A Modeling Approach for the Performance Management of High Speed Networks *

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Introduction

- Large high speed networks require sophisticated modeling, management and planning tools for use by network operators.
- Classify and characterize customer traffic in terms of average and burstiness.
- Evaluate network performance in terms of delay and losses.
- Predict effect of adding new customers on overall performance.
- Manage existing facilities optimally.
- Plan network expansion.

As data networks have become larger, higher speed and more complex, there has arisen a growing need for advanced modeling, management and planning tools which will assist network operators maintain and improve network performance. This paper describes a two year effort to develop a system tool that will simulate and analyze packet switched networks carrying bursty traffic which can be used for a variety of networks and services such as frame relay, SMDS or B-ISDN. This network tool will be based upon models of network elements. The information needed to drive the tool and describe its elements will vary from detailed to sketchy. Some information will be derived from historical measurements and some from expected characteristics of users based on responses to a questionnaire answered at the time of subscription to the data service. Part of our work was to understand the effect of the quality of data on the precision and generality of the tool.

By using the tool, operators will be able to alter network elements and configurations in the model, analyze performance data, and then identify the consequences of network changes, including the evaluation of proposed configurations. The tool will also be used to compare the performance of different networks, to compare different configurations of the same network, and to evaluate network performance during specified intervals.

A Packet Switched Network

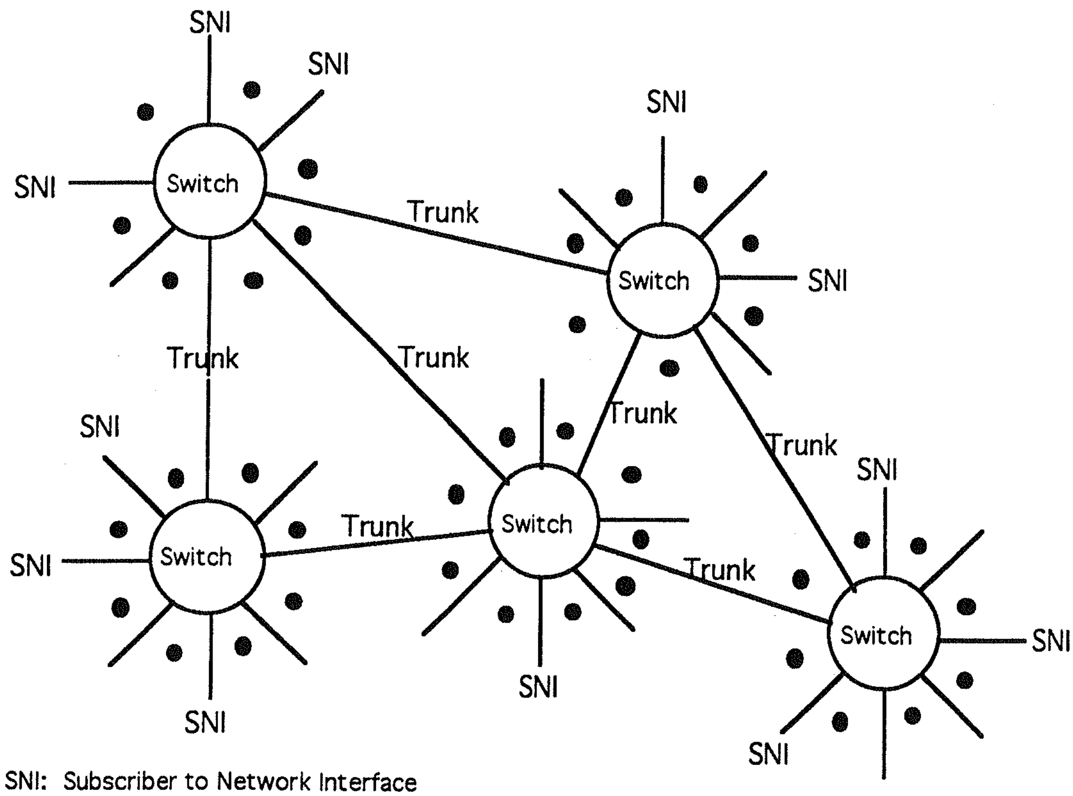

Figure 1: A Packet Switched Network

In general a packet switched network can be represented as a set of nodes interconnected by a set of links. This gives a topological characterization of the network. Customers can send traffic over the network by connecting to one of the nodes via an input interface and sending messages to an output interface at the same or some other node. This is illustrated in Figure 1. The nodes are the network switches, the links are customer access lines or the trunks connecting switches.

Classification and Characterization of Traffic

1. **LAN Interconnection Traffic**

2. **Video applications**
   a. Video Telephony, Multimedia Teleconferencing
   b. Compressed Real Time Packet Video

3. **Teletex(Correspondence exchange)**

4. **Facsimile**

5. **High-resolution graphics – Electronic Imaging, Medical Imaging, CAD–CAM**

1. LAN Interconnection Traffic According to recent results of LAN traffic analysis studies conducted at Bellcore, three protocols have the following traffic characteristics: 80% of TCP packets are 50 bytes, 10% are 500 bytes and10% are 1Kbytes long; 80% of UDP packets are 150 bytes, 10% are 1.5 Kbytes; 80% of ND traffic are 1550 bytes and 20% are 50 bytes. The utilization due to ND traffic is comparable to that of UDP traffic, while TCP traffic represents a small fraction of the total network traffic. The combination of these traffic types results in a "trimodal " distribution for the overall traffic. The largest percentage of packets are less than 200 bytes in size while there are considerable spikes at lengths of 1 Kbyte and 1550 bytes.

2. Video applications

Video Telephony, Multimedia Teleconferencing: The data rate ranges from 1.5 Mbits/sec to 140 Mbits/sec depending on the encoding, the compression, and the service quality requirements.

Compressed Real Time Packet Video: Transmission of good quality compressed video requires at least 1.5 Mb/sec with small video packet loss and delay. For 1.5 Mbits/sec transmission rates, we conclude that a video frame can be segmented into 4 to 5 packets of 1500 bytes with inter arrival times of approximately 8 msec.

3. Teletex(Correspondence exchange): With a transmission rate of 9.6 Kbps, page data volume of 20 Kbits and packet sizes of 1500 bytes, if we consider one page as a burst, then the mean burst period is about 2 secs while the packet inter arrival time is 1 sec.

4. Facsimile: A bit rate of 2 Mbits/s (for compressed data) and more (for uncompressed data) is desirable in order to achieve acceptable transmission times for compatibility with the operation speeds of document preparation equipment of the near future.

5. High-resolution graphics – Electronic Imaging (Medical Imaging, CAD–CAM applications: Based on the typical transmission speed of work stations which is about 0.5 Mbits/sec and using as a packet size the maximum allowable Ethernet packet of about 1530 bytes, while assuming that a single image or graph represents a burst, we conclude that the burst length is on the order of minutes and the mean packet inter arrival time is approximately 25 msec.

The QNA Method for Network Analysis

The flows of packets to a switch are characterized by:

1. The mean of the interarrival time and packet length, namely $E\{T_a\}$ (or the average arrival rate $\lambda = 1/E\{T_a\}$) and $E\{L\}$.

2. The squared coefficients of variation of the interarrival time and packet length, namely v_T^2 and v_L^2. The variability parameter or squared coefficient of variation, is defined as $\text{Var}(T)/E^2(T)$.

The service is characterized by:

1. An average service time $E\{X\}$ (or the average service rate $\mu = 1/E\{X\}$) and v_X^2.

2. The service discipline is assumed to be FIFO.

The basic methodology due to Whitt [1], is to merge all traffic flows (characterized by the above four parameters) entering an input interface of a node or queue into one combined flow. After service, the parameters of the departure flow, different from those of the input, can also be approximately characterized. This four parameter characterization can be used for all flows within the network. The output of QNA gives the mean and variance of the delay at each queue by using approximate GI/G/m formulas.

The mean waiting time of the queue is approximated by

$$E\{W\} = g\frac{\rho E\{X\}(v_T^2 + v_X^2)}{2(1-\rho)} \tag{1}$$

where W is the packet waiting time of a queue; λ is the packet arrival rate to the queue; X is the packet service time of the queue, which is equal to the packet length L (in bits or bytes) divided by the service rate R_s (in bits or bytes per second) of the queue, L/R_s; $\rho = \lambda E\{X\}$ is the load on the queue; v_T^2 is the squared coefficient of variation of the packet inter-arrival time; v_X^2 is the squared coefficient of variation of the packet service time, which is equal to that of the packet length v_L^2. Also g, which is a function of the traffic and service parameters for $v_T^2 < 1$, is $g = 1$ if $v_T^2 > 1$.

Once the traffic parameters for a queue have been estimated, we can use formula (1) to determine an approximation to the mean waiting time. After the queue service, we want to know the departure traffic parameters, which are input parameters to the next queue. If there is no loss in the queuing system, the packet rate, the mean packet length and the variability parameter of the packet length are the same as those of the input traffic parameters, but the variability parameter of the packet inter-departure time. v_D^2, is changed.

Modeling Delay

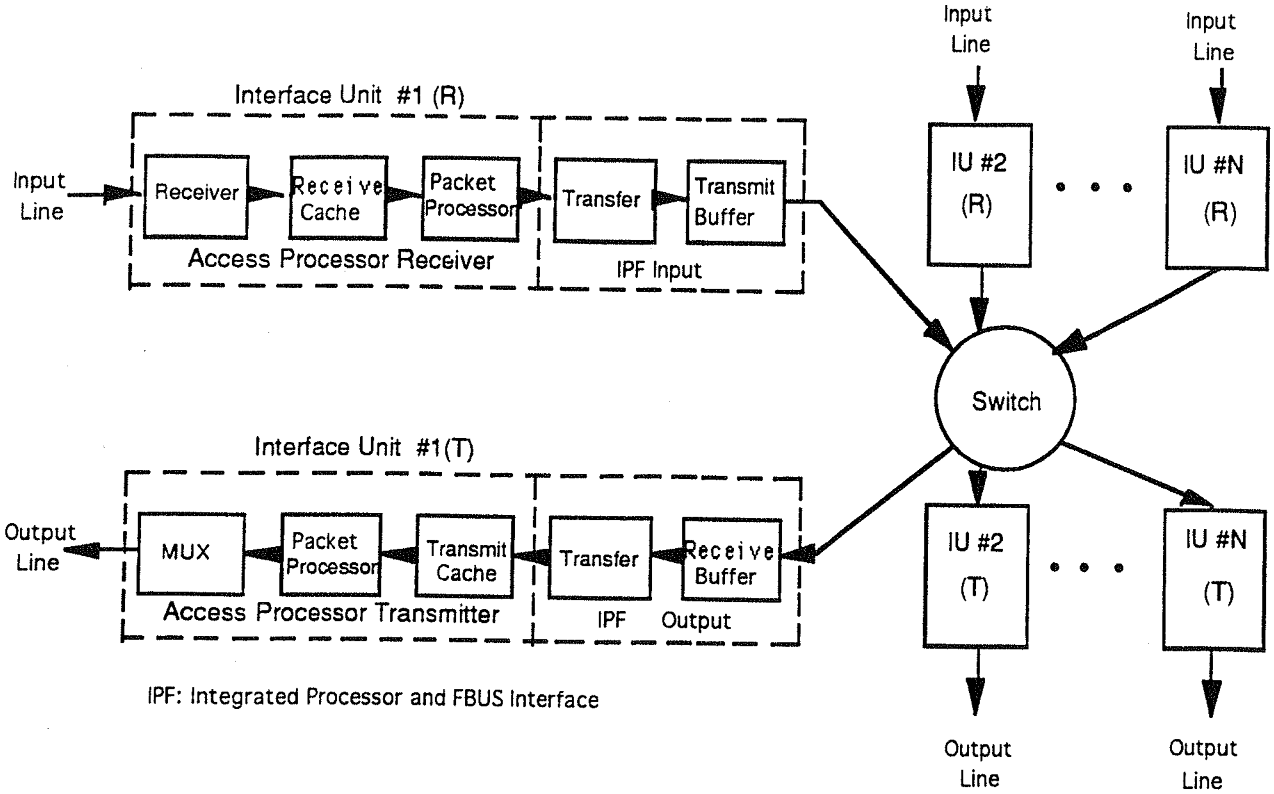

Figure 2: Switch Transport Process

Many switches are based on shared bus or polling architectures and can be modeled by using a round robin service discipline (i.e., the server visits each station in order and serves at most one packet during each visit). A typical switch of this type is shown in Fig.2. It consists of three basic modules: an input, a switch and an output module. An input packet (or a frame of data) from a CPE access line or a trunk connected to the switch is detected and verified by an Access Processor (AP) in the input module. Valid packets are forwarded to the transmit buffer of the cyclic server and wait for the switch server to switch them to their destination output module.

Traffic to a switch node either comes from a previous node through a trunk or enters at the node through an access line from CPEs. Each traffic stream is charactized by the four parameters mentioned previously. When traffic streams in an input line are combined, the QNA method [1] is used.

The input and the output modules are packet processors, which are simple GI/G/1 queues. The switch module is modeled as a non-exhaustive cyclic queue with at-most-one packet served per polling of the queue. We combined some basic methods and developed a better delay model for the switch. We first discuss the switch model for Poisson input traffic and then modify it for general input traffic.

The Cycle Times of Queue i

Kuehn's approximations,

$$c''_{\sim i} = \frac{s_0 + h_i}{1 - \rho_0 + \rho_i} \tag{2}$$

$$c'_{\sim i} = \frac{s_0}{1 - \rho_0 + \rho_i} \tag{3}$$

Our approximations,

$$c''_{\simeq i} = \frac{s_0 + h_i(1 - d_1) + d_3}{1 - \rho_0 + \rho_i(1 - d_1) + d_2} \tag{4}$$

$$c'_{\simeq i} = \frac{s_0}{1 - \rho_0 + \rho_i(1 - d_1) + d_2} \tag{5}$$

Define C''_i to be the conditional cycle time random variable for queue i, given that a packet from queue i is served in the cycle; C'_i to be the conditional cycle time random variable for queue i given that no packet from queue i is served in the cycle. And c''_i and c'_i are their mean values. Kuehn [2] has approximate formulas (2) and (3) for these mean cycle times, where $\rho_0 = \sum_{i=1}^{N} \rho_i$ is the total switch load; $s_0 = \sum_{i=1}^{N} s_j$ is the mean of the total polling time (or total switch-over time). The relationship among the cycle times is $c''_{\sim i} \geq c''_i \geq c_0 \geq c'_i \geq c'_{\sim i}$, where $c_0 = s_0/(1-\rho_0)$ is the mean cycle time.

We found approximations (4) and (5) for c''_i and c'_i, where $d_1 = \sum_{j=1}^{N} \rho_j/(1+\rho_j)$; $d_2 = \sum_{j=1}^{N} \rho_j^2/(1+\rho_j)$ and $d_3 = \sum_{j=1}^{N} h_j \rho_j/(1+\rho_j)$, by solving the following eqations,

$$c''_{\simeq i} = s_0 + h_i + \sum_{j \neq i} \delta''_j h_j \tag{6}$$

$$c'_{\simeq i} = s_0 + \sum_{j \neq i} \delta'_j h_j. \tag{7}$$

Here, $\delta''_j = \lambda_{2j} c''_{\simeq j}$ and $\delta'_j = \lambda_{2j} c'_{\simeq j}$. Equations (4) and (5) also are upper and lower bound for the conditional cycle times.

By combining those two approximations, a better approximation for the conditional cycle times can be acquired. Let $\epsilon''_{ij} = \min(\lambda_i c''_{\sim j}, \lambda_i c''_{\simeq j}, \delta''_i, 1)$ and $\epsilon'_{ij} = \min(\max(\lambda_i c'_{\sim j}, \lambda_i c'_{\simeq j}, \delta'_i), 1)$, which gives a better approximation for the conditional probabilities that a packet from queue j is served in a cycle, given that a packet from queue i is served in the cycle or given that no packet from queue i is served in the cycle. Then, the cycle times $c''_{\cong i}$ and $c'_{\cong i}$ can be aquired by replacing δ''_j and $\delta_{j'}$ with ϵ''_{ij} and ϵ'_{ij} in (6) and (7).

Mean Waiting Time of the Cyclic Queues

Method One:

$$E_1\{W_i\} \approx \frac{r_0}{1-\lambda_i c''_{\cong i}}. \tag{8}$$

$$r_0 \approx \frac{(1-\rho_0)C_{NE}}{\sum_{i=1}^{N} \frac{\rho_i(1-\rho_0-\lambda_i s_0)}{1-\lambda_i c''_{\cong i}}} \tag{9}$$

Method Two:

$$E_2\{W_i\} = b \cdot E_K\{W_i\} \tag{10}$$

For GI/G/1 Queue,

$$E_G\{W_i\} = g_i\left(\frac{c''_{\cong i}\rho_i(v_{T_i}^2-1)}{2(1-\rho_i)} + E_2\{W_i\}\right) \tag{11}$$

A packet upon arrival to an input queue will find either that there is a head-of-line (HOL) packet or there is none. It sees different residual times in those two cases. Based on renewal theory and M/G/1 queue theory, Kuehn [2] derived his formula for the mean waiting time for queue i, $E\{W_i\} = c'^{(2)}_{\sim i}/(2c'_{\sim i}) + \lambda_i c''^{(2)}_{\sim i}/(2(1-\lambda_i c''_{\sim i}))$. Another formula developed by Boxma and Meister [3] is, $E_{BM}\{W_i\} \approx r/(1-\lambda_i c''_{\sim i})$ $r \approx (1-\rho_0)C_{NE}/((1-\rho_0)\rho_0 + \sum_{i=1}^{N}\rho_i^2)$, which is based on two assumptions: 1. $p_i = \lambda_i c''_{\sim i}$ is the utilization observed at queue i; 2. All arrival packets see approximately the same mean residual time r. r is a parameter determined by the conservation law, where $C_{NE} = \rho_0 \sum_{i=1}^{N} \lambda_i h_i^{(2)}/(2(1-\rho_0)) + \rho_0 s_0^{(2)}/(2s_0) + s_0(\rho_0 + \sum_{i=1}^{N}\rho_i^2)/(2(1-\rho_0))$. The errors in the assumptions induce errors in the mean waiting time. If either one could be improved, the errors would be reduced.

The first method is to use $c''_{\cong i}$ as the approximation for the conditional cycle time instead of $c''_{\sim i}$ in Boxma-Meister's formula, then the mean waiting time (8) gives a better approximation, especially when the switch load is high and queues are highly unbalanced, due to a more accurate approximation for the mean cycle time.

A second method is to use Kuehn's formula as the approximation for the mean waiting time, but replace $c''_{\sim i}$, $c'_{\sim i}$, $c''^{(2)}_{\sim i}$ and $c'^{(2)}_{\sim i}$ with $c''_{\cong i}$, $c'_{\cong i}$, $c''^{(2)}_{\cong i}$ and $c'^{(2)}_{\cong i}$. The result is given in (10), where b is a parameter to be determined by the conservation law, $b = C_{NE}/(\sum_{i=1}^{N}\rho_i(1-\lambda_i c_0)E_K\{W_i\})$ In most situations, the second method gives better approximations than other methods.

Load	Simulation	Method 1	Method 2	BM	Kuehn
0.2	.4047(.012)	.4077(.74)	.4135(2.1)	.4073(.64)	.3860(-4.6)
0.4	.9959(.029)	.9975(.16)	1.026(3.0)	.9932(-.27)	.8683(-13.)
0.6	2.548(.068)	2.479(-3.)	2.562(.54)	2.436(-4.4)	1.941(-3.1)
0.8	11.13 (.25)	10.40 (-3.)	10.62(-2.)	9.602(-10.)	6.917(-35.)
0.85	22.18 (.65)	21.62(-2.5)	21.94(-.1)	18.82(-15.)	13.43(-39)
0.9	139.9(23)	163.2 (17.)	164.4(18.)	95.7(-32)	91.7(-34)

Table 1: Waiting Times for the First Queue

Load	Simulation	Method 1	Method 2	BM	Kuehn
0.2	.3814(.013)	3.800(-.5)	.3766(-1.3)	.3801(-.34)	.3576(-6.2)
0.4	.8225(.031)	.8383(1.9)	.8219(-.07)	.8411 (2.3)	.7282(-11)
0.6	1.651(.044)	1.735(5.1)	1.692(2.48)	1.768(6.7)	1.421(-16.)
0.8	3.469(.086)	4.325(25.)	4.272(23.5)	4.935(42.)	2.800 (8.7)
0.85	4.255(.038)	5.959(40.)	5.931(39.0)	7.790(83.1)	5.856(37.6)
0.9	5.335(.056)	9.006(69.)	9.069(70.)	22.83(328.)	11.88(123.)

Table 2: Waiting Times for the Second Queue

Tables 1, 2, 3 and 4 show some numerical results for delay in a 4 queue system with constant polling times $s_i = 0.05$ and exponential packet service times with means $h_i = 1$. The arrivals to the queues are Poisson with average rates $\lambda_1 = 2\lambda_2 = 4\lambda_3 = 4\lambda_4$. These values were chosen to yield an unbalanced system for which the load at queue 1 is twice that at queue 2 and four times the load at queues 3 and 4. Tables 1, 2 and 3 give the waiting time of the first queue, the second queue, and the third and fourth queues respectively. The first column is the total switch load. Simulation results from a special purpose simulation program are given in the second column, with the 95th percentile confidence interval range shown in parentheses. The mean waiting time delay for each method and their percentage errors (compared with the simulation results) are shown in the remaining columns. The results show that improved methods one and two give better waiting time approximations than both Boxma-Meister's method and Kuehn's method. The difference between method 1 and 2 is not large.

Load	Simulation	Method 1	Method 2	BM	Kuehn
0.2	.3780(.013)	.3665(-3.0)	.3584(-5.2)	.3671(-2.9)	.3413(-9.7)
0.4	.7487(.024)	.7655(-2.2)	.7277(-2.8)	.7709(-2.9)	.6478(-13.)
0.6	1.307(.023)	1.441(10.3)	1.336(-2.2)	1.484(13.5)	1.140(-13.)
0.8	2.259(.022)	2.906(28.6)	2.655(17.5)	3.422(54.8)	2.567(13.6)
0.85	2.569(.024)	3.621(40.9)	3.314(29.)	4.911(91.2)	3.653(42.2)
0.9	2.910(.027)	4.723(62.)	4.351(50.)	12.31(323.)	6.270(115.)

Table 3: Waiting Times for the Third and Fourth Queues

Load	Q1 Sim	Method 1	Method 2	Q2 Sim	Method 1	Method 2	Q3,4 Sim	Method 1	Method 2
0.2	.560(.015)	.510(-8.9)	.516(-7.9)	.446(.016)	.433(-2.9)	.430(-3.7)	.407(.024)	.393(-3.4)	.385(-5.4)
0.4	1.66(.067)	1.32(-21)	1.35(-19)	1.11(.023)	1.01(-9.0)	.990(-11)	.887(.016)	.850(-4.1)	.812(-8.4)
0.6	5.74(.226)	3.39(-41)	3.47(-40)	2.74(.084)	2.19(-20)	2.14(-22)	1.71(.032)	1.66(-2.9)	1.55(-9.1)
0.8	29.7 (6.17)	14.4 (-52)	14.6(-51)	6.98(.628)	5.73(-18)	5.67(-19)	3.24(.094)	3.47(7.1)	3.22(-.62)
0.85	61.3 (8.31)	29.8(-51)	30.1(-51)	9.48(.451)	7.96(-16)	7.93(-16)	3.80(.089)	4.36(15)	4.05(6.6)

Table 4: Waiting Times for Non-Poisson Arrivals

In the preceding discussion, the mean waiting times of the cyclic server queues are for Poisson arrival traffic. If the input traffic is general with two parameters, λ_i and $v_{T_i}^2$, the formula for the waiting time must be modified. According to the GI/G/1 theorem [1], the mean waiting time, $E_G\{W_i\}$, for general input traffic can be approximated by (10),where $g_i = \exp\left(-\frac{2(1-\rho_i)(1-v_{T_i}^2)^2}{3\rho_i(v_{T_i}^2+v_{L_i}^2)}\right)$ if $v_{T_i}^2 < 1$ and $g_i = 1$ if $v_{T_i}^2 > 1$; $\rho_i = \lambda_i c''_{\cong i}$ is the utilization observed at queue i. Equation (11) gives us the waiting time for general input traffic.

Table 4 shows the results for the same system as Tables 1 to 3 except that the arrivals are not Poisson. The inter arrival times have a squared coefficient of variation equal to 2. For simulation, a hyper exponential distribution was used to generate the inter arrival times. No comparison can be made to the Kuehn and Boxma-Meister results since these are not applicable to non-Poisson traffic. The results are not as accurate as for the Poisson arrival case, but are adequate for engineering purposes.

The traffic parameters after the switch can be estimated, and used as the input parameters for the output queueing modules.

Equivalent Capacity for Single and Multiple Sources

A) A single source model is based on a two-state fluid-flow model, with idle and burst periods defined as the times during which the source is idle or active, respectively.

B) A traffic source is completely characterized by three parameters, the peak rate R_{peak}, the utilization ρ defined as the ratio of the average traffic rate to the peak rate and the average burst period b.

C) For a single source and a server with a finite buffer size B and capacity c, the loss probability ϵ can be calculated.

D) An equivalent capacity $\hat{c}$ for a server with one source can be obtained by expressing c for a specified loss probability ϵ.

E) For multiple sources, an equivalent capacity $\hat{c}_i$ for each of the sources can be obtained using (14) and the three parameters for each source. The total equivalent capacity (i.e., the capacity needed to keep within the desired loss probability) is

$$\hat{C}_{(F)} = \sum_{i=1}^{N} \hat{c}_i. \tag{12}$$

A fluid flow approximation technique was used to characterize, for a given grade of service(GOS), both the effective bandwidth requirement of a single connection and the aggregate bandwidth needed by multiplexed connections. The purpose of such an expression is to provide a computationally simple approximation for the "equivalent capacity" or bandwidth requirement of both individual and multiplexed connections based on the GOS [4].

C) The ϵ has the form:

$$\epsilon = \beta \times \exp\left(-\frac{B(c - \rho R_{peak})}{b(1-\rho)(R_{peak} - c)c}\right) \tag{13}$$

where β is a function of ϵ, ρ, c and R_{peak}.
D) Since β is typically close to 1 (in fact, it is always smaller), we can approximate β by 1 in (13) and get the explicit upper bound for $\hat{c}$ given by

$$\hat{c} = \frac{\alpha b(1-\rho)R_{peak} - B + \sqrt{[\alpha b(1-\rho)R_{peak} - B]^2 + 4B\alpha b\rho(1-\rho)R_{peak}}}{2\alpha b(1-\rho)} \tag{14}$$

as the equivalent capacity for a server with a single source, where $\alpha = \ln(1/\epsilon)$.
E) The major advantage of (12) is its computational simplicity and its explicit dependence on source parameters. However this linear characteristic is only accurate when $\beta \simeq 1$, so the sum is typically an overestimate of the capacity needed.

Conclusions

- Objective: Develop a Network Management Tool
- Model Customer Traffic and Network Switches
- Route Traffic based on Delay and Loss Measures
- Display Network Status and Performance

This paper has described an approach being used to develop a tool to enable network operators to better manage large communication networks. This tool is based on the ability to model the network switches for delay and packet loss as well as customer traffic for both its average demand and burstiness. These models are then used in a general routing procedure, described in [5], to obtain optimal paths along which to route traffic through the network. This routing algorithm takes into account both delay and packet loss objectives to route the traffic. A graphical interface, not reported on here, is also being developed in conjunction with the models which enables the network operator to see the network and its performance and to accept or override the actions proposed by the tool. We are currently working on extending the applicability of the model to new services and switches.

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