Traffic Control and Architecture for High-Quality and High-Speed Internet

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Preface

Several tens of bit per second: this mere value was the maximum speed of the Internet when it was born as a result of a DARPA project. As the Internet is widely deployed, the speed of access networks is improved thanks to ADSL and FTTH technologies, and the capacity of core network routers now reaches several tens of Gbps, today's Internet has become a part of the daily life of residential users as well as business users, who use e-mails and Web for their business and entertainment. This means that the Internet has already become a life infrastructure.

Compared with the improvement of speed and capacity, the changes in quality are much smaller. The Internet is operated on "best effort" basis. This comes from the design principle of IP, the connectivity is the most important attribute of the Internet. This contributed to the rapid growth of the Internet, but it also results in no quality of service (QOS) guarantee especially for multimedia traffic. Toward the future Internet as an actual infrastructure, QOS consideration is required. We should provide traffic control mechanisms for achieving high QOS and network architecture based on the in-depth consideration to the traffic control while operational cost and equipment cost remains minimum.

This thesis discusses on traffic control and architecture for high-quality and high-speed Internet. The discussion focuses on five subjects; admission control methods for (1) connections and (2) a partial data of connections, i.e. a burst, (3) application aware communication control, (4) network architecture for application level QOS and (5) technologies for realizing high speed routers.

Because Asynchronous Transfer Mode network or an ATM network has been expected to integrate both circuit switch and packet switch, it is necessary to guarantee QOS for accommodating multimedia traffic, i.e. circuit switch friendly traffic such as telephone or movie streams. As ATM networks are based on connection-oriented principle, it is suitable for controlling QOS by controlling network traffic load.

Call Admission Control or Connection Admission Control (CAC) is, thus, firstly addressed. CAC enables networks to achieve required QOS values in the ATM network while keeping network utilization at its maximum. In the CAC, CAC performance depends on a cell loss estimation and a CAC procedure. For that, a classical M/M/1 model is no longer fit to ATM traffic since ATM has to accommodate itself to various traffic characteristics generated from variable bit rate streaming, ftp data transfer, Web data retrieval and computer communication such as LAN interconnection. The CAC proposed in this thesis has good advantages; only two parameters and small time are

needed for accept/reject new connection request. The proposed CAC uses “virtual cell loss rate” instead of cell loss rate as cell loss estimation. Simulation results show that the proposed CAC has appropriate accuracy in estimation and offers appropriate network utilization.

More bursty traffic, however, does not fit to the connection-by-connection CAC. Burst transfer methods have been developed for that purpose. It reserves and releases bandwidth for each piece of burst data at the time when the burst is ready to send. This, however, may cause large latency and waist of bandwidth during the reservation if the bandwidth is not available at least in one link. The more the number of links are on the end-to-end path, the larger the expected delay to grab the all bandwidth on the path simultaneously. To make the latency small, burst server architecture is proposed. In the architecture, a burst server, which stores and forwards the bursts, is placed between links and the reservation methods are modified. Bandwidth reservations can succeed either if the bandwidth on all links between sending and receiving terminals are available, or all links between burstservers/sending/receiving terminals. Numerical results show that the latency is much improved and utilization of networks is also improved two times larger than conventional architecture at maximum.

These contribute packet level QOS improvement. On the other hand, for considering user level QOS, not only packet but throughput and contents retrieval time also must be improved.

One of the today’s most important and popular applications of the Internet is web (World Wide Web). “8-second rule” reveals users have very little tolerate against waiting time of contents displayed. The waiting time consists of combination of network delay and server delay. It is necessary to cooperate networks and servers to reduce the waiting time. The web has a feature that the content that is to be retrieved next can be predicted from the current contents. This feature implies pre-fetch retrieval may be able to hide transfer delay (latency) to users/browsers. However, there is a tradeoff between the latency and the traffic load due to pre-fetching as many contents as potentially expected to be used. If the pre-fetch is done when network is not congested, i.e. is in idle, network can be well utilized and latency can be small as well. The network cache architecture for network friendly pre-fetch retrieval is therefore proposed. Some numerical results show that the network cache improves 40% latency without causing congestion.

To meet the high throughput requirements such as from storage networking and high speed wide-area LAN interconnection, TCP must be improved. TCP has been developed since 1980s, and has never essentially improved by today. TCP has to

provide new functions to meet their requirements for today's high-speed link, fairness consideration, and future services. Although many TCP modifications has been proposed, it is difficult to introduce new TCP to end hosts/servers, but easy to the intermediate node. TCP relay node (TCP Bridge) is proposed to be set in a network. It can thus give new functions such as reliability and high throughput applications without changing existing IP network and user terminals/servers. TCP overlay network then is proposed. In the TCP overlay network TCP Bridges are cooperated each other and change TCP characteristics to appropriate one for the links between TCP Bridges. Since TCP Bridge has a little experience, there are many issues to be solved. One of these is a congestion control issue. If a trivial congestion occurs on sending side of a TCP Bridge, it causes serious rather than non-trivial congestion on receiving side of the TCP Bridge. To prevent this problem, buffer control in TCP Bridge is proposed and discussed. Simulation results shows that proposed control method improve the problem and can achieve throughput two times higher than the case without any control.

As well above mentioned traffic control, network itself must be improved in its speed to accommodate today's huge traffic. One of essential bottleneck in developing high-speed router is IP address table search. Longest Prefix Matching (LPM) search must be used to search the table. Although a lot of quick search methods are developed in full matching search, they cannot apply to LPM search. Algorithmic approach is very cost effective but has lower speed. One of the solutions for quick search is to develop hardware search engine, i.e. Ternary CAM (T-CAM). T-CAM is tri-state Contents Addressable Memory. However, it causes high cost and small capacity, and to accommodate large IP table, many T-CAM chips are necessary, which increases the cost. The idea is proposed that cache architecture is employed with algorithmic search and hardware search engine. Instead of ordinary CAM as used by conventional caching architecture, T-CAM is employed to reduce cache miss-hit ratio. Because caching-in/out rule is not obvious in LPM search, the rule is carefully invented and investigated for a validation. Performance evaluation is shown to disclose the proposed architecture can achieve at least ten times smaller miss-hit ratio than the conventional cache architecture.

Today's Internet has become a part of the daily life of residential users and business users as a life infrastructure. Internet must have high quality and capability of multimedia data traffic. Toward the future Internet, this thesis discusses new architecture and control for high QOS with the networks while operational cost and equipment cost remains minimum. We believe these discussions contribute to realize next generation high quality, high-speed Internet.

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I dedicate this thesis to my parents, my wife, Kyoko, and my daughter, Yume who have continuously loved and supported to me.

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CHAPTER 1
INTRODUCTION

1.1 Toward high-quality network with new traffic control and network architecture

Several tens of bit per second: this mere value was the maximum speed of the Internet when it was born as a result of a DARPA project. As the Internet is widely deployed, the speed of access networks is improved thanks to ADSL and FTTH technologies, and the capacity of core network routers now reaches several tens of Gbps, today's Internet has become a part of the daily life of residential users as well as business users, who use e-mails and Web for their business and entertainment. This means that the Internet has already become a life infrastructure.

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This thesis discusses on several traffic control methods and new architecture based on those methods for establishing high QOS Internet. The discussion focuses on five subjects; admission control methods for (1) connections and (2) a partial data of

connections, i.e. a burst, (3) application aware communication control, (4) network architecture for application level QOS and (5) technologies for realizing high speed routers.

A transport network that people expected to become an essential part of broadband multimedia network is an Asynchronous Transfer Mode network or an ATM network. ATM has been initially developed by ITU, international standard body, and by telephone operators and careers. Next, The ATM Forum [71] has been funded for industrial de facto standard in 1991. In 1990s, although packet switching technologies could not achieve high-speed routers, a fixed length packet, which is called a cell used in ATM, is the key technology to realize high-speed packet switching. Because ATM has been expected to integrate both circuit switch and packet switch, it is necessary to guarantee QOS for accommodating multimedia traffic, i.e. circuit switch friendly traffic such as telephone or movie streams. As ATM networks are based on connection-oriented principle, it is suitable for controlling QOS by controlling network traffic load i.e. by controlling numbers of connections.

Traffic control methods can be divided into two categories, that is, reactive control and preventive control, and the most efficient use of network resources can be achieved by combining the two. Traffic load control such as call admission control or Connection Admission Control (CAC) is, of course, a form of preventive control, which is by its nature more effective than reactive control for use in high-speed networks [3].

CAC is, thus, firstly addressed. CAC enables networks to achieve required QOS values while keeping network utilization at its maximum. In the CAC, CAC performance depends on a cell loss estimation and a CAC procedure. A Classical M/M/1 model is no longer fit to ATM traffic since ATM has to accommodate itself to various traffic characteristics. The traffic is generated from not only telephone

conversation but variable bit rate streaming, ftp data transfer, Web data retrieval and computer communication such as LAN interconnection. Such traffic characteristics are categorized into two; Constant Bit Rate (CBR) and Variable Bit Rate (VBR). VBR is characterized with three parameters; peak rate, sustainable rate (average rate) and burst length. Traffic is called bursty if the ratio of peak rate divided by average rate is large. It is easy to achieve no cell loss by multiplexing traffic on peak rate basis, but this makes network utilization quite low in case of bursty traffic. If we accept very low cell loss rate, utilization can drastically be improved by using statistical multiplexing. Many queueing theories suggest that in statistical multiplexing traffic load must be kept lower to handle more bursty traffic with the same buffer overflow rate. This means a tradeoff between utilization resulted from statistically multiplexing gain and QOS, i.e. cell loss rate. Network, thus, requires accurate estimations of cell loss rate for multiplexing a new call (connection), before giving the connection an admission to setup the connection. The problem is that in real network, the estimation must be calculated in practical time. This generates another tradeoff between accuracy of the estimation and calculation time to the estimation.

The traffic of an admitted connection must be guaranteed up to the point that it declared at the admission. The traffic that exceeds the declared value is not guaranteed even though it comes from the traffic source of the admitted connection. The mechanism to distinguish a traffic data into guaranteed part and non-guaranteed part is referred to as “policing”. Because cell loss due to the policing may larger than cell loss due to network congestion, it is required for user to declare the traffic characteristic values. Cell loss rate resulted from the multiplexing may vary between connections each of which have different characteristics. To meet QOS requirements for all accommodated connections, cell loss rate estimation for each connection are required.

Various call admission control schemes have been proposed [3]-[7] these days. A cell loss measure based on "queueing model" is sensitive to burst length that is difficult to be declared and policed. More simple and robust estimation mechanism is required for practical use. Furthermore, many preceding works which fall into this category use average cell loss probability defined over the mixture of different traffic classes [4]-[8]. In most practical situations where a variety of applications share a link, it is usually difficult for an admission control method based on the average cell loss probability to reflect a specific quality requirement for individual calls. Individual cell loss probability has therefore been introduced in [9], [10]. However, it is not sufficient for practical use. Several other studies have proposed simple and fast control procedures using virtual bandwidth methods but for similar reasons they are insufficient to ensure that all individual cell loss probabilities fall within a specific degree of quality. Moreover, while each of these studies notes that virtual bandwidth methods fail to guarantee a specific quality in heterogeneous traffic multiplexing, none gives any solution to the problem [11]-[14]. In [10], the problem is referred to as "interference" between different types of bursty traffic and, though a solution is offered, [10] itself admits that the solution is still incomplete.

The objective to propose a new CAC is thus to give network a simple and practical cell loss estimation considering the traffic QOS requirements of each request. The proposed method has three features; the estimation "virtual cell loss rate" is calculated in "buffer less fluid flow model" instead of "queueing model" and simulation results show that the estimation is sufficiently accurate for practical use. The estimation and virtual bandwidth derived from the estimation are well meets for individual estimation and interference even the traffic is the mixture of heterogeneous ones.

More bursty traffic, however, does not fit to the connection-by-connection CAC. Looking at data transfer such as LAN interconnection, traffic can be so bursty that we seldom can expect meaningful statistical multiplexing gain. For such traffic burst bandwidth reservation protocol is more appropriate. The reservation protocol includes admission control for each burst. Before sending a burst, necessary bandwidth, i.e. peak rate must be reserved on all links between a source and a destination. After sending the burst, the reserved bandwidth is released immediately. It is obvious that admission control for the reservation protocol for bursts is the same with the one for CBR connections. For small bursts or short bursts, latency of the reservation is the key performance measure. Since reservation succeeds only when all links are simultaneously reserved, insufficient resources even of a link result in the failure of reservation. This failure causes another reservation attempt after a certain backoff time, which results in large delays at the source terminals and throughput degradation. This problem becomes serious as the number of links on the path increases. One study [31] shows that the multipath scheme improves the blocking probability if terminals manage many different paths to the same destination.

To solve the blocking problem based on an ATM single path, there are two approaches that provide feasible solutions. One is to reduce the bandwidth used for each burst transfer in order to get smaller burst blocking probabilities [32]. This approach, however, may result in large transfer delays at the source terminals. The other approach is to reduce the number of links that must be reserved simultaneously. A method is proposed based on this approach. In the method, networks have special servers – so-called burstservers – and use it by storing and forwarding bursts. Because only a reservation between terminals and burstservers is necessary for transfer data, a number of links reserved simultaneously is reduced.

These are the efforts taken to guarantee QOS on packet networks, and the author are sorry to say that the efforts has not achieved much success until now. Considering the situation the Internet, a network developed as best effort network, focused on QOS improvements rather than QOS guarantees. In addition to this improvement of both packet level QOS and user level QOS such as throughput and contents retrieval time must be achieved, instead of packet level QOS guarantee.

Since Web must be a today's most popular application, it is reasonable to focus Web application as to improve user level QOS. In terms of application level QOS, web QOS can be measured with the latency for the web contents shown up on a user browser. For example, it is reported in [36] as 8-second rule that users who suffer more than 8-second latency from a web cite will leave the cite. This is vital for Web-based e-commerce business or network service provider. The latency of a web site generally consists of two delays; network delay and server delay. Either one may cause above-mentioned problem. Internet technology is developing so as to solve such problems.

On the network side, some Internet providers give service level agreement (SLA) of QOS to dedicated users. However, it is not sufficient even to employ a new network QOS mechanism such as Differentiated Services. Problems lays in that finer granularity is required in SLA, monitoring methods are not established in QOS values, and operation difficulties remain in Differentiated Services. Moreover Differentiated Services are defined for a certain pair of source and destination, and need prior procedure for bandwidth reservation. It is, thus, hard to apply for Web in which users browse many different servers spreading over the world.

On the server side, reducing CPU load and treating large number of accesses simultaneously, cache and mirror technologies such as Digital Island [38], Akamai [39]

are developed and deployed. While mirroring is useful in case of predicted access bursts such as concert ticket reservation, it costs much for other purposes because of the cost of copying contents. Moreover, the benefit of mirroring arises only in the contents provider side. There is no merit for the user side.

Caching seems attractive to reduce server and network congestion because the most popular contents at the moment will be automatically and naturally distributed to the caching servers. Caching server is widely deployed both as forward cache combining with proxy servers and as reverse cache in front-end of servers. While reverse cache reduces the number of accesses to servers, it has no benefits on a network. Although forward cache is useful for very popular contents access, it requires more "hit ratio" for less popular contents. "Hit ratio" is defined as a rate of number of accesses for which cache server has a requested content over number of all request accesses. After miss-hit, i.e. access not hit on cache servers, cache servers set up a new connection to origin Web servers. Because Web contents size is not so large, say several 10K bytes in average [42], miss-hit cause extra delay due to connection setup delay as large as contents retrieval delay.

If the contents are retrieved prior to user's real retrieval the so-called pre-fetch -, the latency of which user is able to recognize can be small. In [43], [44], pre-fetch has effects on latency by 45% reduction by using client pre-fetch and by 60% reduction by using cache server pre-fetch.

This pre-fetch, however, should be used carefully to prevent network congestion and the waste of server resources. Because the pre-fetch is based on prediction of the link on the page of which users are supposed to click, the pre-fetched data may eventually not be used and causes the waste of network resources, causing congestion on network resources and/or server resources [45], [46]. If network can control the

pre-fetch access so as to control network load, it may achieve both less congestion for networks and low latency for users

Including Web application, almost all application uses TCP protocol. Although TCP provides high reliability transport to the application, TCP has a potential performance bottleneck in terms of throughput in its acknowledge based feedback mechanism. It is, thus, required to improve TCP throughput as the next QOS of user level. To meet the high throughput requirements such as from storage networking and high speed wide-area LAN interconnection, TCP must be improved. TCP has been developed since 1980s, and has never essentially improved by today. TCP has to provide new functions to meet their requirements for today's high-speed link, fairness consideration, and future services.

The speed of the networks must keep pace with the ever-increasing traffic, providing the traffic control mentioned above. Various efforts for high-speed networks contribute to the improvement of network speed in layer 1, 2 and 3 such as WDM, Ethernet, and IP networks. On the other hand, very little efforts on network side for layer 4 such as TCP protocol are being made for high-speed network because layer 4 is end-to-end protocol and a network usually does not process layer 4 protocols. In other words, TCP can be a bottleneck for high-speed data transfer. For example, TCP throughput is limited in 18 Mbps with typical implementations for 30 msec round trip time (RTT), which is a typical RTT between Tokyo and Osaka in the Internet. This means that an application can never expect large throughput even a 1 Gbps network is installed between Tokyo and Osaka. Almost of all applications use TCP and require high throughput. To improve TCP throughput, many proposal are presented. Many of them focus on the improvements of TCP rate control. Famous improvement includes TCP-Tahoe, TCP-Reno and TCP-Vegas implementation of rate control. Their design

principle is to sustain TCP-friendly nature. TCP is expected to have congestion control, to share the bandwidth in fair manner and to be robust against QOS degradation such as packet loss and packet re-ordering. Further High-Speed TCP, TCP-FAST achieved high throughput by ignoring fair-share nature. These approaches could be successful to the specified networks. For example, High-Speed TCP shows high throughput only if the connection alone exists in a link. Another example is that TCP-Vegas shows better link utilization than TCP-Reno although TCP-Vegas has poor throughput if it is mixed with TCP-Reno.

These approaches do not fit the practical situations. As mentioned before, TCP is an end-to-end protocol and is implemented in clients and servers. It is not easy for application to change TCP implementations because the application does not know whether it uses leased line network or best effort public network, or whether it is multiplexed with TCP-Reno, TCP-Vegas or some other kind of TCP implementations. Who should choose the best TCP implementation for the application considering the communication environment?

As a result of the TCP improvement to reach its bedrock, new approaches based on TCP overlay network are proposed. In the architecture TCP is relayed in network, i.e. TCP is terminated and initiated to the next destination at TCP relay node, which is herein called TCP Bridge. The overlay network architecture will be discussed for high throughput. By relaying TCP, RTT can be short in individual TCP loop because TCP bridge terminates TCP flow control, i.e. feedback loop. Packet loss probability in individual TCP loop can also become small. TCP throughput estimation [53] shows that smaller RTT and smaller packet loss lead to higher throughput.

There are some disadvantages of TCP relay schemes. One problem is that if TCP connection is accidentally collapsed in intermediate TCP Bridge, no protocol is

provided to recover TCP connections between both ends of the collapsed port of the TCP Bridge. Another problem is that if latency is more important performance criteria than throughput, for example, small data transaction, extra delay due to TCP relay may cause QOS degradation since TCP Bridge work in store and forward basis.

While the first problem might be a fundamental problem for TCP overlay network architecture, it will not be a problem in practical sense. Separate-TCP such as Wireless TCP [69], which is based on TCP relay schemes, is lately proposed and is broadly spreading to mobile wireless networks (so what?). Furthermore, because one end of TCP often happens to become collapsed, for example, client computer being hung up, almost all applications using TCP provide application layer handshake, i.e. acknowledgement mechanism in case against such TCP collapse. The second problem, the increase of latency, occurs depending on implementation of TCP relay architecture. If TCP Bridge is explicitly specified or not specified by an application when TCP connection is set up, the problem may not arise because throughput sensitive applications only can specify TCP bridge. Since TCP Bridge has a little experience, there are many issues to be solved. One of these is congestion control issue. If a trivial congestion occurs on sending side of a TCP Bridge, it causes serious rather than non-trivial congestion on receiving side of the TCP Bridge. To prevent this problem buffer control in TCP Bridge is proposed and discussed. Simulation results show that proposed control method improve the problem and can achieve throughput two times higher than the case without any control.

Above-mentioned traffic control technologies, including CAC, pre-fetch caching, TCP overlay network, are necessary to prevent network congestion. In other words, such traffic engineering technologies must be employed in the next-generation high-quality Internet.

Besides these traffic control or traffic engineering technologies, high-speed switching technologies are required to reduce congestion by reducing network utilization. Recent high-speed routers have several hundreds interfaces of Gbps lines and a capacity of terra (T) bps. Such routers are very expensive because they use expensive hardware chips for protocol processing. One of the bottlenecks of the protocol processing is a search process in IP address table lookup. The table stores IP packet forwarding information and must be searched by Longest Prefix Matching (LPM) search method. Although Full Matching search methods are well developed and less expensive, LPM search has not been developed yet. For example of LPM search, there are supposed to be three entries in IP version 4 address tables, entry A: 63.0.1.*, B:63.0.2.* and C:63.0.2.4 where "*" means "don't care," i.e. wildcard. When IP packet having 63.0.*x*.*y* arrives, the table is searched using IP address as a search key. In the case of *x*=1, entry A must be the search result. In the case of "*x*=2 and *y*=3," and "*x*=2 and *y*=4," the search results are entry B and C, respectively because entry C match longer than B in the case of "*x*=2 and *y*=4". The situation becomes even more difficult because the number of table entry is increasing day by day. For example, more than 64K entries are stored in border gateway routers. It is necessary to develop search methods for a large table in enough high-speed. There are two approaches; one is algorithmic approach and the other is hardware approach. Algorithmic approach has advantages of low cost and robust for table size. Previous researches are based on Patricia tree algorithm and are mainly devoted for improvement of Patricia tree search [70]. On the other hand, a hardware-based approach has advantages in its speed and simple implementation. Some venders have already sold a search chip, which is based on Ternary Contents Addressable Memory (T-CAM) [64][68]. While T-CAM has not only "0" and "1" state but also "don't care" state for LPM search, T-CAM is likely to

work as CAM. This is the reason why this is fast and simple. Because the chip is expensive and its capacity is limited, for example, 4K entries, this approach cannot adopt the increase of the table size. Because neither approach is able to achieve both fast and low cost search, a new approach is required.

On the other hand, if the requirement of search speed is not so rigid, i.e. is not in wire-speed, it is possible to have high-speed, low cost and large capacity search methods. They intend to search all table entries with always the same speed. In real networks, entries searched in a short time are expected not so spread and may have locality and successiveness because several packets for the same destination are likely to arrived at the routers in a short time. It is, therefore, possible to develop a new search method with low cost and enough speed by considering a practical packet arrival process.

One of the feasible methods is a cache method where recently used entries only are stored in the cache table and all entries are stored in the main table. Expensive but fast search is employed for the cache table such as T-CAM chips, and slow but low cost search is employed for the main table (herein called full table). This method has characteristics that it can search fast if cache hits, but the search becomes slow if the search entries are widely spread due to cache miss-hit. The previous idea of this cache method is here referred host-address-cache methods since host address is stored in the cache as the entry [67]. The host-address-cache, however, is expected poor performance because its cache covers very narrow IP address. For example B:63.0.2.* covers 256 IP addresses but C:63.0.2.4 covers only an IP address. This may cause thrashing between cache and full table and result in high miss-hit ratio if it is used in backbone routers, which receive widely variety of IP address. It is therefore important to improve miss-hit ratio in a cache table in order to realize low cost and high speed LPM search. The idea

is proposed that cache architecture is employed with algorithmic search and hardware search engine. Instead of ordinary CAM as used by conventional caching architecture, T-CAM is employed to reduce cache miss-hit ratio. Because caching-in/out rule is not obvious in LPM search, the rule is carefully invented and investigated for a validation. Performance evaluation is shown to disclose the proposed architecture can achieve at least ten times smaller miss-hit ratio than the conventional cache architecture.

Today's Internet has become a part of the daily life of residential users and business users as a life infrastructure. Internet must have high quality and capability of multimedia data traffic. Toward the future Internet, this thesis discusses new architecture and control for high QOS with the networks while operational cost and equipment cost remains minimum. We believe these discussions contribute to realize next generation high quality, high-speed Internet.

1.2 Organization of this thesis

This thesis consists of five approaches for high quality network. Connection admission control (CAC), burst transfer methods, network caching architecture, TCP overlay network architecture and high speed IP address table search methods are individually addressed.

Chapter 2 discusses CAC, which enables networks to achieve required QOS values while keeping network utilization at its maximum. The CAC proposed in this chapter has advantages; only two parameters and small time are needed for accept/reject new connection request. The proposed CAC uses "Virtual cell loss rate" instead of cell loss rate as cell loss estimation. Simulation results show that the proposed CAC has appropriate accuracy in estimation and offers appropriate network utilization.

In Chapter 3, burst transfer methods are addressed for more bursty data. To accommodate bursty data in an efficient manner, it is suitable to reserve and release bandwidth for each piece of burst data at the time when the burst is ready to send. This, however, may cause large latency to the burst if the bandwidth is not available. The more the number of links are on the end-to-end path, the larger the expected delay to grab the all bandwidth on the path simultaneously. To make the latency small, burst server architecture are proposed. In the architecture, a burst server, which stores and forwards the bursts, is placed between links and the reservation methods are modified. Bandwidth reservations can succeed either if the bandwidth on all links between sending and receiving terminals are available, or all links between burstservers or sending/receiving terminals. Numerical results show that the latency is improved in this architecture.

In Chapter 4, we prove the fact that if we can take account of application feature into congestion control, both network utilization and user QOS can be improved. One of the today's most important and popular applications of the Internet is web (World Wide Web). The web has a feature that the content that is to be retrieved next can be predicted from the current contents. This feature implies pre-fetch retrieval may be able to hide transfer delay (latency) to users/browsers. There is a tradeoff between the latency and the traffic load due to pre-fetching as many contents as potentially expected to be used. If the pre-fetch is done when network is not congested, i.e. is in idle, network can be well utilized and latency can be small as well. The network cache architecture for network friendly pre-fetch retrieval is therefore proposed.

Chapter 5 describes TCP overlay network architecture. TCP has been developed since 1980s, and has never essentially improved by today. TCP has to provide new functions to meet their requirements for today's high-speed link, fairness consideration,

and future services. I believe it is easy to introduce new TCP functions not to end hosts/servers, but to the intermediate node. TCP relay node (TCP bridge) can thus give new functions such as reliability and high throughput applications without changing existing IP network and user terminals/servers.

In Chapter 6, improvement of IP router in protocol processing speed is discussed. To accommodate huge Internet traffic, high-speed routers are the key. IP address table search is one of a bottleneck to realize high speed protocol processing in such routers. Longest Prefix Matching (LPM) search must be used to search the table. Although a lot of quick search methods are developed in Full matching search, they cannot apply to LPM search. Algorithmic approach is very cost effective but has lower speed. One of the solutions for quick search is to develop hardware search engine, i.e. Ternary CAM (T-CAM). T-CAM is tri-state Contents Addressable Memory. However, It causes high cost and small capacity, and to accommodate large IP table, many T-CAM chips are necessary, which increases the cost. The idea presented in this thesis is to solve the trade off with cache architecture with algorithmic search and T-CAM. Because of LPM search, caching-in/out rule is not obvious. The rule is carefully invented and performance evaluation is shown in this chapter.

The last chapter concludes this thesis.

CHAPTER 2
A CALL ADMISSION CONTROL FOR ATM NETWORKS BY USING SIMPLE QUALITY ESTIMATE

2.1 Introduction

The asynchronous transfer mode (ATM) is an important component of the Internet. ATM has useful features, for example connection oriented connection, traffic control to each connection inside networks [1], [2]. Since ATM, however, involves a statistical multiplexing scheme, queueing delay, cell loss, or other degradation of quality of service (QOS) may easily occur in cases of network congestion. Bursty traffic, such as image data transfers, has a particularly serious impact on communication quality because of statistical load fluctuations. Traffic control is, thus, necessary to avoid congestion. Traffic control methods may be divided into two categories (reactive control and preventive control) and the most efficient use of network resources may be achieved by combining the two. Call admission control is, of course, a form of preventive control, which is by its nature more effective than reactive control for use in high-speed networks [3].

We are interested here in developing a practical control system, a very important function of which is the guaranteeing of a specific QOS for each call. It is also very important that the decisions to accept or reject calls be made in real time. That is, a simple and fast call admission control should be able to estimate individual cell loss probability rather than average cell loss probability. From the economical aspect, it is expected for the control to obtain statistical multiplexing gain. The call admission

control we report here is based on a measure which estimates traffic characteristics in a link. The measure is calculated from user declaration parameters of a traffic description. Burst traffic is modeled here as on-off traffic and is basically characterized by three parameters: peak rate, mean rate, and burst length. A measure based on "queueing model" is sensitive to burst length, which is difficult to be declared and policed. A measure based on a "bufferless fluid flow model" is simply calculated by two parameters of peak rate and mean rate, which are easily declared by users and easily policed [4]. We use a call admission control using a measure based on a bufferless fluid flow model. Many preceding works which fall into this category use average cell loss probability defined over different traffic class mixes [4]-[8]. In many cases of practical interest, where a variety of applications share a link, it may be difficult for an admission control based on the average cell loss probability to reflect a specific quality requirement for individual calls. In [9] and[10], individual cell loss probability has been introduced. However, it is not sufficient to discuss the use of the measure for practical control. We will show the comparison between average and individual cell loss probabilities, and show link utilization results using the measure of individual cell loss probability. Several other studies have proposed simple and fast control procedures using virtual bandwidth methods but, for similar reasons, they are insufficient to ensure that all individual cell loss probabilities lay within a specific degree of quality. Further, while each of these studies note that virtual bandwidth methods fail to guarantee a specific quality in heterogeneous traffic multiplexing, none gives any solution to the problem [11]-[14]. In [10], the problem is referred to as "interference" between different types of bursty traffic and, though a solution is offered, it is still incomplete. That is to say, they seek to improve the virtual bandwidth method by using a virtual link capacity scheme, the basic idea of which is to give a margin to every virtual bandwidth and to

execute acceptance judgments using subdivided link capacities. They fail, however, to mention how to allocate bandwidths so as to meet the various QOS requirements.

In Section 2.2 of this thesis, after briefly describing our traffic model, we introduce a "link overflow model," a kind of "bufferless fluid flow model," and compare the virtual cell loss probability calculated for it with cell loss probability calculated for a queueing model. In Section 2.3, we show the difference between individual cell loss probabilities and average cell loss probability. Section 2.4 proposes a bandwidth allocation scheme, based on the virtual bandwidth scheme, which employs virtual link capacities. We summarize the results of our study in Section 2.5.

2.2 Quality estimation measure – virtual cell loss probability

2.2.1 Definition and features of virtual cell loss probability

The traffic model we use here is characterized as follows. Bursty traffic sources such as video sources might generate bursty traffic, i.e., bursty cell streams. Such sources have both active and idle periods. In active periods, cells are generated at a constant rate ***MAX***, the peak rate. In idle periods, no cells are generated (see Fig. 1-1). The number of cells generated in an active period is denoted by ***B***, and ***AVG*** is the mean rate. The respective source's being active or idle are, then, probabilities for ***AVG***/***MAX*** and 1 - ***AVG***/***MAX***. In constant bit rate (CBR) traffic, ***AVG*** is equal to ***MAX***.

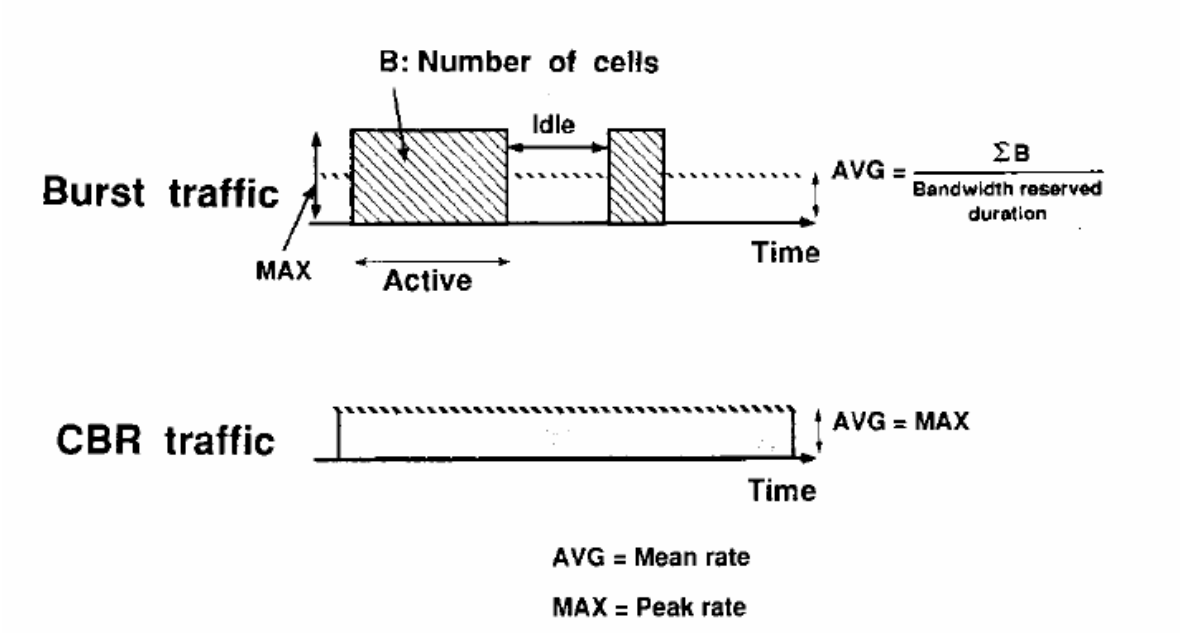

Fig. 2-1 Traffic model

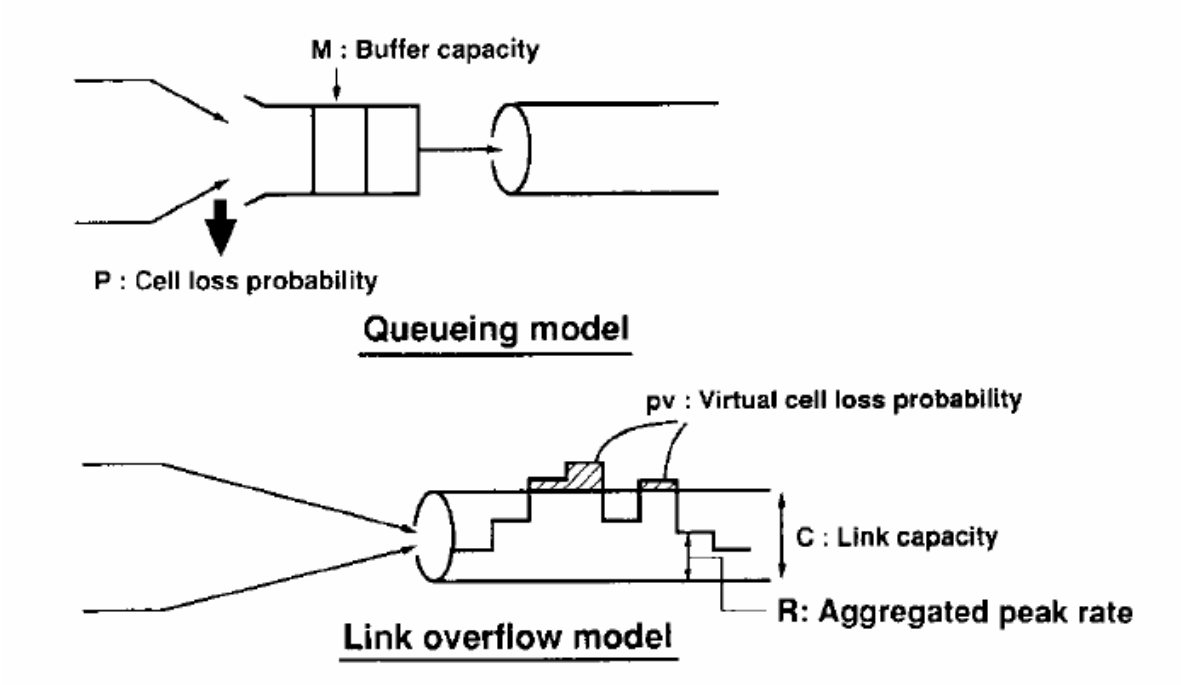

Fig. 2-2 Real cell loss probability ***P*** versus virtual cell loss probability ***pv***

Although real ATM networks have buffers in each switch, for simplicity's sake, we use a bufferless model. In our model, quality is measured in terms of the virtual cell loss probability, which is based on a "link overflow model," a logically bufferless fluid flow model, derived from the traffic characteristic parameters ***MAX*** and ***AVG*** alone [4], [7], [8]. In the link overflow model, cell losses due to overflow occur if and only if an aggregated peak rate ***R*** exceeds link capacity ***C*** (Fig. 1-2), where ***R*** is defined by a load with ***n*** sources being active, i.e., ***n*·*MAX***. Cell loss probability in the model is the ratio of excess traffic ***OF*** and traffic load ***ρ***. The virtual cell loss probability (***pv***) is defined in (1), where the number of sources multiplexed in the link is denoted by ***N***.

$$pv=OF/\rho \tag{2-1}$$

$$OF = \sum_{(n \cdot MAX-C)=>0}^{n=N} p(n)(n \cdot MAX-C) \tag{2-2}$$

$$\rho = N \cdot AVE \tag{2-3}$$

where p(n) is the probability that n out of N sources are active: that is.

$$p(n) = \binom{N}{n} \left(\frac{AVG}{MAX}\right)^n \left(1 - \frac{AVG}{MAX}\right)^{N-n} \tag{2-4}$$

One of the advantages of ***pv*** is that it is easy to declare and police both ***MAX*** and ***AVG***[1]. Another advantage is its explicit formulation. This means that it is simple and easy to estimate the quality provided to the users.

In the bufferless model, when the aggregated peak rate ***R*** is smaller than link capacity, i.e., ***R*** < ***C***, cell loss is assumed to never occur. Even if the condition ***R*** < ***C*** holds true, due to a short-term load fluctuation caused by simultaneous cell arrival from different calls, traffic load may instantaneously excess the link capacity. It is the reason why buffers should have an appropriate capacity in a real system to prevent cell loss due to the short-term load fluctuation. The necessary buffer capacity is derived from anM/D/1-S queueing model [16], [17]. For example, about 100 cells are necessary[2] for a cell loss probability of 10^{-9} and a traffic intensity of 0.9.

The cell loss probability observed in a real system is less than the quality measure ***pv***, since the buffer can save cells that should normally be discarded in a bufferless model.

---

1 A sliding-moving window scheme can police ***MAX*** and ***AVG***.

2 The buffer would cause cell delay, but the delay might be quite small in high-speed networks.

2.2.2 Comparison between real and virtual cell loss probability

Simulation results show that, for identical traffic, the virtual cell loss probability ***pv*** in the link overflow model is always larger than real cell loss probability ***P*** in the queueing model. Fig. 1-3 shows ***pv*** and ***P*** for $\bar{B}/M$[3], where

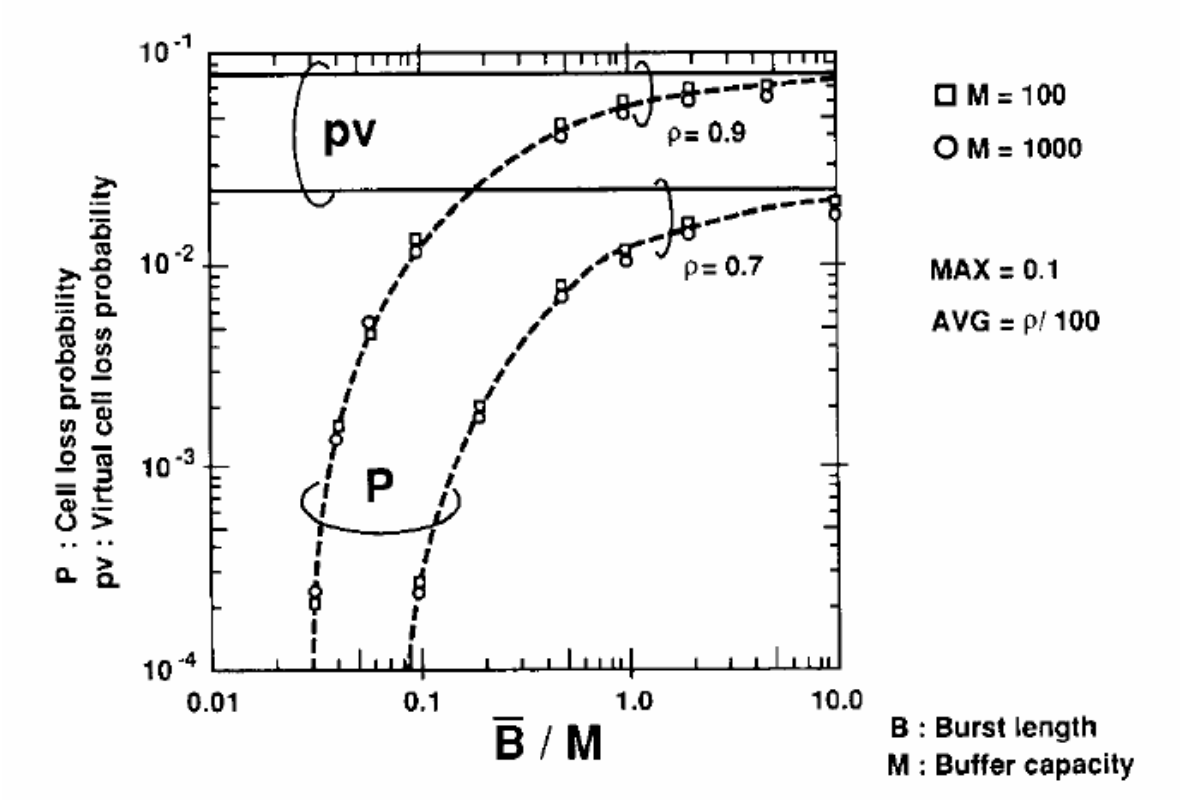

Fig. 2-3 Cell loss probability: ***P*** versus ***pv***

$\bar{B}$ is the average burst length, ***M*** is buffer capacity, ***MAX*** and ***AVG*** are normalized by the link capacity ***C***, and the link utilization ρ is fixed. In this case, 100 homogeneous traffic sources which have a peak rate of 0.1 and mean rate of 0.009 are

---

[3] Even with the same load and with the same ***MAX*** and ***AVG*** parameters, the value of $\bar{B}/M$ determines ***P*** [8], [13]-[15].

multiplexed on the link. The figure shows that ***pv*** provides the upper bound of ***P*** as $\bar{B}/M$ increases. Furthermore, ***P*** converges to the value of ***pv*** when $\bar{B}/M$ goes to infinity. For example, ***pv*** is almost the same as ***P*** for $\bar{B}/M$ = 208.33, which corresponds to the ratio of one burst of still picture containing 1 Mbyte (= 20,833 cells) and buffer capacity of 100 cells.

From a control point of view, this means that ***pv*** is a conservative quality measure of ***P***[4], and a good quality measure because ***pv*** is robust to variation of $\bar{B}$. The same results are expected in the cell loss probability under 10^{-4}, which is hard to obtain due to computer processing power.

The efficiency of statistical multiplexing under ***pv*** < 10^{-9} results in Fig. 1-4, where non-statistical mode means a maximum utilization under a peak rate multiplexing, i.e., ***MAX*** < ***C***. The figure shows that we can expect statistical multiplexing gain for almost all kinds of traffic. It also shows that we should not expect an economical use of resources for traffic whose peak rate is bigger than 0.1. To reduce peak rate, a user can employ a traffic shaping mechanism [3]. The features of the call admission control, based on ***pv*** and ***P***, are summarized in Table 1-1. Consequently, we use the virtual cell loss probability as the quality measure for call admission control.

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4 When ***B*** is smaller than ***M***, ***P*** becomes too sensitive, and such characteristics become intractable for the call admission control.

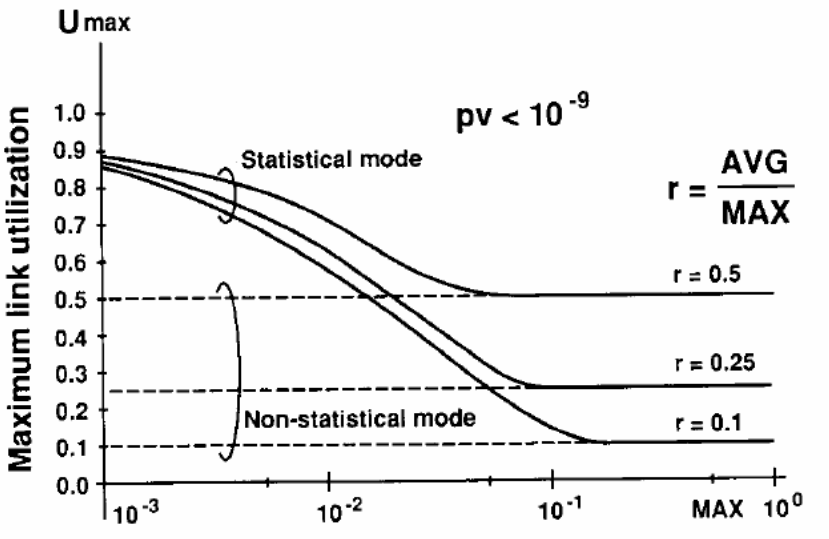

Fig. 2-4 Maximum link utilization versus ***MAX*** (homogeneous traffic)

Table 2-1 Traffic control scheme comparison based on ***pv*** versus ***P***

TRAFFIC CONTROL SCHEME COMPARISON BASED ON *pv* VERSUS *P*

	Required Parameters	Quality Estimation	Link Utilization
pv Virtual cell loss probability	*MAX* *AVG*	Robust	*See Fig. 4.
P Cell loss probability	*MAX* *AVG* *B*, *M* distribution parameters	Sensitive	High

2.2.3 Extension to a network model

We discuss how to estimate virtual cell loss probabilities in intermediate links, where the traffic usually has different characteristics from those of a sending terminal. Generally, traffic characteristics are changed at every queue[8], [14], [18], and their estimation is too complicated. To evaluate such changes, we employ a queueing network model shown in Fig. 1-5(a) for output-buffer type switches [19], [20]. Fig. 1-5(b)-(d) show a queue length Q$\boldsymbol{i}$ instead of the cell loss probability in simulation results, where a subscript $\boldsymbol{i}$ indicates $\boldsymbol{i}$-th stage of the network model [8], [21]. In Fig. 1-5(c) and (d), using the average queue length $\boldsymbol{q}$ and the variance σ from the average, $\boldsymbol{q} + 3\boldsymbol{\sigma}$ is indicated as the approximate tail distribution of the queue length. It is found that traffic is still bursty at the second stage, but is less bursty than at the first stage. Such effects may be caused by a decrease of ***MAX***. The reason is thought to be that several bursts are mixed and interleaved with each other in a queueing buffer, and they leave the buffer to reach the next queueing buffer in a sparse form (as illustrated in Fig. 1-6). Other simulation results also show that the average queue length at the third stage is smaller than that at both the first stage (Q1) and second stage (Q2), but almost the same as Q2[5]. Similar results concerning the queue length distribution for interrupted Poisson process traffic [18] confirm the above discussion.

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[5] A study in [14] evaluates a two-stage queueing network model, and explains similar behavior as the "smoothing effect." That study, however, does not give the traffic behavior after the second stage.

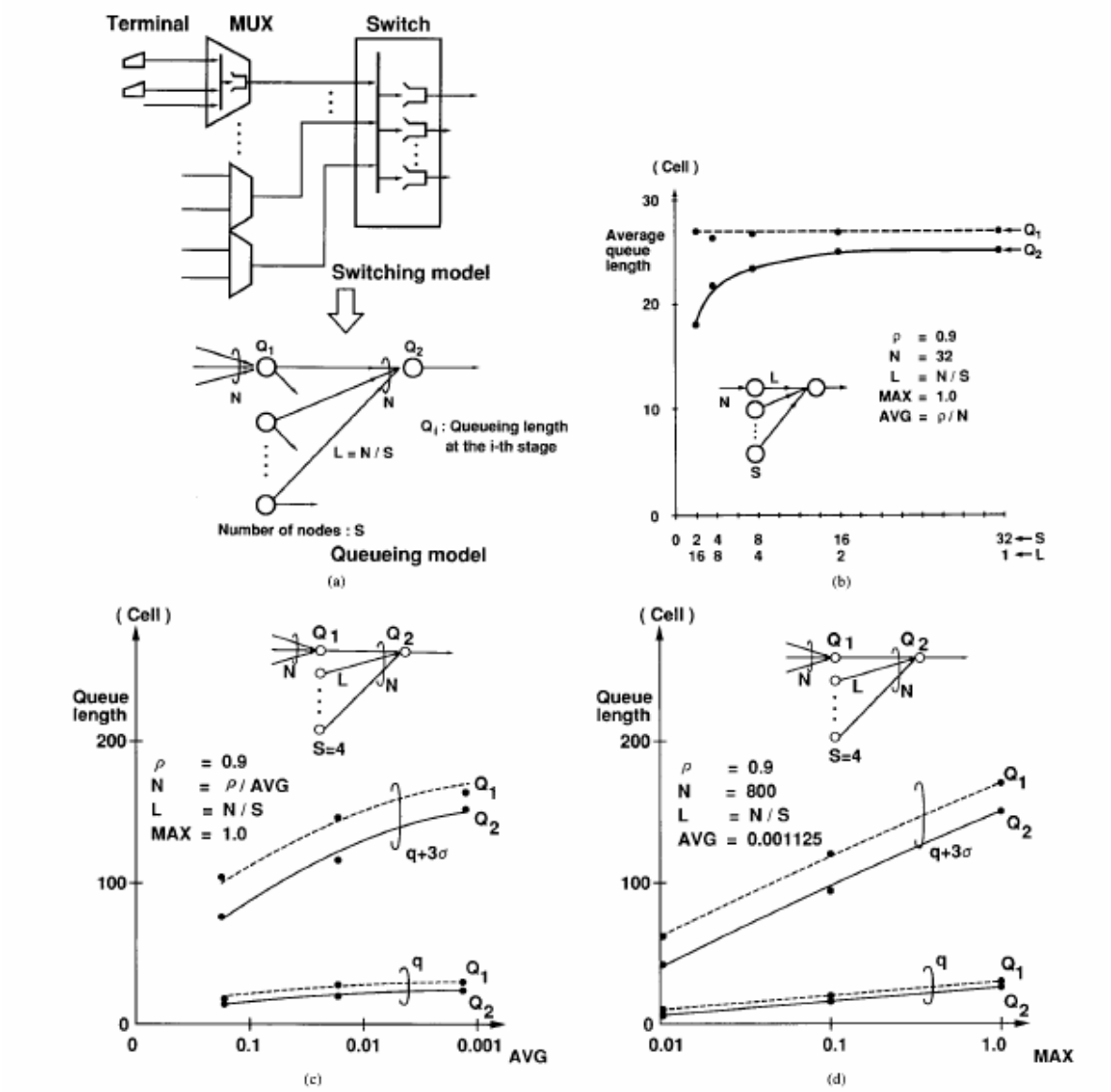


Fig. 2-5 Network model. (a) Queueing network model. (b) Queue length versus number of nodes S. (c) Queue length versus ***AVG***. (d) Queue length versus ***MAX***

It would be reasonable to predict that the average queue length in the ***i***-th (i > 3) stage (Q2,) is smaller than Q1. However, the queue length difference between the first

and ***i***-th (> 1) stages is small. We should assume that traffic characteristics are the same in all stages in order to implement a simple and practical control, although the quality estimation at the ***i***-th stage is rather conservative. We conclude that the virtual cell loss probability ***pv***, which is calculated using the peak and mean rates of the sending terminal, should be directly applied to all intermediate links.

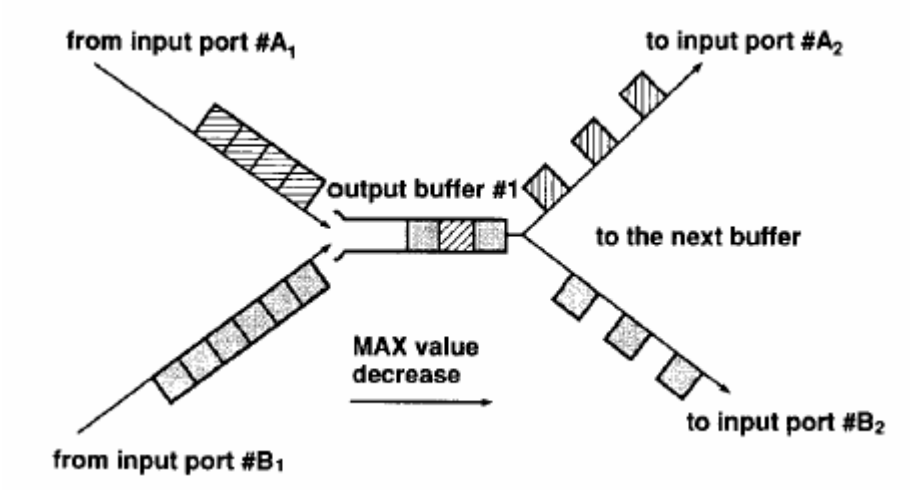


Fig. 2-6 Change into less bursty

2.2.4 Extension to a heterogeneous traffic environment

In the case of a heterogeneous traffic environment, it is necessary to extend the virtual cell loss probability defined in the case of homogeneous traffic of Section 2.2.1. At first, it is convenient for control to categorize traffic types according to their characteristics, i.e., ***MAX*** and ***AVG***. The peak rate of the traffic category ***j*** is denoted ***MAXj***, and the mean rate ***AVGj***. The virtual cell loss probability (***PV***) for heterogeneous traffic is defined in (5) [8].

$$PV=OF/\rho \tag{2-5}$$

$$OF = \sum_{n_i \in \{\sum_{i=1}^{K} n_i \cdot MAX_i - C \geq 0\}}^{n_1=N_1} \cdots \sum^{n_K=N_K} \left[\prod_{i=1}^{K} p_i(n_i)((\sum_{j=1}^{K} n_j \cdot MAX_j) - C) \right] \tag{2-6}$$

$$\rho = \sum_{j=1}^{K} n_j \cdot AVG_j \tag{2-7}$$

$$p_j(n_j) = \binom{N_j}{n_j} \left(\frac{AVG_j}{MAX_j} \right)^{n_j} \left(1 - \frac{AVG_j}{MAX_j} \right)^{N_j - n_j} \tag{2-8}$$

where

K: Number of categories

Nj: Number of category j calls

The virtual cell loss concept based on the link overflow model is, thus, easily extended to a case of heterogeneous traffic.

On the other hand, ***PV*** calculation steps would dramatically increase as the number of categories ***K*** increases. From a practical point of view, in order to reduce calculation steps, we present in Section 2.2.4 another approach to implement a call admission control scheme based on an indirect calculation of ***PV***.

2.3 Guaranteeing a specific QOS – individual virtual cell loss probability

2.3.1 Characteristics of individual multiplexed traffic

From a practical point of view, call admission control should guarantee a specific QOS for every call using a measure which evaluates individual cell loss probability observed for each call.

When different categories of traffic, i.e., traffic sources with different characteristics, are multiplexed, even if their required qualities (cell loss rates) are the same[6], the cell loss probabilities observed for each individual category are usually different [22]-[24]. The cell loss probability for traffic type ***j*** is denoted by ***Pj***, and the average value of ***Pj*** (***j*** = 1, … , ***K***) is denoted ***P***. It is easy to predict that ***Pj*** is relatively larger for more bursty traffic and relatively smaller for less bursty traffic when various kinds of bursty traffic sources are multiplexed. The intuitive reason is that more bursty traffic causes congestion by itself, so that cells tend to be discarded more frequently. We need to extend the virtual cell loss probability concept in order to consider a specific quality for individual traffic, because the virtual cell loss probability mentioned above corresponds to an average of the individual cell loss probabilities for individual traffic sources.

2.3.2 Definition of individual virtual cell loss probability

In order to investigate individual loss characteristics and to use them in a call admission control scheme, a quality measure, individual virtual cell loss probability, ***PVj*** should be employed. The individual virtual cell loss probability ***PVj*** is based on the

---

6 In this chapter, we focus on individual traffics (all of which require the same QOS).

same conceptual model as the average virtual cell loss probability ***PV***. Fig. 1-7 illustrates the individual link overflow model, where several Type 1 and Type 2 calls are multiplexed. The individual virtual cell loss probability for two types of traffic is shown in [9]. The individual virtual cell loss probability for type ***j*** among ***K*** types of traffic results in ***PVj*** in (9) [10], [25].

$$pv_j = OF_j / \rho_j \tag{2-9}$$

$$OF_j = \sum_{n_i \in \{\{\sum_{i=1}^{K} n_i \cdot MAX_i - C \geq 0\}\}}^{n_1=N_1} \cdots \sum^{n_K=N_K} \left[\prod_{i=1}^{K} p_i(n_i) (((\sum_{j=1}^{K} n_i \cdot MAX_i) - C) MAX_j \cdot n_j) / (\sum_{i=1}^{K} n_i \cdot MAX_i) \right]$$

(2-10)

$$\rho_j = n_j \cdot AVG_j \tag{2-11}$$

$$p_i(n_i) = \binom{N_i}{n_i} \left(\frac{AVG_i}{MAX_i} \right)^{n_i} \left(1 - \frac{AVG_i}{MAX_i} \right)^{N_i - n_i} \tag{2-12}$$

Where

K: Number of traffic categories

N: Number of type j call

MAX: Type j peak rate

AVG: Type j mean rate

C: Link capacity

Underlined terms in (10) correspond to individual loads that instantaneously exceed the link capacity, OFj, in Fig. 1-7. When ***MAX*** and ***MAX/AVG*** increase, traffic

becomes more bursty [8], [13], [15], [26]. From (10), ***PVj*** is large for relatively large ***MAXj*** and ***MAXj***/***AVGj***.

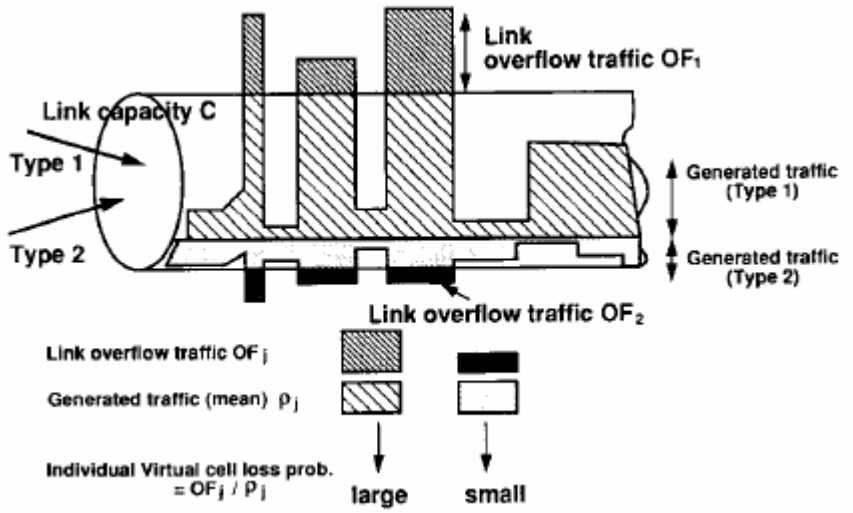


Fig. 2-7 Individual virtual cell loss probability

2.3.3 Comparison between real and virtual individual cell loss probabilities

Comparing ***PVj*** with ***Pj*** shows that ***PVj*** is reasonable as a quality measure for ***Pj***. ***Pj*** was evaluated by a simulation technique. Fig. 1-8 shows ***PVj*** and ***Pj*** as a function of ***B/M***. In Fig. 1-8, Type 1 traffic is more bursty than Type 2. Comparison between virtual and real cell loss probabilities of Type 1 traffic, ***PV1*** and ***P1***, has a similar behavior as those in the average case (as shown in Section 2.2.2). We can obtain the same conclusion about Type 2 traffic behavior. Consequently, the individual virtual cell loss probability ***PVj*** is suitable for quality estimation for individual traffic.

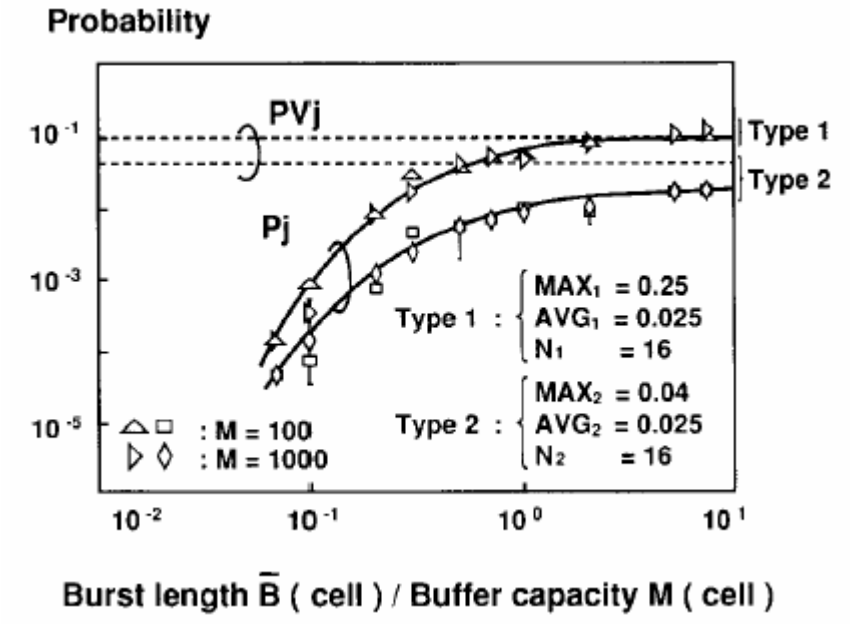


Fig. 2-8 Individual real cell loss probability Pj versus individual virtual cell los probability ***pv***j

2.3.4 Comparison between average and individual virtual cell loss probabilities

It is necessary to know how much difference there is between individual and average virtual cell loss probabilities, i.e., ***PVj*** and PV. Fig. 1-9 shows the ***PVj*** characteristics (solid line) for two kinds of calls, Type 1 and Type 2, and ***PV*** (dashed line) defined by (5), where Type 1 traffic is more bursty than Type 2. The difference is large between ***PVj*** and ***PV***. The ***PVj*** ratio between Type 1 and Type 2 is about 10 times. The same characteristics hold true for three or four kinds of calls. In that case, for example, the ***PVj*** ratio between the most (***MAX*** = 0.2) and the least (***MAX*** = 0.02) bursty traffic types is at least 10 times. Consequently, the difference between ***PVj*** and

PV is significant and it is necessary to base the call admission control, not on the average, but on the individual virtual cell loss probability, in order to guarantee a specific QOS for every traffic types. In the next Section, such a call admission control is practically developed.

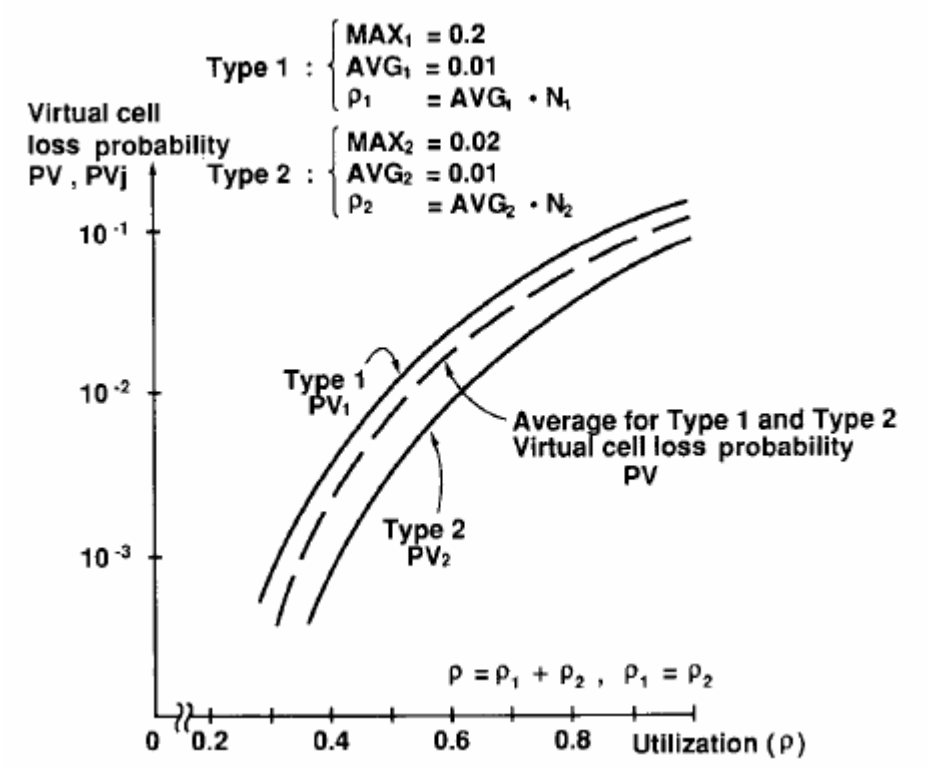


Fig. 2-9 Individual virtual cell loss probability characteristics

2.4 Call admission control scheme

2.4.1 Virtual bandwidth method

A call admission control scheme is developed, based on the above-mentioned individual virtual cell loss probability. Call admission control schemes using virtual bandwidth methods have been proposed in many papers as practical control methods [11]-[15], [27]. In the virtual bandwidth method, each call has its own virtual bandwidth, and a new call is accepted only if the total virtual bandwidth is smaller than the link capacity [see Fig. 1-10(a)]. The virtual bandwidth ***VB*** is defined as the bandwidth necessary for one call to guarantee a specific cell loss probability in a homogeneous traffic environment [13], [27]. Fig. 1-10(b), which is obtained using (1), provides ***VB*** for a cell loss probability of 10^{-9}. Although a virtual bandwidth method is very practical, we found two problems in it. After briefly introducing these problems and the way to solve them, we discuss the details in the following sections.

Problem 1: The virtual bandwidth method has the disadvantage that it is not certain to guarantee a specific cell loss probability [12]-[14], [26] because of interference between different traffic categories during statistical multiplexing, although it has the advantage of being quite simple. To avoid such uncertainty, the virtual bandwidth method should be executed based on the virtual link capacity scheme [13].

Problem 2: The relation between the required QOS and the provided VB is not clear. A study in [28] proposes a scheme to realize multiple QOS by giving each call a different VB and priority. This scheme, however, requires a rather complicated algorithm, and it is not clear whether it guarantees the required QOS. We take a similar approach, but we emphasize simplicity and the respect of the QOS instead of emphasizing efficiency.

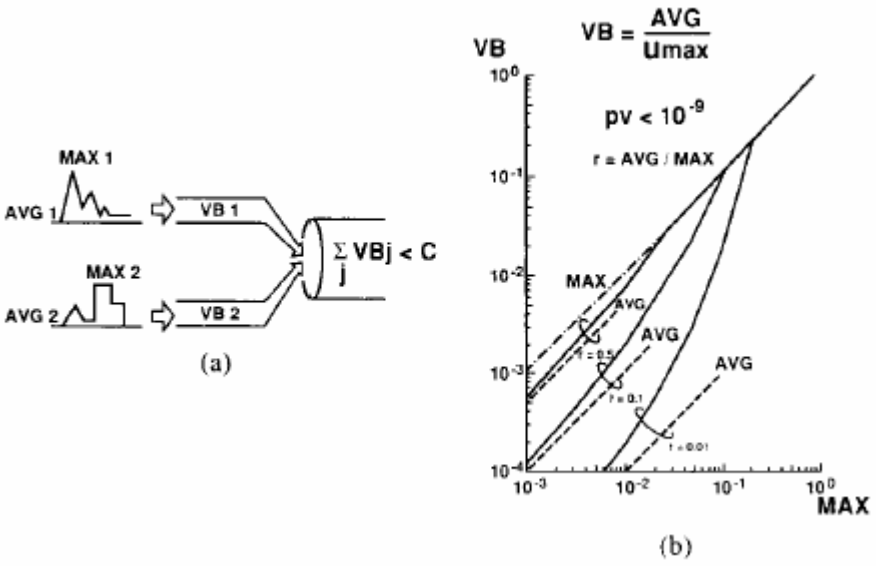


Fig. 2-10 Virtual bandwidth (***VB***) allocation

2.4.2 Virtual link capacity

First, we describe the problem of the virtual bandwidth method, then explain the virtual link capacity scheme.

Fig. 1-11 shows four admissible call regions, corresponding to a cell loss probability guaranteed at 10^{-9} for both types of calls. These regions are represented in Table 1-2.

The PVj-region is the exact admissible call region in which the cell loss probability is under 10^{-9}. It is more concave than the KB-region[7], as shown in Fig. 1-11. This means that a call lying in the shaded area of Fig. 1-11, if accepted by the call admission control scheme, might have a cell loss probability bigger than 10^{-9}. The reason why the PVj-region is concave seems to be that statistical multiplexing is interfered with and it results in a reduction in statistical multiplexing gain for every call. (We call such phenomenon "interference.")

In order to ensure a specific quality, we use the modified virtual bandwidth VB' instead of ***VB***. ***VB'*** is function of ***VB*** and is larger than ***VB***. For example, ***VB'*** = ***VB*** + 0.1 ·***VB***. In Fig. 1-1l(a), where interference has been taken small, the KB'-region is completely included in the PVj-region. On the other hand, in Fig. 1-1l(b), where interference has been taken large, the VB'-region is not included within the PVj-region.

In order to use the virtual bandwidth method with ***VB'***, we divide traffic into two categories: a small interference and a large interference category. In the case of the small interference, the KB'-region is completely included in the PVj-region as shown in Fig. 1-11(a), and in the case of the large interference, it is not. In the virtual link capacity scheme [13], traffic sources belonging to the large interference category (which we, from now on, call large interference traffic) require an amount of virtual bandwidth equal to ***MAX*** and are allocated to a predefined part of the total link capacity ***C***. On the other hand, those belonging to the small interference category (which we, from now on, call small interference traffic) require an amount of virtual bandwidth equal to ***VB*** (<***MAX***) calculated based on the rest of the total link capacity and are allocated to that

---

[7] We use here individual virtual cell loss probability ***PVj*** to derive admissible call region. When we use the average virtual cell loss probability PV by (6) instead of ***PVj***, of course, we have the same problem.

capacity. We describe this again in Section 2.4.5. Note that the link capacity is not physically but virtually divided; that is, bandwidth allocation is executed in each separate capacity and cell transfer is executed using the whole link capacity shared by the two categories. Using the combination of this virtual bandwidth method and virtual link capacity, call admission control can ensure a specific quality if traffic is appropriately divided.

The virtual link capacity concept and the division of traffic appeared in the paper [13]. However, the quality measure in the paper is not individual cell loss quality but average cell loss quality. We use this concept but should revise the division of traffic presented in [13], which we call now “traffic clustering. “

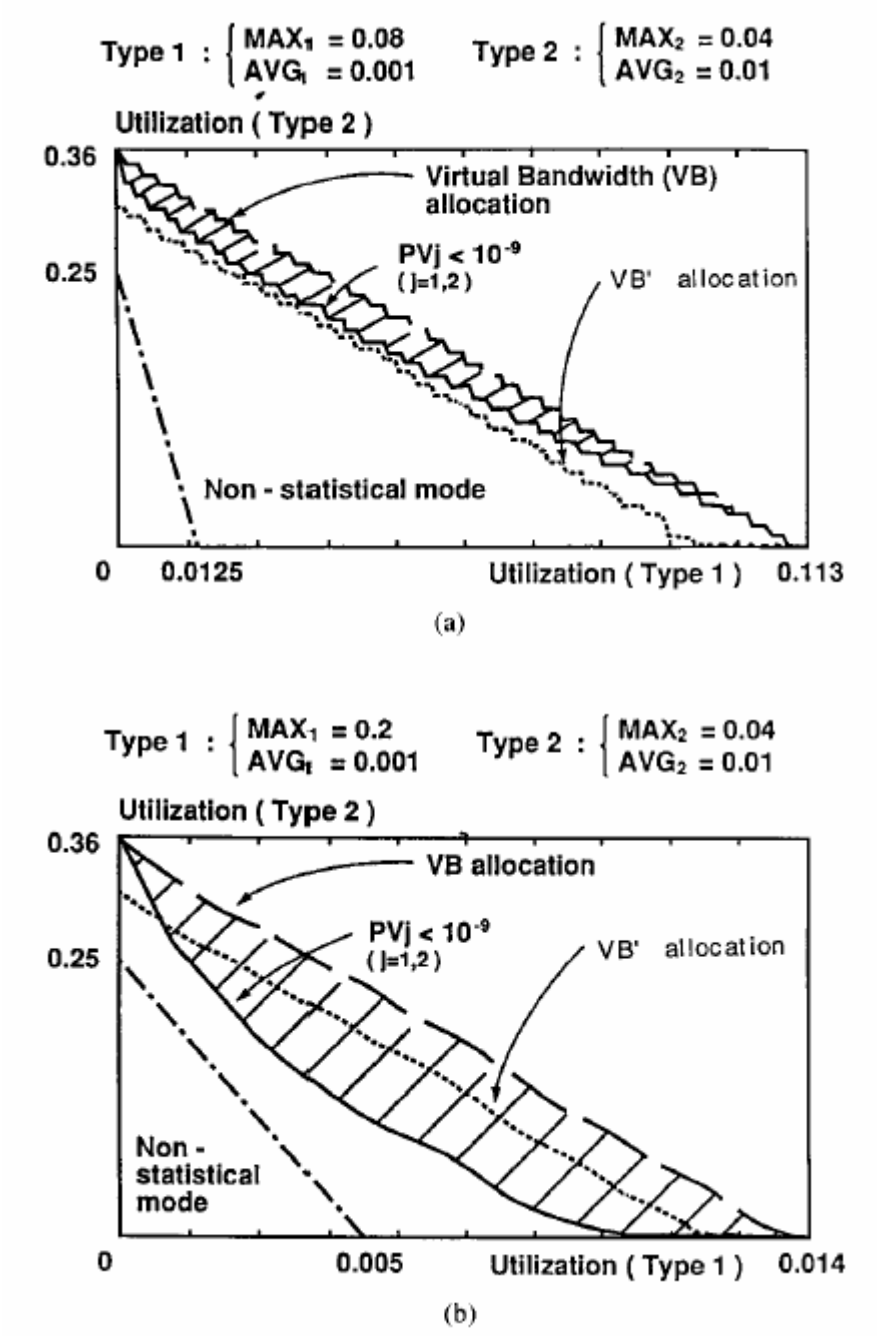


Fig. 2-11 Admissible call region (two types). (a) Interference: small. (b) Interference: large

Table 2-2 Regions

Region Name	Delimited by the Horizontal and Vertical Axis, and
VB-region	the dashed line
PV_j-region	the solid line
VB'-region	the dotted line
Nonstatistical region	the dot-dashed line

2.4.3 Traffic clustering

Because traffic-clustering results depend on modified virtual bandwidth definitions, we employ two kinds of definition for the modified virtual bandwidth. Results given by both definitions are, however, essentially similar. As mentioned above, Fig. 1-11 shows the admissible call region in the case of the small interference (a) and the large interference (b), where the modified virtual bandwidth $\boldsymbol{VB'} = \boldsymbol{VB} + 0.1 \cdot \boldsymbol{VB}$. Type 1 traffic in (b) is more bursty than that in (a), and Type 2 traffic is the same in (a) and (b). Similar results are obtained when three kinds of traffic sources are multiplexed.

In another example, to guarantee $\boldsymbol{PVj} < 10^{-9}$, we define ***VB'*** to be equal to ***VB*** calculated for $\boldsymbol{pv} < 10^{-10}$. In that case, traffic-clustering results are shown in Fig. 1-12, where Type 1 traffic is indicated by a circle if both Type 1 and 2 are the small interference traffic, and by "x" if both are not. Traffic sources that have large ***MAX*** such as $\boldsymbol{MAX} > 0.1$ and traffic sources that are nearly CBR, which even have statistical multiplexing gain, tend to virtually reduce the link capacity for statistical multiplexing and they are prone to large interference [10], [25].

We can derive some boundary conditions for traffic clustering from evaluation results in Figs. 1-11 and 1-12. Advancing in such an evaluation, the boundary conditions will become more strict, and the intensity of interference will be estimated.

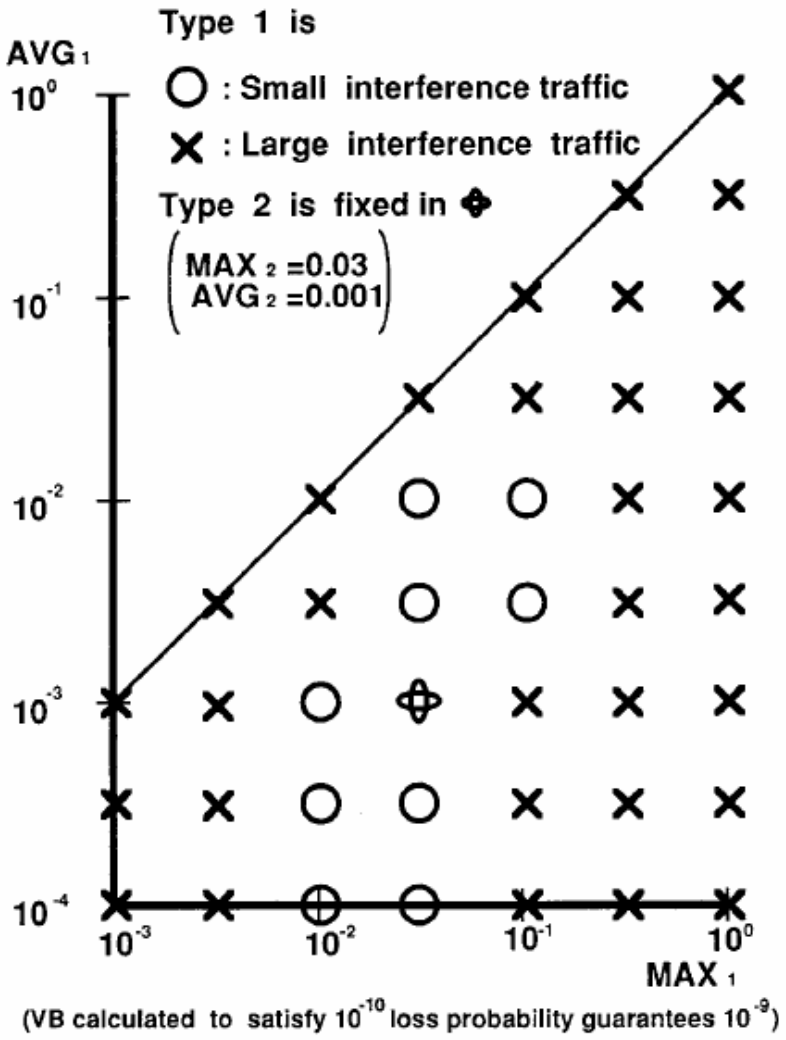


Fig. 2-12 Traffic clustering

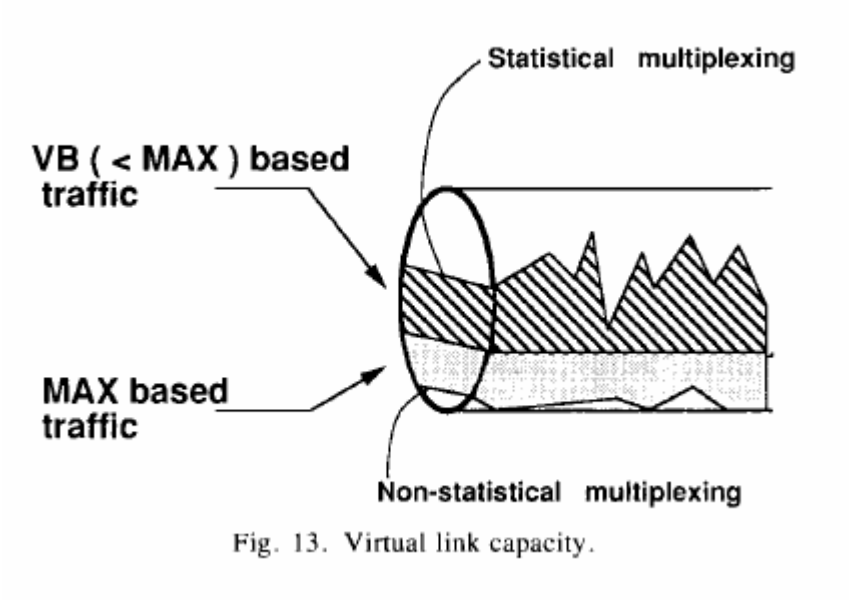


Fig. 13. Virtual link capacity.

Fig. 2-13 Virtual link capacity

2.4.4 Quality class and bandwidth allocation

The relation between bandwidth allocation and QOS classes to be provided by the network is discussed. If statistical multiplexing is adopted for all services, statistical quality degradation will occur for all services. The deterministic quality class should also be supported by the network.

Therefore, we consider that networks should treat at least two QOS classes, a "deterministic class" and a "statistical class" [26], [13]. The deterministic class implies an individual virtual cell loss probability ***PVj*** of zero, and for the statistical class traffic,

a ***PVj*** of a specific value[8]. The specific value, such as 10^{-9}, is determined by user requirements. Users can select among these QOS classes. To prevent not only buffer overflow but also link overflow, ***MAX*** is allocated to the deterministic class traffic. Basically, the virtual bandwidth as mentioned in Section 1.4.1 is allocated to the statistical class traffic. In order to prevent cell loss probability fluctuations for the deterministic class, deterministic class traffic is transmitted with a higher priority in each cell. The proposed priority control scheme provides excellent service quality (***PVj*** = 0) for the deterministic class traffic, and statistical multiplexing gain for the statistical class traffic while ensuring the required quality ***PVj***; on a long-term average. In the next section, a call admission control based on the previous discussion is summarized and evaluated.

2.4.5 Call admission control

E. Call Admission Control This section summarizes a call admission control scheme using the VB method and a virtual link capacity scheme in order to guarantee individual cell loss probability. Table 1-3 shows virtual bandwidth allocation and virtual link capacity allocation. The link capacity is divided into two capacities, ***Cmd*** and ***C'***, for two QOS classes. The "deterministic class" traffic sources are allocated in the subcapacity ***Cmd***. For the "statistical class," the subcapacity ***C'*** is virtually divided to two capacities, ***Cms*** and ***Cv***, in which traffic sources of large and small interference categories are allocated, respectively, as illustrated in Fig. 1-14. Each divided capacity, which should be determined according to the forecasted traffic intensity, is basically fixed, but can be adaptively changed with long-term traffic intensity variation. The

---

[8] For the statistical class, the ***PVj*** value can fluctuate time by time during a call [29], but it is guaranteed to meet a specific value over a long period.

virtual bandwidth ***VB*** can be calculated before call request. For example, it is convenient to make a table which has sets of (***VB***: ***MAX***, ***AVG***, ***Cv***). The decision procedure for call request requires only a few steps to look at this virtual bandwidth table. Fig. 1-14 shows the admissible call regions, under two kinds of multiplexed traffic sources, where both Type 1 and 2 are in the statistical class and both are of small interference category in (a), but only Type 1 is of large interference category in (b). Unfortunately, in (b), the efficiency of link utilization is not good when Type 1 traffic is the major load, because the scheme multiplexes traffic sources with large interference in non-statistical mode. In (a), however, it is possible to guarantee a specific QOS and to provide a good link utilization value. This control is, thus, quite simple and is able to ensure a specific quality for every traffic.

Table 2-3 Bandwidth allocation

Quality Class	Traffic Characteristics		Bandwidth Allocation		Virtual Link Capacity
		Interference	Priority	Reserved Bandwidth	
Deterministic :$PV_j = 0$	CBR Burst	—	High	MAX	$C_{md} = \Sigma$ MAX
Statistical :$PV_j < 10^{-N}$ on an average	CBR	Large	Low	VB = MAX	$C_{ms} = \Sigma$ MAX
	Burst	Small		$VB <$ MAX*	$C_r = \Sigma\ VB$

C'

—: do not care
*: VB should be calculated based on virtual link capacity C_r.

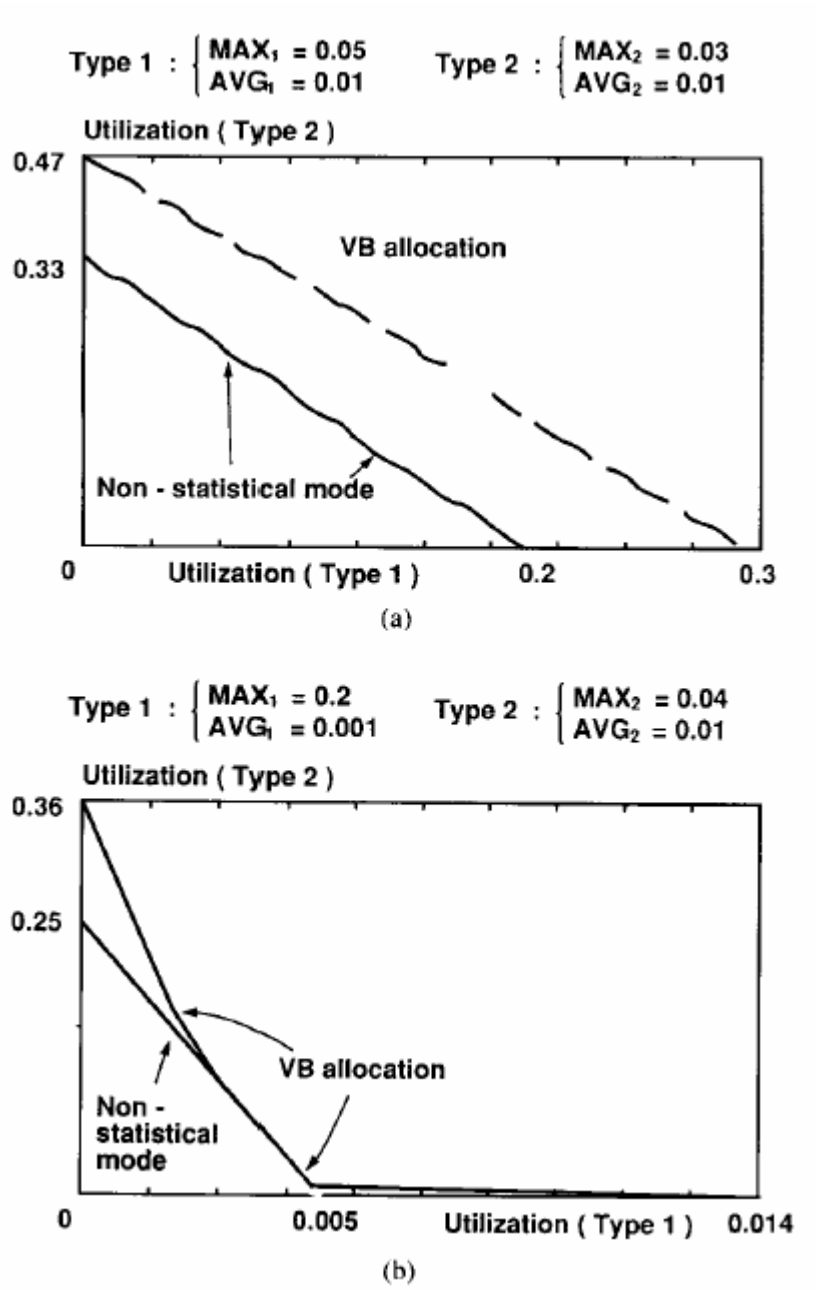


Fig. 2-14 Virtual bandwidth allocation with virtual link capacity. (a) Interference: small. (b) Interference: large

2.5 Conclusion

This chapter has indicated a practical call admission control scheme for ATM networks, in order to keep cell loss probabilities observed for individual traffic sources under a specific value. For quality estimation, we should use a quality measure, the "virtual cell loss probability" based on the link overflow model (bufferless fluid flow model). The measure is simply calculated using two traffic characteristic parameters only: peak and mean rate, and is good estimation of real cell loss probability for homogeneous and heterogeneous traffic when burst length is larger than buffer capacity. The results of link utilization for peak rate show that we cannot expect statistical multiplexing gain for traffic which has a larger peak rate than 10% of the link capacity, so that traffic shaping is useful for such traffic. For quality estimation in intermediate node or link, it is reasonable to use the same traffic characteristic parameters as those at the sending terminal. Furthermore, from the viewpoint of individual call quality, practical control should be based on the measure "individual virtual cell loss probability," to which the virtual cell loss probability is extended and which corresponds well to the real individual cell loss probability. For the implementation of call admission control, the virtual bandwidth allocation method should be employed to develop a simple and practical call admission control. Before the call setup request, each traffic is given its own virtual bandwidth, which should be determined from both the traffic characteristics and the required QOS classes such as "deterministic" or "statistical" quality class. For statistical class traffic, traffic should be divided into two clusters and be allocated in virtually divided link capacity, separately. The traffic clustering is executed based on interference intensity, which is defined as a reduction factor of statistical multiplexing gain in heterogeneous traffic, and depends on traffic characteristics. In one cluster, traffic has statistical multiplexing gain in a heterogeneous

traffic environment and is statistically multiplexed. In the other, traffic is provided with an amount of virtual bandwidth corresponding to its peak rate. For deterministic class traffic, traffic should have a high priority and should be non-statistically multiplexed in order to logically avoid any cell loss. Call admission control based on these methods is simple, practical, and guarantees a specific quality to all traffic sources.

In this conclusion, the author would like to thank Colleague of NEC for their helpful guidance and advice.

CHAPTER 3
BURSTSERVER ARCHITECTURE FOR BURST BANDWIDTH RESERVATION PROTOCOL

3.1 Introduction

To increase network utilization for accommodating more bursty traffic, burst-by-burst admission control is here discussed. High-speed bursty data traffic in ATM networks is increasing more and more as the demand to interconnect ATM LANs has been expanding. In LAN interconnection, the intermediate links between LANs are shared by many users and may become a bottleneck. The high cost of intermediate links, which may be leased from the network carrier, make it desirable for LAN users to use links efficiently. Many research works [33], [34], however, have pointed out that high-speed bursty data traffic can not be multiplexed to gain statistical multiplexing of the cell level.

The Fast Reservation Protocol (FRP) [30] is one traffic control method which provides high speed bursty data users both with high quality of service and with statistical multiplexing gain in the burst level [31]. In FRP, each bursty data source reserves a bandwidth for itself on a virtual channel (VC) path from the source to the destination before transmission, and releases the bandwidth for other data traffic after the transmission has been completed.

In wide area networks, for example, LAN interconnection for remote LANs, however, bursts may suffer from high blocking ratios in FRP because each burst must simultaneously succeed in making bandwidth reservations over all links on the path. A

reservation is blocked when just one link on the path is not available. This results in another reservation attempt after a certain backoff time. Then, large delays at the source terminals and throughput degradation occur. This problem becomes serious as the number of links on the path increases. One study, [31], shows that the multipath scheme improves the blocking probability, if terminals manage many different paths to the same destination.

To solve blocking problems based on an ATM single path, there are two approaches that provide feasible solutions. One is to reduce the bandwidth used for each burst transfer in order to get smaller burst blocking probabilities [32]. This approach, however, may result in large transfer delays at the source terminals. The other approach is to reduce the number of links which must be reserved simultaneously.

This thesis proposes a "**Burstserver**" architecture based on the latter approach, which reduces the number of links reserved simultaneously by dividing all the links on the path into small sets, and obtains higher network throughput due to low reservation blocking ratios.

3.2 Characteristics of the bandwidth reservation method

3.2.1 Blocking in burst bandwidth reservation

Let us focus on bursty data transfers such as, Internet Protocol (IP) packet transfers and Ethernet packet transfers between remote LANs. These packets in ATM networks are delimited into a series of ATM cells, which is defined as a "**burst**". In addition, a series of cells are called a burst, if all the cells go to the same destination via

the same ATM connection. The bursts defined here are assumed to require high peak rates for ATM transfer.

In FRP [30] or the burst bandwidth reservation protocol [31], bursts are sent after they succeed in reserving a bandwidth along the established connection path. Each source/destination pair establishes the virtual channel (VC) connection with no bandwidth before generating a burst because it takes much more time to set up a connection than to reserve a bandwidth. When a burst is generated, it has to reserve necessary bandwidths in all the links on the path. The reserved bandwidth is released after the burst is sent. Each burst is identified by a beginning-of-burst cell and an end-of-burst cell, which are detected by looking at the ATM cell headers [35].

Reservation blocking, which causes delays because of successive reservation attempts after a certain backoff time can occur frequently as the number of links on a path increases. In such cases, simultaneous link reservations can be difficult as shown in Fig. 3-1. The figure shows a communication diagram for sending a burst to the destination via Link #1, #2 and #3. Suppose that Link #1, #2 and #3 have traffic streams independent of each other. Then, busy periods occur randomly on each link. In (a), the first several reservations are denied by any one of the three links. It is so difficult to reserve all three links simultaneously that the burst suffers many reservation attempts and an extended waiting delay in order to complete the bandwidth reservation.

3.2.2 Store and forward burst transfer

One reasonable solution is to logically divide the network look into smaller parts in order to avoid reserving many links simultaneously. At the first attempt in Fig. 3-1 (a) and (b), the generated burst tries to reserve bandwidths over all Links #1, #2 and #3. Then, Link #2 rejects the reservation. If the burst goes first only to Link #1, the next

attempt could start at Link #2, and would try to reserve bandwidth only for Link #2 and #3 regardless of Link #1. Figure 1 (b) shows reservations made in such a manner, i.e. the burst is sent first on Link #1, and is stored at a relay station between Link #1 and Link #2, then the next attempt arises for link #2 and so on. This transfer method seems to intuitively solve the simultaneous reservation problem. In fact, delay quality is drastically improved in Fig. 3-1 (b) as compared with (a).

In order to realize this procedure and implement a relay station such as depicted in Fig. 3-1 (b), the ATM **burstserver** architecture is proposed, which introduces a burst store-and-forward mechanism into the ATM network. It is necessary to consider the amount of storage space for bursts because the cell buffer capacity in current ATM nodes is not large enough for this purpose. In addition, a function which performs a burst bandwidth reservation protocol must be considered. The reservation protocol tries to reserve the rest of the links on the path for the stored burst in the same way as the source terminal does. It is a useful option to reserve B-server buffer storage as well as link bandwidth in order to avoid buffer overflow.

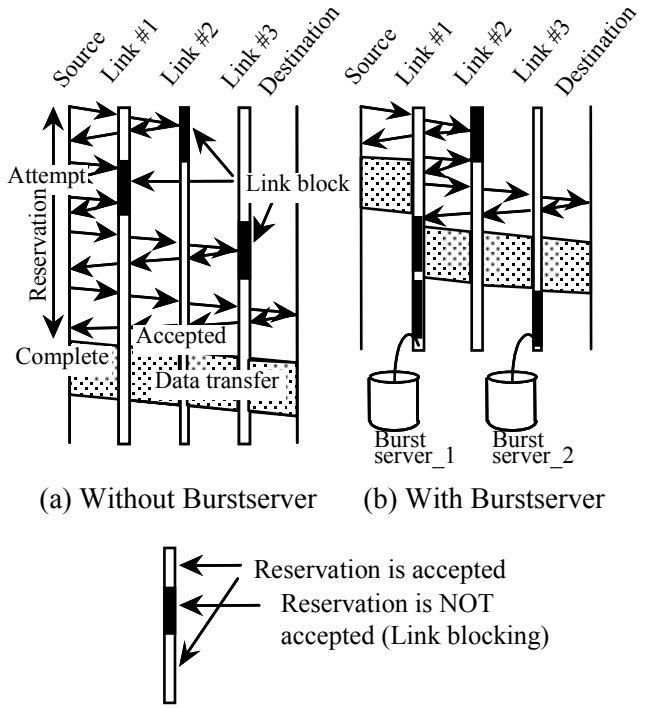


Fig. 3-1 Diagram with/without burstserver

3.3 Burstserver

3.3.1 Basic functions

A burstserver is composed of a large capacity cell storage unit, which is able to read/write cells at the same speed of the link where the burstserver is attached, and a bandwidth reservation protocol function. A sample network configuration is illustrated in Fig. 3-2. Burstservers (referred to simply as **B-servers** from now on) may be placed not at all the nodes but at the appropriate nodes.

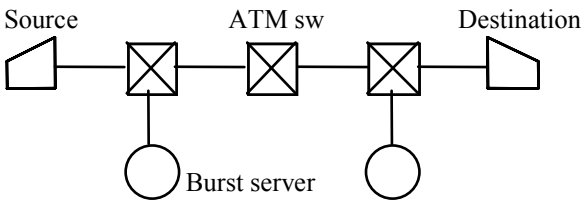


Fig. 3-2 Network structure

In the data transfer phase, the burst transfer protocol treats a series of cells as a burst by detecting both a beginning-of-burst cell and an end-of-burst cell identifier as well as the VC identifier (VCI) in the ATM cell header. The burst in the B-server is also treated as a series of cells. The bursts are separately stored into the corresponding areas in terms of VCI. No upper layer protocol, other than the ATM layer protocol and a reservation protocol, is necessary in the B-server.

3.3.2 Burstservers and connectionless servers

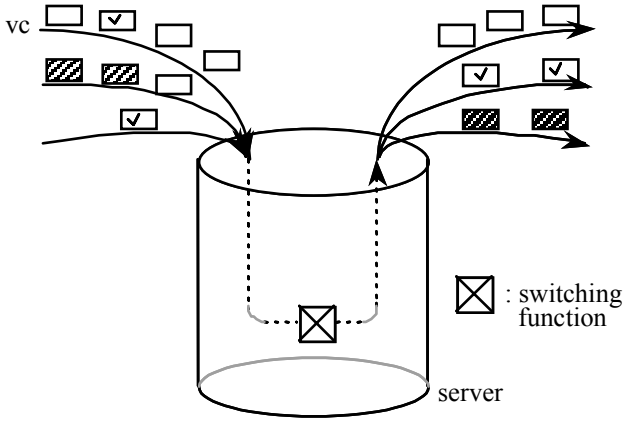
Burstservers have significant differences from so-called connectionless servers, which may be used for low speed and small amount of data rather than for high speed and bulk data.

Connectionless servers (CLS) are defined to share certain VCs among many paths. CLS data transfer protocols must switch AAL or upper layer packets to the appropriate destinations as illustrated in Fig. 3-3 (a). It is necessary to check the ATM payload for switching information. It is thus very difficult to implement CLS functions in hardware. Further, it is not possible to guarantee the requirements for quality of service (QOS) such as the cell loss ratio or cell delay.

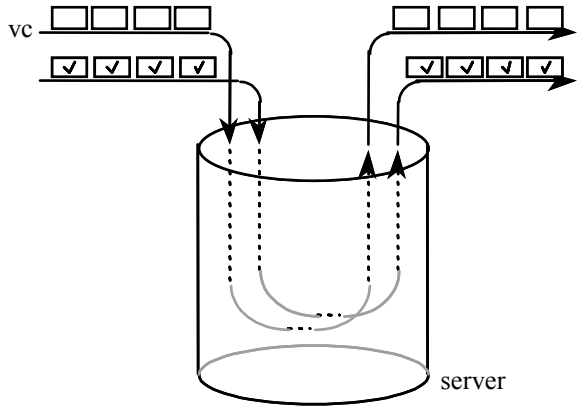
On the other hand, B-servers treat only ATM layer protocols and never take care of ATM payload. B-servers require no switching function in burst-by-burst nor in

cell-by-cell, because each B-server knows the VC path to which the burst should be sent as shown in Fig. 3-3 (b). The ATM protocol function can be easily implemented by hardware. For storage, high-speed read/write is required and a huge capacity is desirable. In practice, a serial/parallel converter can realize high-speed read/write. It is possible to use static RAMs of several tens of megabytes, a relatively inexpensive option. Low cost storage such as magnetic disc arrays can help to reduce the cost of B-servers.

Note that the B-server protocol guarantees the specified QOS, but CLS protocols can not.



(a) Connectionless server



(b) Burstserver

Fig. 3-3 Burstserver and connectionless server

3.3.3 Best effort procedure

Two feasible procedures in the reservation phase to decide usage of the burstserver are proposed.

One is the “best effort” way, where a reservation goes straight toward the destination until it is blocked. A flow chart of this method is drawn in Fig. 3-4. This means that B-servers attached to less busy links may be skipped. For example, the B-server on the right side of Fig. 3-1 (b), named server_2, is not used.

The source terminal establishes VCs for all or part of the B-servers, from the source to a B-server and from the B-server to another B-server, and so on, to the destination (See an example in Fig. 3-5). These VCs should have no bandwidth at the time of connection setup. At each input port in a switch, a VC connection table, which maps input VCIs into output VCIs of particular output switching ports, is set during VC establishment. An example of the connection table for the switch in Fig. 3-5 is shown in Fig. 3-6. In Fig. 3-6 (a), input VCI #2 is mapped either towards the next switch (output VCI #5 and output port #3) or towards the B-server (output VCI #100 of output port #7). Another entry with the input VCI #200 and output VCI #5 of output port #3 must exist at input port #7 (Fig. 3-6 (b)).

If the reservation is blocked, the reservation cell is sent back towards the source according to the flowchart in Fig.3-4. On the way back from the blocked link, the reservation cell gives switching information; indicating which of the two output VCIs in the VC connection table must be used at each ATM switch. This mapping is realized by marking a sign in the “availability” field of the corresponding output VCI in Fig. 3-6. The sign is represented by “OK”. Note that this decision should be done separately for each burst reservation.

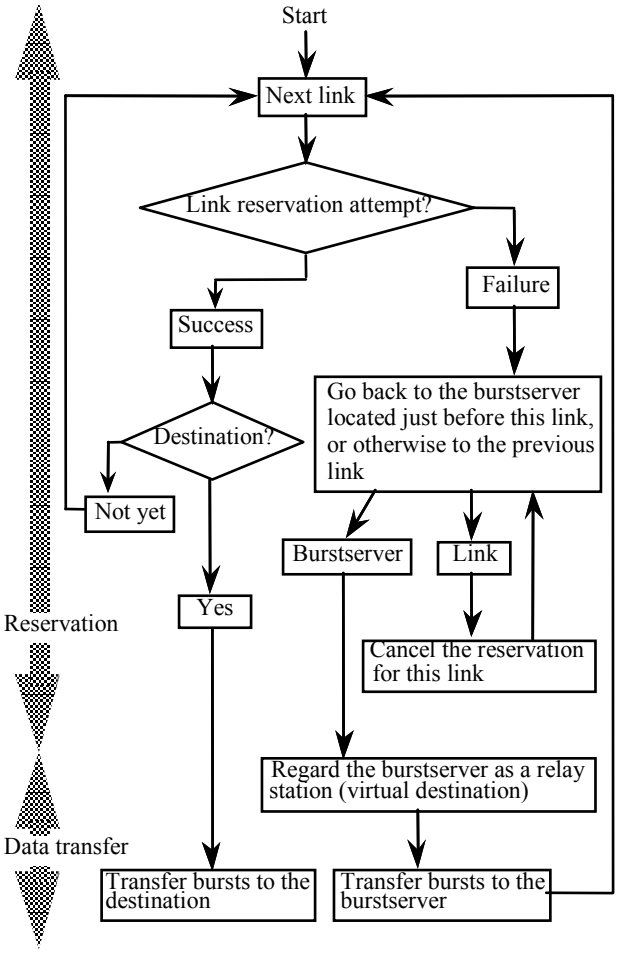


Fig. 3-4 Flowchart of procedure-1 “Best Effort”

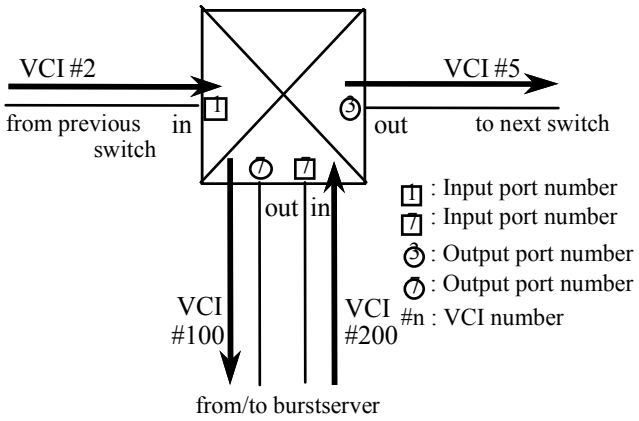


Fig. 3-5 Routing from/to burstserver

(a)

VC connection table of Input port 1.

Input VCI	Availability	Output Port	Output VCI
#2	—	#3	#5
	OK	#7	#100

(b)

VC connection table of Input port 7.

Input VCI	Availability	Output Port	Output VCI
#200	OK	#3	#5

(a) (b) Best effort procedure

(c)

VC connection table of Input port 1.

Input VCI	Availability	Output Port	Output VCI
#2	OK	#7	#100

(d)

VC connection table of Input port 7.

Input VCI	Availability	Output Port	Output VCI
#200	OK	#3	#5

(c) (d) Step-by-step procedure

Fig. 3-6 VC connection table

3.3.4 Step-by-step procedure

The second way to use the B-server is the "step-by-step" method. In this method, all B-servers along the connection setup path should be used for each burst. VCs are setup like chains from the source to a neighbor B-server, from the neighbor B-server to the next B-server, and so on to the destination, and the route is fixed identically. Figure 3-6 (c) and (d) corresponds to the VC connection table using this method. The source terminal sends a reservation to the nearest B-server regardless of the network status. If the traffic load is not heavy throughout the network, this procedure may lose the opportunity that a burst may succeed to reserve a bandwidth at all the necessary links and so may take less time to transfer the burst. This method, however, can be implemented much more simply than the best effort way.

3.4 Performance evaluation study

3.4.1 Model

Burstservers could give high-speed bursty data an improvement in performance such as throughput, delay, and fairness of them. This section is devoted to a performance evaluation based on the models described below.

To make the analysis simple, the "step-by-step" procedure (Proc-2) is employed, where every burst has to be stored and forwarded in every B-server even in low load as described in Section 3.3.3. This procedure is compared with the conventional procedure (Proc-0), where bursts can use no B-servers. A set of links between neighbor B-servers or terminals is called a "hop". For example, a system with two B-servers on the path like Fig. 3-2 has three hops. Congestion in and around B-servers is assumed to occur so rarely that overflow probability in B-server storage and the probability of contention in

the input or output of B-servers are negligible in the analysis. Queueing delays of data and reservation cells in switch buffers are negligible since high-speed links, say, 150 Mbps, and peak rate bandwidth allocation, i.e. no statistical multiplexing, are assumed here. Protocol processing delays are ignored.

Burstservers are placed symmetrically in the dedicated path, which is composed of K links. The number of B-servers is denoted as S. The number of hops is $S+1$, and the number of links in one hop is $K'=K/(S+1)$. Link blocking ratio on each link B, mean delay for burst emission H, mean backoff delay Db, and propagation delay on one way Dp are defined. Bursts are homogeneously generated randomly and each has its peak rate of $1/m$, which is denoted as the value normalized by the link speed. The maximum number of bursts which can be multiplexed in one link is m. Each link is assumed to have identical and independent load ρ.

First, the elapsed delay is defined as the time from the first attempt to send a burst at source terminal to the finish of receiving the burst at the destination terminal. Such mean elapsed delay T for K links and S burstservers in Proc-0 and Proc-2 is calculated by Eq. 3-1. Dp is set to be zero. H is equal to the amount of a burst divided by its peak rate and is added to the elapsed delay in each time when bursts go out from the B-server or from the source terminal. W is mean waiting time before success reservation in one hop and is calculated by Eq. 3-2 from the Bernoulli trial model. It is assumed that a burst is blocked in the probability G in each hop, and not blocked in $1-G$. Such G is defined in Eq. 3-3 since each link is assumed to behave identically and independently. Equation 3-4 is an Erlang B formula.

$$T=(W+H)\cdot(S+1) \tag{3-1}$$

$$W=Db\cdot G/(1-G) \tag{3-2}$$

$$G=1-(1-B)^{K/(S+1)} \tag{3-3}$$

$$B=\mathrm{Erl}(m,\lambda)=(\lambda^{m}/m!)/(\Sigma^{m}_{j=0}(\lambda^{j}/j!)) \tag{3-4}$$

Second, throughput $\boldsymbol{\rho}$ is calculated by Eqs. (3-4) – (3-7). Throughput degradation results from frequent retransmission due to reservation blocking. $\boldsymbol{\lambda r}$ is defined as retransmission load. When $\boldsymbol{\lambda r}$ becomes large, it grabs almost all of the load since the load $\boldsymbol{\lambda}$ is kept to be constant.

$$\lambda r=\lambda \cdot G \tag{3-5}$$

$$\rho=\lambda \cdot (1-G) \tag{3-6}$$

$$\lambda=\rho+\lambda r \tag{3-7}$$

3.4.2 Numerical results

Figure 3-7 (a)(b)(c) show throughput $\boldsymbol{\rho}$ vs. mean elapsed delay $\boldsymbol{T}$ curves for $\boldsymbol{m}$=100, 10, 5, respectively. The source/destination pair has $\boldsymbol{K}$=16 links with B-servers of $\boldsymbol{S}$ =0, 1, 3, 7 and 15 on the path. The back off time $\boldsymbol{Dp}$ is set to be five times of the emission time $\boldsymbol{H}$. In these figures, four curves of $\boldsymbol{S}$=0, 1, 3 and 7 start at left bottom in a low offered load. As the offered load increases, the curves go to the right and up, and after certain values they still go up but turn to the left. On the return to the left, most all of the offered load is blocked. Throughput degradation and large backoff delay occur. Throughput is much limited when bursts require higher speed transfer, comparing Fig. 3-7 (b)(c) with (a). Note that throughput decreases although delay increases as offered load increases.

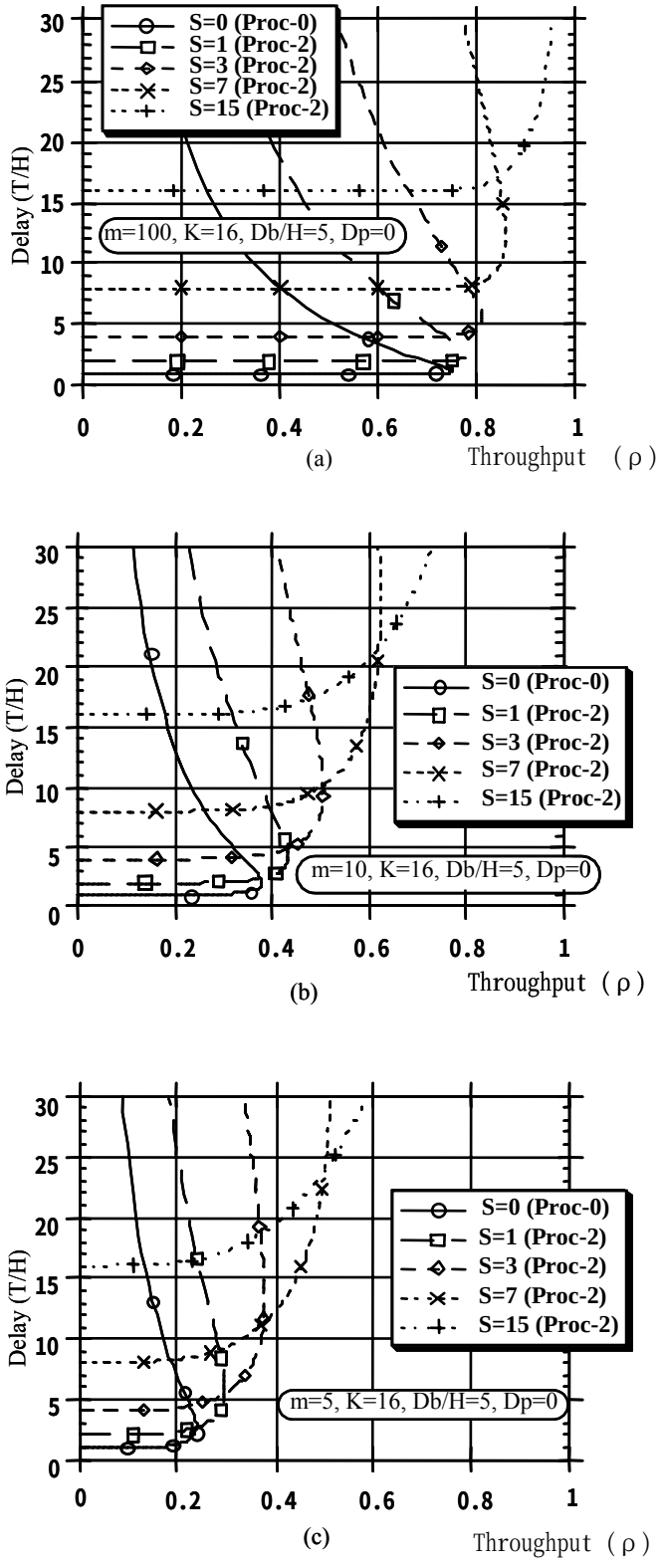


Fig. 3-7 Throughput versus delay characteristics

Since B-server architecture can reduce the number of links in one hop, it is possible to reduce blocking ratio drastically and provide high throughput. For example, for m=10 and S=1, the system can achieve a 14% improvement in throughput compared with no B-server case (S=0). For m=5 and S=1, it can get a 23% improvement. If bursts require higher peak rate, i.e. m reduces, B-servers show improvement in throughput more significantly for a high-speed burst transfer. When S=15 B-servers can be used, throughput characteristics is similar to the ideal one, that is, throughput reaches 1.0.

Figure 3-8 shows the existence of the unfairness problem lying between long and short distance data transfer by evaluating throughput vs. offered load characteristics for some K'*s*. In the figure, m=10 and there are four different hops for four different pairs of B-servers as shown in Fig. 3-9. These hops are denoted #1, #2, #3 and #4, each of which has K'=1, 2, 3 and 4 links on their hops, respectively, and the offered load to each link is kept equal by appropriate background traffic. The figure shows the disadvantages of long distance transfer. Throughput of more numbers of links is suppressed by that of less numbers of links. That is, throughput decreases in long distance traffic but increases in short distance traffic as the offered load increases. This is the reason that throughput in Fig. 3-7 is characterized by the number of B-servers. This unfairness problem becomes more serious if short distance traffic is relatively heavier than long distance traffic and if over all traffic in the network is high. The number of links K', therefore, should be small to provide high throughput for both long and short distance communications.

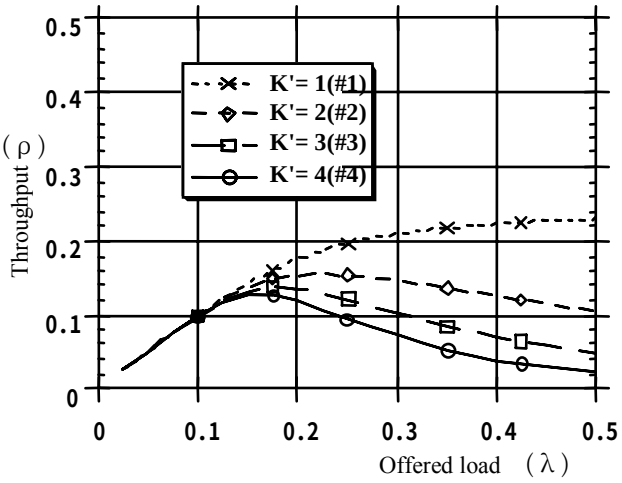


Fig. 3-8 Offered load versus throughput

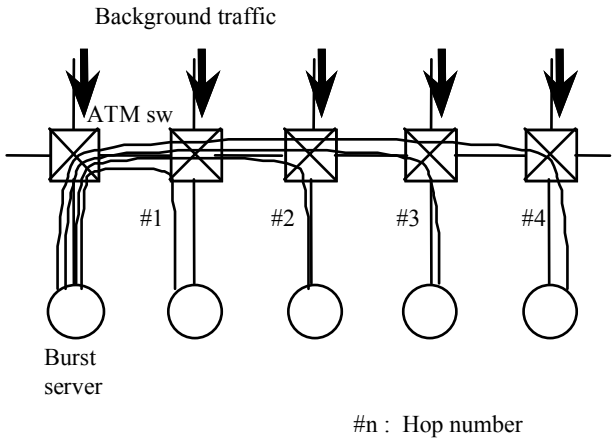


Fig. 3-9 Long hop and short hop model

Figure 3-10 shows maximum throughput vs. number of links in one hop ***K'***. The offered load is given adaptively to get the maximum throughput. In the figure, results

for 1/***m***=0.2, 0.1 and 0.01 are plotted. It shows throughput is limited by ***K'***. ***K'*** has strong effects on the throughput, especially in high peak rate burst. A desirable strategy is to introduce B-servers to maintain ***K'*** within 1 or 2 for high-speed data.

Figure 3-11 shows maximum throughput vs. peak rate. It shows that the peak rate also strongly effects throughput characteristics. Burstservers are required in order to realize higher speed data transfer by maintaining throughput in the same value. In ***K'***=2, 4, 8, and 16, high peak rate transfer, such as in range between 1/***m***=1.0 and 0.1, results in very low throughput.

In a real situation, however, the characteristics may be less pessimistic as in Fig. 3-11. Since independence assumption in the link blocking is employed in this analysis, blocking of high peak rate bursts depend more strongly on the neighbor link blocking than does blocking of low peak rate bursts.

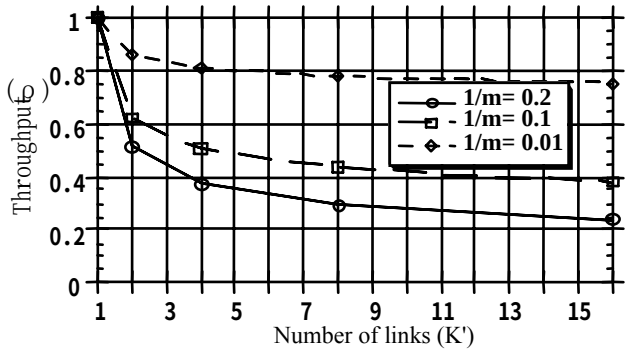


Fig. 3-10 Number of links versus throughput

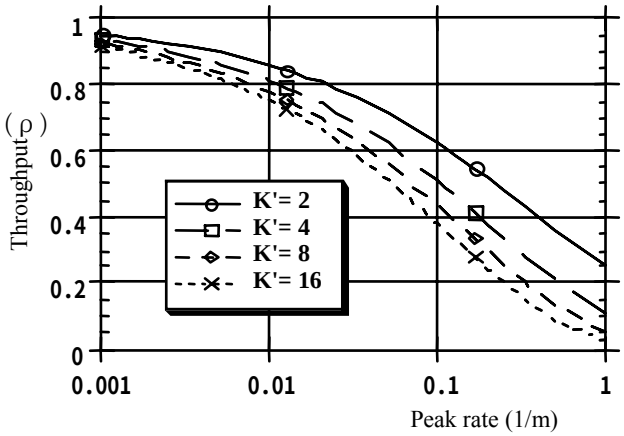


Fig. 3-11 Peak rate versus throughput

Burstservers are desirable for high throughput although mean elapsed delay ***T*** is larger in Proc-2 than in Proc-0 in low traffic loads in Fig. 3-7, which is because every burst in Proc-2 must stay at every B-server. For example, ***S***=15 model in Fig. 3-7, ***T***/***H*** at minimum must be 16. The Best effort procedure, Proc-1, also has great advantages over Proc-0, and is expected to be much better than Proc-2 in both delay and throughput. Since Proc-1 in low load skips B-servers in nature if they are not necessary, both delay and throughput performance are supposed to be better than the corresponding ones shown in Fig. 3-7.

In the above analysis, propagation delay ***Dp*** is set at zero for the sake of simplicity. It is possible to see throughput go down more if ***Dp*** becomes significant compared with the burst emission delay. In such a situation, the B-server architecture is more useful to reduce the reservation time.

As a conclusion in the performance evaluation, if the network has many links or spreads in a wide area, the network is likely to have the unfairness problem and QOS

degradation, which results in smaller throughput. Implementation of B-servers is one good solution to improve the unfairness problem and throughput degradation.

3.5 Conclusion

The burstserver architecture is proposed for high-speed bursty data transfer in order to improve QOS and throughput for large networks or wide area networks. Because connection/call level admission control could not achieve statistical multiplexing gain with high-speed bursty data multiplexing, burst-by-burst admission control (FRP) is required for such bursty data.

In FRP, success of simultaneous link reservation is too difficult in burst-by-burst bandwidth reservation. This results in delay and throughput degradation. For long distance traffic, for example, in ATMLAN interconnection, the degradation can be serious.

First, the proposed burstserver can achieve higher throughput by decreasing the number of links simultaneously reserved. Second, it can reduce the unfairness problem in which long distance communications suffer a disadvantage against short distance communications. The burstserver provides a good solution to improving the unfairness problem and maximizing throughput in order to enhance high-speed bursty data transfer.

CHAPTER 4
PROACTIVE CACHING NETWORK ARCHITECTURE

4.1 Introduction

These are the efforts taken to guarantee QOS on packet networks, and the author are sorry to say that the efforts have not achieved much success until now. Considering the situation the Internet, a network developed as best effort network focused on QOS improvements rather than QOS guarantees. In addition to this improvement of both packet level QOS and user level QOS such as throughput and contents retrieval time must be achieved, instead of packet level QOS guarantee.

Since Web must be a today's most popular application, it is reasonable to focus Web application as to improve user level QOS. In new services such as shopping and video transmission, just "connecting" to the Internet is not acceptable for today's users; it is essential to connect with "some degree of quality." For example, according to [36], a user will leave a site if it is not displayed within 8 seconds. That is, in terms of application level QOS, web QOS can be measured with the latency for the web contents shown up on a user browser. Considering such statistics and user behavior, delays in the network or server are very serious problems for providers involved in activities such as e-commerce.

Delays that are usually felt by users are the sum of the delays in the network and server, which are due to problems in the network, or problems with the server.

For the network in recent years, ISPs that provide a SLA (Service Level Agreement) in their QOS, and/or guaranteed Differentiated Service (DS) in their QOS

have emerged. However, quantization and evaluation methods of SLA are still not confirmed, and a DS only guarantees the QOS pipe between ground that was previously set as a SLA in a single ISP, so it is not realistic to access the WWW servers around the world.

On the other hand, on the server side, attempts are made to rectify delays such as applying mirroring and caching techniques, and distributing the access traffic in order to reduce load so as to deal with simultaneous multiple accessing. First though, mirroring needs to create a copy in advance. So when access can be comprehensively predicted, it is effective, but when it cannot be, it is difficult to realize because of cost problems. Furthermore, mirroring is a technique that only makes contents on a specific server high-speed, and it does not satisfy all client users.

Caching is the attractive method to reduce load because it can in nature distribute contents that are most popular (centralized access) at a specific time. Present cache servers are used mainly as forward caches with a proxy server, or as a reverse cache preliminary of a server farm. However, a forward cache is said to have trouble with a low hit ratio, and a reverse cache can reduce server load but does not connect network traffic, and does not relate to an improved response during congestion on the net.

Dealing with the network side, as noted above, and the server side separately, this can not bring benefits to client users. So in this section, as shown in Figure 4-1, by caching within the network, we aim to reduce traffic load on the network while simultaneously reducing the access load of the server, and offer possibilities for a user to improve total response.

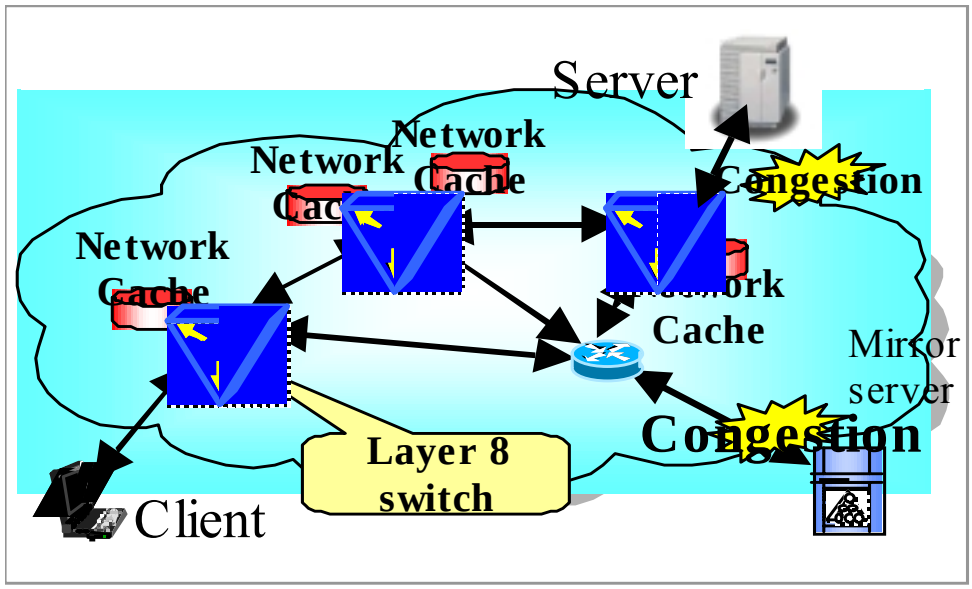


Fig. 4-1 Proactive caching network architecture

4.2 Load balancing technologies

4.2.1 Load balancing network and redirect method

The load balance network in real is configured with a switch that can process the protocol at the application level and re-direct operations (Figure 4-2).

As Figure 4-3 shows, the switch to analyze a URL by re-direct, in other words, snatch TCP flow of the HTTP, and judge if the cache server has contents with even a small possibility and work to turn the flow to the cache server, is called a Layer 5 or Layer 7 Switch [37]. (This kind of caching is called transparent caching.) If the cache server responds with a mis-hit, it will perform a proxy operation.

In this case, a delay by the TCP connection relay may be a problem. A Layer 7 switch must be a TCP relay switch that terminates the TCP once for URL analysis and from the switch resets the cache server, then the cache server sets the TCP to the destination (Figure 4-2); this creates both a communication processing overhead and a bigger delay.

In other words, problems are caused by the combination of a Layer 7 switch and the cache server unable to control delays caused by an increase in the number of TCP relays and pre-fetch traffic, which will be described in detail later. Correspondingly, in Section 4.3.3, a new switch to solve this problem will be proposed.

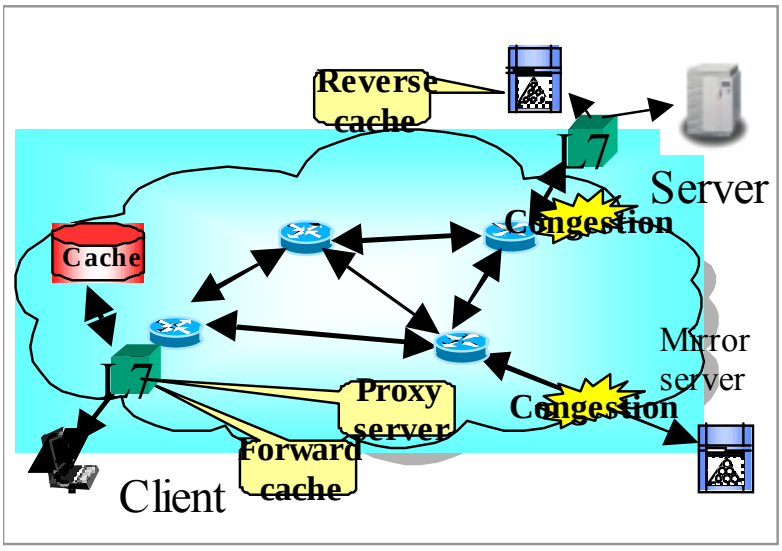


Fig. 4-2　Load balance and redirect

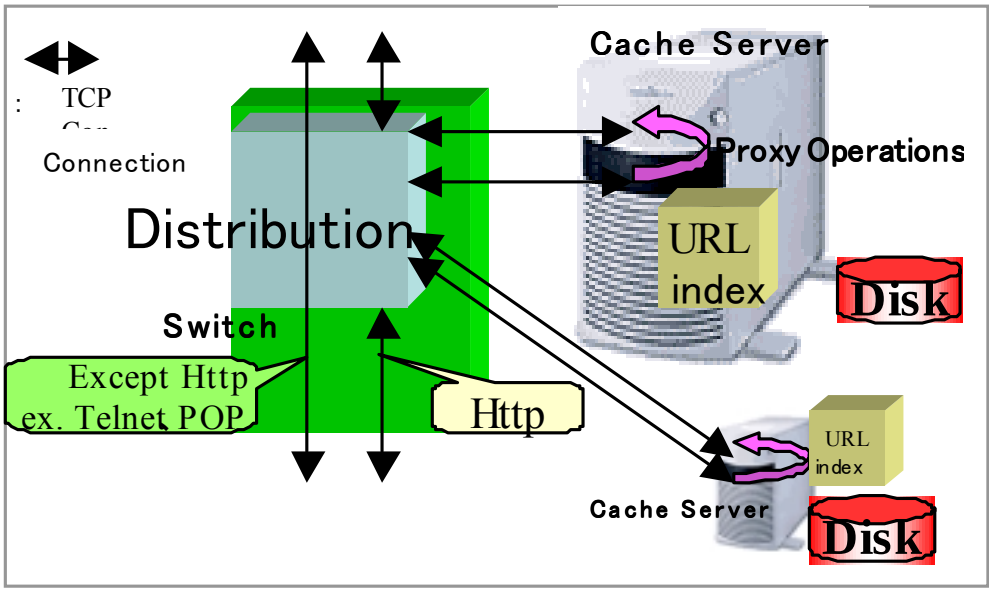


Fig. 4-3　Redirect

4.2.2 Mirroring and caching

Originally, mirroring was used as method to provide contents while avoiding overload to the server and net congestion, Digital Island [38], Akamai [39] are famous services. Mirroring avoids an increase of traffic caused by accessing servers in the far, by locating a copy of the content description near the user, and using a DNS operation that will be described later. The “Near” and ”Far” used here are defined as follows. “Near” can better satisfy conditions in a shorter response time. For example, it can be defined as near if the server 3 hops away on a 150M bps line has a shorter response time than a server 2 hops away on a 64K bps line. Mirroring by distribution has a very high load reduction effect, and an improved response time can be expected by decentralizing the servers with a copy near the users.

However, mirroring is not possible when people flock to contents that have not been predicted on the server side. For example, when access is over a short period or during a sudden burst such as when a popular website introduces daily news, TV, or a bulletin board, it is almost impossible to predict which contents to mirror. And in P2P (Peer to Peer) transmissions as are being discussed by napstar [40], gnutella [41] etc, you cannot predict which servant (server and client) the traffic will focus on at all. It is also impossible to balance the load by mirroring.

On the other hand, caching can reduce extreme loads that occur suddenly, as noted above, because it automatically decentralizes the arrangement of the most popular contents, in other words, where access is concentrated. First, we will categorize the original caching method, and evaluate on its cons and pros.

4.2.3 Caching method

Caches can be functionally categorized into two - the forward cache and the reverse cache.

The reverse cache is mainly located as the preliminary step of a server group such as server farm etc, and the access request is backlogged just before server (Figure 4-2). It can effectively use the CPU resources to retrieve that static contents such as un-animated images, text, etc., that are processed by the cache server, and process script-type, dynamic contents etc., which are impossible to cache, on the server. In such cases, it is common to locate them on the server side between the network and access link server because there is no advantage for the net side, even though it is at the network edge. Thus, it can reduce traffic to server in itself, but does not attempt to reduce traffic of the link connecting the network and server. Furthermore, its weakness is that it cannot avoid a deteriorating response if congestion occurs at the access link.

On the other hand, a forward cache is mainly used on the Net edge near to the user, mainly through a proxy server or transparent cache as shown in Figure 4-2. Forward caching is usually located within the Net to reduce traffic in the network as it reduces traffic exchanged between far location servers. The response time will be very good because it is near the user when the cache is hit.

Adversely, because access goes to the original server when it isn't hit, traffic is concentrated on one part of the network as the load is focused to the original server. Furthermore, as shown in Figure 4-3, because the cache server always redirects, delays will be bigger when mis-hits are set with a relay TCP Connection. Especially on the Web it is said that the volume of most transmissions of download contents is in the several Kbytes per TCP session, which is called short-lived traffic [42], so TCP connection delays are often at the same level as content transfers.

Thus, by reducing mis-hits is the key to improving response when using a cache. This is done by pre-fetch. Pre-fetch operation of a cache is noted below and its weak points are discussed.

4.2.4 Pre-fetch for caching

Here, conventional pre-fetch technology and its problems are discussed.

Caching is advantageous when distributing contents, but its effect determines caching performance, in other words, the hit ratio. Hit ratio is defined as ratio that the cache can answer a request. To raise the hit ratio, contents that have a high possibility of being accessed are cached. Normally, when the maximum hit ratio of cache is low, its upper statistics are 40%-60%. One way to raise the hit ratio is by pre-fetch.

There are two main ways of Pre-fetch. One is pre-fetching in order to refresh, preventing the cached contents from getting old, in expectation of their use in the future. The other is pre-fetching in order to analyze related links and foresee a request before it is actually made by a client.

According the literature [43], latency can be reduced by 45% when pre-fetching is performed by the client himself. And, latency can be reduced up to a maximum of 60% when perfect caching and pre-fetching by the cache server is assumed [44]. But since pre-fetching causes traffic, it can also be said that pre-fetching is a reverse technique by basically increasing the improved response to traffic. Generally, an increase in traffic increases the possibility of queuing delays and packet disposal by resend. Hence, for pre-fetch to be considered advantageous, traffic must be light at the start or pre-fetch must have a very high efficiency.

Thus, efforts are being made to raise pre-fetch efficiency. One method is to predict future access with higher precision. Methods like server side pre-fetch and client

side pre-fetch are employed, but accurate prediction of objects that have never been accessed before is difficult when the user is accessing new pages one after another. Raising efficiency has its limits. Furthermore, increasing unnecessary traffic on the Net may occur instead.

Especially, speculative pre-fetch raises burstiness, causes queuing in the network, and deterioration of the response [46]. To rectify this, it is necessary to control the rate of pre-fetching, and in turn reducing traffic burstiness to a level lower than that without pre-fetch. There is also the possibility of negating the effect of the cache when the client aborts access via cache server and uselessly accesses the Web server by the cache server [47]. This phenomenon can be seen when cache server performs a pre-fetch. To avoid this, pre-fetching needs to control the rate and abortion process.

In conventional techniques, controlling the rate is needed to guarantee prefetch validity, but what is essentially needed is to realize the favorable utilization of the network. This is not necessarily limited to controlling rate.

As stated above, conventional caching techniques have a negative side such as increasing traffic and consequently when trying to improve the response by pre-fetch may actually result in a worse response, but these weaknesses are caused by cache servers pre-fetching independently of the network. In the next Section, techniques that organically combine the Net and cache to solve these weaknesses are discussed.

4.3 Proactive caching network

4.3.1 Resource engineering

As stated in the previous section, in order to solve conventional problems, and to realize WWW access with a good end-to-end response, the following are needed.

1: Increase the optimum distribution of dynamic contents that use caching,

2: Perform pre-fetch caching that reflects the traffic condition of the whole network,

As a practical method to realize this, we propose a system and method that applies a caching technique within the network to distribute dynamic contents, and perform cache operation based on traffic conditions.

As a component to realize a resource engineering that uses network caching, there are cache servers and switches but each must understand traffic conditions of the network and the positional information of the original server, and use it effectively. This realistic method is called **resource engineering** because it is manages the resources of the whole network; not just the traffic engineering of network but also the engineering of its contents at the same time.

The method shown below uses network caching as a practical method of resource engineering.

4.3.2 Proactive caching

By network caching, the pro-active caching performs a cache operation that doesn't load onto the Net, but in cooperation with it, rather than performing the usual cache operation independently of the Net. Some methods are described below.

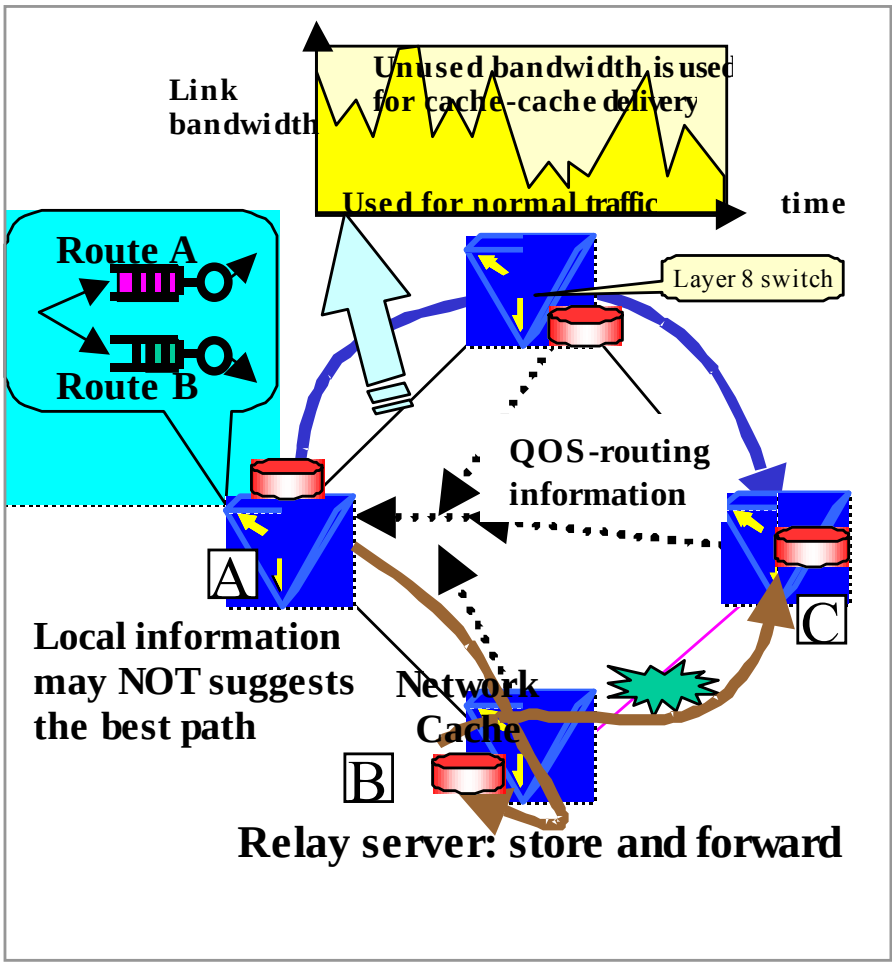


Fig. 4-4 Proactive caching

[Route/Server Selection]

When a mis-hit, or pre-fetch operation is performed, instead of pre-fetching the fixed upstream cache server, a server is chosen from among the server group, including any mirror servers, that can download most effectively at the point in time when the pre-fetch is to be performed. The best server is determined by considering how near the server load and server are, and the degree of congestion of the pass. In conventional methods, there is a method that refers to the utilization ratio of the link and the connected cache [48], and another that uses RTT (Round Trip Time) and hop number

[49]. However, in the first case, it only looks at a part of utilization ratio of the link in the pass. And the other only receives congestion information on the pass that it is using at the time, and cannot receive the availability of other links on the net, so it is ill-informed to choose a route and server. Therefore, we propose a method to process information from the routing protocol, such as the extended OSPF, considering QOS information by the cache server or switch connected to the cache, and choosing the route or server based on this information (Figure 4-4).

[Traffic Priority]

In assuming that user response time should be first considered, data transmission of a pre-fetch is assumed to have a lower priority when compared with data transmission of a cache mis-hit. Thus, the cache server should perform data transmission in consideration of priority. Furthermore, the problem that predictive pre-fetch raises access burstiness, causing queuing in the network, and deteriorating of the response [50] is noted, but for this problem as well, by lowering the priority of pre-fetch traffic, a reduction in burstiness as seen from other traffic can be expected as a result.

[Store and Forward by the Relay Server]

Pre-fetching does not require real-time, so the usability of whole net can increase by transmitting when load is low. When there are plural links in the pass between caches to transmit the contents, conventional real-time transmitting such as Differentiated Service, transmission starts after confirming the entire link band at the same time, and needs to have all the links on the pass vacant, but in this proposed method, transmission can be done when there is a partially vacant band in the pass. The proposed method uses the cache server (Contents Relay Server) in the route as shown in

Figure 4-4. The server gathers the contents transmitted between caches temporarily, and forwards the gathered contents by checking the link condition of the forward side. For example, when there is the relay server (B) on the route between the transmitting server (A) and the receiving server (C), the transmission of the contents from A is done by confirming a vacant band in the link between A and B; the transmitting from B to C will also be done in the same manner. It is possible to renew the cache of B at that time. In this method, transmitting can be done if there is a vacancy in the band of the link to the relay server. More effective use of the network is possible than by conventional means.

[Contents Routing]

To effectively use the network, circulating traffic according availability of the network is needed. In the expanded form of conventional routing, QOSPF (QOS-OSPF) is becoming standardized by IETF etc. But since short-lived best effort traffic is targeted here, route flapping chooses the best route based on the congestion information at that specific time, so if traffic is concentrated on one route, another route is immediately chosen. This circumventing is generally difficult. And when the cache has a mis-hit, a way to choose the server when multiple conventional servers exist is important in order to shorten the response. Although some options can be thought of, only by using the DNS, the appropriate server can be chosen by taking into account the actual conditions. For example, one method is to sort appropriate servers for a client by using its IP address as a clue for the DNS (FQDN) requests. At that point in time, the DNS server route, even without knowing the location of the client, and the traffic condition of the network around the client, static sorting is still done to a certain level no matter what the conditions of the net are. Next, in consideration of the circumventing of the route flapping, expanding the DNS function, and the network availability conditions, a protocol that responds to the FQDN dynamically is used.

4.3.3 Layer 8 switch

A switch that realizes the prediction functions (called Layer 8 Switch) is proposed to operate resource engineering used in network caching. As shown in Figure 4-5, the Layer 8 Switch functions as a cache server with pro-active functions, and the ability to terminate the TCP, extracting the URL, and forward the HTTP request.

Regarding conventional Layer 7 Switch, it was noted in Section 4.2.1 that delays caused by the increase in the number of TCP relay columns can be problematic, and Section 4.2.3 pointed out the problem of traffic increase caused by pre-fetch, but Layer 8 Switch has a cache function in the switch, and as noted in Section 4.3.2, and has solved these problems by a pro-active pre-fetch operation.

The operation of the Layer 8 Switch is as below. The cache server function within Layer 8 Switch exchanges content information with other cache servers by using the ICP protocol etc, and gains topology and congestion information of the network from the QOS expanded routing protocol. Based on this information, the URL forwarding table is dynamically updated. The URL forwarding table is the table that is displayed to which the HTTP request for each URL should be forwarded, contains expired information etc. of each URL, and pro-actively operates pre-fetch, etc., using congestion information. Of course, it can easily access to the cache contents in its own switch by memory reference etc., within the equipment without using TCP, reducing the number of TCP relay columns, and possibly improving the response.

Regarding the architecture of the Layer 8 Switch, for example, a cache function is installed on the server card in the switch, and the forwarding function is on the line interface in the switch. Software processing is done using the processor on the server card until analysis of the URL, which was noted above, but once deciding which cache server to forward to, it can shift to high-speed transmission by the hardware in the line interface.

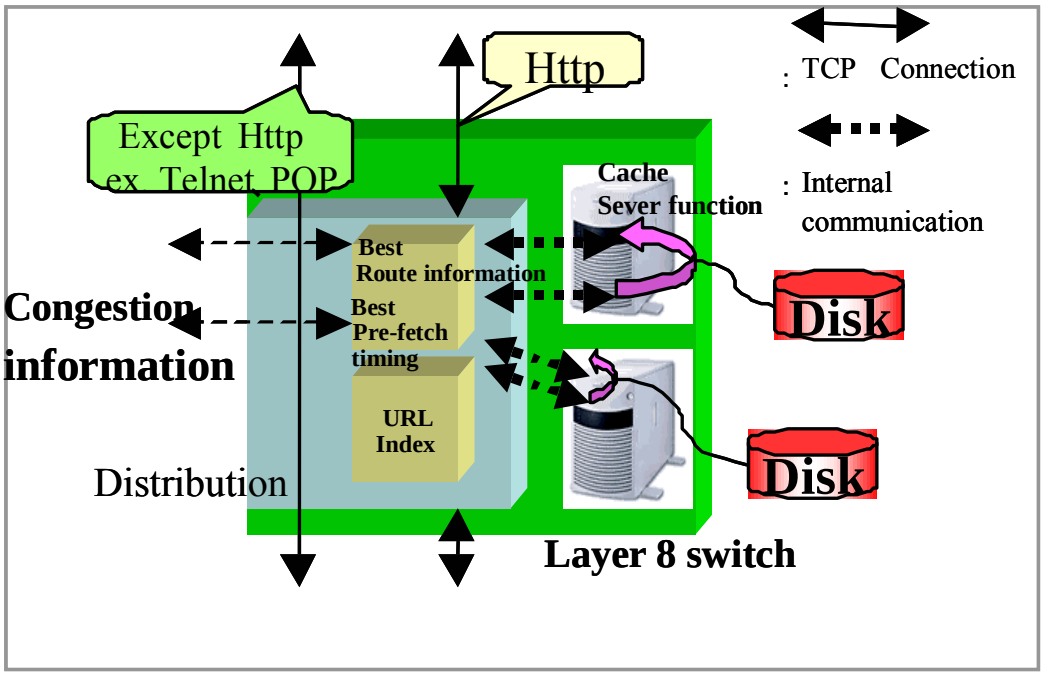


Fig. 4-5 Layer 8 switch

4.4 Simulation results

4.4.1 Simulation model

Proposed method is here evaluated by computer simulation. Because proposed pre-fetch caching is supposed to be done only when networks are not congested, both low miss-hit ratio and low latency for miss-hit retrieval due to well pre-fetching and less congestion in network, respectively, are expected to be achieved. Three models are compared; (1) no pre-fetch: neither user nor network does pre-fetch, (2) greedy pre-fetch: user or network pre-fetch regardless of network congestion, and (3) proposed pre-fetch with traffic smoothing: pre-fetch is only performed when network is not congested. Since it is difficult to construct a general model of Web traffic, a set of real measured traffic data is used as a simulation input. The data set was originated from NEC laboratory with access logs in the proxy cache server, and was collected in the period from 2003/3/1 to 2003/3/7.

User response time is set as a performance measure. We can vary bottleneck link bandwidth. From the cache log, throughput of a contents retrieval is calculated from file size divided by processing time if the access results in cache-miss. Each Web page has five links in average with binominal probability distribution. This number of links was derived from a calculation where volume of waste pre-fetch traffic must be twice as large as real used traffic. The scheduling of (3) is controlled with keeping throughput of the pre-fetched traffic is controlled keeping under a certain fraction of the link bandwidth as shown in Fig. 4-6.

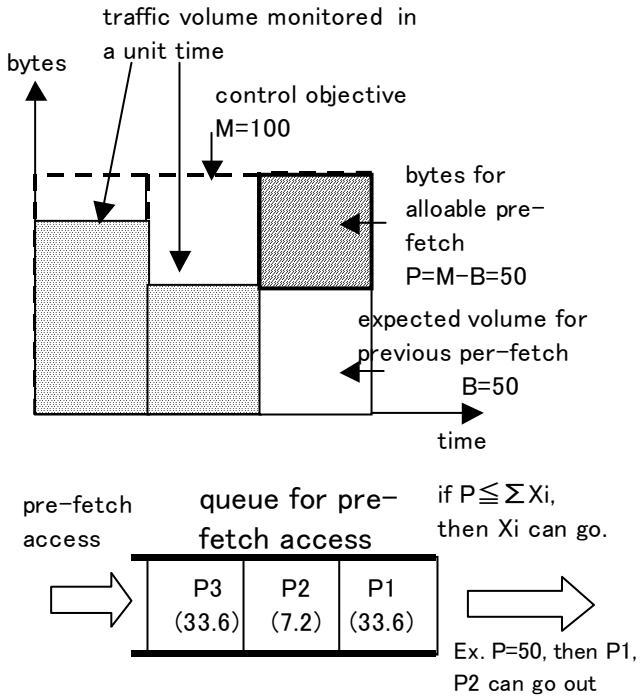


Fig. 4-6 Proposed pre-fetch scheduling

4.4.2 Results and discussions

The simulation results show that good user response time can be achieved without network congestion. Table 1 and Fig. 6 shows user response time between three algorithms. In Table 4-1 and Fig. 4-7, (1) no pre-fetch, (2) greedy pre-fetch, and (3) proposed pre-fetch with traffic smoothing. Bottleneck link bandwidth varies from 100% to 3.3%. As the value decreases, response time increases. Note that (2) greedy pre-fetch has poor result when bandwidth is small because the mechanism causes congestion by means of its pre-fetch traffic. The reason is proved in Table 4-2 and Fig. 4-8. In the table, cache hit ratio is shown and both models (2) and (3) have done effective

pre-fetching. In this simulation, the scheduling of (3) is controlled with keeping throughput of the pre-fetched traffic is controlled keeping under 40% of the link bandwidth.

As shown in Fig. 4-7, proactive caching model achieve about 40% better response time compared with no pre-fetch and greedy pre-fetch model.

Table 4-1 Response time (sec)

Link bandwidth	3.3%	10.0%	16.7%	100%
(1) no pre-fetch	6.69	1.97	1.48	1.15
(2) greedy pre-fetch	10.46	1.95	1.32	0.97
(3) pre-fetch with scheduling	6.44	1.71	1.24	0.97

Table 4-2 Hit ratio (%)

Link bandwidth	3.3%	10.0%	16.7%	100%
(1) no pre-fetch	0.53	0.53	0.53	0.53
(2) greedy pre-fetch	0.66	0.70	0.71	0.72
(3) pre-fetch with scheduling	0.59	0.67	0.69	0.72

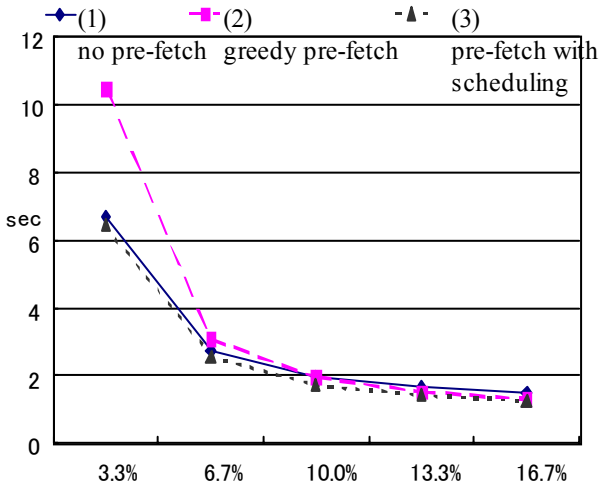


Fig. 4-7　User response time

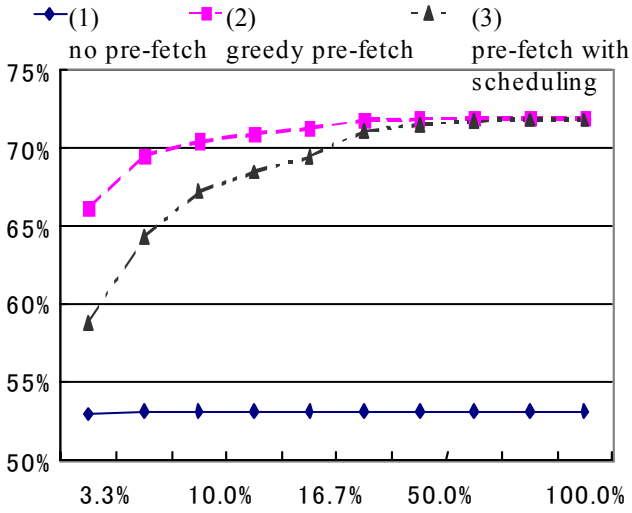


Fig. 4-8　Cache hit ratio

4.5 Conclusion

Considering the situation the Internet, a network developed as best effort network focused on QOS improvements rather than QOS guarantees. In addition to this improvement of both packet level QOS and user level QOS such as throughput and contents retrieval time must be achieved, instead of packet level QOS guarantee.

Since Web must be a today's most popular application, it is reasonable to focus Web application as to improve user level QOS. Web QOS can be measured with the latency for the web contents shown up on a user browser. That is, delays that are usually felt by users are the sum of the delays in the network and server, which are due to problems in the network, or problems with the server.

A method to locate a cache in the network and a network architecture that uses a pro-active cache switch that can balance traffic control in the net and caching is proposed. By the proposed method, improving the response time not only of the specific servers but also the entire WWW transmission in the network is possible.

The aim of pro-active cache network is a dynamic autonomous load balance, and the effective use of the network by considering the circulation of contents at a specific time, in other words, resource engineering. In this thesis, we introduced a framework for a resource engineering method and a partial actual method. Then, we evaluated a pre-fetch model as a proactive caching instance, and showed that the proposed pre-fetch with traffic smoothing; pre-fetch is only performed when network is not congested, can achieve both low miss-hit ratio and low latency for miss-hit retrieval. This reveals that use response time can be 40% smaller than that of conventional models.

In conclusion, the author would like to express our appreciation to colleagues in NEC for their helpful suggestions during this research.

CHAPTER 5
TCP OVERLAY NETWORK ARCHITECTURE AND TCP CONTROL

5.1 Introduction

The quality of the "best-effort" class internet service has become increasingly less expensive. Yet, quality control that is cost effective for particular customers who desire high quality is now in need. This thesis proposes TCP-based overlay network architecture, and discusses the possibility of new services.

With the spread of connection environments such as ADSL, optical fibers, and FTTH, today's networks are able to maintain a user-side communication bandwidth of a few megabits per second to several tens of megabits per second. Nevertheless, in actual communication, the communication bandwidth cannot be sufficiently utilized. One of the reasons for this is the special characteristics of TCP communication which is dominant of the Internet traffic. Because the TCP communication throughput is mainly determined by the network RTT (Round Trip Time) and packet loss rate **[51]**, any throughput higher than that throughput cannot be achieved, even with a broadband link. An overlay network based on TCP proxy technology, thus, has been proposed as to solve this problem **[52]**.

While normal communication is achieved using a single end-to-end TCP loop, TCP proxy communication is achieved by dividing the end-to-end TCP loop into multiple TCP loops. With the TCP loop divided, each TCP loop receives the full benefit

of, for example, (1) an small RTT, and (2) a small packet loss rate, thereby improving the TCP communication throughput.

Nevertheless, the problem exists that even though the theoretical throughput of TCP proxy communication found from the RTT and packet loss rate should be multiplied that of normal communication, the expected value is not achieved by simple TCP relaying. This problem originates in the TCP rate control algorithm. The TCP algorithm, in broad terms, consists of operations that result in rapid rate drops and operations that result in gradual rate climbs. Compared to the amount of time required for the rate to drop, the amount of time required for the rate to be restored is extremely long. When the TCP proxy buffer becomes full, the upstream (receiving side of the TCP proxy) TCP transmission rate drops and, because the transmission rate cannot rapidly increase even if sufficient buffer space becomes available, TCP relay buffer overflow causes performance degradation. A full TCP reciever buffer on the side of the receiving terminal does not readily occur such buffer full with conventional communication methods that does not relay the TCP. This is because the applications that use the data normally load and process the data from the buffer at sufficiently high speeds.

In comparison, communication methods that relay the TCP involve a TCP proxy buffer that decreases in availability according to the transmission rate of the downstream (sending side of TCP proxy) TCP loop. When there is congestion downstream that causes a sharp drop in the downstream transmission rate, the loading rate from the buffer also drops, thereby increasing the possibility of overflow of the proxy's TCP receiver buffer. Due to such differences in the buffer loading mechanism, TCP proxy buffer overflow has become a serious issue with regard to its adverse effect on throughput.

This thesis takes into consideration measures that will solve this problem and, in an effort to maintain TCP friendliness, adopts a method which does not involve modifications to the TCP algorithm itself.

To summarize the above, while communication methods that relay the TCP offer better performance due to advantages such as (1) an small RTT, and (2) a small packet loss rate, when upstream degradation in throughput occurs as the result of downstream congestion caused by a full proxy receiver buffer, the theoretical rate cannot be fully achieved. This thesis proposes a measure for improving throughput while maintaining TCP friendliness.

This chapter first describes architecture of TCP overlay network, then the basic nature of TCP proxy (hereafter called TCP Bridge) communication and the problem regarding the decrease in proxy communication performance with respect to the theoretical value. Comparing conventional method, we propose a stealth buffer control method, i.e. a proxy congestion control algorithm, to solve the issue. Lastly, this chapter evaluates the proposed method through simulation and prototyping, thereby illustrating the method's effectiveness.

5.2 TCP overlay network

5.2.1 Architecture

[Problems in current network with conventional traffic control mechanism]

There have been many approaches to providing quality of service, such as the IETF standards Diffserv and IntServ. Nevertheless, these approaches are problematic since they involve layer 2 or 3 technologies, which comprehensively drive up all switch and router expenses in contradiction to the lower cost demanded by users who do not

necessarily desire high quality. While innovative methods such as active nets have also been proposed to iron out this problem, issues related to the time required for migration, for instance, still remain.

[TCP Overlay network]

This thesis proposes a network architecture that, taking into consideration realistic migration and cost effectiveness, provides enhanced quality of service by realizing the minimum quality of service features using the existing network and then constructing on top of that network an overlay network which performs TCP relay (Figure 5-1). The proposed architecture performs communication by terminating the conventional end-to-end based TCP session at an appropriate spot in the network and then reestablishing connection. The flow relayed in the network at the time is based on a flow control format that reflects the quality demanded.

[Advantages of TCP Overlay network]

By implementing window flow control on the side of the network, the proposed method makes it possible to provide fair-share bandwidth services, a larger number of quality control services for services that, for instance, maintain bandwidths, and improvements in network reliability, such as illegal flow detection and elimination

The above-described effects can be achieved by implementing specific TCP flow control at the relay point of a flow that passes through a congested node, thereby eliminating the need for direct control using the device such as an ether switch or router that is the congested node. Consequently, the advantage for TCP layer based control is that the cost increase that results from comprehensive replacement of all user-required router and switch functions is not necessary. That is, by facilitating deployment of required locations only and by enabling gradual increases in scale in response to service demands, this method is a practical method with superior cost performance.

Furthermore, this method does not involve any changes to the existing terminal interface, including the TCP itself, thereby enabling terminal communication after proposed network introduction using the same conventional method. Additionally, while there have speed-related problems with TCP termination to date, chips that perform TCP termination at a speed of 10 Gbps or higher have been announced in recent years, increasing the potential for realization of multiple high-speed TCP termination within the network.

[Differences from Conventional Overlay Networks]

The main purpose of conventional network TCP termination methods was either to provide a security service using the upper layer, i.e. the application layer, or to guarantee connectivity with existing protocols and improve link layer communication special characteristics. Conventional methods such as SOCKS proxy and HTTP proxy of the proxy system or mobile TCP gateway of the protocol conversion system were designed to actively control quality, with only a few of these methods performing TCP termination.

Designed with the purpose of active quality control, the proposed method, on the other hand, actively performs quality control for both wired and wireless systems, facilitating ISP construction of new quality services for users.

5.2.2 New services for TCP overlay network

With use of this TCP overlay network, realization of various services that were difficult cost-wise using conventional layer 2 or 3 network control can be expected. Examples of such services are described below.

(1) Quality differentiated services: (i) Though the guarantee of absolute quality similar to Diffserv/IntServ is difficult, the potential for relative quality differentiation **[54]** that is not dependent on layers 2 or 3 is possible, (ii) Small end-to-end delay due to

the short RTT and using optimum TCP between the TCP Bridge is possible. (iii) A LAN-WAN seamless service that achieves optimum throughput without terminal knowledge of the geographical location of the party terminal by relaying the TCP tuned to LAN to a wide-area VLAN.

(2) Network monitoring service: An accurate charge for volume of packets (goodput) **[53]**; Illegal TCP detection and elimination, including DOS attack countermeasures.

5.3 Congestion control in TCP Bridge

First, we focus and try to solve the problems; while communication methods that relay the TCP offer better performance due to advantages such as (1) an small RTT, and (2) a small packet loss rate, when upstream degradation in throughput occurs as the result of downstream congestion caused by a full proxy receiver buffer, the theoretical rate cannot be fully achieved.

A simple description of the TCP Bridge (TCP proxy) communication mechanism which enhances throughput is given below (Fig. 5-1).

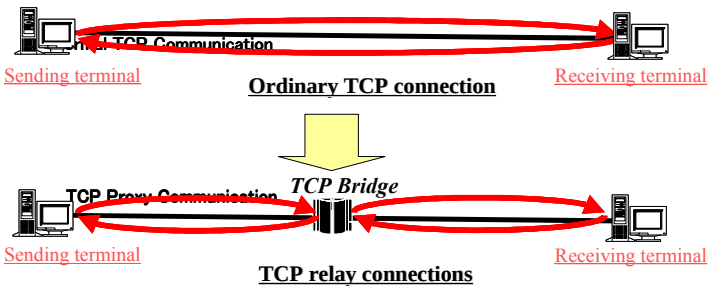


Fig. 5-1 TCP proxy communication

The upper and lower diagrams of Figure 5-1 illustrate normal TCP communication and TCP proxy communication, respectively. The TCP theoretical communication throughput ***B(p)*** is, according to reference **[53]** in the bibliography, generally found as follows:

$$B(p)=\mathrm{Min}\left(\frac{1}{RTT\sqrt{\frac{2bp}{3}}+To\min\left(1,3\sqrt{\frac{3bp}{8}}\right)p(1+32p^2)},\frac{W_{max}}{RTT}\right) \quad (5\text{-}1)$$

Where, the unit for ***B(p)*** is pps (packets per second), ***p*** is the packet loss rate, ***RTT*** is Round Trip Time, ***Wmax*** is the receiver buffer volume and To is the timeout wait time.

The left term in the above equation is predominantly determined by ***RTT*** and the packet loss rate ***p***, and the right term – which is generally referred to as the bandwidth delay product – is determined by the TCP receiver buffer capacity and ***RTT***. Here, the receiver buffer capacity is an item that can be determined by the user and, assuming that a buffer can be sufficiently prepared, throughput is determined by the right term in the equation. For this reason, when TCP is relayed as illustrated in Figure 5-1, both ***RTT*** and the packet loss rate ***p*** decrease, thereby improving throughput ***B(p)***.

5.4 Buffer control mechanism

5.4.1 Conventional method

TCP proxy communication enables improvements in throughput. Depending on the network environment, however, the problem exists that the rate does not improve to the extent of the theoretical value found from RTT and the packet loss rate. The cause of this discrepancy is related to the proxy transmission and receiving process. This section explains the cause of rate decreases due to this processing.

With TCP proxy communication, data is sent from the sending terminal and temporarily accumulates in the buffer of the next proxy. The next TCP sends the data accumulated in the proxy buffer to either the next proxy or the receiver terminal. If the receiver buffer of the proxy or receiver terminal is full, the operation described below is performed through TCP congestion management.

Figure 5-2 illustrates the operation performed when the receiver buffer is full.

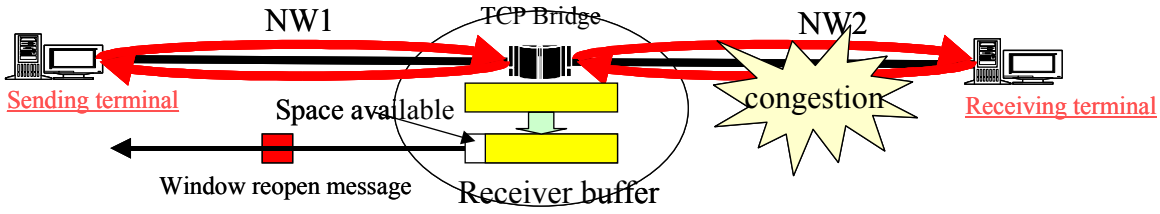


Fig. 5-2 Operation when receiver buffer is full

This operation occurs when the residual space in the TCP receiver buffer reaches zero. At this time, the terminal to receive TCP communication notifies the sending terminal of the zero status using an advertised window. The sending terminal sends the minimum (congestion window or advertised window) amount of data and stops transmission when it receives the zero advertised window.

There are two types of triggers used to restart communication. The first is a notification from the receiver terminal that communication is possible. The moment space becomes available in the TCP buffer, the receiver terminal sends a window reopen signal (advertised window ≠0 in ACK), thereby indicating that data receiving is possible. The second is a probe signal from the sending terminal. After communication stops due to the zero advertised window, the sending terminal sends 1

byte of data every 5 seconds to the receiver terminal to check the terminal status. When the sending terminal receives a window reopen signal from the receiver terminal, communication begins once again **[55]**.

Regardless of the trigger used, problems related to the TCP congestion control algorithm, such as those described below, occur.

Method 1: Receiving Side Sends One Reopen Signal

In this method, the TCP receiver side sends one window reopen signal when space in the TCP buffer becomes available. The problem, however, is that when ACK of the window reopen signal is discarded in the network, there is no means for restarting communication immediately. Even though communication does not stop completely since the sending side issues a probe signal every 5 seconds, because communication does stop for those 5 seconds, throughput deteriorates greatly.

Method 2: Receiving Side Sends Multiple Reopen Signals (BSD OS **[56]**)

In this method, the TCP receiver side sends multiple window reopen signals when space in the receiver buffer becomes available. In contrast to Method 1, this method does not result in a 5-second suspension of communication since the next window reopen signal arrives at the TCP sending side if the first reopen signal is discarded. However, when space in the receiver buffer becomes available, the receiver terminal sends an ACK of the same ID over and over again, causing the sending terminal to regard these ACKs as duplicate ACKs. The TCP sending side assumes the packet has been discarded and decreases the congestion window by half. Once the congestion window decreases, TCP cannot increase the size rapidly even when space becomes available in the receiver buffer. As a result, throughput deteriorates.

With BSD, after notification of the zero advertised window, communication restarts using a congestion window of half the previous size.

Method 3: Receiving Side Sends Multiple Reopen Signals (Linux OS)

In this method, the resulting behavior is basically the same as Method 2. However, with Linux, the size of the congestion window at the time communication restarts depends on the amount of time communication was suspended after the sending side received the zero advertised window. With Linux, the congestion window decreases by half each time a certain amount of time passes after communication suspension. Furthermore, when communication does restart, duplicate ACKs are received, causing the congestion window to decrease by half once again. As a result, the degree of throughput deterioration is greater than that of the BSD OS.

With Linux, after notification of the zero advertised window, communication restarts using a congestion window of half the previous size or less (time-dependent factor).

In this manner, while TCP detailed behavior varies according to OS, when the receiver buffer becomes full TCP throughput deteriorates to half or less.

The frequency at which the receiver buffer becomes full when proxy communication is not performed is high since the rate at which the data in the receiver buffer decreases is dependent upon the application speed of the terminal. However, when proxy communication is performed, the data in the receiver buffer decreases at a rate that is more dependent on the bottleneck rate of the destination network than that of the proxy. Consequently, the frequency at which the receiver buffer becomes full is

greater with TCP proxy communication, and this high frequency is conceivably the cause of rate deterioration.

To resolve the receiver buffer problems described in the previous section, i.e. to prevent decreases in throughput, this thesis proposes a method (*stealth buffer*, hereinafter referred to as "s-buffer") which adjusts the window size to be advertised on the TCP receiver side. The purpose of this method is to decrease to the extent possible the number of times the receiver buffer overflows (becomes full).

The prerequisites of the proposed method were as follows: The proposed method was to realize improvements in performance by devising a TCP proxy buffer management method without tinkering with the sending terminal or receiver terminal. To this end, one of the advantages of TCP proxy communication is that throughput improvement can be realized by simply inserting a proxy in the network without modifying the terminals. The proposed method, therefore, was to involve changes to the TCP proxy buffer management method only without requiring any user changes.

In conventional methods, the sending side is notified of the total receiver buffer space capacity using an advertised window (Figure 5-3). With this type of management method, the receiver buffer instantly becomes full when the congestion window opens up. When the receiver window becomes full, the congestion window decreases to half its original size or less. With TCP, once the congestion window decreases due to the control algorithm, the congestion window cannot be restored rapidly even when the amount of receiver buffer space and, consequently, the advertised window size increase. This causes the transmission rate to decrease.

5.4.2 Proposed method

To avoid the problems of conventional methods, the proposed method eliminates the instantaneous filling of the receiver buffer by using an advertised window value that

is smaller than the conventional value (Figure 5-3). By reducing the size of the advertised window in this manner, the method reduces the number of times a zero advertised window is sent to the TCP sending terminal, thereby preventing congestion window decreases.

Figure 5-3 shows in graph format an example of the advertised windows for both the proposed method and convention method when the maximum value of the receiver buffer is 60MSS. The horizontal axis and vertical axis in the graph indicate the amount of receiver buffer used and the advertised window value at that time (solid line: conventional method, dashed line: proposed method), respectively. The unit of measurement for both the vertical and horizontal axes is MSS (Maximum Segment Size).

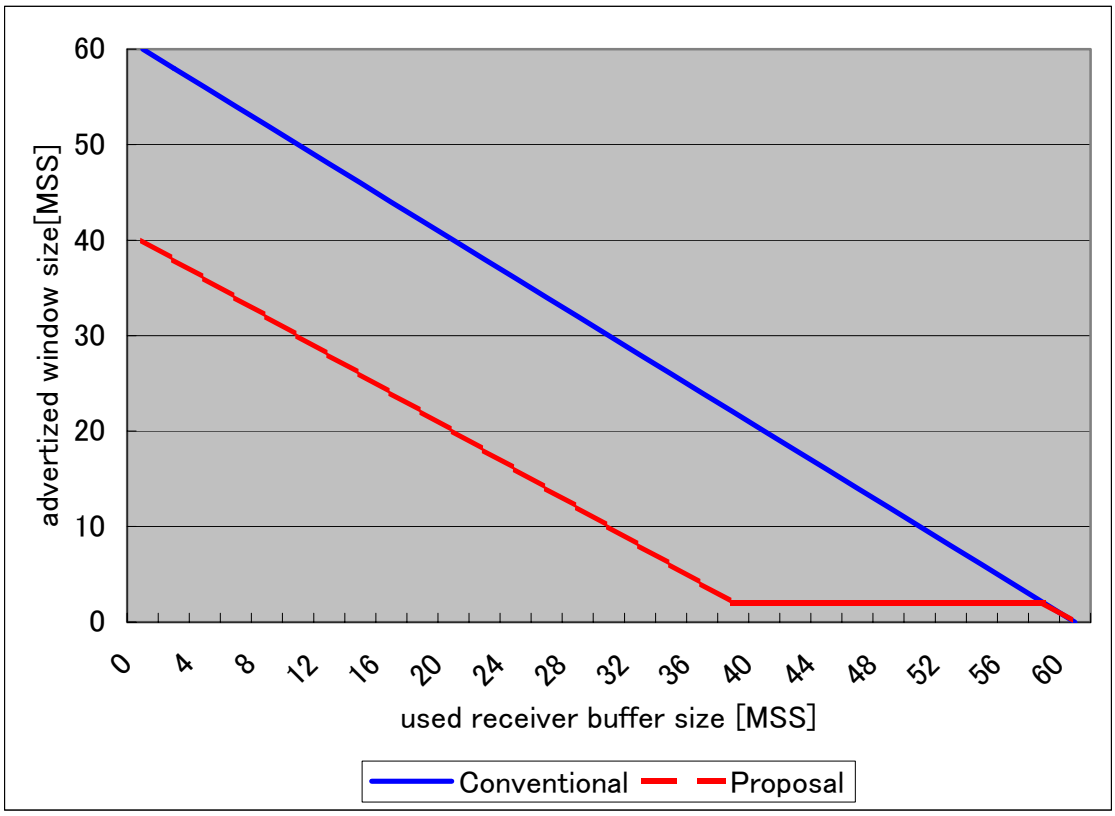


Fig. 5-3 Advertised window values in proposed method and conventional method

In the proposed method, the value assigned to the advertised window is that described below.

First, threshold ***X*** is set, parameter ***Y*** is defined and the receiver buffer space capacity ***Z*** is assigned.

When $Z > (X+Y)$:

Advertised window = $Z - X$

When $Z < (X+Y)$ and $Z > Y$:

Advertised window $=Y$

When $Z < Y$:

Advertised window =0

5.3.3 Comparison

The purpose of this method is to suppress the decrease in throughput on the TCP proxy receiver side. The proposed method reduces the number of times the receiver buffer becomes full, thereby preventing the rate of the TCP loop on the data sending side from decreasing to a value lower than that of the proxy. That is, the proposed method is effective when there is a bottleneck between the data sending side and proxy that causes the receiver buffer to become full frequently, i.e. when a packet loss rate exists at a certain level.

Conventional methods always indicate the maximum value in the advertised window, resulting in the risk of a decrease in the congestion window on the sending side. Conversely, the proposed method performs communication by temporarily decreasing the size of the advertised window, but without decreasing the size of the congestion window. Although an instant increase in the advertised window size is possible, increasing the size of the congestion window takes time. The proposed method, which uses the advertised window to control the rate and prevent decreases in the congestion window size, is more capable of maintaining a high communication rate.

Table 5-1 summarizes the characteristics of the conventional method and the proposed method.

Table 5-1 Conventional Method and Proposed Method Characteristics

	Conventional Method	Proposed Method
Advertised window	Indicates max	Less than max
Congestion window	Easily reduced	Easily maintained

The proposed method involves two parameters: the receiver buffer threshold ***X*** and the amount ***Y*** that is communicated when the receiver buffer space decreases to a value less than or equal to the threshold.

The threshold ***X*** affects the apparent total buffer amount. This is because when the receiver buffer is empty, the value notified is equal to "Reception buffer amount – ***X***." When this value is less than or equal to the bandwidth delay product, performance peaks at that value. For this reason, the threshold set should be a value greater than or equal to the bandwidth delay product.

The parameter ***Y*** is the amount notified when the amount of space in the receiver buffer has decreased. The value set should be greater than or equal to 3MSS, and should be higher for networks with a higher packet loss rate. If the value is set to 2MSS, for example, when one or more packets are lost during communication when the buffer size is less than or equal to the threshold value, a timeout error occurs. Normally, even if a packet is lost, if the same ACK number subsequently arrives three times in a row, the "fast retransmit" function is activated and retransmission is performed before the timeout occurs. However, when parameter ***Y*** is set as 2MSS, duplicate ACKs are not received three times in a row, even after a single packet loss, thereby resulting in non-activation of the retransmit function and the system waiting for the timeout. Thus, it is advantageous to set a value at a level greater than or equal to 3MSS.

Additionally, the amount of time that transpires without the occurrence of buffer overflow during a period in which the receiver buffer is sluggishly decreasing is determined by ***X***/***Y***. This is because this value is equivalent to the number of times communication is performed after the buffer decreases to a value less than or equal to threshold X. This value is proportional to the number of times the proxy receiver buffer becomes full. The above is summarized as follows:

Parameter X

Taking into consideration the window size to be advertised, a smaller value is best. However, since a larger number decreases the number of times the receiver buffer becomes full, there is a trade-off.

Parameter Y

To mitigate the loss between the sending terminal and proxy, a larger value is best. However, since a smaller number decreases the number of times the receiver buffer becomes full, there is a trade-off.

5.4 Simulation results

The following section reports the results of a conventional method and proposed method performance comparison study that was conducted using Network Simulator 2 (hereinafter referred to as “ns2”)**[57]**.

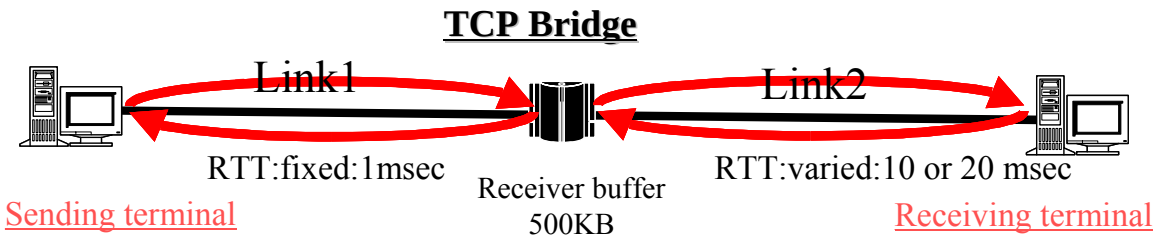


Fig. 5-4 Network model of simulation 1

Simulation 1

Simulation was first conducted to study the basic nature of the proposed method and therefore presumed a network such as that indicated in Figure 5-4. The conventional method used for comparison purposes was that of a BSD OS (the congestion window decreases by half when multiple ACKs are received after the receiver buffer becomes full).

The presumed network consisted of a single TCP proxy in end-to-end communication. Throughput was then measured while changing the Link 1 and Link 2 packet loss rates and the Link 2 RTT parameters. The results are shown in part in Figures 5-5 and 5-6.

For the proposed method, 300KB – which is larger than the bandwidth delay product – was set as the receiver buffer threshold (***X***), and 5MSS (>3MSS) as parameter ***Y***.

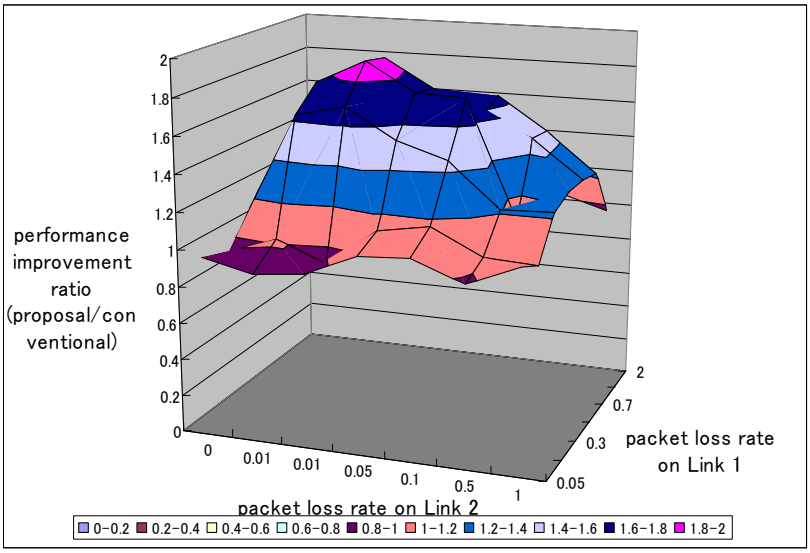


Fig. 5-5　Performance improvement: RTT of link 2 is 10ms

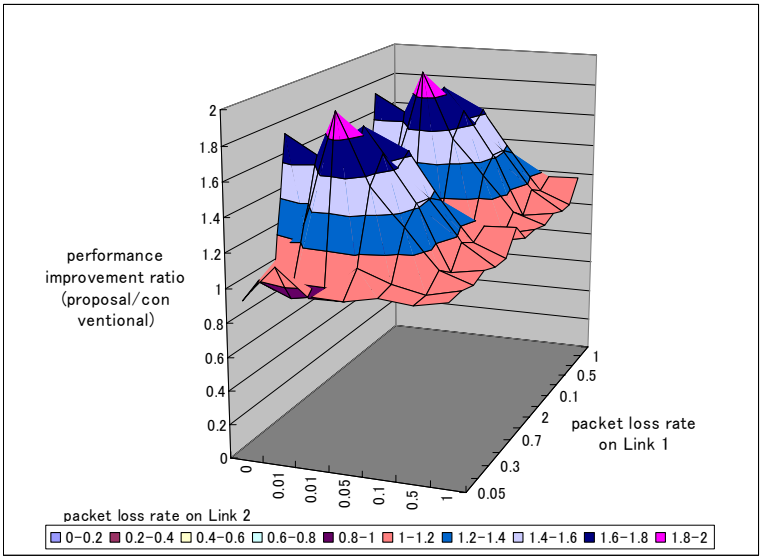


Fig. 5-6　Performance improvement: RTT of link 2 is 20ms

In Figures 5-5 and 5-6, the horizontal axis indicates the packet loss rate (%) of Link 2, the vertical axis the packet loss rate (%) of Link 1 and the depth axis the performance comparison (proposed method/conventional method).

Figures 5-5 and 5-6 show only the results achieved when RTT of Link 2 equals 10ms and 20ms. Nevertheless, the study confirmed the existence of a range in which the proposed method is effective for all Link 2 RTTs tested (values less than or equal to 100ms, in units of 10ms).

Simulation 2

Next, the effect of the proposed method under realistic network conditions was confirmed by simulation.

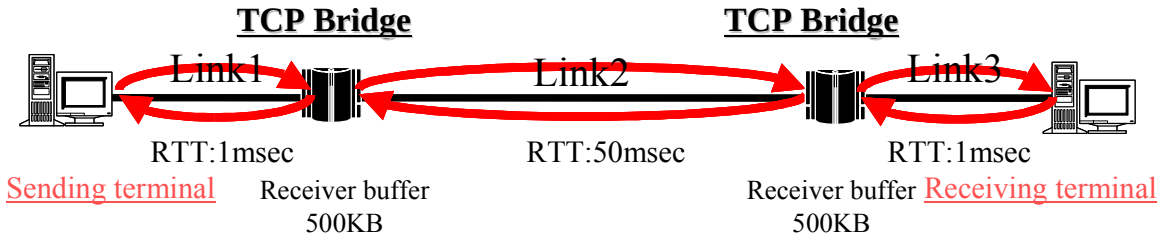


Fig. 5-7 Network model of simulation 2

In Simulation 2, an environment similar to LAN was established and a Link 1 RTT value of 1ms and a Link 2 RTT value of 50ms were presumed. This latter value was used since it is the intermediate value between the Japan Tokyo – Osaka RTT (approx. 30ms) and the US west coast – east coast RTT (approx. 70ms).

The proxy receiver buffer size was set as 500KB. While the receiver buffer default value is normally 64KB, special units designed for proxies and high speeds use a value between hundreds of KBs to several MBs. Because the TCP proxy is a special device for relaying, the 500KB value – which is of the same level as that of special units designed for proxies and high speeds - was employed.

Under such conditions, the Link 2 packet loss rate was set at about 0.01% within the normal wired network range, and throughput was measured while changing the Link 1 (= Link 3) packet loss rate as shown in the table. The results are indicated below.

The parameters of the proposed method were the same as those used in Simulation 1.

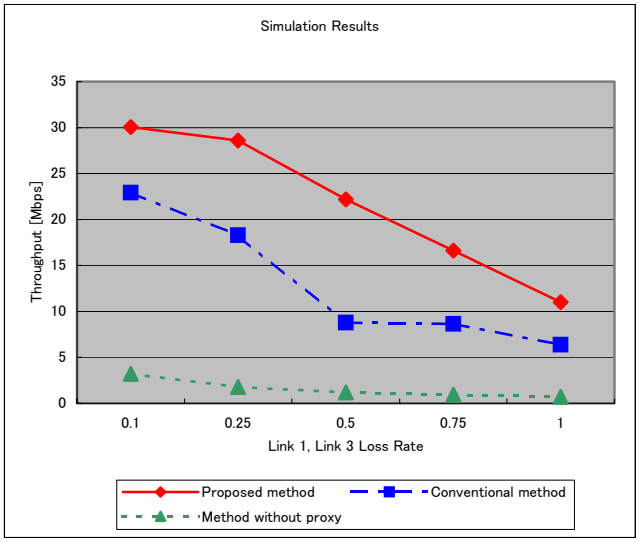


Fig. 5-8　Simulation 2 results

Table 5-2 Simulation 2 results

Packet Loss Rate	0.1%	0.25%	0.5%	0.75%	1%
Proposed Method	30.1	28.6	22.2	16.6	11.0
Conventional Method	22.9	18.3	8.78	8.62	6.39
Without Proxy	3.19	1.77	1.18	0.957	0.710

The simulation results are shown in Figure 5-8 and Table 5-2. In Figure 5-8, the horizontal axis indicates the Link 1 (= Link 3) packet loss rate. With these results, a throughput comparison study was conducted between three methods: the proposed method, conventional method and a non-proxy method (providing TCP end-to-end communication using the network of Figure 5-5). In addition, the conventional method employed was the BSD method. This method was chosen since the receiver buffer problem results in a smaller rate decrease with BSD.

The results indicated that TCP proxy communication can greatly improve throughput in comparison to communication performed without a proxy. Additionally, a comparison between the proposed method and conventional method indicated that, under simulation conditions, a performance level nearly twice that of the conventional method (BSD) can be achieved. These results confirmed the effectiveness of the proposed method.

5.5 Prototyping and evaluations

To confirm the effect of the above-described proposed method in an actual system, the proposed method was installed in UNIX and a simple evaluation conducted.

For the performance comparison, communication was conducted by inserting a TCP proxy between two terminals in the same manner as the network of Figure 5-4. Network conditions were established by placing a Network Emulator (NE) between the TCP proxy and terminal. Under this environment, proxy communication was conducted based on the proposed method and conventional method, respectively, and the results were compared.

The results confirmed that a performance level equivalent to approximately twice that of the convention method can be achieved in the effective range of the proposed method

5.6 Conclusion

This chapter addressed about TCP overlay network architecture and its throughput performance issues, then proposed a TCP proxy congestion control algorithm designed to improve TCP throughput, and described the results of proposed method evaluation conducted through simulation and prototyping.

In order to deploy high quality communication mechanism into existing the "best effort" internet, TCP overlay network is a desirable solution. TCP overlay network can improve throughput performance and also can introduce many new future services.

TCP Bridge (TCP proxy), however, must be careful to relay TCP communications to prevent throughput degradation due to buffer overflow in TCP Bridge. With conventional methods, congestion from downstream causes throughput degradation upstream. Conversely, with proposed stealth buffer management, the number of times the receiver buffer becomes full is decreased by restricting the

advertised window through proxy buffer management. The proposed method, therefore, facilitates stable TCP relaying without decreasing throughput.

Results of ns-2 based simulation conducted to verify the effect of the proposed method confirmed that the proposed method results in a performance level that is approximately twice that of conventional TCP relaying. While the results are based on a two-stage proxy system, the proposed method resolved the problems that occurred with the proxy transmission/receiver buffer, indicating that the effect can be further improved when the number of proxy stages is increased. Additionally, the operation and effect of the proposed method were confirmed through prototyping.

Future plans include identification of a detailed method for determining each parameter and clarification of the relationship between the number of proxy stages and performance.

This work was partly supported by Ministry of Public Management, Home Affairs, Posts and Telecommunications of Japan, for which the author would like to express his gratitude.

CHAPTER 6
CACHING ARCHITECTURE FOR LONGEST PREFIX MATCHING IP FORWARDING TABLE SEARCH

6.1 Introduction

With the increase in Internet traffic, acceleration of the backbone router is increasingly demanded. In order to speed up the router, acceleration of the protocol processing of the packet is needed. One area causing a bottleneck in the acceleration of the packet processing is the search for the IP forwarding table to decide the output path of packet. Because searching for the Longest Prefix Match (LPM) is needed when searching for the IP forwarding table, conventional full match high-speed searching methods such as hashing etc. cannot be used. Therefore, acceleration of LPM searching is becoming an increasing challenge. Many attempts in acceleration have been conducted in recent years, but while high speed searching is demanded, the search for a high capacity table needs to be done at a low cost.

First, we will explain the LPM search method as previously suggested, and then discuss weaknesses from the IP forwarding table capacity viewpoints. The IP forwarding table in the Ipv4 backbone net that is widely used now consists of huge entries over 64K, and the number of table entries is increasing daily. Furthermore, with regards to the VPN (Virtual Private Network), when thousands of entries are prepared for only hundreds of VPN users, it still needs to search for a high capacity table for millions of entries alone. In order to search the high capacity table by LPM, complicated algorithms and much calculation time are needed. For LPM searching, the

algorithmic approach and hardware accelerator (LSI chip) approach have been suggested. For example, in the algorithmic approach like ones suggested in reports[60][61][62], the algorithm is sped up by using a layered system of an IP address that is actually in use and is characteristic of an integrated address space. In these algorithms, it searches the tree by referring to the memory several times, so memory access becomes bottlenecked and has a speed limit. On the other hand, as shown in reports[64][65][66], in the LSI chip approach (hereinafter called T-CAM method. The LSI chip implemented in the T-CAM method is called a T-CAM chip.) where several LSI chips are laid out and acceleration is performed using a parallel action, it is possible to search at a very high speed because LPM searching is done by the LPM searching chip (T-CAM) [68][64]by adding a special built-in hardware circuit to the content-addressable memory(CAM). Another approach is a hybrid approach, where algorithmic approach is performed on a CPU chip and a CPU internal cache memory[72][73][74]. This can enable high-speed search because of the high-speed internal cache memory access. However, both in LSI chip approach and hybrid approach, the table entry number for packing is limited to 1 chip. Therefore, even if initially equipped with enough capacity chips, a response for bigger IP forwarding tables are needed, and it will be difficult due to net broadening etc. For the router, in the other words the silicon router that processes protocol with the current common hardware, an increase in table capacity, in other words more chips, would mean extensive changes in the hardware, and is not realistic from both the cost and operation viewpoints. As such, in both approaches, searching the high capacity table at high speed and low cost, is difficult.

Conventional methods try to search the entire entries of a IP Forwarding Table, at a uniform and stable speed. However, under present conditions, with regard to the

entries used, characteristics such as locality and continuity can be predicted because packets with the same destination may be arrival continuously. Consequently, entire entries do not need to be searched at the high cost method uniformly; there is high possibility to search in low cost by devising a search method that considers actual packet arrival characteristics.

As the search method considers locality and entry continuity, several search methods using a cache have been suggested. For example, an architecture that has an IP forwarding table and cache table (it is simply called cache) to search only IP addresses that have been recently used when searching at high speed instead of searching all of the IP forwarding table[67]. This method can search at high speed because it easily hits the cache when the same IP address is continuously searched. However, when the IP address is widely distributed, cache mistakes will increase. In this case, a low speed longest prefix matching search at the IP forwarding table is conventionally done. Another method uses an extension of the host address with caching [67] (called the Host address Cache Method), but when there are many traffic flows and they are mixed, the miss-hit rate will be bigger because many different IP addresses would be frequently switched in the cache. In other words, improving the cache hit ratio is needed to realize high speed searching at a low cost.

In this thesis, we propose a LPM Cache Method to improve the cache hit ratio. The proposed LPM Cache Searching method searches the IP forwarding table in a low cost algorithmic approach, and searches in high speed using a T-CAM chip, making high speed LPM searching possible using the cache. Also, in a high speed cache in order to further reduce the cache miss-hit rate dynamically, an entry switching method to locate actual entries that are frequently used is proposed. As shown in Fig. 6-1, the

LPM cache method stores a wider space in one entry than the host address, thus it is possible to lower the miss-hit rate by caching rather than by conventional methods.

Conventional points of issue and the proposed LPM cache method are explained in Section 6-2, below. In Section 6-3, features of the proposed method are stated. In Section 6-4, a quantitative performance evaluation of the proposed method is done through a simulation using actual network traffic and an IP forwarding table.

6.2 Longest Prefix Match cache architecture

6.2.1 LPM search

First, the LPM Search is explained. In an IP forwarding table (hereinafter simply called forwarding table), several entries- prefix, prefix length and output port No., become one entry and constitute the table (Figure 6-1). An address search at the router is an operation to obtain the output port information of an entry with the longest match (searched by the LPM search) within a combination of prefixes and prefix lengths as the retrieval key for the destination IP address (called DestIP) of an arrival packet. In this report, a prefix and prefix length pair are hereinafter called a simple entry, to simplify explanations.

The prefix shows the lead part of the IP address used for IP address matching assessment, and the prefix length show the effective length from the head of the prefix. In other words, the part longer than the prefix length is considered excluded (called the "don't care") from the matching assessment. To simplify explanation, when considering 4 bits of an IP address, if the prefix of an entry is 0000, and 1 is the prefix, the entry will be written as 0*** or 0000/1 hereinafter (* is the don't care part). In the matching assessment, IP addresses (Host addresses) that match this entry are eight- 0000、0001、

0010、…、0100、0101、0110、0111. LPM is the method to hit the entry that has the longest prefix length among several entries that match the prefix at the time. As in Fig. 6-1, when searched with DestIP=0100 as key, Entry No. 1, 4, 5 are prefix matches. The prefix length of individual entries that match are 0,1,2, thus entry No. 5 with the has longest prefix length would be the LPM[1] search result. Then, the packet is the output from #4, which is specified as the output port of Entry 5.

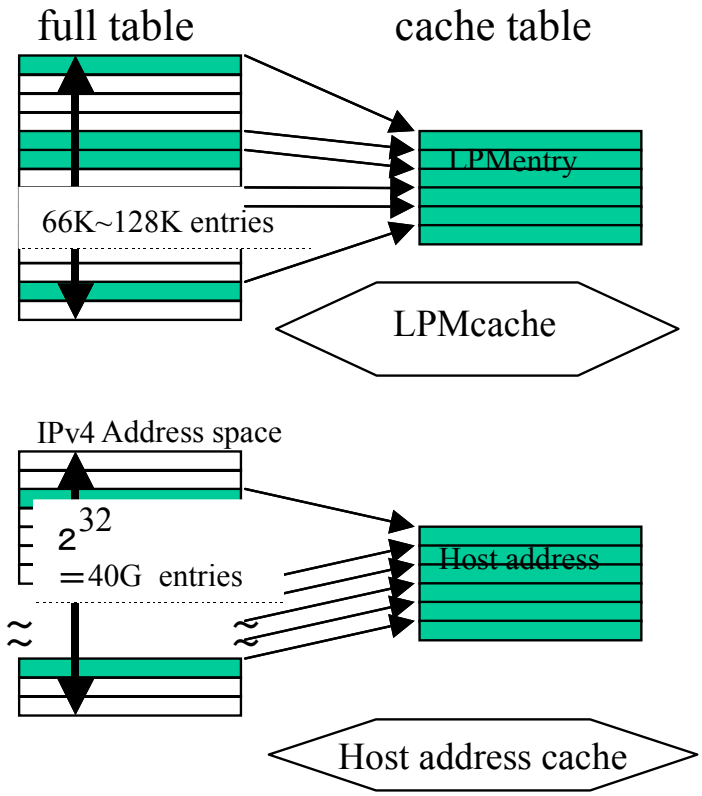


Fig. 6-1 LPM cache system and traditional cache system

Table 6-1 Forwarding table

Entry No.	Prefix	Prefix length	Output Port No.	Aggregated Address
1	****	0	#1	1000, 1001, 1010, 1011
2	11**	2	#2	1100, 1101, 1110
3	1111	4	#1	1111
4	0***	1	#3	0000, 0001, 0010, 0011
5	01**	2	#4	0100
6	011*	3	#2	0110, 0111
7	0101	4	#5	0101

6.2.2 Caching architecture

Searching architecture using caching is explained. First, this paragraph explains the caching architecture. The next paragraph in turn, explains conventional methods and a proposed method based on that architecture.

In the caching architecture, there are two tables- the forwarding table (Full table) and the cache table. The full table is the table that has the whole entries from the forwarding table, and the cache table is the table that has partial entries of the full table as is, or in an expanded form. The full table determines an action of low cost through the use of high capacity memory and an algorithmic approach [60][61][62] search method at the sacrifice of high speed. On the other hand, the cache table uses a T-CAM method, in other words a search method [64][65][66] using a proprietary LSI chip to search at high speed. A proprietary LSI chip has a smaller capacity per chip and is expensive, so it is common for the size of the table size to be smaller than the full table at a realistic scale and cost.

In the table search, the cache table is first searched, and if it doesn't hit, in other words, if there is a miss-hit, the full table is searched. All entries are stored in the full table, so it is sure to be hit,[9] and the search will be completed.

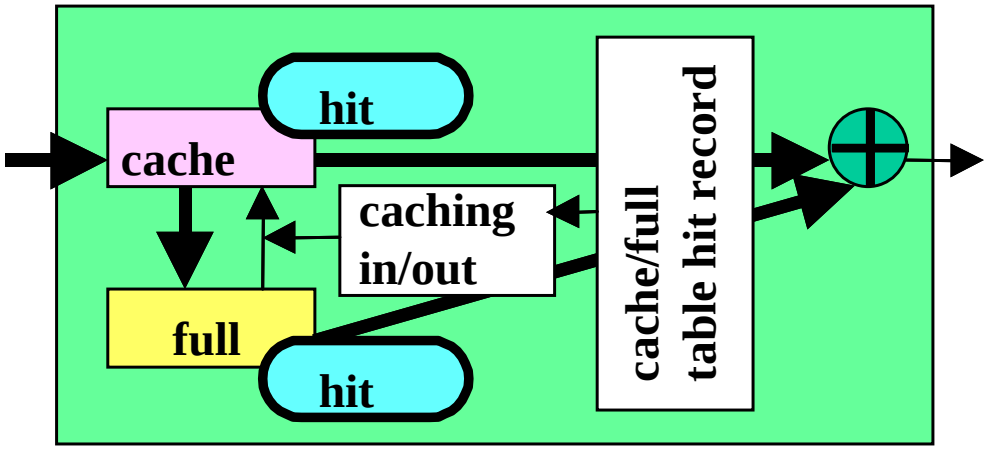


Fig. 6-2 LPM cache search architecture

6.2.3 Host address caching method

The conventional, Host Address Cache Method is described here.

In Host Address Cache Method, the full table retains the prefix and its prefix length as the complete entry. However, the cache table retains the IP address, meaning the host address. Therefore, the cache table searches for a full match search, in other words a host address that completely matches the DestIP. For example, when the full table is as in Figure 6-1, the DestIP＝0100 packet operates the LPM search, and as a

[9] There are cases that do not result in hit such as IP Unreachable address, but is considered here with exceptions of those.

result of this full table search, entry No.5 is hit. By that, the host address 0100 is put in the cache table as an entry.

When the host address method is used, usability of cache may deteriorate because host address distributed in very wide space would be put into a comparatively smaller cache.

6.2.4 LPM caching method

As shown in Fig. 6-1, in contrast to the conventional host address cache method that caches the IP address (in other words, the host address), the LPM cache method caches the entry itself into the forwarding table, in other words, the combination of prefix and prefix length instead of host address. That is to say, the LPM cache method operates an LPM search for part of an entry in the forwarding table at high speed. For example, in the example in paragraph 6.2.3, for the LPM cache method, only entry 1, 01**/2, is inputted into cache table.

As shown in Figure 6-2, in the search procedure for the LPM cache method, the cache table is first LPM searched, and if there is a hit in the cache table, the search is completed. If there is a miss-hit, the search continues to the forwarding table (full table). To improve the hit ratio at the time, the hit condition of the table is recorded, and by switching the appropriate cache entry, an improvement in the cache hit ratio can be realized. This is described in the next paragraph.

6.2.5 Caching rules

The caching system is generally characterized by cache management such as switching the algorithm rule of entries. In this LPM Cache method, in order to correctly perform an LPM search, rules of cache need to be established first. The rules for cache

switching of LPM cache are first stated below, and the selection of a candidate for the switch target will be discussed in next paragraph.

Data switching rules of the cache, which is essential in order to correctly perform the LPM will now be discussed. To increase the hit ratio, the cache table entry and full table entry are switched with appropriate timing. In the LPM cache method, During this switching, the rule is that the entry and its corresponding entry are switched together, not switching the entry singly. This is because the longest match entry within cache table would not be able to guarantee the correct longest match entry if a single entry is moved between tables. For example, in Figure 6-1, if only Entry 4 (prefix=0***) and Entry 5 (prefix=01**) are put in the cache table, in the results of the search of cache table, Entry 5 will be judged as the longest match, and the search would be completed, but this is incorrect since Entry 6 in the full table side is also the correct match. Caching rules needed to avoid such incorrect searches are explained below.

To explain caching rules, terms are defined. When entries have the same prefix, the long entry is called a child, and the short entry is called the parent. For example, for LPM entry of 100.120.0.0/16, 100.0.0.0/8 is the “Parent”, and 100.120.140.0/24 and 100.120.180.0/24 are the” Children”.

At this point, the switching rule will be (refer to in Fig. 6-3)

[When adding to the cache]: Register both the objective entry and all the “Children” into cache together.

[When taking out from cache]: Delete both the objective entry and all “Parents” from the cache.

This rule puts the complete entry of the long prefix into cache so that when only one prefix entry is registered into cache, the shorter prefix entry in the cache is hit,

ending the search, even if the “Child” should be hit, in other words the entry with the longer prefix (fundamentally this should be hit). The same idea applies when removing from the cache as well.

As shown in Figure 6-4, the related parent-child entry, explained in other terms, corresponds to a sub tree with a focused entry at the top spot, and the forwarding table is described as a binary tree that branches out at 0/1. The above-mentioned selection rule can be rephrased as the rule to register a whole sub-tree.

An example of a cache operation using a full table as in Figure 6-1 is explained. First, the cache starts in an empty condition. After receiving the packet destination, in the full table is searched, in other words, the key is the host address. When the host address is 0100, Entry 5 is hit, and output port is determined as #4. Based on this information, the packet is forwarded, and at the same time this entry is LPM cached. As shown in Figure 6-4, in the Entry 5 sub-tree (shaded part in the smaller part of Figure 6-4), Entry 6, and 7 are included. In other words, within Entry 5 is Entry 6 and 7, which have longer prefixes as “Children” on an individual branch, so that Entry 5 is stored with Entry 6 and 7 in the cache. Furthermore, when host address =0000 is received, the cache will be miss-hit, but will still hit 4 in the full table side, so it is necessary to put a sub tree of Entry 4 in other words, Entry 4,5,6,7 into the cache. Since Entry 5,6,7 are already in cache, only Entry 4 needs to be added. Under these conditions, the four entries corresponding to the sub-tree in the large shaded area of Figure 6-4 are registered, and it represents a caching of eight host addresses.

In the LPM cache, of course caching the host address itself is possible. Therefore, it is possible to cache the host address as it is when the cache does not have

enough space to put several entries that comprise a sub-tree[10]. In that case, when there is enough space in cache later, it may delete this host address and store the desired sub-tree.

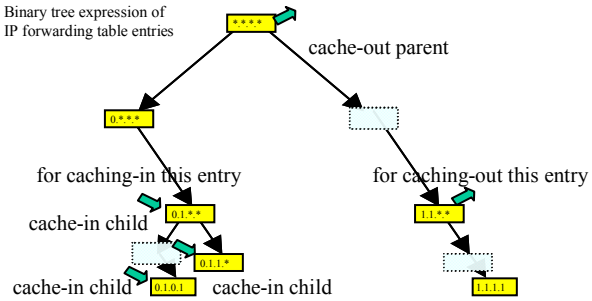


Fig. 6-3 Rules for LPM caching

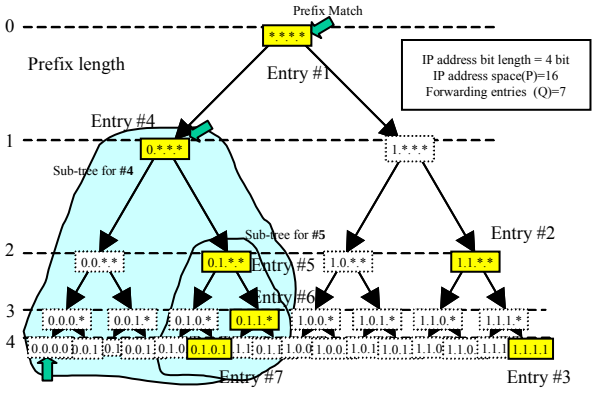


Fig. 6-4 Efficiency of IP address space coverage with LPM cache

10 In that case, prefix length to be maximum value. In the case of IPv4, it is 32 bits.

6.2.6 Selection of caching-in/out entry

For entry switching, selection of which data to switch is important. Data selection is choosing which data should be added to the cache and which data should be deleted from the cache.

Heretofore, for cache switching, selection methods such as FIFO (First In First Out) and LRU (Least Recently Used) are known. These methods choose data based on the history of the individual data remaining in cache. However in the LPM cache, it is not proper to add a one time entry, or delete multiple entries using the conventional selection method as is, since the history of each of those multiple entries is different.

So when applying FIFO or LPU to this proposed method, it will be as below. To realize FIFO, make the oldest existing entry in the cache to be delete target, and delete the entry along with the "Parent" based on the caching rule. In LPU, the entry that has been least hit recently should be the delete target, and it should be deleted with the existing "parent" in the cache. However, when a new entry is registered in the cache, if a "Child" of the entry already exists in the cache, it should be treated as recently being hit and re-registered.

Qualitative comparison of both methods is as below. In FIFO, even for those with a high hit frequency in the cache, they will be eventually be deleted when a new entry comes. Thus, efficiency is not good, but management of cache is easy. On the other hand, in LRU, entries that are often hit are always new, and those not used get older and become delete targets, so the cache can be used effective, but entry management is complicated. Section 6-4 shows the results regarding the qualitative comparison of these two.

Meanwhile, the timing of a switch to the cache is decided by the miss-hit rate and trade off of moving overhead. To reduce overhead, it is assumed that it would be effective if a method could determine the point of moving based on if it has been hit frequently by using threshold value, instead of moving it to the cache right away even if it hits in the full table. Details are explained in Section 6-3.

6.2.7 Policy control of cache selection

For the selection method of cache switching entries, not only should performance be considered but also the possibility of adopting an IP Packet forwarding policy. In contrast to the performance-oriented cache management mentioned in previous paragraph, in the Forwarding Policy Control, settings according to the SLA (Service Level Agreement) and management of the net is possible. Actual control can be realized by functions such as lengthening the cache residence time of specific entries, or letting them remain. For example, in the SLA, an entry including the address of users who contracted a high speed forwarding service can be forwarded in high speed from other packets by remaining in the cache. Also, it is possible to choose an entry to effectively handle traffic for a hosting server.

6.3 Features of the LPM cache method

Features of the proposed LPM cache method are compiled in this section.

6.3.1 Efficiency per entry

Usability of cache in the proposed LPM cache method and conventional host address cache method is assumed as below. As shown in Figure 6-1 (B), when considering a 32 bit address system of IPv4 and a realistic scale of the table of router,

the relation of P, the scale of host address space (2^{32}＝about 40 billion), Q, the number of entries for the forwarding table(Full table), R; the cache entry number will be $P \gg Q \gg R$.

In the host address cache method, a host address is stored in the cache. The cache entry number R is in host address format, so in order to place a part of the address space into R, $P \gg R$. This is said to be very ineffective.

On the other hand, $Q > R$ in the LPM cache method. It can use the cache efficiently because it caches an extended part of the compact forwarding table Q that is combined with the host address space P (Figure 6-1(a)).

6.3.2 Frequency bias of use of individual entry

One of the features of the LPM cache is the use of a biased use frequency entry. The current forwarding table (Full table) consists of an enormous number of entries. The conventional method that proposes searching with the LPM is premised on the treating of all entries equally, but it would be more appropriate to show bias according to use frequency.

A proposed LPM cache method will put a small quantity of entries that are used frequently in the high-speed cache, and majority of entries that are not often used are searched in the full table. Details of the evaluation will be given later, but in fact, space of each IP address space differs largely by entry, and use frequency is a great bias in the forwarding table entry set at the router. For example, there are about 100 “child” entries of for an entry with a prefix length of 16, and another independent entry with a prefix length of 24 does not have a “Parent” or a “Child”. Also, simulation results stated in Section 4 confirm that there are many entries that are hardly used or not used at all within the specific observation period.

6.3.3 LPM search engine LSI

A search engine that can perform a high speed LPM search for a high-speed cache, which is essential to realize a LPM cache search system is discussed here. As a high-speed search cache, a LPM capable T-CAM chip has been realized[68]. Only 16K ～64K per chip can be accommodated, but it can ensure the capacity according to the allowed cost because capacity can be extended by connecting several chips in sequence. The approximate search delay of these chips is a very high speed, 20～60 nano seconds. However, when considering that several entries are often placed in and out, when there is no restriction [64] on the sequence of adding the entry, a search chip that does no sequencing is good for the LPM cache. Without a sequence restriction it lessens the overhead of entry switching, which will be described later.

When comparing cache realization costs of the LPM cache search method and the host address cache search method, cost of the host address cache will be lower when they have the same cache capacity. This is because the host cache address search can be realized using the existing CAM etc. On the other hand, the cost per entry in T-CAM is estimated to be about 2-4 times of CAM. This is because the regions that hold prefix length and additional circuits to find the entries with the longest match are needed when using T-CAM.

6.3.4 Caching-in/out moving overhead

Considerations about cache overhead are discussed here. In the host address cache method, each new host address needs to be put in the cache. Therefore, the overhead for cache input is large.

On the other hand, in the LPM cache, even in different host addresses, entry cache-in/out can be not needed. If the same entry is hit for the addresses, that amount of

overhead will be reduced. Also, by having only entries with a high use frequency remained in the cache, it is possible to further reduce the overhead.

However, in the LPM cache method, the travel distance in one jump is large because caching of the entry is performed for several entries, not by a single entry unit, according the previous rules. This travel distance differs according to entry configuration of each forwarding table, use frequency, and the complicated relationship between “Parent” and “Child”. And, when using search LSI chip like having restriction for order of entry, its overhead will be increased more because switching of existing entry order also will be occurred when switching entries.

Caching-in/out must be waited, when searches are performed. This may be a problem of cache search system. However, this is not a severe problem in practical network because time lag between the consecutive arrived packet headers makes idle time of search system for cache-in/out.

6.4 Performance evaluation

6.4.1 Definition of caching search system performance

In this thesis, as a function of the cache system, the miss-hit rate or hit ratio (=1-miss-hit rate) is used as a gauge. As previously stated, when using only the full table, at least 10 times the search period is needed as compared with using caches together. When the router is configured so that the full table is in a different location from the cache, for example in the case of Figure 6-5, more time is needed to search the full table. Furthermore, when processing a full table search with a CPU using a routing protocol (RIP, OSPF etc) process, the CPU overloads when miss-hitting, and the routing

process may lag. Furthermore, when short packets arrive continuously, and the cache miss-hit still continues, the next packet may become search waiting, or at the end be disposed.

It is clear that the average search period can be improved by reducing the miss-hit rate. Also by lowering the miss-hit rate, the minimizing of the cache capacity is also possible; the effects of improving the miss-hit rate is also large from the actual cost of cache system.

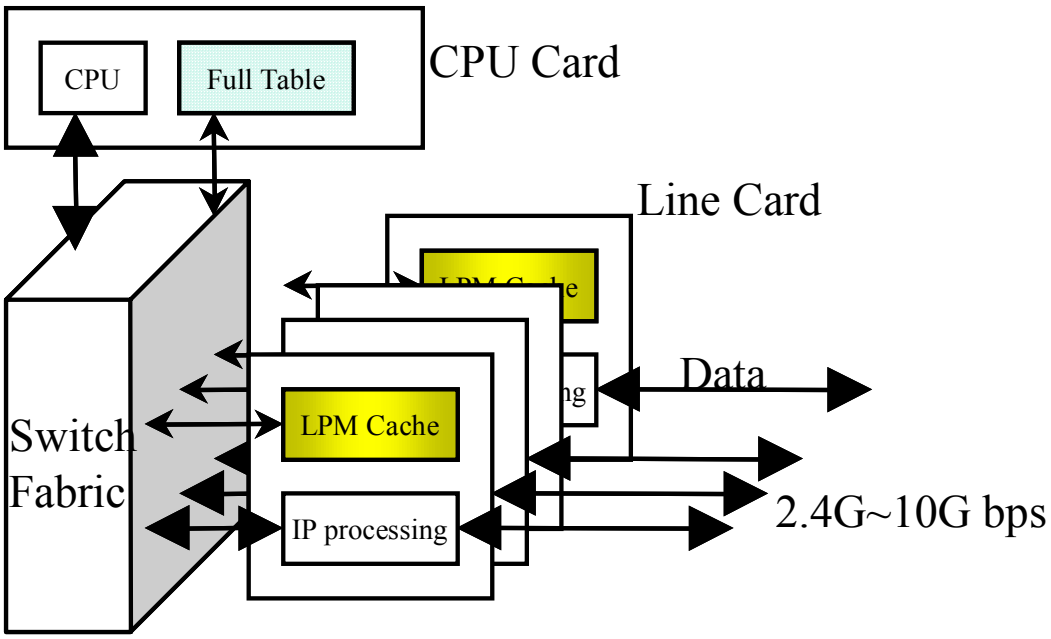


Fig. 6-5 Typical router architecture by using proposed LPM cache search

6.4.2 Simulation model

The miss-hit rate of the proposed method was evaluated by simulation. In the simulation, the forwarding processing at one of inputs of router IF was used as a model.

The data used was a recorded IP address packet that arrived in the router located in the backbone network, and was searched; the forwarding table of a common router was used as the full table. The full table has 68,000 entries; there is a maximum class size for a router. Total number of input packets is about 2,000,000 packets. The data was derived from gains measured on September 30, 1999.

Data characteristics are summarized as follows. Entries hit by the LPM amount for each of the 68,000 full tables was about 3,000, remaining entries did not get hit by the LPM. And, for the frequency of entry moves between "Parent" and "Child", each entry moved at 4.6 entries per one time on the average, including the entry itself.

6.4.3 Miss-hit rate comparison

Regarding the performance evaluation indicator, comparison was done using the miss-hit rate. The miss-hit rate is the ratio that did not have an entry to hit in the cache table search, in other words, it is the probability of a hit in the full table, and gives a hit number for the full table and the entire hit number.

The result of a changed cache capacity and comparison of the miss-hit rate of host address cache and the hit ratio of LPM cache is shown in Figure 6-6. Two methods of evaluation were used, FIFO, LRU. In the evaluation, the entry that was a miss-hit on cache side, in other words an entry that was hit in the full table, is immediately moved to the cache, along with its "Child". Figure 6-6 shows that the LPM cache is always better than host cache in a cache capacity over 2000. The difference is especially noticeable when the cache capacity is large.

However, the proposed LPM cache method is at a disadvantage when the capacity is smaller. Putting an entry with a lower hit rate into the cache appears to be more noticeable when the cache capacity is smaller because the LPM cache is

performing caching at an average of 4.6 entries unit. On the other hand, when the capacity is bigger, as recorded above, the performance of the LPM cache is better because of the effect from the fact that the space the entry covers is wide, rather than using the cache wastefully.

And by using the LPU, instead of FIFO, the host address cache constantly changes the destination IP address, the performance will get better because frequently used entries will tend to remain in cache.

Meanwhile, also in the fixed method that does not switch between caches, performance exceeding the host address cache method was achieved. This is because the actual, frequently used entries are the only parts among many entries in the full table; an area that used frequency has a large bias.

Lastly, results of Figure 6-6 will be evaluated from viewpoint of cache capacity. When a miss-hit ratio of 0.01 is achieved, it has a difference of only thousands of entries. On the contrary to host address cache method however, improvement of miss-hit rate is peaked even with an increased cache size, thus the proposed method can gain more improvement. Viewing the range gained by the data this time, when a gain miss-hit rate of 0.003 is desired, the host address cache method needs about 12K cache size, but LPM cache method needs only about 6K. Even if the expense of cache in LPM cache method is considered, it is understood that the LPM cache method will be advantageous when striving for further improvement of miss-hit rate.

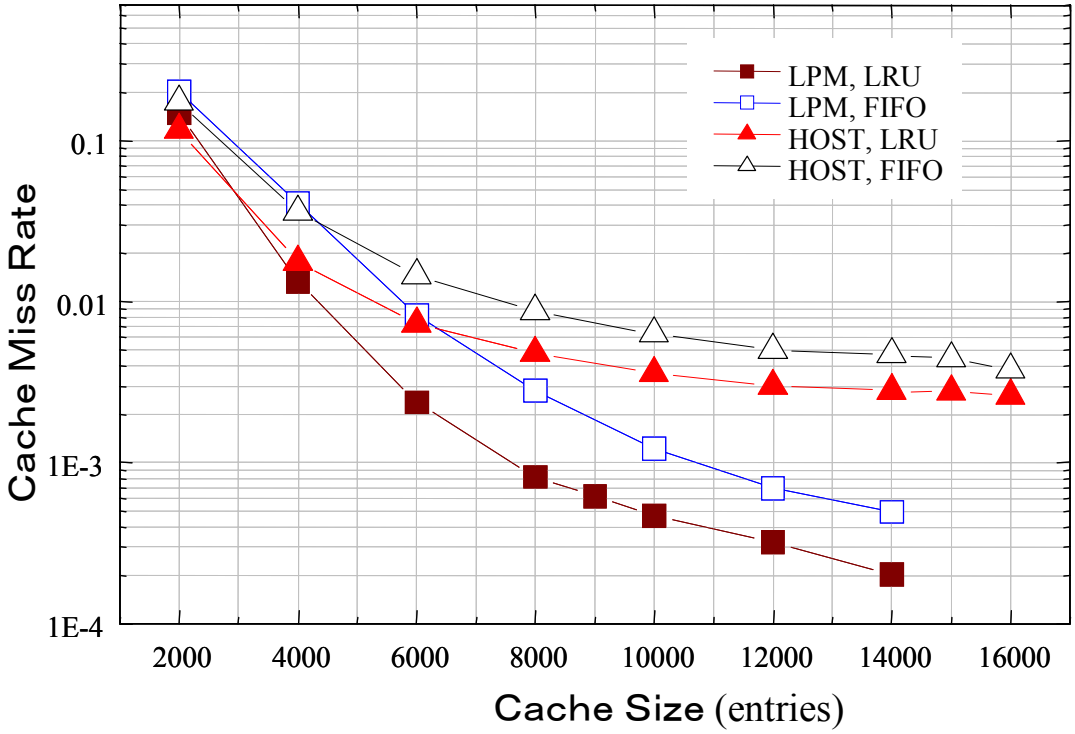


Fig. 6-6 Cache performance (miss hit rate for cache size)

6.5 Conclusion

In this manuscript, an LPM cache method that operates a high capacity IP forwarding search, at a high speed and low cost by using cache has been introduced. Performance was quantitatively evaluated by simulation using data of the router, which is used in actual backbone networks, and its efficiency was demonstrated. In the comparison with the conventional host address cache method, it showed that the miss-hit rate of cache can be improved by almost 10 times. And, it was found that performance could be greatly changed by switching algorithms of cache. In this

evaluation, only the algorithms FIFO, LPU, were used. An examination of algorithms with better performance is planned in the future.

CHAPTER 7
CONCLUSION

Today's Internet has become a part of the daily life of residential users as well as business users, who use e-mails and Web for their business and entertainment. This means that the Internet has already become a life infrastructure.

Compared with the improvement of speed and capacity, the changes in quality are much smaller. Toward the future Internet as an actual infrastructure, QOS consideration is required. We should provide traffic control mechanisms for achieving high QOS and network architecture based on the in-depth consideration to the traffic control while operational cost and equipment cost remains minimum.

This thesis discusses on traffic control and architecture for high-quality and high-speed Internet. The discussion focuses on five subjects; admission control methods for (1) connections and (2) a partial data of connections, i.e. a burst, (3) application aware communication control, (4) network architecture for application level QOS and (5) technologies for realizing high speed routers.

A transport network that people expected to become an essential part of broadband multimedia network is an Asynchronous Transfer Mode network or an ATM network. Because ATM has been expected to integrate both circuit switch and packet switch, it is necessary to guarantee QOS for accommodating multimedia traffic, i.e. circuit switch friendly traffic such as telephone or movie streams. As ATM networks are based on connection-oriented principle, it is suitable for controlling QOS by controlling network traffic load i.e. by controlling numbers of connections. Call Admission Control or Connection Admission Control (CAC) is, thus, firstly addressed. In the CAC, CAC performance depends on a cell loss estimation and a CAC procedure. The CAC proposed in this thesis has good advantages; only two parameters and small

time are needed for accept/reject new connection request. The proposed CAC uses "virtual cell loss rate" instead of cell loss rate as cell loss estimation. Simulation results show that the proposed CAC has appropriate accuracy in estimation and offers appropriate network utilization.

More bursty traffic, however, does not fit to the connection-by-connection CAC. Burst transfer methods has been developed for that purpose. It reserves and releases bandwidth for each piece of burst data at the time when the burst is ready to send. This, however, may cause large latency and waist of bandwidth during the reservation if the bandwidth is not available at least in one link. To make the latency small, burst server architecture is proposed. In the architecture, a burst server, which stores and forwards the bursts, is placed between links and the reservation methods are modified. Bandwidth reservations can succeed either if the bandwidth on all links between sending and receiving terminals are available, or all links between burstservers/sending/receiving terminals. Numerical results show that the latency is much improved and utilization of networks is also improved two times larger than conventional architecture at maximum.

These contribute packet level QOS improvement. On the other hand, for considering user level QOS, not only packet but throughput and contents retrieval time also must be improved. One of the today's most important and popular applications of the Internet is web (World Wide Web). "8-second rule" reveals users have very little tolerate against waiting time of contents displayed. The waiting time consists of combination of network delay and server delay. It is necessary to cooperate networks and servers to reduce the waiting time. . The network cache architecture for network friendly pre-fetch retrieval is therefore proposed. Some numerical results show that the network cache improves 40% latency without causing congestion.

To meet the high throughput requirements such as from storage networking and high speed wide-area LAN interconnection, TCP must be improved. Although many TCP modifications has been proposed, it is difficult to introduce new TCP to end hosts/servers, but easy to the intermediate node. TCP relay node (TCP bridge) is proposed to be set in a network. In the TCP overlay network TCP Bridges are cooperated each other and change TCP characteristics to appropriate one for the links between TCP Bridges. One of issues to be solved is a congestion control issue. If a trivial congestion occurs on sending side of a TCP Bridge, it causes serious rather than non-trivial congestion on receiving side of the TCP Bridge. To prevent this problem, buffer control in TCP Bridge is proposed and discussed. Simulation results shows that proposed control method improve the problem and can achieve throughput two times higher than the case without any control.

As well above mentioned traffic control, network itself must be improved in its speed to accommodate today's huge traffic. One of essential bottleneck in developing high-speed router is IP address table search. Longest Prefix Matching (LPM) search must be used to search the table. The idea is proposed that cache architecture is employed with algorithmic search and hardware search engine. Instead of ordinary CAM as used by conventional caching architecture, Ternary-CAM is employed to reduce cache miss-hit ratio. Because caching-in/out rule is not obvious in LPM search, the rule is carefully invented and investigated for a validation. Performance evaluation is shown to disclose the proposed architecture can achieve at least ten times smaller miss-hit ratio than the conventional cache architecture.

Toward the future Internet, this thesis has discussed new architecture and control for high QOS with the networks while operational cost and equipment cost remains

minimum. We believe these discussions contribute to realize next generation high quality, high-speed Internet.

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Biography

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