UNIT 1

Introduction to distributed systems

Distributed System: Definition

A distributed system is a piece of software that ensures that: *A collection of independent*

Computers that appears to its users as a single coherent system

Two aspects:

(1) Independent computers and

(2) Single system _ **middleware**.

Distributed computing is a field of computer science that studies distributed systems. A **distributed system** consists of multiple autonomous computers that communicate through a computer network. The computers interact with each other in order to achieve a common goal. A computer program that runs in a distributed system is called a **distributed program**, and **distributed programming** is the process of writing such programs.

Introduction:

The word *distributed* in terms such as "distributed system", "distributed programming", and "distributed algorithm" originally referred to computer networks where individual computers were physically distributed within some geographical area. The terms are nowadays used in a much wider sense, even when referring to autonomous processes that run on the same physical computer and interact with each other by message passing. While there is no single definition of a distributed system, the following defining properties are commonly used:

There are several autonomous computational entities, each of which has its own local memory.

The entities communicate with each other by message passing. In this article, the computational entities are called *computers* or *nodes*. A distributed system may have a common goal, such as solving a large computational problem. Alternatively, each computer may have its own user with individual needs, and the purpose of the distributed system is to coordinate the use of shared resources or provide communication services to the users. Other typical properties of distributed systems include the following:

The system has to tolerate failures in individual computers.

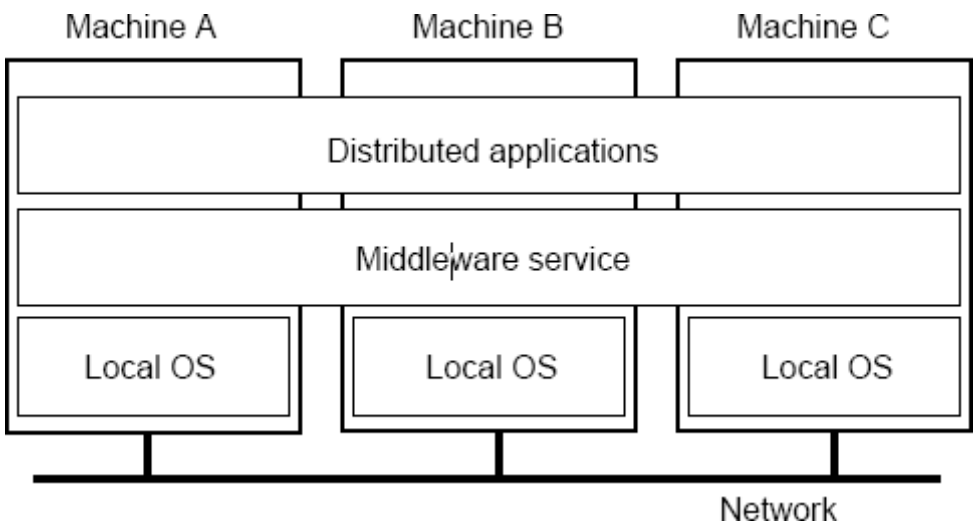
The structure of the system (network topology, network latency, number of computers) is not known

in advance, the system may consist of different kinds of computers and network links, and the system may change during the execution of a distributed program.

Each computer has only a limited, incomplete view of the system. Each computer

may know only one part of the input.

Architecture for Distributed System:



Various hardware and software architectures are used for distributed computing. At a lower level, it is necessary to interconnect multiple CPUs with some sort of network, regardless of whether that network is printed onto a circuit board or made up of loosely coupled devices and cables. At a higher level, it is necessary to interconnect processes running on those CPUs with some sort of communication system. Distributed programming typically falls into one of several basic architectures or categories: client–server, 3-tier architecture, *n*-tier architecture, distributed objects, loose coupling, or tight coupling.

Client–server: Smart client code contacts the server for data then formats and displays it to the user. Input at the client is committed back to the server when it represents a permanent change.

3-tier architecture: Three tier systems move the client intelligence to a middle tier so that stateless clients can be used. This simplifies application deployment. Most web applications are 3-Tier.

n-tier architecture: *n*-tier refers typically to web applications which further forward their requests to other enterprise services. This type of application is the one most responsible for the success of application servers.

Tightly coupled (clustered): refers typically to a cluster of machines that closely work together, running a shared process in parallel. The task is subdivided in parts that are made individually by each one and then put back together to make the final result.

Peer-to-peer: an architecture where there is no special machine or machines that provide a service or manage the network resources. Instead all responsibilities are uniformly divided among all machines, known as peers. Peers can serve both as clients and servers.

Space based: refers to an infrastructure that creates the illusion (virtualization) of one single address-space. Data are transparently replicated according to application needs. Decoupling in time, space and reference is achieved. Another basic aspect of distributed computing architecture is the method of communicating and coordinating work among concurrent processes. Through various message passing protocols, processes may communicate directly with one another, typically in a master/slave relationship. Alternatively, a "database-centric" architecture can enable distributed computing to be done without any form of direct inter process communication, by utilizing a shared database.

Goals of Distributed system

- Connecting resources and users
- Distribution transparency
- Openness
- Scalability

Hardware and Software concepts

Hardware Concepts

1. Multiprocessors
2. Multi computers
3. Networks of Computers

DistinguishingFeatures:

- Private versus shared memory
- Bus versus switched interconnection

Networks of Computers

High degree of node heterogeneity:

- High-performance parallel systems (multiprocessors as well as multicomputers)
- High-end PCs and workstations (servers)
- Simple network computers (offer users only network Access)
- Mobile computers (palmtops, laptops)
- Multimedia workstations

High degree of network heterogeneity:

- Local-area gigabit networks
- Wireless connections
- Long-haul, high-latency connections
- Wide-area switched megabit connections

Observation: Ideally, a distributed system hides these Differences

Software Concepts

- Distributed operating system
- Network operating system
- Middleware

Distributed Computing Model

Many tasks that we would like to automate by using a computer are of question–answer type: we would like to ask a question and the computer should produce an answer. In theoretical computer science, such tasks are called computational problems. Formally, a computational problem consists of *instances* together with a *solution* for each instance. Instances are questions that we can ask, and solutions are desired answers to these questions. Theoretical computer science seeks to understand which computational problems can be solved by using a computer (computability theory) and how efficiently (computational complexity theory). Traditionally, it is said that a problem can be solved by using a computer if we can design an algorithm that

produces a correct solution for any given instance. Such an algorithm can be implemented as a computer program that runs on a general-purpose computer: the program reads a problem instance from input, performs some computation, and produces the solution as output. Formalisms such as random access machines or universal Turing machines can be used as abstract models of a

Sequential general-purpose computer executing such an algorithm. The field of concurrent and distributed computing studies similar questions in the case of either multiple computers, or a computer that executes a network of interacting processes: which computational problems can be solved in such a network and how efficiently? However, it is not at all obvious what is meant by "solving a problem" in the case of a concurrent or distributed system: for example, what is the task of the algorithm designer, and what is the concurrent and/or distributed equivalent of a sequential general-purpose computer?

The discussion below focuses on the case of multiple computers, although many of the issues are the same for concurrent processes running on a single computer. Three viewpoints are commonly used:

Parallel algorithms in shared-memory model

□□ All computers have access to a shared memory. The algorithm designer chooses the program executed by each computer.

□□ One theoretical model is the parallel random access machines (PRAM) are used. However, the classical PRAM model assumes synchronous access to the shared memory.

□□ A model that is closer to the behavior of real-world multiprocessor machines and takes into account the use of machine instructions such as Compare-and-swap (CAS) is that of *asynchronous shared memory*. There is a wide body of work on this model, a summary of which can be found in the literature.

Parallel algorithms in message-passing model

□□ The algorithm designer chooses the structure of the network, as well as the program executed by each computer.

□□ Models such as Boolean circuits and sorting networks are used. A Boolean circuit can be seen as a computer network: each gate is a computer that runs an extremely simple computer program. Similarly, a sorting network can be seen as a computer network: each comparator is a computer.

Distributed algorithms in message-passing model

□□ The algorithm designer only chooses the computer program. All computers run the same program. The system must work correctly regardless of the structure of the network.

□□ A commonly used model is a graph with one finite-state machine per node. In the case of distributed

algorithms, computational problems are typically related to graphs. Often the graph that describes the structure of the computer network *is* the problem instance. This is illustrated in the following example.

An example

Consider the computational problem of finding a coloring of a given graph *G*. Different fields might take the following approaches:

Centralized algorithms

The graph *G* is encoded as a string, and the string is given as input to a computer. The computer program finds a coloring of the graph, encodes the coloring as a string, and outputs the result.

Parallel algorithms

Again, the graph *G* is encoded as a string. However, multiple computers can access the same string in parallel. Each computer might focus on one part of the graph and produce a coloring for that part.

The main focus is on high-performance computation that exploits the processing power of multiple computers in parallel.

Distributed algorithms

The graph *G* is the structure of the computer network. There is one computer for each node of *G* and one communication link for each edge of *G*. Initially, each computer only knows about its immediate neighbors in the graph *G*; the computers must exchange messages with each other to discover more about the structure of *G*. Each computer must produce its own colour as output.

The main focus is on coordinating the operation of an arbitrary distributed system. While the field of parallel algorithms has a different focus than the field of distributed algorithms, there is a lot of interaction between the two fields. For example, the Cole– Vishkin algorithm for graph colouring was originally presented as a parallel algorithm, but the same technique can also be used directly as a distributed algorithm. Moreover, a parallel algorithm can be implemented either in a parallel system (using shared memory) or in a distributed system (using message passing). The traditional boundary between parallel and distributed algorithms (choose a suitable network vs. run in any given network) does not lie in the same place as the boundary between parallel and distributed systems (shared memory vs. message passing).

Advantages & Disadvantage distributed system

1: Incremental growth: Computing power can be added in small increments

2: Reliability: If one machine crashes, the system as a whole can still survive

3: Speed: A distributed system may have more total computing power than a mainframe

4: Open system: This is the most important point and the most characteristic point of a distributed system. Since it is an open system it is always ready to communicate with other systems. An open system that scales has an advantage over a perfectly closed and self-contained system.

5. Economic: Microprocessors offer a better price/performance than mainframes

Disadvantages of Distributed Systems over Centralized ones

1:As it is previously told you distributed systems will have an inherent security issue.

2:Networking: If the network gets saturated then problems with transmission will surface.

3:Software:There is currently very little less software support for Distributed system.

4:Troubleshooting: Troubleshooting and diagnosing problems in a distributed system can also become more difficult, because the analysis may require connecting to remote nodes or inspecting communication between nodes.

Issues in designing Distributed System

□□ Secure communication over public networks ACI: who sent it, did anyone see it, did anyone change it

□□ Fault-tolerance : Building reliable systems from unreliable components □nodes fail independently; a distributed system can “partly fail” [Lamport]: “A distributed system is one in which the failure of a machine I’ve never heard of can prevent me from doing my work.” Replication, caching, naming Placing data and computation for effective resource sharing, hiding latency, and finding it again once you put it somewhere. Coordination and shared state □What should the system components do and when should they do it? □ Once they’ve all done it, can they all agree on what they did and when?

Synchronization

Clock synchronization is a problem from computer science and engineering which deals with the idea that internal clocks of several computers may differ. Even when initially set accurately, real clocks will differ after some amount of time due to clock drift, caused by clocks counting time at slightly different rates. There are several problems that occur as a repercussion of rate differences and several solutions, some being more appropriate than others in certain contexts. In serial communication, some people use the term "clock synchronization" merely to discuss getting one metronome-like clock signal to pulse at the same frequency as another one frequency synchronization and phase synchronization. Such "clock synchronization" is used

in synchronization in telecommunications and automatic baud rate detection.

Process scheduling

Preemptive scheduling is widespread in operating systems and in parallel processing on symmetric multiprocessors. However, in distributed systems it is practically unheard of. Scheduling in distributed systems is an important issue, and has performance impact on parallel processing, load balancing and meta computing. Non-preemptive scheduling can perform well if the task lengths and processor speeds are known in advance and hence job placement is done intelligently

Deadlock handling

Deadlocks in Distributed Systems: Deadlocks in Distributed Systems Deadlocks in distributed systems are similar to deadlocks in single processor systems, only worse. They are harder to avoid, prevent or even detect. They are hard to cure when tracked down because all relevant information is scattered over many machines. People sometimes might classify deadlock into the following types: Communication deadlocks -- competing with buffers for send/receive Resources deadlocks -- exclusive access on I/O devices, files, locks, and other resources. We treat everything as resources; there we only have resources deadlocks. Four best-known strategies to handle deadlocks: The ostrich algorithm (ignore the problem) Detection (let deadlocks occur, detect them, and try to recover) Prevention (statically make deadlocks structurally impossible) Avoidance (avoid

deadlocks by allocating resources carefully)

The FOUR Strategies for handling deadlocks : The FOUR Strategies for handling deadlocks The ostrich algorithm No dealing with the problem at all is as good and as popular in distributed systems as it is in single-processor systems. In distributed systems used for programming, office automation, process control, no system-wide deadlock mechanism is present -- distributed databases will implement their own if they need one. Deadlock detection and recovery is popular because prevention and avoidance are so difficult to implement. Deadlock prevention is possible because of the presence of atomic transactions. We will have two algorithms for this. Deadlock avoidance is never used in distributed system, in fact, it is not even used in single processor systems. The problem is that the banker's algorithm need to know (in advance) how much of each resource every process will eventually need. This information is rarely, if ever, available. Hence, we will just talk about deadlock detection and deadlock prevention.

Load Balancing

Resource Scheduling (RS)

RS continuously monitors utilization across resource pools and intelligently aligns resources with business needs, enabling you to:

□□ Dynamically allocate IT resources to the highest priority applications. Create rules and policies to prioritize how resources are allocated to virtual machines.□□Give IT autonomy to business organizations. Provide dedicated IT infrastructure to business units while still achieving higher hardware utilization through resource pooling.

□□ Empower business units to build and manage virtual machines within their resource pool while giving central IT control over hardware resources.

File Sharing

File sharing is the practice of distributing or providing access to digitally stored information, such as computer programs, multi-media (audio, video), documents, or electronic books. It may be implemented through a variety of storage, transmission, and distribution models and common methods of file sharing incorporate manual sharing using removable media, centralized computer file server installations on computer networks, World Wide Web-based hyper linked documents, and the use of distributed peer-to-peer (P2P) networking. The Distributed File System is used to build a hierarchical view of multiple file servers and shares on the network. Instead of having to think of a specific machine name for each set of files, the user will only have to remember one name; which will be the 'key' to a list of shares found on multiple servers on the network. Think of it as the home of all file shares with links that point to one or more servers that actually host those shares. DFS has the capability of routing a client to the closest available file server by using Active

Directory site metrics. It can also be installed on a cluster for even better performance and reliability. Medium to large sized organizations are most likely to benefit from the use of DFS - for smaller companies it is simply not worth setting up since an ordinary file server would be just fine.

Concurrency Control

In computer science, especially in the fields of computer programming (see also concurrent programming, parallel programming), operating systems (see also parallel computing), multiprocessors, and databases, **concurrency control** ensures that correct results for concurrent operations are generated, while getting those results as quickly as possible. **Distributed concurrency control** is the concurrency control of a system distributed over a computer network.

Failure handling

In a distributed system, **failure transparency** refers to the extent to which errors and subsequent recoveries of hosts and services within the system are invisible to users and applications. For example, if a server fails, but users are automatically redirected to another server and never notice the failure, the system is said to

exhibit *high failure transparency*.

Failure transparency is one of the most difficult types of transparency to achieve since it

is often difficult to determine whether a server has actually failed, or whether it is simply responding very slowly. Additionally, it is generally impossible to achieve full failure transparency in a distributed system since networks are unreliable.

Configuration

Dynamic system configuration is the ability to modify and extend a system while it is running. The facility is a requirement in large distributed systems where it may not be possible or economic to stop the entire system to allow modification to part of its hardware or software. It is also useful during production of the system to aid incremental integration of component parts, and during operation to aid system evolution.

UNIT III

In computer networking, an **Internet socket** or **network socket** is an endpoint of a bidirectional inter-process communication flow across an Internet Protocol based computer network, such as the Internet. The term *Internet sockets* is also used as a name for an application programming interface (API) for the TCP/IP protocol stack, usually provided by the operating system. Internet sockets constitute a mechanism for delivering incoming data packets to the appropriate application process or thread, based on a combination of local and remote IP addresses and port numbers. Each socket is mapped by the operating system to a communicating application process or thread.

A **socket address** is the combination of an IP address (the location of the computer) and a port (which is mapped to the application program process) into a single identity, much like one end of a telephone connection is the combination of a phone number and a particular extension. An Internet socket is characterized by a unique combination of the following:

□□ **Protocol:** A transport protocol (e.g., TCP, UDP), raw IP, or others. TCP port 53 and UDP port 53 are different, distinct sockets.

□□ **Local socket address:** Local IP address and port number

□□ **Remote socket address:** Only for established TCP sockets. As discussed in the Client-Server section below, this is necessary since a TCP server may serve several clients concurrently. The server creates one socket for each client, and these sockets share the same local socket address.

Sockets are usually implemented by an API library such as Berkeley sockets, first introduced in 1983. Most implementations are based on Berkeley sockets, for example Winsock introduced in 1991. Other socket API implementations exist, such as the STREAMS-based Transport Layer Interface (TLI). Development of

application programs that utilize this API is called socket programming or network programming. These are examples of functions or methods typically provided by the API library:

socket() creates a new socket of a certain socket type, identified by an integer number, and allocates system resources to it.

bind() is typically used on the server side, and associates a socket with a socket address structure, i.e. a specified local port number and IP address.

listen() is used on the server side, and causes a bound TCP socket to enter listening state.

connect() is used on the client side, and assigns a free local port number to a socket. In case of a TCP socket, it causes an attempt to establish a new TCP connection.

accept() is used on the server side. It accepts a received incoming attempt to create a new TCP connection from the remote client, and creates a new socket associated with the socket address pair of this connection.

send() and recv(), or write() and read(), or recvfrom() and sendto(), are used for sending and receiving data to/from a remote socket.

close() causes the system to release resources allocated to a socket. In case of TCP, the connection is terminated.

gethostbyname() and gethostbyaddr() are used to resolve host names and addresses.

select() is used to prune a provided list of sockets for those that are ready to read, ready to write or have errors

poll() is used to check on the state of a socket. The socket can be tested to see if it can be written to, read from or has errors.

Data Representation & Marshaling

In computer science, **marshalling** (similar to serialization) is the process of transforming the memory representation of an object to a data format suitable for storage or transmission. It is typically used when data must be moved between different parts of a computer program or from one program to another. The opposite, or reverse, of marshalling is called *unmarshalling* (or *demarshalling*, similar to *deserialization*).

Group Communication

Computer systems consisting of multiple processors are becoming commonplace. Many companies and institutions, for example, own a collection of workstations connected by a local area network (LAN). Although the hardware for distributed computer systems is advanced, the software has many problems. We believe that one of the main problems is the communication paradigms that are used. This thesis is

concerned with software for distributed computer systems. In it, we will study an abstraction, called *group communication* that simplifies building reliable efficient distributed systems. We will discuss a design for group communication, show that it can be implemented efficiently, and describe the design and implementation of applications based on group communication. Finally, we will give extensive performance measurements. Our goal is to demonstrate that group communication is a suitable abstraction for distributed systems.

Client Server Communication

Client–server model of computing is a distributed application structure that partitions tasks or workloads between service providers, called servers, and service requesters, called clients. Often clients and servers communicate over a computer network on separate hardware, but both client and server may reside in the same system. A server machine is a host that is running one or more server programs which share its resources with clients. A client does not share any of its resources, but requests a server's content or service function. Clients therefore initiate communication sessions with servers which await (*listen* for) incoming requests.

RPC- Implementing RPC Mechanism

In computer science, a **remote procedure call** (**RPC**) is an Inter-process communication that allows a computer program to cause a subroutine or procedure to execute in another address space (commonly on another computer on a shared network) without the programmer explicitly coding the details for this remote interaction. That is, the programmer writes essentially the same code whether the subroutine is local to the executing program, or remote. When the software in question uses object oriented principles, RPC is called **remote invocation** or **remote method invocation**.

The steps in making a RPC

1. The client calling the Client stub. The call is a local procedure call, with parameters pushed on to the stack in the normal way.

2. The client stub packing the parameters into a message and making a system call to send the message. Packing the parameters is called marshaling.

3. The kernel sending the message from the client machine to the server machine.

4. The kernel passing the incoming packets to the server stub.

5. Finally, the server stub calling the server procedure. The reply traces the same in other direction

Stub Generation, RPC Messages.

Following figure illustrates the basic operation of RPC. A client application issues a normal procedure call to a *client stub*. The client stub receives arguments from the calling procedure and returns arguments to the

calling procedure. An argument may instantiate an input parameter, an output parameter, or an input/output parameter. In the discussion of this Section, the term *input argument* refers to a parameter which may be either an input parameter or an input/output parameter, and the term *output argument* refers to either an output parameter or an input/output parameter. The client stub converts the input arguments from the local data representation to a

common data representation, creates a message containing the input arguments in their common data representation, and calls the client runtime, usually an object library of routines that supports the functioning of the client stub. The client runtime transmits the message with the input arguments to the server runtime which is usually an object library that supports the functioning of the server stub. The server runtime issues a call to the *server stub* which takes the input arguments from the message, converts them from the common data representation to the local data representation of the server, and calls the server application which does the processing. When the server application has completed, it returns to the server stub the results of the processing in the output arguments. The server stub converts the output arguments from the data representation of the server to the common data representation for transmission on the network and encapsulates the output arguments into a message which is passed to the server runtime. The server runtime transmits the message to the client runtime which passes the message to the client stub. Finally, the client stub extracts the arguments from the message and returns them to the calling procedure in the required local data representation.

Synchronization: - Clock Synchronization

Clock synchronization is a problem from computer science and engineering which deals with the idea that internal clocks of several computers may differ. Even when initially set accurately, real clocks will differ after some amount of time due to clock drift, caused by clocks counting time at slightly different rates. There are several problems that occur as a repercussion of rate differences and several solutions, some being more appropriate than others in certain contexts. In a centralized system the solution is trivial; the centralized server will dictate the system time. Cristian's algorithm and the Berkeley Algorithm are some solutions to the clock synchronization problem in a centralized server environment. In a distributed system the problem takes on more complexity because a global time is not easily known. The most used clock synchronization solution on the Internet is the Network Time Protocol (NTP) which is a layered client-server architecture based on UDP message passing. Lamport timestamps and Vector clocks are concepts of the logical clocks in distributed systems.

Cristian's algorithm

Cristian's algorithm relies on the existence of a time server. The time server maintains its clock by using a radio clock or other accurate time source, then all other computers in the system stay synchronized with it. A time client will maintain its clock by making a procedure call to the time server. Variations of this algorithm make more precise time calculations by factoring in network propagation time.

Berkeley algorithm

This algorithm is more suitable for systems where a radio clock is not present, this system has no way of making sure of the actual time other than by maintaining a global average time as the global time. A time server will periodically fetch the time from all the time clients, average the results, and then report back to the clients the adjustment that needs be made to their local clocks to achieve the average. This algorithm highlights the fact that internal clocks may vary not only in the time they contain but also in the clock rate. Often, any client whose clock differs by a value outside of a given tolerance is disregarded when averaging the results. This prevents the overall system time from being drastically skewed due to one erroneous clock.

Network Time Protocol

This algorithm is a class of mutual network synchronization algorithm in which no master or reference clocks are needed. All clocks equally participate in the synchronization of the network by exchanging their timestamps using regular beacon packets. CS-MNS is suitable for distributed and mobile applications. It has been shown to be scalable, accurate in the order of few microseconds, and compatible to IEEE 802.11 and similar standards.

Reference broadcast synchronization

This algorithm is often used in wireless networks and sensor networks. In this scheme, an initiator broadcasts a reference message to urge the receivers to adjust their clocks.

Mutual Exclusion,

Assumptions

The system consists of *n* processes; each process *Pi* resides at a different processor Each process has a critical section that requires mutual exclusion Basic Requirement If *Pi* is executing in its critical section, then no other process *Pj* is executing in its critical section The presented algorithms ensure the mutual exclusion execution of processes in their critical sections. Mutual exclusion must be enforced: only one process at a time is allowed in its critical section A process that hales in its non critical section must do so without interfering with other processes. It must not be possible for a process requiring access to a critical section to be delayed indefinitely: No deadlock or starvation When no process is in a critical section, any process that requests entry to its critical section must be permitted to enter without delay No assumptions are made about relative process speeds or number of processors A process remains inside its critical section for a finite time Only

Election Algorithms:- Bully & Ring Algorithms

□□ There are at least two basic strategies by which a distributed system can adapt to failures.

□□ Operate continuously as failures occur and are repaired

□□ The second alternative is to temporarily halt normal operation and to take some time out to reorganize the system.

□□ The reorganization of the system is managed by a single node called the coordinator.

□□So as a first step in any reorganization, the operating or active nodes must elect a coordinator.

□□ Similar

□□ Like Synchronization, all processors must come to an agreement about who enters the critical region (i.e. who is the leader)

□□ Different

□□ The election protocol must properly deal with the case of a coordinator failing. On the other hand, mutual exclusion algorithms assume that the process in the critical region (i.e., the coordinator) will not fail.

□□ A new coordinator must inform all active nodes that it is the coordinator. In a mutual exclusion algorithm, the nodes not in the critical region have no need to know what node is in the region.

- The two classical election algorithms by Garcia-Molina
- Bully Algorithm
- Invitation Algorithm
- Ring Algorithm

Election algorithms

We often need one process to act as a coordinator. It may not matter which process does this, but there should be group agreement on only one. An assumption in election algorithms is that all processes are exactly the same with no distinguishing characteristics. Each process can obtain a unique identifier (for example, a machine address and process ID) and each process knows of every other process but does not know which is up and which is down.

Bully algorithm

The bully algorithm selects the process with the largest identifier as the coordinator. It works as follows:

1. When a process *p* detects that the coordinator is not responding to requests, it initiates an election:

a. *p* sends an *election* message to all processes with higher numbers.

b. If nobody responds, then *p* wins and takes over.

c. If one of the processes answers, then *p*'s job is done.

2. If a process receives an *election* message from a lower-numbered process at any time, it:

a. sends an OK message back.

b. holds an election (unless its already holding one).

3. A process announces its victory by sending all processes a message telling them that it is the new coordinator.

4. If a process that has been down recovers, it holds an election.

Ring algorithm

The ring algorithm uses the same ring arrangement as in the token ring mutual exclusion algorithm, but does not employ a token. Processes are physically or logically ordered so that each knows its successor.

- If any process detects failure, it constructs an *election* message with its process I.D. (e.g. network address and local process I.D.) and sends it to its successor.

 If the successor is down, it skips over it and sends the message to the next party. This process is repeated until a running process is located.

 At each step, the process adds its own process I.D. to the list in the message. Eventually, the message comes back to the process that started it:

1. The process sees its ID in the list.

2. It changes the message type to *coordinator*.

3. The list is circulated again, with each process selecting the highest numbered ID in the list to act as coordinator.

4. When the *coordinator* message has circulated fully, it is deleted. Multiple messages may circulate if multiple processes detected failure. This creates a bit of overhead but produces the same results.

Unit V

Distributed Shared Memory (DSM), also known as a **distributed global address space** (**DGAS**), is a term in computer science that refers to a wide class of software and hardware implementations, in which each node of a cluster has access to shared memory in addition to each node's non-shared private memory.

Software DSM systems can be implemented in an operating system, or as a programming library. Software DSM systems implemented in the operating system can be thought of as extensions of the underlying virtual memory architecture. Such systems are transparent to the developer; which means that the underlying distributed memory is completely hidden from the users. In contrast, Software DSM systems implemented at the library or language level are not transparent and developers usually have to program differently. However, these systems offer a more portable approach to DSM system implementation. Software DSM systems also have the flexibility to organize the shared memory region in different ways. The page based approach organizes shared memory into pages of fixed size. In contrast, the object based approach organizes the shared memory region as an abstract space for storing shareable objects of variable sizes. Other commonly seen implementation uses tuple space, in which unit of sharing is a tuple.

Shared memory architecture may involve separating memory into shared parts distributed amongst nodes and main memory; or distributing all memory between nodes. A coherence protocol, chosen in accordance with a consistency model, maintains memory coherence.

Examples of such systems include:

 Delphi DSM

 JIAJIA

 Kerrighed

 NanosDSM

- OpenSSI
- MOSIX
- Strings
- Terracotta
- TreadMarks
- DIPC
- Intel Cluster OpenMP is internally a Software DSM.
- ScaleMP ?
- RNA networks

DSM Architecture & its Types,

- DSM Subsystem
- DSM Server
- KEY Server

DSM Subsystem

- Routines to handle page faults relating to virtual addresses corresponding to a DSM region.
- Code to service system calls which allow a user process to get, attach and detach a DSM region.
- Code to handle system calls from the DSM server.

DSM Server

- In-server :Receives messages from remote DSM servers and takes appropriate action. (E.g. Invalidate its copy of a page)
- Out-server :Receives requests from the DSM subsystem and communicates with its peer DSM servers at remote nodes. Note that the DSM subsystem itself does

not directly communicate over the network with other hosts.

- Communication with key Server.

Key Sever

Each region must be uniquely identifiable across the entire LAN. When a process executes system call with a key and is the first process at that host to do so, the key server is consulted.

Key server‘s internal table is looked-up for the key, if not found then it stores the specified key in the table as a new entry.

Design & Implementations issues In DSM System

There are various factors that have to be kept in mind while designing and implementing the DSM systems. They are as follows:

1. Block Size:

As we know, transfer of the memory blocks is the major operation in the DSM systems. Therefore block size matters a lot here. Block size is often referred to as the **Granularity**. Block size is the unit of sharing or unit of data transfer in the event of network block fault. Block size can be few words, pages or few pages. Size of the block depends on various factors like, paging overhead, thrashing, false sharing, and directory size.

2. Structure of Shared Memory Space:

How the shared memory space is organized with data determines the structure of the shared memory space. It refers to the layout of shared data. It depends upon the type of application the DSM is going to handle.

3. Replacement Strategy:

It may happen that one node might be accessing for a memory block from DSM when its own local memory is completely full. In such a case, when the memory block migrating from remote node reaches, it finds no space to get placed. Thus a replacement strategy of major concern in the design and implementation of the DSM systems. Certain block must be removed so as to place the new blocks in such a situation. Several techniques are used for the replacement of old blocks such as removal of **Least Recently Used** memory blocks.

4. Thrashing:

Sometimes two or more processes might access the same memory block. Suppose two processes need to perform write operation on the same memory block. Since, to accomplish this, the block has to migrate in both directions at a very small interval of time, so it will be transferred back and forth at such a high rate that none of the two processes will be able to perform the operation accurately and completely. As such no real work is done. This condition is called thrashing. A technique should be incorporated, while designing the

DSM systems to avoid thrashing.

5. Heterogeneity:

DSM systems should be able to function on computers with different architectures. Issues Involved in

DSM Issues

- Network Communication
- Consistency
- Data Granularity
- Coherence

Consistency Model

• Strict consistency in shared memory systems.

• Sequential consistency in shared memory systems –Our focus.

• Other consistency protocols

• Casual consistency protocol.

• Weak and release consistency protocol

Unit VI

Desirable features of good Distributed File System

In computing, a **distributed file system** or **network file system** is any file system that allows access to files from multiple hosts sharing via a computer network. This makes it possible for multiple users on multiple machines to share files and storage resources. The client nodes do not have direct access to the underlying block storage but interact over the network using a protocol. This makes it possible to restrict access to the file system depending on access lists or capabilities on both the servers and the clients, depending on how the protocol is designed. In contrast, in a shared disk file system all nodes have equal access to the block storage where the file system is located. On these systems the access control must reside on the client. Distributed file systems may include facilities for transparent replication and fault tolerance. That is, when a limited number of nodes in a file system go offline, the system continues to work without any data loss. The difference between a distributed file system and a distributed data store can be

vague, but DFSes are generally geared towards use on local area networks.

Features

DFS offers many features that make managing multiple file servers much simpler and effective.

Unified Namespace

DFS links multiple shared folders on multiple servers into a folder hierarchy. This hierarchy is same as a physical directory structure on a single hard disk. However, in this case, the individual branch of the hierarchy can be on any of the participating servers.

Location Transparency

Even if the files are scattered across multiple servers, users need to go to only one network location. This is a very powerful feature. Users do not need to know if the actual file location has changed. There is no need to inform everyone about using new paths or server names! Imagine how much time and energy this can save. It reduces downtime required during server renames, planned or unplanned shutdowns and so on.

Continuous Availability

As mentioned, during planned shutdowns, the file resources can be temporarily made available from another standby server, without users requiring to be notified about it. This way downtime related to maintenance or disaster recovery tasks is completely eliminated. This is very useful especially in Web servers. The Web server file locations can be configured in such a way that even when the physical location of the files changes to another server, the HTML links continues to work without breaking.

Replication

It is possible to replicate data to one or more servers within the DFS structure. This way, if one server is down, files will be automatically served from other replicated locations. What's more, users will not even know the difference.

Load Balancing

This is a conceptual extension of replication feature. Now that you can put copies of the same file across multiple locations. If the file is requested by more than one user at the same time, DFS will serve it from different locations. This way, the load on one server is balanced across multiple servers, which increases performance. At a user level, they do not even come to know that the file came from a particular replica on DFS.

Security

DFS utilises the same NTFS security and file sharing permissions. Therefore, no special configuration is required to integrate base security with DFS.

Ongoing hard disk space management

What happens when your hard disk space is exhausted? You typically add another hard disk. Now, this hard disk will have another name. What if this disk is on another server? Things would get worse. With DFS, you can keep adding new directories to the namespace on completely separate servers. Users never have to bother about the physical server name. This way, you can grow your storage in steps without having to worry about destabilizing file access by users.

Unifying heterogeneous platforms

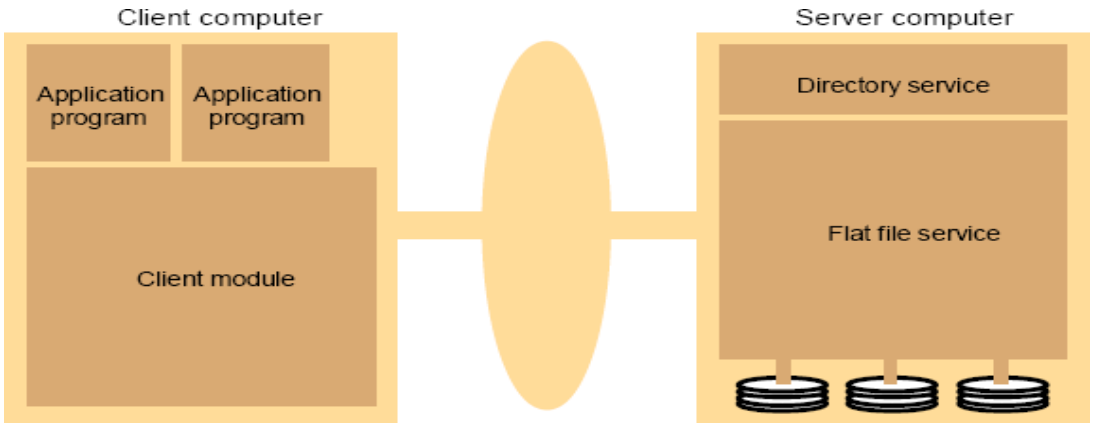
DFS also supports NetWare. This way, administrators can unify data access by combining servers running heterogeneous operating systems from a file access perspective.

Fault tolerance or higher availability of data

DFS works with clustering services. This combination offers higher availability than just using clustering.

File Model

The Distributed File System is used to build a hierarchical view of multiple file servers and shares on the network. Instead of having to think of a specific machine name for each set of files, the user will only have to remember one name; which will be the 'key' to a list of shares found on multiple servers on the network. Think of it as the home of all file shares with links that point to one or more servers that actually host those shares. DFS has the capability of routing a client to the closest available file server by using Active Directory site metrics. It can also be installed on a cluster for even better performance and reliability. Medium to large sized organizations are most likely to benefit from the use of DFS - for smaller companies it is simply not worth setting up since an ordinary file server would be just fine.



File Service Architecture : Following figure shows the architecture of file service. Client computer can communicate with server using client module to flat file service.

File Sharing Semantics

Sequential semantics: Read returns result of last write

Easily achieved if

• Only one server

• Clients do not cache data

BUT

• Performance problems if no cache

• Obsolete data

– We can **write-through**

• Must notify clients holding copies

• Requires extra state, generates extra traffic

Session semantics

Relax the rules

• Changes to an open file are initially visible only to the process (or machine) that modified it.

• Last process to modify the file wins.

Asynchronous Transfer Mode (ATM)

connection-oriented technology, in which a connection is established between the two endpoints before the actual data exchange begins.

ATM provides a highly complex technology, with features intended for applications ranging from global telco networks to private local area computer networks. ATM has been a partial success as a technology, with widespread deployment, but generally only used as a transport for IP traffic; its goal of providing a single integrated end-to-end technology for LANs, public networks, and user services has largely failed. However, as it often happens in technology development, various important ATM concepts have been inherited by other technologies, such as MPLS.

To accelerate the deployment of ATM technology, the ATM Forum, a consortium of service providers and equipment vendors in the communication industries was created to develop implementation and specification agreements. Later, ATM Forum was merged with other industry forums to form MPLS Frame Relay ATM (MFA) forum [23]. In this chapter, we present a brief overview on ATM protocol layers, the current status on Traffic Management, and discuss related technologies such as MPLS, as well as technologies using the ATM protocol layer stack, such as DSL, FTTP, and UMTS.

Keywords: Switching, traffic management, reference model, ATM.

1 Introduction

The purpose of this chapter is to introduce the reader to the basic aspects of asynchronous transfer mode (ATM) networks. The length of this short chapter makes it impossible to cover all important aspects of ATM networks. Much of the material in this chapter is based on existing tutorials on ATM, including [4, 3, 8, 12, 14, 23, 17, 51, 64, 58, 42, 37]. The industrial momentum behind ATM technology and the intensive research interest in ATM has led to a vast and diversified literature in recent years. Most of the cited references are mainly review articles or documents of ATM and MFA Forums [23]. Interested readers in further understanding of the individual topics are referred to the corresponding papers and the references therein.

1.1 Basic Principles

Various network applications are requiring increasingly higher bandwidth and generating a heterogeneous mix of network traffic. Existing networks cannot provide the transport facilities to efficiently support a diversity of traffic with various service requirements. ATM was designed to be potentially capable of supporting heterogeneous traffic (e.g., voice, video, data) in one transmission and switching fabric technology. It promised to provide greater integration of capabilities and services, more flexible access to the network, and more efficient and economical service.

ATM is a switching and multiplexing technology that employs small, fixed-length packets (called cells). Each cell has 5 bytes of header information and a 48-byte information field

(payload). The reason for choosing a fixed-size packet was to ensure that the switching and multiplexing function could be carried out quickly, easily, and with least delay variation. The reason for choosing a small size cell was mainly a result of the need to support delay- intolerant interactive voice service (e.g., phone calls) with a a small packetization delay, i.e., the time needed to fill a cell with PCM (pulse code modulation) encoded voice samples arriving at the rate of 64 Kbps.

ATM is a connection-oriented technology in the sense that before two systems on the network can communicate, they should inform all intermediate switches about their service require- ments and traffic parameters. This is similar to the telephone networks where a fixed path is set up from the calling party to the receiving party. In ATM networks, each connection is called a virtual circuit or virtual channel (VC), because it also allows the capacity of each link to be shared by connections using that link on a demand basis rather than by fixed allocations. The connections allow the network to guarantee the quality of service (QoS) by limiting the number of VCs. Typically, a user declares key service requirements at the time of connection setup, declares the traffic parameters, and may agree to control these parameters dynamically as demanded by the network.

ATM was intended to provide a single unified networking standard that could support both synchronous and asynchronous technologies and services, while offering multiple levels of quality of service for packet traffic.

ATM sought to resolve the conflict between circuit-switched networks and packet-switched networks by mapping both bit streams and packet streams onto a stream of small fixed-size

"cells" tagged with virtual circuit identifiers. Cells are typically sent on demand within a synchronous time slot pattern in a synchronous bit stream: what is asynchronous here is the sending of the cells, not the low-level bitstream that carries them.

In its original conception, ATM was to be the enabling technology of the "broadband in- tegrated services digital network" (B-ISDN) that would replace the existing narrowband "integrated services digital network (ISDN). The full suite of ATM standards provides defi- nitions for layer 1 (physical connections), layer 2 (data link layer), and layer 3 (network) of the classical OSI seven-layer networking model. Because ATM is asynchronous, it provides true bandwidth-on-demand. Additionally, ATM is capable of handling any form of informa- tion (e.g., data, voice, video, audio, e-mail, faxes), moving this information quickly across a network with millions of virtual paths and channels between end-user equipment

ATM allows the user to select the required level of service, provides guaranteed service quality, and makes reservations and preplans routes so those transmissions needing the most attention are given the best service.

1.2 The MFA Forum, ITU, and ANSI

With the objective of accelerating the convergence of standards and industry cooperation, an international consortium called the ATM Forum was founded to ensure interoperability between public and private ATM implementations and to promote the use of ATM products and services. Although it was not a standard body, the ATM Forum worked closely with standard organizations such as the International Telecommunications Union (ITU) and Inter-

net Engineering Task Force (IETF) in developing the definitions for ATM standards. In 2005 the ATM Forum was merged in MPLS Frame Relay and ATM Forum — MFA Forum, which is an international, industry-wide, nonprofit association of telecommunications, networking, and other companies focused on advancing the deployment of multi-vendor, multi-service packet-based networks, associated applications, and interworking solutions [23].

The ITU is rooted in the International Telegraphy Union, founded in Paris in 1865. Its name changed in 1934, and in 1947 the ITU became an agency of the United Nations. The ITU works with public and private organizations to develop earth-linked and satellite communications, while developing standards for all types of telecommunication technology.

The ITU-Telecommunication Standardization Sector (ITU-T) is the leader in defining in- tegrated services digital network (ISDN), B-ISDN, and ATM specifications. The American National Standards Institute (ANSI) is the formal standards body guiding the development of ATM in the UStates.

1.3 New Developments

Numerous telcos have implemented wide-area ATM networks, and many ADSL implementa- tions use ATM. However, ATM has failed to gain wide use as a LAN technology, and its great complexity has held back its full deployment as the single integrating network technology in the way that its inventors originally intended.

Many people, particularly in the Internet protocol-design community, considered this vision to be mistaken. Although there is a need for a unifying protocol at network layer, to be

able to run over all existing and future link-layer technologies, ATM could not do this role. Conveniently, IP already plays the role of such an integrator in a more scalable, more flexible, less complex, and most importantly, less expensive way than ATM could do. Therefore, there was no point in implementing ATM as an integrator at the network layer.

In addition, the need for cells to reduce jitter has disappeared as transport speeds increased (see below), and improvements in voice over IP have made the integration of speech and data possible at the IP layer, again removing the incentive for ubiquitous deployment of ATM. Most telcos are now planning to integrate their voice network activities into their IP networks, rather than their IP networks into the voice infrastructure.

Many technically sound ideas from ATM were adopted by MPLS, a generic layer 2 packet switching protocol. ATM remains widely deployed, and is used as a multiplexing service in DSL networks, where its compromises fit DSL's low-data-rate needs well. In turn, DSL networks support IP (and IP services such as VoIP) via PPP over ATM.

ATM will remain deployed for some time in higher-speed interconnects where carriers have already committed themselves to existing ATM deployments; ATM is used here as a way of unifying PDH/SDH traffic and packet-switched traffic under a single infrastructure.

However, ATM is increasingly challenged by speed and traffic shaping requirements of con- verged networks. In particular, the complexity of SAR imposes a performance bottleneck, as the fastest SARs known run at 2.5 Gbps and have limited traffic shaping capabilities.

Currently it seems like Ethernet implementations (10-Gbit-Ethernet [18], Metro Ethernet [20]) will replace ATM in many locations.

2 ATM Protocol Reference Model

The ATM protocol reference model is based on standards developed by the ITU. Communi- cation from higher layers is adapted to the lower ATM defined layers, which in turn pass the information onto the physical layer for transmission over a selected physical medium. The protocol reference model is divided into three layers: the ATM adaptation layer (AAL), the ATM layer, and the physical layer, as shown in Figure 1 [4]. The three management planes user/control plane, layer management and plane management, are shown in Figure 2 [4].

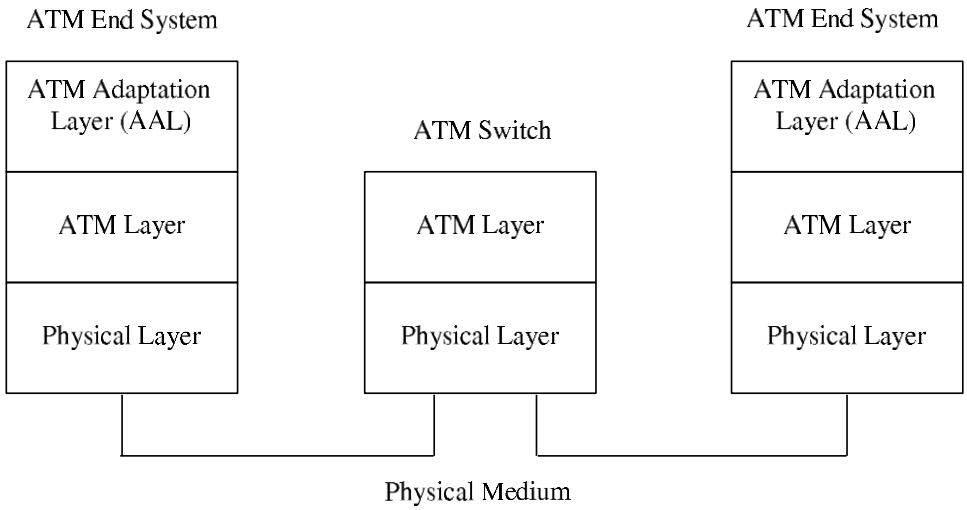


Figure 1: ATM protocol structure

2.1 The ATM Adaptation Layer

The ATM adaptation layer (AAL) interfaces the higher layer protocols to the ATM Layer. It relays ATM cells both from the upper layers to the ATM layer and vice versa. When relaying information received from the higher layers to the ATM layer, the AAL segments

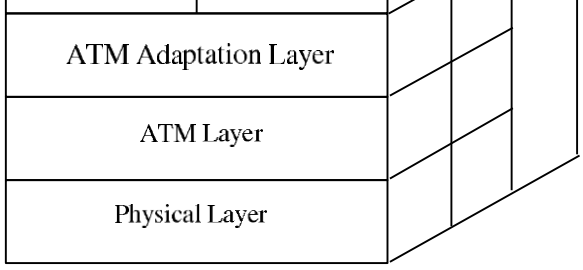


Figure 2: ATM model

the data into ATM cells. When relaying information received from the ATM Layer to the higher layers, the AAL must take the cells and reassemble the payloads into a format that the higher layers can understand. This is called segmentation and reassembly (SAR).

Four types of AALs were proposed, each supporting a different type of traffic or service expected to be used on ATM networks. The service classes and the corresponding types of AALs are as follows:

- *AAL0* AAL0 payload consists of 48 bytes without special field, is also referred to as raw cells.
- *AAL1* AAL1 was designed to support constant bit rate applications. Examples of these types of applications include 64 Kbps voice, fixed-rate uncompressed video, and leased lines for private data networks.
- *AAL2*: AAL2 was initially conceived to support variable bit rate applications that require a bounded delay for delivery. One example of such applications is compressed

packetized voice or video. The requirement on bounded delay for delivery is necessary for the receiver to reconstruct the original uncompressed voice or video. Although, AAL2 was conceived in early years of ATM development, it was not designed. So later when ATM designers needed an AAL for voice traffic, they first labeled it AAL6 and then quickly relabeled it as AAL2. So today, AAL2 is used for carrying voice traffic and allows several small compressed voice packets to be packed in a single 48-byte cell payload.

- *AAL3/4*: AAL3 and AAL4 were conceived for connection-oriented and connectionless data traffic that do not have delay constraints. Both these were to support variable bit rate data applications such as file transfer. However, designers quickly realized that there was little difference between the two types of traffic and so a single AAL called AAL 3/4 was designed. Because of the high complexity of AAL3/4 protocols, a simpler AAL called AAL5 was later proposed and is the common AAL used today. AAL 3/4 is no longer used.

- *AAL5*: AAL5 is designed for data traffic that do not have delay constraints. Examples of applications include IP traffic, LAN, FTP, and network management.

Although each AAL is optimized for a specific type of traffic, there is no stipulation in the standards that AALs designed for one class of traffic cannot be used for another. In fact, many vendors of ATM equipment currently manufacture products that use AAL5 to support all the above classes of traffic, and most activities at the ATM Forum were focused on AAL5. The AAL5 is also important in the internetworking of different networks and services. For

more discussion on the issues in AAL5 design, see [63]. AAL1 is also important, because it is used for streams and for circuit emulation [1].

AAL5 places control information in an 8-octet trailer at the end of the packet. The AAL5 trailer contains a 16-bit length field, a 32-bit cyclic redundancy check (CRC) and two 8-bit fields labeled UU and CPI that are currently unused.

In AAL5, each higher layer packet is divided into an integral number of ATM cells. At the receiving end, these cells are reassembled into a packet before delivery to the receiving host. The last cell contains padding to ensure that the entire AAL5 protocol data unit (PDU) is a multiple of 48 octets long. The final cell contains up to 40 octets of data, followed by zero padding and the 8-octet trailer.

2.2 The ATM Layer

The ATM layer provides an interface between the AAL and the physical layer. This layer is responsible for relaying cells from the AAL to the physical layer for transmission and from the physical layer to the AAL for use at the end systems. When it is inside an end system, the ATM layer receives a stream of cells from the physical layer and transmits cells with new data. When it is inside a switch, the ATM layer determines where the incoming cells should be forwarded to, modifies the corresponding connection identifiers, and forwards the cells to the next link. Moreover, it buffers incoming and outgoing cells, and handles various traffic management functions such as cell loss priority marking, congestion indication, and generic flow control. It also monitors the transmission rate and conformance to the service contract

(traffic policing). Traffic management was a hotly debated topic in the ATM Forum, and we shall address the important issues in more details later.

The fields in the ATM cell header define the functionality of the ATM layer. The format of the header for ATM cells has two different forms, one for use at the user-to-network interface (UNI) [10, 9] and the other for use internal to the network, the network-to-node interface (NNI), as shown in Figure 3. ATM user network interface (UNI) signalling specification version 4.1 [10, 9] was standardized in 2002. At the UNI, the header dedicates four bits to a function called generic flow control (GFC), which was originally designed to control the amount of traffic entering the network. This allows the UNI to limit the amount of data entering the network during periods of congestion. At the NNI, these four bits are allocated to the virtual path identifier (VPI).

The ATM inter network interface (AINI) protocol [5] was designed for use between ATM networks. AINI protocol is based on ATM Forum PNNI signalling [25]. The networks on either side of the AINI may be running any protocol internally. However, the goal in defining this protocol was to facilitate interworking of two networks running PNNI internally in disjoint PNNI routing domains.

Figure 4 gives an illustration of ATM Network Interfaces.

The VPI and the virtual channel identifier (VCI) together, as shown in Figure 5, form the routing field, which associates each cell with a particular channel or circuit, see Figure 6. Each VCI identifies a single flow (channel); the VPI allows grouping of VCs with different VCIs that can be switched together as an entity. However, the VPIs and VCIs have signif-

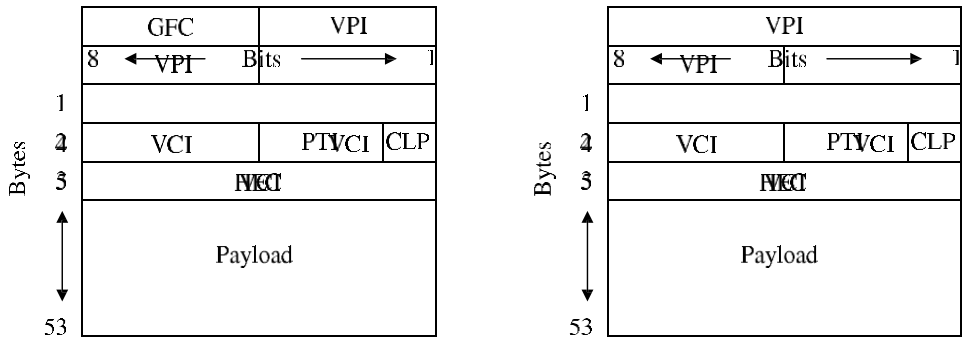


Figure 3: UNI (left) and NNI (right) ATM cell format

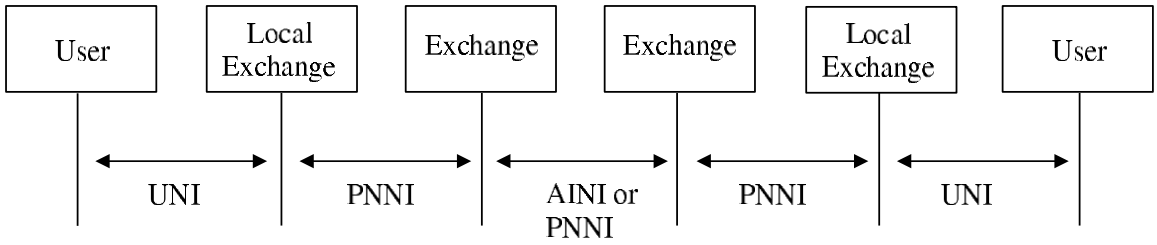


Figure 4: ATM network interfaces

icance only on the local link; the contents of the routing field will generally change as the cell traverses from link to link. For the UNI, the routing field contains 24 bits and thus the interface can support over 16 million concurrent sessions. At the NNI, the field contains 28 bits, allowing for over 268 million sessions to share a link within a subnet. We refer the readers to the discussion of important issues in private network-to-network interface (PNNI) routing to [25, 49].

The payload type indicator (PTI) field is used to distinguish between cells carrying user data and cells containing control information. This allows control and signaling data to be transmitted on a different subchannel from user data and hence separation of user and control data. A particular bit is used by the AAL if the cell is a part of an AAL5 connection.

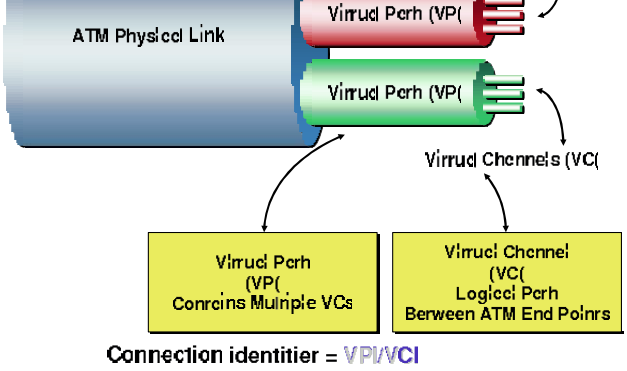


Figure 5: Virtual path and virtual channels

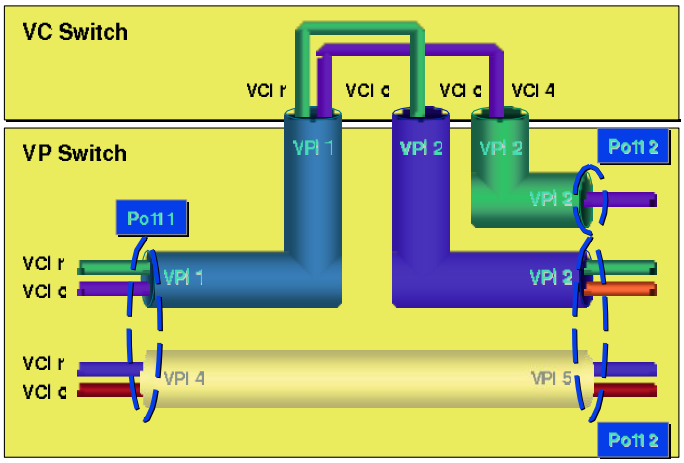


Figure 6: VP and VC switching

Another bit is used to indicate that the cell has experienced congestion.

The cell loss priority (CLP) bit provides the network with a selective discard capability within each VPI/VCI. Cells with a CLP bit setting of 1 are discarded before cells with a CLP bit setting of 0. This bit could be set by a user to indicate lower-priority cells that

could be discarded by the network during periods of congestion. Whereas data applications generally cannot suffer any cell loss without the need for retransmission, voice and video traffic, especially if not compressed, can tolerate minor cell loss. One could, therefore, code voice and video traffic such that some less important cells could be marked with CLP = 1 while other more important cells would be marked with CLP = 0. The CLP bit could also be used by the network to indicate cells that exceed the negotiated rate limit of a user.

The header error check (HEC) field is used to reduce errors in the header that cause a misrouting of the cell for one user into another user's data stream. This field contains the result of an 8-bit CRC checking on the ATM header (this does not include the payload). When a switch or an end system terminates the header, multiple-bit errors will be detected with a high probability. Moreover, a single-bit error can be corrected. This is desirable since ATM is intended for use on fiber optics link, where the error rate is less than 10^{-9} with current modulation techniques. Therefore, single-bit error correction is quite effective in removing most header errors.

2.3 The Physical Layer

The physical layer defines the bit timing and other characteristics for encoding and decoding the data into suitable electrical/optical waveforms for transmission and reception on the specific physical media used. In addition, it also provides cell delineation function, header error check (HEC) generation and processing, performance monitoring, and payload rate matching of the different transport formats used at this layer.

The Synchronous Optical Network (SONET), a synchronous transmission structure, is often used for framing and synchronization at the physical layer. In addition to the optical media and line rates defined for SONET, the ATM Forum has proposed a variety of physical layer standards, such as ATM over twisted-pair wire. This will accelerate the acceptance of ATM as a desktop connection technology since existing cabling plants can be retained and the cost per connection will be reduced. We refer the readers to [54] for a discussion on the ATM physical layer issues.

3 Traffic Management

In order for ATM networks to deliver guaranteed quality of service (QoS) on demand while maximizing the utilization of available network resources, effective traffic management mech- anisms are needed. Almost every aspect of ATM network operation, from signaling requests and routing to network resource allocation and policing, contains some traffic management mechanisms [26].

A set of six service categories are specified. For each one, a set of parameters is given to describe both the traffic presented to the network, and the QoS which is required of the network.

3.1 Generic Functions

To meet the QoS objectives, the following functions [26] form a framework for managing and controlling traffic and congestion in ATM networks and may be used in appropriate combinations depending on the service category.

- Network Resource Management: is used in broadband networks to keep track of the way link resources are allocated to connections. The two primary resources that are tracked by network resource management are capacity (bandwidth) and connection identifiers. Network resource management keeps track of the capacity and controls the allocation of capacity to connections when requested as part of the connection setup process [60]. In ATM, the service architecture allows logical separation of connections according to service characteristics. Although cell scheduling and resource provisioning are implementation and network specific, they can be utilized to provide appropriate isolation and access to resources. Virtual paths are a useful tool for resource manage-ment.

- Traffic policing: is monitoring network traffic for conformity with a traffic contract. An application that wishes to use the broadband network to transport traffic must first request a connection, which involves informing the network about the characteristics of the traffic and the quality of service (QOS) required by the application [39]. This information is stored in a traffic contract. If the connection request is accepted, the application is permitted to use the network to transport traffic.

  The main purpose of this function is to protect the network resources from malicious

connections and to enforce the compliance of every connection to its negotiated traffic contract. The network also has the capability to discard non-conformant traffic in the network (using priority control). Traffic policing in ATM networks is known as usage parameter control (UPC) and network parameter control (NPC) [59].

- Traffic shaping provides a mechanism to control the volume of traffic being sent into a network (bandwidth throttling), and the rate at which the traffic is being sent (rate limiting). For this reason, traffic shaping schemes are commonly implemented at the network edges to control traffic entering the network. The objectives of this function are to achieve a better network efficiency while meeting the QoS objectives and/or to ensure connection traffic conformance at a subsequent interface. Simple traffic shaping schemes like leaky bucket and token bucket rely on shaping all traffic uniformly by rate.

- Connection admission control (CAC): Admission control is the simple practice of discriminating which traffic is admitted into a network in the first place [39]. Admission control in ATM networks is known as connection admission control (CAC) [59].

  Connection admission control is defined as the set of actions taken by the network during the call set-up phase in order to determine whether a connection request can be accepted or should be rejected (or whether a request for re-allocation can be ac- commodated).

- Feedback controls: are defined as the set of actions taken by the network and by

end-systems to regulate the traffic submitted on ATM connections according to the state of network elements. This specification defines one network feedback control mechanism: the ABR flow control. The ABR flow control may be used to adaptively share the available bandwidth among participating users.

- Usage parameter control (UPC): is defined as the set of actions taken by the network to monitor traffic and enforce the traffic contract at the user network. Network parameter control (NPC) is a similarly defined set of actions at the Network Node Interface. The main purpose of UPC and NPC is to protect network resources from malicious as well as unintentional misbehavior, which can affect the QoS of other already established connections, by detecting violations of negotiated parameters and taking appropriate actions. Such actions may include cell discard and cell tagging.

- Cell loss priority control: For some service categories the end system may generate traffic flows of cells with cell loss priority (CLP) marking. The network may follow models which treat this marking as transparent or as significant. If treated as signifi- cant, the network may selectively discard cells marked with a low priority to protect, as far as possible, the QoS objectives of cells with high priority.

- Frame discard: A congested network that needs to discard cells may discard at the frame level rather than at the cell level.

3.2 Quality of Service Attributes

While setting up a connection on ATM networks, users can negotiate with the network the following parameters related to the desired quality of service:

- *Peak-to-peak cell delay variation (peak-to-peak CDV).*

  Cell transfer delay (CTD) is the delay experienced by a cell between network entry and exit points is called the cell transfer delay. It includes propagation delays, queueing delays at various intermediate switches, and service times at queueing points.

  The peak-to-peak CDV is the difference between the $(1-\alpha)$ quantile of the CTD and the fixed CTD that could be experienced by any delivered cell on a connection during the entire connection holding time. The term peak-to-peak refers to the difference between the best and worst case of CTD, where the best case is equal to the fixed delay, and the worst case is equal to a value likely to be exceeded with probability no greater than α.

- *Maximum cell transfer delay (maxCTD).*

  Cell delay variation (CDV) is a measure of variance of CTD. High variation implies larger buffering for delay sensitive traffic such as voice and video.

  The maximum cell transfer delay (maxCTD) specified for a connection is the $(1-\alpha)$ quantile of CTD. The CLR at connection request time is used to place an upper bound on α.

- *Cell loss ratio (CLR)*: The percentage of cells that are lost in the network because of

error or congestion and are not delivered to the destination, i.e.,

$$\text{CLR} = \frac{\#\ \text{Lost Cells}}{\#\ \text{Transmitted Cells}}.$$

Recall that each ATM cell has a cell loss priority (CLP) bit in the header. During periods of congestion, the network will first discard cells with CLP = 1. Because the loss of cells with CLP = 0 is more harmful to the operation of the application, CLR can be specified separately for cells with CLP = 1 and for those with CLP = 0.

All these parameters are described in details in the ”Traffic Management Specification” document [26].

3.3 Traffic Contract

To provide a guaranteed QoS, a traffic contract is established during connection setup, which contains a connection traffic descriptor and a conformance definition. However, it is not necessary for every ATM virtual connection to have a specified QoS. The reason for this is that if only specified QoS connections are supported by ATM, then a large percentage of the network resources will be wasted. This can happen when one or more connections are not utilizing the full capacity of their QoS contracts. Unspecified QoS contracts can be supported by an ATM network on a “best-effort” basis. Such best-effort services are sufficient for supporting most of the existing data applications.

In general, a traffic contract specifies one of the following six service categories:

- *Constant bit rate (CBR)*: This service category is used for emulating circuit switching,

where the bit rate is constant. Cell loss ratio is specified for cells with CLP=0 and may or may not be specified for cells with CLP =1.

- *Real-time variable bit rate (rt-VBR)*: The real-time VBR service category is intended for real-time applications, i.e., those requiring tightly constrained delay and delay variation, as would be appropriate for voice and video applications. rt-VBR connections are characterized in terms of a peak cell rate (PCR), sustainable cell rate (SCR), and maximum burst size (MBS). Sources are expected to transmit at a rate that varies with time. Equivalently the source can be described as "bursty". Cells that are de- layed beyond the value specified by maxCTD are assumed to be of significantly reduced value to the application. Real-time VBR service may support statistical multiplexing of real-time sources.

- *Non-real-time variable bit rate (nrt-VBR)*: The non-real-time VBR service category is intended for non-real-time applications that have bursty traffic characteristics and which are characterized in terms of a PCR, SCR, and MBS. For those cells that are transferred within the traffic contract, the application expects a low cell loss ratio. Non-real-time VBR service may support statistical multiplexing of connections. No delay bounds are associated with this service category.

- *Available bit rate (ABR)*: This service category is designed for normal data traffic such as file transfer and email. Although the standard does not require the cell transfer delay and cell loss ratio to be guaranteed, it is desirable for switches to minimize the delay and loss as much as possible. Depending upon the congestion state of the network,

the source is required to control its rate. The users are allowed to declare a minimum cell rate (MCR), which is guaranteed to the VC by the network. Most VCs will ask for an MCR of zero. Those with higher MCR may be denied connection if sufficient bandwidth is not available.

- *Unspecified bit rate (UBR)*: This service category is designed for those data applications that want to use any left-over capacity and are not sensitive to cell loss or delay. Such connections are not rejected on the basis of bandwidth shortage (i.e., no connection admission control) and not policed for their usage behavior. During congestion, the cells are lost but the sources are not expected to reduce their cell rate. Instead, these applications may have their own higher-level cell loss recovery and retransmission mechanisms. Examples of applications that use this service are email and file transfer. Of course, these same applications can use the ABR service, if desired.

- *Guaranteed frame rate (GFR)*: The GFR service category is intended to support non-real-time applications. It is designed for applications that may require a minimum rate guarantee and can benefit from accessing additional bandwidth dynamically available in the network. It does not require adherence to a flow control protocol. The service guarantee is based on AAL5 PDUs (frames) and, under congestion conditions, the net- work attempts to discard complete PDUs instead of discarding cells without reference to frame boundaries. On the establishment of a GFR connection, the end-system spec- ifies a PCR, and a minimum cell rate (MCR) that is defined along with a maximum burst size (MBS) and a maximum frame size (MFS). The user may always send cells

at a rate up to PCR, but the network only commits to carry cells in complete frames at MCR. Traffic beyond MCR will be delivered within the limits of available resources. There are no delay bounds associated with this service category.

These service categories relate traffic characteristics and QoS requirements to network be- havior. Functions such as routing, CAC, and resource allocation are, in general, structured differently for each service category. Service categories are distinguished as being either real-time or non-real-time. For real-time traffic, there are two categories, CBR and rt-VBR, distinguished by whether the traffic descriptor contains only the peak cell rate (PCR) or both PCR and the sustainable cell rate (SCR) parameters. All service categories, except GFR, apply to both VCCs and VPCs. GFR is a frame-aware service that only applies to VCCs since frame delineation is not usually visible at the virtual path level.

ABR or UBR are usually specified in the traffic contract when the ATM network is providing a best-effort service. Thus, these two classes of traffic are sometimes referred to as best-effort traffic. The attributes for the above service categories are summarized in Table 1.

3.4 Congestion Control Techniques

Congestion control lies at the heart of the general problem of traffic management for ATM networks. In general, congestion arises when the incoming traffic to a specific link is more than the outgoing link capacity. The primary function of congestion control is to ensure good throughput and delay performance while maintaining a fair allocation of network resources to the users [44]. For unspecified QoS traffic such as ABR service, whose traffic patterns

Table 1: ATM Service Category Attributes

	ATM Layer Service Category					
Attribute	CBR	rt-VBR	nrt-VBR	UBR	ABR	GFR
Traffic Parameters						
PCR and CDVT	Specified					
SCR, MBS, CDVT	n/a	Specified		n/a		
MCR	n/a				Specified	n/a
MCR, MBS, MFS CDVT	n/a					Specified
QoS Parameters						
Peak-to-peak CDV	Specified		Unspecified			
MaxCTD	Specified		Unspecified			
CLR	Specified			Unspecified	Network Specific	
Feedback	Unspecified				Specified	Unspecified

are often highly bursty and unpredictable, congestion control poses more challenges than for other services.

As described in [45], one way to classify congestion control schemes is based on the layer of ISO/OSI reference model at which the scheme operates. For example, there are data link, routing, and transport layer congestion control schemes. Typically, a combination of such schemes is used. The selection depends upon the severity and duration of congestion. Figure

7 shows how the duration of congestion affects the choice of the method.

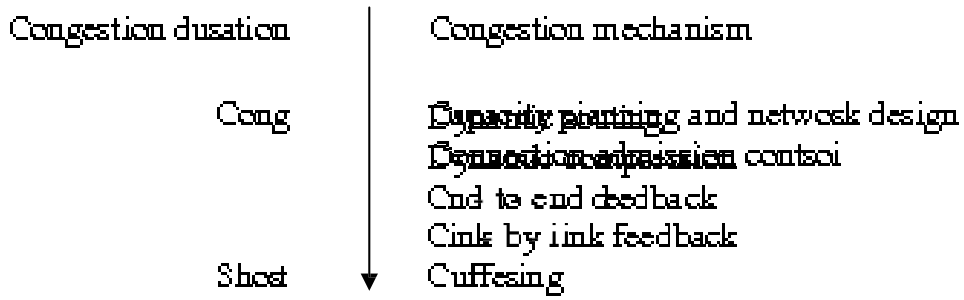


Figure 7: Congestion techniques for various congestion durations

One method to avoid network congestion is to accept a new ATM connection during connec- tion setup phase only when sufficient network resources are available to provide the accept- able QoS. This is called connection admission control (CAC), which is needed for connections where the QoS must be guaranteed. The "busy" tone on telephone networks is an example of CAC. Mechanisms for CAC are currently not standardized and are at the discretion of the network operators.

In addition to CAC, [26] also allows traffic shaping using a generic cell rate algorithm (GCRA) and binary explicit forward congestion indication (EFCI) feedback congestion control. These mechanisms are described next.

Generic Cell Rate Algorithm (GCRA) The GCRA is also called the "leaky bucket" algorithm, which converts a bursty stream into a more regular pattern. This algorithm essentially works by putting all arriving cells into a bucket, which is drained at the sustained cell rate. If too many cells arrive at once, the bucket may overflow. The overflowing cells

are called non-conforming and may or may not be admitted into the network. If admitted, the cell loss priority (CLP) bit of the non-conforming cells may be set so that they will be the first to be discarded in case of overload.

The leaky bucket algorithm is often used by the network to ensure that the input meets the pre-negotiated parameters such as the sustained and peak cell rates. Such “traffic shaping” algorithms are open loop in the sense that the parameters cannot be changed dynamically if congestion is detected after negotiation. In a closed-loop (feedback) scheme, however, sources are informed dynamically about the congestion state of the network and are asked to increase or decrease their input rate.

Feedback Congestion Control As described earlier in Figure 3, four bits of the cell header at the user-network interface (UNI) are reserved for generic flow control (GFC). Originally, the plan was to use these bits to flow control the source. The discussions in ATM Forum eventually led to the development of end-to-end congestion control scheme instead of GFC.

An effective congestion control scheme must satisfy several key criteria. In addition to being able to maximally utilize available bandwidth, a good scheme must also provide fairness of network resources to users. Moreover, it must be scalable to a large number of nodes and links with various capacities, robust against slight mistuning of parameters and loss of control cells, as well as low in switch complexity and buffer requirement.

The ATM Forum initially considered the use of the explicit forward congestion indication (EFCI) bit in the ATM cell headers to mark congestion in the switches [41]. This scheme

was to be based on DECbit scheme [53]. The forum finally adopted an explicit rate-based indication scheme based on [33].

The available bit rate (ABR) method of traffic management works as follows. The sources periodically send resource management (RM) cells, which indicate their current rate and the desired rate. The switches along the path adjust the desired rate down. The destination returns the RM cells to the sources. The sources then adjust their rate to that indicated in the RM cells. The algorithm for deciding the rate allocated by a switch is not specified and is left for the vendors to design. For examples of such algorithms, see [46, 47, 56].

The rate-based congestion control approach and its development at the ATM Forum is described in more detail in [26]. Other reference sources include the review papers of [43, 32].

4 Switch Architecture

Perhaps the most developed aspect of ATM is the switch architecture. Over the past decade, a vast amount of research efforts have been made on studying and designing ATM switches. The field has now become a mature research area and a number of tutorial articles have appeared in the literature. The design of ATM switch architectures is at the discretion of switch vendors. Basic principles of switch design and examines the influence of traffic patterns on the design methodologies are discussed in [34, 38, 29, 61].

ATM switches are high-speed packet switches specialized to process and forward ATM cells (packets). Because ATM is a connection-oriented protocol, ATM switches must establish a

virtual connection from one of its input ports to an output port before forwarding incoming ATM cells along that virtual connection.

A generic ATM switch architecture with N input ports and N output ports is shown in Figure 8. The functions of an ATM switching system may be divided broadly into the three planes as in [34].

- User Plane: The main function of an ATM switch is to relay user data cells from input ports to the appropriate output ports. The switch processes only the cell headers and the payload is carried transparently. As soon as the cell comes in through the input port, the virtual path identifier/virtual channel identifier (VPI/VCI) information is derived and used to route the cells to the appropriate output ports. This function can be divided into three functional blocks: the input module at the input port, the cell switch fabric (sometimes referred to as switch matrix) that performs the actual routing, and the output modules at the output ports.

- Control Plane: This plane represents functions related to the establishment and control of the VP/VC connections. Unlike the user data cells, information in the control cells payload is not transparent to the network. The switch identifies signaling cells, and even generates some itself. The connection admission control (CAC) carries out the major signaling functions required. Signaling information may/may not pass through the cell switch fabric, or maybe exchanged through a signaling network such as SS7.

- Management Plane: The management plane is concerned with monitoring the con-

trolling the network to ensure its correct and efficient operation. These operations can be subdivided as fault management functions, performance management func- tions, configuration management functions, security management functions, account- ing management and traffic management. These functions can be represented as being performed by the functional block switch management. The switch management is responsible for supporting the ATM layer operations and maintenance (OAM) proce- dures. OAM cells may be recognized and processed by the ATM switch. The switch must identify and process OAM cells, maybe resulting in generating OAM cells. As with signaling cells, OAM cells may/may not pass through cell switch fabric. Switch management also supports the interim local management interface (ILMI) of the UNI. The Switch Management contains, for each UNI, a UNI management entity (UME), which may use SNMP.

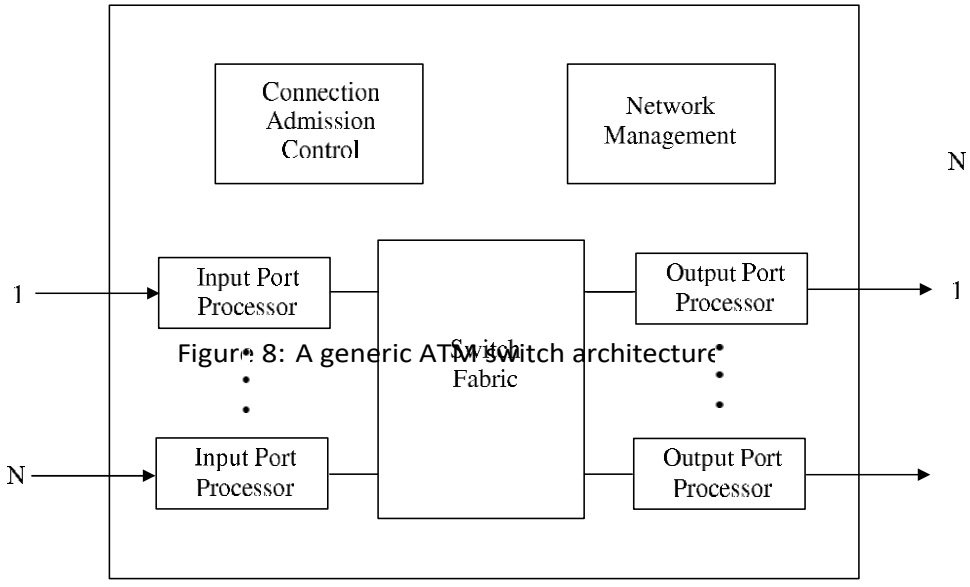


Figure 8: A generic ATM switch architecture

ATM cells containing user data are received at the input ports, and the input port processors prepare the cells for routing through the switch fabric. The fabric in the center of the switching system provides the interconnections between input port processors and output port processors. The output port processors prepare the outgoing user cells for transmission from the switch. User cell forwarding is characterized by parallelism and high-speed hardware processing. The ATM protocol was intentionally streamlined to allow incoming cells to be processed simultaneously in hardware and routed through the switch fabric in parallel. Thus, ATM switches have been able to realize high-end performance in terms of throughput and cell forwarding delay.

An ATM switch contains a set of input ports and output ports, through which it is inter- connected to users, other switches, and other network elements. It might also have other interfaces to exchange control and management information with special purpose networks.

Connection control, sometimes called the control plane, refers to the functions related to the establishment and termination of ATM virtual connections. Connection control functions generally encompass: exchange and processing of signaling information; participation in routing protocols; and decisions on admission or rejection of new connection requests.

The cell switch fabric is primarily responsible for routing of data cells and possibly signaling and management cells as well. Other possible functions include: cell buffering, traffic con- centration and multiplexing redundancy for fault tolerance, multicasting or broadcasting, cell scheduling based on delay priorities, congestion monitoring and activation of explicit forward congestion indication (EFCI). More details about switch fabrics can be found in

[34, 38, 29].

Network management is currently carried out by SNMP (simple network management pro- tocol), the standard protocol for managing data networks. ATM switches typically support an SNMP agent and an ATM MIB (management information base).

4.1 Switch Interface

4.1.1 Input Modules

The input module first terminates the incoming signal (for example a SONET signal) and extracts the ATM cell stream. This involves signal conversion and recovery, processing SONET overhead, and cell delineation and rate decoupling. After that, for each ATM cell the following functions should be performed:

- Error checking the header using the header error control (HEC) field
- Validation and translation of VPI/VCI values
- Determination of the destination output port
- Passing signaling cells to CAC and OAM cells to switch management
- UPC/UNC for each VPC/VCC
- Addition of an internal tag containing internal routing and performance monitoring information for use only within the switch

4.1.2 Output Modules

Output Modules prepare the ATM cell streams for physical transmission by:

- Removing and processing the internal tag
- Possible translation of VPI/VCI values
- HEC field generation
- Possible mixing of cells from CAC and switch management with outgoing cell streams
- Cell rate decoupling
- Mapping cells to SONET payloads and generation of SONET overhead
- Cconversion of the digital bitstream to an optical signal

4.2 Connection Admission Control (CAC)

CAC establishes, modifies and terminates virtual path/channel connections. More specifi- cally, it is responsible for:

- High-layer signaling protocols
- Signaling ATM adaptation layer (AAL) functions to interpret or generate signaling cells
- Interface with a signaling network

- Negotiation of traffic contracts with users requesting new VPCs/VCCs
- Renegotiation with users to change established VPCs/VCCs
- Allocation of switch resources for VPCs/VCCs, including route selection
- Admission/rejection decisions for requested VPCs/VCCs
- Generation of UPC/NPC parameters

If the CAC is centralized, a single processing unit would receives signaling cells from the input modules, interpret them, and perform admission decisions and resource allocation decisions for all the connections in the switch. CAC functions may be distributed to blocks of input modules where each CAC has a smaller number of input ports. This is much harder to implement, but solves the connection control processing bottleneck problem for large switch sizes, by dividing this job to be performed by parallel CACs. A lot of information must be communicated and coordinated among the various CACs [34, 38]. Some of the distributed CAC functions can also be distributed among output modules which can handle encapsulation of high-layer control information into outgoing signaling cells.

4.3 Switch Management

Switch management physical layer OAM, ATM layer OAM, configuration management of switch components, security control for the switch database, usage measurements of the switch resources, traffic management, administration of a management information base,

customer-network management, interface with operations systems and finally support of network management.

Switch management is difficult because management covers an extremely wide spectrum of activities. In addition, the level of management functions implemented in the switch can vary between minimal and complex.

Switch management must perform a few basic tasks. It must carry out specific management responsibilities, collect and administer management information, communicate with users and network managers, and supervise and coordinate all management activities. Manage- ment functions include fault management, performance management, configuration manage- ment, accounting management, security management, and traffic management. Carrying out these functions entails a lot of intraswitch communication between the switch management and other functional blocks.

A centralized switch management can be a performance bottleneck if it is overloaded by processing demands. Hence, switch management functions can be distributed among input modules, but a lot of coordination would be required. Each distributed input module switch management unit can monitor the incoming user data cell streams to perform accounting and performance measurement. Output module switch management units can also monitor outgoing cell streams [34, 38].

4.4 The Cell Switch Fabric

The cell switch fabric is primarily responsible for transferring cells between the other func- tional blocks (routing of data cells and possibly signaling and management cells as well). Other possible functions include:

- Cell buffering
- Traffic concentration and multiplexing
- Redundancy for fault tolerance multicasting or broadcasting
- Cell scheduling based on delay priorities
- Congestion monitoring and activation of explicit forward congestion indication (EFCI)

4.4.1 Concentration, Expansion, and Multiplexing

Traffic needs to be concentrated at the inputs of the switching fabric to better utilize the incoming link connected to the switch. The concentrator aggregates the lower variable bit rate traffic into higher bit rate for the switching matrix to perform the switch at standard interface speed. The concentration ratio is highly correlated with the traffic characteristics, so it needs to be dynamically configured. The concentrator can also aid in dynamic traffic distribution to multiple routing and buffering planes, and duplication of traffic for fault tolerance. At the outputs of the routing and buffering fabric, traffic can be expanded and redundant traffic can be combined.

4.4.2 Routing and Buffering

The routing and buffering functions are the two major functions performed by the cell switch fabric. The input module attaches a routing tag to each cell, and the switch fabric simply routes the arriving cells from its inputs to the appropriate outputs. Arriving cells may be aligned in time by means of single-cell buffers. Because cells may be addressed to the same output simultaneously, buffers are needed. Several routing and buffering switch designs have aided in setting the important switch design principles. All current approaches employ a high degree of parallelism, distributed control, and the routing function is performed at the hardware level.

Traditionally switching has been defined to encompass either space switching or time switch- ing or combinations of both techniques. The classification adopted here is slightly different in the sense that it divides the design approaches under the following four broad categories
[34]: (1) shared memory, (2) shared Medium, (3) fully interconnected, and (4) space division.

Shared Memory Approach: Figure 9 illustrates the basic structure of a shared memory switch. Here incoming cells are converted from serial to parallel form, and written sequen- tially to a dual-port random access memory. A memory controller decides the order in which cells are read out of the memory, based on the cell headers with internal routing tags. Outgoing cells are demultiplexed to the outputs and converted from parallel to serial form.

This approach is an output queueing approach, where the output buffers all physically belong to a common buffer pool. The approach is attractive because it achieves 100% throughput under heavy load. The buffer sharing minimizes the amount of buffers needed to achieve a

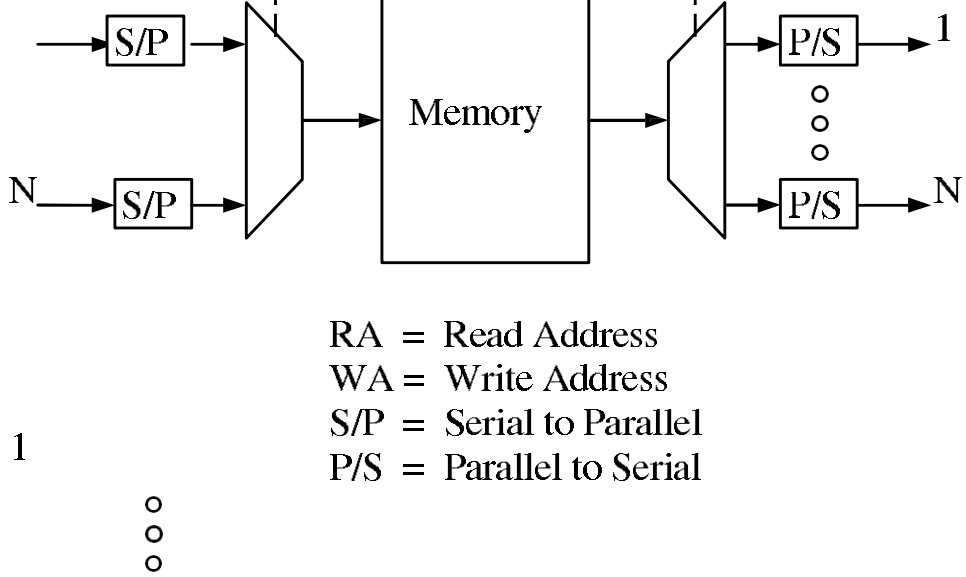


Figure 9: Basic structure of a shared-memory switch

specified cell loss rate. This is because if a large burst of traffic is directed to one output port, the shared memory can absorb as much as possible of it.

The approach, however, suffers from a few drawbacks. The shared memory must operate N times faster than the port speed because cells must be read and written one at a time. As the access time of memory is physically limited, the approach is not very scalable. The product of the number of ports times port speed (NV) is limited. In addition, the centralized memory controller must process cell headers and routing tags at the same rate as the memory. This is difficult for multiple priority classes, complicated cell scheduling, multicasting and broadcasting.

Shared Medium Approach: Cells may be routed through a shared medium, like a ring, bus or dual bus. Time-division multiplexed buses are a popular example of this approach,

and Figure 10 illustrates their structure. Arriving cells are sequentially broadcast on the TDM bus in a round-robin manner. At each output, address filters pass the appropriate cells to the output buffers, based on their routing tag. The bus speed must be at least NV for cells/s to eliminate input queueing.

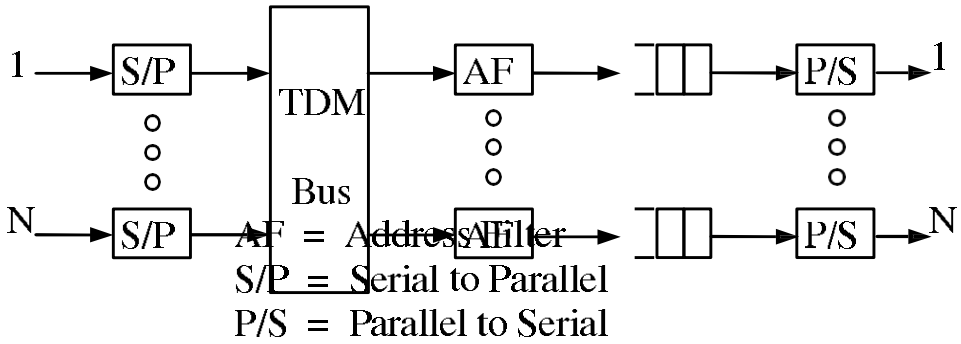


Figure 10: A shared bus switch (adapted from Chen and Liu [34])

The outputs are modular, which makes address filters and output buffers easy to implement. Also the broadcast-and-select nature of the approach makes multicasting and broadcasting straightforward. As a result, many such switches have been implemented, such as IBM's Packetized Automated Routing Integrated System (PARIS) and plaNET, NEC's ATM Out- put Buffer Modular Switch (ATOM), and Fore Systems' ForeRunner ASX-100 to mention a few [52]. The Synchronous Composite Packet Switching (SCPS), which uses multiple rings is also one of the most famous experiments of shared medium switches [55].

However, because the address filters and output buffers must operate at the shared medium speed, which is N times faster than the port speed, this places a physical limitation on the scalability of the approach. In addition, unlike the shared memory approach, output buffers

are not shared, which requires more total amount of buffers for the same cell loss rate.

Fully Interconnected Approach: In this approach, independent paths exist between all N squared possible pairs of inputs and outputs. Hence arriving cells are broadcast on separate buses to all outputs and address filters pass the appropriate cells to the output queues. This architecture is illustrated in Figure 11.

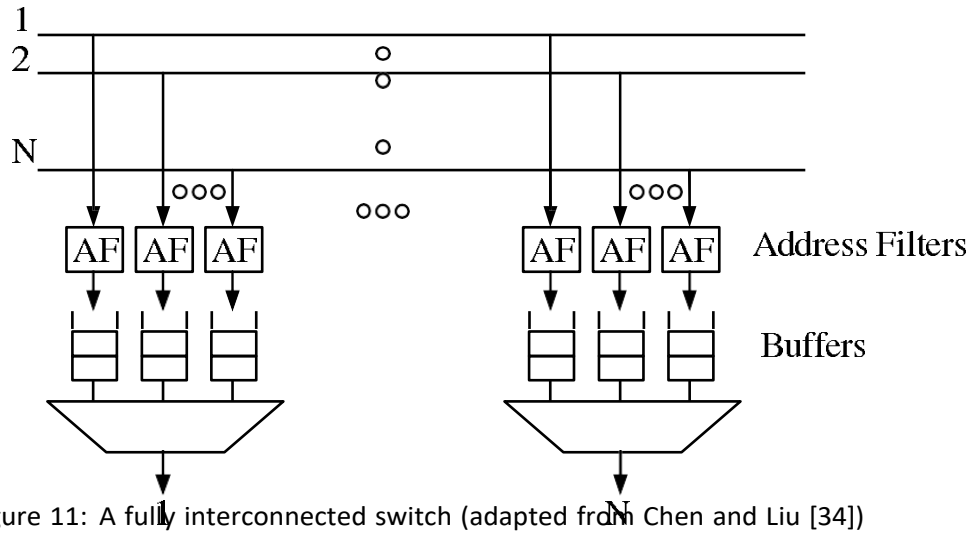


Figure 11: A fully interconnected switch (adapted from Chen and Liu [34])

This design has many advantages. As before, all queueing occurs at the outputs. In addition, multicasting and broadcasting are natural, like in the shared medium approach. Address filters and output buffers are simple to implement and only need to operate at the port speed. Because all of the hardware operates at the same speed, the approach is scalable to any size and speed. Fujitsu's bus matrix switch and GTE Government System's SPANet are examples of switches in which this design was adopted.

Unfortunately, the quadratic growth of buffers limits the number of output ports for practical reasons. However, the port speed is not limited except by the physical limitation on the speed

of the address filters and output buffers.

The *Knockout* switch developed by AT&T was an early prototype where the amount of buffers was reduced at the cost of higher cell loss [52, 55]. Instead of N buffers at each output, it was proposed to use only a fixed number of buffers L for a total of NxL buffers. This technique was based on the observation that it is unlikely that more than L cells will arrive for any output at the same time. It was argued that selecting the L value of 8 was sufficient for achieving a cell loss rate of 1/1 million under uniform random traffic conditions for large values of N.

Space Division Approach: The *crossbar* switch is the simplest example of a matrix-like space division fabric that physically interconnects any of the N inputs to any of the N outputs. Multistage interconnection networks (MINs), which are more tree-like structures, were then developed to reduce the N squared crosspoints needed for circuit switching, multiprocessor interconnection and, more recently, packet switching.

One of the most common types of MINs is the banyan network. It is named for its resem- blance to the roots of the Banyan tropical tree which crossover in complex patterns. The banyan network is constructed of an interconnection of stages of switching elements. A basic 2x2 switching element can route an incoming cell according to a control bit (output address). If the control bit is 0, the cell is routed to the upper port address, otherwise it is routed to the lower port address.

In general, to construct an NxN banyan network, the n^{th} stage uses the n^{th} bit of the output address to route the cell. For N = 2 to the power of n, the banyan will consist of n = log

to the base 2 of N stages, each consisting of N/2 switching elements. A MIN is called self- routing when the output address completely specifies the route through the network (also called digit-controlled routing).

The banyan network technique is popular because switching is performed by simple switching elements, cells are routed in parallel, all elements operate at the same speed (so there is no additional restriction on the size N or speed V), and large switches can be easily constructed modularly and recursively and implemented in hardware.

It is clear that in a banyan network, there is exactly one path from any input to any output. Regular banyans use only one type of switching element, and SW-banyans are a subset of regular banyans, constructed recursively from LxM switching elements.

Delta networks are a subclass of SW-banyan networks, possessing the self-routing property. There are numerous types of delta networks, such as rectangular delta networks (where the switching elements have the same number of outputs as inputs), omega, flip, cube, shuffle- exchange (based on a perfect shuffle permutation) and baseline networks. A delta-b network of size NxN is constructed of bxb switching elements arranged in log to the base b of N stages, each stage consisting of N/b switching elements [55].

Unfortunately, since banyan networks have less than N squared crosspoints, routes of two cells addressed to two different outputs might conflict before the last stage. When this situation, called internal blocking, occurs, only one of the two cells contending for a link can be passed to the next stage, so overall throughput is reduced. A solution to this problem is to add a sort network (such as a Batcher bitonic sort network) to arrange the cells before the

banyan network. This will be internally non-blocking for cells addressed to different outputs [55]. However, if cells are addressed to the same output at the same time, the only solution to the problem is buffering. Buffers can be placed at the input of the Batcher network, but this can cause ”head-of-line” blocking, where cells wait for a delayed cell at the head of the queue to go through, even if their own destination output ports are free. This situation can be remedied by First-In-Random-Out buffers, but these are quite complex to implement. Alternatively, buffers may be placed internally within the banyan switching elements. Thus if two cells simultaneously attempt to go to the same output link, one of them is buffered within the switching element. This internal buffering can also be used to implement a backpressure control mechanism, where queues in one stage of the banyan will hold up cells in the preceding stage by a feedback signal. The backpressure may eventually reach the first stage, and create queues at the banyan network inputs [34]. It is important to observe that internal buffering can cause head-of-line blocking at each switching element, and hence it does not achieve full throughput. Awdeh and Mouftah [30] have designed a delta-based ATM switch with backpressure mechanism capable of achieving a high throughput, while significantly reducing the overall required memory size.

A third alternative is to use a recirculating buffer external to the switch fabric. This technique has been adopted in Bellcore’s Sunshine and AT&T’s Starlite wideband digital switch [55]. Here output conflicts are detected after the Batcher sorter, and a trap network selects a cell to go through, and recirculates the others back to the inputs of the Batcher network. Unfortunately, this approach requires complicated priority control to maintain the sequential order of cells and increases the size of the Batcher network to accommodate the recirculating

cells [34].

As discussed before, output buffering is the most preferable approach. However, banyan networks cannot directly implement it since at most one cell per cell time is delivered to every output. Possible ways to work around this problem include:

- Increasing the speed of internal links
- Routing groups of links together
- Using multiple banyan planes in parallel
- Using multiple banyan planes in tandem or adding extra switching stages

Apart from banyan networks, many types of MINs with multiple paths between inputs and outputs exist. Classical examples include the non-blocking Benes and Clos networks, the cascaded banyan networks, and the randomized route banyan network with load distribution (which eliminates internal buffering). Combining a number of banyan planes in parallel can also be used to form multipath MINs. The multipath MINs achieve more uniform traffic distribution to minimize internal conflicts, and exhibit fault tolerance. However if cells can take independent paths with varying delays, a mechanism is needed to preserve the sequential ordering of cells of the same virtual connection at the output. Since this might involve considerable processing, it is better to select the path during connection setup and fix it during the connection. Special attention must be paid during path selection to prevent unnecessary blocking of subsequent calls.

4.5 Switch Design Principles

4.5.1 Internal Blocking

A fabric is said to be internally blocking if a set of N cells addressed to N different outputs can cause conflicts within the fabric. Internal blocking can reduce the maximum possible throughput. Banyan networks are blocking, whereas TDM buses where the bus operates at least N times faster than the port speed are internally nonblocking. By the same concept, shared memory switches which can read and write at the rate of NV cells per second are internally non-blocking, since if N cells arrive for N different outputs, no conflicts will occur. Hence, to prevent internal blocking, shared resources must operate at some factor greater than the port speed. Applying this to banyan networks, the internal links need to run square root of N times faster than the highest speed incoming link [52]. This factor limits the scalability and throughput of the switch. Coppo et al. [35] have developed a mathematical model for analyzing the optimal blocking probability versus complexity tradeoff.

4.5.2 Buffering Approaches

Buffering is necessary in all design approaches. For instance, in a banyan network, if two cells addressed to the same output successfully reach the last switching stage at the same time, output contention occurs and must be resolved by employing buffering. The location and size of buffers are important issues that must be decided [52].

There are four basic approaches to the placement of buffers. These basic approaches are illustrated in Figure 12. The literature abounds with comparative studies of these, aug-

mented with numerous queueing analysis and simulation results. Uniform random traffic, as well as bursty traffic have been examined. Although each approach has its own merits and drawbacks, output queueing is the preferred technique so far.

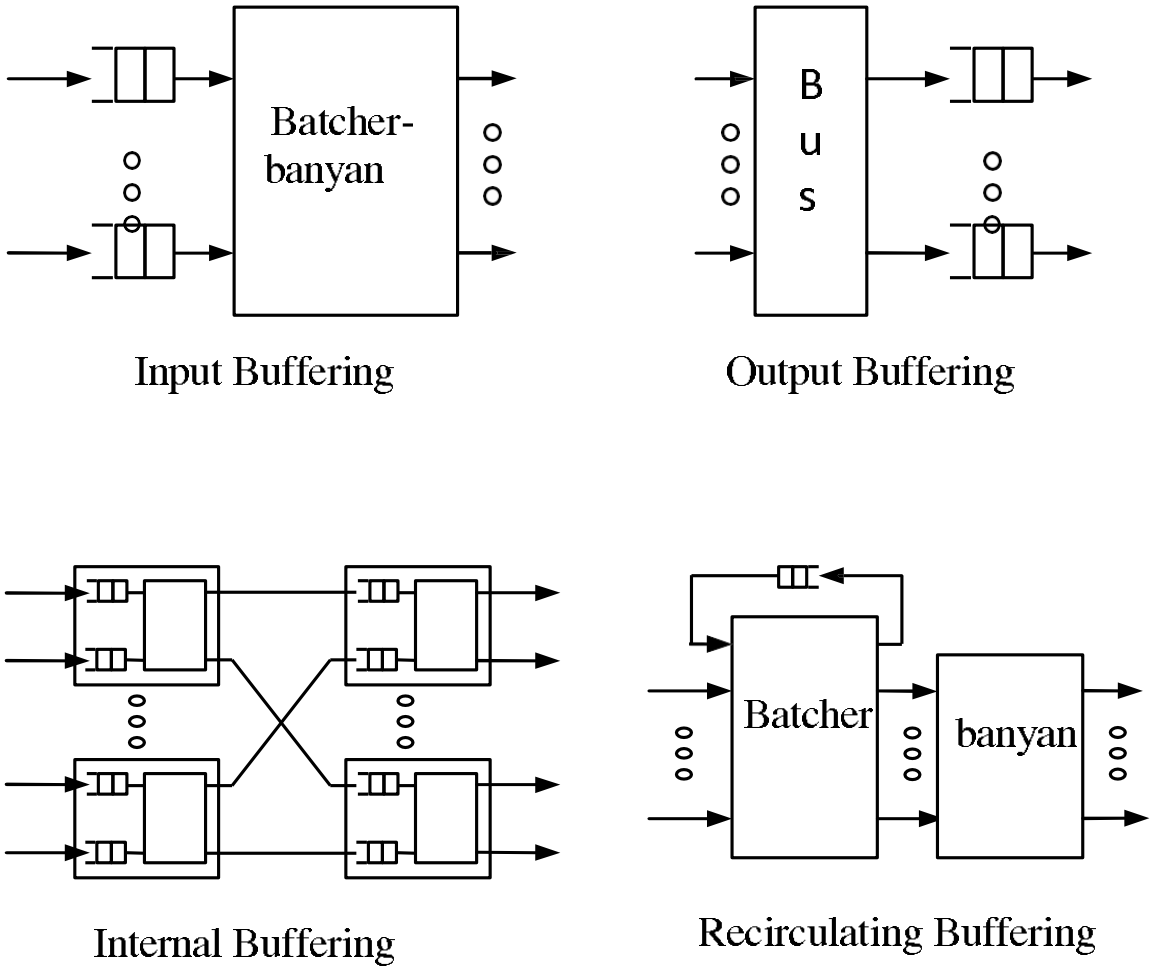


Figure 12: The various buffering approaches (Combined from Chen and Liu [34] and Onvural [52])

Input Queueing: Buffers at the input of an internally nonblocking space division fabric (such as Batcher banyan network) illustrate this type of buffering. This approach suffers from head-of-the-line blocking. When two cells arrive at the same time and are destined to

the same output, one of them must wait in the input buffers, preventing the cells behind it from being admitted. Thus capacity is wasted.

Several methods have been proposed to tackle the head-of-the-line blocking problem, but they all exhibit complex design. Increasing the internal speed of the space division fabric by a factor of four, or changing the first-in-first-out (FIFO) discipline are two examples of such methods.

Output Queueing: This type of buffering can be evident by examining the buffers at the output ports of a shared bus fabric. This approach is optimal in terms of throughput and delays, but it needs some means of delivering multiple cells per cell time to any output. Hence, either the output buffers must operate at some factor times the port speed, or there should be multiple buffers at each output. In both cases, the throughput and scalability are limited, either by the speedup factor or by the number of buffers.

Internal Queueing: Buffers can be placed within the switching elements in a space division fabric. For instance, in a banyan network, each switching element contains buffers at its inputs to store cells in the event of conflict. Again, head-of-the-line blocking might occur within the switching elements, and this significantly reduces throughput, especially in the case of small buffers or larger networks. Internal buffers also introduce random delays within the switch fabric, causing undesirable cell delay variation.

Recirculating Buffers: This technique allows cells to re-enter the internally nonblocking space division network. This is needed when more than one cell is addressed to the same out- put simultaneously, so the extra cells need to be routed to the inputs of the network through

the recirculating buffers. Although this approach has the potential for achieving the optimal throughput and delay performance of output queueing, its implementation suffers from two major complexities. First, the switching network must be large enough to accommodate the recirculating cells. Second, a control mechanism is essential to sequentially order the cells.

4.6 Buffer Sharing

The number and size of buffers has a significant impact on switch design. In shared memory switches, the central buffer can take full advantage of statistical sharing, thus absorbing large traffic bursts to any output by giving it as much as is available of the shared buffer space. Hence, it requires the least total amount of buffering. For a random and uniform traffic and large values of N, a buffer space of only 12 N cells is required to achieve a cell loss rate of 1/10 to the power of 9, under a load of 0.9.

For a TDM bus fabric with N output buffers, and under the same traffic assumptions as before, the required buffer space is about 90 N cells. Also a large traffic burst to one output cannot be absorbed by the other output buffers, although each output buffer can statistically multiplex the traffic from the N inputs. Thus, buffering assumes that it is improbable that many input cells will be directed simultaneously to the same output.

Neither statistical multiplexing between outputs or at any output can be employed with fully interconnected fabrics with N squared output buffers. Buffer space grows exponentially in this case.

5 New Developments

ATM could not fulfill the promise of providing a single integrated technology for LANs, public networks, and user services. IP was shown to provide such integration in a more flexible, more scalable, and less complex way than ATM. However, as it happens usually with technologies, the best ideas are borrowed by other solutions. In the case of ATM, various important concepts are inherited by other technologies, such as MPLS. Whereas other technologies, such as DSL, FTTP, and UMTS use ATM and AAL layers.

5.1 Multiprotocol label switching - (MPLS)

Multiprotocol label switching (MPLS) [23, 19, 21, 22, 57] is a data-carrying mechanism which emulates some properties of a circuit-switched network over a packet-switched network. MPLS has emerged as an elegant solution to meet the bandwidth-management and service requirements for next-generation Internet protocol (IP)based backbone networks. MPLS addresses issues related to scalability and routing (based on QoS and service quality metrics) and can exist over existing asynchronous transfer mode (ATM) and frame-relay networks. MPLS is standardized by IETF in RFC 3031 [57]. For ATM-MPLS network interworking see [6].

Although the underlying protocols and technologies are different, both MPLS and ATM provide a connection-oriented service for transporting data across computer networks. In both technologies connections are signaled between endpoints, connection state is maintained at each node in the path and encapsulation techniques are used to carry data across the

connection. Excluding differences in the signaling protocols (RSVP/LDP for MPLS and PNNI for ATM) there still remain significant differences in the behavior of the technologies.

The most significant difference is in the transport and encapsulation methods. MPLS is able to work with variable length packets whereas ATM transports fixed-length (53 byte) cells. Packets must be segmented, transported and re-assembled over an ATM network using an adaption layer, which adds significant complexity and overhead to the data stream. MPLS, on the other hand, simply adds a label to the head of each packet and transmits it on the network.

Differences exist, as well, in the nature of the connections. An MPLS connection (LSP) is uni-directional, allowing data to flow in only one direction between two endpoints. Estab- lishing two-way communications between endpoints requires a pair of LSPs to be established. Because two LSPs are required for connectivity, data flowing in the forward direction may use a path different from data flowing in the reverse direction. ATM point-to-point connec- tions (Virtual Circuits), on the other hand, are bi-directional, allowing data to flow in both directions over the same path (bi-directional are only SVC ATM connections; PVC ATM connections are uni-directional).

Both ATM and MPLS support tunneling of connections inside connections. MPLS uses label stacking to accomplish this while ATM uses virtual paths. MPLS can stack multiple labels to form tunnels within tunnels. The ATM virtual path indicator (VPI) and virtual circuit indicator (VCI) are both carried together in the cell header, limiting ATM to a single level of tunneling.

The biggest single advantage that MPLS has over ATM is that it was designed from the start to be complimentary to IP. Modern routers are able to support both MPLS and IP natively across a common interface allowing network operators great flexibility in network design and operation. ATM's incompatibilities with IP require complex adaptation making it largely unsuitable in today's predominantly IP networks.

5.2 Technologies exploiting ATM and AAL layers

DSL

DSL or xDSL [13], is a family of technologies that provide digital data transmission over the wires of a local telephone network. DSL originally stood for digital subscriber loop, although in recent years, many have adopted digital subscriber line as a more marketing-friendly term for the most popular version of DSL, ADSL over UNE.

Typically, the download speed of DSL ranges from 128 kilobits per second (kbps) to 24,000 kbps depending on DSL technology and service level implemented. Upload speed is lower than download speed for asymmetric digital subscriber line (ADSL) and equal to download speed for symmetric digital subscriber line (SDSL).

Many DSL technologies implement an ATM layer [15] over the low-level bitstream layer to enable the adaptation of a number of different technologies over the same link.

DSL implementations may create bridged or routed networks. In a bridged configuration, the group of subscriber computers effectively connect into a single subnet. The earliest implementations used DHCP to provide network details such as the IP address to the sub-

scriber equipment, with authentication via MAC address or an assigned host name. Later implementations often use PPP over Ethernet (PPPoE) [50] or ATM (PPPoA) [48], while authenticating with a userid and password and using PPP mechanisms to provide network details.

PPPoA, Point-to-Point Protocol (PPP) over ATM, is a network protocol for encapsulating PPP frames in ATM AAL5. It is used mainly with cable modem, DSL and ADSL services.

PPPoA offers standard PPP features such as authentication, encryption, and compression. If it is used as the connection encapsulation method on an ATM based network it can reduce overhead slightly (around 0.58%) in comparison to PPPoE [50]. It also avoids the issues that PPPoE suffers from, related to having a MTU lower than that of standard ethernet transmission protocols. It also supports (as does PPPoE) the encapsulation types: VC-MUX and LLC based. PPPoA is specified in RFC 2364 [48].

Fiber to the Premises

Fiber to the premises (FTTP) or fiber to the home (FTTH) [17] is a broadband telecommu- nications system based on fiber-optic cables and associated optical electronics for delivery of multiple advanced services such as the triple play of telephone, broadband Internet, and television all the way to the home or business.

Two competing FTTP technologies are active FTTP, also called active Ethernet, and passive optical network (PON) architectures.

Active FTTP networks utilize powered (i.e., “active”) electronic equipment in neighbor-

hoods, usually one equipment cabinet for every 400 to 500 subscribers. This neighborhood equipment performs layer 2/layer 3 switching and routing, offloading full layer 3 routing to the carrier's central office. The IEEE 802.3ah standard enables service providers to deliver up to 100 Mbps full-duplex over one single-mode optical fiber to the premises depending on the provider.

Passive optical network (PON) FTTP networks on the other hand avoid the placement of electronics in the field. PON networks use passive splitters to distribute fiber to individual homes. One fiber is optically split into 16, 32, or 64 fibers, depending on the manufacturer, which are then distributed to residential or business subscribers. In PON architectures, the switching and routing is done at the carrier's central office.

The International Telecommunications Union (ITU) has standardized on two generations of PON. The older ITU-T G.983 standard is based on ATM, and has therefore been referred to as APON (ATM PON) [7]. Further improvements to the original APON standard — as well as the gradual falling out of favor of ATM as a protocol — led to the full, final version of ITU-T G.983 being referred to more often as Broadband PON, or BPON. A typical APON/BPON provides 622 megabits per second (Mbps) of downstream bandwidth and 155 Mbps of upstream traffic, although the standard accommodates higher rates.

UMTS Core Network

ATM is also the data transmission method used within the universal mobile telecommuni- cations system (UMTS) core network [24]. ATM adaptation layer type 2 (AAL2) handles

circuit-switched connections. Packet connection protocol AAL5 is used for data delivery.

6 Conclusion

In this brief chapter, we have discussed several key aspects of ATM. ATM is a cell-oriented switching and multiplexing technology that uses fixed-length cells to carry various types of traffic, such as data, voice, video, multimedia, and so on. through multiple classes of services.

ATM is a connection-oriented technology, in which a connection is established between the two endpoints before the actual data exchange begins.

The ATM protocol reference model is divided into three layers: the ATM adaptation layer (AAL), the ATM layer, and the physical layer, and three planes: user/control plane, layer management and plane management. Four types of AALs were proposed, each supporting a different type of traffic or service that could be used on ATM networks.

ATM was designed to deliver guaranteed quality of service on demand while maximizing the utilization of available network resources. Therefore, effective traffic management mecha- nisms were specified.

ATM has been a partial success as a technology, with widespread deployment, but generally only used as a transport for IP traffic; its goal of providing a single integrated technology for LANs, public networks, and user services has largely failed. This role of integrator in today's networks in played by IP. However, various important ATM concepts are inherited by other technologies, such as MPLS, DSL, and FTTH. It is expected that the best ideas

and lessons of ATM will be used in designing the next Internet.

7 GLOSSARY

AAL - ATM adaptation layer.

ABR - available bit rate.

ANSI - American National Standards Institute.

ATM - asynchronous transfer mode.

ATMF - ATM Forum.

B-ISDN - broadband-integrated services digital network.

BT - burst tolerance.

CAC - connection admission control.

CBR - constant bit rate.

CDV - cell delay variation.

CDVT - cell delay variation tolerance.

CLP - cell loss priority.

CLR - cell loss ratio.

CRC - cyclic redundancy check.

CS - convergence sublayer.

CTD - cell transfer delay.

EFCI - explicit forward congestion indication.

FTP - file transfer protocol.

GCRA - generic cell rate algorithm.

GFC - generic flow control.

HEC - header error check.

IETF - Internet Engineering Task Force.

ISDN - integrated services digital network. ITU -

International Telecommunications Union. LAN - local

area network.

LDP - label distribution protocol.

MAC - medium access control.

MIB - management information base.

MBS - maximum burst size.

MCR - minimum cell rate.

MFA Forum - MPLS frame relay ATM Forum.

MPLS - multi-protocol label switching.

NDIS - network driver interface specification.

NNI - network-to-node interface, or network to network interface.

nrt-VBR - non real-time VBR.

PCR - peak cell rate.

PDH - plesiochronous digital hierarchy. PNNI -

private network-to-network interface. PTI -

payload type identifier.

PVC - permanent virtual circuit.

QoS - quality of service.

rt-VBR - real-time VBR.

RSVP - resource reservation protocol. SAR -

segmentation and reassembly. SCR -

sustainable cell rate.

SDH - synchronous digital hierarchy.

SONET - synchronous optical network.

SSCOP - service specific connection oriented protocol.

SVC - switched virtual circuit.

VBR - variable bit rate.

VC - virtual channel.

VCC - virtual channel connection.

VCI - virtual channel identifier.

VPC - virtual path connection.

VPI - virtual path identifier.

UBR - unspecified bit rate.

UNE - unbundled network elements.

UMTS - Universal Mobile Telecommunication System.

UNI - user-to-network interface.

References

[1] ”AAL1 Circuit Emulation over Packet Switched Networks Version 1.0,” *ATM Forum Techical Committee*, January 2005, http://www.mfaforum.org/ftp/pub/approved-specs/ af-arch-0204.000.pdf

[2] ”Asymmetric Digital Subscriber Line (ADSL),” *IEC Tutorial*, http://www.iec.org/acrobat.asp?filecode=4

[3] ATM, http://www.telecomspace.com/vop-atm.html.

[4] ”ATM Fundamentals,” *IEC Tutorial*, 2005, http://www.iec.org/online/tutorials/atm_fund/index.html.

[5] ”ATM Inter-Network Interface (AINI) Specification Version 1.1,” *ATM Forum Techical Committee*, September 2002, http://www.mfaforum.org/ftp/pub/approved-specs/ af- cs-0125.002.pdf

[6] ”ATM-MPLS Network Interworking Version 2.0,” *ATM Forum Techical Committee*, August 2003, http://www.mfaforum.org/ftp/pub/approved-specs/ af-aic-0178.001.pdf

[7] ”ATM Passive Optical Networks,” *IEC Tutorial*, http://www.iec.org/acrobat.asp? file- code=10.

[8] ”ATM Tutorial,” http://www.npac.syr.edu/users/mahesh/homepage/atm tutorial/.

[9] ”ATM User-Network Interface (UNI) Signalling Specification Version 4.1,” *ATM Forum Techical Committee*, April 2002, http://www.mfaforum.org/ftp/pub/approved-specs/ af-sig-0061.001.pdf

[10] ”ATM User-Network Interface (UNI) Specification Version 4.1,” *ATM Forum Techi- cal Committee*, November 2002, http://www.mfaforum.org/ftp/pub/approved-specs/ af-arch-0193.000.pdf

[11] B-ISDN Intercarrier Interface (B-ICI) Specification, Version 1.0. ATM Forum, May 1994.

[12] ”Converged Data Networks, Bringing Together ATM and MPLS Technologies,” *ATM Forum, White Paper*, April 2004, http://www.mfaforum.org/education/downloads/ CDNwhtpapr.final.pdf.

[13] ”Digital Subscriber Line (DSL) Testing,” *IEC Tutorial*, http://www.iec.org/acrobat.asp?filecode=35

[14] ”Delivering Video over Packet Networks,” *ATM Forum, White Paper*, April 2003, http://www.mfaforum.org/education/downloads/Del.Vid.Final.pdf.

[15] ”DSL and the Evolution of ATM Networks,” *IEC Tutorial*, http://www.iec.org/acrobat.asp?filecode=33

[16] ”EPON,” http://www.ieee802.org/3/

[17] Michael Kunigonis, ”FTTH Explained,” *IEC Tutorial*, http://www.iec.org/acrobat.asp?filecode=51

[18] IEEE Online Standards, http://standards.ieee.org/catalog/olis/lanman.html.

[19] ”Introduction to MPLS,” http://www.riverstonenet.com/support/mpls/intro to .mpls.htm.

[20] Metro Ethernet Forum, http://www.metroethernetforum.org/.

[21] Multiprotocol Label Switching, *IETF Chapter*, http://www.ietf.org/html.charters/mpls-charter.html.

[22] ”Multiprotocol Label Switching,” *IEC Tutorial*, http://www.iec.org/acrobat.asp?filecode=94

[23] MPLS Frame Relay ATM - MFA Forum, http://www.mfaforum.org/.

[24] ”Overview of GSM, GPRS, and UMTS,” http://www.cisco.com/univercd/cc/td/doc/product/wir

[25] ”Private Network-Network Interface Specification Version 1.1 (PNNI 1.1)”, *ATM Forum, Techical Committee*, April 2002, http://www.mfaforum.org/ftp/pub/approved-specs/ af-pnni-0055.001.pdf

[26] ”Traffic Management Specification Version 4.1,” *ATM Forum, Techical Committee*, March 1999, http://www.mfaforum.org/ftp/pub/approved-specs/ af-tm-0121.000.pdf

[27] ”XEPON,” http://grouper.ieee.org/groups/802/3/10GEPON study/index.html

[28] G. J. Armitage, ”Multicast and Multiprotocol support for ATM based Internets,” *ACM SIGCOMM Computer Communication Review,* 25(2): 34-46, April 1995.

[29] R. Y. Awdeh and H. T. Mouftah, ”Survey of ATM switch architectures,” Computer Networks, Vol. 27, No. 12, November 1995, pp. 1567–1613.

[30] R.Y. Awdeh and H.T. Mouftah, ”Design and performance analysis of input-output buffering delta-based ATM switch with backpressure mechanism”, IEE Proceedings: Communications v 141 n 4, Aug 1994. pp 255-264.

[31] H. Badran and H.T. Mouftah, ”ATM switch architectures with input-output buffering: effect of input traffic correlation, contention resolution policies, buffer allocation strategies and delay in backpressure signal”, Computer Networks and ISDN Systems Vol: 26 pp. 1187-1213, 1994

[32] F. Bonomi and K. W. Fendick, ”The rate-based flow control framework for the available bit rate ATM service,” *IEEE Network*, 9(2):25-39, March-April 1995.

[33] A. Charny, D. Clark, and R. Jain, ”Congestion Control with Explicit Rate Indication,” Proc. IEEE International Conference on Communications (ICC’95), June 1995, pp. 1954-1963.

[34] T. M. Chen and S. S. Liu, ”ATM Switching.” Wiley Encyclopedia of Telecommunica- tions, January 2003.

[35] P. Coppo, M. D'Ambrosio and R. Melen, "Optimal cost/performance design of ATM switches", IEEE/ACM Transactions on Networking Vol: 1 Iss: 5 p. 566-75, Oct. 1993.

[36] M. De Prycker, R. Peschi, and T. Van Landegem, "B-ISDN and the OSI protocol reference model," *IEEE Network*, 7(2):10-18, March 1993.

[37] G. Dobrowski and D. Grise, "ATM and Sonet Basics," APDG Publishing, January 2001.

[38] S. Fahmy, "A Survey of ATM Switching Techniques," http://www.cs.wustl.edu/ jain/cis788-95/ atm switching/index.html

[39] P. Ferguson and G. Huston, "Quality of Service: Delivering QoS on the Internet and in Corporate Networks," John Wiley & Sons, Inc., 1998. ISBN 0-471-24358-2.

[40] T. R. Henderson, "Design principles and performance analysis of SSCOP: a new ATM Adaptation Layer protocol," *ACM SIGCOMM Computer Communication Review,* 25(2): 47-59 , April 1995.

[41] M. Hluchyj et al., "Closed-Loop Rate-based Traffic Management," *ATM Forum Contribution* 94-0438R2, July 1993.

[42] O. C. Ibe, "Converged Network Architectures: Delivering Voice and Data Over IP, ATM, and Frame Relay," Wiley, November 2001.

[43] R. Jain, "Congestion Control and Traffic Management in ATM Networks: Recent Advances and A Survey," *Computer Networks and ISDN Systems,* 28(13):1723-1738, October 1996.

[44] R. Jain, ”Congestion control in computer networks: issues and trends,” *IEEE Network*, 4(3):24-30, May 1990.

[45] R. Jain, ”Myths about Congestion Management in High Speed Networks,” *Internet-working: Research and Experience,* Vol. 3, pp. 101-113, 1992.

[46] R. Jain, S. Kalyanaraman, R. Viswanathan, ”The OSU Scheme for Congestion Avoid- ance in ATM networks Using Explicit Rate Indication,” Proceedings WATM’95 First Workshop on ATM Traffic Management, Paris, December 1995.

[47] S. Kalyanaraman, R. Jain, S. Fahmy, R. Goyal, and B. Vandalore, ”The ERICA Switch Algorithm for ABR Traffic Management in ATM Networks,” IEEE/ACM Transactions on Networking, Vol. 8, No. 1, Feburary 2000, pp. 87-98

[48] M. Kaycee, A. Lin, A. Malis, and J. Stephens, ”PPP Over AAL5,” *Request for Com- ments 2364 IETF*, 1998.

[49] W. C. Lee, ”Topology Aggregation for Hierarchical Routing in ATM Networks,” *ACM SIGCOMM Computer Communication Review,* 25(2): 82-92, April 1995.

[50] L. Mamakos, K. Lidl, J. Evarts, D. Carrel, D. Simone, and R. Wheeler, ”A Method for Transmitting PPP Over Ethernet (PPPoE),”IETF Request for Comments: 2516,” February 1999.

[51] D. McDysan and D. Spohn, ”ATM Theory and Applications,” *McGraw-Hill*, 1999.

[52] R. O. Onvural, ”Asynchronous transfer mode networks : performance issues”, Boston : Artech House, 1994. Chapter 7, pp. 207-252.

[53] K. Ramakrishnan and R. Jain, ”A Binary Feedback Scheme for Congestion Avoidance in Computer Networks with Connectionless Network Layer,” ACM Transactions on Computer Systems, Vol. 8, No. 2, May 1990, pp. 158-181.

[54] S. K. Rao and M. Hatamian, ”The ATM Physical Layer,” *ACM SIGCOMM Computer Communication Review,* 25(2):73-81, April 1995.

[55] Thomas G. Robertazzi, ”Performance evaluation of high speed switching fabrics and networks : ATM, broadband ISDN, and MAN technology”, New York : IEEE Press, 1993.

[56] L. Roberts, ”Enhanced PRCA,” *ATM Forum Contribution* 94-735R1, September 1994.

[57] E. Rosen, A. Viswanathan and R. Callon, ”Multiprotocol Label Switching Architec- ture,” *Request for Comments 3031 IETF*, 2001.

[58] G. C. Sackett and C. Metz, ”ATM and Multiprotocol Networking,” McGraw-Hill, Jan- uary 1997.

[59] H. Saito, ”Teletraffic Technologies in ATM Networks,” Artech House, 1993. ISBN 0-89006-622-1.

[60] M. Sexton and A. Reid, Broadband Networking: ATM, SDH and SONET, Artech House Inc., Boston, London, 1997. ISBN 0-89006-578-0.

[61] R. J. Simcoe and T.-B. Pei, ”Perspectives on ATM Switch Architecture and the Influ- ence of Traffic Pattern Assumptions on Switch Design,” *ACM SIGCOMM Computer Communication Review,* 25(2):93-105 , April 1995.

[62] B. Stiller, ”A Survey of UNI Signaling Systems and Protocols for ATM Networks,” *ACM SIGCOMM Computer Communication Review,* 25(2):21-33 , April 1995.

[63] T. Suzuki, ”ATM Adaptation Layer Protocol,” *IEEE Communications Magazine*, 32(4):80-83, April 1994.

Prentice-Hall, 1993.

[64] R. J. Vetter, ”ATM concepts, architectures, and protocols,” *Communications of the ACM*, 38(2):30-38, 109, February 1995.

File Catching Scheme

File caching has implemented in several file system for centralized time-sharing systems to improve file I/O performance. The idea in file caching in there systems is to retain recently accessed file data in main memory, so that repeated accesses to the same information can be handled without additional disk transfers. Because of locality in file access patterns, file caching reduces disk transfers substantially, resulting in better overall performance of the file system. The property of locality in file access patterns can as well be exploited in distributed systems by designing a suitable file-caching scheme. In addition to better performance, a file-caching scheme for a distributed file system may also contribute to its scalability and reliability because it is possible to cache remotely located data on a client node. Therefore, every distributed file system in serious use today uses some form of file caching.

File Application & Fault tolerance

In engineering, **fault-tolerant design**, also known as **fail-safe design**, is a design that enables a system to continue operation, possibly at a reduced level (also known as graceful degradation), rather than failing completely, when some part of the system fails. The term is most commonly used to describe computer-based systems designed to continue more or less fully operational with, perhaps, a reduction in throughput or an increase in response time in the event of some partial failure. That is, the system as a whole is not stopped due to problems either in the hardware or the software. An example in another field is a motor vehicle designed so it will continue to be drivable

if one of the tires is punctured. Distributed fault-tolerant replication of data between nodes (between servers or servers/clients) for high availability and offline (disconnected) operation. Distributed file systems, which also are parallel and fault tolerant, stripe and replicate data over multiple servers for high performance and to maintain data integrity. Even if a server fails no data is lost. The file systems are used in both high-performance computing (HPC) and high-availability clusters.

Naming: - Features, System Oriented Names

Naming in distributed systems is modeled as a string translation problem. Viewing names as strings and name resolution mechanisms as syntax directed translators provides a formal handle on the loosely understood concepts associated with naming: we give precise definitions for such informal terminology as name spaces, addresses, routes, source-routing, and implicit-routing; we identify the properties of naming systems, including under what conditions they support unique names, relative names, absolute names, and synonyms; and we discuss how the basic elements of the model can be implemented by name servers. The resources in a distributed system are spread across different computers and a naming scheme has to be devised so that users can discover and refer to the resources that they need.

An identifier that:

• Identifies a resource

– Uniquely

– Describes the resource

• Enables us to locate that resource

– Directly

– With help

• Is it really an identifier

– Bijective, persistent

An example of such a naming scheme is the URL (Uniform Resource Locator) that is used to identify WWW pages. If a meaningful and universally understood identification scheme is not used then many of these resources will be inaccessible to system users.

Objects:

• processes, files, memory, devices, processors, and networks.

Object access:

• Each object associate with a defined access operation.

• Accesses via object servers

Identification of a server by:

• Name

• Physical or Logical address

• Service that the servers provide.

Identification Issue:

• Multiple server addresses may exist requiring a server to move requiring the name to be changed.

Object Locating Mechanism

Distributed systems based on objects are one of the newest and most popular approaches to the design and construction of distributed systems. CORBA platform is built from several standards published by the organization OMG (Object Management Group), whose objective is to provide a common system for the construction of distributed systems in a heterogeneous environment. The role of the ORB is to deliver the tasks the system the following services: network communication, locating objects, sending notifications too

Mca

objects, the results to clients. The basic features of CORBA are: independence from the programm language by using language IDL and the independence of the system, hardware, communication (IIOP).

Java RMI (Remote Method Invocation) is a second example of a distributed system platform based objects. RMI is a structure built based on Java. The model presupposes the existence of the facility, loca in the address space of the server and client, which causes the object operations. Remote state of the ob is located on a single machine, and the local interface of an object is released.

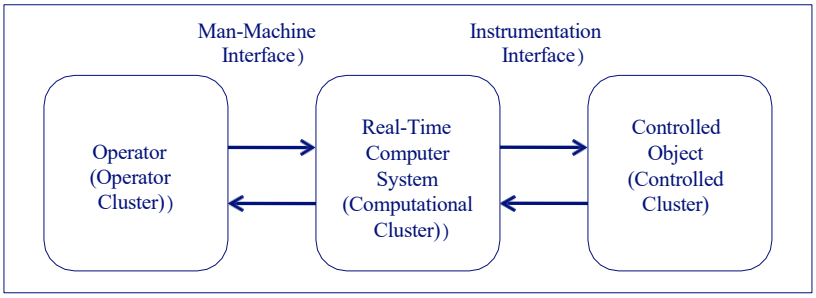
Human Oriented Name

Names allow **us** to identify objects. to talk about them and to access them. Naming is therefore an import issue for large scale distributed systems. It becomes a critical issue when those systems are intended support collaboration between humans. **A** large volume of research has already been published on subject of naming, particularly within the context of name servers and directories. However, it **can** argued that the hierarchical nature ***of*** many of the naming mechanisms so far proposed is too constraining fully support the great flexibility of human naming practice, particularly where group work is concerned.

Mca

When is a Computer System Real-Time?

- The correctness of the system behavior depends on…
  - The logical results of the computations,
  - ***Physical instant*** at which these results are produced.)



<Real-time system>)

The Property of Real-Time System

- A real-time computer system must react to stimuli from the controlled object (or the operator) within time intervals dictated by its environment.
- Deadline : the instant at which a result must be produced.
- Classification or RT-system
  - Soft real-time computer system.
  - Hard real-time computer system.)

Real-Time Entity

- A controlled object, e.g., a car or an industrial plant, changes its ***state*** as a function of time.
- Real-Time Entity (RT-Entity)
  - *A set of state variable* that we are normally interested in.
  - Car : the speed of the car, the position of switches on the dash board, the position of a piston in a cylinder.

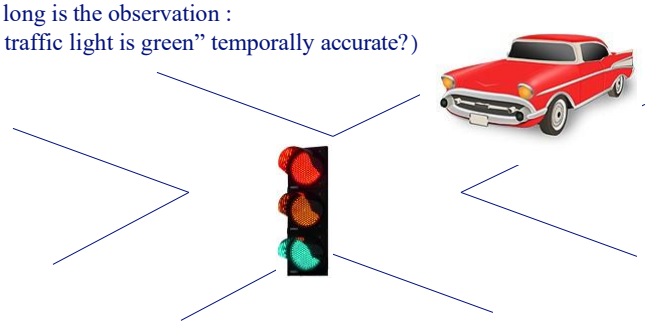
Real-Time Image

- The first functional requirement of a real-time computer system is the ***observation*** of the RT entities.
- Real-Time Image (RT-Image) : An observation of RT-Entity
- The property of RT-Image
  - A given RT-image is only *temporally accurate* for a limited

The Meaning of "Temporally Accurate"

- An upper bound for the accuracy interval is given by the duration of the yellow phase of the traffic light

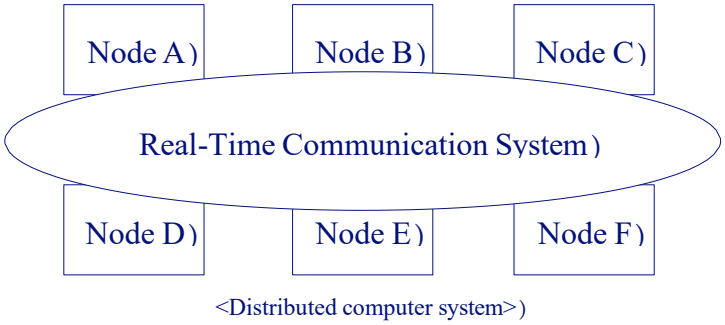
How long is the observation :
"The traffic light is green" temporally accurate?)



<Temporal accuracy of the traffic light information>)

The Distributed Real-Time System

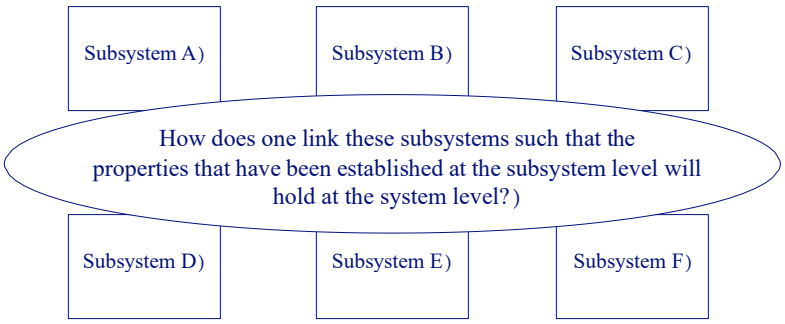
- If the real-time computer system is distributed, it consists of a set of (computer) nodes interconnected



<Distributed computer system>)

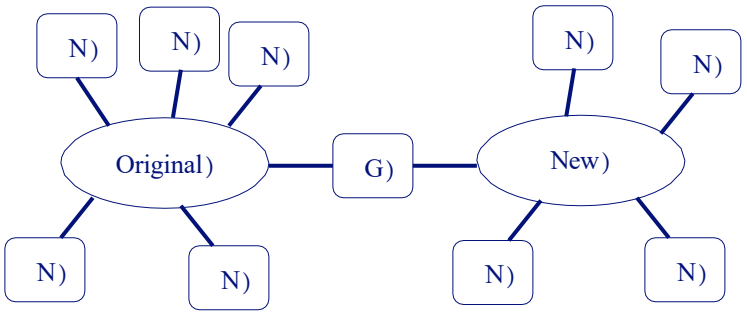
Issue 1 : Composability

- An architecture is said to be ***composable*** with respect to a specified property if the system integration will not invalidate this property once the property has been established at the subsystem level.



Issue 2 : Scalability

- In a scalable architecture, the complexity of reasoning about the proper operation of any system function is independent of the system size.



<Transparent expansion of a node into a new cluster>)

Issue 3 : Dependability

- Error-containment regions.
  - A fault-tolerant system must be structured into partitions in such a way that the consequences of faults that occur in one of these partitions can be detected and corrected or masked before these consequences corrupt the rest of the system.
- Replication
  - Given that the failure hypothesis holds, node failures can be masked by providing actively replicated nodes.
  - The replicas must show deterministic behavior.

Reference

- Hermann Kopetz. *Real-Time Systems: Design Principles for Distributed Embedded Applications*. Kluwer Academic Publishers, 1997.