IP switching in a simplified ATM environment

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ABSTRACT

Key issues in the current development of Internet seem to be its capability to scale and to support new real-time or near real-time applications like video- and audio conferencing. There are two factors that affect these qualities: one is the ability to distinguish which connections should be switched and the other is the effective control over network resources. ATM is a serious attempt to standardize global multiservice networks. This attempt seems to suit well for the future Internet. ATM was originally meant to be an easy and an efficient protocol but it is now turning to be 'yet another ISDN'. More and more features are implemented to ATM resulting in the overloading of the network with management procedures. Therefore a new approach needs to be taken. In this approach a strong reminder of 'what is necessary' needs to be kept in mind. This paper presents an alternative, simpler approach to the ATM traffic management and introduces some suggestions how to map Internet applications to simplified ATM environment using an advanced IP switching concept.

Keywords: ATM, traffic management, IP switching, Internet

1. INTRODUCTION

It is becoming increasingly obvious that Internet will be THE network of the future. In this role Internet will have to mature to adult age. It has to scale and offer reliable and efficient platform for different services. The key issue for a successful implementation of a new network or a protocol is that it should provide something better than earlier networks or protocols. In order to accomplish this, the future Internet must provide an easy and scaleable platform to support a variety of services. Easy in this context means that the threshold for introducing new concepts and services should be low. Scaleable means that services ranging from few kilobits per second to terabits per second can be introduced at the same time. This goal may be reached by adopting asynchronous transfer mode (ATM) as a core networking technology.

Designing a new network protocol, like ATM, is always inspiring work. Many groups and even more people want to leave their mark in the history. This can be observed for instance in ATM Forum contributions list that has had a tendency to double every three years. This can lead to complex solutions resulting in inefficiency and overcomplexity when real-world applications are being implemented. Sometimes the mistake of having designed an excessively complex protocol is easily repaired; but sometimes a lot of tedious work and a strong effort are needed. Therefore a strong reminder of 'what is necessary' needs to be kept in mind.

Internet protocol (IP) switching that is based on the concept of flow switching is one approach to deal with the problems of short vs. long duration Internet traffic on top of ATM. Disregarding the connection based model of ATM, IP switching tries to be swift and able to adapt to changing traffic profiles with the cost of losing consistent quality of service.

Internet services, in our opinion, do not need different traffic classes in the way that is proposed in ATM Forum's latest specification for traffic management [1]. Instead, we are promoting the use of a simpler scheme for managing traffic in an ATM environment. In this scheme most of the segregated controlling functions are replaced by one unified approach. Main idea is to combine the multitude of service classes to only two service classes, which by default could offer an ideal base for current and future Internet applications. We also propose that connection admission control (CAC) and congestion control be based on real-time measurements that is the most suitable method to be used in fractal Internet environment. This way the need for instantaneous complex real time calculations is transferred to simple calculations spanned on a longer time period.

2. SIMPLIFIED ATM

Traffic in the future Internet is a combination of heterogeneous traffic from all existing and forthcoming services. These services have different characteristics when considering tolerance for the delay and cell losses and the burstiness of traffic

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they are emitting. Therefore, it is obvious that we need to be able to switch some part of the traffic on separate connections providing quality of service (QoS) in the network. These differing combinations of service demands and service classes providing the aforementioned requirements as specified in ATM Forum's Traffic Management version 4.0 are presented in Figure 1.

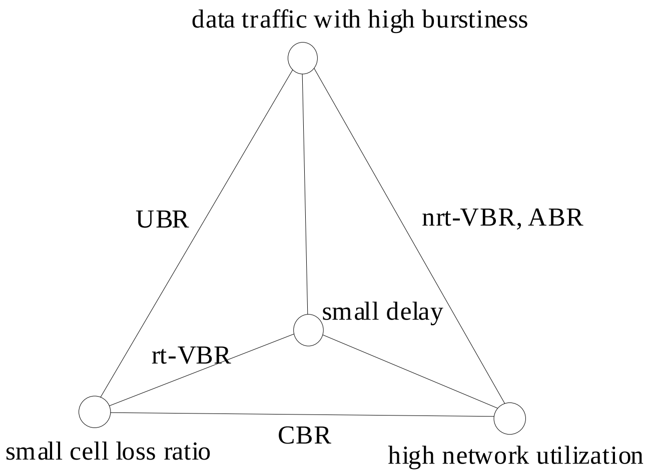


Figure 1: Different service components and their requirements [2]

Figure 1 shows the kinds of service classes that ATM Forum suggests to be used in a network. This leads to the fact that these service classes should also be applied to the Internet.
Choosing the suitable service classes for the network is made difficult by the fact that the more services one implements the more management one needs. This is especially true for the variable bit rate (VBR) connections which are usually treated based on the stochastic model. These stochastic models try to transfer the key characteristics of actual information flow to two or three parameters. It has been stated that these stochastic models do not represent accurately enough all of the real sources they try to model [3]. On the other hand, constant bit rate (CBR) connections are fully expressed with one variable.

2.1 Proposed service classes

Because variability in transmitted cell rate and leads to the stochastic approach we think that it should be left out of consideration. This is because the stochastic model, at its best, can only reveal momentary characteristics in the traffic stream, not the long range behavior in, for example, a video stream. We believe that it is sufficient to consider only cell losses and delays, because the rate variations may either be buffered or discarded depending on choice.
To accommodate the most viable services in the Internet only two major service classes are needed: Nominal bit rate (NBR) and CBR. NBR is purely for non-real-time applications like the traditional Internet services such as file transfer, electronic mail and even web-traffic to a certain extent. In our proposed model the CBR service class, or more accurately the real-time NBR (rt-NBR) is intended only for applications that demand absolute real-time connections or are extremely sensitive to cell loss. Rt-NBR should be implemented with strict rate enforcement and applications demanding rt-NBR should be able to shape the traffic they are sending. In Internet routers working on ATM-transmission technology the rt-NBR connections should be as separate as possible from the other connections. The resource allocation for rt-NBR and other connections is briefly discussed further on in this paper.
One possible suggestion for service classes in the network has been introduced by Kilkki in [4]. We present here our proposal where a nominal bit rate (NBR) is introduced. The proposed NBR should be divided into two to ease the resource allocation in the network.

1. Real-time-NBR (rt-NBR) is a traffic class without rate enforcement and it is meant for the applications which require a minimum capacity but, on the other hand, do not require a small cell loss probability. This could be seen as one way to send VBR traffic: one can make the traffic contract for a minimum cell rate (MCR) and then transmit even more, but still facing the risk of cell loss due to restricted use of buffers.
2. NBR with large buffers is intended applications that need to communicate with other peer entities on a regular basis but can tolerate long delays. NBR can also realize the unspecified bit rate (UBR) traffic class if minimum cell rate required is set to zero.

The proposed traffic classes concerning cell loss probability and delay are presented in Figure 2.

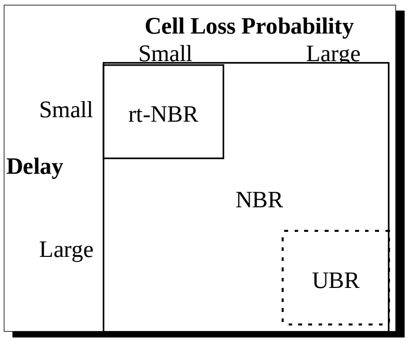


Figure 2: Proposed traffic classes

As the reader may observe from Figure 2 there are no VBR or available bit rate (ABR) categories. If one wishes to use VBR connection one should either be aware of that if the emitted traffic is relatively bursty it may face long queuing delays (NBR) or then shape the traffic so that it falls to rt-NBR or NBR/UBR depending on the loss requirements. As ABR is concerned we feel that the transmission control protocol (TCP) used in the Internet traffic has the ability to handle the flow control on connection basis and that no additional flow control is needed.

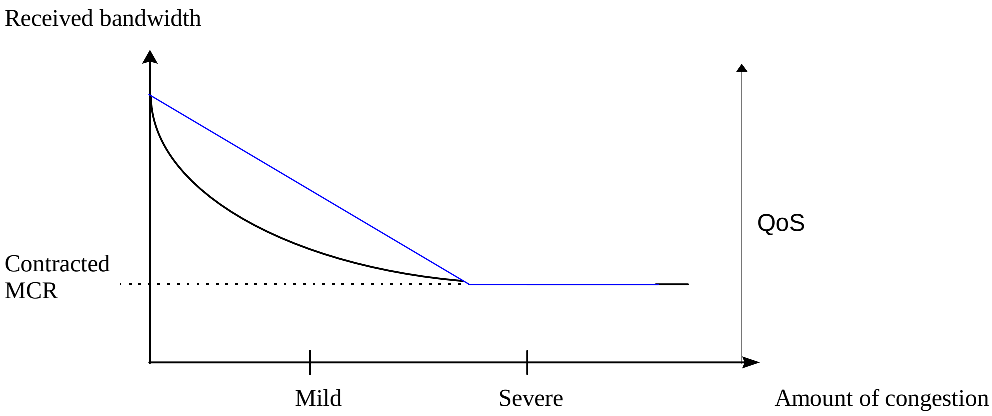


Figure 3: Received bandwidth and QoS related to congestion

Figure 3 shows how the QoS received at contracted MCR could be considered as the hard state contract which would equal circuit switching (or a call) in conventional networks and the ACR (actual cell rate) with MCR deducted could be considered as the soft state contract which effectively resembles the packet switched network.

2.2 Connection admission control

Connection admission control for this kind of scenario is far lighter and somewhat different from those of the proposed solutions for the conventional ATM networks. In the whole, the CAC mechanism should utilize information gathered in real-time measurements of the network. This way CAC can determine the amount of resources that are given to CBR and NBR traffic classes respectively. Capacity aggregation within traffic classes can be done based on a simple adding of the advertised peak cell rate (PCR) values (actual PCR for rt-NBR and MCR for NBR). Buffers are allocated based on the delay bounds. For rt-NBR connections all that one really has to work out is how much the aggregate rt-NBR capacity there is and how much buffer space one has to have for a new rt-NBR allocation and whether this capacity existed. The MCR used with NBR results statistical multiplexing.

2.3 Link management

Congestion control is one of the most vital blocks in network devices. Main task for congestion control is to ensure that network resources are used efficiently and fairly among different competing users. This fairness is a key issue in the Internet

that has no guaranteed provisioning of network resources. Also as we speak about ATM we must also deal with fairness issues. Fairness in this case comes to the picture in the buffering systems of an ATM switch.
Congestion control as a reactive method and resource allocation as a preventive one are linked by the fact that the usage of the other affects the other (Figure 3).

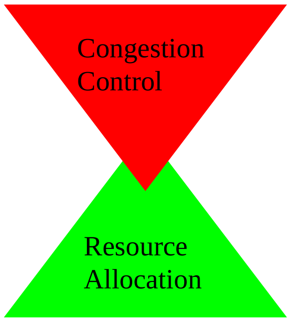


Figure 4:Resource allocation versus congestion control.

Although theoretically it is possible to allocate network resources so that congestion does not exist it is not a practical way of operating. On the other extreme, no allocation of resources at beforehand exist and only congestion control is used to control the use of resources. In practical systems both methods, resource allocation and congestion control, are needed.
Statistical multiplexing leads to the possibility of an overload on the link and can therefore, if persisting, cause switch buffers to overflow. This overflow reduces the performance of the network. Key issues in buffer management are the adequate buffer size and an efficient control algorithm. For the guaranteed traffic buffer requirements are relatively straightforward to solve. The guaranteed traffic together with user parameter control (UPC) and traffic policing should adequately solve most of the problems.
For the best effort traffic the requirements are more based on the control we are suggesting. The question of what is the best suited algorithm for cell discarding is more or less a matter of an opinion. Several relatively equal algorithms exist. Some of them are based on the actual queue occupancy and some on average occupancy. We believe that some of these methods suit quite well for certain types of traffic but to handle different services different algorithms should coexist.

Table 1: Combinations of the buffer management algorithms

	FBA	**EPD**	**PPD**	**RED**
UBR	X			X
IP over CBR		X		
Video over CBR		X	X	

Short descriptions of the algorithms shown in Table 1 are presented on the following chapters. Also in Table 1 are presented the authors' suggestions to combinations of service types and suitable buffer management algorithms to go with them. The natural cause for us to choose these algorithms presented here, is that they suit well for the frame based Internet traffic which is of the essence since we are considering sending IP based traffic in our simplified ATM-environment. Other methods of cell or frame discarding might be implemented if additional non-IP based traffic is to be transmitted in the network.

2.3.1 Early packet discard

Early packet discard (EPD) was proposed by Romanow in [5]. EPD is a scheme where the switch monitors the instantaneous queue length. After a certain predetermined threshold level, the switch starts to discard entire packets - not individual cells from multiple packets. This method is predictive in the manner of operation. It tries to avoid congestion by starting cell discarding before the buffer is full. In this way there is no corrupted packets in buffer and no bandwidth is wasted. Short simulations done by the authors indicate that this method is one of the best ones when traffic relayed in ATM-network is frame based, especially conventional (no multimedia) TCP/IP-connections are quite well managed on EPD alone.

2.3.2 Partial packet discard

Partial packet discard (PPD) is similar to the EPD but it reacts to overflow in the buffer not predicts its possibility. This drop tail congestion control method is far from optimal when communication has frame based structure. Nevertheless it is well suited for connections which have rear possibility of buffer overflow and which do not utilize frame based communication.

Together with EPD and PPD the management of real-time traffic in simplified ATM traffic should be quite well handled, although further research on EPD and PPD needs to be done using actual network traffic.

2.3.3 Fair buffer allocation

Fair buffer allocation (FBA) was proposed by Heinänen and Kilkki [6]. FBA adapts the fundamental idea of EPD with the key issue in the best effort service - fairness. In FBA the switch monitors both the instantaneous queue length of an individual connection *[Y_i]* and the aggregate value of the buffer usage *[X]*. Decision of whether a cell should be discarded is based on two rules; buffer occupancy is exceeding the threshold level *[R]* and the connection is using more than its fair share *[W_i]* of resources. In the calculation of fair share only active connections *[N_a]*, i.e. those having at least one cell in buffer, are taken into account. To smooth the operations in the boundary areas a smoothing factor is calculated based on the buffer size *[K]*, buffer occupancy and threshold level presented in (1).

$$X \geq R \wedge \begin{cases} W_i = \dfrac{Y_i N_a}{X} \\ W_i > Z\left(1 + \dfrac{K - X}{X - R}\right) \end{cases} \tag{1}$$

The weakness of the FBA is that it only works with best effort services. It is not intended to be used with services that need guaranteed bandwidth. Therefore, it should be used when realizing fair UBR services [7].

2.3.4 Random early detection

Random early detection (RED) was proposed by Floyd and Van Jacobson in [8] to be used in Internet routers. RED implements a two parameter approach to the congestion control. First it measures average queue size using exponential decay filter and compares it to the fixed threshold values. If the threshold is smaller than the average queue size, a packet discard probability is calculated and marking is executed based on this probability. If the upper threshold is exceeded all packets are marked. Random early detection is based on the assumption that the fairness among competing connections can be achieved if connections are treated in a random manner. RED also prevents the 'global synchronization' -phenomena present in router systems that use the source quench -messages to all sources sending. RED tries limit the speed for those sources that use the most of the bandwidth and, therefore, it can be stated that the fast connections lose more cells than the slower ones.

3. IP SWITCHING

3.1 Concepts of switching and routing

People quite often mix the concepts of routing and switching. One quite accurate and straight forward explanation is presented in [9]: *'Forwarding (switching) consists of taking packet, looking its destination address, consulting a table, and sending the packet in a direction determined by the table. Routing is the process by which forwarding tables are built.'* This definition quite clearly implies that both switching and routing are viable for an operational network.

Open systems interconnection (OSI) reference model (Figure 5) shows the different levels in which communications networks operate. In the case of Internet people see things from the angle of IP, which is network layer protocol, and protocols up from that. Lower levels may be chosen freely but routers (layer 3 forwarding devices), the basic components of IP networks, operate at a network level. Switching is the main function of the network layer if we keep strictly with the definitions.

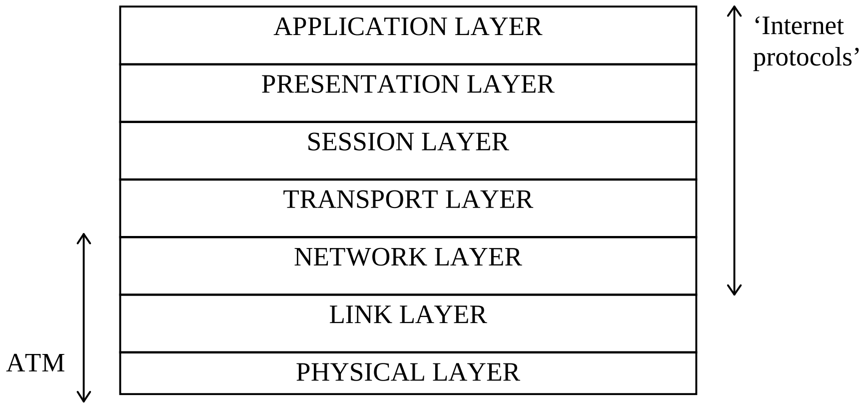


Figure 5: OSI reference model for communications networks

What are then layer two and layer one forwarders? People used to call them bridges and repeaters but definitions change as the networks change. In the case of ATM, layer two forwarder is considered to be the ATM- switch.
So, what has happened is that layer three functionalities have, as ATM has been introduced, dropped down to layer two. By this we mean that addressing is based on the link layer addresses and routing is done according to these addresses.

3.2 Introduction to IP switching

The ongoing growth of Internet requires better performance from routers which do the transmitting of IP datagrams. Routers are an essential part in the Internet and their limited ability to perform routing is creating bottlenecks in the current Internet. Router is a device that usually has separate interfaces to the different networks; local or wide area, as the case may be. Information is usually queued to the route processor in IP level packet by packet. This queue is served based on first come first served (FCFS) scheduling. So a router inherently utilizes statistical multiplexing since its resources are shared between all users. All the links connected to the router are utilized from the common pool by the route processor on a need to use -basis. As is easily seen from the above, the routers tend to use bandwidth resources as efficiently as they in the whole can be used but they waste routing processor resources quite heavily. Current routers make forwarding decisions for each packet separately, but it would obviously be more efficient to make this decision only once, especially if a data flow, or a flow of packets sent to the same destination from one source, contained a large number of packets sent to a specified location for a relatively long time.
Problems in routers relate to their structure. As said route processor is the common resource which is heavily utilized since each packet has to go through it. This sets serious scaling problems in terms of aggregated capacity and possibility to offer QoS. Another huge problem in the current Internet is that the number of hops between two locations may be dozens or even more. This affects routing and especially to the delay which comes from traversing through number of queues and route processors. Also as the IP is not the only network protocol in the world more than one protocol has to be implemented in order to be able to use a router in a mixed environment.
Switching comes to ease these problems. As switches are usually connection oriented they release some of the delay constraints from the links between two routers. Additionally the network protocols are transparent to layer two switches. Switches are usually constructed so that they contain a possibility to offer QoS in addition to the best effort services. QoS in a router based network is claimed to be possible with resource reservation protocol (RSVP) which uses link by link reservation mechanism. It means that routers have to adopt parallel queues for high priority traffic that gets first served by route processor. This indicates a movement towards real layer three switches.
Two approaches to IP switching exist; these are the flow-based and topology-based solutions. These concepts are clarified in Figure 6.

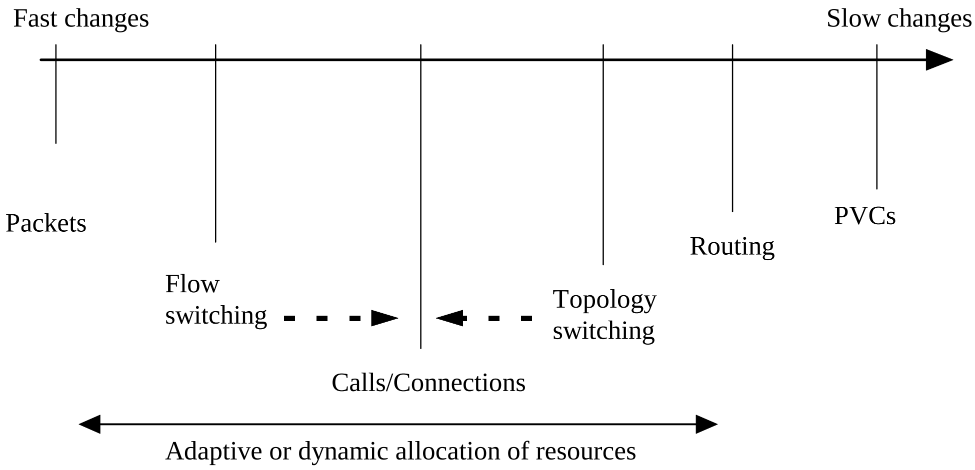


Figure 6: IP switching schemes

Figure 6 illustrates the attack that IP switching has launched against conventional ATM. The concept of the call is modified to two directions: Flow switching may include several flows during a call. This would actually mean that one establishes several calls even to the same destination during one session. Topology switching relies on pre-defined connections based on routing information and traffic monitoring. Here calls might include several connections to the same part of the network. On the other hand topology based switching might be keeping connections up even if no data is being transmitted.

Two major technological solutions have been introduced to implement IP switching. These are the Ipsilon's IP switching and Cisco's Tag switching. Several other solutions including Toshiba's Cell Switch Router (CSR), Telecom Finland's Switching IP through ATM (SITA) and IBM's Aggregate Route-Based IP switching (ARIS) have emerged and standardization has started in the Internet community. All aforementioned solutions aim to offer, one way or the other, the flexibility of routing combined with the speed of ATM switching.

In this paper we base our work on flow-based IP switching because it is, at the moment, the only technology that is commercially available. The basic component in flow based IP switching is the combined router and ATM-switch. A simplified illustration of such a device is illustrated in Figure 7.

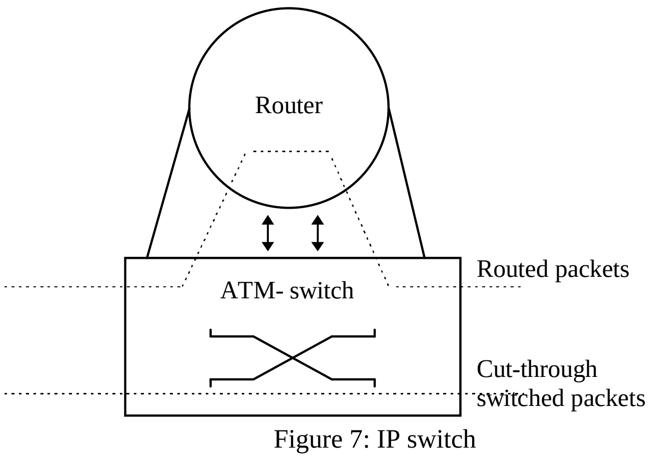


Figure 7: IP switch

The main idea behind IP switching is to provide more bandwidth and a dedicated QoS to applications which need it the most. In Internet world these applications are currently defined by TCP/UDP-port numbers, although in the future special flow specifiers may also be used to identify flows.

The concept of flow, or a series of packets with identical source and destination addresses, and possibly some other parameters in TCP/IP-packets, has been introduced since 1986 in [10] when a parameterizable methodology for traffic flows was introduced. Ever since the concept of flow has been studied and a lot of current work in Internet protocols has adopted the concept [11, 12]. Flow based IP switching is naturally based on detecting IP flows. An IP flow is a series of IP packets that share some common properties, usually the IP address and TCP/UDP-port number. This concept of an IP flow is illustrated in Figure 8.

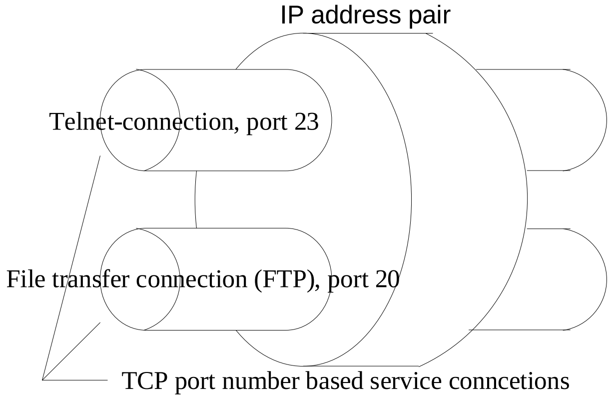


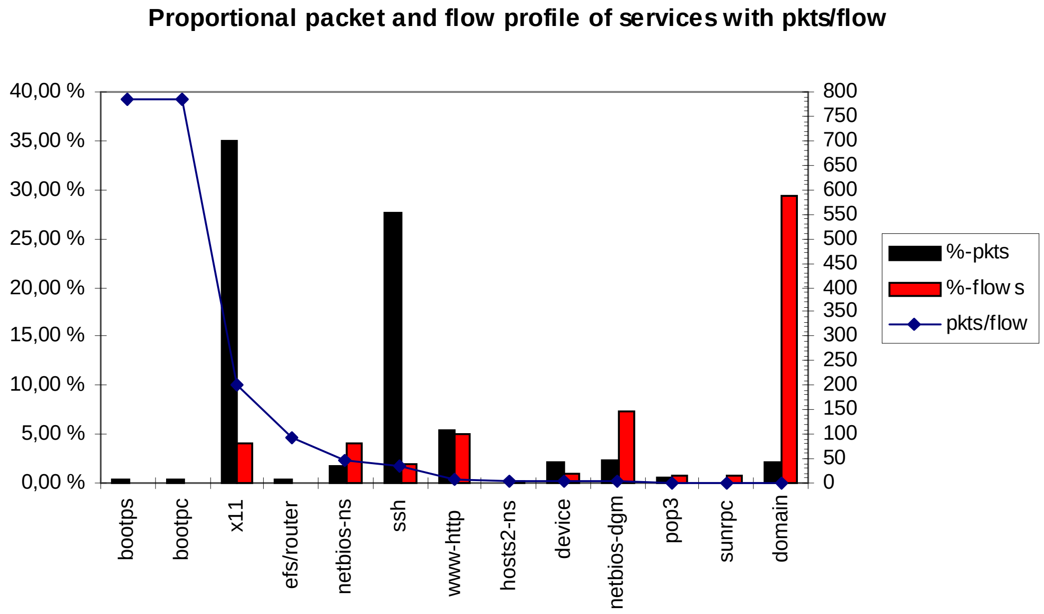
Figure 8: IP flows

Flow switching can be done on any level of granularity. Figure 8 illustrates a deep level of granularity where different TCP-port numbers to an equal IP address constitute different flows. Different IP flows might be considered to require different QoS, e.g. error ratio, transfer delay etc. However, QoS has not been realized in current networks and practically every service used in Internet receives only best-effort QoS. In ATM VPs and VCs are used to manage the available bandwidth and provide different connections different kind of QoS.

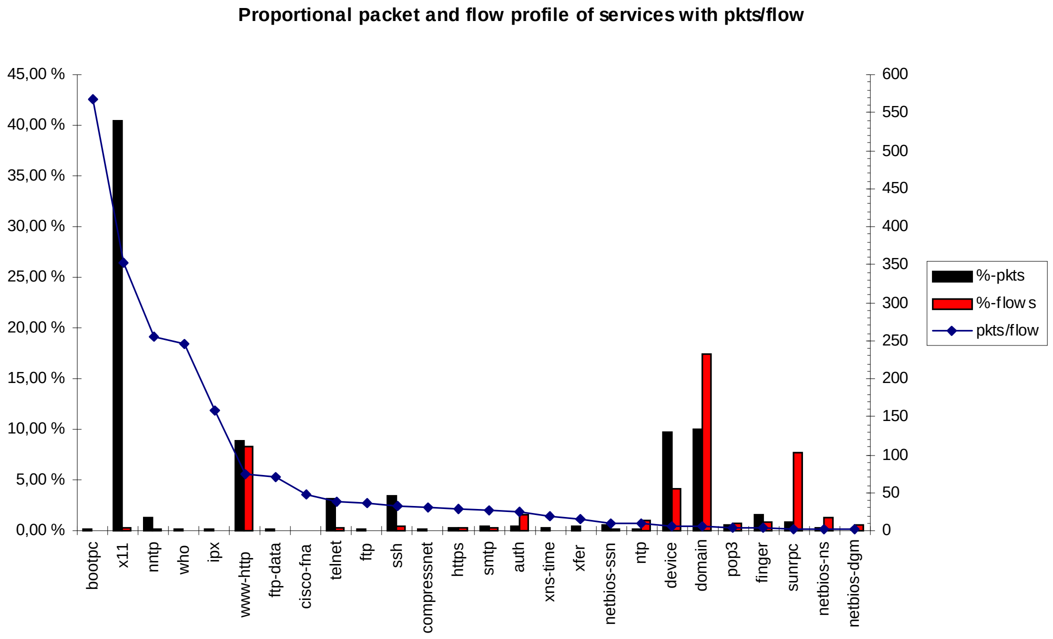
IP switching aims to decrease the workload of routers by assigning long-lived IP flows which contain a relatively large amount of data to transfer to separate virtual paths (VP) or virtual channels (VC). This way it is possible to assign certain services on separate VPs or VCs and offer them at least a better QoS than which can be realized in a default connection. The flow based IP switching is presented in more depth in [13, 14, 15].

3.3 Properties of Internet traffic

The behavior of Internet traffic is self-similar in nature when concerning such parameters as frame arrivals and traffic processes. We have measured two bridged networks which are of different size (contain a different amount of users) and obtained the port numbers and IP addresses from the TCP/IP-packets. Here it should be noted, that unlike in some other studies we have also included the UDP/IP-traffic as eligible for switching. We first analyzed the distribution of TCP-port numbers from the packets and then from the flows formed. Flows were formed if two packets with identical IP address pairs and TCP-port numbers arrived within 60 seconds from each other. After this we determined how many packets on an average were transmitted on respective flows. The results of these analyses are illustrated in pictures 8a) and 8b).



a) small network



b) large network

Figure 9a) and 9b): Packet and flow profiles in networks of different size compared to sent packets/flow

The measurements made indicate that the service profiles relayed in the networks differ slightly. Comparing to other traffic measurements and analysis made in [10, 13, 16] we conclude that differing service profiles exist depending where (geographically or topologically) we are situated in the network and at what time the measurements are made. If we exclude the unassigned port numbers in figures 9a and 9b and focus on the services that might be regarded as interactive we see that although similarities exist the profiles look different although these two networks measured are bridged together. Considering measurements and other traffic analyses performed [10, 13, 16] it seems that although the individual behavior of protocols does not

change the proportions at which they present themselves in actual networks varies with network size, time and place. Also the technology used varies today and even if an ideal situation existed, where network components would have equal capabilities the differences in service profile would be incontestable. This implies that in the wide area sense establishing consistent and absolute QoS throughout a internet connection is practically impossible. Given a conforming environment some sort of guarantees of the effort offered to a connection might be given.
In the future Internet there will be an increasingly larger amount of applications that need a specific QoS. ATM, either simplified or conventional, seems to be able to fulfill such a need. The classifying of Internet flows to ATM is currently a heavily debated issue and in the next chapter we propose some basic guidelines how such mapping of flows to ATM traffic classes might be advanced.

3.4 Classifying flows in simplified ATM

Internet traffic flows can easily be understood as ATM connections. The resources available in an ATM-switch for the Internet traffic should be divided between switched and non-switched traffic according to the proportion between these traffic types according to Figure 10. In the Internet this process should be resource driven i.e. the available resources should dictate what traffic should be switched and what routed.

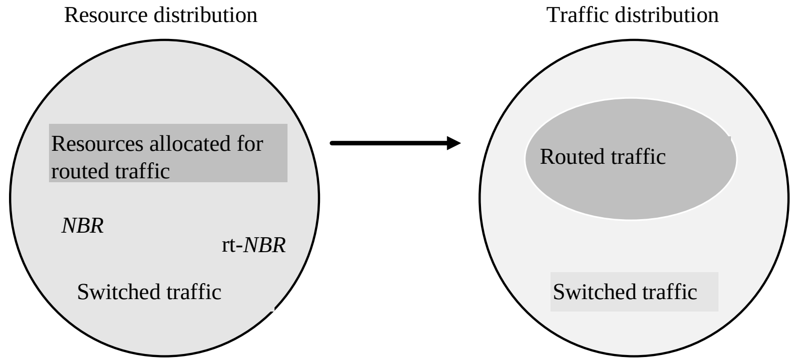


Figure 10: The resource driven traffic shaping

The concept depicted in Figure 10 is based on the fact that routers use a somewhat constant time for the routing of one packet. This offers an absolute amount of resources to be used by the non-switched traffic. The resources of routers should be optimized in a way that guarantees an optimal performance to the routed part of the traffic. This way we could offer an improved QoS to the switched part of the traffic and also an optimal performance to the routed traffic.
Some IP switching solutions disregard ATM signaling altogether. This leads to very difficult situations concerning establishing consistent QoS throughout the connection, although, having the advantage to keep CAC local and therefore enabling flexible QoS in the connection. However the preference would be that some kind of signaling should be preserved together with an appropriate method of CAC designed to understand the dynamic nature of Internet traffic. RSVP might be one answer but in any case this is an issue that demands further research.
Considering this we suggest that two effort-classes to be introduced to the customer. Guaranteed effort would utilize our proposal for rt-NBR traffic class and best effort would be mapped to NBR with varying MCR depending on the service requirements. Further on, if we wish to avoid the load of signaling in the networks, we may map our effort classes quite directly to the defined and accepted RSVP QoS classes[11]. The mapping is illustrated in Figure 11.

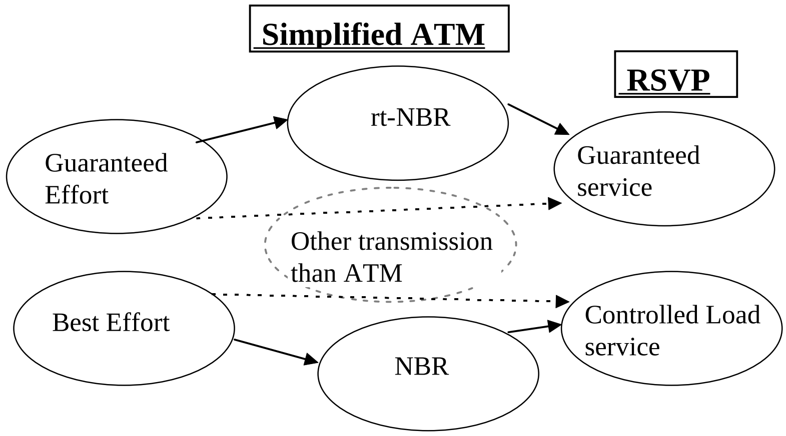


Figure 11:Mapping of QoS classes to simplified ATM and RSVP

Figure 11 also suggests that if ATM is not used as the transmission technology, hence no traffic classes, the mapping of guaranteed effort and best effort traffic to RSVP might still be used. The key issue of how to determine specific parameters from services to traffic classes remains an open issue and demands investigations in the future.

4. CONCLUSIONS

Internet is gaining wide popularity which is resulting in a traffic increase. Several solutions to handle the increase in traffic have been proposed. Of all the work, the ATM-concept seems to be the most promising. However, ATM traffic management has turned to a complex web and is now facing a big dilemma whether to pursue ahead or turn around and see where did it go wrong. We believe that the time to turn around has come. ATM's strength should lie on its simplicity not in the complexity of the traffic management. We have suggested a simplified approach to the traffic management with only two traffic classes: rt-NBR and NBR. We also propose simple, measurement based methods to handle connection admission control and link management. We also propose the use of flow based IP switching to transmit Internet traffic over simplified ATM environment. To back up our claims we have conducted and compared different traffic measurements and gathered that the dimensions of components of Internet traffic are on a constant process of change. We propose a method to allocate router resources between routed and switched traffic and further on specify some frame of reference as to how the different service categories in the Internet might be mapped to our simplified ATM approach. We firmly believe that further research should be done on Internet traffic and simple and efficient traffic management of ATM.

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