Simulation of Simple Integrated Media Access (SIMA) with a Model for the User Behavior

COST 257

Kaisa Kyläkoski
Laboratory of Telecommunications Technology
Helsinki University of Technology

December 1998

Table of Contents

Foreword

This work was done within the project COST 257, funded by Tekes, Nokia Telecommunications and Sonera.

The course of the work was guided by discussions with Kalevi Kilkki, Jussi Ruutu and Jussi Penttinen from the Nokia Research Center and supervised by Professor Jorma Virtamo at the Laboratory of Telecommunications Technology, Helsinki University of Technology.

Goal

The purpose of this work is to examine the performance of Simple Integrated Media Access (SIMA) with simulation. Specifically of interest is to see how the users influence the dynamics of the system.

Integrated services

Whereas previously telecommunication and data communication services used their own dedicated networks today it is increasingly popular to increase network efficiency by using just one network. Another process is the way in which one network – the Internet – is used by more and more different applications.

Different applications can use the same network if all data is in the form required by network protocol. Network protocol and network's other properties determine how well an application functions in an integrated network environment. Applications that could use similar network can have different service requirements in terms of e.g. reliability, delay or delay variation.

In the Internet, for which SIMA has been designed, currently all packets are equal and the Quality of Service (QoS) is mostly dependent on the congestion. This means that no one can be sure of the QoS they'll receive and nor can they influence it in a significant way. Due to current flat-rate pricing policy there is no cost in sending more traffic to the network and so there is no mechanism to stop the Internet congestion from getting worse.

The generally offered solution to Internet congestion is differential pricing. This way one could control the amount of traffic and make use of the fact that the users themselves have both different needs, willingness to pay and expectations for the Quality of Service they receive. One way of implementing differential pricing is to create service classes that have guaranteed properties. Simple Integrated Media Access has another approach and this will be described in the following section.

SIMA[1]

The primary idea of the SIMA service is to maximize the exploitation of network resources with a simple control scheme while keeping the ratios of QoS levels offered to different flows unchanged under changeable traffic conditions. The maximization is based on three key features: all flows with different QoS requirements share the total capacity of every link, the network attempts to avoid any unnecessary packet discarding, and flow (or call) level blocking can be totally avoided. The approximate constancy of QoS ratios and simplicity are achieved by using 8 priority levels which make possible a fair packet discarding scheme inside the network without keeping track on the traffic of every flow.

[1] http://www-nrc.nokia.com/sima/

When the network operator offers the SIMA service, a customer first pays for some Nominal Bit Rate (NBR, kbit/s) and then he/she can trade the speed for QoS. The user is entitled to him/herself determine whether the flow is a real-time (rt) or non-real-time (nrt) one. The quality of the flow depends on two issues: the NBR to actual bit rate ratio, and total load in the network. The fairness of the SIMA service is based on the fact that all flows with the same actual bit rate to NBR ratio perceive similar QoS.

If the user sends traffic by using a constant bit rate, the SIMA service offers 7 different quality levels (for variable bit rate traffic the levels are less distinct but basically the same).

7 = reserved for non-SIMA services with resource reservation
6 = excellent quality: negligible packet loss ratio
5 = high quality: packet losses only during exceptional traffic peaks
4 = good quality: small packet loss ratio even during busy hour
3 = moderate quality: usually small packet loss ratio except during busy hours
2 = satisfactory quality: from time to time very high packet loss ratio
1 = suitable for best-effort traffic during busy hour
0 = unusable during busy hour, but suitable for best-effort traffic during non-busy hours

Table 1 Priority levels of SIMA

The implementation of the SIMA service consists of two main parts: access nodes and core network nodes. The traffic measurement of every flow is performed at access nodes whereas at the core network nodes the traffic control functions do not need to know anything about the properties of separate flows.

Let us suppose that there is an IP flow (i) at an access node. A nominal bit rate, NBR_i, is associated with the flow. At the user/network interface there is a measuring device which measures the momentary bit rate of the flow at the arrival of the j:th packet (or cell). This rate is denoted by $MBR_{i,j}$. The device gives every packet (or cell) a priority, $PL_{i,j}$, based on the $MBR_{i,j}$ to NBR_i ratio:

$$x = 4.5 - \frac{\ln\left(\frac{MBR_{i,j}}{NBR_i}\right)}{\ln(2)}$$

$$PL_{i,j} = \begin{cases} 6 & if \quad x \geq 6 \\ Int(x) & if \quad 0 < x < 6 \\ 0 & if \quad x \leq 0 \end{cases} \tag{1}$$

where Int(x) is the integer part of x.

Since the bit rate of every connection may change significantly in several time scales, the operator must apply an averaging measuring principle to determine the instantaneous cell rate of each connection. If we suppose that the moving average is calculated at every time slot, the measured load generated by a connection (i) at the instant of transmission of j:th cell is:

$$\rho_{i,j} = (1-\alpha)^{N_{i,j}} \rho_{i,j-1} + \alpha \tag{2}$$

where $N_{i,j}$ is the distance between j:th and (j-1):th cells in time slots and α is a parameter which defines the time scale of measurement.

The proper value for parameter α depends on the buffer capacity reserved for the service class used by the connection. With real-time services (with small delay variation) the buffer should be small, and thus the value of α must be quite high. On the contrary, when using a non-real-time service the user may want to send bursts of cells without high cell loss ratio. As a consequence α must be much smaller (or the averaging period should be much longer).

The key issue in the implementation of the SIMA service in a high capacity core network is the packet or cell discarding system before the actual buffering. At any instant there is an accepted level of priority (PL_a): if an incoming packet or cell has the same or higher priority, it is accepted, otherwise it is discarded. The calculation of PL_a is based on the buffer occupancy levels of the real-time buffer (M_rt) and non-real-time buffer (M_nrt).

All the packets or cells which have been accepted in the scheduling unit are placed either in the real-time or non-real-time buffer. Both buffers may apply the First In First Out (FIFO) principle. All packets (or cells) in the real-time buffer shall be transmitted before any packet (or cell) in the non-real-time buffer.

Implementation

Ideally the simulation would have a great number of different traffic sources (different applications), and both the traffic sources and the protocols would be represented in detail. The users could also be modeled in detail and with fair amount of complexity. In order to simulate a sensible length of time simplification is necessary.

The most important simplification is that the simulation model has discrete time. The discretization period was chosen as 10 ms. Discrete time means that the function of the relevant protocol (TCP) can only be approximated. Similarly the model uses an approximation of the SIMA.

There is no generally accepted reasonably valid models of the various kinds of traffic sources that exist in the Internet. Some (but not all) of the salient characteristics of such traffic have been described in abstract terms. For sources that use adaptive congestion control the traffic is always dependent on the traffic present in the network at an earlier instant.

The applications chosen for this simulation model are WWW browser, IP phone and Video phone. These all produce a notable amount of traffic compared to for instance email and are already widely used or will probably be so in the near future. The phone applications represent real-time applications while WWW traffic is non-real time and uses TCP. The traffic models used to descibe the traffic produced by these models will be described in the following sections.

Since virtually all degrees of congestion, including none at all, are observed with non-negligible probability in the Internet, it is necessary to simulate a number of configurations that represent different amounts of traffic.[2]

Round-Trip-Time (RTT), delays

Round-Trip-Times in the Internet are widely varying. As a rule of thumb the RTTs within Finland vary between 10 and 50ms and RTTs for Finland – USA – Finland between 100 and 250ms. A simulation can either model the variation or assume constant values.

The choice of RTT for the model and the presence/absence of delays affect most the function of TCP which in reality uses a *measured* RTT as a *basis* of its timeout value.

For this model the following combination is used. The packets move in the network in a deterministic manner (propagation delay) and RTTs are multiples of 10ms. But at each SIMA node a queueing delay is added to the packet. According to SIMA specification the priority level accepted PL_a is dependent on the buffer occupancy and as this is proportional to the average delay for a packet, in this model the converse is assumed to be true. When PL_a is known it can be linearly transformed to (combined) buffer occupancy. The linear transformation needs to be such that buffer occupancy of 1 (full buffer) corresponds to PL_a greater than 7 (or 6) and buffer occupancy of 0 to PL_a = 0. Since all the simulations that were carried out involved only one SIMA node between transmitter and receiver the combined effects of delays in the nodes were not studied.

SIMA node

(SIMA node referes to the scheduling unit)

Based on the buffer occupancy SIMA node determines the lowest accepted priority level. Packets (cells) that have lower priorities are discarded. The accepted packets (cells) are buffered according to whether they are real-time or non-real-time. All real-time packets (cells) are served before non-real-time packets (cells).

The approximation of SIMA node places all cells that arrive during one unit of time (the discretization period) to a priority queue and serves as many from the top as its capacity allows. The priority of the last packet (cell) accepted is the PL_a.

User reaction

Users can react to unsatisfactory QoS by either terminating the connection or by choosing a new higher NBR. The first is the reaction of a user with low willingness to pay and the latter the reaction of a user with high willingness to pay. (Reducing the transmission speed would give the traffic a higher priority, but transmission speed is an important part of the Quality of Service.)

[2] Vern Paxson and Sally Floyd: Why We don't Know How To Simulate The Internet. Proceedings of the 1997 Winter Simulation Conference, Atlanta 1997.

The NBR chosen by the user represents his/her willingness to pay. As a simplification one might assume that the willingness to pay is combined with higher quality expectations. It should, however, be remembered that the user might have to pay for instance double the current price in order to receive a higher priority class and that there is limited number of classes, i.e. after receiving the highest class paying more does not improve the situation.

The issue of QoS of the applications is presented in following sections.

WWW user (TCP)

Characterization of WWW usage is an ongoing project.[3] So far the research results for individual user behavior are rare and concentrate on the usefulness of caching or explaining the observed self-similarity of WWW traffic. The following results exist.
File sizes (requested and from entire Web): Heavy tailed (Pareto) with average HTML size of 4–6 KB and median of 2 KB, images have an average size of 14 KB[4]
Reading time per page: Heavy tailed distribution with an average 30 seconds, median of 7 seconds, and standard deviation of 100 seconds [5]
Session time outs: 25 minutes, with mean time of 9 minutes [5]

WWW traffic is a good example of traffic that uses the Transport Control Protocol (TCP). TCP is used for both flow control and congestion control. Flow control ensures (among other things) that all packets transmitted are received. Congestion control entails the dynamic change of transmission rate so that the TCP source avoids sending traffic when it is probable that it would not reach the receiver within the timeout period.

In reality (the presently used) HTTP protocol opens a separate TCP connection for each file.

Analytical solutions for steady state TCP throughput as function of dropped packets are not applicable to this model since it is the dynamical behavior that is of interest. Modeling the protocol behavior as such, in detail, is not a feasible solution either because the information with discretized time is not detailed enough. It is worth

---

[3] James E. Pitkow. Summary of WWW charcterizations. WWW7

[4] Carlos R. Cunha, Azer Bestravos and Mark E. Crovella. *Characteristics of WWW Client-based Traces*, Technical Report BU-CS-95-010, Boston University, July 1995 ;T. Bray. *Measuring the Web.* The World Wide Web Journal 1(3), 1996; A. Woodruff, P. Aoki, E. Brewer, P. Gauthier and L. Rowe. *An investigation of documents from the World Wide Web,* The World Wide Web Journal, 1(3), 1996.; M. Arlitt and C. L. Williamson. *Web server workload characterization: The search for invariants.* In Proceedings of the ACM Conference on Measurement and Modeling of Computer Systems, April 1996. pp. 126 - 137.

[5] L.D. Catledge and J.E. Pitkow. *Characterizing browsing strategies in the World Wide Web.* Computer Networks and ISDN Systems 26(6), 1995; Carlos R. Cunha, Azer Bestravos and Mark E. Crovella. *Characteristics of WWW Client-based Traces*, Technical Report BU-CS-95-010, Boston University, July 1995

noting that various implementations of TCP have been found to exhibit wide variation.[6]

This part of the model could have been partioned into two parts, a WWW user and a WWW server. However, a necessary simplification would have been to assume that the requests (representing small amounts of data) would arrive without loss. Thus, for the sake of simplicity the model is one block that creates the traffic requests, transmits the traffic and monitors the quality of transmission. Despite this duality the block represents a WWW user not a WWW server. This means the supposition that the user that is receiving, not sending, traffic is in fact paying for the transmission by some method of reversal of charges. What is lost is the information of the delays for the requests, which could in reality have a significant influence on QoS.

The traffic to be sent with TCP is generated with a bursty traffic source. A burst corresponds to a "click" and consists of the pulses that correspond to the documents requested. The size of the documents is Pareto distributed. This corresponds to research results[7] and also makes it possible to match both mean and median of recent (spring 1998) measurements by Markus Peuhkuri of Funet WWW traffic. The other parameters (average time between clicks, number of documents per request) were chosen without support from research results.

The length of sessions is not addressed. Once the session begins it will go on as long as the QoS is sufficient. The justification for this choice is that the most important point of interest is the QoS for the real-time services while there are TCP connection on. There is also little information about the length of actual sessions but it seems that lengths that exceed simulated time (one hour) are not an uncommon occurrences.

At the level of accuracy of this simulation model this rather simple model is sufficient and the actual differences between users and their usage patterns are not addressed.

Since with the discretized time it is impossible to achieve a model that includes all of TCP characteristics the main goal is that the source adapts to the capacity available. The model uses the basic linear growth and the exponential reduction of the transmission window.

Momentary bit rate is measured as a moving average of the past transmissions.

The QoS for the TCP connection is divided into two parts. First, the delay attached to the packets at SIMA nodes is examined. Secondly the length of time for the transmission of files requested by one "click" is compared to a tolerance value. When tolerance values are not exceeded, the parameter value grows. When tolerance values for either are exceeded, a parameter value is increased and its value is compared to an additional tolerance value. If this tolerance value is exceeded the session is terminated for a random period.

---

[6] Vern Paxson: Automated Packet Trace Analysis of TCP Implementations

[7] James E. Pitkow. Summary of WWW charcterizations. WWW7

IP phone (Real-time source)

IP phone is an example of a real-time application. Quality of an IP phone connection depends on its implementation and the congestion in the network. Congestion causes both delay and lost packets. If data arrives after its appropriate playback time, either because it was delayed in the network or because it was dropped and subsequently retransmitted, it is essentially useless. Proper implementation can, however lessen the effect of these.

The voice source is coded so that the traffic is continuous bit- and packet-rate. The packet length is dependent of the codec used, sampling frequency, frame length and the number of coded frames in one packet. Frame length indicates the number of samples per coded frame. For example if we are sampling 8000 samples/s and the frame length is 20 ms we have 160 samples in a frame. The packetization interval is constant and typically 1-2 frame lengths, i.e. 20-40 ms.[8]

Mean opinion testing (MOS) provides a means of evaluating the subjective performance of voice and/or video transmission equipment. The MOS scores are derived by averaging the responses of a large number of listeners.

5	excellent quality	no noticeable impairments
4	good quality	only very slight impairments
3	fair quality noticeable but acceptable impairments	
2	poor quality	strong impairments
1	bad quality	highly degraded speech

Table 2 The five point scale of the MOS test of speech quality

Packet loss is a serious impairment of the Internet of today on realtime communication. Packet loss as high as 10% to 30% is not uncommon. For realtime voice over IP a packet is considered lost when its playback point is missed. In literature speech clipping of 16 to 64 ms has been noted to cause noticeable quality degradation unless the percentage of speech clipped is under 2%. For higher percentages of speech clipped at the same clip lengths the quality degrades, but intelligibility is still maintained. For speech clipping durations of >64 ms the quality of the speech is degraded seriously and intelligibility is reduced [9]. The effect of packet losses on the perceived quality of speech is dependent of the packet size. Listening tests have suggested[10] that in the sense of robustness to packet losses the optimal packet length is between 16-32ms. The duration makes a tradeoff between the number of speech losses per second and the probability of totally losing a phoneme.

For bit error ratios there exist the following results. If the error ratio in the transmission of voice on a digital 64kbit/s connection is 10^{-6} or less, during an

[8] Tomi Yletyinen, The Quality of Voice over IP, Master's Thesis 1998
[9] Gruber, J., Strawczynski, L., "Subjective Effects of Variable Delay and Speech Clipping in Dynamically Managed Voice Systems", IEEE Transactions on Communications, vol.COM-33,no.8, pp.801-809, Aug.1985.
[10] Jayant, N.,Christensen, S., "Effects of Packet Losses in Waveform Coded Speech and Improvements Due to an Odd-Even Sample-Interpolation", IEEE Transactions on Communications, Vol. COM-29, No.2, Feb. 1981.

arbitrary period of time, the recipient notices no degradation of quality. If the error ratio is 10^{-5}, the degradation of quality is negligible; at an error ratio of 10^{-4} the disturbance is considered somewhat irritating; and at a bit error ratio of 10^{-3} the degradation of quality is severe. In practice, bit errors normally occur in "bursts", and the time aspect must therefore be taken into account when the quality level is established. In Recommendation G.281, the ITU-T defines the following parameters for a switched 64 kbit/s connection between two subscibers:

- *degraded minute, DM*: less than 10% of a number of one-minute intervals have a bit error ratio of 10^{-6} or worse;
- *errored second, ES*: less than 8% of a number of one-second intervals are impaired by bit errors;
- *severely errored second, SES*: less than 0.2% of a number of one-second intervals have a bit error ratio of 10^{-3} or worse.[11]

It is hard to find references to packet loss tolerance. For voice digitized at a 64kb/s rate and packetized (ATM cell, 48 bytes), each packet corresponds to 6 ms of speech. In [12] it is assumed that a maximum loss of two packets out of 100 can be tolerated even in the worst case, i.e., when the two packets are consequtive. In [13] a emulated Internet study lead the writers to conclude that packet losses can be tolerated up to 15 %.

Given the cell size of 50 bytes, the bit error ratios given earlier correspond to cell error ratios, with the assumption of independent bit error probabilities, as presented in Table 3.

BER	CER
10^{-6}	0.0004
10^{-5}	0.0040
10^{-4}	0.0392
10^{-3}	0.3298

Table 3 Bit error rate and cell error rate correspondance

As with the TCP source the phone source is also modeled in a fairly coarse way. The parts of the source represent the two callers. One caller "talks" for an exponentially distributed time after which there is a pause with constant length after which the other caller "talks". The length of the call is exponentially distributed as is the length of the pause in between calls.

The QoS requirements in terms of delay and lost packets are a currently researched topic. In the model the QoS is solely determined by the ratio of delivered packets to the sum of delivered and undelivered packets calculated from 20 second long intervals. If the ratio calculated is larger than the tolerance value, with a certain probability the user reacts. The reaction is either the termination of the call or increase of NBR. As long as the call is in progress each instance of non-satisfactory QoS increases the probability of reaction. The probability reaches one within certain amount of instances (parameter value).

[11] Understanding Telecommunications 1, Ericsson, Telia
[12] Domenico Ferrari: Client Requirements for Real-Time Communication Services. IEEE Communications Magazine Nov 1990
[13] Marko Luoma, Markus Peuhkuri, Tomi Yletyinen. Quality of service for IP voice services – is it necessary ?, Voice, Video & Data Communications '98

Real-time source (Video phone)

Video Phone is a real-time application and differs from IP phone by the amount of data that is transmitted and by the fact that in the video phone both callers transmit data continuously. The bit rate can vary. It is comonly given as multiples of 64 kbps. Reasonable quality can be obtained with 6*64 kbps.

The QoS requirements in terms of delay and lost packets are a currently researched topic. At the rate of 15000 cells/s the cell loss ratio between 10^{-7}-10^{-6} corresponds to MoS quality 4, 10^{-5} to quality 3 and 10^{-4} to quality 2.[14] (Packet has average size 8 cells)

Momentary bit rate is not measured, but is assumed constant.

In the model the QoS is solely determined by the ratio of delivered packets to the sum of delivered and undelivered packets calculated from 30 second long intervals. If the ratio calculated is larger than the tolerance value, with a certain probability the user reacts. The reaction is either the termination of the call or increase of NBR. As long as the call is in progress each instance of non-satisfactory QoS increases the probability of reaction. The probability reaches one within certain amount of instances (parameter value).

Simulations

Duplication of an earlier result

In the paper Simple Integrated Media Access (SIMA) with TCP by Kilkki and Ruutu presented in the 4th INFORMS Telecommunications Conference, the following result is given.

The capacity is divided between 5 TCP-like sources that have different NBRs. The throughput of a TCP-like source is found to be a linear function of the NBR of the source. That is to say that TCP-like sources divide the available capacity according to their respective NBR values.

To assure the validity of the approximation of the SIMA node and TCP constructed for these simulations we should arrive at a similar result. These simulations were run with five TCP-like greedy sources (N.B. TCP-like greedy sources are only used in this simulation) and one SIMA node. The capacity of the SIMA node had values of 50, 100,150 and 200 and the maximum traffic generated by a source in a discretization period was 50 (the unit here corresponds to 50 bytes).

Had the results corresponded exactly to the earlier ones the values of received/capacity/NBR should be constant for each simulation. This not quite the case, but the values did not vary too much.

[14] Hanna-Maija Karjalainen: Videopalvelun subjektiivinen laatu ATM-verkossa. Diplomityö. 1997

even NBRs (Capacity 150)

NBR	50	50	50	50	50
transmitted	14087720	13554710	13549850	13548490	13547840
received	11165300	8007380	8024620	8006510	7979340
rec./trans.	0.792556	0.590745	0.592229	0.590952	0.588975
rec./cap./NBR	0.004135	0.002966	0.002972	0.002965	0.002955

uneven NBRs (Capacity 50)

NBR	200	100	50	25	12
transmitted	13394510	5741920	5700690	3651610	8589660
received	10190730	1394630	1401610	22690	1250770
rec./trans.	0.760814	0.242886	0.245867	0.006214	0.145613
rec./cap./NBR	0.002831	0.000775	0.001557	5.04E-05	0.005791

uneven NBRs (Capacity 100)

NBR	200	100	50	25	12
transmitted	17738260	10477970	6445830	6036410	8322940
received	17624970	2324130	2730200	2060770	3748050
rec./trans.	0.993613	0.221811	0.423561	0.34139	0.450328
rec./cap./NBR	0.002448	0.000646	0.001517	0.00229	0.008676

uneven NBRs (Capacity 150)

NBR	200	100	50	25	12
transmitted	17999870	15620240	12636310	7300080	6122940
received	17999870	8715640	8119640	2746640	1097180
rec./trans.	1	0.557971	0.642564	0.376248	0.179192
rec./cap./NBR	0.001667	0.001614	0.003007	0.002035	0.001693

uneven NBRs (Capacity 200)

NBR	200	100	50	25	12
transmitted	17999920	17999920	15718220	12639860	6668780
received	17999870	14129290	8955810	7460000	1931220
rec./trans.	0.999997	0.784964	0.569773	0.590196	0.289591
rec./cap./NBR	0.00125	0.001962	0.002488	0.004144	0.002235

Table 4 The combined results of the simulation

One duplex SIMA node

Description

This (nearly) simplest possible model consists of one duplex SIMA node and a number of both real-time and non real-time sources. The following is a set of simulations with four different source configurations and three different stages of reaction.

The number of sources was chosen so that their combined average rates (when on) corresponded to 0.5, 1 or 2 times the capacity of the SIMA node.

0	6*1,6+2=11,6
1	6*1,6+2*2+10=23,6
2	14*1,6+4*2+2*10=50,4
3	28*1,6+8*2+4*10=100,8

Table 5 Average traffic of configurations

The capacity of the node had the default value of 50 cells/DI. A video source that is on produces 10 cells/DI, IP phone 2 cells/DI and a TCP source on the average 1,6 and at maximum 50 (the actual rate is dependent on the available capacity).

The purpose of these simulations is to find out the nature of the traffic under different loads. Table 6 presents the users defined by their parameters. The simulated time was 1 hour, the results were recorded after an initial period of 10 minutes, so that the period where all the real-time sources have their initial break is excluded.

	WWW user(TCP)	
	NBR	
1	6	
2	12	
3	25	
	IP phone	
	NBR	**Priority**
1	2	4
2	4	5
	Video phone	
	NBR	**Priority**
1	10	4
2	20	5

Table 6 Users defined by parameters

Each configuration was run with the three settings

A. no source adjusted traffic flow
B. the only reaction model was to break transmission
C. for real time + high NBR users the reaction is to buy more (double) NBR.

A	
	WWW user(TCP)
	QoS control
1	0
2	0
3	0
	IP phone
	Quality parameter
1	10^5
2	10^5
	Video phone
	Quality parameter
1	10^5
2	10^5

Table 7 Quality control parameters for setting A

B				
	WWW user(TCP)			
	QoS control	**Constant**	**Constant - times**	**Delay limit**
1	1	30	10	250
2	1	30	10	250
3	1	30	10	250
	IP phone			
	Quality P.	**W. buy NBR**	**QoS ratio**	**Action**
1	100	0	0.04	break
2	50	0	0.04	break
	Video phone			
	Quality P.	**W. buy NBR**	**QoS ratio**	**Action**
1	100	0	0.04	break
2	50	0	0.04	break

Table 8 Quality control parameters for setting B

C				
	WWW user(TCP)			
	QoS control	**Constant**	**Constant - times**	**Delay limit**
1	1	30	10	250
2	1	30	10	250
3	1	30	10	250
	IP phone			
	Quality P.	**W. buy NBR**	**QoS ratio**	**Action**
1	100	0	0.04	break
2	50	0.75	0.04	buy NBR
	Video phone			
	Quality P.	**W. buy NBR**	**QoS ratio**	**Action**
1	100	0	0.04	break
2	50	0.75	0.04	buy NBR

Table 9 Quality control parameters for setting C

Configuration 0

One IP phone connection (duplex) with NBR 4 and for each direction 6 TCP connections with NBRs two of each NBR 6, 12 and 25.

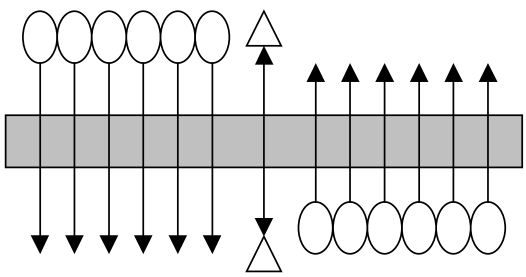


Figure 1. Schematic representation of configuration 0

A. no source adjusted traffic flow

	WWW user(TCP)						
	NBR	**1**	**2**	**3**	**4**	**5**	**6**
1	6	36,3	38,8	8,3	6,9	4,9	4,8
2	12		29,3	15,0	20,1	14,0	21,6
3	25			12,6	16,3	22,1	49,0

Table 10 The priorities received by WWW sources (as percentage of instants)

	IP phone		
	NBR	**Priority**	**QoS ratio**
2	4	5	0 - 0.035

Table 11 QoS ratios for the real-time sources

Load exceeded the capacity at one of the nodes in 3942 instances.

	1	2	3	4	5	6
1	1,5	1,3	2,4	3,1	88,0	3,7

Table 12 Priority limit distribution

B. The only reaction model was to break transmission

	WWW user(TCP)						
	NBR	**1**	**2**	**3**	**4**	**5**	**6**
1	6	4,9	20,1	21,1	22,1	15,9	15,9
2	12		15,0	38,7	15,8	11,5	19,0
3	25			11,4	20,2	21,1	47,4

Table 13 The priorities received by WWW sources (as percentage of instants)

	IP phone			
	NBR	**Priority**	**QoS ratio**	**Interrupt**
2	4	5	0 - 0.035	none

Table 14 QoS ratios for the real-time sources

Load exceeded the capacity at one of the nodes in 2955 instances.

1	2	3	4	5	6
0,3	1,1	2,6	2,4	90,2	3,4

Table 15 Priority limit distribution

C. For real-time, high NBR users the primary reaction is to buy more NBR

Because there were no interruptions, results are the same as for B.

Configuration 1

One video phone connection (duplex) with NBR 20, Two IP phone connections (duplex) with NBRs 2 and 4 respectively and for each direction 6 TCP connections with NBRs 6, 12 and 25.

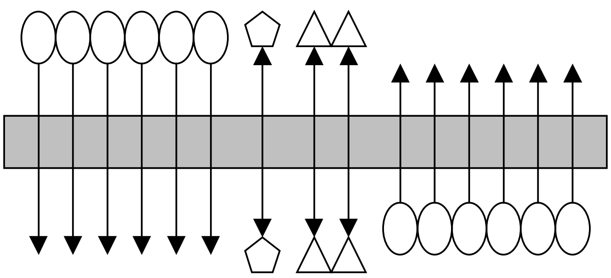


Figure 2. A schematic representation of configuration 1

A. no source adjusted traffic flow

	WWW user(TCP)						
	NBR	**1**	**2**	**3**	**4**	**5**	**6**
1	6	34,4	40,6	8,6	7,1	4,7	4,6
2	12		28,6	16,8	19,2	13,8	21,5
3	25			12,8	16,3	20,9	50,0

Table 16 The priorities received by WWW sources (as percentage of instants)

	IP phone		
	NBR	**Priority**	**QoS ratio**
1	2	4	0 - 0,085
2	4	5	0 - 0,042
	Video phone		
	NBR	**Priority**	**QoS ratio**
2	20	5	(no values)

Table 17 QoS ratios for the real-time sources

Load exceeded the capacity at one of the nodes in 6562 instances.

1	**2**	**3**	**4**	**5**	**6**
1,0	1,1	1,6	61,2	27,4	7,6

Table 18 Priority limit distribution

B. The only reaction model was to break transmission

	WWW user(TCP)						
	NBR	**1**	**2**	**3**	**4**	**5**	**6**
1	6	4,0	22,9	24,4	22,6	13,6	12,5
2	12		31,3	17,9	18,5	12,9	19,4
3	25			12,7	19,3	20,8	47,3

Table 19 The priorities received by WWW sources (as percentage of instants)

	IP phone			
	NBR	**Priority**	**QoS ratio**	**Interrupt**
1	2	4	0 - 0.09	none
2	4	5	0 - 0.06	none
	Video phone			
	NBR	**Priority**	**QoS ratio**	**Interrupt**
2	20	5	0 - 0.025	3

Table 20 QoS ratios for the real-time sources

Load exceeded the capacity at one of the nodes in 4750 instances.

1	2	3	4	5	6
0,1	1,2	1,2	56,8	38,9	1,8

Table 21 Priority limit distribution

C. For real-time, high NBR users the primary reaction is to buy more NBR

Since none of the interruptions were changed to buys, the results are as for B.

Configuration 2

Two video phone connections (duplex) with NBR 10 and 20, four IP phone connections (duplex) with NBRs 2 and 4 and for each direction 14 TCP connections with NBRs 6, 12 and 25.

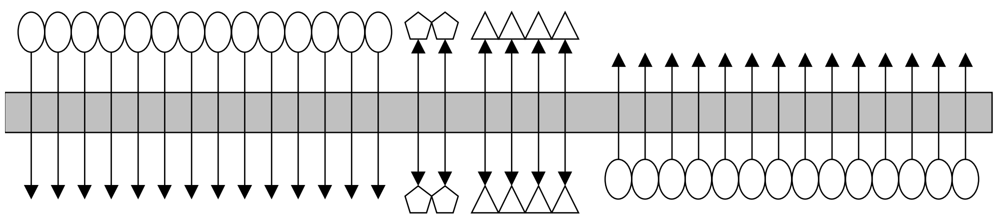


Figure 3. A schematic representation of configuration 2

A. no source adjusted traffic flow

	WWW user(TCP)						
	NBR	**1**	**2**	**3**	**4**	**5**	**6**
1	6	6,4	24,6	23,7	20,9	13,3	11,2
2	12			16,7	26,0	25,4	31,9
3	25				22,9	24,1	53,1

Table 22 The priorities received by WWW sources (as percentage of instants)

	IP phone		
	NBR	**Priority**	**QoS ratio**
1	2	4	0 - 0,18
2	4	5	0 - 0,06
	Video phone		
	NBR	**Priority**	**QoS ratio**
1	10	4	0 - 0,08
2	20	5	0 - 0,09

Table 23 QoS ratios for the real-time sources

Load exceeded the capacity at one of the nodes in 27745 instances.

1	2	3	4	5	6
0,7	3,5	3,5	52,2	37,2	3,0

Table 24 Priority limit distribution

B. The only reaction model was to break transmission

	WWW user(TCP)						
	NBR	**1**	**2**	**3**	**4**	**5**	**6**
1	6	14,1	16,7	25,8	20	13,6	9,7
2	12			14,0	25,9	25,9	34,2
3	25			4,3	15,9	24,7	55,1

Table 25 The priorities received by WWW sources (as percentage of instants)

	IP phone			
	NBR	**Priority**	**QoS ratio**	**Interrupt**
1	2	4	0 - 0,15	1
2	4	5	0 - 0,08	none
	Video phone			
	NBR	**Priority**	**QoS ratio**	
1	10	4	0 - 0,09	6
2	20	5	0 - 0,055	4

Table 26 QoS ratios for the real-time sources

Load exceeded the capacity at one of the nodes in 19673 instances.

1	2	3	4	5	6
0,3	1,2	2,5	68,3	20,7	7,1

Table 27 Priority limit distribution

C. For real-time, high NBR users the primary reaction is to buy more NBR

Since none of the interruptions were changed to buys, the results are as for B.

Configuration 3

Four video phone connections (duplex) with NBR 10 and 20, eight IP phone connections (duplex) with NBRs 2 and 4 and for each direction 28 TCP connections with NBRs 6, 12 and 25.

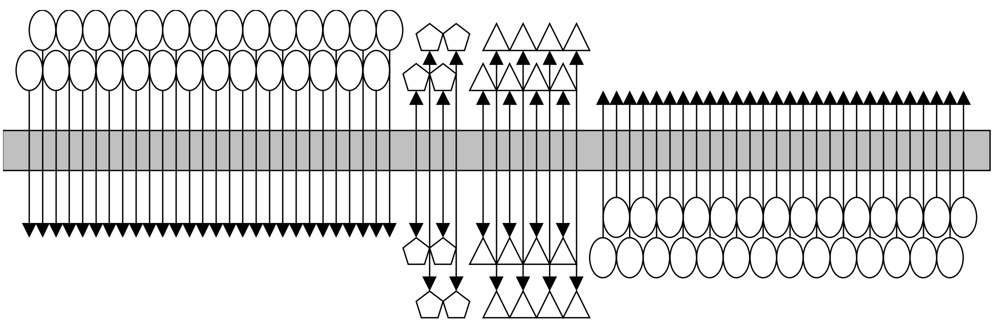


Figure 4. A schematic representation of configuration 3

A. no source adjusted traffic flow

	WWW user(TCP)						
	NBR	**1**	**2**	**3**	**4**	**5**	**6**
1	6	15,1	55,5	17,4	5,7	3,7	2,7
2	12		0,8	53,3	19,6	11,6	14,7
3	25			6,3	26,2	23,9	43,5

Table 28 The priorities received by WWW sources (as percentage of instants)

	IP phone		
	NBR	**Priority**	**QoS ratio**
1	2	4	0 - 0,25
2	4	5	0 - 0,128
	Video phone		
	NBR	**Priority**	**QoS ratio**
1	10	4	0 - 0,163
2	20	5	0 - 0,1

Table 29 QoS ratios for the real-time sources

Load exceeded the capacity at one of the nodes in 68483 instances.

	1	2	3	4	5	6
1	1,2	6,2	8,7	56,3	24,2	3,3

Table 30 Priority limit distribution

B. The only reaction model was to break transmission

	WWW user(TCP)						
	NBR	**1**	**2**	**3**	**4**	**5**	**6**
1	6	3,3	68,9	17,4	4,8	3,2	2,4
2	12		4,5	50,0	19,4	12,0	14,2
3	25				38,7	21,3	40,0

Table 31 The priorities received by WWW sources (as percentage of instants)

	IP phone			
	NBR	**Priority**	**QoS ratio**	**Interrupt**
1	2	4	0 – 0,13	none
2	4	5	0 – 0,12	1
	Video phone			
	NBR	**Priority**	**QoS ratio**	**Interrupt**
1	10	4	0 – 0,17	6
2	20	5	0 – 0,12	7

Table 32 QoS ratios for the real-time sources

1	2	3	4	5	6
0,5	8,5	7,0	63,7	15,1	5,2

Table 33 Priority limit distribution

C. For real-time, high NBR users the primary reaction is to buy more NBR

This simulation could not be run due to insufficient memory.

Conclusions drawn from the results

The C setting was the one of greatest interest and it was never simulated due to either lack of traffic or lack of computer memory.

One duplex SIMA node – 48 sources

Description

Since in the previous results there were rather few interruptions the question of the influence of the users reactions on the system's functions did not become clear. Both possible alterations to induce more insatisfaction of QoS were used in the following simulations – more traffic sources and tighter QoS parameters.

Due to computer memory capacity problems the simulated time had to be shortened to 50 minutes.

There were six types of traffic sources; 14 TCP sources with lower NBR and 14 with higher both directions, 8 phone sources with lower NBR and quality requirements and 8 with higher and 2 video phone sources with lower NBR and quality requirements and 2 with higher.

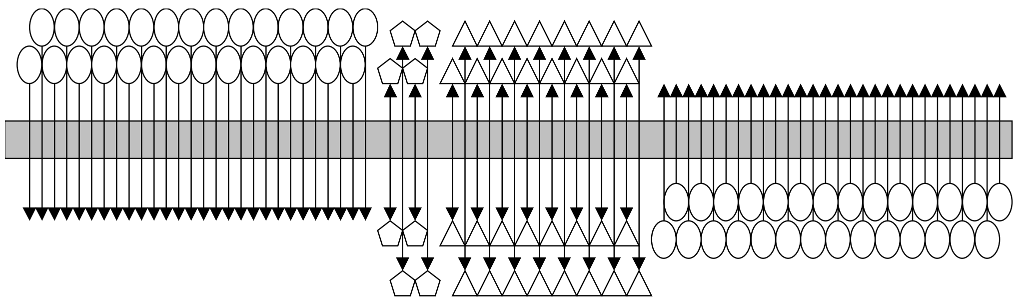


Figure 5. A schematic representation of the configuration

	WWW user(TCP)	
	NBR	
1	6	
2	12	
	IP phone	
	NBR	**Priority**
1	2	4
2	4	5
	Video phone	
	NBR	**Priority**
1	10	4
2	20	5

Table 34 Users defined by parameters

There were two simulations, one with no quality control (A) and one with quality control (B). Both simulations were run with three different random seeds. The real-time sources with lower quality expectations responded only by termination whereas the sources with higher quality requirements bought more NBR with the probability 90% and terminated the connection with probability 10%.

A	
	WWW user(TCP)
	QoS control
1	0
2	0
3	0
	IP phone
	Quality parameter
1	10^5
2	10^5
	Video phone
	Quality parameter
1	10^5
2	10^5

Table 35 Quality control parameters for A simulations

B				
	WWW user(TCP)			
	QoS control	**Constant**	**Constant – times**	**Delay limit**
1	1	30	10	250
2	1	30	10	250
3	1	30	10	250
	IP phone			
	Quality P.	**W. buy NBR**	**QoS ratio**	**Action**
1	50	0	0.04	break
2	10	0.9	0.04	break/buy
	Video phone			
	Quality P.	**W. buy NBR**	**QoS ratio**	**Action**
1	50	0	0.04	break
2	10	0.9	0.04	break/buy

Table 36 Quality control parameters for B simulations

The probes recorded the throughput per source type, load distribution, distribution of the priority limit in the SIMA node and the priority distributions for traffic that arrived at the SIMA node and traffic that passed the SIMA node.

Buys and Breaks

The lack of user influence in previous simulations was evident also now, with more traffic and higher quality standards. The combined user reactions in the B simulations are as follows in Table 36.

	B1	B2	B3
phone low – break (8)	15	11	13
phone high – break (8)	-	1	-
phone high – buy (8)	4	6	4
video low – break (4)	3	4	3
video high – break (4)	1	-	-
video high – buy (4)	-	1	-

Table 37 User reactions in simulations B1, B2 and B3

As the differences between A and B simulations tend to be small, one would be tempted to wish for a more significant number of user reactions. Additional work in finding an interesting number could have been done. It is, however, necessary to remember the connection to reality, would a real-time application user actually try to make further connections within an hour if she had terminated twice because of insatisfactory quality of service?

Throughput per source type

Information of individual traffic sources was not examined, but groups composed of the six different sources. Questions of interest are the differences in throughput between the low/high quality sources and the difference between A and B simulations.

Two different throughputs were calculated. One is the ratio of received bytes to transmitted bytes and the other the ratio of received bytes of a group to all received bytes.

Consistently (the case of video 2 in simulation 1 and 2 is caused by a small amount of traffic) the sources with higher NBR had a higher throughput.

The differences between A and B simulation were very small and not consistent.

A1		tcp 1	tcp 2	phone 1	phone 2	video 1	video 2
	% of transmitted	0.146278	0.210599	0.136609	0.154396	0.215601	0.136518
	throughput	0.678894	0.749854	0.924349	0.965011	0.692175	0.966906
			0.356877				0.643123
B1		tcp 1	tcp 2	phone 1	phone 2	video 1	video 2
	% of transmitted	0.153294	0.204901	0.154271	0.176257	0.209819	0.101458
	throughput	0.717673	0.763302	0.93833	0.970558	0.938742	0.974161
			0.358195				0.641805
A2		tcp 1	tcp 2	phone 1	phone 2	video 1	video 2
	% of transmitted	0.150991	0.180872	0.111684	0.120012	0.237911	0.19853
	throughput	0.633182	0.74378	0.913449	0.964029	0.917158	0.965736
			0.331863				0.668137
B2		tcp 1	tcp 2	phone 1	phone 2	video 1	video 2
	% of transmitted	0.153224	0.191353	0.115704	0.135285	0.17229	0.232145
	throughput	0.667522	0.747025	0.928123	0.970979	0.929864	0.974672
			0.344576				0.655424
A3		tcp 1	tcp 2	phone 1	phone 2	video 1	video 2
	% of transmitted	0.219685	0.207388	0.155929	0.140857	0.119394	0.156748
	throughput	0.700217	0.808438	0.922671	0.962627	0.930251	0.963661
			0.427072				0.572928
B3		tcp 1	tcp 2	phone 1	phone 2	video 1	video 2
	% of transmitted	0.21143	0.213123	0.154541	0.165246	0.072228	0.183432
	throughput	0.710988	0.814539	0.932398	0.972548	0.942564	0.971179
			0.424553				0.575447

Table 38 Throughput of the source groups

Load distribution

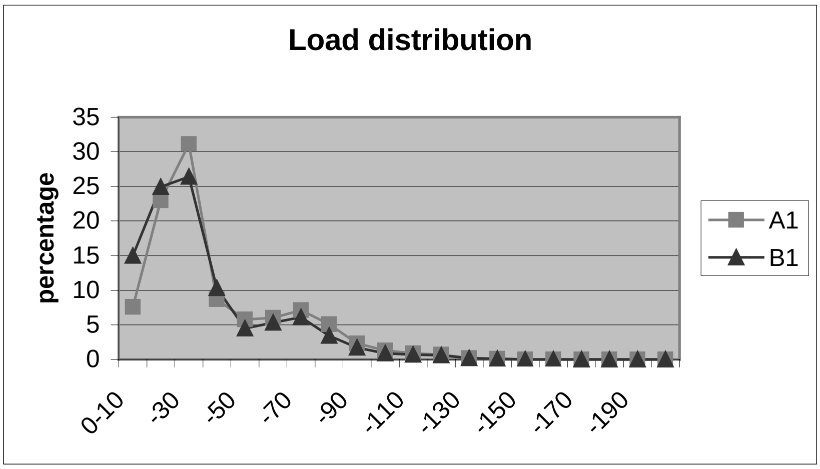
For each time interval the incoming traffic was measured and gathered in a histogram. Additionally the number of time units (out of 300000) when load was 0 was recorded.

Simulation	
A1	1821
B1	2379
A2	571
B2	512
A3	3139
B3	4049

Table 39 Number of no-load time instances

Load distributions for all the simulations had two peeks. At most of the time instances the load was lower than the capacity limit (of 50). In addition to a significant peek in this interval there was a lower peek slightly above the capacity limit.

Figure 6. A chart of the load distributions of two respective simulations

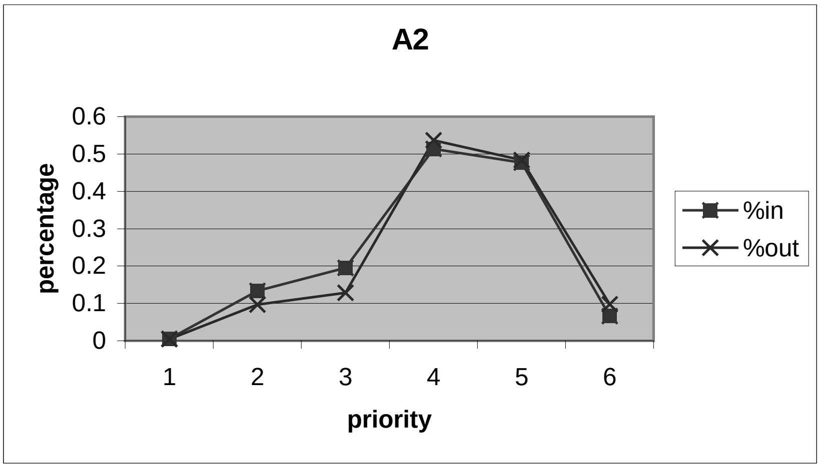


The one consistent difference between A and B simulations was that for B simulations the number of time instances where load was lower than capacity was slightly higher.

Priority distributions for traffic that arrived at the SIMA node and traffic that passed the SIMA node.

As could be expected the proportion of lower priority traffic is slightly lower for the outgoing traffic since the traffic that exceeds capacity will necessarily have a low priority. The difference between A and B simulations is negligible.

Figure 7. Chart of simulation results



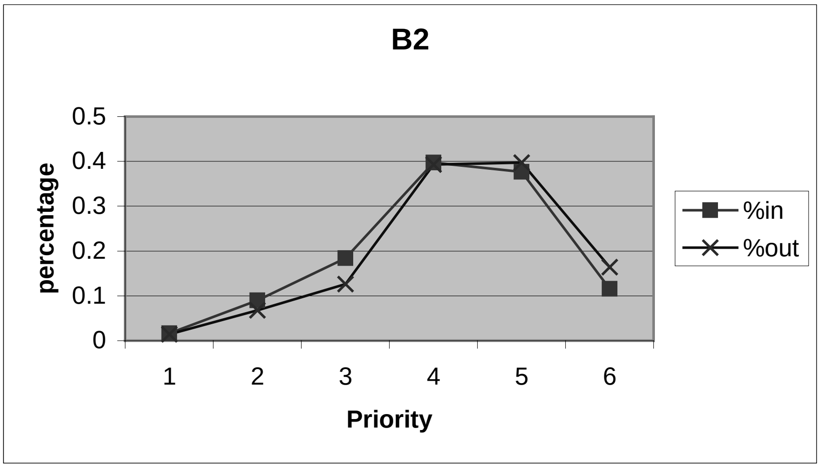


Figure 8. Chart of simulation results

Distribution of the priority limit in the SIMA node

The priority limit in the simulation was defined so that in a time interval there could be packets with the limit priority and some would be passed and others discarded.

In the simulations all the priorities were limits at some time, but for the majority of time the priority limit was 4. Priority 4 was rather common, representing half of the real-time sources.

The difference between A and B simulations was again small.

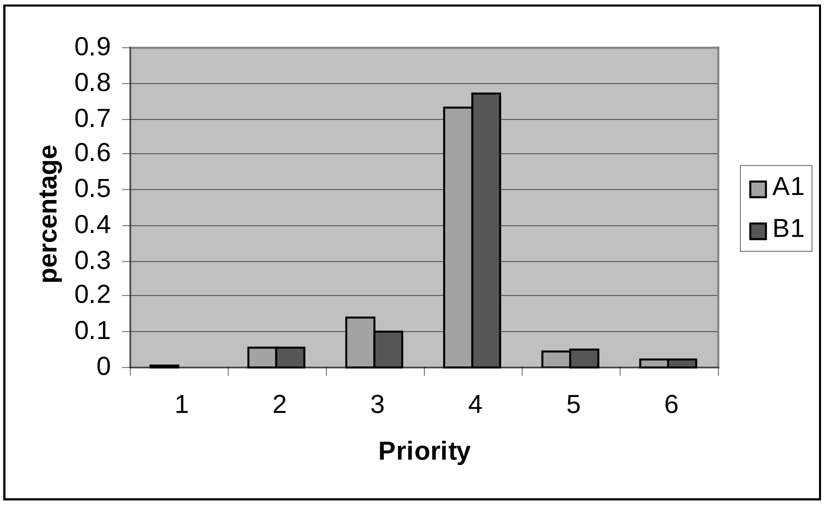


Figure 8 Priority limit distribution

Conclusions drawn from the results

It is clear that with this rather low number of sources and a short simulated time it is hard to find result that is clear. But the presented results give very little evidence of significant differences between the model with user reaction and the one without.

Conclusions and suggestions for further work

The simulation results showed that a discrete time approximation of SIMA can work. However, this level of traffic and simulated time no real difference between models with user reactions and without user reactions could be detected.

The simulations presented in this report were performed with the program package BONeS DESIGNER. This program sets limits to the complexity of the models, as the simulation time grows considerably as more pieces are added to the model. Also memory allocation became a problem for the larger configurations. There is no hope in constructing a configuration with more SIMA nodes and/or sources with this program, computer facilities and model.

After the analysis of the simple case of one SIMA node is completed it would be necessary to proceed to small networks. These would exhibit the properties of traffic that has passed multiple SIMA nodes. In order to accomplish this the present model would need further simplification or a more efficient formulation.

To better model the users more information about the QoS of real-time applications would have to be incorporated into the model. With such information one could also define utility functions which could then represent various kinds of consumers.

Documentation of the simulation model

General

Discretization interval (DI): 10 ms
Cell: Approximates ATM cell – 50 bytes
TCP packet: in reality has a range of sizes. Here has (for the purposes of the transmission window) the constant size of 500 bytes, 10 cells
SIMA packet: Data structure that corresponds to the transmission of one source during a DI

- Group (integer)
- Source (integer)
- Priority (integer)
- NBR (real)
- Destination (integer)
- Passed (binary)
- Size (integer), number of cells
- Non real-time (binary, 1 for non real-time, 0 for real-time)
- Delay (real)

SIMA node (S node II)

Parameters

Maximum Queue Size (integer)
Maximum number of packets in a queue, Default 1000
Capacity – init (integer)
Capacity at the beginning of the DI, The number of cells / DI, 2Mbps – 50, 34MBPS – 850, 155Mbps – 3875, default 50
Capacity (memory, integer)
remaining capacity in DI
limit (memory, integer)
the lowest priority accepted in DI, initialization value 0
buffer size (integer)
used in the linear transformation of priority limit to delay, default 1 Mbyte - 20000

Input ports

Clear (trigger)
Begins the output of the node
Incoming (SIMA packet)

Output ports

Output (SIMA packet)
onwards (trigger)
signals the end of the output of the node

Description

Incoming packets are separated based on their *passed* value.
Incoming packets that have *passed* value of 1 are placed in a Priority FIFO Queue. Packets with the *passed* value 0 are placed to a Simple FIFO Queue(1).
After the *Clear* trigger the memory parameters *Capacity* and *Limit* are initialized and the packets in the Priority FIFO Queue are released one by

one. The packet *Size* is read and as long as there is more capacity left than the *Size* the packets are passed on to a Simple FIFO Queue (2) and the *Size* is substracted from the remaining capacity. If there is not enough capacity left for the entire packet, it is divided into two parts. The one corresponding to the capacity that is left is passed on to Simple FIFO Queue (2) and the other part to *Output*. *Capacity* is set to 0.
After the capacity is used up all the rest of the packets are placed in a Simple FIFO Queue (3).
The priority of the last packet accepted is stored in memory as *Limit*.
After the Priority FIFO Queue is emptied, the accepted packets are released from their queue (2) and given their delays. The delay is the lowest priority accepted multipled by a constant for the non realtime packets and for a real-time packets is zero.
After the packet gets its delay it goes to the *Output* port.
Then the discarded packets waiting in a Simple FIFO Queue (1) are released to the *Output*. Then the discarded packets waiting in a Simple FIFO Queue(3) are released to the *Output* and receive a *passed* value of 0. Once the Simple FIFO Queue(3) has emptied the SIMA node has finished for the DI and trigger is inserted to *Onwards* port.

Phone source (Primary and Secondary)

Parameters

Probability (1/(E+1)) (real)
Parameter for the geometric distribution that determines the length of "a line" of conversation, default E=5s – 500

Source id (integer)

NBR – value (real)

MBR (integer)
Number of cells sent during DI, 64kbps – 1.6, default 2

Mean for the interval between calls (real)
The mean of both the length of the call and the pause between calls, default 4*60s

QoS ratio (real)
The limit of acceptable value of the ratio undelivered / (undelivered + delivered), default 0.04

Destination (integer)

Undelivered packets (integer, memory)
Initialization value 1

Delivered packets (integer, memory)
Initialization value 1

Willingness to buy NBR (real)
probability of choosing buying NBR instead of breaking off transmission

Traffic (memory, integer)

Binary memory (memory, binary)
directs the pulses between primary and secondary sources

help (integer, memory)

Quality parameter (integer)
number of instances of substandard quality before reaction

End probability (real)

Probability of ending connection when quality is below desired. At those instants grows at a inversely proportional rate to Quality parameter N p*=(sqrt(p)+1/N)^2

Input ports

Feedback (SIMA packet)

Uniform pulses (trigger)

Output ports

Output (SIMA packet)

Break (trigger)

Buy (trigger)

(Trigger for secondary (trigger)**)**

Description

Primary Phone Source is responsible for beginning the call and monitoring its quality. Primary Phone Source produces the *SIMA packet*s of source 1 and triggers for source 2.

At initialization *traffic* gets its first value (geometric distributed, parameter *probability 1/(E+1)*) and the length of the first break before the first call is set (exponentially distributed with the expected value of *Mean for the interval between calls*). At the end of the break the length of the call is generated from the same distribution.

Uniform pulses pass gate if call is in progress. After random delay (uniformely distributed) value of *traffic* is examined, if *traffic* is greater than 0, *traffic* is substracted by 1 and according to value of *binary memory* either a pulse is deferred to secondary source or a packet is created at primary. If *traffic* is not greater than 0 then *traffic* gets new value as does *binary memory*.

The quality of the call is controlled by keeping count of received and non-received cells (with *undelivered packets* and *delivered packets*). Each 20 seconds the ratio undelivered packets/(undelivered packets + delivered packets) is calculated (if any packets were received during the period). If it is greater than the limit value *QoS ratio, end probability* grows and call is terminated with that probability (or if *Willingness to buy NBR* is greater than 0, then an additional random switch determines whether the call is terminated or NBR is increased).

Video Phone source (Primary and Secondary)

Parameters

Source id (integer)

NBR – value (real)

MBR (integer)

Number of cells sent during DI, 6*64kbps – 10, default 10

Mean for the interval between calls (real)

The mean of both the length of the call and the pause between calls, default 10*60s

QoS ratio (real)

The limit of acceptable value of the ratio undelivered / (undelivered + delivered)

Willingness to buy NBR (real)

Destination (integer)

Undelivered packets (integer, memory)

Initialization value 1

Delivered packets (integer, memory)

Initialization value 1

help (integer, memory)

Quality parameter (integer)

number of instances of substandard quality before reaction

End probability (real)

Probability of ending connection when quality is below desired. At those instants grows at a inversely proportional rate to Quality parameter N p*=(sqrt(p)+1/N)^2

Input ports

Feedback (SIMA packet)

Uniform pulses (trigger)

(Interrupt from secondary (trigger)**)**

Output ports

Output (SIMA packet)

(Interrupt (trigger)**)**

Break (trigger)

Buy (trigger)

Description

Primary Video Phone Source produces the *SIMA packet*s of source 1 and triggers for source 2. Both sources monitor their own quality and is able to interrupt the call.
At initialization the length of the first break before the first call is set (exponentially distributed with the expected value of *Mean for the interval between calls*). At the end of the break the length of the call is generated from the same distribution.
Uniform pulses pass gate if call is in progress. After random delay (uniformely distributed) a pulse to secondary source and a packet created at primary source.
The quality of the call is controlled by keeping count of received and non-received cells (with *undelivered packets* and *delivered packets*). Each 30 seconds the ratio undelivered packets/(undelivered packets + delivered packets) is calculated (if any packets were received during the period). If it is greater than the limit value *QoS ratio, end probability* grows and call is terminated with that probability (or if *Willingness to buy NBR* is greater than 0, then an additional random switch determines whether the call is terminated or NBR is increased).

TCP source (TCP source II)

Parameters

Mean delay between bursts (real)

Delay between WWW browsing “clicks”, default 20

Mean number of pulses (real)

Average number of files / “click”, default 5

Inter-pulse Time (during burst) (real)

Default 0

Traffic (integer, memory)

Number of cells waiting for transmission

Window (integer, memory)

Number of cells that can be transmitted in DI, initialization 10

RTT (integer)

multiples of DI

MBR memory (real, memory)

Capacity of access line (integer)

Given as cells/DI, maximum number of cells than can be transmitted in DI, since the average amount of traffic generated with the default values above is 16 cells/DI this must be sufficiently higher, default 50

alfa (real) default 48

gamma (real) default 1.3

parameters for Perto distribution with the default values E=8000 bytes =160and medium 1500 bytes=30

QoS control (binary)

0 for no quality control 1 for quality control

Source id (integer)

NBR-value (integer)

Delay limit (real)

tolerance limit for the delay attached to SIMA paketti'sDefault 5

Last traffic=0 (memory, real)

initialization value 0

Constant (real)

tolerance value for the length of time that the transmission of the files requested by one click takes, Default 30

Too long calculator (memory, integer)

Constant - times (integer)

Tolerance value for TLC, Default 10

Mean (real)

Mean for the length of the break before beginning another TCP session after an interruption, default 5*60s

Alpha

The constant for the measurement, default 0,99

Input ports

Feedback (SIMA packet)

uniform pulses (trigger)

Output ports

Output (SIMA packet)

Description

The traffic to be sent is created by a bursty traffic source. If the gates 1 and 2 are open for each pulse a geometrically distributed (with the parameter *probability*) random number of cells are added to *Traffic*.

Gate 1 ensures that new pulses do not go through if traffic created by earlier requests is still in transmission. Gate 2 enables the break in session if QoS values become too low (and Quality control is on).

At the beginning of each *DI* after random delay (uniform distribution between 0 and 0.001) the source checks if the value of *Traffic* is greater than zero.

If it is then the traffic to be sent during *DI* is, if *Traffic* is greater than *Window/RTT*, the minimum of *Window/RTT* and *capacity of access line* and else it is *Traffic*. This value is placed as Size for the outgoing packet.

If *Traffic* is equal to zero then the *Window* and *MBRmemory* are set to the initialization values. The value of last traffic=0 is substracted from Tnow and value is compared with constant. If smaller than *TLC* is substracted by one. If greater then increased by one and *TLC* compared to *constant-times* . if *TLC* is greater session terminated is for a random period (exponentially distributed with mean *mean*)

The MBR is calculated as (1-alpha)*MBRmemory+alpha*"traffic to be sent during DI"
The cells arriving at the input port are examined for their *passed* value. If the value is 0 then *Window* is halved *Size* times and *Size* is added to *Traffic*. If the value is 1 then the *Window* grows with *Size**packet size/*Window* and the value of Delay is compared to *Constant*. If greater the TLC grows by one, smaller substracted by one.