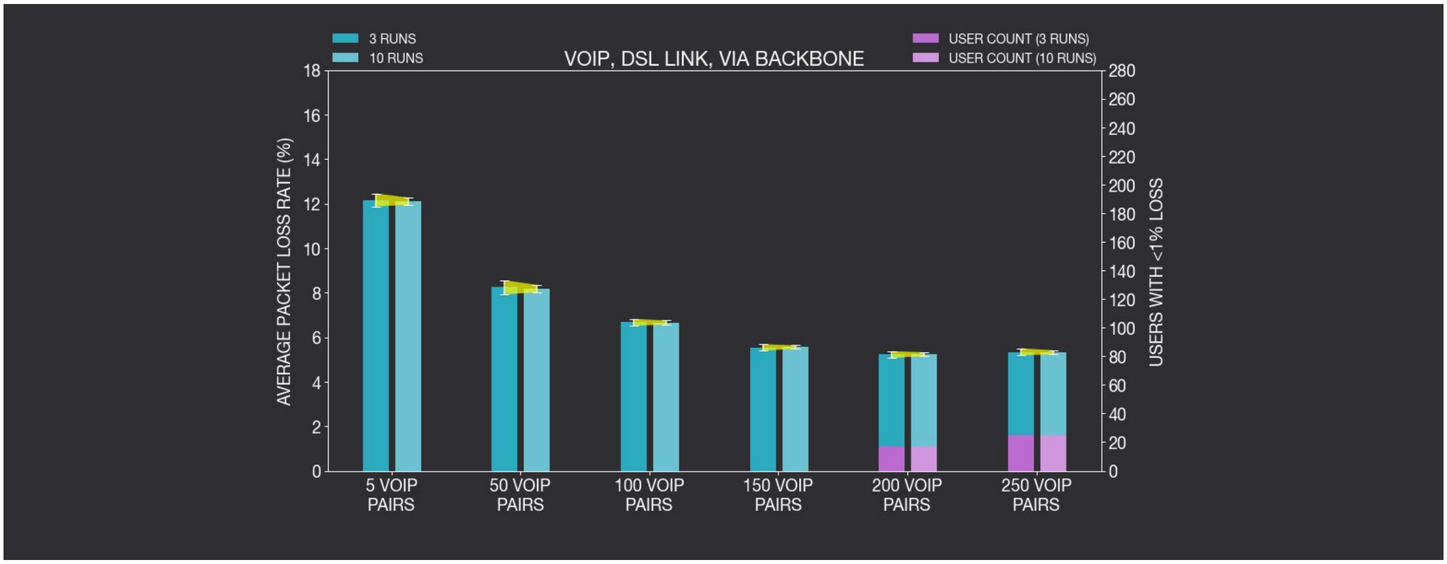
To begin with, we made all the users communicate with each other via the same access router. This situation produced the best results with a loss rate of nearly zero percent. With the upgraded links, no packets were delayed due to the queuing delay and queue sizes never went to their maximum limit thus all users can establish VoIP calls without any degradation at the access router level. This can also be verified from the plots where it shows negligible losses and all the users can communicate with each without any degradation.

We also ran a scenario with mixed geographical user positions where some of them communicate via the same access router while others doing so via a relatively longer route i.e through the backbone router. As expected, only a small proportion of users communicating via the backbone experienced a lower loss rate than the threshold value. We observed that 140 users can use VoIP services without any degradation in this particular setup with a loss rate falling the wide range beginning from 0% and peaking at 14.8%. However, the overall loss rate remained relatively lower at 5.7%. Both scenarios are plotted below.



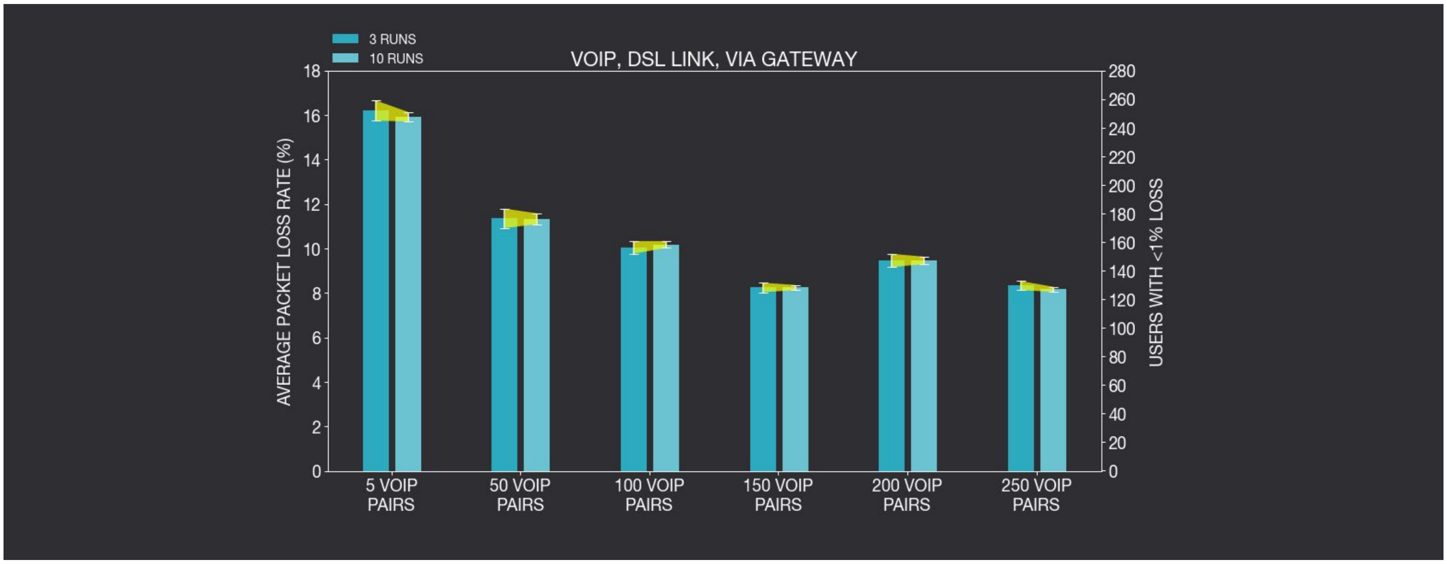
Next, we investigated at the backbone router level where users from different access routers connect with each to utilize the VoIP services. As mentioned earlier, for the upgraded links queues are formed at the backbone router and this point acts as a bottleneck even when users are only surfing the internet, addition of UDP traffic further adds to the trouble. Due to queue formation in the output buffers of the backbone routers, packets are either dropped or have to wait for very long in the queues which results in the delayed packets delivered to the destination. Since, any packet arriving later than 200ms is considered to be causing the degradation in the VoIP services, users would not be able to utilize the VoIP services.

At this point, we carefully analyzed the average queuing time at access and backbone routers and found out that access routers do not cause any trouble after the links have been upgraded to DSL but packets have to wait for a relatively longer time in backbone routers which is way greater than the acceptable delay. However, an increase in the number of VoIP calls in the network, caused the flooding of the UDP packets and they started to fill the queues at the backbone routers. This increase in the number of VoIP calls resulted in some of the users experiencing less than 1% packet loss rate and overall packet loss rate started to decrease gradually. By running the simulations several times, we established that at most twenty-five users can communicate via VoIP through backbone routers with the minimum overall loss rate. If we increase the number of VoIP user pairs beyond 250, the overall loss rate starts to increase but the number of successful calls remains the same. Hence, 250 user pairs gave us the sweet spot with minimum loss rate and the maximum number of users able to communicate via backbone routers.



The plots show that the best case scenario presented an average loss rate of 5.3% with a wide range of variation from 0.6% to 15.8%. On the contrary, the worst case had a relatively smaller range between 9.8% to 14.8% with an average loss rate of 12.1%.

As a final step, we increased the geographical distance to maximum which meant that users would be communicating via the gateway. Establishing VoIP calls via gateway only worsened the situation by adding more traffic and thus causing even larger queue delays at the output ports of the gateway. In this scenario, packets have to go through the backbone routers which are the first bottleneck point and then face further delay at the second bottleneck point formed at the gateway. In addition, there is some propagation delay from the links as well thus most of the packets arrived late at the destination. As mentioned earlier, any packet arriving later than 200ms from the time of creation is deemed lost thus all the users experienced degraded VoIP services via gateway router. However, by increasing the number of VoIP calls, the overall loss rate showed a decreasing trend.



As it is evident from the plot, overall average loss rate was as high as 15.9% with variation ranging from 13.9% to 18.5%. On the other hand, with the maximum number of VoIP calls going on in the network, an average loss rate observed was 8.2% with low and high at 1.1% and 18.7% respectively.

In conclusion, after rigorous testing we found that the worst case scenario occurs between the users at the largest geographical distance i.e via the gateway router and virtually no VoIP call can be established without degradation.