Efficient Congestion Management Framework using Compression Techniques

by

KHO Lee Chin

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Supervisor: Associate Professor Yuto LIM

School of Information Science
Japan Advanced Institute of Science and Technology

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Abstract

In this dissertation, congestion control management mechanisms are investigated to introduce a new direction for aiding in solving network congestion. Existing mechanisms that directly or indirectly help to eliminate the congestion problem were being investigated.

The well-known empirical model of TCP connection throughput model has been extended in this thesis to include the effect of compression of part of the traffic throughput. The derived model can help to estimate the performance of congested networks when compression is applied.

The proposed generic ECM framework can be applied to minimize the impact of network congestion by orchestrating different existing congestion management mechanism with the newly introduced compressed MPLS mechanism. The ECM framework presented in this research offers an overall idea on how to help in improving congested networks throughput by reducing the impact of network congestion. In other words, ECM framework can indirectly help congestion points or links of the entire network (Internet). ECM framework can be re-engineered to cooperate with other possible mechanisms (i.e., network coding) for further reduction of network congestion.

The ECM/C model is presented in this research which offers an overall idea on how to utilize compression in the networking model to help in improving congested networks throughput for limited resource devices.
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List of Symbols

Symbols in Chapter 4

\( \alpha \) The number of packets from the beginning of the TD period until and including the first lost packet (unit: packets)

\( \beta \) The number of sent packets in the last round in that specific TD period (unit: packets)

\( \lambda \) The average input rate of that TCP connection (unit: packets/second)

\( \mu \) The service or processing rate of that network device (unit: packets/second)

\( b \) The slope of additive increase in a TCP congestion control scheme

\( p \) Loss probability of sent packets

\( r \) The duration of the round trip time of this specific round in the TD period considered (unit: seconds)

\( t_c \) The compression time (unit: seconds)

\( x \) The number of rounds in a TD period until and including the round where the TD occurs. It is either the number of rounds in that specific TD period or the same number minus one (unit: packets)

\( A \) The time duration of the TD period (unit: seconds)

\( B \) The number of packets in buffer before compression (unit: packets)

\( B' \) The final number of packets in the buffer (unit: [packets])

\( R(p) \) Relationship between the throughput of TCP connection (\( R \)) and loss probability (\( p \))

\( RTT \) Round-trip time of one packet start when the packet of TCP connection is completely transmitted from the sender until its ACK from the receiver is completely received by the sender again

\( T \) Number of packets sent per unit time regardless the eventual fate, either lost or received (unit: packets per second)

\( W \) The maximum window size of that specific TD period, which is the window size of the last or before last round, depending on where did the TD occurred

\( Y \) The number of sent packets in that specific TD period
Symbols in Chapter 6

$S_c$  The size of encoded data (unit: byte)
$S_o$  The size of source data (unit: byte)
$\alpha$ The size of one input symbol (unit: bits)
$R$  The sum of repetitions of all output symbols and codeword
$P_s$ The sum of repetitions of all output symbols that were not compressed
$P_c$ The sum of repetitions of all output codeword that have been compressed
$Q$  The sum of entries in the dictionary
$l_{cmax}$ The codeword length that has the maximum number of bits of a codeword
$l_{cmin}$ The codeword length that has the minimum number of bits of a codeword
$\bar{l}_c$ The codeword length that has the average number of bits of a codeword
List of Abbreviations

ABR  Available Bit Rate
ACK  Acknowledgement
Africa  Adaptive and Fair Rapid Increase Congestion Avoidance
AIMD  Additive Increase Multiplicative Decrease
AQM  Active Queue Management
ATM  Asynchronous Transmission Mode
B    Buffer size
BBN  Bolt, Beranek and Newman
BDP  Bandwidth-Delay Product
BIC  Binary Increase Congestion control
BISDN  Broadband Integrated Services Digital Network
BMP  BitMaP
CC   Congestion Control
CA   Congestion Avoidance
CAD  Connection Admission Control
CBR  Constant Bit Rate
CDV  Cell Delay Variation
CER  Cell Error Ratio
CID  Context IDentifier
CLP  Cell Loss Priority
CLR  Cell Loss Rate
CP   Compression Percentage
CR   Compression Ratio
CRTP  Compression Real Time Protocol
CTCP  Compound TCP
CTD  Cell Transfer Delay
D    Delay-based
DiffServ  Differentiated Services
DNS  Domain Name System
DOOR  Detection of Out-of-Order and Response
DSACK  Duplicate Selective ACKnowledgement
ECM  Efficient Congestion Management
EFCI  Explicit Forward Congestion Indicator
ESRT  Event-to Sink Reliable Transport
FDDI  Fiber Distributed Data Interface
FIFO  First In First Out
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<td>FR</td>
<td>Fast Recovery</td>
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<td>FWestwood</td>
<td>Fuzzy Westwood</td>
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<td>GFC</td>
<td>Generic Flow Control</td>
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<tr>
<td>HEC</td>
<td>Head Error Control</td>
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<td>HSTCP</td>
<td>High-Speed TCP</td>
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<td>HTTP</td>
<td>Hypertext Transfer Protocol</td>
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<td>ICTCP</td>
<td>Incast Congestion control for TCP</td>
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<td>IoT</td>
<td>Internet of Things</td>
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<td>IP</td>
<td>Internet Protocol</td>
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<td>IPHC</td>
<td>Internet Protocol Header Compression</td>
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<td>ISDN</td>
<td>Integrated Services Digital Network</td>
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<td>ISP</td>
<td>Internet Service Provider</td>
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<td>JPEG</td>
<td>Joint Photographic Expert Group</td>
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<td>L</td>
<td>Loss-based</td>
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<td>LAN</td>
<td>Local Area Network</td>
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<td>LBE</td>
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<td>MAN</td>
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<td>MAC</td>
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<td>MCR</td>
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<td>MIMA</td>
<td>Multiplicative Increase Multiplicative Decrease</td>
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<td>MSR</td>
<td>Molecular Sequence Reduction</td>
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<tr>
<td>MTU</td>
<td>Maximum Transmission Unit</td>
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nrt-VBR  non-real Variable Bit Rate
PCR    Peak Cell Rate
PTI    Payload Type Identifier
QoS    Quality of Service
RCP    Receiver Centric Protocol
RED    Random Early Detection
RLE    Run Length Encoding
ROHC   RObust Header Compression
RSVP   Resource reservation Protocol
RTO    Retransmission timeout
RTP    Real Time Protocol
RTT    Round Trip Time
rt-VBR real time Variable Bit Rate
SACK   Selective ACKnowledgement
SCP    Sender Centric Protocol
SF     Shannon-Fano
SMTP   Simple Mail Transfer Protocol
SNR    Signal to Noise Ratio
SSL    Secure Socket Layer
STCP   Scalable TCP
TCP/IP  Transmission Control Protocol / Internet Protocol
TCPW BR Transmission Control Protocol Westwood with Bulk Repeat
TCPW CRB Transmission Control Protocol Westwood with Combined Rate
         and Bandwidth estimation
TD     Triple Duplicate
TD-FR  Time Delayed Fast Recovery
TDP    Triple Duplicate Period
TFRC   TCP Friendly Rate Control
TM     Traffic Management
Tri-S  Slow Start and Search
UBR    Unspecified Bit Rate
UDP    User Datagram Protocol
UNI    User Network Interface
UPC    Usage Parameter Control
VCI    Virtual Channel Identifier
Vegas A Vegas with Adaptation
VJHC   Van Jacobson’s Header Compression
VoIP   Voice over Internet Protocol
Veno   VEgas and ReNo
VPI    Virtual Path Identifier
W      Window Size
WAN    Wide Area Network
WSN    Wireless Sensor Network
XCP    eXplicit Control Protocol
Chapter 1

Introduction

This chapter first briefly introduces the congestion problems in the communication networks. The background of some of the existing congestion control techniques are presented here together with the research motivation. The objective and contribution of this research are also provided, which is followed by the dissertation outline.

1.1 Congestion Problems in Communication Networks

A computer network is collection of autonomous computers interconnected by a single technology. Two computers are said to be interconnected if they are able to exchange information. The data in some communication networks is transmitted in the form of messages, which could be referred to as packets during the transmission process. These packets are sent across the network based on the communication protocols, and reconstructed at the destination [1].

The size and complexity of communication network can be classified into four main types: small network, Local Area Network (LAN), Metropolitan Area Network (MAN), and Wide Area Network (WAN). Small networks commonly connect the sub-assemblies or devices, while LAN connects the distributed terminals and computer equipment in restricted areas, such as University campuses. MAN is a high speed network that is used to interconnect LANs in a geographical region, such as within a city. Meanwhile, WAN is multiple communication connections for a large geographical area, for example, Internet is connected via a large global network of service providers using the servers, modems, routers and switches.

Internet has had a big influence on human daily life for years. It is used at home for personal usage entertainment, communication, banking, shopping, and devices control. It is also used at work as professional tool. Fig. 1.1 illustrates the growth of Internet based on the time-line from year 2004 to 2015 [2]. One important time-stone is in 2010, when mobile broadband connections exceeded fixed broadband connection. The time-line also shows that the number of Internet users increased from one billion in 2005 to almost three billion by 2015.

The rapid growth of Internet users increases traffic loads, at the same time, networks are targeting full utilization to maintain revenue and service pricing. Consequently, the problem of network congestion is likely to occur in certain situations. Network congestion is one of the unending problems that can occur when capacity of an underlying sub-network is insufficient for the demanded amount of data. The growth of demand
will eventually go beyond the Service Provider’s ability to efficiently cope with the huge data traffic. As a result, the network might face tremendous and unpredictable network congestion. When network congestion occurs, the quality of service (QoS) and energy efficiency in the network will degrade. In other words, if traffic load ever goes beyond the Internet’s capacity, Internet will face tremendous and unpredictable network congestion. When a resource (buffer or bandwidth) is shared with multiple users who contend to access that resource beyond its availability, network congestion may occur. In other word, the network congestion can be defined as when the incoming traffic demand to some sub-network (node or link) exceeds the capacity of that sub-network (buffers and output capacity), congestion may occur.

Network congestion occurs due to various reasons. It can be generally categorized into predictable events and unpredictable events or alternatively into random congestion and recurrent congestion. Predictable events generate additional Internet demand to the already existing traffic load, like when users accessing global online debut of big branded products. Unexpected life threatening events such as floods, tornadoes, and earthquakes can increase the network traffic erratically and/or suddenly remove vital network resources. This kind of unpredictable events can last for few hours, days or weeks. Random congestion can occur when a number of users are sharing a sub-network and all become highly active within a very short period of time. Random congestion is mainly due to bad design of network resource that is over-utilized above the safe statistical nature of traffic. Finally, recurrent congestion occurs due to the normal pattern of daily activities that create an overall significant traffic load in recurrent time. For instance, higher usage on business hours for commerce and trading servers, similarly higher usage of residential users in the evening to entertainment service providers.

During congestion, the quality of service (QoS) in Internet will be degraded. When a network is congested, the delay time and packet loss increase, and the throughput decreases. The delay time might increase due to packets suffering from long queuing till timing-out. For example, if the worst case for satellite network transmission is 160 ms round-trip, when a voice call experiences 200 ms, the talk will be overlapping. Packet loss in congestion happens when a packet is dropped because of long waiting time due to unavailability of the next buffer. Although the lost packets would be retransmitted with Transmission Control Protocol (TCP), still, the overall network throughput decreases.
1.2 Research Background and Motivation

Network congestion basically can be resolved by three possible methods: increasing physical output bandwidth, increasing physical size of local buffering and implementing better network flow management. When physical output of a network is increased, i.e., the transmission rate is increased, the problem of congestion which mainly caused by an extremely slow link would be solved. However, it is usually not that simple, when some links are upgraded without sufficient studying and planning, worse congestion can reappear in some nearby link due to bad network resources balancing.

When the capacity of buffering is insufficient, incoming packets which wait too long before being allowed to enter the buffer might experiences times-out and get dropped. If the buffer size is enlarged, more packets will be allowed to enter and stored, this can mitigate the congestion problem. However, larger buffer size can introduce longer delay again causing timing-out and congestion again.

Network congestion is a dynamic problem; the two manual methods mentioned are not always useful and definitely not responsive enough. That leaves only the last method which has been actively pursued by researchers for nearly three decades, which is the network flow management to handle congestion. In short, the congestion management. The goal of any network flow management is to ensure that the network is utilized as efficiently as possible. In other word, the highest possible network throughput shall be achieved while trying to avoid over-utilization and its associated problems such as congestion.

Different flow management approaches has been introduced over the years to address the problem of congestion. Congestion control handles the level of traffic entry to the system. The initial model of congestion control is based on microeconomic theory and convex optimization theory, where individuals controlling flow rates can interact to obtain an optimal network-wide rate allocation. Later many distributed network optimization algorithms extended that initial model. However, the initial model has a weakness: it assumes the entire flow is controlled by the same parameter, and different flows will be controlled by different parameters.

Congestion avoidance such as Random Early Detection (RED) and fair queuing are commonly used. In RED mechanism, the average queue size is monitored and the packets are pre-dropped according to statistical probabilities to save space for other packets and explicit messages are sent to inform the sender to decrease the transmission rate. RED is simple to implement without global synchronization and can provide high link utilization. However, the performance is highly sensitive to the controlling parameters; threshold, drop probability and weight in RED algorithm and might cause buffer overflow or slow responsiveness to congestion. Fair queuing is a scheduling algorithm which allows the capacity of the link to be fairly shared by multiple connections; it can prevent high speed data connection from swamping links of insufficient capacity. Priority schemes, is one of the network congestion mending functionalities which allow some packets to be transmitted with higher priority than others. These schemes cannot solve network congestion directly. They only help to relieve congested network through providing some services.

A wide range of network congestion mechanisms have been introduced by researchers around the world; some of current network congestion mechanisms have included fuzzy logic or neural network. Adaptive route changing algorithms have been also presented to alleviate congestion by modifying the routes to steer traffic in the most effective way.
The utilization of network coding in congestion control has also been studied. TCP Vegas with online network coding (TCP VON) uses one of the examples of congestion control mechanisms.

From the point of view of congestion control mechanisms, the term congestion control is wider than just a way to get rid of congestion, but rather to the way for better utilization of the network resources. The goal of researchers is still almost the same, which is to improve the efficiency of network resources usage while providing safety measures or margins to avoid over-utilization.

Here, I emphasize again, congestion control actually refers to the way of effectively use the network resources, not just congestion management itself. Network technologies have reached blazing speeds, Ethernet technology advanced from IEEE802.3a 10 Mbps in 1985 to IEEE802.3ba 100 Gbps in 2010 [3]. Also, wireless LAN technology increased from IEEE802.11a 54 Mbps in 1999 to IEEE802.11n 600 Mbps in 2009 [4]. Still, the congestion sometime occurs due to the unbalance load distribution.

Most of the existing congestion control mechanisms is about ‘how to eliminate congestion’, focusing on one or more scenarios of special cases. But congestion is a dynamic problem which might occur due to different network scenarios and events. This motivates us to propose an efficient congestion management framework, which is about ‘how to better utilize the available mechanisms’ by adaptive selection of the suitable congestion mechanism, trying to improve resources utilization. The research introduces a new approach for solving congestion, compression is studied to reduce the effective size of data and possibly to improve performance.

To implement compression in the proposed congestion management framework, the following research problems are carefully considered.

- Lengthy compression time even for the fastest existing hardware compression schemes.
- Highly sensitive compression degree depending on the fragile existing correlation in any data based on bot spatial and temporal locality principals of information theory.
- Enormous temporally memory requirements for compression and sometimes decompression as well.
- The conflicting nature of the mentioned above three problems, meaning improving one will definitely degrade the other two.

### 1.3 Research Vision and Objectives

Congestion can occur at any time and place in the Internet. Fig. 1.2 shows the possibility of congestion in the Internet. Current congestion management is usually implemented as an integral part of flow management which ensures the network resources are being fully utilized all the time. However, there will never be a perfect flow control mechanism that can make sure that the network resource is fully utilized and at the same time avoid congestion completely. This can easily be realized from the researches’ continuous development of new traffic control mechanisms.

The vision of this research is to propose an efficient congestion management framework to minimize the impact of congestion, without wasting much network resources. To accomplish this vision, the research objectives are summarized as follows:
• To review the existing network congestion management, especially the ATM and TCP/IP networks.

• To derive an empirical model throughput of congested network with compression capability.

• To design an efficient congestion management framework making use of the compression.

• To propose an efficient congestion management with compression for small devices.

1.4 Summary of Contribution

This research investigates the efficiency of compression techniques in networks from the perspective of throughput. The network congestion management framework (ECM) is proposed for network devices. Although the framework model is quite simple, the work provides valuable insight contribution to the existing research on congestion management. The main contributions of this research can be summarized as follows:

• An empirical model for network throughput with compression capability to study the feasibility of compression in networks.

• A generic congestion management framework is proposed that can be applied to minimize the impact of network congestion by using the existing mechanisms coupled with compression on top of Multi-Protocol Label Switching-Traffic Engineering (MPLS-TE).

• A simple compression scheme with lightweight decompression for devices with limited resources is proposed.

Both ECM framework and ECM/C model proposed show opportunities of using compression in networking such as in the Internet. Both models can co-exist and be applied in different parts of the network simultaneously, or with limited overlapping.
1.5 Structure of the Dissertation

The dissertation is organized as follow:

Chapter 1 (this chapter) introduces the basic concept of congestion and the congestion problems in networks. The motivation and some research backgrounds are described. Also a summary of the contributions of this research are presented.

Chapter 2 reviews some of the existing network congestion management mechanisms used in the Internet. First, the devices in general networks are distinguished, and then traffic management methods are reviewed. Almost 34 of TCP variants are used to explain in-depth end-to-end congestion control.

Chapter 3 reviews some of the common existing compression schemes and provides thorough classification. Example schemes are later explained and discussed to show the suitable chances of application. Finally, the generic compression metrics are presented together with derivation of a relation between overall compression degree and part-wise compressed data compression degrees.

Chapter 4 formulates an empirical model for connection throughput of compression capable networks. This throughput model is the extension of an existing model with finite buffer. The operating curves of the model are also shown.

Chapter 5 proposes an Efficient Congestion Management (ECM) framework for networking devices to minimize the impact of network congestion when it occasionally occurs. The architecture of ECM mainly consists of a congestion classifier. The new congestion control mechanisms introduced together with the compression.

Chapter 6 introduces an ECM/C model together with Dictionary-based Lightweight DeCompression (LDC) scheme that is proven to minimize the impact of congestion in small limited resources devices. Those devices which have small bandwidth and small buffering capabilities are the main concerned here. The four stages of LDC; dictionary building, encoding, dictionary loading and decompression are explained. Analysis of the LDC scheme performance due to the affected parameters is also presented.

Chapter 7 summarizes the work in this dissertation and provides insight into the future work.
Chapter 2

Network Congestion Management

In this chapter, the background knowledge of network congestion management is discussed. Some network architectures can suffer from congestion while other designs managed to avoid any possibility of occurrence. An example of each network class is presented in this chapter. Some of the existing congestion management techniques are also introduced while talking about the congestion vulnerable class of networks.

2.1 Introduction

Network congestion can occur when different data streams flowing in from multiple links to a router, require a single resource. Since congestion can only happen in nodes with multiple input links and at least one output resource, it can happen in any device except for end devices. For example, when core Internet devices encounter more traffic than they can handle. The single resource which is competed for here might be memory like a router buffer for traffic, or bandwidth of the output link. In both cases, data experiencing prolonged delays (buffer waiting) would eventually timeout (be dropped). When reliable data is required and the network is congested, the more data being re-sent after dropping, the more time need for successful transfer. This would waste the already consumed energy, bandwidth and memory resources which that data already utilized until the point of dropping. Not to mention the additional overall delay that data would accumulate besides the retransmission delay.

Congestion can occur mainly due to one of two reasons. First, when the amount of data traffic admitted into the whole network doesn’t exceed the whole network capacity for data transfer. The second is attributed to poor balance routing of data traffic across the network links. Known routing techniques are still far from perfection, thus frequently fail to balance traffic among the links. Imperfect routing causes congestion by over utilized some parts of the network for some profit (financial, political, energy ...), while other parts are underutilized.

Fig. 2.1 illustrates an example of Internet congestion. Internet is comprised of three elements: end device, edge device, and core device. An end device can be any machine with some application running. An edge device (router) provides an entry point to the lower level devices for the higher level devices. Core devices (extremely fast big routers) form the core Internet and interconnect it to lowest level edge devices (router). In Internet links, data only flows from one end device to another end device. According to what was previously mentioned, congestion in Internet would only occur in edge or core devices but
not end devices.

Any network that might experiences congestion will be operating in one of the three main stages; normal traffic flow stage and two congestion stages. When congestion is detected, the network will go through a congestion mending stage, followed by the recovery stage. Some networks will additionally go through congestion avoidance stage after optional congestion prediction. The naming of those stages has varied a lot throughout the literature. Fig. 2.2 (a) shows the network congestion state diagram of the operation of congestion vulnerable networks in the different congestion stages.

To handle the network operation during those stages, four different basic functions must be provided by the traffic flow management. Those four functions are; normal flow control, congestion detection, congestion mending manoeuvre and congestion recovery flow control. The correspondence between the stages and the functions is clear from the naming. Two additional optional functions are provided in the networks capable of running in optional avoidance stage. Those two functions are congestion prediction and avoidance flow control. Both congestion management and its functions have also been referred to in the literature with different names. The functions responsible for controlling the flow during the four network congestion stages are shown with solid boxes in Fig. 2.2 (b). Those functions are basic normal flow control, congestion mending manoeuvre and congestion recovery flow control, besides the optional avoidance flow control.

Both the congestion detection and congestion prediction functions, which is showed with dotted boxes in Fig. 2.2 (b), are responsible for switching the operation of the network from normal flow stage into either congestion mending stage or congestion avoidance stage, respectively. Switching between either stage is usually straight forward after carrying out the congestion mending manoeuvre or avoidance flow control, respectively. During the avoidance stage, congestion detection can also switch the network operation into the congestion mending stage.

Many techniques and standards have been introduced to resolve network congestion problems or avoid it. Some network standards were designed from the beginning to avoid any possibility of congestion occurrence, ATM and MPLS are among those network standards. In the next subsection, ATM as an example network that totally avoids any congestion is introduced. The following subsection will introduce TCP network as an
example of network architectures experiencing congestion and how it is managed. The transmission control protocol (TCP) and Internet Protocol (IP) are actually two different network protocols, TCP and IP are normally implemented together. The terms TCP/IP or TCP are going to be used interchangeably throw out this thesis and considered to be equivalent referring to either the whole Internet or only the transport layer.

2.2 Congestion Free Networks

ATM is one of the technologies that are targeted to meet the Broadband Integrated Services Digital Network (BISDN) requirements. The BISDN is an extension of Integrated Service Digital Network (ISDN). In ISDN, the end-to-end digital connectivity for data such as audio, video, and data applications are provided. While in BISND, a wide range of applications that require higher transmission rate are supported. The BISND that uses ATM technique is also called as ATM networks. Almost all other networks mainly targeting voice connection services are also congestion free networks such as Public Switched Telephone Network (PSTN). Some data network as MPLS are also designed to provide congestion free traffic.

2.2.1 Asynchronous Transfer Mode (ATM)

According to the ATM reference model as shown in Fig. 2.3, it consists of three main layers, which are physical layer, ATM layer, and ATM adaption layer [5, 6]. The figure shows the ATM layers side by side with the corresponding OSI model layers. The physical layer is constructed by two sub-layers; physical medium (upper sub-layer) which provides bit transmission capability, and transmission convergence (lower sub-layer) which provides the functions of transmission frame generation and extraction, transmission frame adaption, cell rate decoupling, cell delineation, and Head Error Control (HEC) signal generation and confirmation. The ATM layer is mainly responsible for the cells traffic management and congestion control in the network. In ATM adaption layer, the seg-
mentation is performed to split the higher layer information into a size of cells at the end devices of sender and reassembles back into data units at the end device of receiver devices before being delivered to higher layer.

![ATM Reference Model](image)

Figure 2.3: ATM Reference Model (courtesy of [7])

The information in the ATM network is transmitted in a short fixed length of 53 byte cells. The 5 bytes are used for ATM header and the remaining 48 bytes are the payload of encapsulation information. The ATM cell structure is shown in Fig. 2.4, where it consists of six header fields, Generic Flow Control (GFC), Virtual Path Identifier (VPI), Virtual Channel Identifier (VCI), Payload Type Identifier (PTI), Cell Loss Priority (CLP), and Header Error Control (HEC). The GFC field is used to control the traffic flow across the User Network Interface (UNI). The VPI defines the virtual paths between sender and receiver for a particular cell. The VCI is used to identify a channel path for a particular cell. By combining both information in the field of VPN and VCI, a virtual circuit for a specified ATM cell is identified. In an idle cell, all the bits of VPI and VCI are set to 0’s. The PTI is responsible to identify the type of ATM cell that follows, to indicate whether the cell experienced congestion in its journey, and to determine the last cell in a block for ATM adaptive layer for user ATM cells. The CLP is used as priority indicator, and HEC is used for error detection.

<table>
<thead>
<tr>
<th>GFC (4 bits)</th>
<th>VPI (4 bits)</th>
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<tbody>
<tr>
<td>VPI (4 bits)</td>
<td>VCI (4 bits)</td>
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<tr>
<td>VCI (8 bits)</td>
<td></td>
</tr>
<tr>
<td>VCI (4 bits)</td>
<td>PTI (3 bits)</td>
</tr>
<tr>
<td>HEC (8 bits)</td>
<td></td>
</tr>
<tr>
<td>Data Payload (48 bytes)</td>
<td></td>
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</table>

Figure 2.4: The ATM cell structure
The ATM network provides several services to the users, according to the traffic type and the transmission method. The ATM service includes Constant Bit Rate (CBR), real time Variable Bit Rate (rt-VBR), non-real time Variable Bit Rate (nrt-VBR), Unspecified Bit Rate (UBR), and Available Bit Rate (ABR) [5, 8-10]. CBR provides the service of transmitting a constant bit rate of information. This service is mainly for the applications that need a fixed and continuous data during the connection lifetime and have a tight upper bound of transfer delay, such as Skype, YouTube, Live TV and so on. The rt-VBR service is for the real time applications that have limited time delay and variable data rate transmission. On the other hand, the nrt-VBR service is for the applications that consist of bursty traffic characteristics and do not require tightly constrained delay variation. The UBR service is for tolerate variable delays and cell losses acceptable applications. During congestion, no feedback mechanism is concerning, the cells are lost and their source will not reduce the transmission rate to overcome the congestion. In ABR, some parameters in the used applications need to be specified, for instances Peak Cell Rate (PCR), Minimum Cell Rate (MCR), Cell Loss Rate (CLR), and so on. An explicit feedback message is used in the ABR to inform the status of congestion and/or the rate of to be transmitted data.

In ATM network, QoS is an important issue [8-10]. When a connection is established in ATM network, a contract about the given services that is related to various QoS parameters must be agreed by both user and the network. These parameters include Peak Cell Rate (PCR), Minimum Cell Rate (MCR), Cell Loss Rate (CLR), Cell Transfer Delay (CTD), Cell Delay Variation (CDV), and Cell Error Ratio (CER). The PCR is the maximum rate of cell that sender plans to send. The MCR is the minimum rate that a user can accept. CLR is the percentage of cells loss in the network. The cells are considered lost even though they did reach the destination, if they were received with an invalid header, or the content of cells has been corrupted by errors. The measured CTD can be defined as the time elapse between the departure time of a cell from sender end devices and arrival time at destination. The CTD includes propagation delays, internal delays (switching, processing, and internal transmission link), external queuing and transmission delays. The CDV indicates the uniformity of cells deliver, and CER is the fraction of cells that is corrupted during delivery.

Until here, the basic structure of ATM network has briefly explained. The following section will focus on the traffic management mechanism on ATM network.

### 2.2.1 Traffic Management in ATM network

The ATM network is designed to support variety of services and applications. The control of ATM mainly involves providing proper differentiated QoS for the network applications. As the traffic management of ATM, it is used to ensure the network and the end devices are out from congestion problems, so that high network performance is achieved. The traffic management is also used to promote the efficient network resources utilization. It ensures the efficient and fair operation in the networks, meanwhile fulfills the demand and QoS that users desired. In short, traffic management is a set of mechanisms that ensures the network resources are fully utilizes and meet the various QoS as part of traffic contract. The traffic management mechanisms for ATM networks are depicted in Fig. 2.5, which include connection admission control, usage parameter control, traffic shaping, selective cell discard, explicit forward congestion indication, and network resource management [8-10].