Exercise 7

Communication Network Evaluation

For each team, please

- Complete the task within the next two weeks
- Prepare a short presentation (max. 20 min) for the session in two weeks
- Recommended: answer all questions in a written report. This is similar to the final task for CNII and therefore a good way to practice and to get some feedback.

- Do not underestimate the workload! This exercise is way more complex than the previous ones.

1 Introduction

A network operator contacted your team and asked for consultancy. Currently the operator is running a small Internet Protocol (IP) network with simple data services using Integrated Services Digital Network (ISDN) for dial up. Besides the data services he would like to offer a simple Voice over IP (VoIP) service to selected customers. The simple voice service should enable customers to make phone calls to other customers within the operator's network. Further, the operator thinks about upgrading all ISDN dial up nodes to DSL (Digital Subscriber Line).

Your task is to support the operator's decision process by giving him the performance figures he requires. The next Sections will first describe in detail the operator's current network and infrastructure as well as his service requirements. Finally, the operator has listed all information that he would like to have from you. A good consultant would of course make further suggestions and present more investigations than was asked for.

2 Current Network

The topology of the network is as follows: A gateway is connected to the Internet. 5 backbone routers are connected directly to the gateway via full-duplex links with a fixed propagation delay of 10 ms and a bit rate of 10 Mbps. Each backbone router is directly connected to 5 access routers via a full-duplex link with a fixed propagation delay of 15 ms and a bit rate of 2 Mbps. To each access router 25 users can connect via an ISDN connection which has a propagation delay of 40 ms and a bit rate of 64 kbps.

Each router is output port buffered and the buffer has a maximum capacity of 40 IP packets, regardless of their size.

The routers currently support only IP packets containing TCP segments with a Maximum Segment Size (MSS) of 1000 Byte.

3 Current Services

The operator is currently only offering simple best effort data services. Recent measurements in his network have shown that all customers are currently only surfing the Internet. Thereby they all follow a simple pattern: Online users select content from the Internet and download it. The size of the content is random. Measurements have shown that the file size is exponentially distributed with a mean value of 150 kB. After the users have downloaded a file they pause (possibly reading the content they have downloaded and looking for something new to download) for a random time and then start the next file transfer. The pause time is also exponentially distributed with a mean value of 30 s.

The measurements have also shown that all customers download their data via TCP NewReno connections and have an upper bound on the congestion window of 100 packets.

4 Envisioned Changes

4.1 Voice Service

The operator wants to enable customers to make phone calls with each other. However, simultaneous use of the access link for data and voice is foreseen. The customers use a simple voice codec: packets are generated at a constant packet rate of 50 packets per second and a voice packet consists of 32 Bytes of payload plus protocol headers (Real-time Transport Protocol (RTP) over User Datagram Protocol (UDP) over Internet Protocol version 4 (IPv4) is used).

VoIP is an interactive real-time service, which means that users are willing to accept only a limited end-to-end delay. The maximum acceptable end-to-end delay is 200 ms, i. e. the time when the voice data is required in the application of the receiver is at maximum 200 ms after the packet is generated at the sender. If a packet arrives too late it will be considered as lost.

The voice codec can cope with few errors. Error rates below 1 % do not lead to hearable service degradation. However, if the error rate exceeds this threshold, the customers feel disturbed and will not be willing to pay for the service. The average error rate over the whole session time can be used here for simplicity, even though usually error rates averaged over smaller periods of time would need to be considered.

Decide on a reasonable number of active VoIP clients. Think about the influence of their position within the network on the link utilization.

4.2 DSL Access

The access via ISDN is slow and not state of the art. Hence, the operator would like to upgrade all access links to DSL connections. While this will not lead to a reduction of the propagation time (it will still be 40 ms), the bit rate on these links could be increased to 2 Mbps. It is not expected that the user behavior will change.

5 Operator's Information Requirements

The operator wants at least the following information from you:

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?
2. What would be the worst case packet loss rate of a Voice over IP service between any two users in the current network? Is that sufficient?
3. What is the average bit rate that a customer would use surfing the Internet if the access links have been upgraded to DSL?
4. What would be the worst case packet loss rate of a Voice over IP service between any two users in case all access links have been upgraded to DSL? Would that be sufficient?

6 Hints

This section gives you a few hints for the OMNeT++ simulator with the INET framework.

- OMNeT++ has already some built-in functionality for processing data (e.g. calculate confidence intervals) generated by the simulations, but it is also possible to use other tools, e. g. Matlab, grep/awk/sed, spreadsheet software etc. OMNeT++ can export the generated data

- In your project properties you have to set up the reference to the INET framework, otherwise you will not be able to use the INET framework models.

- The INET framework and your project must be set to the same build configuration (i. e. gcc-debug or gcc-release). The setting can be changed using the project's context menu → Build Configurations → Set Active.

- Using gcc-release in connection with command line interface instead of the standard Tcl/Tk GUI can greatly improve the performance of the simulation (can be changed in Run → Run Configurations).

- Switching off selected unneeded logging functionality (e. g. `**.vector-recording=false`) can also speed up the simulation.

- To model the VoIP client which tolerates a maximum end-to-end delay of 200 ms, you can derive a subclass from UDPBasicApp. The NED definition for this would look like this:

```
import inet.applications.udpapp.UDPBasicApp;
simple VoIPApp extends UDPBasicApp
{
        @class(VoIPApp);        // this means that we want to use our VoIPApp C++ class,
                                // otherwise UDPBasicApp class would be used by default
}
```

- The behavior of the subclass is programmed in C++ (i. e. create a VoIPApp.cc file and overwrite the appropriate functions from UDPBasicApp)

- In your derived class, you can add logging for sent and received packets (`recordScalar()` or use the signal mechanism)

- By default, OMNeT++ uses the run number as part of the seed for its random number generator, i. e. when a run with the same number is executed again, it will give the same results.

- If you calculate mean values, it is a must to calculate confidence intervals. Without these, mean values do not tell you anything. Hence, you should run the simulations several times with different seeds (e. g. by using the `repeat` parameter in the ini file)

- On computers with more than one core and/or Hyper-Threading, multiple simulation runs can be executed in parallel. This can be specified in the run configurations dialog.