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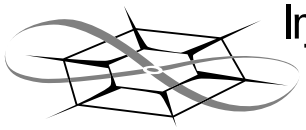

Information and Telecommunication Technology Center

Technical Report

Impact of Traffic Aggregation on Network Capacity and Quality of Service

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Abstract

The impact of traffic handling mechanisms on network capacity and supporting of Quality of Service (QoS) in the Internet is studied. The emergence of applications with diverse throughput, loss and delay requirements requires a network that is capable of supporting different levels of service as opposed to the single best-effort service that was the foundation of the Internet. As a result the Integrated Services (per-flow) and Differentiated Services (Diffserv) models have been proposed. The per-flow model requires resource reservation on a per-flow basis while the Diffserv model requires no explicit reservation of bandwidth for individual flows and instead relies on a set of pre-defined service types to provide QoS to applications. Flows are grouped into aggregates having the same QoS requirements and the aggregates are handled by the network as a single entity with no flow differentiation. We refer to this type of handling as semi-aggregate or class-based. The Best-Effort model does not perform any differentiation and handles all traffic as a single aggregate. Each of these traffic handling models can be used to meet service guarantees of different traffic types, the major difference being in the quantity of network resources that must be provided in each case. The cross-over point at which the three approaches of aggregate traffic management, semi-aggregate traffic management and per-flow traffic management become equivalent is found. Specifically, we determine the network capacity required to achieve equivalent levels of performance under these three traffic management approaches. We use maximum end-to-end delay as the QoS metric and obtain analytic expressions for network capacity based on deterministic network analysis. One key result of this work is that on the basis of capacity requirements, there is no significant difference between semi-aggregate traffic handling and per-flow traffic handling. However Best-Effort handling requires more capacity that may be several orders of magnitude greater than per-flow handling.

Chapter 1

Introduction and Motivation

When the Internet first came into being it was used primarily as a research tool and delivered uniform best-effort service to all users. The majority of traffic carried at this time was primarily data, which did not have very stringent requirements on timely delivery. During the last decade the Internet has evolved into being more of a commercial entity than a research network and has experienced tremendous growth in both the volume of traffic carried as well as diversity in the type of traffic carried. The engineering philosophy behind the Internet was based on the model of a homogenous community that had common interests rather than on a model of service providers and customers [49]. The best-effort Internet can be considered as consisting of just one user group in which everyone is allowed to use the network for any purpose and limits are imposed only when the capacity is not enough to satisfy demand. It is also assumed that all users behave agreeably during times of congestion by limiting their usage. The major tool that was used to engineer the Internet was over-engineering (often referred to as "throwing bandwidth at the problem") which refers to providing more bandwidth than the aggregate demand so that every subscriber is given ample access to network resources. The recent growth in network usage both at the commercial and public level coupled with the advances in high-speed applications however tends to stretch the limits of over-booking as more and more customers are demanding and using more bandwidth from the networks while at the same time having high expectations on the service that they receive.

The emergence of applications with diverse throughput, loss and delay requirements requires a network that is capable of supporting different levels of service as opposed to the single best-effort service that was the foundation of the Internet. Quality of Service (QoS) has become the buzzword and um-

brella term that captures the essence of this shift in paradigm. IP Telephony is a good example of an application that is driving the push towards QoS on the Internet and is in fact being touted as today's killer application for the Internet [44, 77]. Latency rather than bandwidth is the primary issue in providing voice services in the Internet, thus the traditional approaches of simply over-engineering may not work as well for this type of application. To provide a network that caters to these different levels of service requires changes to network control and traffic handling functions. Control mechanisms allow the user and network to agree on service definitions, identify users that are eligible for a particular type of service and let the network allocate resources appropriately to the different services. Traffic handling mechanisms are used to classify and map information packets to the intended service class as well as controlling the resources consumed by each class. Notable results of the effort to provide Quality of Service in the Internet are the definition of Integrated Services and Differentiated Services by the Internet Engineering Task Force (IETF) [7, 8, 9, 21, 45, 46] and Asynchronous Transfer Mode (ATM) by the ATM Forum [3].

The Integrated Services per-flow model uses resource reservation to provide delay and throughput guarantees. The per-flow model is based on the idea that bandwidth must be explicitly managed in order to meet application requirements therefore resource reservation and admission control are a must [9, 10]. Advocates of the per-flow model claim that high fidelity interactive audio and video applications need higher quality and more predictable service than that provided by the best-effort Internet and that this can only be achieved through explicit resource reservation [12].

The Differentiated Services model takes a different approach from the per-flow model in that it does not promote the use of resource reservation. Proponents of Diffserv argue that a simple priority structure will be sufficient to provide Quality of Service in the Internet. One of the arguments against resource reservation is that in the future bandwidth will be infinite, therefore there will be no need for reservations. Advances in fiber-optic communication may suggest that bandwidth will be so abundant, ubiquitous and cheap that it will not benefit network operators to undertake resource reservation however, one cannot ignore the fact that increases in available bandwidth are always followed by corresponding development of applications that consume and exhaust this bandwidth [9, 35]. Trends in the history of communications indicate that regardless of how much bandwidth is made available, applications are always created that quickly exhaust the supply.

Another argument against resource reservation models is that simple priority will be sufficient to meet the needs of real-time traffic. This may be

true under some conditions but not always. For instance with fixed network capacity if the number of high priority real-time transmissions increases then they will all have degraded performance. A third argument against resource reservation is that it is too expensive because reservation of resources is wasteful in that not all the reserved resources are used. This is true if all of the resource is exclusively reserved and thus it must be ensured that there is a limit on how much guaranteed traffic is allowed and provisions must be made for non-real time traffic to utilize bandwidth unused by real-time traffic [35]. Lastly, it has been suggested that delay bounds are not necessary and throughput bounds are enough. However, guaranteeing minimum throughput does not automatically result in better delay performance. Delay bounds must be explicitly guaranteed and enforced.

Opponents of reservation contend that the issue boils down to one of provisioning and that reservation-enabled networks can only provide satisfactory service if the call blocking rate is low. It is believed that by adequate provisioning, a best-effort network can achieve the same performance as a reservation-based network [12, 34]. As an example consider IP telephony users who require the network to guarantee to carry their calls with a maximum end-to-end latency that is no larger than 100msec. If an IP network is provisioned to accommodate N users simultaneously with the end-to-end latency within 100msec, an increase in traffic beyond N would result in the service of all the current users being degraded and the resources wasted since no user would attain acceptable performance [60]. Thus, significant over-provisioning is required. The higher the quality of guarantee, the more over-provisioning that must be done for the same level of user satisfaction and hence the lower the efficiency of network utilization. Consequently the quality of guarantees must be traded-off against the efficiency of network resource usage. The case for over-provisioning is that declining prices in bandwidth will make the extra capacity required in a best-effort Internet more economical than the complexity and increased network management to support reservations.

Neither a pure best-effort model such as the current Internet, nor a pure guaranteed service model such as the Integrated Services model can provide an efficient solution in a multiple service environment [49]. Having a large number of service classes increases the management overhead and impairs cost efficiency. An integrated network must balance the trade-off between performance and flexibility while ensuring that performance of traffic with real-time guarantees is not degraded. Providing QoS in the Internet requires providers to re-evaluate the mechanisms that are used for traffic engineering and management in their networks. Over-engineering is an attractive option because it is simple and it has been said that within a well-defined scope

of deployment it can prove to be a viable solution [34]. Recent proposals are calling for more active traffic management in the Internet that will be used to make more efficient use of resources while allowing providers to offer varying levels of service suited to the different applications being supported. These traffic management mechanism range from simple admission policies to complex queuing and scheduling mechanisms within routers and switches.

We can envision several alternative paths for carrier networks to follow in their quest to provide QoS. These are:

1. Inefficient use of network bandwidth with no traffic management. This approach assumes that bandwidth is abundant and cheap and thus traffic management is not needed.
2. Moderately efficient use of network bandwidth with simple traffic management
3. Efficient use of network bandwidth with complex traffic management. With this approach the assumption is that the cost of bandwidth justifies the use of traffic management.

Knowledge of the network capacity required to achieve comparable user perceived performance will indicate the importance of traffic management as the network evolves. For example, if an aggregate network capacity of 10Gb/s is needed given no traffic management while only 100Mb/s is needed when the traffic is controlled, then the cost of traffic management can be justified. However, if the difference in required capacities is "small" then it may not be time to deploy complex traffic management functionality. There is a need for a clearer understanding of the issues surrounding the provision of QoS in IP-based networks as well as guidelines on how traffic management and network capacity can be used to provide QoS.

In this thesis we consider the issue of finding the point at which the three approaches of no traffic management, simple traffic management and complex traffic management become equivalent. Specifically, we would like to determine the network capacity required to achieve equivalent levels of performance under a variety of traffic management schemes. This knowledge would help network engineers and decision-makers determine the suitability of IP QoS traffic management as well as the type of traffic management to use.

In Chapter 2 we provide a discussion on the correspondence between traffic management schemes and traffic aggregation and consider some of the questions that need to be addressed in comparing traffic management strategies.

We also provide a formal statement of the problem to be addressed by this thesis. Chapter 3 provides a review of the current literature that relates to this work. In Chapter 4 we provide a review of the theoretical foundations of the thesis while Chapter 5 provides an overview of the analytic framework in terms of the network applications and traffic management schemes that were studied. Chapter 6 discusses the methodology and results for the simple case of a single-link while Chapter 7 extends this to carrier-size networks by considering different combinations of traffic handling mechanisms in the edge and core of the network. Chapter 8 builds on the analysis in Chapters 6 and 7 to derive bounds on the capacity requirements that may be easier to use. In Chapter 9 we apply the analysis to networks that implement path aggregation and show the relationship between link utilization, end-to-end delay and network capacity. Chapter 10 provides a methodology and results for sensitivity analysis of the traffic handling schemes. We end with conclusions and directions for future work in Chapter 11.

Chapter 2

Traffic Aggregation, Quality of Service and Network Capacity

The Internet's need to support traffic with diverse requirements and with differing levels of service coupled with the transition of the Internet from a research network to a commercial one has resulted in the re-definition of the Internet's architecture. The major change is in the definition of new services and traffic handling mechanisms that can be used to provide differentiated and guaranteed quality of service in the Internet. The challenge facing the deployment of integrated services is to satisfy the strict delay and loss guarantees required for real-time services while realizing the economics of statistical multiplexing which are essential for high-speed bursty data. One objective is to be able to support both voice, video and data traffic on one network in such a way that the performance of voice is equivalent to that on a Public Switched Telephone Network (PSTN) network.

Providing guaranteed QoS today can be achieved in one of three ways. The first technique is to over-provision the network which is the classical "throw bandwidth at the network" solution. This is based on the premise that bigger bandwidth pipes mean less congestion and hence better performance. The second alternative is to reduce delay by introducing the notion of precedence and treating certain types of traffic with higher priority than others. Delay for higher priority traffic in this case will be better than lower priority best-effort but will depend on the traffic load in each priority level. The last technique is to use dedicated resources for each flow in the network, recently referred to as "throwing hardware at the network". This gives the most predictable performance [6, 49].

The above solutions can be related to the level of aggregation of flows used

by traffic handling mechanisms within the network. We define three levels of aggregation as shown in Figure 2.1. As can be seen from the figure, in a

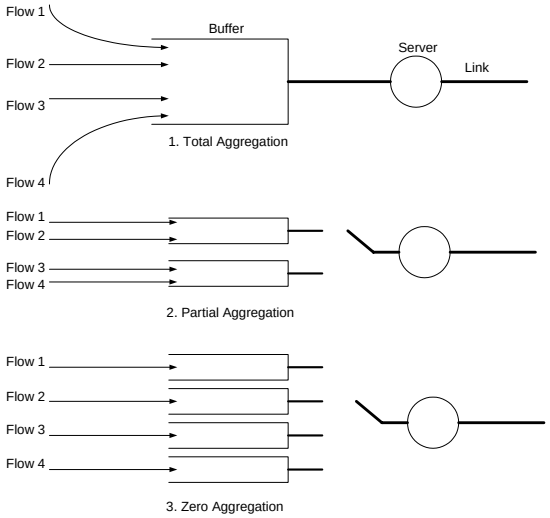

Figure 2.1: Levels of Aggregation

total aggregation environment, all flows are enqueued in the same buffer and share the buffer and link resources. This is the simplest and most prevalent form of traffic handling. The link must be configured with enough capacity to meet the most stringent QoS and the typical approach to maintaining QoS in this situation is to add more capacity to the link - "throwing more bandwidth".

In the partial aggregation environment, flows are divided into classes based on some criteria, the most obvious one being to group flows with similar QoS requirements. In this way, the QoS needs of a class of flows can be ensured in isolation from other flows. This type of aggregation corresponds to the precedence solution. In an environment with zero aggregation, each flow is assigned its own set of resources and thus attains its QoS independent of other flows. This is the best means of ensuring QoS but it is also the most complex to administer. This environment corresponds to the dedicated resources solution. The common term for zero aggregation is per-flow queueing.

Scheduling mechanisms are used to achieve the levels of aggregation that we have outlined. Total aggregation can be achieved with First-In-first-Out (FIFO) scheduling in which packets are served in the order of arrival to a queue. For partial aggregation Priority Queueing (PQ) and Class Based Queueing (CBQ) are typical approaches. Priority Queueing imposes a strict service order by assigning each queue to a fixed priority level and serving the queues accordingly. With Class-Based Queueing, flows are mapped to

classes based on some predefined attribute and service weights are assigned to each class. Per-flow queueing can be implemented using (Weighted) Fair Queueing, (Weighted) Round Robin and their many variants.

Given the levels of aggregation and the associated scheduling mechanisms which we couple under the umbrella term of traffic handling, the question we address in this thesis is that of determining the equivalence of the different traffic handling mechanisms in terms of their ability to support traffic with varying QoS requirements. Of particular interest is the trade-off between the complexity of traffic handling mechanisms and the network capacity required to support QoS.

In addition to the traffic aggregation in traffic handling, the solution to providing QoS depends on the network capacity. It is widely accepted that the use of aggregate schemes may necessitate the provisioning of more network capacity than per flow schemes but it is not clear just how much more capacity is needed nor is it clear how the complexity of per-flow management measures up against the cost of additional capacity with aggregate traffic handling. To provide an adequate answer to this problem requires some quantification of the gain in performance obtained by using complex traffic handling with smaller network capacity versus using simple traffic handling with abundant network capacity. A pertinent issue also has to do with the sensitivity of the selected solution to changes in network conditions such as load or delay requirements. Suppose that using aggregate traffic handling requires high capacity links but the resulting network is insensitive to fluctuations in network traffic whereas using a complex scheme with limited capacity results in a network that is very sensitive to network variations, what would be the better option? It is issues such as these that need to be addressed.

Based on the foregoing discussion, four objectives have been identified. The first objective is to examine the trade-off between complexity of traffic handling and the required network capacity by comparing the bandwidth required for a given level of performance under traffic handling schemes that range from complex to simple. A second objective is to determine to what extent the analytical methods we intend to use are able to scale with network size and capacity and what modifications if any must be made to ensure that they do. In evaluating the performance under different traffic handling schemes we must ensure that the analysis is robust and scalable. Results obtained should be consistent in any network topology or configuration. If the analysis is not robust or scalable then it will provide results that are misleading. A third objective is to provide insight into how connection-less networks such as the Internet can be used to support traffic with diverse QoS

requirements and to provide the analytic framework for deciding on a traffic handling and capacity provisioning strategy. A final objective is to study the sensitivity of the traffic handling algorithms to changes in network load and traffic mix.

We anticipate two main results from this thesis. The first result is a quantification of the trade-off between complexity of traffic management and network capacity. Such a quantification would take the form of a graph showing the trend in the capacity requirements of the different traffic handling requirements. The simplest representation is the capacity required by the three traffic handling models for the same network load and performance as shown in Figure 2.2.

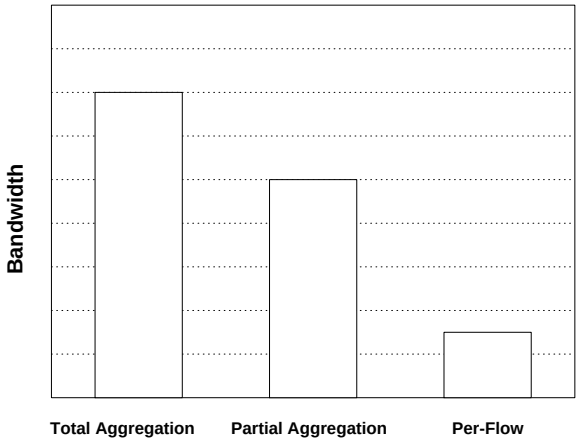

Figure 2.2: Simple Traffic handling and Network Capacity Trade-off

From Figure 2.2 we can obtain quantification of the extra bandwidth required by aggregate schemes when compared to a per-flow scheme. By taking measurements of the required capacity for equivalent performance over a variety of network loads we can obtain a graph that shows how the difference in performance depends on the network load (level of utilization in the network). A hypothetical example of such a plot is shown in Figure 2.3.

In this figure, we plot the difference in capacity (delta_C) of three traffic handling schemes A,B,C as a function of network load with reference to a per-flow scheme such as WFQ. From the plot we are able to immediately identify the points and regions where the different mechanisms provide equivalent performance and are also able to assess how this equivalence translates into a difference in network capacity requirements.

A second result that we anticipate is in the difference in sensitivity of the traffic handling parameters to network conditions and one way of illustrating

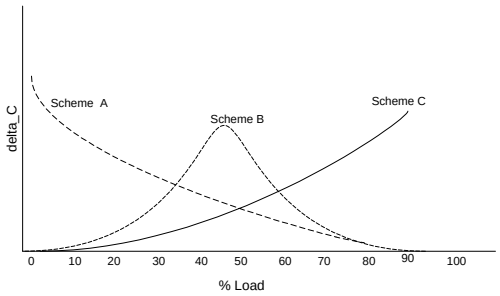

Figure 2.3: Traffic handling and Network Capacity Trade-off with varying Network Load

this difference is as shown in Figure 2.4. In this figure, the design point represents the point at which the delay objectives are satisfied for a given network capacity and load and the figure illustrates how the delay perceived by a candidate traffic class may vary when the network load is varied above and below the design point for three traffic handling schemes. The sensitivity can thus be measured by the ratio of change in delay to change in network traffic and this can be used to determine which scheme is more preferable. It is apparent that we would like to pick the scheme with the least sensitivity especially at loads above the design point and in this case Scheme B would be the likely candidate.

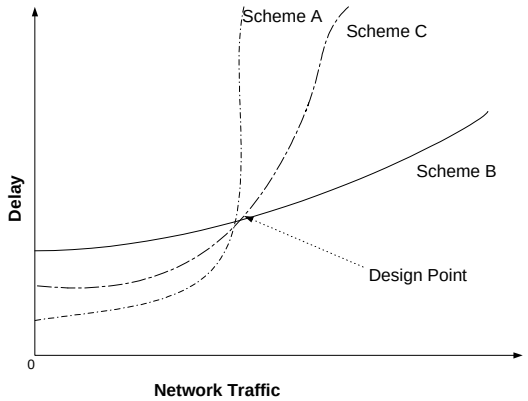

Figure 2.4: Comparison of Traffic Handling Sensitivity

By combining the observations from the capacity-traffic handling trade-off and the sensitivity analysis, we can provide a quantitative answer to the issue of selecting an appropriate traffic handling mechanism that meets the objectives of supporting traffic with diverse requirements in an efficient manner. In the next chapter we look at some related issues and studies that have been published in the literature.

Chapter 3

Background

In this section we provide an overview of existing research and results which are related to this thesis. We begin by looking at Quality of Service and the different service models that are used to define QoS. This is followed by a discussion of scheduling mechanisms and how they can be used to provide QoS. We also touch on the issue of traffic aggregation and how this impacts QoS. Lastly we look at how QoS affects network design.

3.1 Quality of Service

The exponential growth of the Internet and the proliferation of bandwidth-demanding applications coupled with the significance of network availability to business achievements have resulted in the need for providing predictable and consistent system performance [22]. It has also resulted in the creation of a new buzzword within the networking community: Quality of Service. Quality of Service has become one of the most widely used terms in the networking community despite the fact that there is no single definition of the term. In a book on Quality-of-Service, Paul Ferguson and Geoff Huston say [34]:

> "Quality of Service is one of the most elusive, confounding and confusing topics in data networking today. Why has such an apparently simple concept reached such dizzying heights of confusion? After all, it seems that since the entire communications industry appears to be using the term with some apparent ease and with such common usage, it is reasonable to expect a common

level of understanding of the term."

The problem with defining QoS (as it relates to telecommunications) is that it is used by the different players in the industry such as the customers, the equipment vendors, the network engineers, the researchers and the marketers to mean very many different things. As a result it is very difficult to come up with a consensus on what QoS really is. From the network provider's perspective, QoS can be defined in terms of the way in which the services delivered to customers are differentiated based on the allocation of network resources. The customer's definition of QoS can be captured in terms of a utility or benefit function, which relates the customer's perception of quality to the value of that quality [76]. A QoS-enabled network should provide service guarantees appropriate for various application types while making efficient use of network resources. For service providers an important aspect of providing QoS is to classify network applications according to their service needs. The literature abounds in the ways in which traffic is classified and we cite as an example the general classifications [60]:

- Quantifiable traffic requiring high quality guarantees

  Traffic in this category includes IP telephony and other interactive multimedia traffic. The resources required by such traffic are easily quantified and the performance guarantees required are strict so that resources must be explicitly reserved.

- Non-quantifiable persistent traffic requiring high quality guarantees

  Mission critical traffic from client-server sessions falls in this category. Resources still need to be reserved for this traffic using some form of prediction to ensure that strict performance guarantees are met.

- Non-quantifiable, non-persistent traffic requiring low to medium quality guarantees

  Traffic such as this, which includes web surfing, cannot quantify its resource requirements and does not have strict guarantees so that the overhead of resource reservation is unwarranted.

- Best-effort traffic

  This is traffic that is not quantifiable, is not persistent and does not need any service guarantees. Most e-mail and some web-surfing applications fall in this category.

Thus, the challenge facing the deployment of integrated services is to satisfy the strict delay and loss guarantees required for real-time services such as telephony and video-conferencing while realizing the economics of statistical multiplexing which are essential for high-speed bursty data such as e-mail and web browsing [41, 42, 60, 64]. To provide these different levels of service requires changes to network control and data handling functions. Control mechanisms allow the user and network to agree on service definitions, identify users that are eligible for a particular type of service and let the network allocate resources appropriately to the different services. Data handling mechanisms are used to classify and map traffic to the intended service class as well as controlling the resources consumed by each class.

There are currently three main network service models for the delivery of integrated services in high-speed communication networks: ATM, IETF Integrated Services and IETF Differentiated Services. One of the design objectives of ATM was to provide loss and delay QoS guarantees by defining service classes for the different traffic types. Currently, there are five service categories defined [3]. The Constant Bit Rate (CBR) and real-time Variable Bit Rate (rt-VBR) services are intended for real-time traffic that has stringent delay and timing constraints while non-real time Variable Bit Rate (nrt-VBR), Available Bit Rate (ABR) and Unspecified Bit Rate (UBR) are for non-real time traffic with varying degrees of loss and delay assurances. To provide QoS guarantees, the ATM model relies on the use of bandwidth reservation through virtual connections that may be either static or switched.

The IETF Integrated Services model was another attempt at providing QoS in the Internet and is concerned with the time-of-delivery of traffic so that per-packet delay is what determines QoS commitments [9]. There are two types of services defined: the Controlled Load Service(CLS) and the Guaranteed Service (GS). The Controlled Load Service provides better-than-best-effort delivery when the network is lightly loaded while the Guaranteed Service provides real-time traffic with delay constraints guaranteed bandwidth and bounds on delay. Integrated Services relies on the reservation of resources based on dynamic signaling between the sources and the networks. The signaling is based on the resource reservation protocol (RSVP) [10]. One of the concerns with the model is that it requires each node in the network to maintain state on a per-flow basis and this poses some scalability problems for high-speed links supporting a large number of concurrent flows.

The Differentiated Services model takes a different and simpler approach to defining services by using Per-Hop Behaviors (PHBs) which govern the way in which network elements handle traffic [7, 8]. It requires no explicit reservation of resources and relies on priority mechanisms within network

elements to provide QoS to a small number of pre-defined service types. In addition to the priority mechanisms, Differentiated Services also relies on packet classification according to desired service type at the edges of the network. This aggregation of traffic at the edges of the networks reduces the need for nodes in the network core to maintain per-flow state. The Differentiated Services effort represents a renewed interest and focus on simple QoS guarantees by defining services that map to different levels of sensitivity to loss and delay. The reasons for this approach are [42]:

- upgrading the Internet to perform per-flow differentiation is a daunting task that will take a long time and interim solutions are needed.
- deployment of per-flow capabilities will not be widespread initially.
- proper network engineering and broad traffic classification can offer the same functionality as explicit QoS guarantees.
- the number of applications requiring strict guarantees is not significant enough to warrant explicit QoS and provisioning and priority schemes are adequate to provide the guarantees required by these applications.
- applications can be made to adapt to network congestion.

The potential for aggregation provided by Diffserv may prove to be beneficial in the backbone of the Internet by reducing the amount of per-flow state that is maintained. Thus Diffserv provides a scalable architecture but it is hard to provision and does not provide easily quantifiable guarantees.

3.2 Scheduling Mechanisms

Due to the different traffic characteristics and Quality of Service requirements of network traffic the coexistence of voice, video and data in the same network poses new issues in packet scheduling, admission control and bandwidth sharing [20, 42, 58, 84, 95]. In order to provide QoS, there must be classification mechanisms to separate traffic into service classes and there must be buffer management and scheduling mechanisms which handle the traffic from separate flows accordingly. Data traffic is generally very bursty and relatively insensitive to delay but may be sensitive to loss. On the other hand real-time traffic such as voice and video is delay sensitive but can tolerate some loss. Voice is usually less bursty and has smaller bit rates whereas

video is generally burstier and has higher bit rates. This diversity in traffic characteristics and QoS means that traffic should be divided into classes that reflect these attributes and should be treated separately and differently based on the class. Scheduling creates a policy of isolation and sharing.

There are four criteria that can be used when comparing different buffer management and scheduling schemes [42, 93]: fairness, isolation, efficiency and complexity. Fairness refers to the way in which bandwidth is shared among competing flows. Fairness is not a direct measure of QoS, rather it measures how the network assigns resources during periods of congestion [43]. Thus fairness does not capture the user experience and may not be a good indicator of service quality. A better measure might be the number of customers that receive poor service at any time so that the goal of scheduling should be to maximize the number of customers receiving good service during times of congestion. Isolation between flows is needed to protect flows from excess traffic of other sources. The way in which network resources are utilized is captured by the efficiency of the scheduling mechanism and one way of quantifying efficiency is to measure the number of flows that can be accommodated under different scheduling schemes for a given level of service. Complexity refers in part to the processing that is required in the implementation of a scheduler. The ideal scheduler is one that is fair, provides maximum isolation has high efficiency and minimal complexity. The reality however is that all these properties cannot be attained simultaneously and trade-offs have to be made. In general fairness, isolation and efficiency are achieved with complex schedulers. In the sections that follow we consider some examples of common scheduling mechanisms and compare them on the basis of the above metrics and their ability to provide service differentiation.

The First-In First Out (FIFO) scheduler is the simplest mechanism possible in which packets are served in the order in which they arrive. FIFO scheduling by itself does not provide isolation between flows but using buffer management can help to improve the isolation and fairness properties of FIFO [41]. FIFO is the primary model of queueing and complex queueing systems are used only when it is determined that FIFO is inadequate. FIFO can provide high cost-efficiency because the buffers and links are used very efficiently and it is fair if all users behave in the same way and have the same attributes, although this falls short of the requirements of fairness since we desire fairness to prevail even when customers have different characteristics. FIFO does not support service differentiation very well since all customers are treated the same but has the advantage of not requiring any per-flow information to be maintained. Another disadvantage is that it does not easily provide rate or delay guarantees and hence cannot provide fair access to link bandwidth.

The Fixed Priority Queueing Scheduler provides a coarse level of granularity by assigning traffic to a fixed priority level and serving traffic according to its priority. Separate queues are maintained for each class and lower priority queues are served only when higher priority queues are empty. Thus service differentiation is provided through the different priority levels but per-flow guarantees cannot be achieved. Priority queueing can be used to provide differentiation by having each service class in a separate queue. Thus traffic that requires lower delay would be placed in a high priority queue and would have a bounded delay. Delays in the lower priority queues will depend on the traffic in the high priority queue and the maximum delay for lower priority traffic can be bounded by restricting the load of the high priority queue.

In Class Based Queueing (CBQ) traffic is divided into classes based on some criteria such as application type and each class is assigned a proportion of the link capacity with excess bandwidth being shared fairly among all the classes [20, 34, 38, 49]. Different scheduling policies may be used between the classes. Class-based queueing (CBQ) attempts to solve the starvation problem of strict priority queueing in which low priority queues may be denied resources when the high priority load is high. CBQ requires traffic to be classified into relatively large aggregates according to a principle that depends on the service model. In CBQ the importance of a packet depends on the aggregate load level of the class - the larger the number of users the less the importance of individual packets. Thus the quality perceived by a flow depends on the aggregate load level as well as the weight assigned to its class and this makes it difficult to determine precisely the performance that will be perceived by an individual flow.

Latency-Rate schedulers are those that provide both rate and delay guarantees [40, 80, 93]. Notable examples of these schedulers are Weighted Fair Queueing(WFQ), Self-clocked Fair Queueing(SCFQ), Weighted Round Robin and Rate-Controlled Static Priority(RCSP) and their many variants. Fair queueing is a service discipline designed to allocate link capacity among multiple connections sharing a link by distributing bandwidth fairly among all connections and redistributing unused bandwidth fairly among active connections [18]. Details on fair queuing can be found in the paper [27] which illustrates the ability of fair queueing to minimize delays significantly over FIFO by allocating bandwidth fairly to competing flows. In Weighted Fair Queueing(WFQ), the bandwidth allocation is based on some pre-determined weights for each flow or group of flows within an aggregate. A generalization of Weighted Fair Queueing called Packetized Generalized Processor Sharing (PGPS) has received a lot of attention in the research community and has become the standard by which other schedulers are measured [28, 32, 33, 62, 63, 67, 68, 80, 96]. PGPS extends fair queueing by making

the scheduler work-conserving. Each connection i is assigned a proportional rate parameter ϕ_i which determines the minimum guaranteed rate of a connection g_i. Using this approach, the maximum delay of a traffic stream can be bounded based on its own traffic characteristics independent of other streams. PGPS is used to ensure that throughput and delay bounds can be guaranteed for any type of traffic that is regulated provided that the guaranteed rate is greater than the expected long-term average rate of the flow. Thus PGPS provides minimum bandwidth guarantees to each connection, provides deterministic end-to-end delay bounds to traffic that is regulated and ensures fairness in the amount of service provided by a server to competing connections.The RCSP scheduler decouples the rate guaranteeing and bandwidth allocation mechanisms of fair queueing by having a rate controller separate from a priority scheduler [94].

Aggregation refers to the combination of different flows sharing a common path in the network in such a way that individual flows within an aggregate are not visible to network elements. The advantages of aggregation are the reduction in time and space requirements in network nodes but this comes at the expense of losing the isolation between flows which may be necessary to protect flows from each other. The level of aggregation in a network is directly influenced by the scheduling mechanism. Thus using FIFO schedulers results in total aggregation of all flows on the one hand whereas using WFQ schedulers results in no aggregation. Class-based and priority systems provide partial aggregation based on the manner in which classes or priority levels are defined.

For Differentiated Services a fundamental issue is that core nodes should not maintain per-flow information. Thus WFQ on a per-flow basis is not an attractive approach for Differentiated Services since it requires per-flow processing. In addition, it may require the use of a signaling mechanism to adjust weights dynamically when network conditions change. A third concern with WFQ is that it presents a lot of computational effort. For services that have equal priority and roughly equivalent QoS requirements FIFO is a simple and adequate solution. Priority queueing can be used to effectively separate real-time traffic from non-real-time traffic. When the traffic types have different QoS requirements with the potential to overload their allocation and are intolerant of interference from other sources, FIFO and priority queueing become inappropriate and fair queueing schedulers must be used. Using fair queueing schedulers, delay bounds for real-time traffic are met by allocating sufficient bandwidth and using small buffers. In WFQ, delay bounds can be provided but the bounds are coupled to allocated bandwidth. The delay bounds are inversely proportional to the allocated bandwidth and as a result, flows that require low latency must have a large

amount of bandwidth allocated. This can lead to inefficiency when these flows are of low bit-rate [64]. We will now present some results on the use of different scheduling strategies.

- In [87] it is determined that for the same network capacity, a single FIFO queue performs better than a 2-level priority system when the ratio of packet lengths is small. As disparity in packet length increases, the priority system performs better.
- In [65] the research focused on the optimal buffer allocation for given bandwidth for two types of systems:
  - Lossless segregated system: one in which each connection is allocated its own buffer and bandwidth with no resource sharing.
  - Lossless multiplexing scheme: all connections share the same resources.

  They find that for guaranteed lossless services, sources should be divided into groups according to the time scales determined by their leaky bucket parameters. They also observe that sources with slow time scales should be allocated bandwidth equal to their peak rate and no buffer space while sources with fast time scales should be allocated bandwidth equal to their mean rates and buffer space equal to their token bucket size.
- In [69] the authors compare the delay performance of FIFO to WFQ for sources generating Constant Bit Rate(CBR) traffic. They find that for high bandwidth flows the delays with FIFO are two orders of magnitude larger than with WFQ and delays for FIFO decrease significantly with a decrease in utilization whereas WFQ is not affected. They conclude that for networks supporting traffic with the same packet size FIFO is adequate while in networks with variable packet sizes WFQ is more appropriate. At low levels of utilization, the difference between FIFO and WFQ reduces and is not very significant
- The work in [61] uses a priority queueing structure to compare the performance of two types of service: Premium and Assured. The Premium service is a low latency service and is thus given priority over the Assured service which is intended to provide guarantees on throughput rather than delay. Their results show that with Premium service the delay is about 2 orders of magnitude less than Assured service, although the delay with Assured service is still within 100msec which is

considered acceptable for voice. Using Assured service yields more efficient utilization in that more calls can be accepted than using Premium service

- In [86] the use of the controlled load service for support of audio and video services is studied. The key variables in the study are geographical scope of the network, link capacities and reserved traffic load and they assume an architecture in which priority is given to the reserved flows over best-effort traffic. Their results show that there is a tradeoff between link size, packet size and network size. They find that end-to-end delay guarantees are feasible over a wider range of parameters for local and long distance networks compared to transatlantic networks.

- The work in [64] compares Generalized Processor Sharing(GPS), strict priority and FIFO in terms of the admissible region of each policy. The admissible region is the number of sources of each class admitted without violating the QoS requirement. Their results show that when loss probability is the QoS metric, strict priority and FIFO outperform GPS. However, when delay is the constraint, the difference in performance is dependent on the relative traffic mix. When there is more traffic with looser delay constraints, FIFO performs worse than GPS whereas when the number of low QoS traffic decreases, FIFO performs better. The explanation for this behavior is that with FIFO having fewer lower priority sources allows more higher QoS to be queued while under GPS more low QoS sources can be admitted due to their looser constraints

- In [13] a comparison is made between a static priority system and a weighted fair queueing system for support of audio and video traffic using a premium IP service. The general conclusion is that the static priority system performs better than WFQ. This is based on the observation that with static priority, the number of hops before the traffic reaches its maximum delay bound is larger than that with WFQ for networks in which the core link capacities are much greater than edge link capacities.

- In [4] the question of whether to provide a single class of relaxed real-time service using FIFO or multiple levels differentiated by their delay characteristics using priority queueing is investigated. From their results, at low load levels, the priority scheme offers no advantages over FIFO. With increasing load, the benefits of priority scheduling increase. In general the conclusion is that multiple service levels increase the load levels at which the network can satisfy the needs of all classes.

- The authors in [39] examine the use of Rate Controlled Service (RCS) for Intserv Guaranteed services. They also suggest the definition of a service called Guaranteed Rate(GR) which has less stringent delay guarantees and which is served in a WFQ manner only when there is no Guaranteed traffic. The GR traffic improves utilization and provides a service that bounds delays, with the bounds depending on the Guaranteed traffic.

- In [20] the authors conclude that in a single network node carrying flows with the same packet lengths, FIFO provides better jitter performance than WFQ. This is because FIFO shares delays evenly between the flows whereas in WFQ delays are assigned to the flows that send large bursts and cause momentary surges in the queue. Over multiple hops however, the jitter bound for FIFO increases significantly, although it is still better than WFQ.

- In [16] and [17] the authors provide analytical results on end-to-end delay bounds for networks of arbitrary topologies using strict priority schedulers. They conclude that in order to meet delay objectives of high priority traffic, the utilization of traffic in the high priority queue is severely limited by the maximum hop count of the network as well as by the ratio of input to output interfaces at a network node.

- The work in [97] extends that of [16, 17] by considering how utilization can be improved under aggregate scheduling by incorporating timing information in packet headers. This results in two new scheduling mechanisms that schedule packets based on the time-stamps with one scheme being static in that it does not alter the time-stamps while the other dynamically adjusts the time-stamps of each packet. These two schemes are found to improve network utilization over FIFO with the improvement depending on the granularity of the time stamps. The improvement comes at the cost of increased complexity in processing the time stamps.

- The authors in [33] observe that the design of GPS schedulers is based on deterministic QoS guarantees which are overly conservative and may lead to limitations on capacity. In their work, QoS delays are probabilistic and the goal is to maximize the bandwidth available to best-effort traffic while just meeting the guarantees of traffic with QoS requirements. In one case they consider lossless multiplexing in which the QoS guarantee is deterministic and no violation of the QoS delay requirement is allowed. In the second case they consider statistical multiplexing in which there is a small probability that the delay requirement

may not be met. The general result is that the use of statistical guarantees increases the capacity of the network in that it is able to support more sources than in the lossless multiplexing case.

- In [75] the focus is on use of Random Early Discard (RED) with two thresholds as a means for providing throughput assurance to TCP flows. In an over-provisioned network, all flows achieve their target rates but excess bandwidth is not shared fairly. In an under-provisioned network degradation in service is not fair.

- The authors in [70] investigate the use of priority scheduling and threshold dropping to provide loss and delay guarantees. In general threshold dropping requires 30-70% more bandwidth than priority scheduling to provide the same delay performance. When traffic is extremely bursty and a small amount of loss is allowed, threshold discarding performs better than priority scheduling. This is one of the few papers encountered where a comparison is made between FIFO scheduling and Priority queueing on the basis of additional capacity required to equalize their performance.

- In [50] the authors conclude that adequate provisioning is necessary in a FIFO-with-RED network to ensure that QoS guarantees are met and when the network is under-engineered, it cannot meet the requirements.

- In [81, 82] the authors address the issue of whether one can obtain the high utilization, efficiency and isolation of stateful networks using mechanisms that are as scalable and robust as those of stateless algorithms. Examples of stateless solutions to providing QoS are Random Early Drop (RED) and the Differentiated Services model while stateful solutions are weighted fair queueing and the Integrated Services model. The authors propose a technique called Dynamic Packet State(DPS) in which each packet carries in its header information that is initialized by edge routers and used by core routers to process the packet. DPS coordinates the actions of edge and core routers through a distributed scheduling algorithm and allows a network to approximate the performance of a network with per-flow management without incurring the overhead. DPS is associated with a mechanism called Core Stateless Fair Queueing (CSFQ) in which edge routers perform per-flow processing while core routers use FIFO scheduling and do not maintain per-flow state, but use information that is inserted into packets by edge routers (DPS) to perform probabilistic dropping. A comparison of CSFQ to FIFO, Deficit Round-Robin (DRR), Flow Random Early Discard (FRED) and RED is provided on the basis of fairness in bandwidth allocation. In general, CSFQ and DRR had the best performance

with DRR being slightly better followed by FRED,RED and FIFO in that order. If we assume that there is a relationship between allocated bandwidth and delay then essentially these results suggest that more capacity is required with FIFO.

3.3 Network Design

In this section we consider how the provision of QoS impacts the design of communication networks. We begin by noting that QoS mechanisms can be broadly split into signaling mechanisms and data handling mechanisms. Signaling conveys information that relates to call setup and tear-down of the resources required by a call between end-hosts. Signaling can be applied on a per-flow basis as with plain RSVP and ATM or on an aggregate basis as with RSVP tunnels or aggregate RSVP. Data handling can also be done on a per-flow basis as with Intserv or on an aggregate basis as with Diffserv. In providing QoS, there is a trade-off between efficiency in usage of network resources and strictness of QoS guarantees. Figure 3.1 illustrates this trade-off [5]:

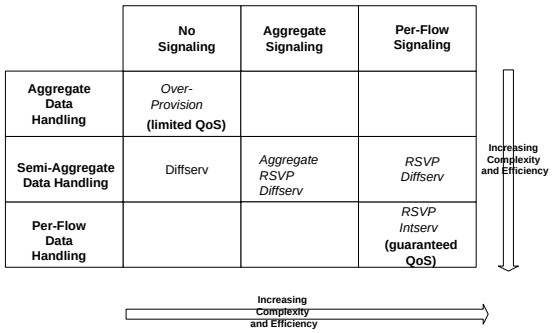

Figure 3.1: QoS trade-off in Communication Networks

From the figure we note that the simplest strategy which is also the least efficient is to perform no data handling and no signaling while over-provisioning the network. On the other end of the scale is the Intserv approach with per-flow signaling and data handling. An integrated network must balance the trade-off between performance and flexibility while ensuring that performance of traffic with real-time guarantees is not degraded. There is also a tradeoff to be made between the cost of bandwidth and the cost of extra mechanisms to provide QoS. One of the issues that arises is that real-time applications operate at time-scales that are much smaller than non-real-time applications thus networks cannot operate at reasonable utilization levels

while providing to all traffic a service that is suitable for real-time traffic [76].

In [48], a "stupid" network is described as one in which control passes from the center(core) of the network to the boundary(edge) so that the center is based on abundant infrastructure - cheap bandwidth and switching- while the boundary uses more intelligent network elements. Congestion in a "stupid" network is dealt with by adding more connections, more bandwidth or more switching power. Some service providers have adopted this model and in [88], an ISP is quoted as saying it does not currently face QoS delivery challenges because it ensures that its network capacity is far in excess of user bandwidth requirements. Another example is given of the network service provider Qwest which does not see the need to deploy CoS or QoS solutions since it has enough fiber and bandwidth to ensure that everyone's traffic is routed at the highest priority. One of the problems with this approach is that the problem is not just about capacity and thus "throwing bandwidth" is not a long-term solution. Network traffic is changing not only in volume but also in its nature and networks need to respond to the changing nature of QoS requirements using more sophisticated traffic handling mechanisms [78].

An emerging model is one in which per-flow traffic handling mechanisms are used in the edge of the network and aggregate mechanisms are used in the core [31, 60, 77, 85]. We will now present some examples of current research relating to network design for the provision of QoS.

- The work of [20] introduces the notion of an Expected Capacity Framework which provides differential service by requiring users to submit to a service profile. The service profile is used to determine which packets are eligible for service when the network gets congested. For the method to work, the core of the network must be provisioned with enough capacity to carry the traffic of outstanding profiles and it is not enough to use a simple summing over all profiles to determine the required capacity. On the other hand traffic inside the core may not be as bursty as at the edges so that the provisioning problem is not as severe.

- The paper in [12] addresses the question of whether the Internet should retain its Best-Effort only architecture or adopt an architecture that supports reservations. They consider the incremental bandwidth that is required to make a best-effort network perform as well as a reservation capable network. The methodology adopted here is to consider network performance in terms of utility - the value that a user obtains from the

network - and use this as the basis of comparison between reservation and best-effort networks. The study restricts itself to identical flows and uses different assumptions on the distribution of network load : random (Poisson and exponential) and deterministic. The trade-off between reservation and best-effort is captured by the relation:

$$R(C) = B(C + \Delta C) \tag{3.1}$$

where R(C) is the normalized utility with a reservation network and $B(C + \Delta C)$ is the normalized utility of a best effort network with incremental bandwidth ΔC. The bandwidth gap ΔC is the amount of additional capacity needed to make a Best-effort network perform the same as a reservation-enabled network. In general, their results indicate that the incremental bandwidth depends on whether the applications are adaptive or non-adaptive, as defined by their utility functions. Adaptive applications require less incremental bandwidth. They also show that the link capacity at which incremental bandwidth is not required depends on the assumptions on the network load probability distribution. With a Poisson distribution in which the network load is tightly controlled, incremental bandwidth ceases to have value at a lower link capacity than with either the exponential and deterministic distributions. In fact for rigid applications which have strict delay requirements, the incremental bandwidth increases with the link capacity for exponential and deterministic network load distributions. The general conclusion is that providing a definite answer to the choice between reservation and best-effort will depend on load patterns in the future Internet.

- The work in [29] addresses the issue of levels of aggregation and the main conclusion is that the division of traffic into two classes, a Real-Time class for audio and video and a non-Real-Time class for data is adequate to meet the stringent delay QoS requirements of the audio and video. The best QoS is achieved when flows with identical characteristics are aggregated. This poses the question as to where to set the limits on what can and should be aggregated. Aggregation strategies range from type aggregation to class aggregation and full aggregation. There is a tradeoff between having a simpler network with a small number of classes and having a more complex network with a larger number of classes. The first case provides good statistical multiplexing while the second provides better isolation but poses scalability problems.

- In [30], the authors compare the "fat dumb pipe" (best-effort) model with a differentiated services model. The fat dumb pipe model uses

over-provisioning to achieve QoS resulting in inefficient network usage. They differentiate between absolute service differentiation in which the network tries to meet the same goals as the Intserv network but without the use of per-flow state and relative service differentiation in which assurances are based on a relative ordering of each application and its requirements.

- In [47], the authors compare the use of flow aggregation with no aggregation for provision of QoS guarantees. They find that flow aggregation systems require more bandwidth than those with no aggregation but that systems with no aggregation are more complex to administer. They also note that the benefits of aggregation increase with the number of flows and as the number of flows increases, the bandwidth required by the aggregate systems approaches that of the non-aggregation system.

- In [36] the authors address the cost versus benefit of using an integrated services intranet vs an over-provisioned best-effort intranet. They find that using a two class network provides marginal savings while a three class network provides 60% savings in capacity. The two class network differentiates between voice and web traffic while the three class network breaks the web traffic into two types: one requiring QoS and the other not.

- In [57] a comparison is made between class-level and path-level aggregation. In class-level aggregation, flows belonging to the same class are queued together and a jitter controller (regulator) is used to ensure that all flows within a class experience the same (maximum) delay. In path-level aggregation, flows which share the same end-to-end path are queued together. They find that the multiplexing gain for class-level aggregation is higher but the use of the jitter controller results in increased delays and requires more buffering. Both schemes are sensitive to the path length with class-level aggregation requiring more bandwidth as the path length increases while for path-level aggregation, the performance deteriorates with increasing path length. They conclude that the better multiplexing gain with the path-level approach is worth the increased delays due to the jitter control.

- In [73], the efficiency due to flow grouping is analyzed. This is based on the observation that aggregation of flows inside the core network will resolve the scalability issues associated with handling numerous flows individually inside the core. Two aspects to aggregation are considered: how should resources be allocated to aggregated flows and which flows should be grouped together. The analysis shows that for homogeneous

flows, aggregation requires less resources than handling flows individually. For flows that have different rate and burstiness parameters and the same delay requirements, aggregation requires more resources.

Per-flow mechanisms handle each flow separately whereas aggregate mechanisms handle multiple flows as one entity. Per-flow handling enhances the quality of service experienced by a flow but it imposes a heavy burden on the network to maintain state about each individual flow. In the core of a network, the number of flows may be in the millions, making such per flow handling impractical. With aggregate handling, the amount of state maintained is reduced significantly. In aggregate handling the QoS seen by an individual flow is thus compromised by the presence of other flows. Over-allocation of resources can help to improve the QoS in aggregate data handling but at the expense of reduced efficiency in network utilization.

The finer granularity of per-flow management has the benefits of fairness and efficiency but at the cost of greater complexity. The choice of trade-off is a function of the scalability requirements of the network environment. In a smaller network, per-flow management may be appropriate but in a larger network, the number of flows may be of such magnitude as to make per-flow management almost impossible and aggregate schemes a more viable option. Scalability requirements are likely to introduce the need for aggregation especially in the core of the backbone where there are a large number of flows, links are high speed and the use of per-flow management may be prohibitively expensive [42, 60, 77].

In this chapter we have seen that there are many facets to providing a complete understanding of the issues surrounding the deployment of integrated services networks. The nature of applications and their quality of service guarantees is one aspect. The type of scheduling mechanisms used and the network architecture are other aspects that must be taken into account. In the next chapter we provide an overview of an analytic method that can be used to determine worst-case bounds on end-to-end delays in networks. We then show how this analysis can be used to compare the capacity requirements of different traffic handling mechanisms in the chapters that follow.

Chapter 4

Network Analysis using Network Calculus

4.1 Principles of Network Calculus

The Network Calculus approach to network analysis is deterministic and does not depend on probabilistic descriptions of traffic unlike most of the research discussed in the last chapter where the results depend on the assumed traffic models. Network Calculus is used primarily with envelope bounded traffic models to provide worst-case analysis on network performance. We have chosen this method of analysis because it allows us to obtain results that can be applied to any type of traffic provided we can bound the traffic process at the input to the network. This method is especially appealing since many services defined by the ATM Forum and the IETF are based on traffic that is regulated before it enters the network [3, 45].

Network Calculus is based on the idea that given a regulated flow of traffic into the network, one can quantify the characteristic of the flow as it moves from element to element through the network. The roots of Network Calculus can be found in the pioneering work of Cruz in which the idea of calculating end-to-end performance bounds using regulated traffic was first introduced [23, 24]. Cruz later extended his work to introduce the notion of service and arrival curves to characterize the quality of service in networks [25, 72]. Following on from Cruz's work other researchers have made significant contributions to formalizing the theory of Network Calculus [14, 15, 52, 54, 55, 56, 74].

Arrival curves are used to describe the input to a network element in that

a flow $x(t)$ is constrained by an arrival curve $A(t)$ if and only if for all times $s \leq t$:

$$x(t) - x(s) \leq A(t-s)$$

Equivalently,

$$x(t) \leq \inf_{0 \leq s \leq t} \{A(t-s) + x(s)\}$$

The simplest arrival curve is $A(t) = Rt$ which would describe a flow with a constraint on the peak rate R. Another type of arrival curve is an affine curve of the form $A(t) = \sigma + \rho t$ where σ is called the burstiness parameter and represents the maximum amount of traffic that can arrive in a burst and ρ is an upper bound on the long-term average rate of the flow. With this arrival curve a source can send σ bits at once but no more than ρ bits/sec over a long period of time. A concise notation for traffic that is regulated in this sense is $A \sim (\sigma, \rho)$. The IETF Integrated Services model uses affine arrival curves, called T-SPECs, of the form [9]:

$$A(t) = min(M + pt, rt + b)$$

where M is the maximum packet size, p is the peak rate, b is the burst tolerance, and r is the sustainable rate. This model corresponds to traffic that is regulated by two token buckets: one for the peak rate and one for the sustainable rate. For the case where only the average rate is regulated, we set the peak rate to infinity. Some ATM services such as CBR and VBR also use regulated traffic which can be represented by arrival curves similar to the IETF model. Figure 4.1 is a graphical representation of the IETF arrival curve model.

For flows that follow the IETF arrival curves, the aggregate arrival rate to a queue is obtained by summing the burstiness and rate parameters. Thus the aggregate arrival rate $A(t)$ for N flows is given by:

$$\begin{aligned} A(t) &= min\left(M_{max} + \sum_{i=1}^{N} p_i t, \sum_{i=1}^{N} r_i t + \sum_{i=1}^{N} b_i\right) \\ M_{max} &= \max_{i=1..N} \{M_i\} \end{aligned}$$

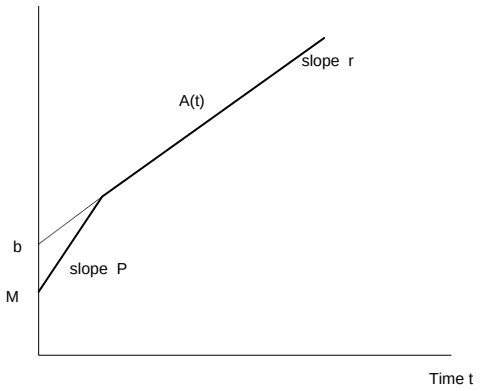

Figure 4.1: IETF Arrival Curve

Service curves are used to abstract the details of packet schedulers. Let $x(t)$ and $y(t)$ be the total traffic input and output respectively of a flow at time t. A system offers a service curve $S(t)$ to the flow if for any t, there exists $s \leq t$ such that the backlog at time s, $B(s) = 0$ and $y[s+1,t] \geq S(t-s)$. An equivalent definition can be obtained by observing that:

$$\begin{aligned} B(s) &= x[1,s] - y[1,s] \\ y[s+1,t] &= y[1,t] - y[1,s] \end{aligned}$$

Then,

$$\begin{aligned} y[s+1,t] &\geq S(t-s) \\ => y[1,t] - y[1,s] &\geq S(t-s) \\ => y[1,t] - x[1,s] &\geq S(t-s) \end{aligned}$$

Thus $S(t)$ is a service curve if for any t there exists $s \leq t$ such that $y[1,t] - x[1,s] \geq S(t-s)$. Note that $S(t)$ must be non-negative, non-decreasing with $S(0) = 0$.

One simple example of a service curve is $S(t) = Rt$ which guarantees that each flow is served at a rate of at least R bits/sec during a busy period. For a FIFO queue this is the service curve that is seen by the aggregate flow. Another example is the rate-latency service curve which guarantees both a

delay and throughput and is used by the IETF Integrated Services model. This service curve is given by:

$$S(t) = R[t-T]^{+} = \begin{cases} R(t-T) & t \geq T \\ 0 & otherwise \end{cases}$$

where T is the latency of the scheduler. An example of a rate-latency scheduler is Packetized Generalized Processor Sharing (PGPS) which is commonly referred to as Weighted Fair Queueing (WFQ) which has a latency $T = L/g + L_{max}/C$ where g is the guaranteed rate for the flow, L is the maximum packet size of the flow being served, L_{max} is the maximum packet size of all flows and C is the link capacity [62].

Another example of a service curve is that of a priority scheduler. In this case we have separate service curves for each priority queue. Assuming P priority levels with $1 > 2 > 3..... > P$ and arrival curves of the form $A(t) = \sigma + \rho t$, the service curve for priority p is given by:

$$S_p(t) = [C - \rho_H(p)]t - [\sigma_H(p) + L_{max}(p)] \tag{4.1}$$

where

$$\rho_H(p) = \sum_{j=1}^{p-1} \rho_j \tag{4.2}$$

$$\sigma_H(p) = \sum_{j=1}^{p} \sigma_j \tag{4.3}$$

$$L_{max}(p) = \max_{j \geq p}\{L_j\} \tag{4.4}$$

where C is the link capacity, ρ_j is the aggregate average rate of priority level j, σ_j is the aggregate burstiness and L_j is the maximum packet size for priority j.

Using the arrival and service curves, bounds are derived that determine the input-output relationship for regulated traffic as it passes through basic network elements. There are three fundamental bounds that are used in the theory of network calculus for lossless systems with service guarantees.

The first bound says that the backlog is bounded by the vertical deviation between the arrival and service curves. More formally, if a flow with input $x(t)$ and output $y(t)$ that is constrained by a service curve $A(t)$ traverses a system with a service curve $S(t)$, the backlog $x(t) - y(t)$ satisfies:

$$x(t) - y(t) \leq \sup_{s \geq 0}\{A(s) - S(s)\}$$

The second bound is on the maximum delay which is given by the maximum horizontal deviation between the arrival and service curves. Formally, for a flow constrained by arrival curve $A(t)$ through an element with service curve $S(t)$, the maximum delay d_{max} is given by:

$$d_{max} \leq \sup_{t \geq 0}\{\inf\{\tau \geq 0 : A(t) \leq S(t+\tau)\}\}$$

Figure 4.2 shows how these two bounds are evaluated using the IETF arrival and service curve models. In the figure d_{max} is the maximum delay

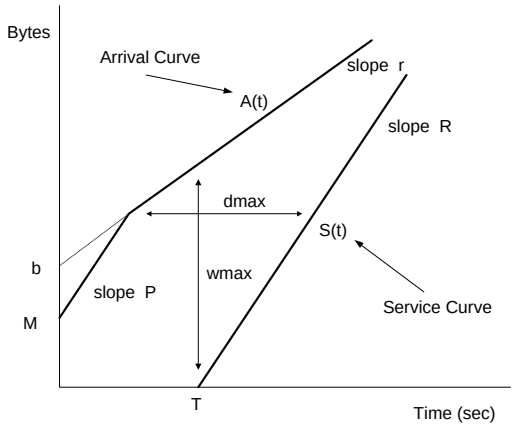

Figure 4.2: IETF Arrival and Service Curves

and w_{max} is the maximum backlog. The last bound applies to the output due to a constrained flow. If a flow with arrival curve $A(t)$ traverses a system with service curve $S(t)$, the output flow is constrained by an arrival curve $A^*(t)$ given by:

$$A^*(t) = \sup_{\tau \geq 0}\{A(t+\tau) - S(t)\}$$

Using arrival curves, service curves and the fundamental bounds the end-to-end delay and other performance measures can be obtained for networks of arbitrary topologies. We will use the examples of FIFO, Priority Queueing(PQ) and Weighted Fair Queueing(WFQ) to illustrate how delay bounds can be calculated for the case of traffic regulated by burstiness σ and rate ρ. We will use the notation $A(t)$ for the arrival curves, $S(t)$ for the service curves and C for the link capacity.

- FIFO

  In this case we have the aggregate arrival curve $A(t) = \sigma + \rho t$ and the service curve $S(t) = Ct$ as shown in Figure 4.3.

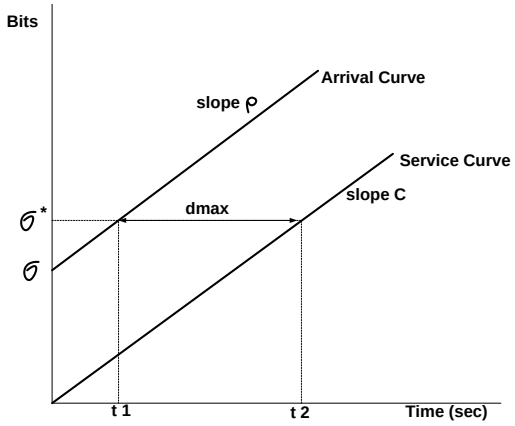

Figure 4.3: Arrival and Service Curves for FIFO

With reference to the figure we have:

$$\begin{aligned} t_1 &= \frac{\sigma^* - \sigma}{\rho} \\ t_2 &= \frac{\sigma^*}{C} \\ t_2 - t_1 &= \frac{\sigma^*}{C} - \left(\frac{\sigma^* - \sigma}{\rho}\right) \end{aligned}$$

From this we find that $t_2 - t_1$ is maximum when $\sigma^* = \sigma$ so that the maximum delay is :

$$d_{max}^{FIFO} = \frac{\sigma}{C} \tag{4.5}$$

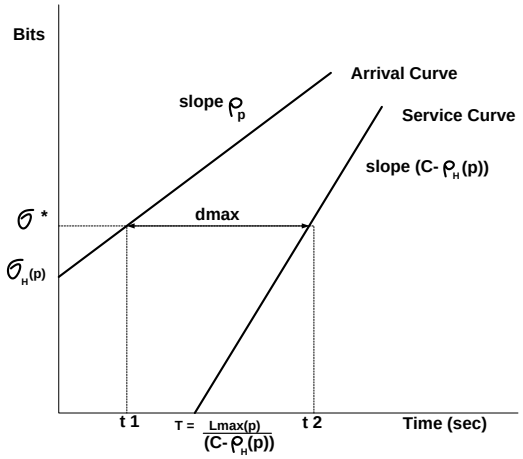

Figure 4.4: Arrival and Service Curves for PQ

- PQ

  The aggregate arrival and service curves for traffic of priority level p are shown in Figure 4.4 from which we have:

$$\begin{aligned} t_1 &= \frac{\sigma^* - \sigma_H(p)}{\rho_p} \\ t_2 &= \frac{\sigma^*}{C - \rho_H(p)} + \frac{L_{max}(p)}{C - \rho_H(p)} \\ t_2 - t_1 &= \frac{\sigma^*}{C - \rho_H(p)} + \frac{L_{max}(p)}{C - \rho_H(p)} - \left(\frac{\sigma^* - \sigma_H(p)}{\rho_p}\right) \end{aligned}$$

  where ρ_p is the aggregate average rate of priority p and $\rho_H(p)$, $\sigma_H(p)$ and $L_{max}(p)$ have been defined in Equations 4.2, 4.3 and 4.4 respectively. Thus $t_2 - t_1$ is maximum when $\sigma^* = \sigma_H(p)$ giving the delay for priority p as:

$$d_{max}^{PQ}(p) = \frac{\sigma_H(p) + L_{max}(p)}{C - \rho_H(p)} \tag{4.6}$$

- WFQ

  We will consider per-flow WFQ and class-based WFQ (commonly referred to as Class-Based Queueing). With (per-flow)WFQ, each flow has its own queue and guaranteed rate while in CBQ, there are several per-class queues each shared by flows belonging to the same class and

service between the queues is done using a fair-queuing scheduler. The equations for both types are similar with the observation that for CBQ, the arrival and service curves apply to the aggregate of all flows sharing a class.

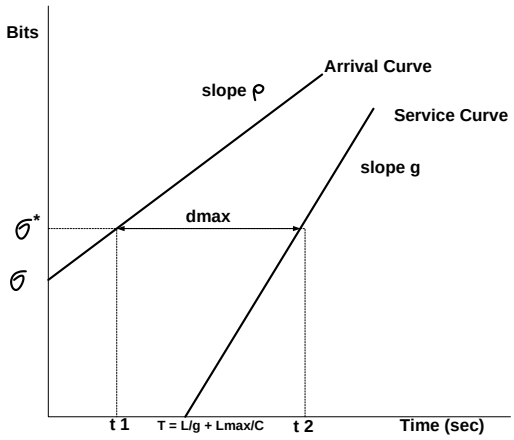

Figure 4.5: Arrival and Service Curves for WFQ

With reference to Figure 4.5 for a single WFQ flow we have:

$$\begin{aligned} t_1 &= \frac{\sigma^* - \sigma}{\rho} \\ t_2 &= \frac{\sigma^* + L}{g} + \frac{L_{max}}{C} \\ t_2 - t_1 &= \frac{\sigma^* + L}{g} + \frac{L_{max}}{C} - \left(\frac{\sigma^* - \sigma}{\rho}\right) \end{aligned}$$

where g is the guaranteed rate for the flow. From this we find that the delay is maximized when $\sigma^* = \sigma$ so that:

$$d_{max}^{WFQ} = \frac{\sigma + L}{g} + \frac{L_{max}}{C} \tag{4.7}$$

For CBQ the delay for class p is thus:

$$d_{max}^{CBQ}(p) = \frac{\sigma_p + L_p}{g_p} + \frac{L_{max}}{C} \tag{4.8}$$

where σ_p is the aggregate burstiness for class p, g_p is the guaranteed rate for class p, L_p is the maximum packet size for flows in class p and L_{max} is the maximum packet size over all flows.

One of the concerns with the Network Calculus approach is that it provides worst-case bounds on end-to-end delay which may underestimate the utilization possible when analyzing networks with bursty traffic. We do not expect this to have a significant impact on the nature of the results that will be obtained since we are primarily concerned with differences in performance between the different traffic handling mechanisms.

Network Calculus has been applied to a variety of network problems such as admission control, design of network regulators and deriving bounds on end-to-end delay [23, 24, 28, 32, 53, 62, 63] . Most of the available literature focuses on simple network topologies with at most two classes of service and we would like to explore and enhance the usefulness of this approach to large networks carrying a diverse mix of traffic as well as in addressing an aspect of network performance evaluation that has received limited attention. In the next section we extend the results of this section to show how network calculus can be used for obtaining bounds on end-to-end delays.

4.2 End-to-End Delay Analysis

In this section we look at how network calculus can be used to calculate maximum end-to-end delays in networks. We consider traffic that is regulated by a single token bucket with burstiness σ and average rate ρ. The general approach to computing end-to-end delays in a multi-node network is to sum the individual delays at each node. For a class of servers known as Latency-Rate (LR) servers [80] which provide a guaranteed rate to each connection and which can offer a bounded delay, a tighter bound can be obtained by using the "pay bursts once" principle [52]. The "pay bursts once" principle uses the approach of considering the entire network as a whole rather than looking at individual network elements in isolation. Using the "pay bursts once" principle for a network of LR-servers, the end-to-end delay for a flow k over a network of M servers in series is given by:

$$D_{E2E_k} = \frac{\sigma_k}{\min_m(g_k^{(m)})} + \sum_{m=1}^{M} \theta_k^{(m)} \tag{4.9}$$

where $\theta_k^{(m)}$ is the latency experienced by connection k at server m and $g_k^{(m)}$ is the guaranteed rate for connection k at server m. Examples of LR schedulers are Generalized Processor Sharing, Weighted Fair Queueing, Self-Clocked Fair Queueing, Weighted Round Robin. For a Generalized Processor

Sharing(GPS) scheduler, the latency is $\theta_k^{(m)} = \frac{L_{max}}{C^m}$ while for Packet Generalized Processor Sharing (Weighted Fair Queueing) the latency is [62]:

$$\theta_k^{(m)} = \frac{L_{max}}{C^{(m)}} + \frac{L_k}{g_k^{(m)}} \tag{4.10}$$

where L_k is the maximum packet size for connection k, L_{max} is the maximum packet size in the network and $C^{(m)}$ is the capacity of the link at the m^{th} server. The end-to-end delay with WFQ is thus[1]:

$$D_{E2E_k}^{WFQ} = \frac{\sigma_k}{\min_m(g_k^{(m)})} + \sum_{m=1}^{M} \left(\frac{L_k}{g_k^{(m)}} + \frac{L_{max}}{C^{(m)}} \right) \tag{4.11}$$

For non-LR schedulers, the end-to-end delay is given by:

$$D_{E2E_k} = \sum_{m=1}^{M} \theta_k^{(m)} \tag{4.12}$$

where $\theta_k^{(m)}$ is the maximum delay for connection k at server m. For a FIFO queue, $\theta_k^{(m)} = \theta^{(m)}$ and we recall from the previous section that:

$$\begin{aligned} \theta^{(m)} &= \frac{\sigma^{(m)}}{C^{(m)}} \\ \sigma^{(m)} &= \sum_{k \in N(m)} \sigma_k \end{aligned}$$

where $\sigma^{(m)}$ is the aggregate burstiness of all flows at node m, $C^{(m)}$ is the link capacity at node m and $N(m)$ is the set of connections that flow through server m. Thus for a series network of M FIFO servers we have the end-to-end delay given by:

$$D_{E2E}^{FIFO} = \sum_{m=1}^{M} \frac{\sigma^{(m)}}{C^{(m)}} \tag{4.13}$$

[1]Tighter bounds can be obtained by omitting the factor L_k/g_k in the last node. Refer to [63] for details.

For a static priority system (PQ) we have from Equation 4.6:

$$\theta_p^{(m)} = \frac{\sigma_H^{(m)}(p) + L_{max}(p)}{C^{(m)} - \rho_H^{(m)}(p)}$$

where $\theta_p^{(m)}$ is the latency experienced by a connection of class p at node m and the other terms are as defined in Equations 4.2, 4.3 and 4.4. Thus, for a series network of M static priority servers we have the end-to-end delay for traffic of priority level p given by:

$$D_{E2E_p}^{PQ} = \sum_{m=1}^{M} \frac{\sigma_H^{(m)}(p) + L_{max}^{(m)}(p)}{C^{(m)} - \rho_H^{(m)}(p)} \tag{4.14}$$

For a Class-Based Queueing System with P classes, we assume separate FIFO queues for each class with WFQ service between the queues so that each queue p gets a guaranteed rate g_p. If we assume that the network routing is such that the set of flows in a class share the same path end-to-end and do not merge with other flows (even if they are of the same class), then we can use the "pay-bursts once" approach to calculating the end-to-end delays as was done for WFQ. This type of flow grouping has been referred to as path-level aggregation in [57] and has also been studied in [73]. We consider however the more general case where flows in a class do not share the same end-to-end path so that the composition of flows belonging to a class varies at each node. From Equation 4.8, we have at the m^{th} node:

$$\theta_p^{(m)} = \frac{\sigma_p^{(m)} + L_p^{(m)}}{g_p^{(m)}} + \frac{L_{max}}{C^{(m)}}$$

Then the end-to-end delay for connections of class p going through a series of M CBQ servers is given by:

$$D_{E2E_p}^{CBQ} = \sum_{m=1}^{M} \frac{\sigma_p^{(m)} + L_p^{(m)}}{g_p^{(m)}} + \frac{L_{max}}{C^{(m)}} \tag{4.15}$$

In this chapter we have shown how network calculus is used to obtain bounds on delay and queue length in network elements. We have also shown

how to obtain delay bounds for four common schedulers. In the next chapter we provide an overview of the notation used in subsequent chapters as well as the applications used in our studies.

Chapter 5

Analytic Framework

5.1 Notation

We begin by introducing the notation that will be used throughout the thesis. We distinguish between traffic types and traffic classes by considering traffic types to be unique categories of flows such as voice, video and e-mail while traffic classes are groupings of traffic types. For instance Real-Time traffic is a class of traffic which may contain the traffic types voice and video. Using delay as an example, parameters for traffic types are denoted in one of two ways:

- D_k : delay for traffic of type k
- $D_{k,p}$: delay for traffic of type k when it is assigned to class p

Parameters for classes are denoted using the subscript $'class\ \ p'$. Parameters that pertain to an aggregation of flows of type k have a bar over them. For instance the average rate of a flow of type k would be denoted ρ_k whereas the aggregation of average rates would be denoted $\overline{\rho_k}$.

Notation and parameters used in the sections that follow are:

- K - number of traffic types
- P - the number of classes or priority levels
- C - link capacity (Mbps)

- w_T - total load on a link expressed as a fraction of link capacity
- w_k - link capacity allocation for type k with $\sum_{k=1}^{K} w_k = w_T < 1$
- D_k - maximum delay per node for traffic of type k (sec)
- $D_{k,p}$ - maximum delay per node for traffic of type k when assigned to class p.
- $D_{class\ p}$ - delay of traffic in class or priority-level p. We will use $D_{class\ p}$ instead of $D_{k,p}$ when we need to refer to the class or priority-level delay without specific reference to the traffic type.
- $D_{min} = \min_k\{D_k\}$
- D_{E2E} - maximum end-to-end delay
- σ_k - burstiness of traffic of type k (bits)
- ρ_k - average rate of traffic of type k (b/s)
- L_k - maximum packet size for flows of type k
- L_{max} - maximum packet size over all flows

5.2 Application Characterization

We consider three aspects of application characterization. The first is the identification of the applications that are likely to prevail in a network offering differentiated and guaranteed quality of service. Having identified the applications the second aspect to characterization is the specification of the nature of quality of service guarantees that are required for each application. The third aspect of characterization is with respect to the way in which the application is described to the network, often referred to as traffic modeling. In Table 5.1 we list the four applications that we chose and their characteristics.

Application	**RT/NRT**	**Rate type**	**QoS**
Telephony	RT	Stream	low delay
Interactive Video	RT	Stream	low delay, low loss
E-mail	NRT	Burst	delay tolerant
WWW	NRT	Burst	delay tolerant

Table 5.1: Network Applications

From Table 5.1 we identify two classes of traffic with voice and video belonging to the Real-Time (RT) traffic class and e-mail and WWW traffic belonging to the Non-Real Time (NRT) traffic class. Our choice of these applications was based on the fact that they are representative of current network usage and they provide diversity in their attributes and QoS. The quality of service metric that we use is end-to-end delay and specific values are given when we discuss numerical results. We recognize that typically e-mail and WWW traffic are considered to be adaptive applications that do not have strict delay requirements. We thus used delay objectives for e-mail and WWW that are an order of magnitude higher than those of voice and video to reflect the fact that while they may be adaptive, users of email and WWW applications have certain expectations on delay.

For characterization of the traffic sources we used the burstiness constraint model of Cruz [23] in which traffic is characterized by two parameters, a burstiness parameter σ and an average rate parameter ρ. We assume that the network uses regulator elements or shapers to ensure that the traffic entering it conforms to these parameters. We chose to use this bounded model for the traffic processes so that the results obtained are general and applicable to a variety of situations and do not depend on specific traffic assumptions. The IETF and ATM Forum have defined network elements which can convert an arbitrary traffic process into a process that is bounded in this way [45, 3]. We have chosen parameters for each class as shown in Table 5.2. The rate and packet size parameters are based largely on

Type	**Average Rate ρ (Mbps)**	**Burstiness σ (Bytes)**	**Packet Size (Bytes)**
Voice	0.064	64	64
Video	1.5	8000	512
E-mail	0.128	3072	512
WWW	1.0	40960	1500

Table 5.2: Traffic Class Parameters

values quoted in standards documents as well as some of the literature we surveyed. The burst parameters were chosen based on the literature and our own judgment. We later ran some tests using values that were based on the criterion of how many seconds worth of buffering the network could provide to each flow. While this did not make a significant difference to the trend of the results it did shed some light on the importance of the burst parameter.

5.3 Traffic Handling Mechanisms

We classified traffic handling mechanisms as simple, intermediate and complex depending on whether they are used for total aggregation, partial aggregation or per-flow handling respectively. We identified four candidate traffic handling mechanisms as shown in Table 5.3:

Classification	**Mechanisms**	**Abbreviation**
Simple	First-In-First-Out	FIFO
Intermediate	Strict Priority Queueing	PQ
	Class-Based Queueing	CBQ
Complex	Weighted Fair Queueing	WFQ

Table 5.3: Traffic Handling Mechanisms

We chose these mechanisms because they are representative of current and future implementations in network routers and switches. In WFQ each flow is assigned its own guaranteed rate. The CBQ and PQ schedulers have two classes a real-time(RT) class for voice and video and a non-real time(NRT) class for email and web traffic. In CBQ each class is assigned a guaranteed rate while for PQ there is no guaranteed bandwidth and service is strictly based on priority with the RT class having highest priority. In FIFO all flows share the same queue and there is also no per-flow guaranteed bandwidth. For each scheme the end-to-end delay that a flow obtains will depend on the traffic handling scheme. Let $D_{k,p}^{X}$ be the per-node delay for traffic of type k when it is assigned to class p using scheme X. For WFQ each flow consititutes a class while with FIFO there is only one class so that the delay for each flow will be the minimum over all specified delays. With CBQ and PQ we have two classes and the delay seen by a given traffic flow will be the minimum over all traffic flows in its class. Thus we have:

$$\begin{aligned} D_k^{WFQ} &= D_k \quad \forall\, k \\ D_k^{FIFO} &= \min_k\{D_k\} = D_{min} \quad \forall\, k \\ D_{k,p}^{CBQ/PQ} &= \min_{j \in class\ p}\{D_j\} \end{aligned}$$

The general approach to comparing the performance of the different traffic handling schemes will be to use WFQ as the reference mechanism and compare the other three schemes to it. We chose this approach because WFQ and its variants are considered to be the best schemes as far as ensuring service

guarantees in networks and the ability to provide guarantees on a per-flow basis makes it easier to establish consistent terms-of-reference. In the next chapter we apply network calculus to determine the capacity requirements under the four schemes for the case of a single link.

Chapter 6

Traffic Aggregation in a Single Network Node

6.1 Analysis and Methodology

In this section we describe the methodology for analysis of capacity requirements of the four traffic handling schemes in a single network element. Since we are using WFQ as the reference we begin by using equation 4.11 with $M = 1$,to find the guaranteed rate g_k^{WFQ} for each traffic type:

$$g_k^{WFQ} = \max\left\{\frac{\sigma_k + L_k}{D_k}, \rho_k\right\} \tag{6.1}$$

where we have assumed that for high-speed networks the factor L_{max}/C is negligible compared to the per-node delay. The number of connections for type k that can be supported using WFQ is then given by:

$$N_k = \left\lfloor \frac{w_k * C}{g_k^{WFQ}} \right\rfloor \tag{6.2}$$

where $\lfloor x \rfloor$ is the largest integer less than or equal to x and w_k is the fraction of link capacity allocated to flows of type k. For WFQ the minimum capacity required is thus given by:

$$C^{WFQ} = \sum_{k=1}^{K} N_k g_k^{WFQ} \tag{6.3}$$

We then determine how much capacity would be required to support the same traffic using the other three schemes by using Equations 4.13, 4.14 and 4.15 with $M = 1$. For CBQ with P classes, the required bandwidth C_{CBQ} is found as :

$$C^{CBQ} = \sum_{p=1}^{P} \sum_{k \in p} \frac{N_k \sigma_k + L_p}{D_{class\ p}} \tag{6.4}$$

where we have again assumed that the factor L_{max}/C is negligible. For Priority Queueing with P priority levels such that $1 > 2 > .....P$, the required capacity C^{PQ} is found as:

$$C^{PQ} = \max_{p=1...P} \left\{ \sum_{j=1}^{p} \sum_{k\ \in\ class\ j} \frac{N_k \sigma_k + L_{max(p)}}{D_{class\ p}} + \sum_{j=1}^{p-1} \sum_{k\ \in\ class\ j} N_k \rho_k \right\} \tag{6.5}$$

For FIFO, the capacity C_{FIFO} is given by:

$$C^{FIFO} = \sum_{k=1}^{K} \frac{N_k \sigma_k}{D_{min}} \tag{6.6}$$

In the next section we use Equations 6.3, 6.4, 6.5 and 6.6 to compare the capacity requirements of the four schemes under varying conditions.

6.2 Numerical Results for Single Network Node

We carried out several tests to demonstrate the applicability of the analysis presented in Section 6.1. The first test considers how the capacity requirements of each scheme are influenced by the traffic composition. In the next test we look at how the delay guarantees provided by each scheme are affected by changes in the traffic once an operating point has been established. In the third test we consider how annual projections on the growth of voice and WWW traffic impact the capacity requirements of the four schemes. Lastly we consider how changes in the delay guarantees and burstiness parameters impact the capacity requirements. For all the tests unless otherwise stated, the maximum delay for each traffic class is as shown in Table 6.1.

Traffic Type	**Delay (sec)**
Voice	0.002
Video	0.005
E-mail	0.5
WWW	0.5

Table 6.1: Maximum Delay for Single-Link Analysis

We recognize from Table 5.1 that typically email and WWW traffic are considered to be adaptive applications that do not have strict delay requirements. We thus used delay objectives for E-mail and WWW that are an order of magnitude higher than those of voice and video to reflect the fact that while they may be adaptive, users of E-mail and WWW applications have certain expectations on delay. We present the results obtained in the sections that follow.

6.2.1 Capacity Requirements with Varying Voice Load

In this section we present results on the difference in bandwidth requirements of the four schemes under varying load conditions. We use the indices $1, 2, 3, 4$ to represent *voice*, *video*, $e-mail$ and WWW traffic respectively. Using the notation w_T for the total load on the link and w_k for the fraction of link capacity allocated to traffic type k, we used three different values for video load: $w_2 = 0, 0.1, 0.2$. For each of these three values, the voice load w_1 was varied from 0.05 to $(w_T - w_2)$. We used 5 different weights to control how the remaining bandwidth after the voice and video were accounted for was shared between e-mail and WWW traffic. Denoting the weight vector as

$\alpha = [0.1, 0.3, 0.5, 0.7, 0.9]$, in each case the e-mail and WWW allocation was calculated as:

$$w_3 = \alpha * (w_T - (w_1 + w_2)) * C \tag{6.7}$$

$$w_4 = (1 - \alpha) * (w_T - (w_1 + w_2)) * C \tag{6.8}$$

where C is the link capacity. Using these parameters allows us to examine the effects of varying the proportions of the four traffic classes. We set the link load w_T equal to 0.9 and for each video load setting w_2, each voice load setting w_1 and each weight α, we calculated the capacity required by WFQ, CBQ, PQ and FIFO using the methodology presented in Section 6.1. We plot the capacity requirements in terms of the minimum number of OC-3 links required by each scheme and unless otherwise noted the results are shown for $\alpha = 0.5$. The results are presented in the form of bar graphs plotted on two separate y-axes. The top axis plots WFQ, CBQ and PQ results only and allows for a better comparison of the WFQ, CBQ and PQ results while the bottom axis plots the results for all four traffic handling schemes. Figures 6.1 and 6.2 show the capacity requirements with varying voice load for video loads of 0 and 0.2 respectively.

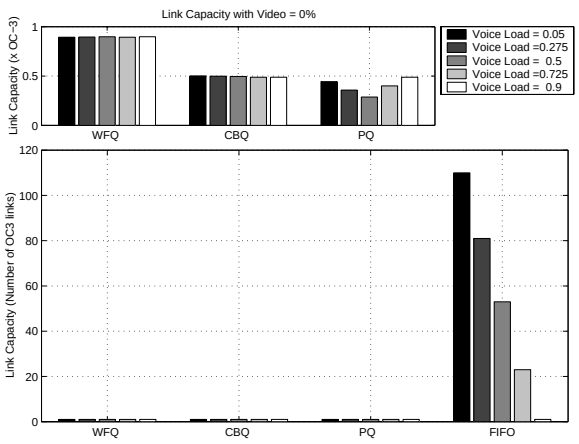

Figure 6.1: Capacity Requirement with No Video

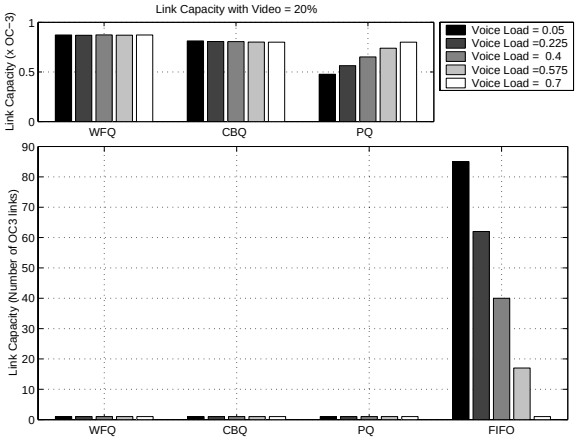

Figure 6.2: Capacity Requirement with 20% Video

We make the following observations:

- WFQ,CBQ and PQ require one OC-3 or less to meet the guarantees of all traffic types while FIFO requires from 20 to 110 OC-3 links when there is e-mail of WWW traffic.

- For FIFO, the amount of voice traffic significantly affects the bandwidth requirements. When the proportion of voice traffic is small, the bandwidth requirements are higher and vice versa. This is because when the voice load is small, the e-mail and WWW traffic proportions increase and more capacity is required to equalize the performance of the e-mail and WWW to that of voice in order to guarantee the delay objectives of voice traffic. With no video traffic, the capacity for FIFO is more than 100 times that of the other schemes when voice is 5% and equal when voice is 90%.

- WFQ requires more capacity than CBQ or PQ. This is not a very intuitive result and can only be understood by closer examination of the underlying equations. For the particular parameters we have used here, the guaranteed rate for e-mail and WWW as determined by Equation 6.1 is equal to their average rates. We thus have for WFQ:

$$C^{WFQ} = \sum_{k \in RT} N_k \left(\frac{\sigma_k + L_k}{D_k} \right) + \sum_{k \in NRT} N_k \rho_k$$

For CBQ we have:

$$C^{CBQ} = \sum_{k \in RT} \frac{N_k \sigma_k + L_{RT}}{D_{RT}} + \sum_{k \in NRT} \frac{N_k \sigma_k + L_{NRT}}{D_{NRT}}$$

where D_{RT}, D_{NRT} are the delays of the RT and NRT classes respectively and L_{RT}, L_{NRT} are the maximum packet sizes in each class respectively. From the given parameters, $D_{RT} = D_1$, $D_{NRT} = D_3 = D_4$, $L_{RT} = L_2$ and $L_{NRT} = L_4$. Substituting and subtracting C^{WFQ} from C^{CBQ} we obtain:

$$\begin{aligned} C^{CBQ} - C^{WFQ} &= N_2 \sigma_2 \left(\frac{1}{D_1} - \frac{1}{D_2} \right) + \frac{L_2}{D_1} + N_3 \left(\frac{\sigma_3}{D_3} - \rho_3 \right) \\ &\quad + N_4 \left(\frac{\sigma_4}{D_4} - \rho_4 \right) + \frac{L_4}{D_3} - \frac{N_1 L_1}{D_1} - \frac{N_2 L_2}{D_2} \\ &< N_2 \sigma_2 \left(\frac{1}{D_1} - \frac{1}{D_2} \right) + \frac{L_4}{D_3} + \frac{L_2}{D_1} - \sum \frac{N_k L_k}{D_k} \end{aligned}$$

where we have used the fact that for e-mail and WWW $\rho_k > \frac{\sigma_k + L_k}{D_k}$ to obtain the inequality.

From this we see that when there is no video traffic ($N_2 = 0$), the WFQ capacity will be greater than CBQ and this is due to the $\sum \frac{N_k L_k}{D_k}$ terms in the WFQ equations. As the video load increases, the difference between CBQ and WFQ capacity should decrease as evidenced by Figure 6.2. We expect that there should be a value of video load that causes the CBQ capacity to be greater than that of WFQ. The same kind of reasoning applies to PQ.

- CBQ, PQ and FIFO capacities are impacted by the weight α. This can be seen by comparing Figures 6.3 and 6.4 which are results obtained when $\alpha = 0.1$, with Figures 6.1 and 6.2. The weight α controls how the link capacity under WFQ is shared between email and WWW traffic. With a smaller α, the allocation to WWW increases so that for the same voice load, a smaller α will result in more capacity required to support the WWW traffic. This is most noticeable for FIFO where the WWW traffic must obtain the same delay guarantees as voice.

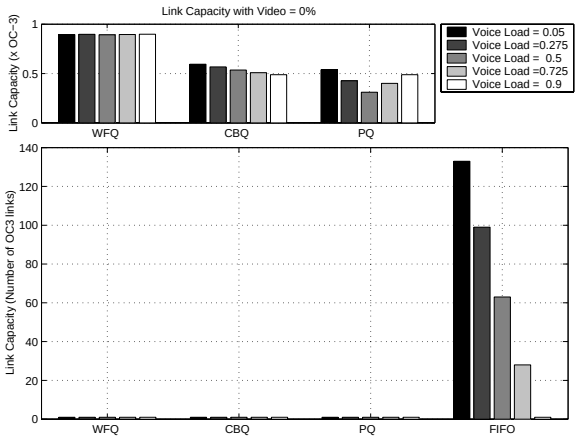

Figure 6.3: Capacity Requirement with No Video ($\alpha = 0.1$)

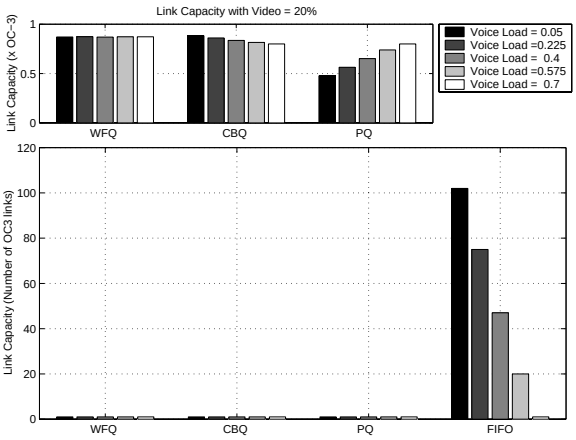

Figure 6.4: Capacity Requirement with 20% Video ($\alpha = 0.1$)

For PQ, the weight α is significant only when the video load is 0 or 10% and the weight is less than or equal to 0.5. Specifically, we observe that the PQ capacity is non-monotonic for these cases. This non-monotonic behavior stems from the nature of the capacity equation for PQ in which we consider the needs of both the low priority(NRT) and high-priority(RT) queues in determining the link capacity. In general we find that the capacity is determined by the NRT queue when the voice load is low (below 0.5) and determined by the RT queue otherwise. When

α is high, the amount of WWW traffic reduces and the influence of the NRT queue on the capacity is diminished. The same applies when the video load is 20% in which case we observe that the capacity is solely determined by the RT queue and capacity increases with increasing voice load.

- For CBQ and PQ, increasing the video load increases the capacity requirements while for FIFO, more video traffic reduces the capacity requirements. With CBQ and PQ the video traffic must obtain the same performance as voice so having more video traffic requires more capacity since the burstiness of the video is much higher than that of voice. For FIFO, the effect is reversed since more video traffic reduces the amount of email and WWW traffic and thus less capacity is required to ensure that the WWW traffic, which has the highest burstiness, attains the same performance as the voice.

6.2.2 Capacity Requirements with Varying WWW Load

In this case the roles of voice and WWW were interchanged from the previous case and the capacity required for each video load w_2, each WWW load w_4 and each weight α was calculated for each of the four traffic handling schemes. Figures 6.5 and 6.6 show the results when there is no video traffic and when the video load is 0.2 respectively.

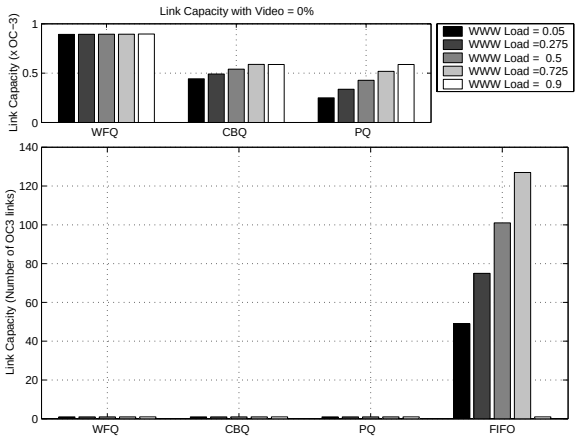

Figure 6.5: Capacity Requirement with no Video

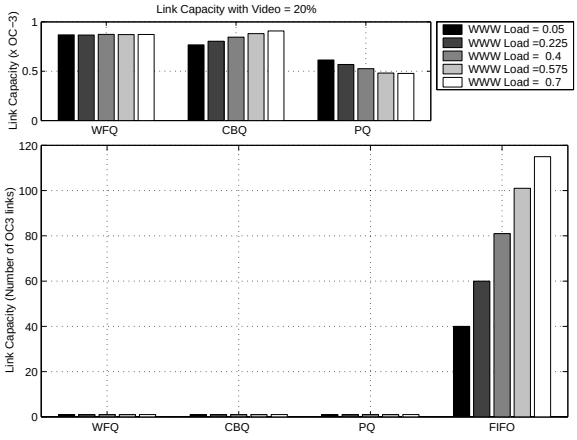

Figure 6.6: Capacity Requirement with 20% Video

From analysis of the results we observe:

- WFQ, CBQ and PQ are able to provide guarantees with one OC-3 link or less while FIFO requires 40 to 120 OC-3 links.

- With FIFO, increasing the WWW load increases the required capacity when voice traffic is present. When there is no voice traffic and no video traffic, the capacity requirements of FIFO decrease significantly and one OC-3 link is sufficient. For FIFO increasing the WWW load increases the capacity required since more capacity is required to ensure that the WWW traffic attains the same performance as the voice traffic. When video traffic is 0 and WWW traffic is 0.9 we see in Figure 6.5 that the capacity requirements are significantly reduced because there is no voice traffic so that the minimum delay in the queue is that of the email and WWW traffic. Comparing Figures 6.5 and 6.7 shows that increasing α results in a slight increase in capacity since the proportion of voice is reduced while the proportion of email traffic is increased, resulting in more capacity since the email has a higher burstiness than the voice traffic.

- The CBQ capacity shows a slightly non-monotonic behavior for some load combinations such as in Figure 6.8. We find that with 0.1 video load, the CBQ capacity is greater for WWW load equal to 0.65 than it is for WWW load equal to 0.8. The reason for this is that with 0.8 WWW load and 0.1 video, there is no e-mail and voice traffic so that the delay requirement for the RT queue is in fact the video delay which is much higher than the voice delay and hence requires less capacity. With 0.65 WWW load, the voice load is 0.135 and thus the RT queue delay is now the more stringent voice delay which requires more capacity to support the video traffic.

- For PQ with no video traffic the capacity is non-monotonic only when α is equal to 0.1. This is because there is a higher proportion of voice traffic so when the WWW load is low, the capacity requirements are determined by the RT queue and as the WWW load increases, the capacity is dictated more by the NRT queue. As the video load increases, the capacity is influenced more by the RT queue so that capacity decreases as WWW load goes up as shown in Figure 6.6. With $\alpha = 0.5$ and no video traffic Figure 6.5 shows that the influence of the voice traffic is less so that the capacity is controlled more by the NRT queue and the capacity increases with increasing WWW load.

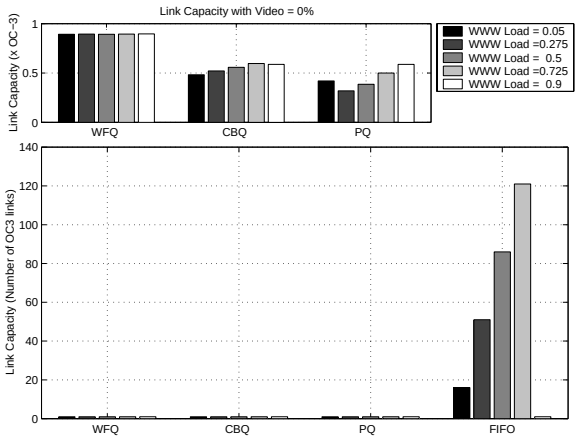

Figure 6.7: Capacity Requirement with no Video ($\alpha = 0.1$)

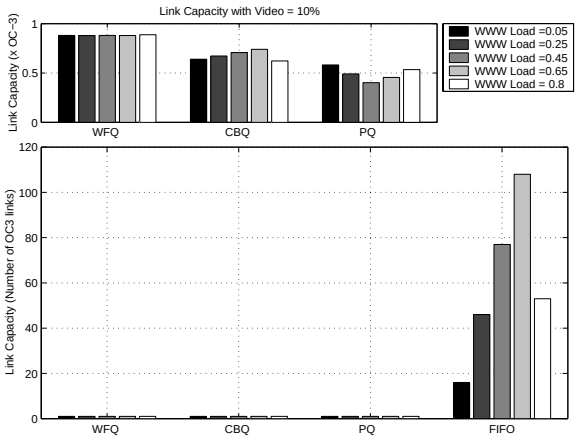

Figure 6.8: Capacity Requirement with 10% Video($\alpha = 0.1$)

6.2.3 Delay Performance with changes in load

The goal of this analysis was to explore the ability of the three schemes to provide acceptable delay QoS guarantees when the traffic submitted exceeded the traffic for which the network was designed. We considered two scenarios: one in which voice was the dominant traffic type and another in which WWW traffic was dominant. For each scenario, the capacity required by each of the four schemes was calculated using the procedures in Section 6.1. The number of sources, the link capacities and the delay performance are collectively referred to as the design point. For each scenario, the volume of either voice or WWW traffic was varied and the delay for each traffic type was calculated using the design point capacities. We note that in practice call admission procedures would be used to restrict the number of flows admitted but since we are testing the sensitivity of the traffic handling schemes we assume no call admission control. Instead we consider two approaches for bandwidth allocation under WFQ. In the first method which we call WFQ1, an increase in the traffic of a particular class is handled by re-distributing the bandwidth share of that class (as determined by the load w_k at the design point) equally among the sources (old and new). In the second approach called WFQ2, an increase in voice traffic is accommodated by "stealing" bandwidth from the e-mail and WWW classes and giving this to the new voice traffic to guarantee the voice traffic its delay QoS. Under WFQ2 an increase in WWW traffic is handled in the same way as with WFQ1. We present our results in the form of plots of the ratio of actual delay to desired delay as a function of the % change in voice or WWW load. The results obtained are presented in the paragraphs that follow.

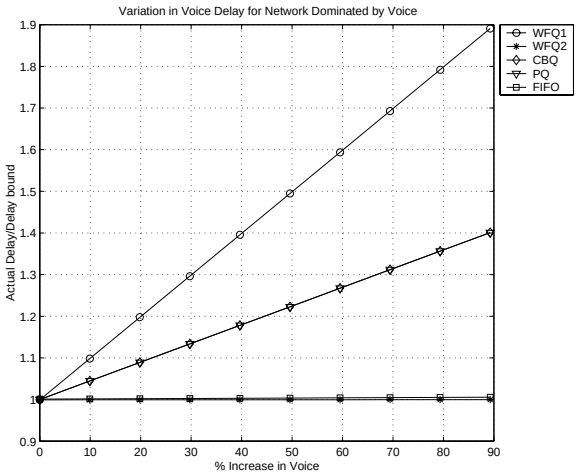

Figure 6.9: Variation in Voice Delay with increase in Voice load

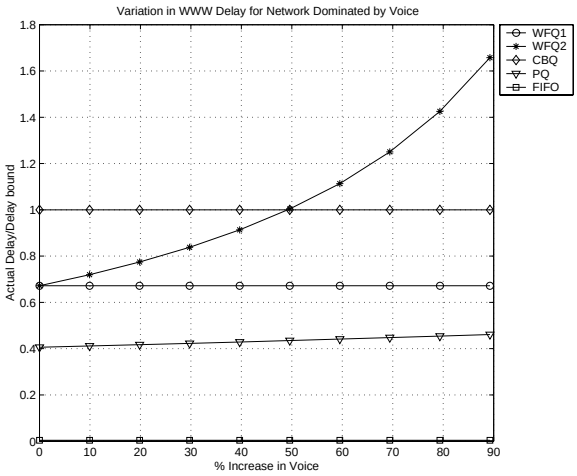

Figure 6.10: Variation in WWW Delay with increase in Voice load

- Network dominated by voice with increase in voice traffic

  Figure 6.9 shows the results obtained for a design point with voice as the dominant traffic type corresponding to $w_1 = 40\%$, $w_2 = 10\%$, $w_3 = 15\%$ and $w_4 = 15\%$ with changing voice traffic. In this case the delay guarantees provided by WFQ1 to voice increase as the voice load increases since less and less bandwith is now available to each individual connection. CBQ and PQ delay guarantees for voice also increase as the voice load increases. WFQ2 and FIFO delay guarantees are not significantly affected by the increased voice traffic. Under WFQ1 the delay guarantees of video, email and WWW are not impacted by the increased voice traffic while under CBQ and PQ, the delays for video are increased but are still within the specifications. Under WFQ2 Figure 6.10 shows that the delays for email and WWW are increased since bandwidth is taken from them to support the increase in voice traffic. We note that for email delay guarantees are violated for an increase in voice traffic greater than 83% while for WWW the violation occurs at an increase of 50%. The impact of the stolen bandwidth is more pronounced on the WWW traffic because of its higher burstiness.

- Network dominated by voice with increase in WWW traffic

  When the WWW traffic increases Figure 6.11 shows that FIFO is not able to provide the delay guarantees required by voice while all other traffic types are unaffected. For WFQ, CBQ and PQ, the voice traffic is isolated from the WWW traffic and there is no impact on the voice delay. CBQ delay for email is affected by the increase in WWW traffic

and the delay required by email is violated by the slightest increase in WWW traffic as shown in Figure 6.12. With PQ, the delay of email and WWW increases with increasing WWW load but is still within their specifications. For all four schemes, the delay requirements of video traffic are not affected by the increase in WWW traffic. Thus when we increase the WWW traffic, FIFO is now the most sensitive and we cannot meet the delay objectives for voice.

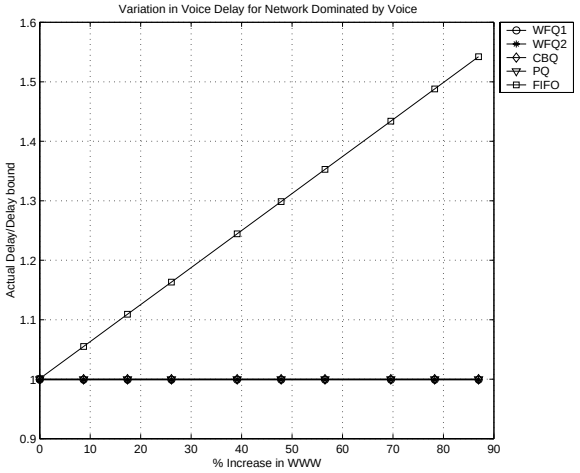

Figure 6.11: Variation in Voice Delay with increase in WWW load

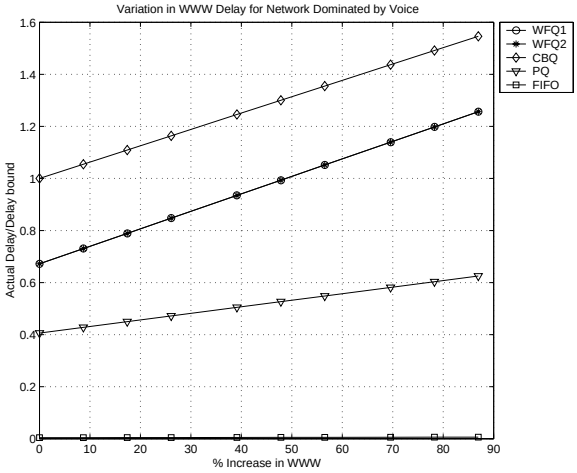

Figure 6.12: Variation in WWW Delay with increase in WWW load

- Network dominated by WWW with increase in voice traffic

  The impact on the voice and video delay is the same as in the case of the network dominated by voice. For email and WWW, the delay guarantees are violated under PQ at a lower value of increase in voice load (10%) due to the increased WWW load.

- Network dominated by WWW with increase in WWW traffic

  In this case video is the only traffic type that has its delay guarantees met under any scheme. For voice, all schemes fail to meet the delay requirements while for email CBQ and PQ fail to meet the guarantees. For WWW CBQ and PQ also fail to meet the delay objectives while WFQ fails when the increase in WWW load is greater than 50%.

The picture emerging from these results is that the traffic handling schemes are both sensitive to the type of traffic that dominates the network at the design point as well as to the type of traffic that increases the load on the

network. In general FIFO is the least sensitive to increases in voice traffic and the most sensitive to increases in WWW traffic when considering the delay objectives of voice. WFQ, CBQ and PQ are all sensitive to increases in the voice load and if the goal is to maintain the delay objectives of voice at all costs, the use of a scheme like WFQ2 can achieve this with a corresponding exponential increase in the delay of e-mail and WWW traffic. The value of these results is best demonstrated by taking into account the permissible variances in the delay objectives which means using statistical objectives as opposed to deterministic ones. The use of statistical delay objectives is discussed in Chapter 10.

6.2.4 Required Capacity with Projections on Traffic Growth

In this part of the analysis we calculate the capacity required to support yearly projections on growth in voice and WWW traffic. Current industry estimates are that voice traffic on the Internet will grow at a rate of 5-15% each year. The trend in WWW traffic has been almost a 100% increase in traffic per year [19]. We assumed the two scenarios in Section 6.2.3 of either voice or WWW being the dominant traffic type. Using the same procedures as before, we calculated the capacity required over a 5 year period assuming a 15% growth in voice traffic per year and a 100% growth in WWW traffic per year. The results obtained are shown in Figures 6.13 and 6.14 with the capacity expressed in terms of the minimum number of OC-3 links.

For the network dominated by voice we find from Figure 6.13 that the capacity required for WFQ increases to 4 times the initial capacity, CBQ by a factor of 3 and PQ by a factor of 2 after the 5 year period. FIFO capacity increases to 8 times the initial capacity, reaching 400 OC-3 links after 5 years. We note that the rate of increase in capacity for WFQ, CBQ and PQ is the same as evidenced by the slopes of the plots. The difference is that for WFQ the increase in capacity occurs after the second year while for CBQ and PQ the increase in capacity occurs after the third and fourth years respectively. For FIFO the increase in capacity is more exponential than linear and capacity increases right after the first year. When we start with a network dominated by WWW traffic as in Figure 6.14, the capacity of WFQ increases by a factor of 7 while CBQ and PQ capacity increases by 5 after the 5 year period. FIFO capacity increases by a factor of 13. In both cases FIFO is affected the most by the increase in traffic especially since we are increasing the volume of WWW traffic by a substantial amount.

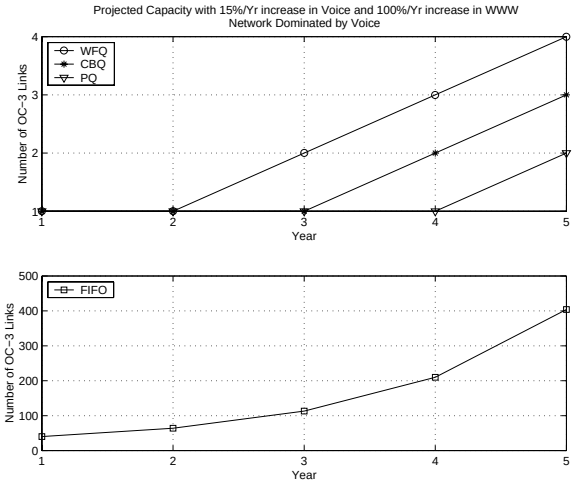

Figure 6.13: Network Capacity with Projections on Voice and WWW Traffic

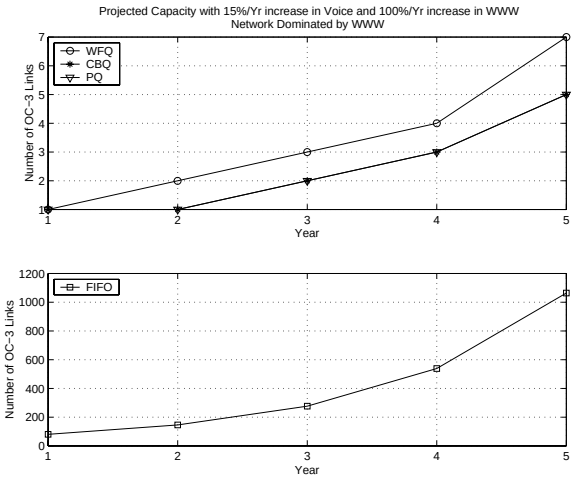

Figure 6.14: Network Capacity with Projections on Voice and WWW Traffic

To complete the picture we considered a hypothetical future situation in which the growth of WWW traffic is 15% and that of voice is 100%. This corresponds to the hypothesis that eventually growth in voice traffic will outpace growth in WWW traffic. The projections on capacity in this case are shown in Figures 6.15 and 6.16.

We find in this case that for a Voice-dominated network, CBQ and PQ capacity increase the least by a factor of 4 while WFQ increases by a factor of 7. FIFO capacity increases by only a factor of 1.5. For a WWW-dominated network, WFQ capacity increases by a factor of 4 and CBQ and PQ by 2 while the FIFO capacity increases by a factor of 1.6. We conclude that WFQ is affected more by the volume of voice traffic than the aggregate schemes while FIFO is affected most by the volume of WWW traffic when voice traffic is present in the network.

Comparing Figures 6.13 and 6.15 we observe that for FIFO a 100% increase in WWW requires more capacity than a 100% increase in voice while for WFQ, CBQ and PQ the reverse is true. The FIFO results can be attributed to the higher burstiness of WWW coupled with the much lower delay QoS for voice. This would lead us to conclude that FIFO is more sensitive to WWW traffic while WFQ, CBQ and PQ are more sensitive to the voice traffic. We also note that a network dominated by WWW requires more than 2 times the capacity of one dominated by voice under FIFO whereas under WFQ, CBQ and PQ the effect is reversed. This again points to FIFO being more sensitive to WWW traffic whereas the other schemes are more

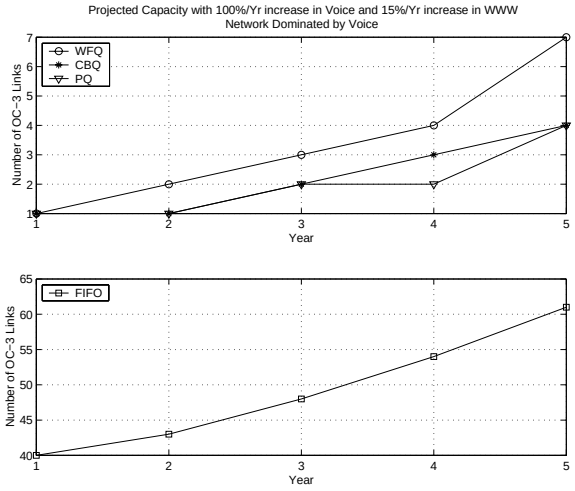

Figure 6.15: Network Capacity with Projections on Voice and WWW Traffic

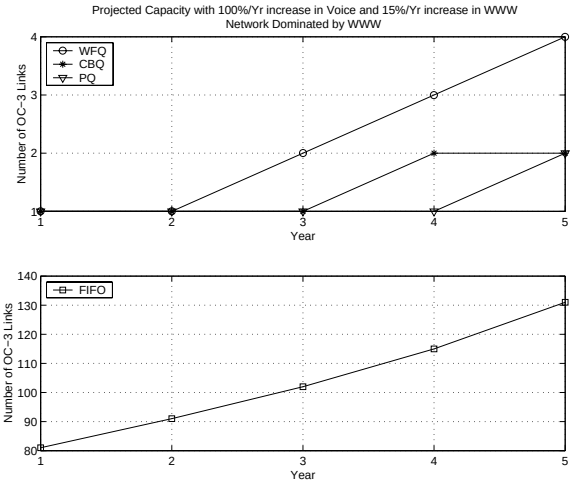

Figure 6.16: Network Capacity with Projections on Voice and WWW Traffic

sensitive to the voice traffic.

6.2.5 Capacity Requirements with Varying Delay Guarantees

In this section we look at how the value of delay guarantees required by voice and WWW impact the capacity requirements. We used load values of $w_1 = 0.4$, $w_2 = 0.1$, $w_3 = 0.15$ and $w_4 = 0.15$ and obtained the capacity required by the four traffic handling schemes for different values of voice and WWW delays. We present our results for CBQ, PQ and FIFO in terms of the ratio of their capacities to the WFQ capacity. In Table 6.2 we show results for varying voice delay.

Voice Delay(sec)	$C^{WFQ}(Mbps)$	C^{CBQ}/C^{WFQ}	C^{PQ}/C^{WFQ}	C^{FIFO}/C^{WFQ}
0.001	121.23	1.08	0.88	99.6
0.0015	121.23	0.86	0.67	66.5
0.002	121.74	0.76	0.57	49.7
0.0025	121.64	0.7	0.5	39.8
0.003	121.57	0.66	0.46	33.29

Table 6.2: Capacity as a function of Voice Delay

From the table we see that the ratio of CBQ, PQ or FIFO capacity to WFQ capacity decreases as the voice delay increases. PQ has the smallest

ratios followed by CBQ and FIFO has the largest ratios. The FIFO ratios are inversely related to the voice delays - tripling the delay from 0.001 to 0.003 causes the ratio (and hence FIFO capacity) to decrease by a factor of 3. Table 6.3 shows results when WWW delay is varied. One noticeable

WWW Delay(sec)	$C^{WFQ}(Mbps)$	C^{CBQ}/C^{WFQ}	C^{PQ}/C^{WFQ}	C^{FIFO}/C^{WFQ}
0.025	112.3	2.3	1.8	21.8
0.05	119	1.5	1.01	23.3
0.1	119	1.1	0.6	27.4
0.15	121.3	0.99	0.56	32.3
0.2	120.8	0.93	0.56	36.5
0.25	121.8	0.89	0.56	41.63
0.375	121.74	0.83	0.56	49.74
0.5	121.74	0.76	0.56	49.74
0.625	121.74	0.76	0.56	49.74
0.75	121.74	0.76	0.56	49.74

Table 6.3: Capacity as a function of WWW Delay

difference between Tables 6.2 and 6.3 is that the WFQ capacity is more significantly affected by the WWW delay than it is by the voice delay. This is because the results in Table 6.2 have the WWW delay equal to 0.5sec for which value the guaranteed rate is the average rate ρ while for values of WWW delay less than 0.25sec in Table 6.3 the guaranteed rate is dependent on the delay and is greater than the average rate. Thus a smaller delay value requires a larger guaranteed rate which limits the number of WWW flows that can be carried while not exceeding the 0.15 load limit. For CBQ and PQ, the ratios are greater for smaller values of delay since more capacity is required to support the NRT queue. For FIFO we see the reverse effect since at smaller WWW delay values there are fewer WWW flows which reduces the total burstiness in the FIFO queue and minimizes the capacity required to achieve the voice delay requirements. We note that for each scheme there is a value of WWW delay above which the delay ceases to have an impact on the capacity. For WFQ and FIFO this is 0.25sec since the guaranteed rate is constant and thus the number of WWW flows remains constant. For CBQ it is 0.5 because the e-mail delay is 0.5sec and with WWW delays greater than 0.5, the NRT queue capacity is determined by the e-mail delay. For PQ the capacity does not change above 0.15sec because the capacity is being determined by the RT queue.

6.2.6 Capacity Requirements with Varying Burstiness

In this section we examine how varying the input burstiness affects the capacity requirements. We varied the burstiness by quantifying the burstiness in terms of the time taken to transmit a burst at each traffic type's average rate. This would allow us to obtain different levels of burstiness while using consistent terms of reference. We conducted analyses for different burst durations ranging from 10ms to 100ms. Figure 6.17 shows results for a burst duration of 10ms for each traffic type.

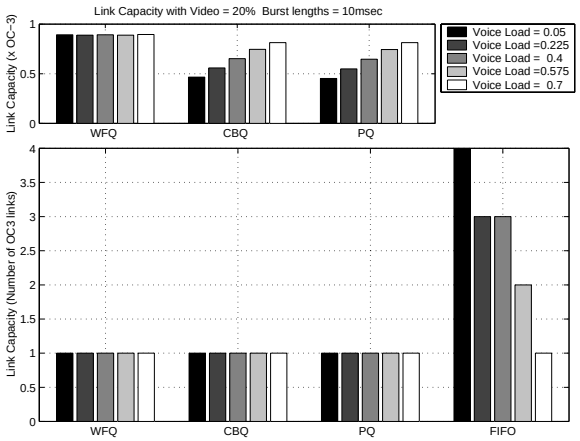

Figure 6.17: Link Capacity with 10ms bursts

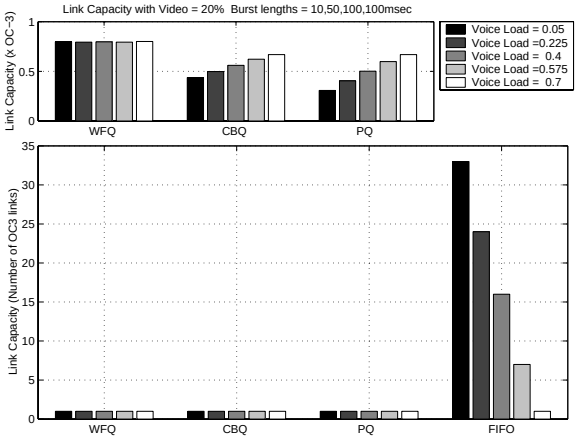

Figure 6.18: Link Capacity with varying burst sizes

This corresponds to burst sizes of 80 bytes for voice, 1875 bytes for video, 160 bytes for e-mail and 1250 bytes for WWW. We note in this case that the capacity requirements are of the same order of magnitude with FIFO requiring at most 4 times the capacity of WFQ, CBQ and PQ. In Figure 6.18 we used burst durations of 10ms for voice, 50msec for video and 100msec for email and WWW. This corresponds to burst sizes of 80 bytes for voice, 9375 bytes for video, 1600 bytes for e-mail and 12500 bytes for WWW. We see now that the gap between FIFO and the other schemes has widened and that FIFO now requires at most 30 times more capacity. Comparing these results with previous results based on the default parameters in Table 5.2, which correspond to burst durations of 8msec for voice, 42.7msec for video, 192msec for email and 327.7msec for WWW, we see that the increase in burstiness of e-mail and WWW affects FIFO the most and the difference in capacity with the default parameters is at most 85. These results demonstrate in part that as the traffic types become less and less homogeneous in their burstiness parameters, the difference in capacity between FIFO and the other three

schemes widens whereas WFQ, CBQ and PQ are not affected much by the disparity in burst sizes.

6.3 Summary

In this chapter we have developed and used an analytic method for evaluating the capacity requirements of different traffic handling schemes. We focused on the case of a single link and have obtained results that demonstrate several ways in which the analysis can be used. We have looked at the impact of the traffic mix, the effect of growth in network traffic both on the delay performance and annual traffic requirements and we have considered how the burstiness parameter affects the capacity requirements. Our key finding is that there is no significant difference between WFQ, CBQ and PQ on the basis of capacity required to provide the same delay QoS. For FIFO we have seen that the capacity requirements are influenced by the aggregate burstiness and the most stringent delay QoS and only when the aggregate burstiness is very low do we get FIFO capacity that is the same order of magnitude as the other three schemes. In the next section we extend our analysis to carrier sized networks.

Chapter 7

Network Analysis

In this chapter we provide the analysis and methodology needed to evaluate traffic handling schemes in carrier-sized networks. We begin by defining the topology of carrier-sized networks and then consider what approaches can be used to perform the analysis. Next we develop the analysis for the edge and core portions of the network after which we provide some numerical results.

7.1 General Network Topology

We consider a network architecture that has two distinct hieararchical layers - an edge and a core - as shown in Figure 7.1. The core is the backbone of the network and consists of high-speed switching elements. The edge portion of the network provides access to the core network and serves to aggregate traffic into the network core. In some cases the edge is divided into two layers: an access layer and a distribution layer [34].

The parameters and notation we use to describe a topology are:

- Traffic Handling Scheme X - Y - X where X is the traffic handling scheme in the edge of the network and Y is the traffic handling scheme in the core.
- Number of core nodes N_{core}
- Number of edge nodes per core node N_{edge}
- Reference capacity of edge links C_{edge} (homogeneous across the network)

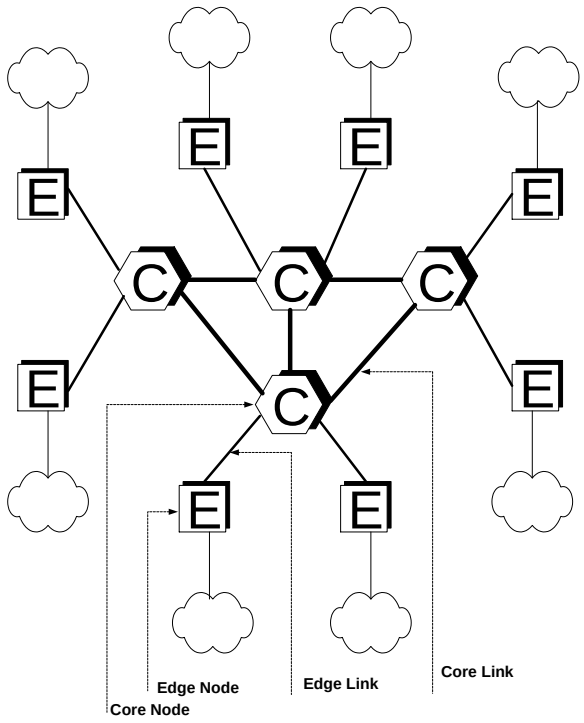

Figure 7.1: Carrier Topology

- Core Connectivity Matrix A

  This matrix describes the way in which the core nodes are connected. For N_{core} core nodes, the matrix is N_{core} x N_{core} and has elements a_{ij} where $a_{ij} = 1$ if a link exists from core node i to core node j and 0 otherwise. Note that links are unidirectional.

- Core Traffic Distribution Matrix T_k

  This matrix describes how the traffic belonging to a particular type k, incident at a core node is distributed to the destination core nodes in the network. There is one matrix for each traffic class and each matrix is of size N_{core} x N_{core}. The elements of the matrix are fractions of the per-class traffic destined for a specific core node. For example let $N_{core} = 3$ and suppose τ_k is the distribution matrix for voice traffic. Then $\tau_k^{12} = 0.5$ means 50% of voice traffic incident at core node 1 is destined for core node 2 etc.

- Traffic Allocation

  The traffic allocation for each traffic type denoted w_k, is specified with respect to each edge link so that it measures how much bandwidth on a single edge-link has been allocated to a specific type. More precisely,

the allocation is the fraction of the edge link capacity assigned with $0 \leq w_k < 1$ and $\sum_k w_k < 1$. It is used together with the guaranteed rate to determine how many sources of each type may be supported on one edge link. The total allocation is denoted w_T and is equal to $\sum w_k$.

7.2 Analysis of Network Capacity Requirements

As noted in Section 4.2, in performing the end-to-end analysis we can either analyze each node separately and accumulate delays [23, 24] or we can analyze the end-to-end path as a whole [62, 63]. The first approach can lead to very loose bounds on delay however the second approach is applicable only when all the nodes in the network are rate-guaranteeing and a minimum guaranteed rate can be identified. Since we are using traffic handling schemes that vary in their ability to provide rate guarantees, the analysis of the end-to-end path will be done in the following manner [11, 90, 91]:

1. divide the path into regions containing rate-guaranteeing and non-rate guaranteeing segments
2. analyze the non-rate guaranteeing segments by accumulating delays additively
3. analyze the rate-guaranteeing segments using the path approach
4. sum the delays in the rate-guaranteeing and non-rate guaranteeing segments to obtain the end-to-end delays

For the edge-to-core node ratio, we calculate the number of edge nodes per core node based on a fixed total external load on the core and assign traffic to each edge node to meet a prescribed utilization level on the edge links. For rate-guaranteeing schemes such as WFQ there are two possibilities for bandwidth allocation in the network: core nodes allocate same bandwidth as edge nodes to each connection or core nodes allocate more bandwidth than edge nodes. We will use the first approach for simplicity which will mean that the delays in the core are the same as in the edge. The second approach may be more useful when optimizing the delay budget since we can reduce the delays in the core by allocating more bandwidth.

Since the goal is to compare the network capacity required by different traffic handling schemes, we must ensure that there is a uniform basis for

comparison. As with the single node case we use a network with WFQ in both the edge and core as the reference. We will refer to this network as the reference network and use the notation $WFQ*$ to distinguish it from other networks. The approach is thus to calculate the amount of traffic that can be supported in a $WFQ*$ network and compare the capacity required to support the same traffic for various combinations of traffic handling schemes in the edge and core.

7.3 Preliminary Equations and Parameters

In addition to the parameters introduced in Section 6 we define the following parameters:

- M_T - maximum number of nodes traversed by any flow in the network
- M - maximum number of core nodes traversed by any flow in the network
- D_k^{E2E} - maximum end-to-end delay for traffic of type k
- D_k - maximum delay per node for traffic of type k. We assume that the delays are uniformly distributed over the nodes to give:

$$D_k = \frac{D_k^{E2E}}{M_T}$$

Using Equation 4.11, we calculate the guaranteed rate with WFQ* g_k^{WFQ*} as :

$$g_k^{WFQ*} = \max\left\{\rho_k, \frac{\frac{\sigma_k}{M_T} + L_k}{D_k}\right\} \tag{7.1}$$

where we have again assumed that the factor $\frac{L_{max}}{C}$ in Equation 4.11 is negligible. From Equation 7.1 we observe that the value assigned to g_k^{WFQ*} will depend on the maximum number of nodes traversed and we can find the critical value of M_T by solving the inequality:

$$\rho_k \geq \frac{\frac{\sigma_k}{M_T} + L_k}{D_k} \tag{7.2}$$

$$\rho_k \geq \frac{(\sigma_k + M_T * L_k)}{D_k^{E2E}} \tag{7.3}$$

This gives

$$M_T^* = \frac{\rho_k * D_k^{E2E} - \sigma_k}{L_k} \tag{7.4}$$

So, when the number of nodes traversed is less than M_T^*, $g_k^{WFQ*} = \rho_k$ and when the number of nodes traversed is greater than M_T^*, $g_k^{WFQ*} = \frac{\frac{\sigma_k}{M_T} + L_k}{D_k}$. This value of M_T will become important when we compare capacity requirements for networks of varying size. The number of sources of each type that can be supported at an edge node using WFQ* is given by:

$$N_k = \left\lfloor \frac{w_k * C_{edge}}{g_k^{WFQ*}} \right\rfloor \tag{7.5}$$

where C_{edge} is the edge link capacity and w_k is the proportion of capacity on the edge link allocated to traffic of class k. We can now proceed to calculating the edge and core capacity required to support this traffic.

7.4 Edge Capacity Requirements

The analysis of capacity in the edge is similar to the Single-Node analysis of Section 6 thus we will omit the details and present the equations. For WFQ, the edge capacity is given by:

$$\begin{aligned} C_{edge}^{WFQ*} &= \sum_{k=1}^{K} N_k * g_k^{WFQ*} \\ &= \sum_{k=1}^{K} w_k C_{edge} \end{aligned} \tag{7.6}$$

For CBQ and PQ with P classes/levels we have:

$$C_{edge}^{CBQ} = \sum_{p=1}^{P} \sum_{k \in p} \frac{N_k \sigma_k + L_p}{D_{class\ p}} \quad p = 1, 2, ...P \tag{7.7}$$

$$C_{edge}^{PQ} = \max_{p=1...P} \left\{ \sum_{j=1}^{p} \sum_{k\ \in\ class\ j} \frac{N_k \sigma_k + L_{max}(p)}{D_{class\ p}} + \sum_{j=1}^{p-1} \sum_{k\ \in\ class\ j} N_k \rho_k \right\} \tag{7.8}$$

For FIFO, the capacity is given by:

$$C_{edge}^{FIFO} = \sum_{k=1}^{K} \frac{N_k \sigma_k}{D_{min}} \tag{7.9}$$

In the next section we explore how using different mechanisms in the edge and core affects the core capacity.

7.5 Core Capacity Requirements

The capacity required in the core for each scheme has the same general form regardless of the mechanism in the edge. The key differentiating factor is the change in burstiness for a given traffic type induced by the delay in the edge. Since edge capacities are calculated to meet the prescribed QoS objectives for each class, we can assume that the edge delay will be bounded by the per-node maximum delay D_k. For a WFQ core, using the traffic distribution matrices T^k, the minimum required capacity on the link $l(i,j)$ between the core node-pair (i,j) is given by:

$$C_{core(i,j)}^{WFQ} = N_{edge} * \sum_{k=1}^{K} \sum_{(x,y)} \tau_k^{(x,y)} * N_k^x * g_k \tag{7.10}$$
$$\{(x,y) : l(i,j) \in Path(x,y)\}$$

where $\tau_k^{(x,y)}$ represents the distribution factors of flows between core nodes (x,y) whose path $Path(x,y)$ includes the link $l(i,j)$ and N_k^x is the number of

sources of class k whose edge node is attached to core node x. The value of g_k will depend on the traffic handling in the edge: when the edge uses WFQ, g_k will be the same as g_k^{WFQ*} in Equation 7.1 and when the edge uses any other mechanism, it will be:

$$g_k = \max\left\{\rho_k, \frac{\frac{\sigma'_k}{M} + L_k}{D_k}\right\} \tag{7.11}$$

where M is the number of core nodes and σ'_k is the burstiness after passing through the edge portion of the network which is given by:

$$\sigma'_k = \sigma_k + \rho_k D_k^{edge} \tag{7.12}$$

$$D_k^{edge} = \begin{cases} D_k & WFQ \\ \min_{j \in class\, p}\{D_j\} \quad k \in class\, p & CBQ, PQ \\ \min_k\{D_k\} & FIFO \end{cases} \tag{7.13}$$

To calculate the core capacity with CBQ, PQ and FIFO, for each link $l(i,j)$ let:

$$\overline{\rho}_k^{(i,j)} = \sum_{(x,y)} \tau_k^{(x,y)} N_k^x \rho_k \tag{7.14}$$

$$\overline{\sigma}_k^{(i,j)} = \sum_{(x,y)} \tau_k^{(x,y)} N_k^x \sigma_k^{h(x,y)} \tag{7.15}$$

where $\{(x,y) : l(i,j) \in Path(x,y)\}$. Note that $h(x,y)$ is the number of hops traversed by the traffic before reaching link $l(i,j)$ and $\sigma_k^{h(x,y)}$ is the associated burstiness which will depend in part on the traffic handling mechanism used in the edge. To be more specific, $\sigma_k^{h(x,y)}$ will be given by:

$$\sigma_k^{h(x,y)} = \begin{cases} \sigma'_k + h(x,y) * \rho_k * D_{class\, p} & \forall\ k\ \in\ p\ for\ CBQ/PQ \\ \sigma'_k + h(x,y) * \rho_k * D_{min} & for\ \ FIFO \end{cases} \tag{7.16}$$

with σ'_k defined as before.

Then by inverting Equations 4.13, 4.14, 4.15 the core bandwidth required on link $l(i,j)$ for each scheme is calculated as follows:

$$C_{core(i,j)}^{CBQ} = N_{edge} * \sum_{p=1}^{P} \sum_{k \in p} \frac{\overline{\sigma_k}^{(i,j)} + L_p}{D_{class\ p}} \tag{7.17}$$

$$C_{core(i,j)}^{PQ} = N_{edge} * \max_{p=1....P} \left\{ \sum_{m=1}^{p} \sum_{k \in\ class\ \ m} \frac{\overline{\sigma_k}^{(i,j)} + L_{max}(p)}{D_{class\ p}} + \sum_{m=1}^{p-1} \sum_{k \in\ class\ \ m} \overline{\rho}_k^{(i,j)} \right\} \tag{7.18}$$

$$C_{core(i,j)}^{FIFO} = N_{edge} * \sum_{k=1}^{K} \frac{\overline{\sigma}_k^{(i,j)}}{D_{min}} \tag{7.19}$$

where we again assume the term (L_{max}/C) in Equation 4.15 is negligible. The equations obtained in this chapter for calculating capacity requirements in edge-core networks are a new result and one of the key contributions of this thesis. In the next section we use the analysis of this section to provide numerical results on capacity requirements of the four schemes for different edge-core combinations.

7.6 Numerical Results for Network Analysis

7.6.1 Topology Construction

Carrier network topologies were constructed by varying the number of core nodes and the number of core links per node. For a given core node value N_{core}, the number of links per core node n_{link} was varied according to:

$$n_{link} = 2, 3....., \;\; N_{core} - 1$$

This resulted in $(N_{core} - 2)$ different topologies per core node value. For each value of N_{core} and n_{link}, the core nodes were connected according to the following algorithm:

- Number the core nodes from 1 to N_{core}
- Connect the i^{th} core node to core nodes $i+1$, $i+2$,..... $i+n_{link}$ where the addition is modulo N_{core}. This means that the entries $(i, i+1)$, $(i, i+2)$,..... $(i, i+n_{link})$ in the connectivity matrix would be set to 1.

An example of a 5 node topology with 3 links per core node is shown in Figure 7.2. Thus a topology is uniquely identified by the pair (N_{core}, n_{link}).

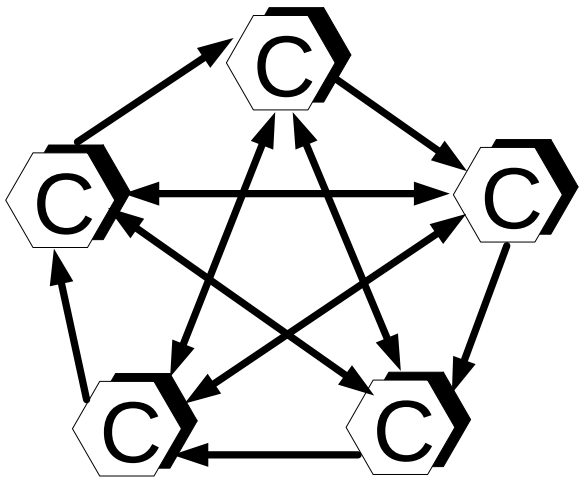

Figure 7.2: Topology with 5 Core Nodes and 3 links per node

Note that the case when the number of links per core node is equal to $(N_{core} - 1)$ is the familiar full-mesh topology. For each topology, we used

the same external load on the core network by fixing the total number of edge nodes N_{edge} and setting the number of edge nodes per core node to $N_{edge} = N_{edge}^{total}/N_{core}$ where N_{edge}^{total} is the total number of edge nodes in the network. We used a value of $N_{edge}^{total} = 60$ for the results reported here. Once a topology had been constructed, routes were set-up within the core using Djikstra's shortest path algorithm [83]. The maximum number of hops in the core was noted in each case and the mean number of hops $\overline{hops}$ was calculated as:

$$\overline{hops} = \frac{\sum_{n=1}^{N_{routes}} hops(n)}{N_{routes}}$$

where N_{routes} is the total number of routes and $hops(n)$ is the number of hops in the n^{th} route.

7.6.2 Parameters

The end-to-end delay values used for each traffic type are shown in Table 7.1.

Traffic Type	**Delay (sec)**
Voice	0.02
Video	0.05
E-mail	0.5
WWW	0.5

Table 7.1: Maximum End-to-End Delay for Edge-Core Analysis

Traffic within the core was distributed symmetrically[1] so that each core node sends an equal amount of traffic to every other core node and the traffic distribution matrix is specified by:

$$T_{ij} = \begin{cases} \frac{1}{(N_{core}-1)} & i \neq j \\ 0 & i = j \end{cases}$$

We used a maximum load on each edge link (w_T) of 90%. Using the procedures outlined in the methodology, the capacity required in the edge

[1] We ran some tests with random distribution of traffic in the core but this did not have a significant impact on the results obtained.

and core for all possible combinations of edge-core traffic handling schemes was calculated for different topologies with core nodes ranging from 3 to 20. The results obtained are presented in the sections that follow.

7.6.3 Capacity Requirements with Symmetric Traffic Distribution

The purpose of this analysis was to find out how bandwidth requirements of the four schemes are affected by changes in the voice and WWW load on the edge links. We used three values of video load (0,0.1,0.2) and in one case we varied the voice load while in the other case we varied the WWW load within the limits of the maximum load w_T. The remaining edge bandwidth was equally divided between email and WWW for the first case and email and voice for the second. For each load setting we used the reference all-WFQ network to establish how many flows of each type could be handled by the edge nodes and then calculated the core capacity. We then calculated the capacity required to support the same flows using different combinations of edge and core traffic handling schemes. We present our results in the form of bar graphs that show the mean core link capacity stacked on top of the mean edge capacity. We consider three different aspects of the results: the impact of edge traffic handling, the impact of the network diameter and the impact of network connectivity.

Impact of Edge Traffic Handling

For this section we will use the specific case of 20 core nodes in a full-mesh topology with a video load of 0.1 to illustrate the results. For this case, each core node supports 3 edge nodes and each core node sends 5.2% (= 1/19) of the total incoming traffic to every other core node. For the reference WFQ network, we thus expect the core link bandwidth to be at least $(3 * OC3)/19$ which is 15.8% of an OC-3 link. Figure 7.3 shows the capacity required for different core schemes when WFQ is used in the edge and voice load is varied while Figure 7.4 shows results for varying WWW load.

In these and subsequent figures we show bar-graphs plotted on two separate y-axes to allow for better observation of the WFQ, CBQ and PQ results. Due to the difference in magnitude between FIFO and the other three schemes, plotting all the results on the same scale as in the lower y-axis sometimes masks the differences between WFQ, CBQ and PQ. In addition we have combined the edge and core results on one graph to better illustrate the dif-

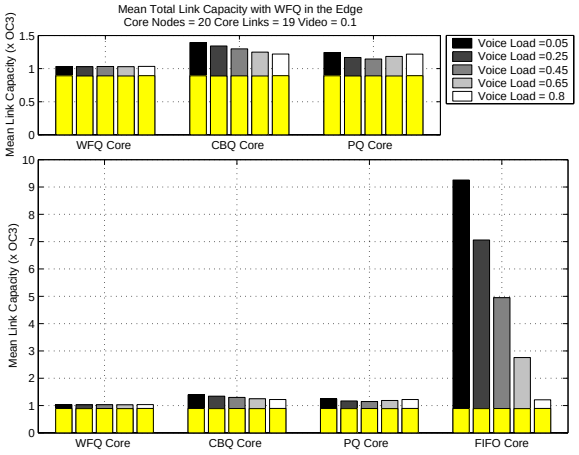

Figure 7.3: Edge and Core Capacity with WFQ in the Edge

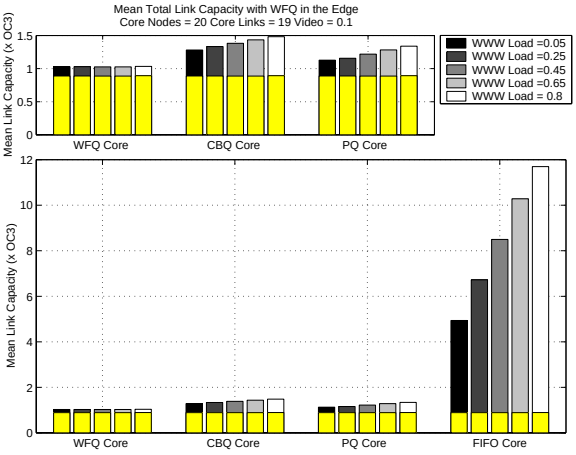

Figure 7.4: Edge and Core Capacity with WFQ in the Edge

ferences between the four schemes. To distinguish the two, the core capacity is shaded according to the value of voice or WWW load and is stacked on top of the corresponding edge capacity.

We observe from Figure 7.3 that with a WFQ core, the bulk of the capacity is in the edge (approximately 1 OC3)and the mean link capacity of a WFQ core is less than an OC-3, the exact value being 0.15 of an OC-3 which agrees with our expectations. The core capacity increases marginally with increasing voice load. With CBQ and PQ, slightly more capacity is needed in the core than with a WFQ core but it is still less than an OC-3. For a FIFO core, the bulk of the capacity is in the core for lower voice loads, ranging from 9 OC-3s when the voice load is 0.05 to less than 1 OC-3 when the voice load is 0.85. We note that the FIFO core capacity depends on the voice load and when the voice load is at its maximum, the capacity with FIFO is comparable to the other three schemes. The difference in capacity between FIFO and the other schemes is attributed to the fact that with FIFO the performance of all traffic types must be equalized to that of the most stringent delay QoS and thus more capacity is required to achieve this when the voice load is low and the email and WWW load are high. When the voice capacity is at its maximum, email and WWW traffic are zero and essentially the FIFO queue is equivalent to the CBQ and PQ queues. When WWW load is varied as in Figure 7.4, the CBQ, PQ and FIFO capacity increases with increasing WWW load and in general the capacity requirements are slightly larger than with corresponding variations in voice load. In Figures 7.5 and 7.6 we present results with CBQ in the edge.

We notice from Figure 7.5 that the edge capacity is dependent on the

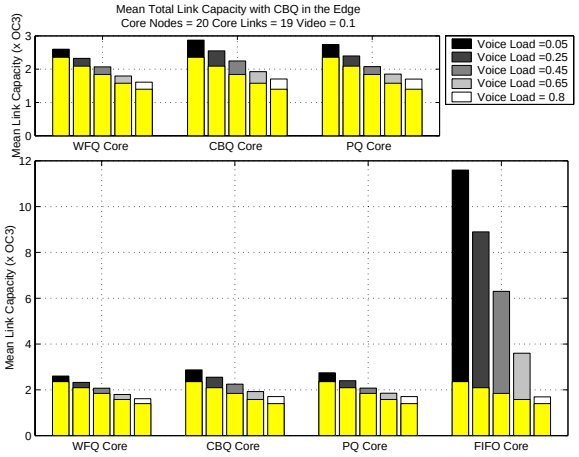

Figure 7.5: Edge and Core Capacity with CBQ in the Edge

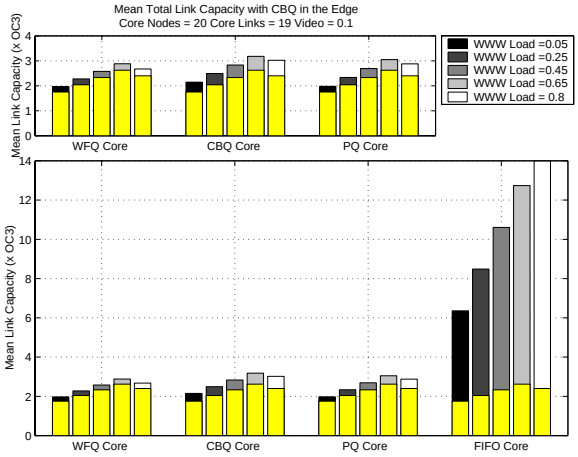

Figure 7.6: Edge and Core Capacity with CBQ in the Edge

voice load, being larger for smaller values of voice load which is attributed to the correspondingly higher values of email and WWW load. The edge capacity with CBQ is about twice that with a WFQ edge. We note that the core capacity with WFQ, CBQ and PQ is less than or equal to 1 OC-3, with CBQ having the larger core capacity. The FIFO core capacity follows the same trend as with the WFQ edge, ranging from 10 to 1 OC-3. With varying WWW load, Figure 7.6 shows that the CBQ edge capacity increases with increasing WWW load as long as there is voice traffic present. When there is no voice (WWW = 0.8), the edge capacity drops slightly because now the RT queue delay is determined by the video requirements which are less stringent than the voice delay and thus less capacity is required for the RT queue. The core capacities follow the same trend as with the WFQ edge.

With PQ in the edge, the variation in capacity for varying voice load is shown in Figure 7.7. The key difference between having CBQ in the edge versus PQ in the edge is the variation in edge capacity with voice load. We note that the variation is non-monotonic, with the capacity decreasing with increasing voice load up to a load of about 0.45 after which it starts to increase with increasing voice load. When the WWW load is varied as in Figure 7.8 the PQ edge capacity increases linearly with increasing WWW load and the trend of core capacities is the same as with WFQ and CBQ.

When FIFO is used in the edge, the network capacity is dominated by the edge capacity as shown in Figures 7.9 and 7.10.

With varying voice load as in Figure 7.9, the edge capacity depends greatly on the voice load and ranges from 40 OC-3s when voice load is 0.05 to 2

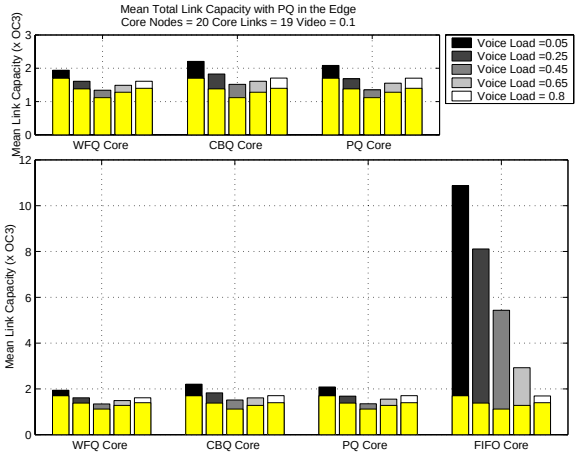

Figure 7.7: Edge and Core Capacity with PQ in the Edge

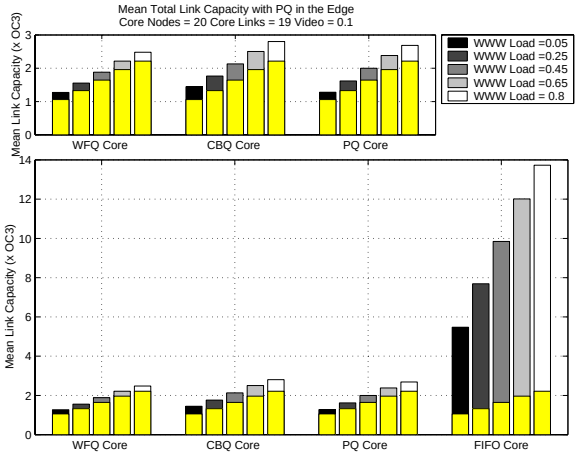

Figure 7.8: Edge and Core Capacity with PQ in the Edge

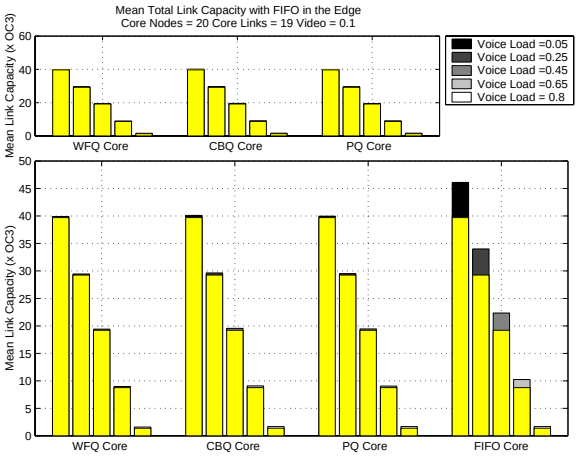

Figure 7.9: Edge and Core Capacity with FIFO in the Edge

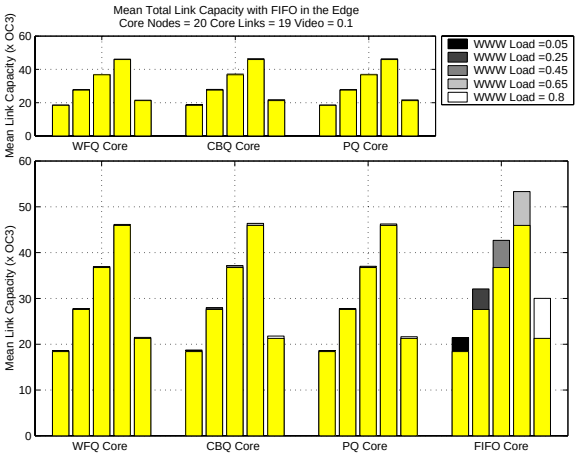

Figure 7.10: Edge and Core Capacity with FIFO in the Edge

OC-3s when the voice load is 0.8. With WFQ, CBQ and PQ cores, the core capacity is of the order of 1 OC-3 whereas with a FIFO core the core capacity ranges from 8 to 1 OC-3. With varying WWW load as in Figure 7.10, the FIFO edge capacity increases with increasing WWW load as long as there is voice traffic present. When there is no voice traffic (WWW= 0.8), the edge capacity starts to drop since the minimum delay is now the video delay which is larger than the voice delay and thus less capacity is required to provide delay guarantees. The FIFO core capacity also increases with increasing WWW load whereas for WFQ, CBQ and PQ the core capacity is not impacted much by the WWW load.

		Core Traffic Handling			
		WFQ	**CBQ**	**PQ**	**FIFO**
Edge Traffic Handling	**WFQ**	107	201	144	1497
	CBQ	191	256	195	1818
	PQ	146	210	149	1700
	FIFO	1212	1269	1224	2318

Table 7.2: Network Capacity for 20 Node Full-Mesh Network (in equivalent OC3 links)

To illustrate the implications of these results in a network-wide sense, we calculated the total network capacity for the 20 core node network in a full-mesh configuration for the specific case of voice load of 0.45 on the edge links. The results are shown in Table 7.2 where the capacity is in multiples of OC-3 capacity. Looking at the table we see that the all-WFQ network requires the least network capacity while the use of FIFO in either the edge or core requires large amounts of capacity. Using CBQ and PQ with WFQ requires at most 2 times the capacity of the all-WFQ network. The general conclusion to be drawn from these results is that any combination of WFQ, CBQ and PQ in the edge and core will require capacity of the same order of magnitude and on the basis of capacity requirements there is no significant difference between these three schemes.

Impact of Network Diameter

In this section we consider how the network diameter, measured in terms of the maximum core hops traversed by a flow, affects the network capacity. We use the cases of WFQ and FIFO in the edge for illustration since the CBQ and PQ results are comparable to WFQ. In Table 7.3 we show the WFQ capacity and the ratios of CBQ, PQ and FIFO capacity to WFQ capacity for the case of WFQ in the edge with 20 core nodes. For all four schemes

Max Hops	C^{WFQ} **(x OC3)**	C^{CBQ}/C^{WFQ}	C^{PQ}/C^{WFQ}	C^{FIFO}/C^{WFQ}
1	54	2.8	1.72	28.5
2	85	3.52	2.17	40.08
3	102	3.96	2.24	40.21
4	113	4.0	2.47	46.06
5	156	4.66	2.8	51.8
7	198	5.27	3.36	62.6
10	281	6.17	4.11	74.6

Table 7.3: Core Capacity as a function of Network Diameter for WFQ Edge

the capacity increases with network diameter although the rate of increase is not the same. For instance with a WFQ core the capacity required by a 10-hop network is 5.2 times that of a 1-hop (full-mesh) network while for CBQ the factor is 11.5, for PQ it is 12.4 and for FIFO it is 13.6. When FIFO is used in the edge, the results obtained are shown in Table 7.4. In

Max Hops	C^{WFQ} **(x OC3)**	C^{CBQ}/C^{WFQ}	C^{PQ}/C^{WFQ}	C^{FIFO}/C^{WFQ}
1	77	1.74	1.15	15.5
2	99	2.5	1.57	22.12
3	117	3.13	1.83	28.65
4	145	3.8	2.15	34.4
5	171	4.4	2.4	39.8
7	213	5.3	2.84	49.9
10	297	6.67	3.58	62.54

Table 7.4: Core Capacity as a function of Network Diameter for FIFO Edge

this case the difference in capacity between a 10-hop and 1-hop network is now 3.8 for WFQ, 14.8 for CBQ, 12 for PQ and 15.6 for FIFO. We note however that for the same hop-count, the capacities with FIFO in the edge are greater than with WFQ in the edge. Another way to look at the impact of the network diameter is to consider the utilization in the core. We define the core utilization as:

$$\mu = \frac{N_{edge} \sum_{k=1}^{K} N_k \rho_k}{C_{core}}$$

where N_{edge} is the total number of edge nodes, N_k is the number of flows of type k, ρ_k is the average rate of flows of type k and C_{core} is the total core capacity. We will use the results in Table 7.3 to discuss how network

diameter affects the utilization. For WFQ, the utilization ranges from 0.73 in a full-mesh network to 0.14 in a 10-hop network. For CBQ the range is 0.25 to 0.02, for PQ it is 0.4 to 0.03 and for FIFO it is 0.025 to 0.001. In general the utilization decreases with increasing hop count. This is because with a fixed end-to-end delay budget, increasing the number of hops reduces the maximum delay per node which results in more capacity per link to support the same external load. For CBQ, PQ and FIFO there is the added effect of accumulation in burstiness which increases the capacity requirements as the network increases in diameter. For the type of topologies used here, small hop counts are achieved by having more links per core node and intuitively one would expect to have lower utilization when there are more links in the network. The difference comes about because the capacity per link is much higher in networks with more hops which makes the total network capacity exceed that of networks with smaller hops, leading to reduced utilization. In Chapter 9 we look at the relationship between capacity, delay and utilization for CBQ, PQ and FIFO in more detail.

Impact of Network Connectivity

In this section we look at how the network connectivity affects the capacity requirements. For a given number of core nodes, N_{core}, the connectivity is determined by the number of links per core node which ranges from 2 to $N_{core}-1$. With $N_{core}-1$ links per node, the network is the highly-connected full mesh topology with a maximum of 2 hops between edge nodes whereas with 2 links per core node the network is poorly connected with a larger number of hops between edge nodes. We use the cases of 5 and 20 core nodes to illustrate the results for the case of FIFO in the edge. In Figures 7.11 and 7.12 we present results for a network with 5 core nodes and 2 and 4 links per core node respectively.

Comparing the two graphs we notice that in a full-mesh topology with 4 links per core node, both the edge and core capacity are less than in the topology with 2 links per core node. The reason for this is that in both networks the end-to-end delay objective is the same so that the network with more links per core node (smaller diameter) has a higher per-node delay than that with fewer links per core node (larger diameter). A higher per-node delay translates to less capacity required to achieve the delay objective.

Figures 7.13 and 7.14 show the results with 20 core nodes. Comparing the two graphs we observe results similar to the 5 node case. A more interesting comparison is to compare Figure 7.11 with 7.13 and Figure 7.12 with 7.14 which have the same number of links per node but different number of

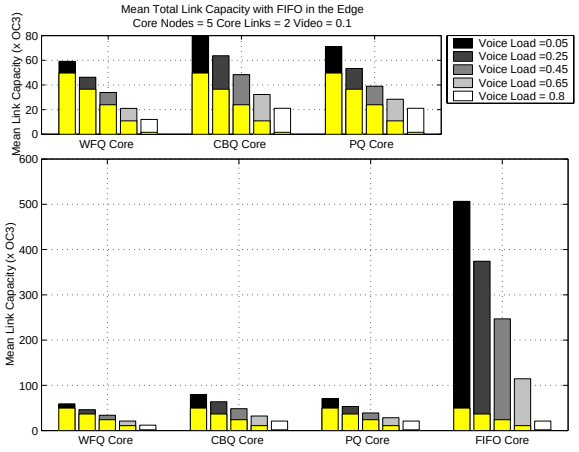

Figure 7.11: Edge and Core Capacity with FIFO in the Edge: 5 nodes poorly-connected

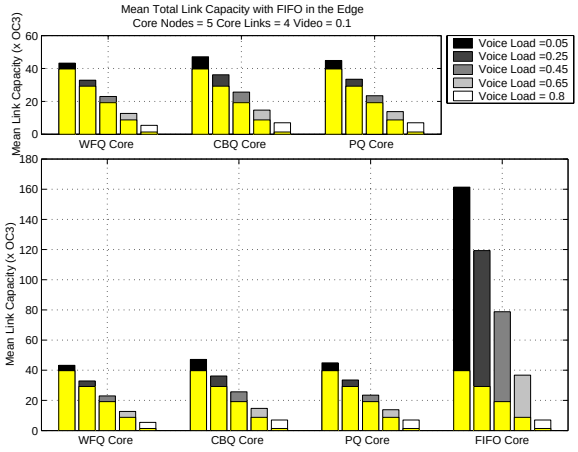

Figure 7.12: Edge and Core Capacity with FIFO in the Edge: 5 nodes highly-connected

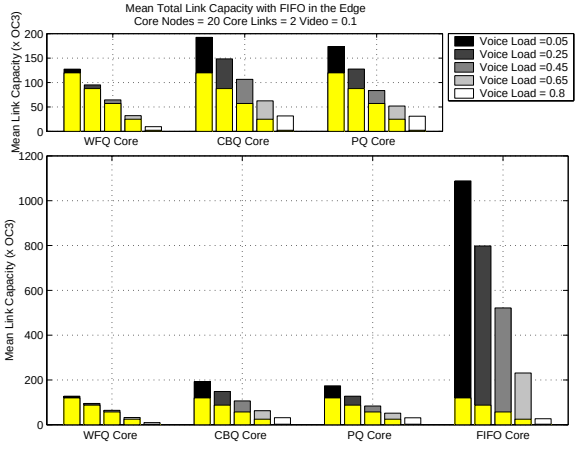

Figure 7.13: Edge and Core Capacity with FIFO in the Edge: 20 nodes poorly-connected

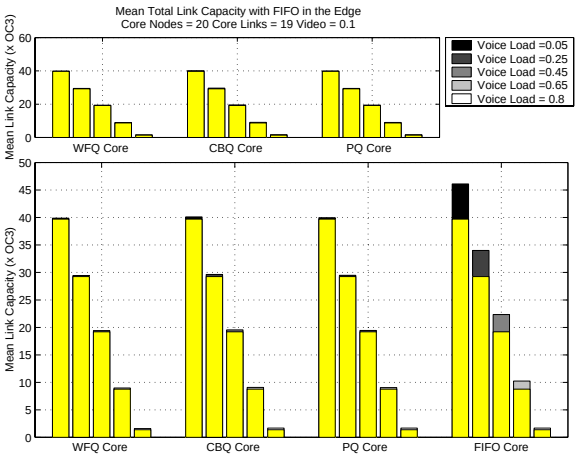

Figure 7.14: Edge and Core Capacity with FIFO in the Edge: 20 nodes highly-connected

core nodes. We find that with 2 links per core node, the capacity with 20 core nodes is significantly higher (by almost a factor of 2) than that with 5 core nodes for all the core schemes whereas with a full mesh topology the capacity with 20 nodes is significantly lower only for the FIFO core case. In fact looking at the full mesh topologies we notice that the edge capacity is the same irrespective of the number of nodes which is to be expected since the network diameter is the same for all full-mesh topologies. In a full-mesh topology, the FIFO core capacity with 20 nodes is less than with 5 core nodes because of the way in which traffic is distributed inside the core: recall that we are using a symmetric distribution in which the traffic sent from one core node to another is equal to the total load from the edge divided by the number of core nodes minus 1. Thus with 20 nodes the traffic between nodes is less than with 5 core nodes. With 2 links per core node, the edge capacity is significantly greater with 20 core nodes since the network diameter is larger and hence the delay per node is much smaller than with 5 core nodes, resulting in more capacity to meet delay objectives. The capacity in the core is less with with 20 nodes again due to the symmetric distribution of traffic in the core.

7.6.4 Effect of Projections on Traffic Growth

Using procedures similar to those in Section 6.2.4, we investigate the impact on core capacity of projected annual growth of 15% in voice traffic and 100% in WWW traffic over a period of 5 years. We use a network with 20 core nodes for illustration. Figure 7.15 shows the capacity required with WFQ in the edge for the case of initial voice load of 40%, video 10%, email 15% and WWW 15%. We note that the WFQ capacity increase by a factor of 2 to two OC-3 links after the 5 year period while CBQ and PQ both increase by a factor of 4 although the CBQ capacity increases slightly faster than PQ capacity between the second and third years. The FIFO capacity increases the most by a factor of almost 9. With FIFO in the edge Figure 7.16 shows the same trend as before for WFQ. The variation in CBQ and PQ capacity is the same in this case and the capacity increases by a factor of 3. The FIFO capacity increases by a factor of almost 14 although the net capacity is less than with WFQ in the edge.

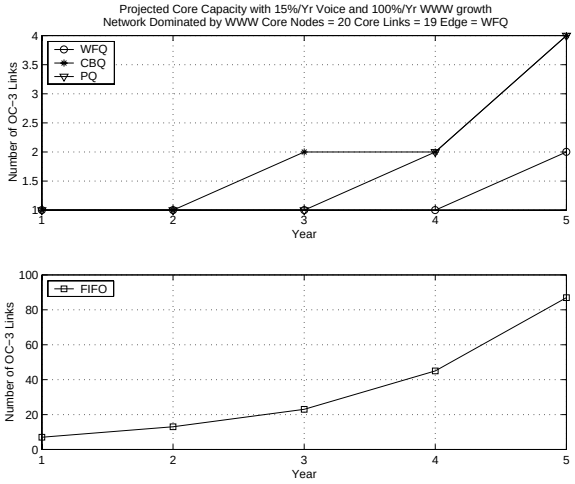

Figure 7.15: Core Capacity for 20 node network with WFQ in the Edge

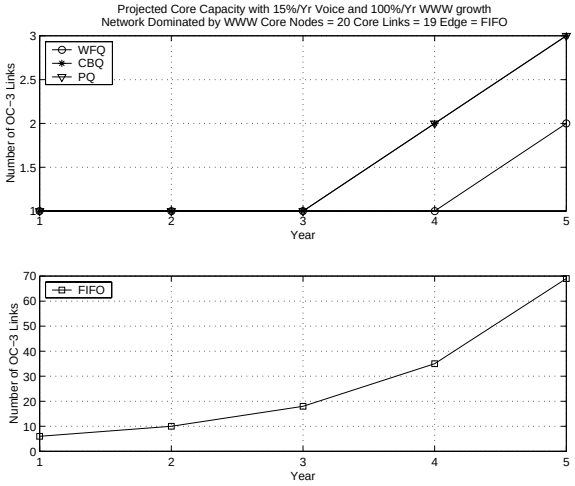

Figure 7.16: Core Capacity for 20 node network with FIFO in the Edge

7.6.5 Impact of Burstiness on Edge and Core Capacity

We tested the impact of burstiness on the edge and core capacity, focusing on WWW burstiness for illustration. We varied the WWW burstiness to give us values that ranged from 1 to 1000 times the burstiness of voice. We used a 20-node full-mesh topology with fixed load of 40% voice, 15% video, 15% email and 15% WWW. For each burstiness value we calculated the edge and core capacity and plotted these against the ratio of WWW to voice burstiness. The results are shown in Figures 7.17 and 7.18.

We notice that for both the edge and core, the difference in capacity between FIFO and the other three schemes widens as the WWW burstiness increases. This further reinforces the observation that the aggregate burstiness is the key reason for the difference in capacity between FIFO and WFQ, CBQ or PQ. For CBQ, the capacity also increases with increasing WWW burstiness but the impact is not as great as in the FIFO case since the delay requirements in the NRT (email and WW) queue are not as stringent as in the FIFO queue. For PQ, the burstiness has no effect because the capacity is solely determined by the RT queue.

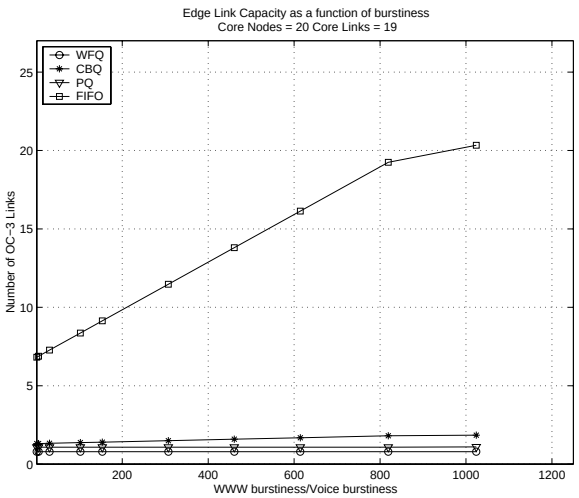

Figure 7.17: Edge Capacity as a function of WWW burstiness

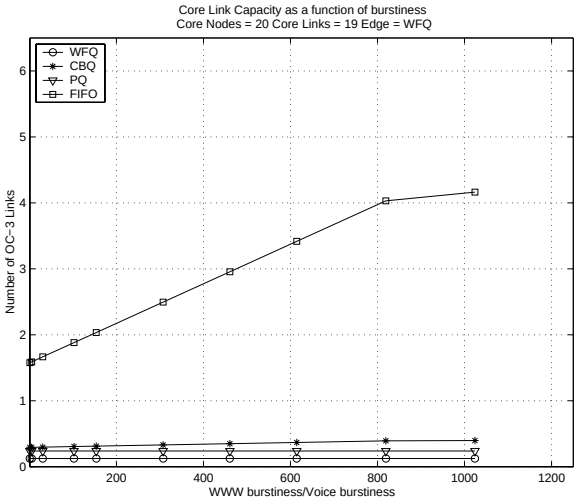

Figure 7.18: Core Capacity as a function of WWW burstiness

7.6.6 Effect of Delay Ratios

In this section we look at the effect of voice and WWW delay requirements on core capacity. We use a full mesh network with 10 core nodes for illustration with a load of 40% voice, 10% video, 15% email and 15% WWW. Results are presented in terms of the ratio of CBQ, PQ and FIFO core capacity to WFQ core capacity. Table 7.5 shows the effect of the voice delay on core capacity when WFQ is used in the edge. The WFQ capacity is not impacted

Voice Delay	$C^{WFQ}(Mbps)$	C^{CBQ}/C^{WFQ}	C^{PQ}/C^{WFQ}	C^{FIFO}/C^{WFQ}
0.01	81.73	4.04	3.0	55.16
0.015	81.73	3.28	2.2	37.01
0.02	81.73	2.93	1.89	27.9
0.025	81.73	2.75	1.7	22.57
0.03	81.73	2.65	1.6	18.9

Table 7.5: Capacity as a function of Voice Delay with WFQ Edge

by the voice delay while for CBQ, PQ and FIFO the capacity decreases with increasing voice delay. For CBQ and PQ the decrease in capacity is because a larger value of voice delay requires less capacity to support the RT queue. For FIFO, a larger value of voice delay also reduces the capacity requirements of the entire queue. Note that for FIFO the relationship is linear in that when the delay is increased by a factor of 3 the capacity reduces by a factor 3. When FIFO is used in the edge, Table 7.6 shows that the WFQ capacity increases with increasing voice delay. This is because as the voice

Voice Delay	$C^{WFQ}(Mbps)$	C^{CBQ}/C^{WFQ}	C^{PQ}/C^{WFQ}	C^{FIFO}/C^{WFQ}
0.01	110	2.47	1.89	31.02
0.015	113	1.99	1.42	20.4
0.02	116	1.76	1.2	15
0.025	119	1.63	1.08	11.96
0.03	122	1.55	1.01	9.89

Table 7.6: Capacity as a function of Voice Delay with FIFO Edge

delay increases there is a corresponding increase in the burstiness of traffic through the FIFO edge and this requires more guaranteed bandwidth in the WFQ core. For CBQ, PQ and FIFO, the capacity decreases with increasing voice delay as before although the capacity requirements are now less than with a WFQ edge. Increasing the WWW delay with a WFQ edge produces results similar to the single-link case as shown in Table 7.7. With FIFO in

WWW Delay	$C^{WFQ}(Mbps)$	C^{CBQ}/C^{WFQ}	C^{PQ}/C^{WFQ}	C^{FIFO}/C^{WFQ}
0.25	81	3.49	2.1	21.78
0.375	81	3.28	1.89	27.95
0.5	81	2.93	1.89	27.97
0.625	81	2.93	1.89	27.97
0.75	81	2.93	1.89	27.97

Table 7.7: Capacity as a function of WWW Delay with WFQ Edge

the edge, increasing the WWW delay reduces the WFQ capacity as shown in Table 7.8. This is because a larger WWW delay reduces the guaranteed

WWW Delay	$C^{WFQ}(Mbps)$	C^{CBQ}/C^{WFQ}	C^{PQ}/C^{WFQ}	C^{FIFO}/C^{WFQ}
0.25	122	1.96	1.15	11.42
0.375	124	1.84	1.13	14.2
0.5	116	1.76	1.2	15.1
0.625	113	1.83	1.25	15.69
0.75	111	1.86	1.27	16

Table 7.8: Capacity as a function of WWW Delay with FIFO Edge

rate for WWW traffic. The CBQ, PQ and FIFO capacities do not change for values of WWW delay above 0.375 - the change in ratios is due to the changing WFQ capacity.

7.7 Summary

In summary the results of this chapter are:

- Any combination of WFQ, CBQ and PQ in the edge and core requires capacity of the same order of magnitude and on the basis of capacity there is no significant difference between the three traffic handling schemes.
- With FIFO in the edge, the edge capacity dominates the total network capacity and again there is no significant difference between WFQ, CBQ and PQ in the core.
- When voice and video are the only traffic types in the network, there is no difference between the four traffic handling schemes.
- For the same delay budget and the same number of core nodes, increasing the network connectivity increases the edge capacity but decreases the core capacity.
- For the same delay budget and the same number of core links per node, a larger network (more core nodes) requires more capacity due to the increased network diameter.
- With WFQ in the edge, WFQ core capacity is not affected much by increases in voice delay while increases in WWW delay decrease the WFQ core capacity. CBQ, PQ and FIFO core capacity decrease with increasing voice delay with WFQ in the edge
- With FIFO in the edge, WFQ core capacity increases with increasing voice delay while CBQ, PQ and FIFO core capacity decreases.
- Increasing the WWW delay has no effect on WFQ core capacity with a WFQ edge and reduces the WFQ core capacity when FIFO is used in the edge.
- The aggregate burstiness is a critical factor for FIFO; as the aggregate burstiness increases, the difference between FIFO capacity and the other three schemes increases.
- With WFQ in the edge, CBQ, PQ and FIFO core capacity is not impacted much by WWW delay while with FIFO in the edge, there is a cutoff delay above which the core capacity does not change with increase in WWW delay.

In the next chapter we look at how the analysis can be used to provide bounds on the capacity requirements of the four traffic handling schemes.

Chapter 8

Bounds on Capacity Requirements

In this section we would like to derive bounds on the ratio of capacity required by CBQ, PQ and FIFO to that required by WFQ. The goal is to provide some simplification to the process of comparing the four schemes and to identify the parameters that are most important in performing the comparison. We begin with the case of a single link and then consider the more general network case.

8.1 Single-Link

From the analysis in Section 6, we note that the (minimum) WFQ capacity is given by:

$$\begin{aligned} C^{WFQ} &= \sum_{k=1}^{K} N_k g_k^{WFQ} \\ &= \sum_{k=1}^{K} N_k \overline{g^{WFQ}} \end{aligned}$$

where $\overline{g^{WFQ}}$ is the average guaranteed rate taken over all flows. The CBQ capacity is given by:

$$
\begin{aligned}
C^{CBQ} &= \sum_{p=1}^{P} \sum_{k \in p} \frac{N_k \sigma_k}{D_{class\ p}} + \sum_{p=1}^{P} \frac{L_p}{D_{class\ p}} \\
&= \sum_{k=1}^{K} N_k \overline{\gamma^{CBQ}} + \sum_{p=1}^{P} \frac{L_p}{D_{class\ p}} \\
&< \sum_{k=1}^{K} N_k \overline{\gamma^{CBQ}} + \sum_{p=1}^{P} \frac{L_p}{D_{min}}
\end{aligned}
$$

where $\overline{\gamma^{CBQ}}$ is defined as:

$$
\overline{\gamma^{CBQ}} = \frac{\sum_{k=1}^{K} \frac{N_k \sigma_k}{D_{k,p}}}{\sum_{k=1}^{K} N_k}
$$

We thus obtain the ratio of CBQ to WFQ capacity as:

$$
\frac{C^{CBQ}}{C^{WFQ}} < \frac{\overline{\gamma^{CBQ}}}{\overline{g^{WFQ}}} + \frac{\sum_{p=1}^{P} \frac{L_p}{D_{min}}}{\sum_{k=1}^{K} N_k \overline{g^{WFQ}}} \tag{8.1}
$$

The capacity with PQ is given by:

$$
C^{PQ} = \max_{p=1...P} \left\{ \sum_{j=1}^{p} \sum_{k \in class\ j} \frac{N_k \sigma_k + L_{max(p)}}{D_{class\ p}} + \sum_{j=1}^{p-1} \sum_{k \in class\ j} N_k \rho_k \right\} \tag{8.2}
$$

$$
= \max_{p=1...P} \left\{ \sum_{j=1}^{p} F_j \overline{\gamma^{PQ}}(p) + \sum_{j=1}^{p-1} F_j \overline{\rho_H}(p) + \frac{L_{max(p)}}{D_{class\ p}} \right\} \tag{8.3}
$$

where F_p is the number of flows in priority levels greater than or equal to p, $\overline{\rho_H}(p)$ is the average rate of all flows of priority greater than p and $\overline{\gamma^{PQ}}(p)$ is given by:

$$
\overline{\gamma^{PQ}}(p) = \frac{\sum_{j=1}^{p} \sum_{k \in class\ j} \frac{N_k \sigma_k}{D_{class\ p}}}{F_p} \tag{8.4}
$$

The ratio of PQ to WFQ capacity is thus:

$$\frac{C^{PQ}}{C^{WFQ}} = \max_{p=1...P} \left\{ \frac{\overline{\gamma^{PQ}}}{\overline{g^{WFQ}}} \frac{\sum_{j=1}^{p} F_j}{\sum_{k=1}^{K} N_k} + \frac{\overline{\rho_H}(p)}{\overline{g^{WFQ}}} \frac{\sum_{j=1}^{p-1} F_j}{\sum_{k=1}^{K} N_k} + \right. \quad (8.5)$$

$$\left. \frac{\frac{L_{max(p)}}{D_{class\ p}}}{\sum_{k=1}^{K} N_k \overline{g^{WFQ}}} \right\} \quad (8.6)$$

For FIFO similar analysis yields:

$$\frac{C^{FIFO}}{C^{WFQ}} = \frac{\overline{\gamma^{FIFO}}}{\overline{g^{WFQ}}} \quad (8.7)$$

where $\overline{\gamma^{FIFO}}$ is given by:

$$\overline{\gamma^{FIFO}} = \frac{\sum_{k=1}^{K} \frac{N_k \sigma_k}{D_{min}}}{\sum_{k=1}^{K} N_k} \quad (8.8)$$

We defer discussion of the nature of the bounds until after the next section where we derive the bounds for the network case.

8.2 Edge-Core Network

In this section we look at how the core capacity with different edge-core traffic handling mechanisms compares to that required by a network using WFQ in both the edge and core. Specifically we obtain bounds on the ratio of maximum core capacity for different edge-core combinations compared to an all WFQ network. The comparison is done for topologies having the same number of core nodes and we make several assumptions on the traffic in the network:

- The load from the edge on each core node is the same so that $N_k^m = N_k$ for each core node m
- Traffic is distributed symmetrically within the core and the distribution is the same for each traffic type so that $\tau_k^{i,j} = \tau_k = \tau = 1/(N_{core} - 1)$ for all traffic types k and all core nodes (i, j)

- There is an upper bound on the number of flows carried by a link (i, j) that depends on the connectivity of the network which is given by:

$$\begin{aligned} N_k^{max}(i,j) &= \sum_{k=1}^{K} N_k * \tau_{max} \\ \tau_{max} &= Link_{flow}(i,j) * \tau \end{aligned}$$

  where $Link_{flow}(i, j)$ is the number of core source-destination pairs whose traffic is routed over link (i, j).

In the reference all-WFQ network, the maximum capacity on a core link is found using Equation 7.10 to be:

$$C_{core}^{WFQ*} = N_{edge} * \sum_{k=1}^{K} N_k * g_k^{WFQ*} * \tau_{max} \tag{8.9}$$

This is the reference case and we will now consider maximum core capacity for different edge-core combinations.

8.2.1 WFQ Core

When the edge does not use WFQ, the maximum WFQ core capacity is given by:

$$C_{core}^{WFQ} \leq N_{edge} \tau_{max} \sum_{k=1}^{K} N_k g_k \tag{8.10}$$

where we recall that g_k is given by:

$$g_k = \max \left\{ \rho_k, \frac{\frac{\sigma_k'}{M} + L_k}{D_k} \right\} \tag{8.11}$$

where σ_k' is the burstiness after passing through the edge portion of the network defined in Equation 7.12 and M is the number of core nodes. Thus the ratio of WFQ core capacity to the reference WFQ* network is:

$$\frac{C_{core}^{WFQ}}{C_{core}^{WFQ*}} < \frac{\overline{g}}{\overline{g^{WFQ*}}} \quad (8.12)$$

8.2.2 CBQ Core

For CBQ using Equation 7.17 the maximum core capacity is given by:

$$\begin{aligned} C_{core}^{CBQ} &\leq N_{edge} * \tau_{max} \sum_{p=1}^{P} \sum_{k \in p} \frac{N_k \sigma_k^{(M)} + L_p}{D_{class\ p}} \\ &\leq N_{edge} * \tau_{max} \sum_{k=1}^{K} \frac{N_k \sigma_k^{(M)}}{D_{k,p}} + \sum_{p=1}^{P} \frac{L_p}{D_{class\ p}} \end{aligned}$$

where $\sigma_k^{(M)}$ is the burstiness of a flow that traverses M core nodes which is given by:

$$\begin{aligned} \sigma_k^{(M)} &= \sigma_k' + (M-1)\rho_k D_{k,p} \\ &= \sigma_k + \rho_k D_k^{edge} + (M-1)\rho_k D_{k,p} \end{aligned}$$

Substituting for $\sigma_k^{(M)}$ we have:

$$\begin{aligned} C_{core}^{CBQ} &\leq N_{edge} * \tau_{max} \sum_{k=1}^{K} N_k \left[\frac{\sigma_k}{g_k^{WFQ*} D_{k,p}} + \frac{\rho_k}{g_k^{WFQ*}} \frac{D_k^{edge}}{D_{k,p}} + (M-1) \frac{\rho_k}{g_k^{WFQ*}} \right] \\ &\quad + \sum_{p=1}^{P} \frac{N_{edge} L_p}{D_{class\ p}} \\ &< N_{edge} * \tau_{max} \sum_{k=1}^{K} N_k \overline{\gamma^{CBQ}} + \overline{\Delta^{CBQ}} + (M-1)\overline{\rho} + \sum_{p=1}^{P} \frac{N_{edge} L_p}{D_{min}} \end{aligned}$$

where $\overline{\gamma^{CBQ}}$ and $\overline{\Delta^{CBQ}}$ are defined as:

$$\overline{\gamma^{CBQ}} = \frac{\sum_{k=1}^{K} \frac{N_k \sigma_k}{D_{k,p}}}{\sum_{k=1}^{K} N_k} \quad (8.13)$$

$$\overline{\Delta^{CBQ}} = \frac{\sum_{k=1}^{K} \frac{\rho_k D_k^{edge}}{D_{k,p}}}{\sum_{k=1}^{K} N_k} \quad (8.14)$$

We thus obtain:

$$\frac{C_{core}^{CBQ}}{C_{core}^{WFQ*}} < \frac{\overline{\gamma^{CBQ}} + \overline{\Delta^{CBQ}} + (M-1)\overline{\rho}}{\overline{g^{WFQ*}}} + \sum_{p=1}^{P} \frac{L_p}{D_{min}} \sum_{k=1}^{K} N_k \overline{g^{WFQ}} \quad (8.15)$$

8.2.3 PQ Core

The maximum capacity for a PQ core is given by:

$$C_{core}^{PQ} \leq N_{edge} \max_{p=1..P} \left\{ \sum_{j=1}^{p} \sum_{k \in j} \frac{\tau_{max} N_k \sigma_k^{(M)} + L_{max}(p)}{D_{k,p}} + \sum_{j=1}^{p-1} \sum_{k \in j} \tau_{max} N_k \rho_k \right\}$$

where $\sigma_k^{(M)}$ has been defined in Equation 8.13. Proceeding as we did with the single-link case we obtain:

$$C_{core}^{PQ} \leq N_{edge} \max_{p=1..P} \left\{ \sum_{j=1}^{p} F_j \overline{\gamma^{PQ}}(p) + \overline{\Delta^{PQ}}(p) + (M-1)\overline{\rho}(p) \right. \quad (8.16)$$

$$\left. + \sum_{j=1}^{p-1} F_j \overline{\rho_H}(p) + \frac{L_p}{D_{class\ p}} \right\} \quad (8.17)$$

where F_p is the number of flows of priority greater than or equal to p, $\overline{\rho}(p)$ is average rate over flows of priority greater than or equal to p, $\overline{\rho_H}(p)$ is the average rate over flows of priority greater than p and $\overline{\gamma^{PQ}}(p)$ and $\overline{\Delta^{PQ}}(p)$ are given by:

$$\overline{\gamma^{PQ}}(p) = \frac{\sum_{j=1}^{p} \sum_{k\ \in\ class\ j} \frac{N_k \sigma_k}{D_{class\ p}}}{F_p} \quad (8.18)$$

$$\overline{\Delta^{PQ}} = \frac{\sum_{j=1}^{p} \sum_{k\ \in\ class\ j} \frac{N_k \rho_k D_k^{edge}}{D_{class\ p}}}{F_p} \quad (8.19)$$

The ratio of PQ to WFQ* core capacity is thus:

$$\frac{C_{core}^{PQ}}{C_{core}^{WFQ*}} < \max_{p=1..P} \left\{ \frac{\overline{\gamma^{PQ}}(p) + \overline{\Delta^{PQ}}(p) + (M-1)\overline{\rho}(p)}{\overline{g^{WFQ*}}} \frac{\sum_{j=1}^{p} F_j}{\sum_{k=1}^{K} N_k} \right. \quad (8.20)$$

$$\left. + \frac{\overline{\rho_H}(p)}{\overline{g^{WFQ*}}} \frac{\sum_{j=1}^{p-1} F_j}{\sum_{k=1}^{K} N_k} + \frac{\frac{L_p}{D_{class\ p}}}{\sum_{k=1}^{K} N_k \overline{g^{WFQ*}}} \right\} \quad (8.21)$$

8.2.4 FIFO Core

The maximum capacity in a FIFO core is given by:

$$C_{core}^{FIFO} \leq N_{edge} \tau_{max} \sum_k \frac{N_k \sigma_k^{(M)}}{D_{min}} \quad (8.22)$$

Substituting for $\sigma_k^{(M)}$ we obtain:

$$C_{core}^{FIFO} \leq N_{edge} \tau_{max} \sum_k N_k \left(\overline{\gamma^{FIFO}} + \overline{\Delta^{FIFO}} + (M-1)\overline{\rho} \right) \quad (8.23)$$

where $\overline{\gamma^{FIFO}}$ and $\overline{\Delta^{FIFO}}$ are given by:

$$\overline{\gamma^{FIFO}} = \frac{\sum_{k=1}^{K} \frac{N_k \sigma_k}{D_{min}}}{\sum_{k=1}^{K} N_k} \quad (8.24)$$

$$\overline{\Delta^{FIFO}} = \frac{\sum_{k=1}^{K} \frac{N_k \rho_k D_k^{edge}}{D_{min}}}{\sum_{k=1}^{K} N_k} \quad (8.25)$$

Thus we obtain the ratio of FIFO to WFQ core capacity as:

$$\frac{C_{core}^{FIFO}}{C_{core}^{WFQ*}} < \frac{\overline{\gamma^{FIFO}} + \overline{\Delta^{FIFO}} + (M-1)\overline{\rho}}{\overline{g^{WFQ*}}} \quad (8.26)$$

Looking at the bounds on capacity we notice that the CBQ, PQ and FIFO bounds have certain elements in common. In each case the capacity is related to a parameter $\overline{\gamma}$ which is determined by the ratio of burstiness within an aggregate to the minimum delay requirements of that aggregate. For edge-core networks, the delay in the edge influences the capacity bounds and this is captured by the parameter $\overline{\Delta}$ which is a function of the ratio of edge delay to core delay. We also note the factor (M-1) which captures the effect of the network diameter on the capacity bounds. Since we know that the average rate ρ is always less than or equal to the guaranteed rate g, we expect that the ratios will be lower-bounded by (M-1). We have thus established bounds on the capacity requirements that are related to the ratios of delay objectives of the traffic that is aggregated within the network. This is a new result which is able to relate differences in capacity to differences in delay requirements. In the next section we present some numerical results on the accuracy of the bounds.

8.3 Numerical Results on Capacity Bounds

To demonstrate the use of the capacity bounds we will focus on the case of a single link loaded with 40% voice, 10% video, 15% email and 15% WWW and look at how the bounds change for varying voice delay. Starting with CBQ we show in Table 8.1 the guaranteed rate and $\overline{\gamma^{CBQ}}$ along with the simplified bounds obtained in this section and the bounds obtained from the exact expressions in Chapter 7. We will refer to the bounds as the simple and exact bounds respectively.

Voice Delay	$\overline{g^{WFQ}}$	$\overline{\gamma^{CBQ}}/\overline{g^{WFQ}}$	C^{CBQ}/C^{WFQ}	
(sec)	(Mbps)		Simple	Exact
0.001	0.45	0.99	1.25	1.08
0.0015	0.41	0.8	0.88	0.86
0.002	0.37	0.72	0.78	0.76
0.0025	0.34	0.66	0.71	0.7
0.003	0.31	0.64	0.68	0.66

Table 8.1: Ratio of CBQ to WFQ Capacity as a function of Voice Delay

We find that the simple bounds are accurate and we note that $\overline{\gamma^{CBQ}}$ can be used to provide reasonable estimates in the absence of information about packet sizes. The results for PQ are shown in Table 8.2. We note that for this particular case the PQ capacity is maximized by the needs of the

Voice Delay	$\overline{g^{WFQ}}$	$(F_1\overline{\gamma^{PQ}})/(N\overline{g^{WFQ}})$	C^{PQ}/C^{WFQ}	
(sec)	(Mbps)		Simple	Exact
0.001	0.45	0.79	0.89	0.88
0.0015	0.41	0.61	0.68	0.67
0.002	0.37	0.52	0.57	0.57
0.0025	0.34	0.47	0.51	0.5
0.003	0.31	0.44	0.47	0.46

Table 8.2: Ratio of PQ to WFQ Capacity as a function of Voice Delay

RT class with p equal to 1 so we do not tabulate the NRT parameters. We observe that the bounds are determined by the $\overline{\gamma^{PQ}}$ parameter along with the ratio F_1/N where F_1 is the number of flows in priority level 1 and N is the total number of flows. For FIFO the results are shown in Table 8.3.

Voice Delay	$\overline{g^{WFQ}}$	Simple	Exact
(sec)	(Mbps)	$\overline{\gamma^{FIFO}}/\overline{g^{WFQ}}$	C^{FIFO}/C^{WFQ}
0.001	0.45	101.3	99.6
0.0015	0.41	66.7	66.5
0.002	0.37	50.2	49.7
0.0025	0.34	40.07	39.8
0.003	0.31	33.8	33.3

Table 8.3: Ratio of FIFO to WFQ Capacity as a function of Voice Delay

For FIFO the bounds are determined solely by $\overline{\gamma^{FIFO}}$ and we see a good match between the simple and exact bounds. The importance of the bounds presented here is that they can be used to perform preliminary assessments of capacity requirements based on knowledge of just the burstiness, average rate and delay parameters. For edge-core networks the impact of the network diameter as well as distribution of delay between the edge and core portions can also be easily assessed using the bounds. We leave investigation of the use of the bounds for edge-core networks for future work. In the next chapter we look at the relationship between network capacity, end-to-end delay and utilization for CBQ, PQ and FIFO.

Chapter 9

Aggregation, Network Capacity and Utilization

In this chapter we focus on how for CBQ, PQ and FIFO the accumulation of burstiness as a flow propagates through a network affects the delay at each subsequent node and how this ultimately affects the capacity and utilization. Our goal is to find an expression for end-to-end delay that captures the tradeoff between link utilization, network capacity and end-to-end delay. We expect such information to be useful in addressing such questions as what is the maximum utilization possible for a given end-to-end delay? We will assume a network that uses path-level aggregation so that flows within a class share the same end-to-end path and are queued independently of flows in the same class that are not on their path. This differs from the analysis in Chapter 7 where flows were aggregated only according to the class they belonged to and not according to how they were routed. The more general case assumes no knowledge of routing within the network and we will discuss this using results presented by other researchers.

We have seen from Chapter 7 that delays incurred in network elements will impact the burstiness of traffic as it flows through subsequent elements. Burstiness is accumulated at each node according to the general equation [23, 52]:

$$\begin{aligned} \sigma_k^{(m)} &= \sigma_k^{(m-1)} + \rho_k * \theta_k^{(m-1)} \\ \sigma_k^{(1)} &= \sigma_k \end{aligned} \tag{9.1}$$

where σ_k is the original burstiness, $\sigma_k^{(m)}$ is the burstiness at the input to

node m and θ_k^m is the latency at the m^{th} node. If the latency at each node is the same we have:

$$\sigma_k^{(m)} = \sigma_k + (m-1) * \rho_k * \theta_k \tag{9.2}$$

In this section we examine how this accumulation of burstiness impacts the end-to-end delay and the maximum allowable utilization for FIFO, PQ and CBQ networks. We start by finding a general expression for the delays in these three systems.

We begin by recalling the delay in a single server with link capacity C for each of these three systems.

1. FIFO

$$\begin{aligned} \theta^{(m)} &= \frac{\overline{\sigma}^{(m)}}{C^{(m)}} \\ \overline{\sigma}^{(m)} &= \sum_{k \in S(m)} \sigma_k \end{aligned}$$

2. Priority Queueing

$$\theta_p^{(m)} = \frac{\overline{\sigma}_H^{(m)}(p) + L_{max}(p)}{C^{(m)} - \overline{\rho}_H^{(m)}(p)}$$

3. Class-Based Queueing

$$\theta_p^{(m)} = \frac{\overline{\sigma}_p^{(m)} + L_p^{(m)}}{g_p^{(m)}} + \frac{L_{max}}{C^{(m)}}$$

A general expression that covers all three cases takes the form:

$$\theta_p = \frac{\Omega_p}{C_p}$$

where, for FIFO:

$$\begin{aligned} \Omega_p &= \Omega = \overline{\sigma} \\ C_p &= C \end{aligned}$$

For Priority Queueing (PQ):

$$\begin{aligned} \Omega_p &= \overline{\sigma}_H(p) + L_{max}(p) \\ C_p &= C - \overline{\rho}_H(p) \end{aligned}$$

For Class-Based Queueing (CBQ):

$$\begin{aligned} \Omega_p &= \overline{\sigma}_p + L_p \\ C_p &= g_p \end{aligned}$$

where we have assumed that the term L_{max}/C is negligible. We will use the notation $D_p^{(m)}$ to represent the delay experienced by traffic class p at node m and $\overline{\sigma_p}^{(m)}$ to represent the aggregate burstiness of class p after passing through node $(m-1)$ (in other words the aggregate burstiness seen by node m) and $\overline{\rho}_p$ to represent the aggregate average rate for traffic of class p. Note that with FIFO there is only one class which is composed of all flows sharing the queue.

Initially we have:

$$\begin{aligned} \overline{\sigma_p}^{(1)} &= \overline{\sigma}_p \\ \Omega_p^{(1)} &= \begin{cases} \overline{\sigma}_p^{(1)} & for\ \ FIFO \\ \overline{\sigma}_H(p)^{(1)} + L_{max}(p) & for\ \ PQ \\ \overline{\sigma}_p^{(1)} + L_p & for\ \ CBQ \end{cases} \end{aligned}$$

After the first node we have:

$$D_p^{(1)} = \frac{\Omega_p^{(1)}}{C_p^{(1)}}$$

$$
\begin{aligned}
\overline{\sigma_p}^{(2)} &= \overline{\sigma_p}^{(1)} + \overline{\rho_p} * D_p^{(1)} \\
&= \overline{\sigma_p}^{(1)} + \frac{\Omega_p^{(1)}}{C_p^{(1)}}\overline{\rho_p} \\
\Omega_p^{(2)} &= \frac{\Omega_p^{(1)}}{C_p^{(1)}}(C_p^{(1)} + \overline{\rho_p})
\end{aligned}
$$

After the second node we have:

$$
\begin{aligned}
D_p^{(2)} &= \frac{\Omega_p^{(2)}}{C_p^{(2)}} \\
&= \frac{\Omega_p^{(1)}}{C_p^{(1)}C_p^{(2)}}\left(C_p^{(1)} + \overline{\rho_p}\right) \\
\overline{\sigma_p}^{(3)} &= \overline{\sigma_p}^{(2)} + \overline{\rho_p} * D_p^{(2)} \\
&= \overline{\sigma_p}^{(1)} + \frac{\Omega_p^{(1)}}{C_p^{(1)}C_p^{(2)}}\left(\overline{\rho_p}C_p^{(1)} + \overline{\rho_p}C_p^{(2)} + \overline{\rho_p}^2\right) \\
\Omega_p^{(3)} &= \frac{\Omega_p^{(1)}}{C_p^{(1)}C_p^{(2)}}\left(C_p^{(1)}C_p^{(2)} + \overline{\rho_p}C_p^{(1)} + \overline{\rho_p}C_p^{(2)} + \overline{\rho_p}^2\right)
\end{aligned}
$$

Proceeding in this manner we obtain after the m^{th} node:

$$
\begin{aligned}
\overline{\sigma_p}^{(m)} &= \overline{\sigma_p}^{(1)} + \frac{\Omega_p^{(1)}}{C_p^{(1)}C_p^{(2)}.....C_p^{(m-1)}} \sum_{x=1}^{m-1} \overline{\rho_p}^x \sum_{i=1}^{\left(\begin{smallmatrix} m-1 \\ x \end{smallmatrix}\right)} \Phi(i, m-1-x) \\
\Omega_p^{(m)} &= \frac{\Omega_p^{(1)}}{C_p^{(1)}C_p^{(2)}.....C_p^{(m-1)}} \sum_{x=0}^{m-1} \overline{\rho_p}^x \sum_{i=1}^{\left(\begin{smallmatrix} m-1 \\ x \end{smallmatrix}\right)} \Phi(i, m-1-x) \\
D^{(m)} &= \frac{\Omega_p^{(1)}}{C_p^{(1)}C_p^{(2)}....C_p^{(m)}} \sum_{x=0}^{m-1} \overline{\rho_p}^x \sum_{i=1}^{\left(\begin{smallmatrix} m-1 \\ x \end{smallmatrix}\right)} \Phi(i, m-1-x)
\end{aligned}
$$

where $\Phi(i, m-1-x)$ is the i^{th} member of the set of $(m-1-x)$-element combinations from the set $\{C_p^{(1)}, C_p^{(2)}, ...C_p^{(m-1)}\}$. Note that we assume that

permutations of the same elements constitute a single combination. For instance $C_1C_2C_3 \equiv C_2C_1C_3 \equiv C_3C_2C_1$ and so on.

The end-to-end delay is then given by:

$$\begin{aligned} D_p^{E2E} &= \sum_{m=1}^{M} D_p^{(m)} \\ &= \sum_{m=1}^{M} \frac{\Omega_p^{(1)}}{C_p^{(1)}C_p^{(2)}....C_p^{(m)}} \sum_{x=0}^{m-1} \overline{\rho_p}^x \sum_{i=1}^{\left(\begin{array}{c} m-1 \\ x \end{array} \right)} \Phi(i, m-1-x) \end{aligned}$$

- Suppose that $C_p^{(m)} \geq C_p$, then

$$\begin{aligned} D_p^{E2E} &\leq \sum_{m=1}^{M} \left(\frac{\Omega_p^{(1)}}{C_p^m} \sum_{x=0}^{m-1} \overline{\rho_p}^x \left(\begin{array}{c} m-1 \\ x \end{array} \right) C_p^{m-1-x} \right) \\ &\leq \sum_{m=1}^{M} \frac{\Omega_p^{(1)}}{C_p^m} (\overline{\rho_p} + C_p)^{m-1} \end{aligned}$$

- Suppose further that $\overline{\rho_p} < \alpha_p * C_p$ with $0 < \alpha_p < 1$. Then,

$$\begin{aligned} D_p^{E2E} &\leq \sum_{m=1}^{M} \frac{\Omega_p^{(1)}}{C_p^m} C_p^{m-1} (1+\alpha_p)^{m-1} \\ &\leq \frac{\Omega_p^{(1)}}{C_p} \sum_{m=1}^{M} (1+\alpha_p)^{m-1} \\ &\leq \frac{\Omega_p^{(1)}}{C_p} \left(\frac{(1+\alpha_p)^M - 1}{\alpha_p} \right) \end{aligned}$$

From this we observe that increasing the number of nodes traversed M, decreases the utilization factor α_p for a given end-to-end delay objective and to maintain the same α_p for different values of M we must either increase the capacity C_p or decrease $\Omega_p^{(1)}$.

The authors in [17] show how bounding the maximum delay at each node affects the allowable utilization for a network using static priority servers for the more general case where there is no assumption of path-level aggregation.

Their results provide very conservative bounds on the allowable utilization. To compare their results, hereafter referred to as the non-path aggregation (non-PA) bound, to the bounds with path-aggregation(PA) we will consider the case of the high-priority queue in a static priority server which is the case that they focus on. The bound on the end-to-end delay in a non-PA network is given by [17]:

$$D_{E2E}^{non-PA} = \frac{\sigma + L}{C}\left(\frac{M}{(1-(M-1)\alpha)}\right) \tag{9.3}$$

From Equation 9.3, we have:

$$D_{E2E}^{PA} = \frac{\Omega_p^{(1)}}{C_p}\left(\frac{(1+\alpha_p)^M - 1}{\alpha_p}\right) \tag{9.4}$$

$$= \frac{\sigma + L}{C}\left(\frac{(1+\alpha)^M - 1}{\alpha}\right) \tag{9.5}$$

We note that the two equations differ in the multiplying factor that is applied to the single-node delay $(\sigma + L)/C$ thus in comparing the two we will focus only on the multiplying factors. We will consider two cases: one with low utilization ($\alpha = 0.1$) and the other with a high utilization factor ($\alpha = 0.9$). The results are shown in Table 9.1:

Nodes	Utilization = 0.1		Utilization = 0.9	
	non-PA	PA	non-PA	PA
1	1	1	1	1
2	2.2	2.1	20	2.9
3	3.75	3.31	*	6.51
10	100	15.7	*	147.7

Table 9.1: Comparison of End-to-End Delay Bounds

Looking at the results we see that for a single node, the two bounds are identical irrespective of the utilization. With 0.1 utilization, the bounds are similar when the number of nodes is low. With 10 nodes however, the non-PA bound is much higher than the PA bound. With 0.9 utilization, the non-PA bounds allow for a network with no more than 2 nodes whereas the PA bound poses no such restriction. We thus find that the non-PA bounds are extremely conservative and it is still an open issue whether better

bounds can be obtained for the general case. Path-level aggregation has been studied through simulation in [57] where the key result was that better end-to-end delay performance was obtained compared to class aggregation with jitter control in which flows do not necessarily share the same path but share per-class queues in network nodes. We have provided new results in this chapter which extend research in this area by providing an analytical framework for studying the performance of path-level aggregation. In the next chapter we look at how sensitivity analysis can be performed using a stochastic description of the delay requirements.

Chapter 10

Sensitivity Analysis of Capacity Requirements

The analysis and comparison of capacity requirements among the different traffic handling schemes is based on the assumption of knowledge of the maximum delay bounds for each traffic type carried in the network. Typically, these values may be obtained based on recommendations in published standards or based on observations of network performance over time. Whatever the source of the values, a network designer must consider how imperfect knowledge of these parameters will affect capacity requirements and the degree to which each traffic handling mechanism is affected by uncertainty in the delay bounds. Uncertainty and sensitivity analysis are two methods that can be used to address such issues.

The first question that we consider is: what is the uncertainty in the capacity requirements given the uncertainty in the delay bounds? This is dealt with mainly by uncertainty analysis which is performed by examining the probability distribution of the capacity given assumptions on the distributions of the delay bounds. The second question is how important are the individual delay bounds for each traffic type with respect to the uncertainty in the capacity? Answering this question requires sensitivity analysis to rank the contributions of each delay bound to the uncertainty in the capacity and determine which bounds are more significant than others for each traffic handling scheme.

We will begin by giving an overview of uncertainty and sensitivity analysis and some methods used to perform such analyses for general models based largely on the work in [71]. We will then discuss how some of these methods can be applied to examine the sensitivities of the traffic handling

schemes to delay bounds for the simple case of a single network link. We will then present some examples validating the analysis with the results of Monte Carlo simulations. Lastly we will discuss practical applications of sensitivity analysis and how the analysis can be used to address architectural issues in the design of edge-core networks.

10.1 Uncertainty and Sensitivity Analysis

Uncertainty analysis is the first step in understanding how a model's output is influenced by its input parameters. Uncertainty analysis aims to quantify the overall uncertainty in the output as a result of uncertainties in the input. One way of presenting uncertainty results is to evaluate the mean and variance of the output given a random sampling of the input values according to their prescribed probability distribution functions. Another way is to consider the probability density function(pdf) of the output. The probabilities extracted from the pdf characterize subjective uncertainty in the output and give quantitative measures of the range of the output. Uncertainty analysis can also be used to estimate the variance of the output given the variance in the input.

Uncertainty analysis can be performed by considering the variation of the output for small one-at-a-time changes in the input parameters. This is also known as local sensitivity analysis. Local sensitivity is usually assessed through the partial derivatives of the output functions with respect to each input variable. Numerical computation of the partial derivatives is accomplished by allowing each variable to vary within a small interval around a nominal value. Local sensitivity analysis assumes that the input-output relationship is linear within the interval of variation and examines the impact of a parameter when all other parameters are held constant.

Local sensitivities provide the slope of the model output in the parameter space at a given set of value and allow for a rapid preliminary analysis. Given a system described by $\mathbf{y} = \mathbf{f}(\mathbf{x})$, an estimate of the i^{th} system output due to changes in the inputs is obtained by a Taylor-series approximation as:

$$y_i(\mathbf{x} + \Delta\mathbf{x}) = y_i(\mathbf{x}) + \sum_{j=1}^{m} \frac{\partial y_i}{\partial x_j}\Delta x_j + \frac{1}{2}\sum_{j=1}^{m}\sum_{k=1}^{m} \frac{\partial^2 y_i}{\partial x_j \partial x_k}\Delta x_j \Delta x_k + ...(10.1)$$

The partial derivatives $\partial y_i/\partial x_j$ are called first-order local sensitivities while

the derivatives $\partial^2 y_i/\partial x_j \partial x_k$ are second-order local sensitivities. A sensitivity matrix S is formed from the first-order sensitivities such that $S = \{s_{ij}\} = \{\partial y_i/\partial x_j\}$. In the absence of a numerical model formulation, a finite-difference approximation can be used to estimate the first -order sensitivity indices by changing one parameter at a time and calculating the model output for each parameter change. Thus,

$$\frac{\partial y_i}{\partial x_j} \approx \frac{y_i(x_j + \Delta x_j) - y_i(x_j)}{\Delta x_j} \tag{10.2}$$

A higher sensitivity index indicates that the model output is more sensitive to changes in that parameter. The local sensitivity indices can also be used as a first step in sensitivity analysis. Given probability distribution functions for the input variables x_j, let $\bar{y}$ denote the output of the system when the inputs are all at their mean values $\bar{x_j}$. Then, the expected value of the output $E\{y\}$ for arbitrary inputs is given by a Taylor series expansion around $\bar{y}$ as follows [2]:

$$E\{y_i\} = \bar{y_i} + \frac{1}{2}\sum_{j=1}^{m} \frac{\partial^2 y_i}{\partial x_j^2} var[x_j] + \sum_{j=1}^{m}\sum_{k=1}^{m} \frac{\partial^2 y_i}{\partial x_j \partial x_k} cov(x_j, x_k) + ... \tag{10.3}$$

where $var[x_j]$ is the variance of x_j and $covar(x_j, x_k)$ is the covariance of inputs x_j and x_k. Similarly, the variance of the output y_i may be estimated by [2]:

$$\begin{aligned} \sigma_{y_i}^2 &= \sum_{j=1}^{m} \left(\frac{\partial y_i}{\partial x_j}\right)^2 var[x_j] + 2\sum_{j=1}^{m}\sum_{k=1}^{m} \left(\frac{\partial y_i}{\partial x_j}\right)\left(\frac{\partial y_i}{\partial x_k}\right) cov(x_j, x_k) \\ &\quad + \sum_{j=1}^{m} \left(\frac{\partial y_i}{\partial x_j}\right)\left(\frac{\partial^2 y_i}{\partial x_j^2}\right) \mu_3(x_j) \end{aligned} \tag{10.4}$$

where $\mu_3(x_j)$ is the third moment of x_j. Simplifications to the above equations can be made when the third moments are insignificant and the input parameters are uncorrelated yielding:

$$E\{y_i\} = \bar{y_i} + \frac{1}{2}\sum_{j=1}^{m} \frac{\partial^2 y_i}{\partial x_j^2} var[x_j] \tag{10.5}$$

$$\sigma_{y_i}^2 = \sum_{j=1}^{m} \left(\frac{\partial y_i}{\partial x_j} \right)^2 var[x_j] \tag{10.6}$$

The variance of the output indicates how much uncertainty in each parameter contributes to the uncertainty of the output. Each parameter contributes an amount ν_{ij} (called the partial variance) to the uncertainty in the output y_i given by:

$$\nu_{ij} = s_{ij}^2 * \sigma_{x_j}^2 \tag{10.7}$$

$$\nu_{ij}\% = \frac{s_{ij}^2 * \sigma_{x_j}^2}{\sigma_{y_i}^2} * 100 \tag{10.8}$$

Thus by ranking the contributions to the variance of each parameter, the most important parameter for each output variable y_i can be identified. This is one way of performing a preliminary sensitivity analysis.

Sensitivity analysis deals with the issue of how the variation of a model's output can be apportioned either qualitatively or quantitatively to different sources of variation. The input to a model is subject to uncertainty that may arise either due to absence of information, errors in measurement or partial understanding of the driving forces and mechanisms of the system under investigation. Quantitative sensitivity analysis is used to rank the importance of model parameters. Sensitivity analysis apportions the output uncertainty to the uncertainty in the inputs which are described by probability distribution functions that determine acceptable ranges for the inputs. The choice of probability distribution functions and the ranges represent the knowledge or lack of knowledge about each input. A global analysis looks at the influence of a parameter in the face of variations in other parameters as well. In addition to the use of partial variances, sensitivity analysis can be accomplished through many other methods and we discuss a few representative methods.

- Scatter-plots

  These are plots of the output against each input which are used to determine the nature of the relationship - linear or nonlinear, monotonic or non-monotonic - between the input and output. Scatter-plots offer a qualitative measure of the relative importance of each input factor.

- Correlation coefficients

Correlation between each input and the model output can provide some insight into the importance ranking of input variables. Correlation coefficients are based on the assumption of a linear relationship between the input and output. For relationships that are not very linear but that are monotonic, the data can be replaced by their rank transformations and correlation coefficients calculated using the ranks.

- Measures of Importance

  The measures of importance are based on the notion that the importance of an input x_j can be assessed by evaluating the conditional probability distribution of the output y conditioned on the input x_j. This stems from the fact that the marginal pdf of the output can be written in terms of the conditional probability distribution :

$$p_Y = \int p_{Y|X}(y|x)p_X(x)dx \tag{10.9}$$

  The importance of an input variable x_j is thus related to how well it controls the output and the expectation is that if fixing the value of x_j substantially reduces the output variance relative to the marginal variance then x_j is an important factor. The output variance is given by:

$$Var[Y] = Var_X[E\{Y|X\}] + E_X\{Var[Y|X]\} \tag{10.10}$$

  The first term in Equation 10.10 is called the variance of the conditional expectation (VCE) and it captures how well x controls y by looking at how closely $E\{Y|X\}$ matches y. Specifically, if when x is varied, the total variation in y is matched by $E\{Y|X\}$, then x is a very important input for the model. The VCE is given by:

$$Var_X[E\{Y|X\}] = \int [E\{Y|x\} - E\{Y\}]^2 p_X(x)dx \tag{10.11}$$

  The second term in Equation 10.10 is called the residual and it measures the remaining variability in y that is attributable to other unknown sources of variation when x is fixed.

  Using the VCE, one measure of importance is the correlation ratio [59] given by:

$$\eta^2(x_j) = \frac{VCE(x_j)}{Var(Y)} \tag{10.12}$$

When the input factors are independent (uncorrelated) the VCE can be calculated as:

$$\begin{aligned} VCE(x_j) &= Var(Y) - E\{Var(Y|X_j)\} && (10.13)\\ &= \int [E\{Y|X_j\}]^2 p_{x_j}(X_j) dx_j - [E(Y)]^2 && (10.14) \end{aligned}$$

The quantity $U_j = \int [E\{Y|X_j\}]^2 p_{X_j}(x_j) dx_j$ can be obtained analytically or estimated numerically using Monte Carlo simulations by:

$$U_j = \frac{1}{n} \sum_{k=1}^{n} y_k y'_k \tag{10.15}$$

where y_k is the k^{th} sample output (i.e. output due to the k^{th} set of input values) and y'_k is obtained by resampling all the inputs except the j^{th} one.

Having presented a general overview of sensitivity and uncertainty analysis we proceed now to consider how we can apply the theory to the specific problem of assessing the impact of uncertainty in delay bounds on network capacity. We will use methods based on local sensitivity and leave global methods for future work. We consider the simpler case of a single link in detail to develop the analysis and later discuss how the analysis can be extended to arbitrary edge-core networks.

10.2 Sensitivity Analysis for a Single Link

Having laid the theoretical foundation for uncertainty and sensitivity analysis we now turn to the application of these concepts to the uncertainty and sensitivity analysis of link capacity for a single link using either Weighted Fair Queueing (WFQ), Class-Based Queueing (CBQ), Priority Queueing (PQ) or FIFO for traffic handling. We begin by recalling the equations for link capacity for each of the four schemes and starting with WFQ we have:

$$C^{WFQ} = \sum_{k=1}^{K} N_k g_k^{WFQ} \tag{10.16}$$

$$g_k^{WFQ} = \max\left\{\frac{\sigma_k + L_k}{D_k}, \rho_k\right\} \tag{10.17}$$

Since we are interested in the impact of the delay values D_k, we will only consider the case where g_k^{WFQ} is determined solely by the delay D_k so that[1]:

$$C^{WFQ} = \sum_{k=1}^{K} N_k \left(\frac{\sigma_k + L_k}{D_k}\right) \tag{10.18}$$

For Class-Based Queueing and Priority Queueing we have:

$$C^{CBQ} = \sum_{p=1}^{P} \left(\frac{\sum_{k\in p} N_k\sigma_k}{D_{class\ p}} + \frac{L_p}{D_{class\ p}}\right) \quad p = 1, 2, ...P \tag{10.19}$$

$$C^{PQ} = \max_p \left\{\sum_{j=1}^{p} \left(\frac{\sum_{k\ \in\ class\ j} N_k\sigma_k}{D_{class\ p}} + \frac{L_{max(p)}}{D_{class\ p}}\right) + \sum_{j=1}^{p-1} \sum_{k\ \in\ class\ j} N_k\rho_k\right\} \quad p = 1, 2, ...P \tag{10.20}$$

$$L_{max}(p) = \max_{j\geq p}\{L_j\} \tag{10.21}$$

where L_j is the maximum packet size for class or priority level j.

For FIFO we recall:

$$C^{FIFO} = \sum_{k=1}^{K} \frac{N_k\sigma_k}{D_{min}} \tag{10.22}$$

[1] In practice we would expect this to hold for only some delay values with the guaranteed rate being equal to the average rate ρ_k in some cases. Based on the delay distribution, the analysis for this more general case would have to be modified to include information about the range of values for which g_k was determined by D_k for each traffic type k.

where $D_{min} = \min_k\{D_k\}$.

We assume that the delay bounds D_k have known independent distributions with means $\overline{D}_k$. We also assume that we assess the impact of the delay distribution of each traffic type by sampling the distribution of interest with all other delay values set to their mean values. Applying Equations 10.5 and 10.6 the expected value and variance of the capacity will be given by:

$$E\{C\} = \overline{C} + \frac{1}{2}\sum_{k=1}^{K}\frac{\partial^2 C}{\partial \overline{D}_k^2}Var[D_k] \quad (10.23)$$

$$Var[C] = \sum_{k=1}^{K}\left(\frac{\partial C}{\partial \overline{D}_k}\right)^2 Var[D_k] \quad (10.24)$$

where $\overline{C}$ is the capacity when the delay bounds are at their mean values. Using this approach we can find the sensitivity indices and partial variances for each traffic handling scheme. Note that in the equations that follow all partial derivatives are evaluated with reference to the mean values of the delay bounds. For WFQ we have:

$$\frac{\partial C^{WFQ}}{\partial D_k} = \frac{-N_k(\sigma_k + L_k)}{\overline{D}_k^2} \quad (10.25)$$

$$\frac{\partial^2 C^{WFQ}}{\partial D_k^2} = \frac{2 * N_k(\sigma_k + L_k)}{\overline{D}_k^3} \quad (10.26)$$

Thus the expected value and variance of the capacity are:

$$E\{C^{WFQ}\} = \sum_k \frac{N_k(\sigma_k + L_k)}{\overline{D}_k} + \sum_k \frac{N_k(\sigma_k + L_k)}{\overline{D}_k^3}Var[D_k] \quad (10.27)$$

$$Var[C^{WFQ}] = \sum_k \left(\frac{N_k(\sigma_k + L_k)}{\overline{D}_k^2}\right)^2 Var[D_k] \quad (10.28)$$

For CBQ, since we are dealing with p classes, we compute sensitivity indices and partial variances for each class as follows:

$$\frac{\partial C^{CBQ}}{\partial D_{class\ p}} = -\left(\frac{\sum_{k\in class\ p} N_k\sigma_k}{\overline{D}_{class\ p}^2} + \frac{L_p}{\overline{D}_{class\ p}^2}\right) \quad (10.29)$$

$$\frac{\partial^2 C^{CBQ}}{\partial D^2_{class\ p}} = 2 * \left(\frac{\sum_{k \in class\ p} N_k \sigma_k}{\overline{D}^3_{class\ p}} + \frac{L_p}{\overline{D}^3_{class\ p}} \right) \quad (10.30)$$

where $\overline{D}_{class\ p}$ is the mean delay for class p. Thus the expected value and variance of the capacity are:

$$\begin{aligned} E\{C^{CBQ}\} &= \sum_p \left(\frac{\sum_{k \in class\ p} N_k \sigma_k}{\overline{D}_{class\ p}} + \frac{L_p}{\overline{D}_{class\ p}} \right) \quad (10.31) \\ &\quad + \sum_p \left(\frac{\sum_{k \in class\ p} N_k \sigma_k}{\overline{D}^3_{class\ p}} + \frac{L_p}{\overline{D}^3_{class\ p}} \right) Var[D_{class\ p}] \\ Var[C^{CBQ}] &= \sum_p \left(\frac{\sum_{k \in class\ p} N_k \sigma_k}{\overline{D}^2_{class\ p}} + \frac{L_p}{\overline{D}^2_{class\ p}} \right)^2 Var[D_{class\ p}] \quad (10.32) \end{aligned}$$

For PQ, we also calculate the sensitivity parameters on a per-class (priority-level) basis to yield:

$$\begin{aligned} \frac{\partial C^{PQ}}{\partial D_{class\ p}} &= -\sum_{j=1}^{p} \left(\frac{\sum_{k \in class\ j} N_k \sigma_k}{\overline{D}^2_{class\ p}} + \frac{L_{max(p)}}{\overline{D}^2_{class\ p}} \right) \quad (10.33) \\ \frac{\partial^2 C^{PQ}}{\partial D^2_{class\ p}} &= 2 * \sum_{j=1}^{p} \left(\frac{\sum_{k \in class\ j} N_k \sigma_k}{\overline{D}^3_{class\ p}} + \frac{L_{max(p)}}{\overline{D}^3_{class\ p}} \right) \quad (10.34) \end{aligned}$$

Note that the values in Equations 10.33 and 10.34 apply when the maximizing class p in Equation 6.5 is unique over all possible delay values. Thus the expected value and variance of the capacity are:

$$\begin{aligned} E\{C^{PQ}\} &= \max_p \left\{ \sum_{j=1}^{p} \left(\frac{\sum_{k\ \in\ class\ j} N_k \sigma_k}{\overline{D}_{class\ p}} + \frac{L_{max(p)}}{\overline{D}_{class\ p}} \right) \right. \quad (10.35) \\ &\quad \left. + \sum_{j=1}^{p-1} \sum_{k\ \in\ class\ j} N_k \rho_k + \sum_{j=1}^{p} \left(\frac{\sum_{k \in class\ j} N_k \sigma_k}{\overline{D}^3_{class\ p}} + \frac{L_{max(p)}}{\overline{D}^3_{class\ p}} \right) Var[D_{class\ p}] \right\} \end{aligned}$$

$$Var[C^{PQ}] = \sum_p \left(\sum_{j=1}^{p} \frac{\sum_{k \in class\ j} N_k \sigma_k}{\overline{D}^2_{class\ p}} + \frac{L_{max(p)}}{\overline{D}^2_{class\ p}} \right)^2 Var[D_{class\ p}] \quad (10.36)$$

The case where the maximizing value of p is not the same over all delay values requires a more detailed examination of the distribution of the delays to determine the range of values for which each class or priority-level influences the capacity in Equation 6.5. In this case the sensitivity coefficients are evaluated as before but using delay values within the range of influence of each class. Example 1 of Section 10.3 illustrates this.

For FIFO, there is essentially one sensitivity index since we use the minimum delay bound over all traffic types. Thus,

$$\frac{\partial C^{FIFO}}{\partial \overline{D_{min}}} = -\frac{\sum_k N_k \sigma_k}{\overline{D}^2_{min}} \tag{10.37}$$

$$\frac{\partial^2 C^{FIFO}}{\partial \overline{D_{min}}^2} = \frac{2 * \sum_k N_k \sigma_k}{\overline{D}^3_{min}} \tag{10.38}$$

$$E\{C^{FIFO}\} = \frac{\sum_k N_k \sigma_k}{\overline{\overline{D}_{min}}} + \left(\frac{\sum_k N_k \sigma_k}{\overline{D}^3_{min}}\right) Var[D_{min}] \tag{10.39}$$

$$Var[C^{FIFO}] = \left(\frac{\sum_k N_k \sigma_k}{\overline{D}^2_{min}}\right)^2 Var[D_{min}] \tag{10.40}$$

The equations obtained for estimating the mean and variance of capacity for each scheme as well as the sensitivity indices are straightforward to apply for WFQ. However, for the other three schemes, the applicability of the equations will depend on the ranges and inter-relations between the delay bounds. This is because for CBQ and PQ we need to know the minimum delay bound within an aggregation level (class or priority level) while for FIFO we need to know the minimum delay bound over all types. When dealing with delay bounds that have distribution functions, the determination of the minimum bound may not be straightforward. The simplest case to deal with is that in which the ranges of the delay bounds are non-overlapping so that the distribution of the minimum is easily identified. For more general cases however, an analytic solution would require the use of order statistics [1, 26] to determine the distribution of the minimum. This may prove to be complex when the number of distributions is more than 2. For this work we use Monte Carlo simulations to estimate the sensitivity indices for such cases and leave the analysis using order statistics for future work.

10.3 Numerical Results on Sensitivity Analysis

We consider several cases to illustrate how the sensitivity of capacity to delay can be evaluated. The first case applies the analysis to the simple case of a single flow from each traffic type with non-overlapping uniformly distributed delay bounds. The second case extends this by increasing the number of flows while maintaining the same delay statistics. The next case considers slight overlap in the delay bounds of voice and video and in the delay bounds of e-mail and WWW by increasing the variance of video and WWW traffic compared to the first case. In the last case, we consider the case where we know with certainty what the maximum delay bound is for each class but where the minimum is not known precisely so that all four types have the same minimum bound. Thus there is overlap between all four types and the variances of video, e-mail and WWW are increased greatly compared to the first case. We present and discuss the results obtained in the sections that follow.

10.3.1 Case 1: Single Flow, Small Variance in Delay

The parameters used for the first case are shown in Table 10.1:

Type	$D_{min}(ms)$	$D_{max}(ms)$	$D_{mean}(ms)$	$Var[D](ms)^2$
Voice	1	2	1.5	0.083
Video	3	5	4.0	0.333
Email	50	100	75	208.33
WWW	100	200	150	833.33

Table 10.1: Delay Bound Statistics for Case 1: Small Delay Variance

The sensitivity coefficients, capacity and variance for this case are shown in Table 10.2[2]:

For WFQ we observe that the greatest sensitivity is due to the video flow while for CBQ and PQ[3] the greatest sensitivity is due to the voice flow. However, the large sensitivity in this case is due to the sharing of the real-time(RT) queue between the voice and video traffic. In fact, assuming that

[2]For CBQ, the capacity is influenced only by voice and email delay values since these are the minima in their respective classes. For FIFO, the capacity is only influenced by the voice

[3]PQ results are comparable to CBQ results

	Parameter	Voice	Video	Email	WWW
WFQ	Sensitivity (Mps/msec)	0.455	4.25	0.00509	0.01509
	$\sqrt{Variance}$(Mbps)	0.13	2.45	0.07	0.43
	Capacity (Mbps)	20.35	20.67	20.33	20.4
CBQ	Sensitivity (Mps/msec)	34	—	0.0647	—
	$\sqrt{Variance}$(Mbps)	9.8	—	0.933	—
	Capacity (Mbps)	52.8	—	56.04	—
FIFO	Sensitivity (Mps/msec)	185.2	—	—	—
	$\sqrt{Variance}$(Mbps)	53.35	—	—	—
	Capacity (Mbps)	277.8	—	—	—

Table 10.2: Analytic Results for Case 1: Single Flow per Traffic Type

there was no video traffic, the sensitivity index would only be 5.56Mbps/msec compared to the 34Mbps/msec with the presence of the video flow. Thus the high sensitivity is also attributable to the video flow. For the non-real-time(NRT) queue, the sensitivity is due to the email delay but as with the RT queue, it is actually the presence of the WWW traffic that influences the sensitivity index more than the email traffic does. For FIFO, the sensitivity is due only to the voice traffic since it has the minimum distribution over all the delay distributions. The largest contributors to the sensitivity index however are email and WWW traffic, which account for about 85% of the sensitivity. The single flow analysis shows that there are two perspectives to understanding and using the sensitivity analysis, especially for the aggregate schemes. On the one hand there is the delay distribution that actually determines the sensitivity of the capacity and on the other there is the traffic type that actually contributes the most to the sensitivity index which depends on the burstiness of each traffic type. Since the sensitivity is linear in the number of sources we expect that for the same number of flows in each traffic type, the trend of the results in Table 10.2 would be preserved. This suggests then that in order to control the variance in capacity, the number of sources for the traffic type with the highest contribution to the sensitivity must be strictly controlled for each traffic handling scheme. For this case WFQ, CBQ and PQ require the number of video flows to be limited while for FIFO, the number of email and WWW flows must be limited.

For the remaining cases we used 50 voice flows, 2 video flows, 100 email flows and 10 WWW flows and ran Monte Carlo simulations to calculate the variance and sensitivity indices of each type. We did this by generating 10 batches of 1000 samples each of delay values for each traffic type and calculating the required capacity for each sample value while keeping the delay of the other three types at their mean values. We then calculated the

variance and sensitivity index for each batch and averaged these over all the batches.

10.3.2 Case 2: Multiple Flows, Small Variance in Delay

For this case we use the same delay statistics as in Table 10.1. The results obtained analytically and from simulation will be discussed separately for each traffic handling scheme. The results for WFQ are shown in Table 10.3:

	Parameter	**Voice**	**Video**	**Email**	**WWW**
Analytic	Sensitivity (Mbps/msec)	22.7	8.5	0.509	0.151
	Capacity (Mbps)	130.31	129.75	130.47	129.9
	$\sqrt{Variance}$(Mbps)	6.5	4.9	7.4	4.4
	% Variance	31	17	39	14
Simulation	Sensitivity (Mbps/msec)	24.5	8.89	0.55	0.163
	Capacity (Mbps)	130.38	129.82	130.39	129.8
	$\sqrt{Variance}$(Mbps)	7.1	5.15	7.97	4.7
	% Variance	31.2	16.2	38.9	13.7

Table 10.3: WFQ Analytic and Simulation Results for Case 2: Multiple Flows, small variance

The results show a good match between the analytic and simulation results. A look at scatter plots of capacity and delay for video and WWW in Figure 10.1 shows that for the delay statistics chosen, the capacity is an approximately linear function of delay hence the good correlation between the analytic and simulation results. Similar plots are obtained for voice and email. We will observe that this correlation does not hold when we use different delay statistics for the other cases.

Looking at the sensitivity indices the capacity is most sensitive to voice traffic followed by video then e-mail and WWW. We find however that e-mail contributes slightly more to the variance in capacity than voice. The variance due to video is comparable to that of WWW.

For CBQ the analytic and simulation results are shown in Table 10.4.

For the analytic results we present results in terms of the RT and NRT traffic classes while for the simulation results we are able to consider the impact of each traffic type separately. For this particular example since there is no overlap between the voice and video traffic we could replace RT

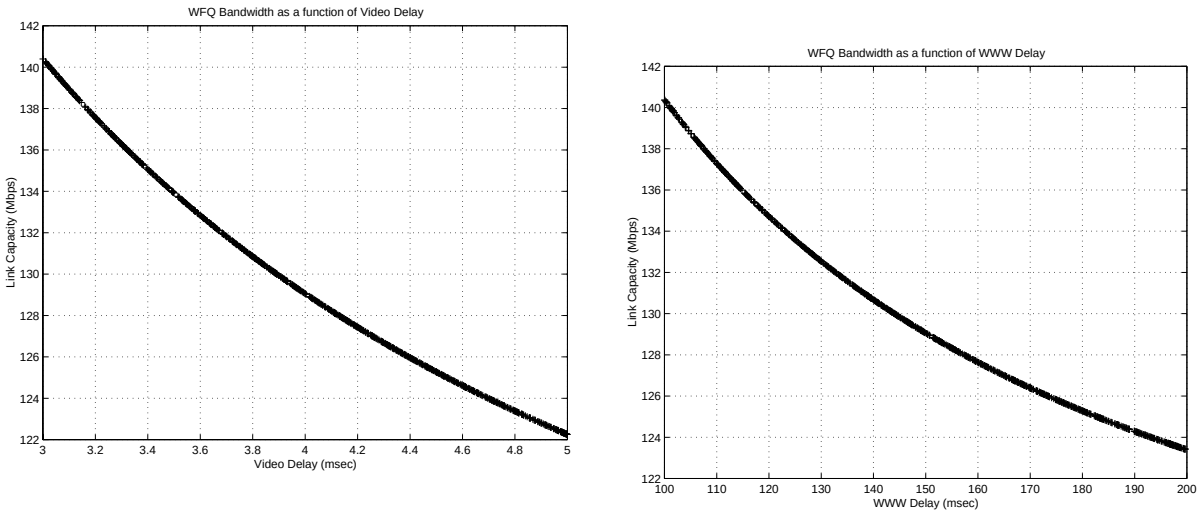

Figure 10.1: WFQ Capacity vs Delay for Video and WWW traffic for Case 2: small delay variance

	Analytic		**Simulation**			
Parameter	**RT**	**NRT**	**Voice**	**Video**	**Email**	**WWW**
Sensitivity (Mbps/msec)	73.6	1.02	79.4	—	1.09	—
Capacity (Mbps)	185.72	184.59	191.3	187	189.7	187
$\sqrt{Variance}$(Mbps)	21.2	14.7	23.0	—	16	—
% Variance	67.3	32.7	67.4	—	32.6	—

Table 10.4: CBQ Analytic and Simulation Results for Case 2:Multiple Flows, small variance

by voice and similarly we could replace NRT by E-mail. Since the behavior of the RT queue is determined solely by the voice traffic we expect in the simulation results to obtain zero contribution to the variance from the video traffic. The same is true of the NRT queue which is influenced by E-mail so that the contribution to the variance from WWW is zero. We again observe a good match between the analytic and simulation results. The analysis shows that 67.3% of the uncertainty in variance is due to the Real-Time class and since the voice delays are the minimum in this class, this is all attributable to voice traffic which the simulations clearly validate. The same is true for the NRT class where e-mail is the controlling traffic type.

For PQ we observe that when dealing with the RT queue, the results are identical to those of CBQ but for the NRT traffic we are faced with the situation in which the capacity is in some cases influenced by the RT traffic and in others by the NRT traffic. Equating the capacity values due to each class in Equation 6.5 we find that for delay values of NRT traffic in the range 50-56.62ms, the capacity is determined by the NRT class whereas for all other values, the delay is determined by the RT class. The scatter plot in Figure 10.2 taken from the simulations confirms this.

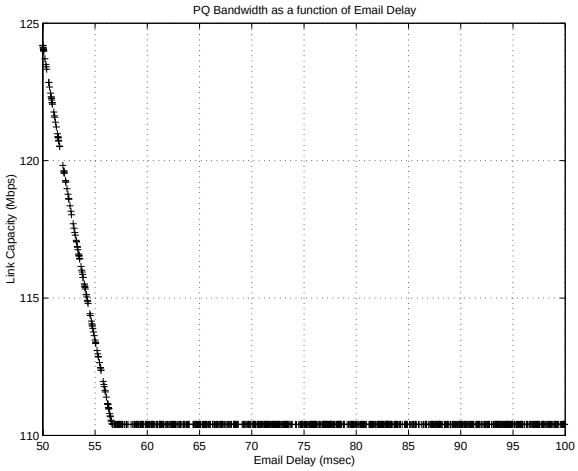

Figure 10.2: PQ Capacity vs Email Delay for Case 2: small delay variance

We see from Figure 10.2 that for delays less than about 57ms, the capacity is a linear function of the email delay whereas for values above this, the capacity does not depend on the email delay. Proceeding with the analysis we obtain the results in Table 10.5.

The results show that voice is the most significant factor influencing the uncertainty in the variance as well as the sensitivity of the capacity. As expected video and WWW do not contribute to the variance in capacity since

	Analytic		Simulation			
Parameter	**RT**	**NRT**	**Voice**	**Video**	**Email**	**WWW**
Sensitivity (Mbps/msec)	73.6	0.209	78.9	—	0.117	—
Capacity (Mbps)	114	114	114.73	110.4	111.24	110.4
$\sqrt{Variance}$(Mbps)	21.2	3.01	23.02	—	2.6	—
% Variance	98.02	1.98	98.7	—	1.3	—

Table 10.5: PQ Analytic Results for Case 2: Multiple Flows, small variance

voice and e-mail have the minimum delay values in each class respectively. The analytic and simulation results for FIFO are shown in Table 10.6.

	Analytic	Simulation			
Parameter	**Voice**	**Voice**	**Video**	**Email**	**WWW**
Sensitivity (Mbps/msec)	2600	2820.2	—	—	—
Capacity (Mbps)	4.07e3	4.077e3	3.92e3	3.92e3	3.92e3
$\sqrt{Variance}$(Mbps)	753.6	821	—	—	—
% Variance	100	100	—	—	—

Table 10.6: FIFO Analytic and Simulation Results for Case 2: Multiple Flows, small variance

Again as expected, both the analysis and simulation show that voice is the only significant factor since all the minimum delay values are due to the voice traffic.

Looking at all four schemes we find that for this case voice delay contributes the most to the sensitivity of the capacity and is also a significant contributor to the variance in capacity. The first two cases have demonstrated that considerable insight into the sensitivity of capacity to delay and uncertainty in the delay variance can be obtained by applying the methodology of Section 10.2. We will consider two more cases that show results when there is more interaction between the traffic classes in terms of their effects on the capacity as well as how the analysis performs in the face of nonlinearity.

10.3.3 Case 3: Increased Video and WWW Variance

In this case, we increased the variance of video and WWW traffic by changing their minimum delay values to coincide with those of voice and email respectively. The parameters are shown in Table 10.7.

Type	$D_{min}(ms)$	$D_{max}(ms)$	$D_{mean}(ms)$	$Var[D](ms)^2$
Voice	1	2	1.5	0.083
Video	1	5	3	1.33
Email	50	100	75	208.33
WWW	50	200	125	1875

Table 10.7: Delay Bound Statistics for Case 3: increased Video and WWW variance

In this case since we have overlap in the delays of voice and video on the one hand and Email and WWW on the other, the analytic results are only directly applicable to WFQ. For the other schemes, applying the analysis would require the use of order statistics to determine the distribution of the minimum. We will thus present analytic and simulation results for WFQ and simulation results only for the other schemes. Table 10.8 shows the WFQ results.

	Parameter	**Voice**	**Video**	**Email**	**WWW**
Analytic	Sensitivity (Mbps/msec)	22.75	15.13	0.509	0.217
	Capacity (Mbps)	146.19	151.64	146.35	148.2
	$\sqrt{Variance}$(Mbps)	6.5	17.5	7.4	9.4
	% Variance	9	62	11	18
Simulation	Sensitivity (Mbps/msec)	24.6	22	0.55	0.288
	Capacity (Mbps)	146.3	154.3	146.3	149.08
	$\sqrt{Variance}$(Mbps)	7.2	26.4	7.98	12.8
	% Variance	5.2	71.5	6.5	16.8

Table 10.8: WFQ Analytic Results for Case 3: increased Video and WWW variance

Comparing the analytic and simulation results we see that the general trend of the results is the same although the match is not as close for video and WWW traffic as it was for the simpler previous case. This is due to the increased nonlinearity in the capacity-delay function as shown in Figure 10.3. This points to a weakness in the analysis which assumes linearity in the capacity-delay function. One way of improving the accuracy would be to estimate the sensitivity indices better by doing piece-wise linear estimations. Note that this is implicit in the Monte Carlo simulations.

From the simulation results, the sensitivity of the capacity is still highest for voice although both video and WWW show an increase in sensitivity compared to case 2. We note also that the uncertainty in variance is now affected most by the video traffic followed by the WWW traffic. This is

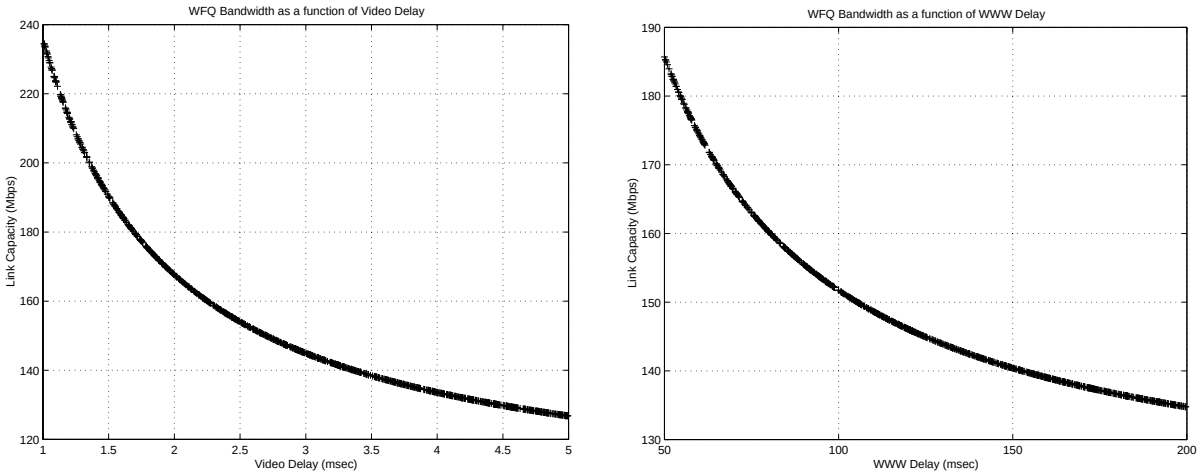

Figure 10.3: WFQ Capacity vs Delay for Video and WWW traffic for Case 3: increased Video and WWW variance

due to the increase in the delay variance of these two types. The voice and email traffic contribute almost equally to the uncertainty in variance. The simulation results for CBQ are shown in Table 10.9.

Parameter	**Voice**	**Video**	**Email**	**WWW**
Sensitivity (Mbps/msec)	79.6	5.36	1.09	0.119
Capacity (Mbps)	191.45	189.95	189.87	189.72
$\sqrt{Variance}$(Mbps)	23.2	9.6	16	7.5
% Variance	57.2	9.7	27.2	5.9

Table 10.9: CBQ Simulation Results for Case 3: increased Video and WWW variance

The sensitivity of the capacity is still highest for voice but we now have sensitivity due to the video and WWW traffic compared to case 2 where there was none. Video and WWW also contribute to the variance in capacity but voice is still the most significant source of uncertainty followed by e-mail. Note also that the net contribution due to the RT class (57.2 + 9.7) is roughly the same as the net contribution due to the RT class in case 2. The simulation results for PQ are shown in Table 10.10.

The voice traffic contributes the most to the uncertainty in variance followed by video. The email and WWW contribution is not very significant. We again note that the net contribution of the RT class to the variance is the same as in case 2. The analytic and simulation results for FIFO are shown in Table 10.11.

Parameter	Voice	Video	Email	WWW
Sensitivity (Mbps/msec)	79.2	5.36	0.119	0.0161
Capacity (Mbps)	114.89	113.33	111.24	110.67
$\sqrt{Variance}$(Mbps)	23.2	9.6	2.6	1.5
% Variance	84.2	14.4	1.0	0.4

Table 10.10: PQ Simulation Results for Case 3: increased Video and WWW variance

Parameter	Voice	Video	Email	WWW
Sensitivity (Mbps/msec)	2830	191	—	—
Capacity (Mbps)	4.08e3	4.03e3	3.92e3	3.92e3
$\sqrt{Variance}$(Mbps)	825.8	339.1	—	—
% Variance	86.4	13.6	—	—

Table 10.11: FIFO Simulation Results for Case 3: increased Video and WWW variance

Voice and video are now the most significant factors contributing to the uncertainty in capacity variance with voice clearly dominating. Email and WWW delays have no impact on the variance because the minimum delay values come from either the voice or video distributions.

10.3.4 Case 4: Increased Email and WWW Variance

For the last case we consider the case where the uncertainty in the Email and WWW delays is greatly increased over that of the previous case. The delay bound statistics are shown in Table 10.12.

Type	$D_{min}(ms)$	$D_{max}(ms)$	$D_{mean}(ms)$	$Var[D](ms)^2$
Voice	1	2	1.5	0.083
Video	1	5	3.0	1.33
Email	1	100	50.5	816.75
WWW	1	200	100.5	3300.08

Table 10.12: Delay Bound Statistics for Case 4: increased Email and WWW variance

As with the previous case where video and WWW had increased variance, analytic results can only be provided for WFQ. Table 10.13 shows the results for WFQ.

We observe that the analytic and simulation results show the same trend in

	Parameter	Voice	Video	Email	WWW
Analytic	Sensitivity (Mbps/msec)	22.75	15.13	1.12	0.336
	% Variance	2.5	17.4	58.8	21.3
Simulation	Sensitivity (Mbps/msec)	24.5	21.7	6.06	0.256
	$\sqrt{Variance}$(Mbps)	7.18	26.4	248	238.8
	% Variance	0.04	0.6	52	48

Table 10.13: WFQ Analytic and Simulation Results for Case 4: increased Email and WWW variance

the ranking of the contribution of each traffic type to the variance although there is marked disparity in the exact numbers for video and WWW traffic. With the current delay statistics the capacity-delay functions for email and WWW are extremely nonlinear as shown in Figure 10.4. The current

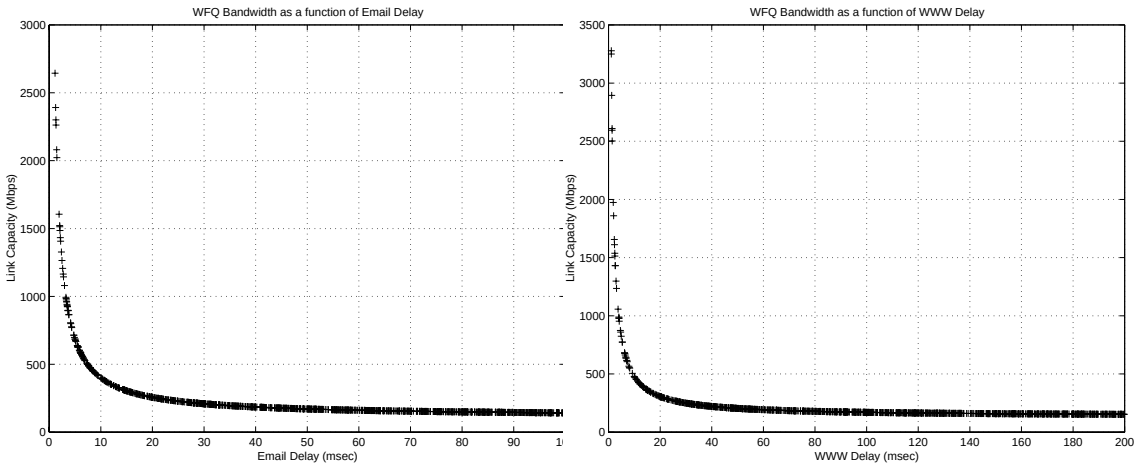

Figure 10.4: WFQ Capacity vs Email and WWW Delay for Case 4: increased Email and WWW variance

analysis is not able to capture the effects of this nonlinearity hence the predicted sensitivity coefficients for Email and WWW are not accurate. This inaccuracy in turn affects the variance results. However we can observe from the simulation that Email is now the most significant factor, contributing to about 50% of the variance, followed by WWW traffic. For CBQ, Table 10.14 shows that email and WWW are the most significant factors contributing to the variance in capacity, together accounting for almost all of the variance in capacity.

In terms of the sensitivity however, voice is still the highest followed by email then video and WWW. The PQ results are shown in Table 10.15. We observe the same trends as for CBQ in the variance contribution of each traffic

Parameter	Voice	Video	Email	WWW
Sensitivity (Mbps/msec)	79.3	5.13	12	3.75
$\sqrt{Variance}(Mbps)$	23.2	9.47	497.94	392.25
% Variance	0.13	0.02	61.5	38.35

Table 10.14: CBQ Simulation Results for Case 4: increased Email and WWW variance

Parameter	Voice	Video	Email	WWW
Sensitivity (Mbps/msec)	32.7	3.04	12	3.85
$\sqrt{Variance}$(Mbps)	11.6	6.32	506.85	402.74
% Variance	0.03	—	61	38.97

Table 10.15: PQ Simulation Results for Case 4: increased Email and WWW variance

type although the sensitivity indices are different. Voice has the highest sensitivity index followed by email, then WWW and Video. For FIFO, voice and video are the most significant factors both in terms of sensitivity and variance in capacity as shown in Table 10.16. This is because only a small

Parameter	Voice	Video	Email	WWW
Sensitivity (Mbps/msec)	2820	182	0.679	0.184
$\sqrt{Variance}$(Mbps)	825.8	336.98	64.17	68.65
% Variance	84.8	14.1	0.5	0.6

Table 10.16: FIFO Simulation Results for Case 4: increased Email and WWW variance

proportion of samples from the email and WWW distributions (0.5% and 0.25% respectively) are small enough to be the minimum value that determine the capacity.

The four cases in this section have validated the methodology and analysis that can be used to perform uncertainty and sensitivity analysis for the four traffic handling schemes. Numerical simulations can be used to perform the sensitivity analysis while the theoretical analysis is valid only when the capacity-delay function is linear. More work is needed to extend and generalize the theoretical formulation to cover all cases.

10.4 Summary

In this chapter we have shown how a stochastic formulation of the delay requirements can be used to provide some understanding of how the capacity required by different traffic handling schemes is affected by uncertainty in the values of the maximum delay bounds. We have developed and demonstrated a methodology for performing uncertainty and sensitivity analysis which can be used to capture how uncertainty in delay bounds translates into uncertainty in the network capacity. A purely analytic solution has been obtained with the accuracy being largely dependent on the linearity of the capacity-delay function. For cases where the functions are not linear, numerical Monte Carlo simulations were used to provide accurate results. In the next chapter we conclude by discussing the relevancy of our results in the context of network planning and design.

Chapter 11

Conclusion

11.1 Implications of Results on Network Architectures

We conclude by discussing the results obtained in the context of current and proposed approaches to the use of traffic handling for multi-service networks.

- Best-Effort Network

  The best-effort network uses FIFO in both the edge and core so that the delay guarantees are uniform across all traffic types. From our results we find that to support the QoS of delay-sensitive applications such as voice requires abundant network capacity when the voice traffic shares a queue with bursty traffic such as email and WWW. When there is no bursty traffic then an all-FIFO network performs just as well as a non-FIFO network. This may seem to suggest the use of separate queues and links (in essence a separate network) for delay sensitive traffic to isolate it from the bursty non-delay sensitive traffic. The appeal of the best-effort network lies in its simplicity and if network capacity is not a constraint, then it may still be the network of choice for some.

- Class-Based Network

  In a class-based network, flows are grouped into distinct classes and resources such as queues and link bandwidth are allocated to the class as whole. Traffic handling approaches for this type of network include Class-Based Queueing, Priority Queueing and Class-Based Weighted Fair Queueing among others. Our results show that using class-based

traffic handling by grouping traffic types with similar delay objectives can meet the delay guarantees of all traffic types with minimal bandwidth. Class-based handling in the edge with best-effort handling in the core requires moderate capacity while using best-effort in the edge increases the bandwidth significantly. The complexity of class-based handling may range from fair to extreme depending on the exact implementation.

- Flow-Based Network

  In a flow-based network, each flow is allocated its own dedicated resources and as such managing the network may prove to be complex when there are numerous flows. Using flow based handling such as WFQ requires minimal capacity and any combination of flow-based handling and class-based handling does not increase the bandwidth significantly. Using flow-based handling with best-effort handling in either the edge or core of the network requires more abundant capacity.

Figure 11.1 summarizes the capacity requirements of combinations of edge and core traffic handling mechanisms.

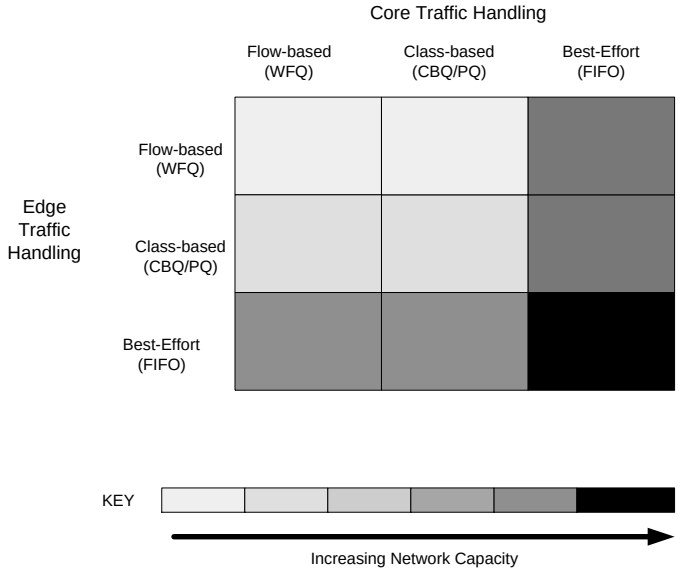

Figure 11.1: Capacity Requirements of Edge and Core Traffic Handling Mechanisms

The diagram uses different shades of gray to show how the capacity requirements change depending on the type of traffic handling used in the edge and core portions of the network. For example, we see that with flow-based

handling in the edge, class-based handling in the core requires comparable capacity to flow-based handling in the core. Given that one objective is to simplify network management, the use of flow-based handling in the core portion of the network may not be practical and the choice between which combination of traffic handling mechanisms to use will depend in part on the availability of bandwidth in the edge and core portions of the network. With moderate amounts of bandwidth in the core, then either class-based or flow-based handling can be used in the edge with best-effort handling in the core portion of the network. If some complexity in the core can be tolerated then with minimal capacity, flow-based or class-based handling can be used in the edge with class-based handling in the core.

11.2 Practical Applications of Sensitivity Analysis

The methodology in Section 10.3 uses a one-at-a-time (OAT) approach by considering how each delay distribution affects the capacity individually. For practical purposes, one would like to use the analysis for the more general case where any number of traffic types may have stochastic delay bounds. Careful examination of the equations for the variance of capacity in Section 10.2 shows that the total variance is simply the sum of the variance due to each individual delay distribution. We can thus use the analysis to upper-bound the standard deviation in capacity for different distributions of the delay parameters. Specifically, suppose the total variance in capacity is $Var[C] = Var[C_1] + Var[C_2] + ... + Var[C_k]$ where $Var[C_k]$ is the variance in capacity due to the k^{th} delay distribution. Then the standard deviation is given by:

$$\begin{aligned} SD(C) &= \sqrt{Var[C_1] + Var[C_2] + ... + Var[C_k]} \\ &< \sqrt{Var[C_1] + Var[C_2] + ... + Var[C_k] + \sum_{i \neq j} SD(i)SD(j)} \\ &< \sqrt{(SD(1) + SD(2) + ...SD(k))^2} \\ &< SD(1) + SD(2) + ...SD(k) \end{aligned}$$

where $SD(i) = \sqrt{Var[C_k]}$. Thus changes in one or more delay parameters can easily be used to assess the effect on the total capacity.

The methodology used for the single-link analysis extends easily to edge-core networks. The OAT approach can be applied to different edge-core architectures and sensitivity indices calculated for each link in the network. From this information the average sensitivity in the edge and core portions as well as variance in the edge and core capacity can be easily calculated. With this in mind we turn now to some ways in which sensitivity analysis can be used in network design and provisioning.

- Planning and Forecasting

  Sensitivity analysis can be used to determine how changes in traffic patterns will impact the capacity requirements of a network. The sensitivity indices can be used to determine the expected capacity and standard deviation of capacity for different traffic conditions which can then be used to predict at what point in time the network capacity will need to be upgraded to support traffic growth. It should be noted that such planning has also been discussed in the context of deterministic delay bounds but the value added by a stochastic formulation is the incorporation of how uncertainty in the knowledge of the delay bounds will impact the capacity requirements. Using the sensitivity analysis allows the network planner to consider how the addition of new services whose delay expectations are not known precisely may impact the capacity requirements.

  Another way to use the sensitivity analysis is to determine how changing trends in delay requirements of different services will impact network capacity requirements. As telecommunication networks and the Internet evolve, different applications will emerge with new delay requirements. There is thus a need for a methodology that can be used to assess how such changes will affect the network design and sensitivity analysis can be used for this.

- Service Provisioning

  By using the sensitivity analysis to determine how imperfect knowledge of maximum delay requirements affects network capacity, a network engineer will be in a better position to define the types of services that the network can offer within the limits of its existing capacity framework.

- Network Management

  Sensitivity analysis can also be used to guide network users on suitable traffic parameters for the type of service they desire. For instance if a user is aware of the rate and burstiness parameters, then the analysis

can be used to define an appropriate range of delay objectives for a given capacity or can be used to determine the capacity requirements needed to satisfy specific delay objectives.

11.3 Summary of Contributions

There are four main contributions made by this thesis:

- The most significant contribution is in developing a methodology that can be used to quantify and compare the capacity requirements of different traffic handling approaches. There has always been a general consensus that larger amounts of capacity are required by FIFO traffic handling compared to per-flow handling without specifics on how to quantify the difference. There have also been notions about the capacity requirements of class-based handling being intermediate between per-flow and FIFO but also with no quantification. Through the work we have done we have established an analytic methodology that has quantified the differences in capacity and that can be used to address a variety of questions relating to design of integrated services networks.

- We began this thesis with the objective of answering the question of how different traffic management approaches compare in terms of their capacity requirements. We have shown that class-based schemes do not differ significantly from flow-based schemes in their capacity requirements which was one of the issues surrounding the Integrated Services vs Differentiated services debate of the last couple of years. We have also shown that there are a number of issues that must be factored in when comparing traffic handling schemes such as the aggregate burstiness, the network size and the connectivity of the network.

- The work on sensitivity analysis provides a first-step in the development of procedures for long-term network planning. By incorporating other parameters such as burstiness, topology and traffic composition into the sensitivity analysis valuable insight can be obtained for use in network design and planning.

- Lastly, we have extended the application of Network Calculus in addressing a significant networking problem.

11.4 Future Work

We have identified several areas for future work:

- In this work we used a deterministic model to characterize the application traffic. This assume worst-case behavior on the part of the sources generating the traffic and is thus a conservative approach. An extension to this work would thus look at the use of stochastically bounded traffic models Ṡome work has been done with stochastically bounded models such as the work in [14, 51, 66, 79, 92].

- Another area for future work is in obtaining better bounds on utilization for networks that use aggregate traffic handling. We reported in Chapter 9 on the work in [17] which provides very pessimistic results on allowable utilization. One issue not considered here is a detailed study on utilization and network topology to determine how our results compared to the bounds in [17].

- In Chapter 8 we developed some analytic results for bounds on capacity requirements in edge-core networks but we did not test the accuracy of the bounds exhaustively. Future work would look at how good these bounds are by conducting analyses on networks of varying topology as was done here. This would provide results and insight into effective uses of the bounds.

- There are several items related to the sensitivity analysis that are left open for future work. The first is obtaining an analytic solution using order statistics to deal with the general cases in which the minimum delay distribution is not easily identified. In view of the fact that the complexity of using order statistics may not justify the effort, Monte Carlo simulations as used here, may still prove to be the best solution. The second item has to do with the use of global sensitivity analysis such as the importance measures described in Section 10.1. A complete uncertainty and sensitivity analysis should incorporate elements of both local and global analysis and we emphasized local analysis in this work. It would complete the work to consider further insight that may be obtained from global analysis.

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