RESOURCE DIMENSIONING IN A GENERAL PACKET-SWITCHED NETWORK ENVIRONMENT

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Abstract—**This paper consider optimization of resource dimensioning in data communication networks, with the aim to provide Quality of Service (QoS) while maximizing resource utilization. The concept of QoS is briefly reviewed. The Differentiated Services (DiffServ) environment is introduced. By means of simulations optimal Provisioning Factors were derived. For simulations the Markov Modulated Poisson Process (MMPP) model was used for UDP traffic, and the Multi-fractal Wavelet Model (MWM) model for TCP traffic. These choices were based on network traffic phenomena like self-similarity, long range dependency and burstiness. A test bed was developed and the traffic models were used to simulate single class traffic and to derive optimal Provisioning Factors (PF) for bandwidth dimensioning for four traffic classes. These Provisioning Factors were evaluated in simulated multi-class traffic. Results conclude that 87% link bandwidth utilization can be achieved while complying with QoS parameters defined for the 4 concerned traffic classes.**

Key Words—**Quality of Service, resource dimensioning, DiffServ, traffic characterization, self-similarity, provisioning factors**

I. INTRODUCTION

Modern communication networks have to support a diversity of protocols for multiple applications over a single network infrastructure. Traffic such as voice, video, data, conferencing, and FTP is carried over a single network, using packet-based transmission.

These applications all have different constraints. For example, real-time data like voice traffic does not require a very high throughput to give adequate performance, and can even tolerate some packet loss, but the demand on delay variation is very strict. Other applications such as the interaction with large databases require a high throughput and strict loss bound, while latency is not significant. Performance measures such as these are collectively referred to as the *Quality of Service* (QoS).

Resource utilization has become a major concern for service providers since it directly affects their profit margins. Current research in this field strives to improve resource utilization while still complying with the required specified QoS guarantees. The goal of our research is to derive optimal resource dimensioning values for a specific dimensioning model by means of suitable simulations.

It is difficult to obtain a general description of network traffic, and hence the desired optimization has to be obtained by means of computer simulations. For these simulations it is important to have good models of network traffic. Various traffic models [1] have been proposed over the last three decades. In our research we have found that it is best to utilize a combination of various traffic models for the simulation experiments.

The paper is organized as follows: In Section II the concept of QoS is introduced, together with a paradigm for classifying data according to the application. Section III reviews DiffServ as a QoS mechanism. Section IV is devoted to traffic characterization and modeling. In Section VI we present simulation results, and Section VII concludes the paper.

II. QOS AND COS

A. *Quality of Service(QoS)*

QoS is sometimes confused with Availability of Service (AoS). AoS describes the time intervals when a specific service is available. On the other hand, QoS is the quality of the service during the times when it is made available [2].

QoS can be described as the state of the measurable characteristics of a network, such as error rate, transmission rate, delay, and delay variation.

B. *Class of Service (CoS)*

The diverging performance needs described above have motivated researchers to aggregate traffic into several distinct classes, to enable differentiated handling over the network. Such an aggregation of traffic, in conjunction with its treatment across the network, is referred to as the *class of service* (CoS).

Based on traffic classes defined by the Internet Engineering Task Force (IETF) [3], Telkom SA defined 4 distinctive traffic classes: Real Time (RT), Interactive Business (IB), Bulk Business (BB) and General Data (GD) [4]. When the traffic load is low compared to the network's capability, all traffic classes perform equally well. However, during times of congestion the higher priority classes receive superior treatment to ensure that the QoS is maintained. The QoS of lower priority classes is degraded which is generally not a problem, since applications utilizing these classes can tolerate a bigger delay and lower throughput. Figure 2 illustrates this effect for the four CoS of Telkom SA.

The mechanism used to reach this class based performance differentiation is discussed in the next section.

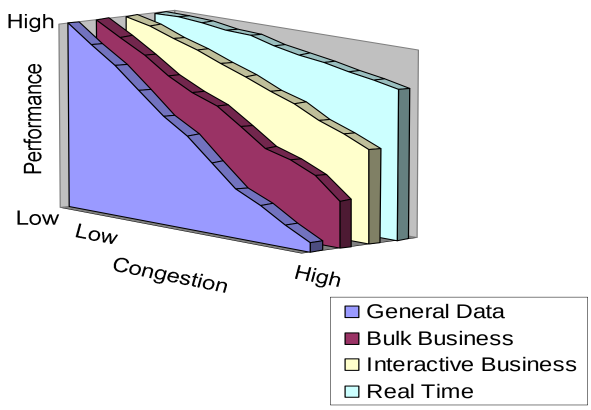

Figure 2. Performance variation for different CoS in a congested network

III. DiffServ

Differentiated Services (DiffServ) is the most commonly used QoS mechanism. However, as it is relative new bandwidth dimensioning for services in a DiffServ environment can be improved upon.

A. *Functionality of DiffSev*

The complete DiffServ architecture was defined in RFC 2475 [5]. DiffServ makes use of Multi Protocol Label Switching (MPLS), a protocol designed to switch, rather than to route. Packets are only regulated at ingress nodes and are then switched at high speed over Label Switched Paths (LSP's). Figure 3 illustrates the connectivity and terminology of the DiffServ domain.

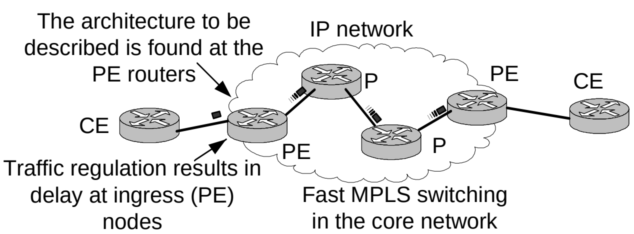

Figure 3. The DiffServ domain
CE – Customer Edge Router; PE – Provider Edge Router; P – Provider Core Router.

At the Provider Edge (PE) router of a DiffServ domain packets are classified, marked, shaped (policed), queued, and scheduled, as illustrated in Figure 4.

The *classifier* identifies traffic and classifies it into four (or five) different classes according to the upper layer protocols, and fields in the IP header, or alternatively using the Experimental (EXP) field in the MPLS header. With regards to DiffServ the most important field in the IP header is the DiffServ Code Point (DSCP). The DSCP is used to specify the quality requirements of each packet. In the DiffServ environment it is also used to identify the CoS to which the packet is belonging.

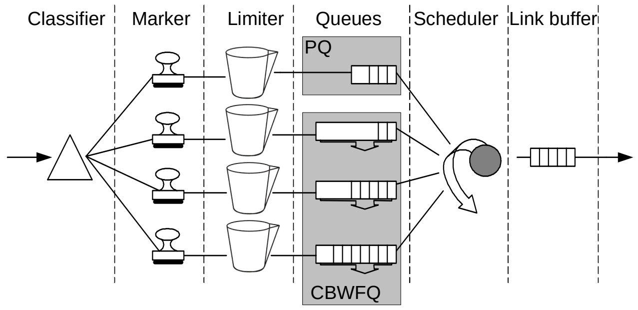

Figure 4. DiffServ PE node architecture.

When a packet enters a DiffServ domain the *marker* sets the DSCP according to the packet's relative importance in the traffic profile. The *rate limiter* could be a *policer* or a *shaper*. A policer is used for the top priority class and drops any out-of-band packets. For lower priority classes a shaper is used. The shaper queues packets rather than dropping them. This is done because the lower classes can tolerate some delay, while the top priority class cannot (or very little so). Many different *queuing* mechanisms exist. In this paper Low Latency Queuing (LLQ) is applied. LLQ is the combination of a priority queue for the top class and Class Based Weighted Fair Queuing (CBWFQ) for the remaining classes. The *scheduler* is essentially the link server programmed to serve the queues according to their priority and fair bandwidth share.

B. *Dimensioning of DiffServ*

QoS is provided by means of the functionality described above, combined with sophisticated provisioning of bandwidth. Telkom South Africa has defined a method of static bandwidth allocation for the DiffServ environment [4][6]. Good utilization and performance is possible if an effort is made to perform the most effective dimensioning process.

During the dimensioning process Provisioning Factors (PF) are used to allocate bandwidth. A PF is a unique numerical value for each traffic class. The PF is multiplied with the requested bandwidth in order to adhere to the QoS specified for the particular class [7]. The traffic is statistically multiplexed, also known as inter-class bandwidth sharing. This allows the service provider to overlap and share the bandwidth of differing classes when the total link bandwidth is determined. Figure 5 illustrates this bandwidth allocation and sharing approach.

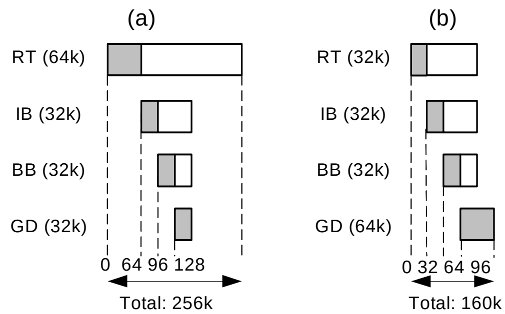

Figure 5. Bandwidth allocation for multiple traffic classes.

Figure 5 shows two scenarios that illustrate how increased demand in a higher class results in a larger required bandwidth. For this example the PF values are arbitrarily chosen as 4, 3, 2 and 1 respectively for the four Telkom QoS classes. In Figure 5(a) the RT class demands 64 kb/s while all the other demand 32 kb/s. In Figure 5(b) The GD class demands 64 kb/s while and the other classes demand 32 kb/s. In both cases a total of 160 kb/s are requested by the customer, but in 5(a) 256 kb/s has to be allocated, while in 5(b) only 160kb/s is required. This indicates the need for more bandwidth to satisfy high QoS guarantees.

The choice of the PF values is critical since it will directly affect the QoS as well as resource utilization, which greatly influences customer satisfaction as well as network provider profitability. Research has shown that the tradeoff between QoS and resource utilization is directly affected by the burstiness of the traffic. For very bursty traffic the task of proper resource dimensioning becomes more challenging.

Figure 6 shows how delay is affected by an increase in utilization. It also indicates that 100% utilization can be achieved. However this occurs only in the special case of an ideal traffic source, i.e. a traffic source generating a continuous flow of packets. As the source becomes bursty, it becomes increasingly difficult to achieve good utilization and hence the delay increases. Typical network traffic is far from ideal and this makes it impossible to achieve 100% utilization. Figure 6 illustrates this concept.

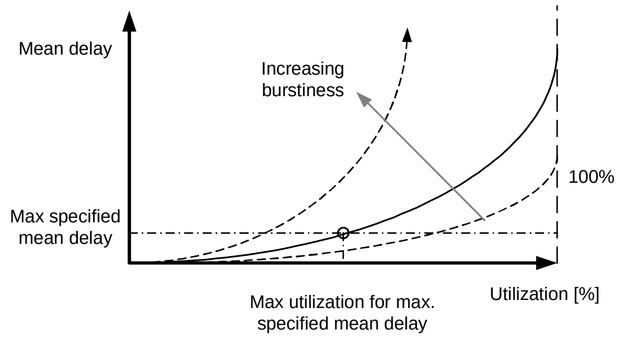

Figure 6. The tradeoff between utilization and mean delay.

Results similar to those for the mean delay in Figure 6 can also be obtained for other QoS parameters. The aim of this paper is to derive and evaluate optimal Provisioning Factors (PFs) that give the best resource utilization for a specified QoS. Current research has not yet led to adequate dimensioning algorithms to determine optimal PFs, and hence simulation methods will be applied. As indicated in Figure 6, the degree of burstiness plays an important role in the tradeoff between utilization and QoS. It is thus important to carefully consider the characteristics, in particular the burstiness, of the specific class-based application traffic that is used in traffic analysis. The identification and modeling of certain traffic characteristics is thus a crucial part of the simulations. The next section is devoted to this important topic.

IV. Traffic Characterization and Modeling

A. Traffic characteristics

Previous research [8] has shown that network traffic exhibits phenomena that are not adequately covered by simple traffic models, including the following:

- *Self-similarity:* Describes the phenomenon that network traffic "looks" the same at different magnifications of the time scale. Usually the Hurst-parameter is used to parameterize the degree of self-similarity [8].
- *Long-Range Dependency (LRD):* Measures the degree to which a certain traffic activity pattern is repeated after a specific time lag. The degree of LRD is expressed by means of the autocorrelation function [9].
- *Burstiness:* Expresses the ratio of the maximum arrivals per sample period to the average arrivals per sample period, as observed over a certain time interval [4].

B. Traffic modeling

Traffic models can be divided into two main categories [10]: Short Range Dependant (SRD) and Long Range Dependant (LRD) models [6]. SRD models include the following:

- Poisson-related processes,
- Markov modulated processes,
- Transform-Expand-Sample (TES) models, and
- Regression models.

LRD models include the following:

- Fractional Brownian Motion,
- Fractional Autoregressive Integrated Moving Average (FARIMA), and
- Wavelet based models.

Previous research indicates that SRD models alone are generally inadequate for modeling network traffic [10][11]. TCP traffic exhibits LRD characteristics, while UDP traffic is best described by a SRD model at a fine time-scale granularity. Therefore, two different models were used for simulations. Extensive work in [10] indicates that the Multi-fractal Wavelet Model (MWM) is best for TCP. On the other hand UDP traffic is best modeled as a Markov Modulated Poisson Process (MMPP). The reader is referred to [10] for a detailed description of the MWM, and [11] for the MMPP.

V. Simulation and Results

A. Simulation Strategy

Vendors cater for individual customer requirements by combining different classes into unique packages. Therefore, single class simulation was first performed to derive dimensioning values for each traffic class, independent of the other classes. The dimensioning values that were obtained from the single class simulations were used in multi class simulations, to evaluate their performance in an environment of statistical multiplexing. The maximum attainable utilization was also measured under these conditions.

B. QoS requirements

The ITU-T has defined delay and loss specifications for all network applications [12]. Furthermore, Telkom SA has aggregated applications into four service classes. The IETF specifications were used to define QoS guarantees for these classes, as shown in Table 1.

TABLE 1. CLASS-BASED DELAY AND LOSS SPECIFICATIONS

Class of Service	Delay	Loss
Real-time (RT)	65ms OW	≤ 1%
Interactive Business Data (IB)	250ms RTT	≤ 0.25%
Bulk Business Data (BB)	400ms RTT	≤ 2%
General Data (GD)	N/a	n/a

OW – One Way   RTT – Round Trip Time

The aim of our simulation, as discussed in the following section, was to find the lowest PF values, while still adhering to the QoS specifications given in Table 1.

C. Simulation environment

Matlab v6.1 R12 was used for the simulations, since it can accept user defined traffic models [13]. Test beds were designed and implemented in Matlab for both the single and multi class simulations.

D. Single class simulation

Figure 7 shows the experimental setup for the single class simulations. The input variables (discussed below) were varied two dimensionally while the measurements, shown in Figure 7, were recorded as a function of time. Sampling periods of 6ms were used. The queue length was used to determine the delay occurring at each time interval. Dropped packets were accumulated throughout the simulations to determine the percentage loss at the end of the simulation. The total number of transmitted packets was then used to calculate the link utilization for each PF value.

The unused bandwidth measurement was only used in the multi class simulation to determine the available bandwidth for lower classes, after the higher priority classes were served.

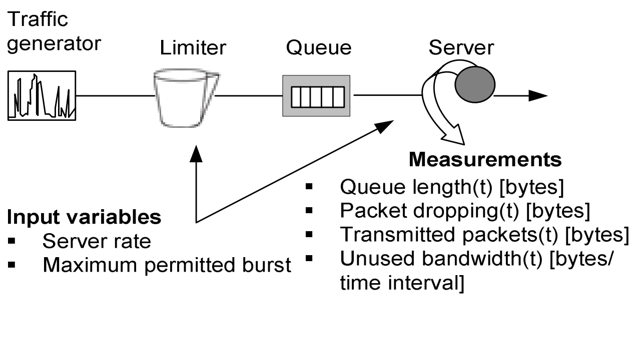

Figure 7. Experimental setup for single class PE node architecture simulation

The applicable traffic generator was used for each class. It was assumed that RT data consists mostly of UDP traffic and therefore the MMPP model was used to generate the traffic for this class. For all other classes, hosting mostly TCP traffic, the MWM was used.

The purpose of the limiter in Figure 7 is to permit, but simultaneously limit, bursts in the traffic stream. The operation of the rate limiter is often described by the token bucket metaphor. Tokens are added to the bucket at a constant rate. Each token permits a certain number of bytes into the queue. As long as there are sufficient tokens in the bucket, packets may enter the queue to be transmitted. In order to limit the size of the bursts the bucket can only accommodate a specified maximum number of tokens, known as the maximum permitted burst (MPB). The MPB is also used as a dimensioning parameter, together with the PF. For more details on token buckets, refer to [14].

Single class simulation was performed as follows:

- A complete traffic trace was generated by the applicable traffic model. The total packet arrivals and size in bytes per 6ms interval were stored in a matrix.
- The average data rate was computed and multiplied with each PF value to be evaluated.
- A nested loop was used to run the complete trace through the test bed for the following:
  - six different PF values ranging from 1 to 5.
  - eight different MPB values ranging between 500 and 20000 bytes (depending on the average rate).
- For each PF-MPB combination the delay and loss measurements were recorded.
- At the end of one entire simulation the delay and loss values were plotted versus the PF values. The different MPB values were plotted as different line types.

A typical single class simulation result is shown in Figure 8. In Figure 8(a) and 8(b) the four line types represent increasing Maximum Permitted Burst (MPB) sizes (as shown in the legend) For illustration purposes only four MPB values were selected from the large number of different MPB values that were actually tested.

In Figure 8(a) the 99% cumulative delay is shown as a function of the PF values, while 8(b) gives the loss versus the PF. The specified minimum loss and delay is indicated on the graphs, and the corresponding PF and MPB values are recorded as indicated in Figure 8. The simulation results show that a PF value of 2.395 and a MPB value of 10 000 are sufficient to give a delay limit of 80ms and a loss limit of 0.08%.

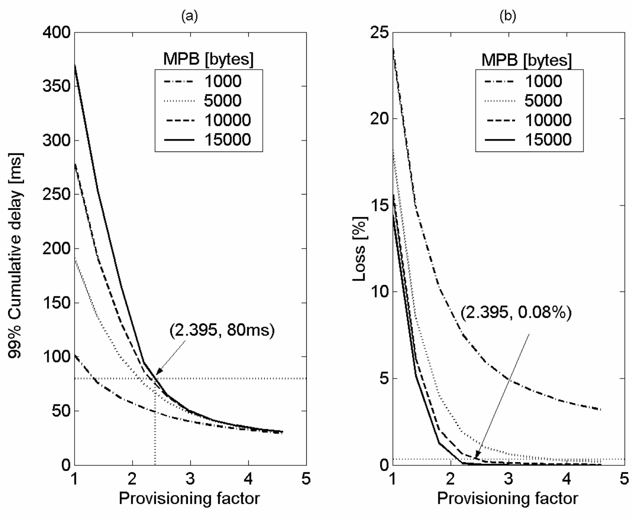

Figure 8. Delay and loss as a function of PF and MPB

These simulations were repeated 50 times and resulted in 50 PF and MPB values, as shown in Table 2 and 3. Results for the single class simulation are discussed in the following paragraphs.

E. Provisioning Factor

The results for the PF values for the various traffic classes are shown in Table 2. These results adhere to the delay and loss limits given in Table 1. The GD class has not been included since no guarantees can be given for this class. The small variance in the PF values indicates the high degree of confidence that may be associated with these results

TABLE 2 SIMULATION RESULTS FOR PROVISIONING FACTORS

	Minimum	Maximum	Mean	Std. deviation
RT	3.25	3.75	3.48	0.12
IB	2.86	3.33	3.10	0.10
BB	1.96	2.42	2.19	0.09

The results shown in Table 2 were verified by applying these values in the dimensioning model and running long traffic traces through the simulation model. Without exception they proved to be adequate. Note however that dimensioning is done at a layer 3 level and therefore these values do not cater for the overheads of layers 1 and 2. Overhead is typically technology dependent, and can be expressed as a percentage of the bandwidth. It is possible to cater for layer 1 and 2 overhead by increasing the experimental PF values by the same percentage.

F. Maximum Permitted Burst (MPB)

In the simulations the MPB values were also recorded during each simulation, and graphs similar to the ones in Figure 8 were created. However, the 50 simulation results for the MPB are very scattered. This can be attributed to the bursty nature of network traffic. However, for practical applications a specific value has to be allocated, and hence available theoretical results were utilized [15]. Equation (1) below shows the MPB in terms of the maximum permitted delay (D), the Provisioning Factor (PF), and the Committed Information Rate (CIR), as agreed upon in the SLA:

$$MPB = PF \times D \times CIR \tag{1}$$

Maximum and minimum values were obtained from the scattered simulation results, and are compared with the theoretical values given by equation (1). This comparison is shown in Table 3.

TABLE 3 MPB VALUES AS A FUNCTION OF THE CIR

	Minimum	Theoretical value	Maximum
RT	0.083	0.140	0.220
IB	0.166	0.248	0.390
BB	0.182	0.286	0.375

In general, the theoretical value may be used. However, it is recommended to use the following policy: If the delay bound is more important than the loss bound, as in the case of real-time applications, a low MPB value should be used. On the other hand when loss is more critical, a high MPB value is recommended. This approach can be explained as follows. When large bursts are permitted to a queue that is being served, little or no packets are discarded, which results in low loss figures. However, the average queuing delay will increase. On the contrary, if only small bursts are permitted the delay will improve due to the shorter queues, but more packets will be lost.

G. Multi class simulation

Single class analysis does not include the effects of statistical multiplexing. Therefore, multi-class simulations were performed to evaluate the PF values that were obtained from the single class experiments, whilst taking into account the effect of statistical multiplexing. Figure 9 illustrates the experimental setup for the multi class simulations.

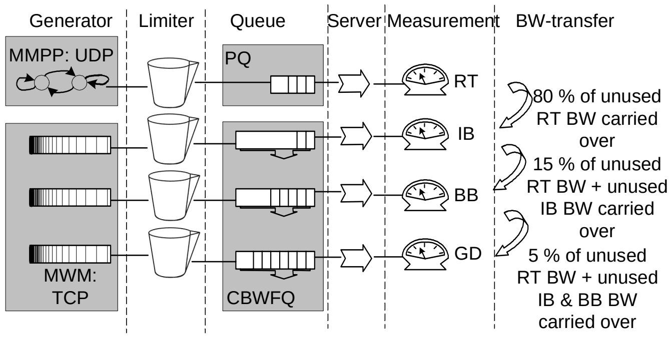

Figure 9. Multi class PE node architecture for simulating bandwidth requirements.

Figure 9 illustrates how the MMPP and MWM models were used for the various classes. It also shows the bandwidth distribution among the classes. Multi class simulations were performed as follows:

- Four traffic traces were generated, using the relevant traffic models.
- The server rate was determined as explained in Section IV, using the mean arrival rate and PFs obtained from the single class simulations.
- The available bandwidth for each class was determined as a percentage of the total server rate and the remaining bandwidth of superior classes, as shown in Figure 9.
- For each 6ms sampling period the delay and loss measurements were recorded, together with the fractions of used and unused bandwidth.
- The results for entire simulation included the per-class delay and loss figures, as well as the total link bandwidth utilization.

Without exception, the recorded loss values of the multi class simulations adhered to the specified limits. However, this observation did not apply to the results for the delay. Only the two top classes, RT and IB, performed within the required delay specification, as shown in Figure 10.

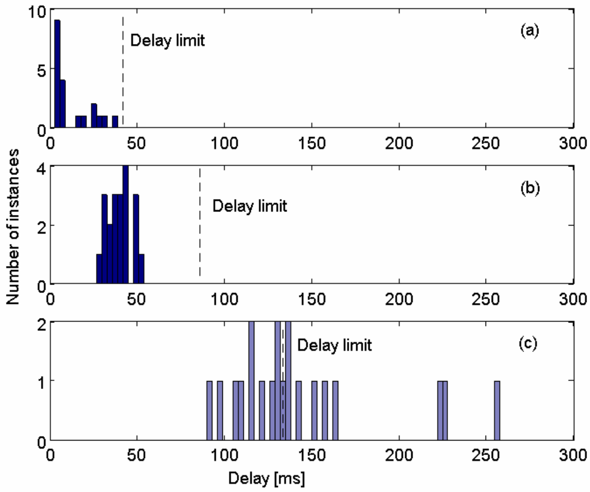

Figure 10. Histograms of delay measurements for (a) RT, (b) IB and (c) BB traffic classes

The BB class did not always remain within the delay limit. However, this problem is not due to a lack of available bandwidth, but rather due to inadequate bandwidth distribution among the classes. This is based on the observation that the other two classes, particularly the IB class, over-performed during the same time periods when the BB class was under-performing. It is thus recommended that the bandwidth distribution among the classes should be revised, or alternatively, that the quality guarantees of the BB class are adapted.

The obtained 87% bandwidth utilization compare very favourably with the 76% that is obtained with the PF values currently used by Telkom.

VI. CONCLUSION

The demand for guaranteed QoS in general packet-switched networks is becoming increasingly important. DiffServ is currently the most popular mechanism to achieve QoS. Characterization of modern network traffic is a challenging task due to the heterogeneous nature of the traffic sources that share a single network. Two traffic models were used to simulate network traffic: the Multi-fractal Wavelet Model (MWM) for modeling TCP traffic, and the Markov Modulated Poisson Process (MMPP) for UDP traffic.

Simulations were performed based on the DiffServ edge nodes, using applicable QoS specifications. The goal of the simulations was to optimize the PF values while complying with specified QoS parameters. In doing this, optimal resource utilization was achieved along with QoS guarantees.

Simulation results included numerical values for class-based Provisioning Factors (PF), and a policy for choosing the most appropriate Maximum Permitted Burst (MPB) values. Single class simulations were first performed to derive PF and MPB values for each individual class. These results were tested to determine the effect of statistical multiplexing in multi class simulations. The results were verified and it was confirmed that the delay and loss results comply with the QoS requirements.

By using the obtained values bandwidth utilization was improved with more than 10% as apposed to values currently being used in practice. It is noteworthy that at a bandwidth utilization of 87% it was still possible to provide all the required QoS guarantees.

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