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# Comparative study of networks using packet and circuit switching within a single network

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# Comparative Study of Networks using Packet and Circuit Switching within a single Network

by

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## ABSTRACT

During the last couple of years, in addition to voice, other types of communications network services are becoming increasingly important. These are interactive data, facsimile, slow scan image, and bulk data. Typically, these services are delivered by separate networks using various kinds of switching technology, such as packet, circuit, or message switching. Recently, much of the focus has been on the integration of all types of communication services within the same switch or network, especially within the telephony and business industry. Integration of the communication services is being realized by integrating packet and circuit switching within the same switch or network. The overall goal of this thesis is to present the key aspects of the integration of circuit and packet switching within the same switch/network.

**KEYWORDS:** packet / circuit switching, SENET, PACUIT, master frame, TASI, TADI, ISDN.

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# CHAPTER 1

# INTRODUCTION

## 1. Introduction

During the last couple of years, in addition to voice, other types of communications network services are becoming increasingly important. These network services are interactive data, facsimile, slow scan image, and bulk data. Typically, they are delivered by separate networks using various kinds of switching technology, such as packet, circuit, or message switching. If these switching technologies are combined within a single Network system the network is generally referred to as a Hybrid, or Integrated Network. This type of network can provide benefits such as the reduction of telecommunication costs, high reliability, and optimum response time, since separate resources such as switching and transmission facilities, would not be required for each network.

With the recent technological developments in electronic/digital switches and transmission media, it is feasible to have an Integrated/Hybrid network in order to utilize the wide variety of communications services and facilities such as message(datagram) delivery, digitized voice, bulk file transportation and interactive digitized

video, using the same network. For example, an Integrated Services Digital Network (ISDN) supplying end-to-end digital transmission in which voice and data services will be provided using common transmission and switching facilities, is currently being developed.

Given the growing interest in realizing the ISDN, especially within the telephony and business industry, it is worthwhile to investigate integrated network systems. There is one study [RUD78] that summarizes the results of the theoretical and experimental work done during the years 1973 - 1978. A number of aspects of integrated circuit-switching and packet-switching systems are discussed in the study, such as:

- performance and cost comparison of the switching techniques;
- multiplexing at the trunk level;
- an integrated interface based on the CCITT X.21 specifications;
- routing for the packet and circuit switched subnetworks;
- a multiple-processor design for the network nodes;
- architecture

However, topics such as flow control for the integrated environment and integrated network performance are not discussed. Moreover, since 1978, different integrated

switching schemes/approaches, implementations, routing strategies, and performance techniques have been published.

The overall goal of this thesis is to study in detail the Integrated network systems developments made after the year 1978. Chapter 2 outlines packet and circuit switching characteristics, types of communication traffic, and reasons for integrating; Chapter 3 describes the approaches to integration; Chapter 4 describes the routing and flow control techniques; Chapter 5 outlines the integrated switching system architecture; and Chapter 6 outlines the ISDN, and some of the commercial switches. Finally the last chapter describes the performance techniques and compares some of the integrated switches and networks.

## CHAPTER 2

## PACKET/CIRCUIT SWITCHING

## 2. Packet and Circuit Switching

### 2.1. Packet Switching

Packet Switching is a data transmission technique in which user information is segmented and routed in discrete data envelopes called packets, each with its own appended control information for routing, sequencing, and error checking. This transmission technique allows a communications channel to be shared by many users, each using the circuit only for the time required to transmit a single packet.

#### 2.1.1. Characteristics of Packet Switching

The characteristics of packet switching are as follows.

- There is no direct electrical connection between the communicating parties.
- It is fast enough for conversational interaction between data machines.
- Messages are stored until delivered.
- It is designed to handle bursts of data.

- The route is established dynamically for each packet.
- There is a negligible delay in setting up the call.
- The packet is returned to the sender if it is undeliverable.
- An overload causes an increased delivery delay (but delivery time is still short). Blocking occurs when saturation is reached.
- Small switching computers are used with no filing facilities.
- There is some protection against loss of packets. End user protocols can be employed in message protection because of the conversational interaction.
- Charges for short transmission can be lower than over the telephone network.
- Lengthy transmission are chopped into short packets. Very long messages must be divided by the users.
- High traffic volumes needed for economic justification.
- The network can perform speed or code conversion.
- Delayed delivery is not permitted without a special network facility.
- Broadcast and multi-address messages are not permitted without a special network facility.

## 2.2. Circuit Switching

Circuit Switching is a process of establishing and maintaining a circuit between two or more users on demand, giving the users exclusive use of the circuit until the connection is released.

### 2.2.1. Characteristics of a Circuit Switching

The following are characteristics of a circuit switching:

- It is the equivalent of wire circuit connection between the communicating parties for a brief period.
- Real-time or conversational interaction between the parties is possible.
- Messages are not stored.
- It is designed to handle long continuous transmissions.
- The switched path is established for the entire conversation.
- There is a time delay in setting up a call, followed by a negligible transmission delay.
- Busy signal occurs if the called party is occupied.
- An overload causes increased delay and/or increased probability of a busy signal.
- Computerized switching offices are used.
- Protection against loss of messages is the responsibility of the end users.

- It is relatively expensive to a user whose transmissions are very short.
- A transmission of any length is permitted.
- It is economical in handling low traffic volumes if the public telephone network is employed.
- Delayed delivery may be permitted if the delay is short.
- It provides end-to-end transmission.

## 2.3. Classes of Communications traffic

There are three general traffic classes - continuous, bursty, and interruptible [ROS82]. Table 1 lists the characteristics and examples of the three traffic classes.

### 2.3.1. Continuous

Continuous traffic is characterized by a continuous flow of information over a fixed communications path, with real-time, end-to-end connectivity. Continuous traffic flows between similar end users and has long holding times. Some delay at the beginning of the traffic can be tolerated but continuous traffic needs a high-quality end-to-end connection. No error correction is feasible because of the real-time connectivity. New services are blocked if network resources are limited. Examples include voice, video, and facsimile transmission.

### 2.3.2. Bursty

Bursty traffic is characterized by discrete messages, transactions, or portions of communications flows that can be handled as complete entities. Bursty traffic operates between dissimilar users and needs error correction and non-blocking service. Examples are Interactive computer operations, query/response, and distributed data-base operations.

### 2.3.3. Interruptible

Interruptible traffic is characterized by a long message that can be interrupted when necessary for the handling of more time-sensitive traffic. Long delays are tolerabled in support of overall economy of transmission.

Interruptible traffic needs error correction and non-blocking service. Examples are bulk digital data, large data files or remote program loads, or overnight electronic mail services, and, in a certain sense, voice traffic. (A typical voice conversation on an expanded time scale has significant interruptions).

Table 1. Classes of Communications traffic

| Interruptible | Bursty | Continuous |
|---|---|---|
| Long data stream | Discrete message | Continuous-real time |
| Near real-time | Not real-time | Connection delay |
| Long delays vs. economy | Delay variable | Fixed delay |
| Indefinite lengths | Short total lengths | Long holding time |
| Nonblocking | Nonblocking | Blockable |
| Error controlled | Error controlled | No error control |
| Arbitrary users | Arbitrary users | Compatible users |
| Bulk Data, Files, Data Base | Interactive, Query/Response | Video, Voice, Facsimile |

## 2.4. Factors Leading to the Combination of Packet and Circuit Switch Networks

Both continuous and bursty traffic match the capabilities of circuit and packet switching, respectively. Interruptible traffic does not match the capabilities of either kind of switching. It is difficult to capitalize on Interruptible traffic's interruptibility in a circuit switched network. Interruptible traffic tends to waste capacity in a packet switched network due to the high overhead associated with the long message lengths.

Integration of circuit and packet switching characteristics in a single network facilitates handling the classes of communications traffic in the most efficient way, depending on that network's characteristics, as well

as the requirements of all the other traffic competing for network resources. Transmission efficiency would be maximized by pooling total capacity and making it available to whichever service has the highest current demand. In an integrated network, the digitized voice message could be transmitted on an interruptible basis via packet switching and delivered to the destination user in the form of a continuous, circuit switched message.[1] Moreover, studies [RUD78][McA78] have shown that circuit switching is more cost- effective for data traffic characterized by long continuous messages, while packet switching is more cost-effective for traffic composed of short messages. Therefore, a case can be made that the user would benefit by providing both circuit and packet switching services in the same network[McD83].

[1]Computer Consoles Inc's DAIS/ARS system

# CHAPTER 3

# APPROACHES TO INTEGRATION

## 3. Approaches to Integrating Circuit and Packet Switching

Following are the different ways that circuit and packet switching can be combined within a single network.

### 3.1. Integration through Shared Transmission

#### 3.1.1. Common Trunking

Both circuit and packet switches have equal access to common transmission facilities via multiplexing or concentrating equipment, shown in Figure 1(a). However, integration at this level does not really provide much improvement in transmission efficiency or utilization, or the exchange of traffic between different user terminal types.

#### 3.1.2. Embedded Network

In the embedded network (Figure 1(b)) the packet switched network is "embedded" in a circuit switched network, with each switch having its own community of users. The packet switched network shares the transmission facilities of the overall network by acting as if the packet switches were user terminals within the circuit switched network.

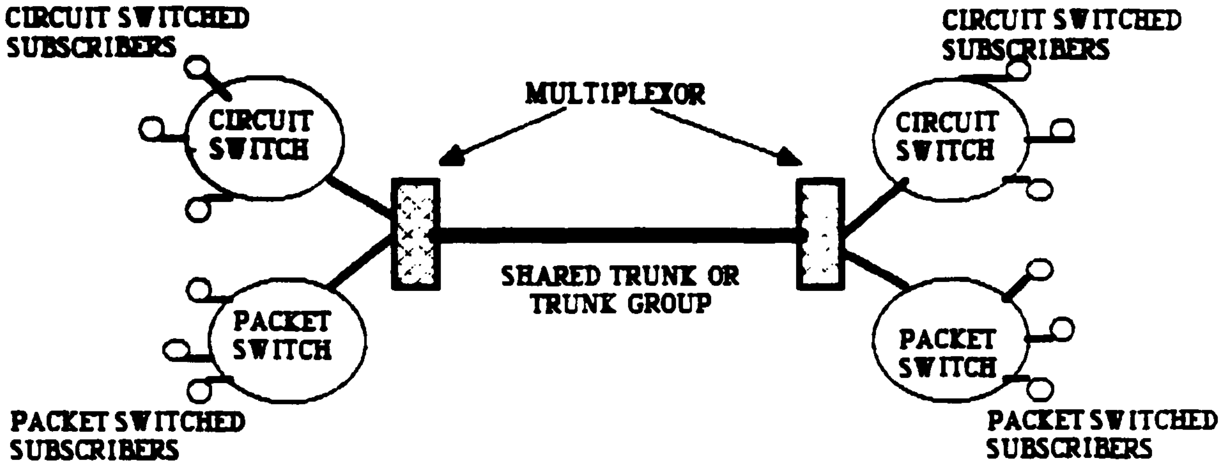



Figure 1(a) INTEGRATION THROUGH COMMON TRUNKING

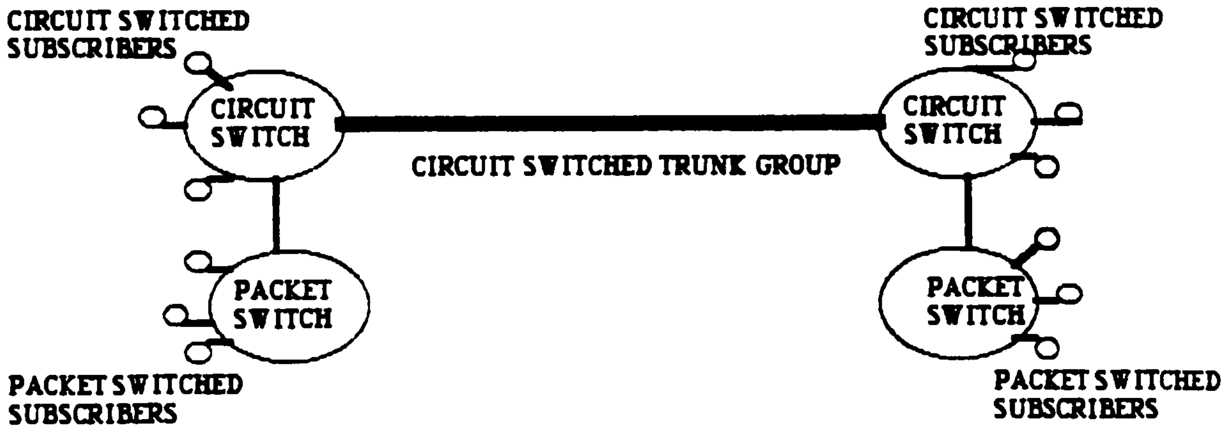



Figure 1(b) INTEGRATION THROUGH EMBEDDING PACKET SWITCHES WITHIN A CIRCUIT SWITCHED NETWORK

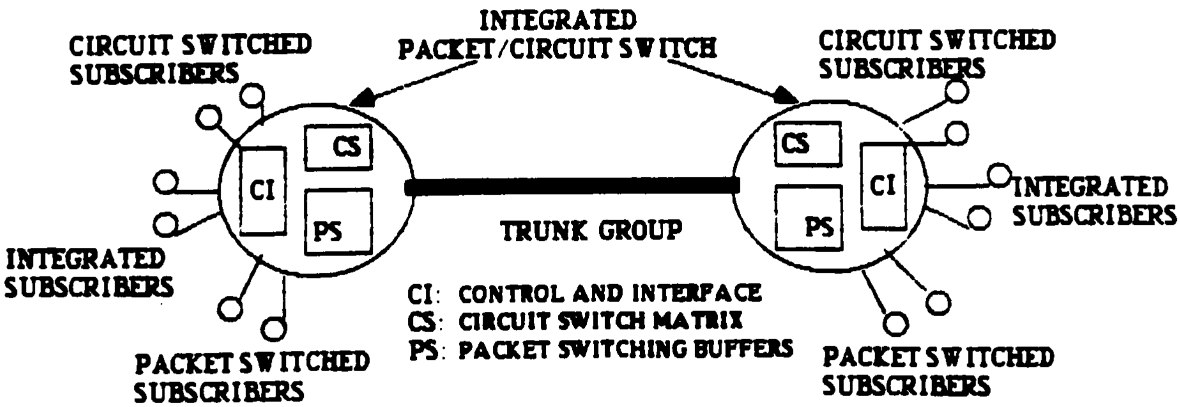



Figure 1(c) INTEGRATION THROUGH COMBINED CIRCUIT AND PACKET MATRICES WITH COMMON CONTROL

### 3.1.3. Combined Matrices

A single processor is used to combine two switching matrices (Figure 1(c)). One of the matrices would be a circuit switching matrix, while the other would be a system of buffers used to hold the information processed as packet switched data. Request for services would come into the single processor, which would decide on the best way to meet the service demand and route the traffic through the most appropriate matrix.

## 3.2. Intergration through Processing and Software

This is the most practical approach utilizing sophisticated switching processors and software to achieve integration. There are three structures utilizing this approach.

The first basically uses a circuit switched technique but adds some aspects of the multiplexing achieved by packet switching.

The second approach uses a packet switched network but limits the dynamic routing features and uses some additional memory for reference tables. This approach achieves a circuit switched connection through the packet network.

The third approach uses dynamically processed master frames, with advanced interswitch signaling, combining the

best features of both technologies.

### 3.2.1. The Circuit Switched Technique

The principle of Time Assignment Speech Interpolation (TASI) is applied (i.e., telecommunications channels are activated by the presence of speech and deactivated for the brief periods when speech is not present). This allows other conversations (talkspurts) to be used on the same circuit during otherwise idle conversation periods. If the number of channels is large enough, most of the idle time on the transmission link can be filled, resulting in an enhancement in transmission capacity greater than twofold.

A disadvantage of this technique is that talkspurts cannot be managed by delay, but must be treated with a loss or blocking strategy. This will tend to decrease overall network efficiency, since talkspurts are either accepted or blocked; they cannot be stored and transmitted later when capacity becomes available. Functional Implementation of time assignment interpolation(TASI) is shown in Figure 2.

A more flexible implementation using the common switch is shown in Figure 3. In this approach when the switch detects a pause in the activity on one of the circuit switched channels, it inserts a packet of data

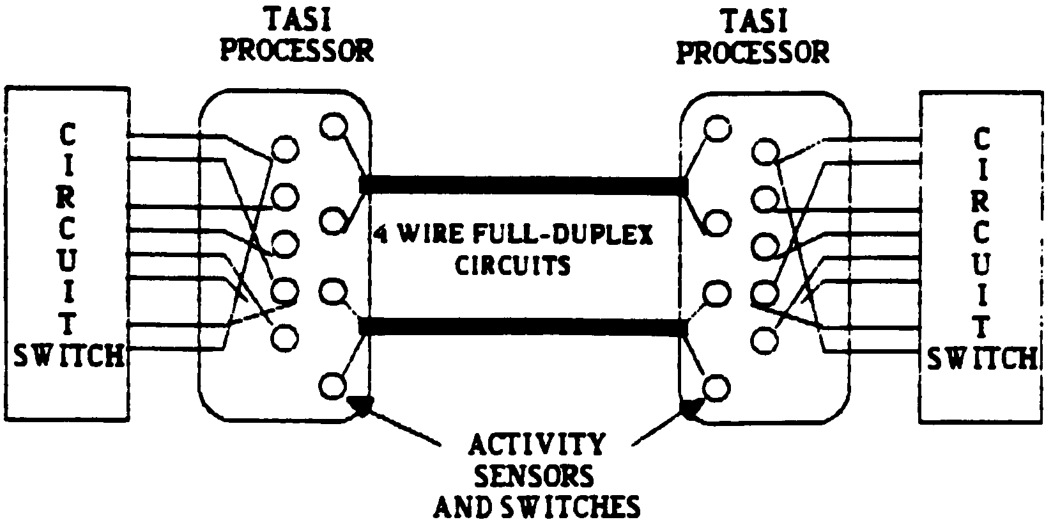



Figure 2. FUNCTIONAL IMPLEMENTATION OF TIME ASSIGNMENT SPEECH INTERPOLATION (TASI)

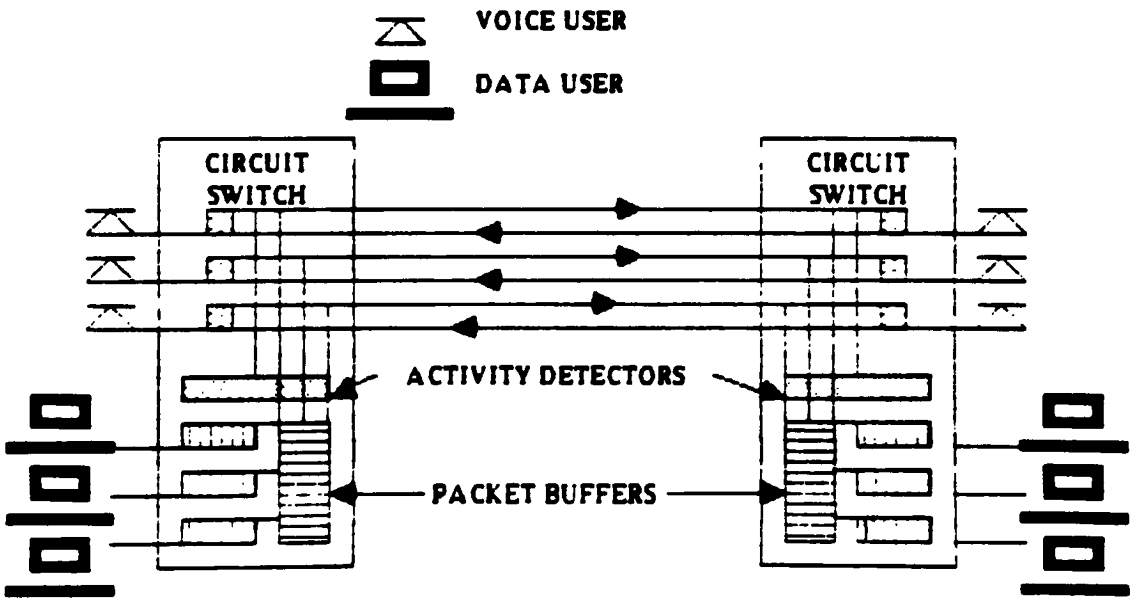


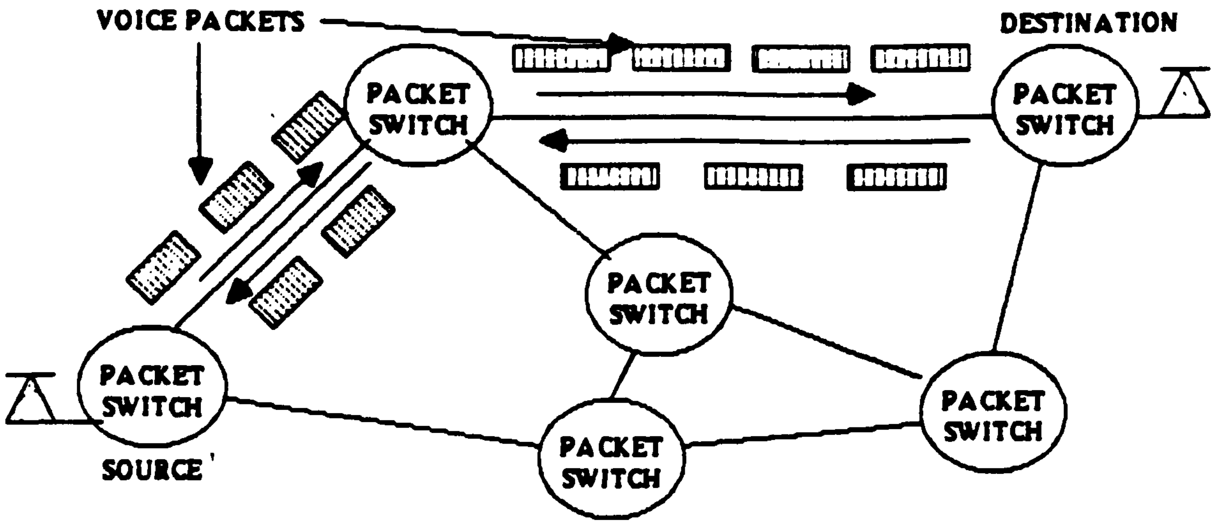

Figure 3. FUNCTIONAL IMPLEMENTATION OF TIME ASSIGNMENT DATA INTERPOLATION (TADI)

destined to the next switch. On a 64k bits/s digital voice channel a 1k bit packet will fit into a time period of about 15 milliseconds, which is less than intersyllable gaps in normal speech. With this time assignment digital interpolation (TADI) approach, the network can carry very large amounts of data traffic with little or no increase in transmission capacity.

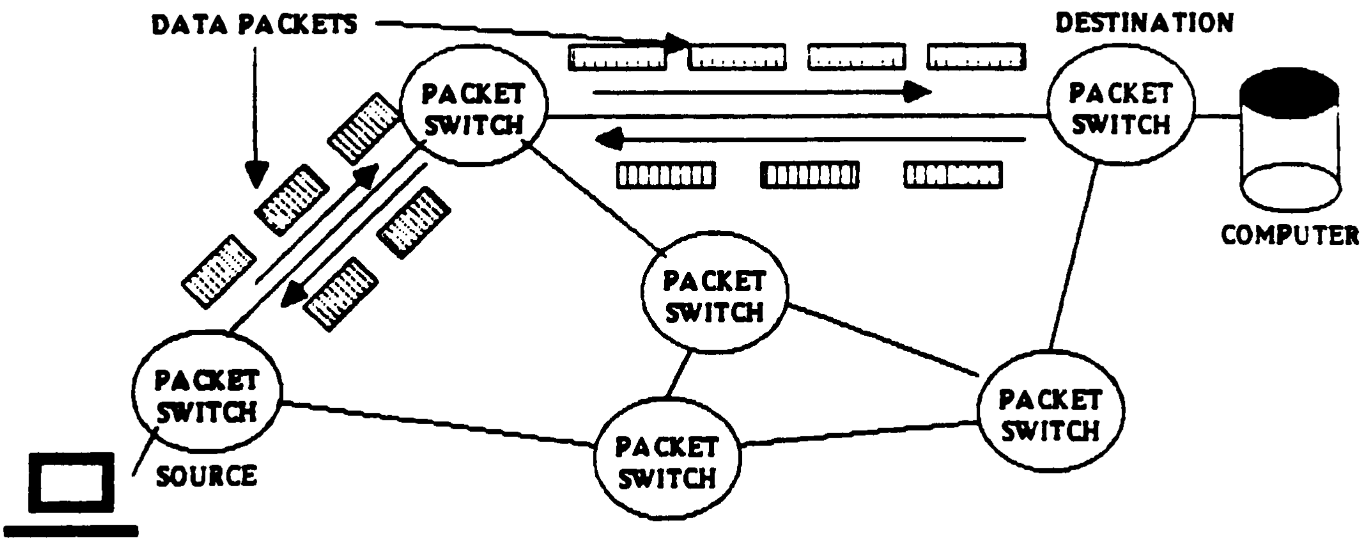
### 3.2.2. The Packet Switched Technique

The packet switched network is used to emulate the circuit switched connections. The circuit switched connection is provided by a continuous stream of packets with only a few bits of overhead in each packet. These packets flow through the network, over a path determined at call set up-time and held constant throughout the connection. Error checking is done on an end-to-end basis, not on node-to-node basis. Acknowledgments and flow control are done on an end-to-end basis. Short data transactions can proceed through the packet network in the normal fashion with dynamic routing, flow control and other features of packet switching. A typical configuration is shown in Figure 4.

The disadvantages of this approach are: more delay due to the reassembly process (for both voice and data) and software complexity of the routing algorithm.




PACKET-SWITCHED VOICE SYSTEM




PACKET-SWITCHED DATA SYSTEM

Figure 4. PACKET SWITCHED NETWORK

### 3.2.3. Master Frame Technique

Integration of circuit and packet switching in a single network is achieved by a master framing technique and dynamic management of the capacity allocation between adjacent switches within the master frame. Master framing is the only technique that actually achieves integration of circuit and packet switching.

The master framing technique is based on the utilization of a dynamic time-division multiplex structure, which acts as a fixed channel allocation for circuit switched traffic but uses any temporary excess capacity to transmit packets associated with bursty and interruptible traffic.

#### 3.2.3.1. Structure and Operation

The dynamic master frame multiplexing depends on very-high processing within the network switches and generally, high capacity trunks. The traffic changes rarely compared to the rate of frame transmission. The initial structure of the frame approximates that of a standard digital time-division multiplexor, with distinct time slots within the individual frames. These slots can be assigned to particular user pairs for the duration of a call. A large buffer in each switch is used to assemble each frame so that any unused time slots are recognized and used to carry packet traffic.

In the master framing approach (Figure 5), the time interval between two successive frames is fixed; that is, for a fixed circuit switched channel, each frame contains the same number of bits, associated with the data rate of the circuit switched channel and the frame interval. Each successive frame contains the same number of data bits.

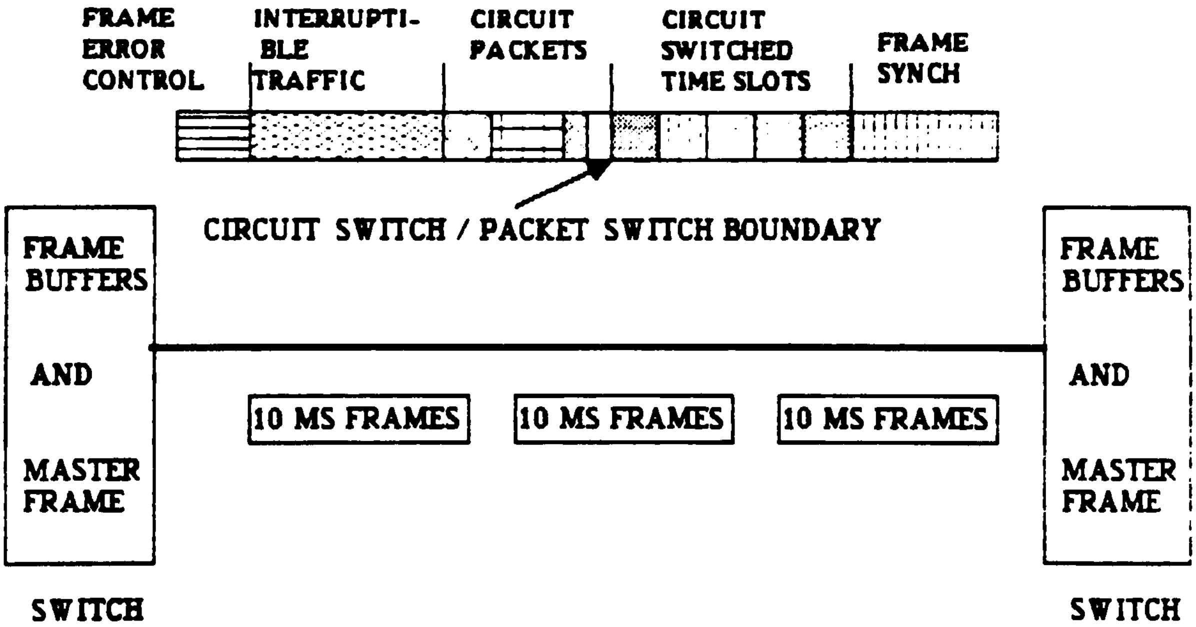



Figure 5. THE MASTER FRAME APPROACH TO INTEGRATED SWITCHING

However, the fraction of each total frames assigned to circuit switched channels need not be the same in each frame. The frame is built in the buffers associated with each of the switches during the time interval when the preceding frame is being transmitted. As the frame is assembled, the subframes associated with each circuit switched connection are placed in the leading part of the frame. Any residual capacity beyond that needed to support the currently active circuit switched channel remains empty.

### 3.2.4. Implementation

SENET and PACUIT are the practical means of implementing a master frame integration of circuit and packet switching.

#### 3.2.4.1. SENET

The implementation called Slotted Envelope NETwork (SENET) views the master frame as a large envelope into which data is placed in the form of both dedicated and dynamically assigned slots. In this implementation, voice is circuit-switched, interactive data are packet-switched, and bulk data are either circuit- or packet-switched.

### 3.2.4.1.1. Design Concept

The SENET concept divides the trunk occupancy into constant-period, self-synchronizing master frames (Figure 6). A start-of-frame (SOF) marker provides the self-synchronizing feature for identifying each master frame. Following this marker, there are two frame regions: Class I and Class II. The Class I region contains those types of traffic normally associated with circuit-switched traffic.

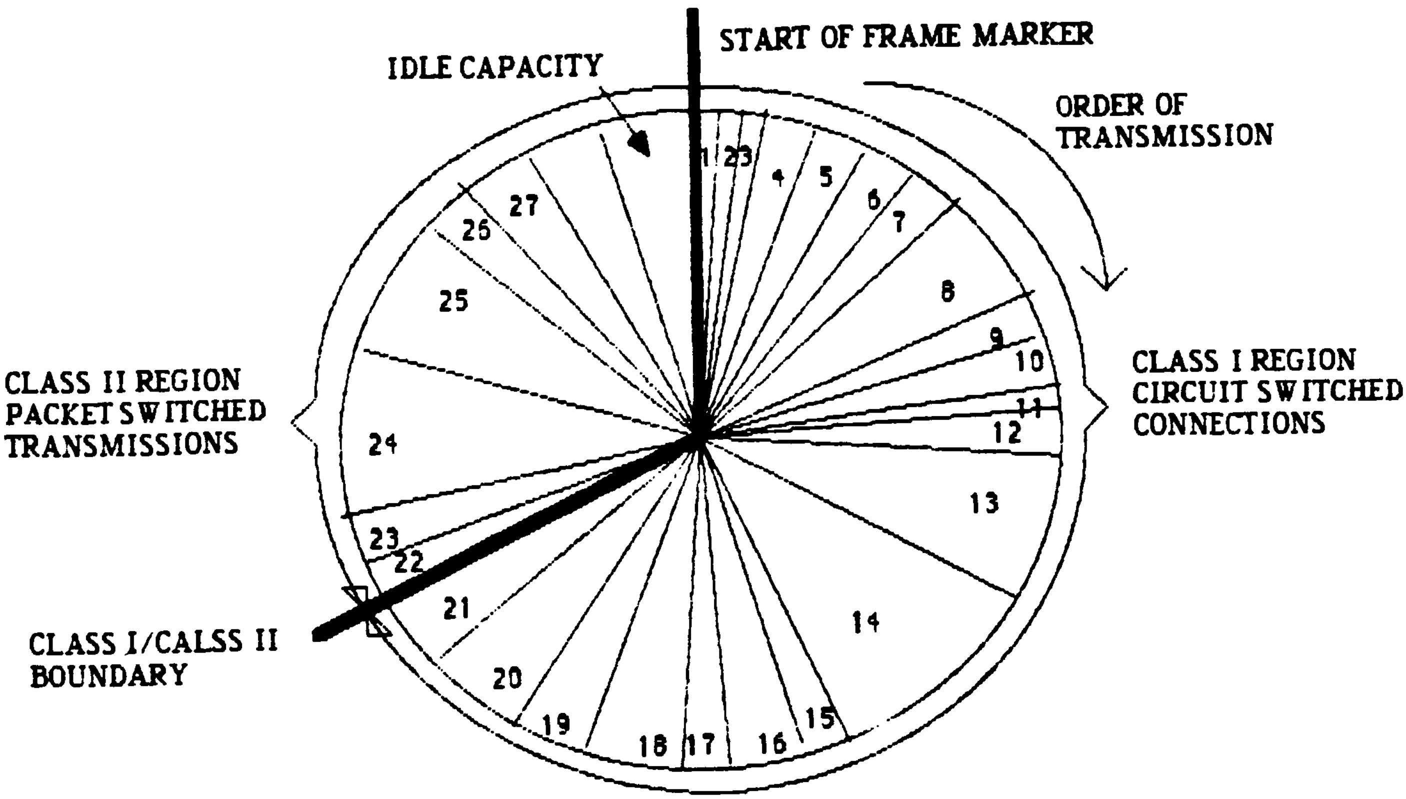



Figure 6. MASTER FRAME ALLOCATIONS FOR THE SENET CONCEPT

The Class II region contains those types of traffic normally associated with packet-switched traffic. Connections in the Class I region are allocated and maintained by frame allocation maps located in switch software at each end of the link between two switches. Changes in these maps are implemented by common channel signaling messages. The common channel signaling messages are handled as Class II packet data.

Within the Class I region, call location tends to reflect primarily the age of the call. The closer a given call is to the SOF marker, the older, in general, that call is. As calls are terminated, Class I region shrinks and the Class I/II boundary moves toward the start-of-frame marker. This shrinking of the Class I region causes the Class II region to expand, thereby allowing more data packets to be transmitted. In this way, the regions react dynamically to changes in one another, resulting in a maximization of the throughput.

For the duration of a call, the minimum Class I region is fixed. However, the number of data packets within the Class II region is dynamic, depending on the available remaining capacity.

Additions to the Class I region are made at the end of the region. As calls terminate, all subsequent allocations move up toward the SOF marker. Thus, the Class I

region vary dynamically, but always in a direction providing the minimum size for Class I calls currently being serviced, and therefore providing the maximum amount of master frame capacity for Class II transmissions.

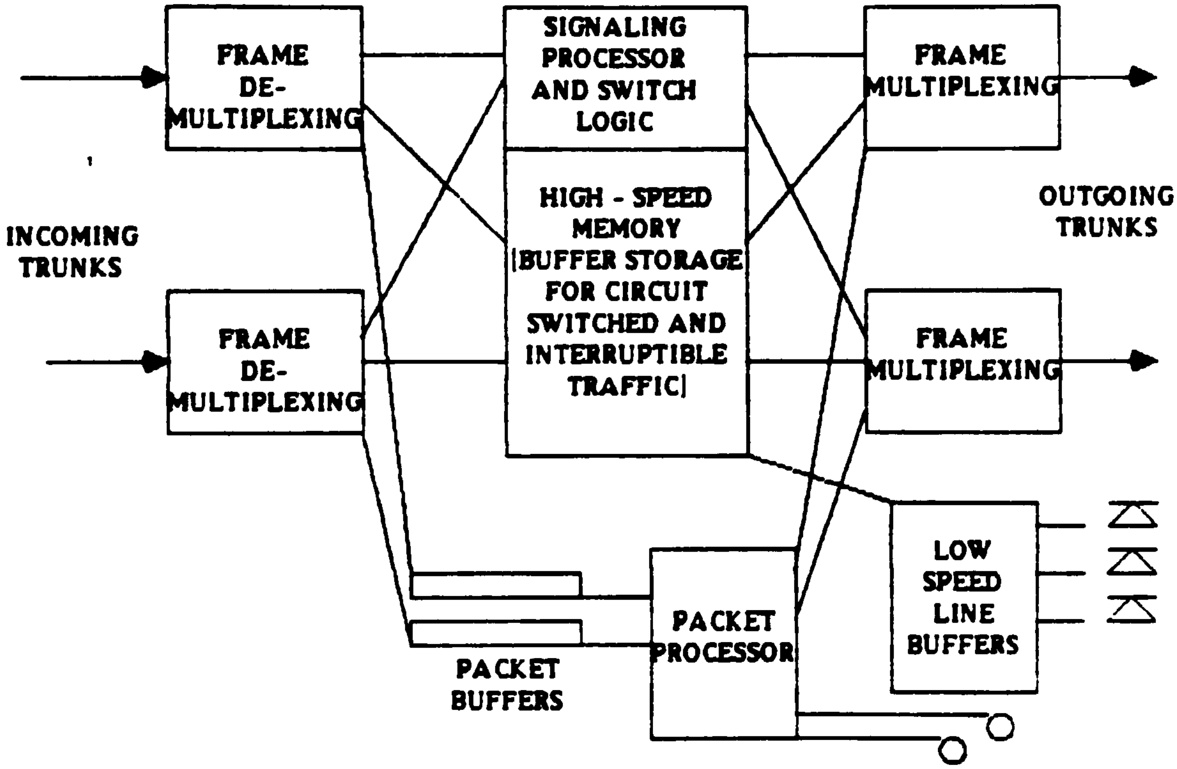
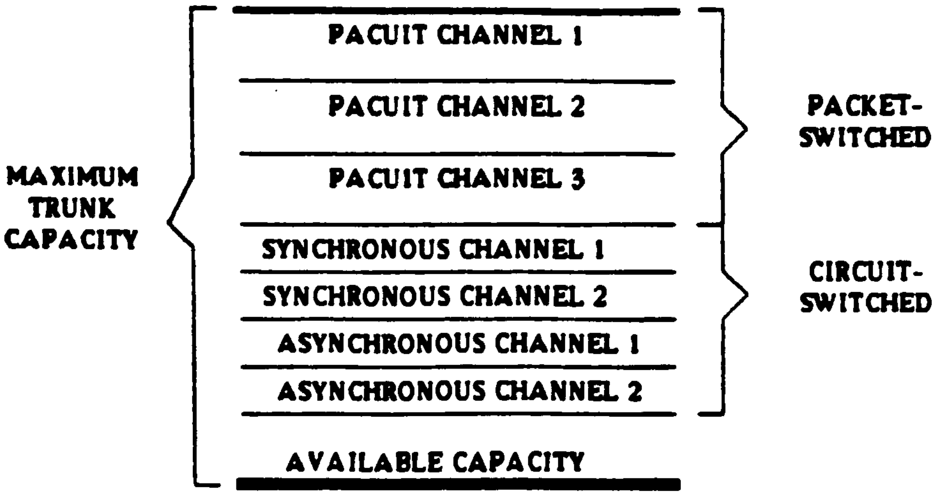



Figure 7. FUNCTIONAL IMPLEMENTATION OF A SENET MASTER FRAME INTEGRATED SWITCH

The main advantage of the SENET scheme with respect to the conventional character-interleaved time-division scheme is the improvement in bandwidth sharing. Conversely the transmission efficiency suffers as a result of the inefficiency in transmitting circuit-switched traffic, since according to author [McD83], the circuit traffic is delayed on the average by a half frame interval at each intermediate node. The SENET technique has not yet been commercially implemented. Although the possible functional configuration for such a switch, using parallel, distributed processing for the frame composition/decomposition functions, has been proposed [ROM82] (Figure 7).

### 3.2.4.2. PACUIT

PACUIT, or "PACket and CirCUIT switching", has already been implemented in commercial product lines. PACUIT network switching components suitable for a user to develop a private switched network are produced by California based TRAN Telecommunications Corporation (Figure 8). These switches combine the master frame concept for integrated switching with time-division circuit switching and packet switching. In this implementation, voice and interactive data are packet-switched, while bulk data messages are circuit-switched.




TRUNK STRUCTURE ON TRANS'S PACUIT NETWORK

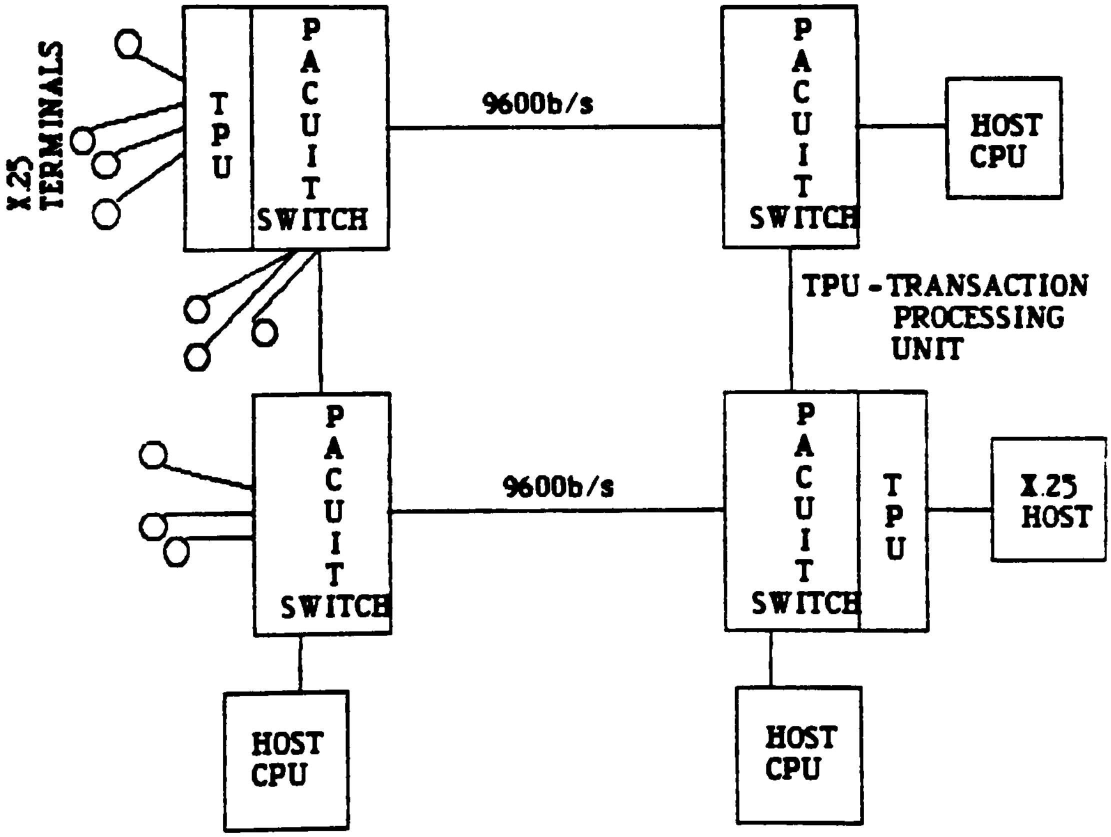



Figure 8. A PACUIT NETWORK IMPLEMENTATION

### 3.2.4.2.1. Design and Operation

The PACUIT equipment multiplexes onto a single digital line asynchronous traffic (from start-stop terminals), synchronous traffic (from buffered terminals), and PACUITS. PACUITS are groups of bits or characters going between different user endpoints located at the same switches. A train of these PACUIT blocks travels from a source switch to a destination switch, through multiple intermediate switches, carrying the data of all users who are using the route. PACUITS are not broken down at intermediate switches, and data can be neither added nor dropped. Also, there is no intermediate error checking or acknowledgment. All error control is done on an end-to-end basis with cyclic redundancy checking at the source and destination switches.

In the PACUIT system, the pacuits leave at regular intervals for a given destination. Each pacuit can carry bits from more than one user at the source node, and for more than one user at the destination node. All links between network switches/nodes are full-duplex. All intermediate switches/nodes operate in a purely time-division circuit-switch mode for through traffic in order to minimize the end-to-end delays.

While the circuit-switched nature lowers the buffer storage in intermediate nodes, the smallness of the

pacuits lowers the buffer storage in the source and destination nodes. The pacuit system uses very small buffers because the pacuits leave at frequent intervals.

If a terminal user does not enter any data between one pacuit departure and the next to that terminal's destination, almost no space is wasted for that terminal in the pacuit. The pacuit is serving multiple users, and it is probable that at least one of them will have something to send. When a user machine disconnects from the node, it is completely deallocated from the pacuit channel.

The main advantages of the PACUIT system are: (1) improvement of traffic controllability, since each source/destination pair is individually flow controlled. (2) lower switch processing load, since traffic is circuit switched through the network rather than reassembling each packet at each intermediate node. The only disadvantage is that of no dynamic bandwidth sharing between packet and circuit traffic.

# CHAPTER 4

# ROUTING TECHNIQUES

## 4. Routing Protocols and Flow control

### 4.1. Routing Protocols

In an Integrated network, the objective of the routing protocol is to find feasible, minimum blocking paths for circuit-switched requests, and minimum delay paths for packet transmissions. These objectives may be achieved with separate routing algorithms, but they might lead to inefficiencies due to high line and processor overhead and lack of proper coordination. Therefore, the following routing schemes have been proposed for Integrated networks.

#### 4.1.1. Distributed algorithm

This algorithm is based on distributed computations. That is, it determines the set of paths to each destination at each node/switch based on exchange of information between nodes. The paths are ranked by increasing values of residual bandwidth and increasing delay. Packets are always routed on minimum delay paths, while circuits are routed on paths with sufficient residual bandwidth to satisfy bandwidth requirements.

Two such algorithms have been proposed for the Integrated networks. Algorithm I [MAS78] and Algorithm II [SHE82].

#### 4.1.1.1. Algorithm I.

In this Distributed Routing algorithm[MAS78], each node has only partial network status information obtained by communicating with its immediate neighbors. Circuit switched routing is used to route a circuit of sufficient bandwidth between a given source-destination pair with a minimal "disturbance" to the network. The "disturbance" is the nodal storage and processing overhead required to maintain the circuit. Each node stores a set of tables with the information of the shortest paths of residual bandwidth to all destinations. Paths are arranged in order of increasing bandwidth.

##### 4.1.1.1.1. Establishing Circuit Routing

To establish a circuit of required bandwidth, the table is first searched at the source node to find the shortest path having sufficient bandwidth to the next intermediate node. If a feasible path is not found, the request is blocked. Otherwise, the request is forwarded to a neighbor node. The source node, while forwarding the request, sets aside the required bandwidth for the ongoing connection and decreases the bandwidth pool on the for-

warded trunk by the required amount. At the same time or periodically, the node transmits its tables to the neighboring nodes. This process is repeated at all the intermediate nodes until the desired destination is reached. This way, the physical connection is established between source and destination.

The probability of the future blocking is minimized by choosing the shortest route based on the total capacity, bandwidth and minimum hop route. If a future blocking occurs the request will wait until the required bandwidth path (shortest) is available.

#### 4.1.1.1.2. Establishing packet routing

To establish packet routing, a scheme of piggybacking on the above described circuit routing is used. This scheme consists of mixing the isolated routing scheme and the quasi-static distributed scheme. In the isolated scheme, there is no exchange of routing information between neighbors. In the distributed quasi-static scheme, there is a slow exchange of information between neighbors.

The quasi-static scheme is implemented using the circuit-switch routing tables. These routing tables specify a set of preferential routes to each destination for each node. A priority ordering and a frequency of

usage is associated with each route. The isolated scheme corrects the queue information. More specifically, the preferred route is used until a threshold is reached. Then the second preferred path is used, and so on.

#### 4.1.1.2. Algorithm II

Another proposed algorithm [SHE82] is a scheme for constructing the routing tables containing lists of non-dominated best paths to all other nodes, at each node. These routing tables are used in a distributed decision process, in the same manner as routing decisions are made in Algorithm I.

##### 4.1.1.2.1. Format of Routing Tables

There are three values associated with each path: delay, bandwidth and reliability. The routing table contains a small list of the possible best paths associated with the above three values, along with a fourth value associated with the next node on the path. These tables are stored at each node.

##### 4.1.1.2.2. Construction of Routing Tables

For constructing routing tables, there are several possible algorithms: In one algorithm, a node uses its own information on links to its neighbors and routing tables sent from those neighbors to construct its own tables. In

this algorithm, a node is dependent on other nodes to construct the table.

In another algorithm, each node maintains a database on the network topology, recording for each link the current measured values of delay, available bandwidth and reliability. It distributes to every other node the updated information containing the measured values on one or more links connected to that node. Then each node operates independently to create its routing table.

### 4.1.2. Centralized Algorithm

This algorithm is based on Centralized Scheduling [LIM84] in a Local Area Network (LAN). That is, scheduling of the routes for Packet and Circuit types of traffic is controlled centrally. The individual nodes control their own access to the routes. This algorithm utilizes the process of service cycle. For one cycle it will service packet type traffic; the next cycle, it will service circuit type traffic and then next cycle packet and so on. Circuit-type traffic has a high priority; that is, if during the servicing of packet type traffic a system detects circuit-type traffic, then the system will stop servicing the packet's data traffic and will start servicing circuit type traffic. During intervals when the system is servicing circuit type traffic packet's data are buffered so that they can be serviced at the end of circuit type

traffic cycle.

## 4.2. Flow Control

Flow control is implemented by procedures which separately control packet and circuit-switched traffic, respectively. These procedures utilize the same set of tables as those used for routing control.

The flow of the packets into the network is regulated on the basis of the value of maximum available bandwidth. Incoming packets directed to a given destination are accepted as long as the maximum available bandwidth to such a destination is above a given threshold; otherwise, they are rejected and must be resubmitted by the user after timeout. In a network, packets may be further flow controlled by conventional flow control strategies. For example, a limit may be set on the number of packets that can be maintained on each internal queue, or sets of buffers can be reserved for packets that have covered a given number of hops.

Flow control on circuit-switched traffic is applied during the call setup phase. At the entry node, the bandwidth request is compared with the available bandwidth. If the request exceeds the availability, the call is refused. Alternatively, if the terminal clock is controllable from the network, the network may "slow down"

the terminal by intermittently starting and stopping its clock, so that the effective terminal rate is reduced to a value acceptable by the network.

# CHAPTER 5

# INTEGRATED ARCHITECTURE

## 5. Integrated Architecture

Since 1978, different integrated switching and network architecture schemes have been proposed [ROM82] [LLM84] [BUN84] [TTY84] [FKD84] [EHG85]. These schemes range from a combination of SENET and TASI like approach to a combination of best features of PBX and LAN.

### 5.1. Integrated Switching Architecture

Integrated Switch Architecture schemes are (1) Flexible-Hybrid scheme, (2) Cut-through scheme, (3) Quasi Cut-through scheme, (4) Synchronous Composite Packet switching scheme, (5) Digital switching scheme, and (6) Burst switching scheme.

#### 5.1.1. Flexible-Hybrid Scheme

The Flexible-Hybrid scheme is based on a combination of the SENET and TASI like approach. In this scheme, the voice is transmitted as packet-switched data without error control via fixed path routing. Interactive and narrative/record data are transmitted as error-protected packet-switched data via independent, or adaptive routing. Bulk, facsimile, and burst data are transmitted as

circuit-switched data in one direction with error control provided via packets or small circuit-switched transmission capacity allocated in the reverse direction. During speech silences, packet voice transmission ceases, and unused capacity is available for other users.

The advantage of this scheme is an increase in transmission efficiency, since the packet switching of voice provides a TASI-like advantage. The disadvantage of this scheme is that for bulk data, the simplex transmission of data requires additional control maps (in addition to the packet data) at either end of the communication link. Further, it is necessary to provide error control in the reverse direction via packets in the packet-switched data region.

### 5.1.2. Cut-Through Scheme

In this architecture, all the trunks in the network are subdivided into subchannels of equal capacity. Each message establishes its own circuit -switched path while traveling through the network. That is, the header of the packet sets up a circuit-switched connection at each intermediate node, so that the remainder of the packet is transferred through the node. If the node has free subchannels, the message is not reassembled. When this happens, the message is said to have made a "cut". If blocking occurs because no free subchannels are found at some

intermediate node, the message is completely reassembled at that intermediate node and completes its journey through the network in a store-and-forward using a packet protocol. This implementation integrates Packet and Circuit switching, not only within the same network, but also within the same message.

Cut-through architecture offers potential advantages with respect to both packet and circuit switching. 1) It eliminates the reassembly and buffering delay of packet switching at light load. 2) It eliminates the call setup delay of conventional circuit switching. These advantages are confirmed by numerical results obtained in a wide range of traffic and network environments. [KER79].

### 5.1.3. Quasi Cut-Through Scheme

Quasi cut-through[LLM84] is a combination of "cut-through" switching and packet switching. It also incorporates an aspect called "partial cut" and a threshold-based rule for the segmentation of messages into packets.

In this scheme, a message is not completely reassembled at an intermediate node if it finds only one message in the node. Thus, the message can start its onward journey as soon as the service of the preceding message is finished. When this happens, a message is said to have made a "partial cut". The only difference between perform-

ing cuts and not performing cuts is that the processor does not hold transmission of a packet until all the bits of that packet have arrived, but starts transmission as soon as a packet's header has been processed. This does not need any extra processor time.

In addition to partial cuts, if the arrived message's length is less than or equal to the threshold value, the message is not segmented into packets. It is routed intact after appending a header for identification and error control. On the other hand, if the arrived message's length is greater than the threshold value, then it is segmented into smaller packets(each with its own header) before routing through the network. The threshold value should be greater than or equal to the maximum allowable packet length. The advantages of this threshold-based segmentation are as follows: (1) The reassembly delay per message decreases because the average number of packets are fewer in the threshold-based segmentation than in the conventional segmentation. (2) The overhead decreases dramatically for the same reason as for reassembly delay. (3) Deadlocks are less likely to occur because there is a smaller number of packets per message. The disadvantages are: (1) The buffer efficiency decreases a little because of few packets of larger length. (2) Because of a few longer packets, error performance degrades a little.

These advantages and disadvantages have been confirmed by the numerical results [IIM84] obtained in simulation studies.

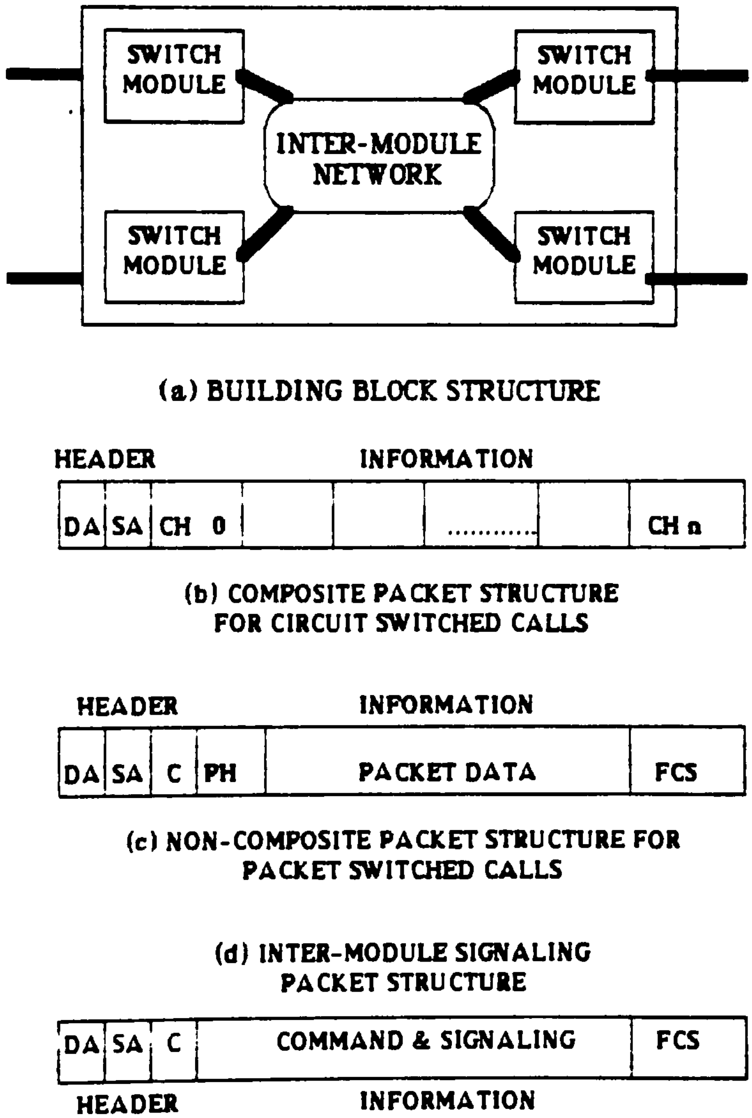
### 5.1.4. Synchronous Composite Packet Switching Scheme

The Synchronous Composite Packet Switching scheme (SCPS) [TTY84] assembles the number of circuit switched channels, which are communicating simultaneously, into quasi-packets called a composite packet and processes them similarly to packet switched channels. The composite packets are assembled, transmitted, and disassembled in a timely and synchronous manner in the system to maintain complete time transparency for circuit switched channels.

The SCPS system utilizes a building block structure as shown in Figure 9 (a). The switching is established by transmitting circuit and packet switched channel messages between switch modules via the common inter-module network. Inter-module command messages and signals for system operation and control are also transmitted between the modules. A similar packet structure as shown in Figures 9(b), (c), and (d), is used for transmitting these three kinds of messages. The non-composite and signaling packets are not transmitted synchronously between switch modules.

The SCPS system has the following advantages: It intergrates circuit and packet switching functions on a

single switching system, it switches an extremely wide range of speeds and traffic characteristics in a unified manner, and it maintains complete time transparency and a short absolute delay time for circuit switched calls.




DA : DESTINATION MODULE ADDRESS
SA : SOURCE MODULE ADDRESS
FCS: FRAME CHECK SEQUENCE
PH : PACKET HEADER
C : CONTROL FIELD
CHn: CHANNELS

Figure 9. SCPS SWITCHING SYSTEM

### 5.1.5. Digital Switching Scheme

This architecture consists of a digital circuit-switched network in which the portion of trunk bandwidth not used by circuit traffic is allocated to packet traffic. In this implementation circuit characters and data characters are processed differently at each intermediate node, as shown in Figure 10. The incoming frame is inspected first by the Switch Processor Module(SPM). Incoming circuit slots are mapped directly by the SPM into appropriate slots of outgoing trunks, as specified by the circuit routing map established at call setup time.

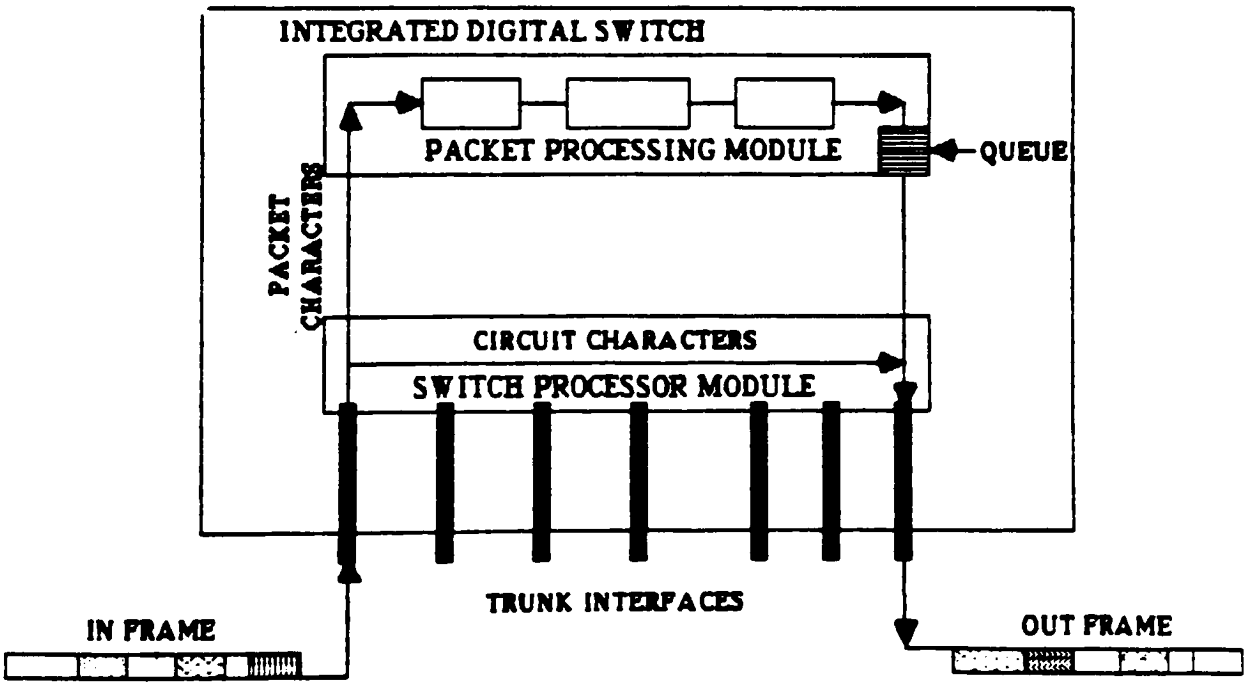



Figure 10. INTEGRATED DIGITAL SWITCH FUNCTIONS

Incoming packets are extracted by the SPM and delivered to a Packet Processing Module (PPM), where packet reassembly, link protocol, flow control, routing, and transfer to an outgoing packet queue are performed.

The SPM is an extremely fast and unsophisticated processor whose only task is transferring characters from incoming slots to outgoing slots. The PPM processes incoming data at a much lower rate and performs several sophisticated operations on each processed packet. The main drawback to this implementation is that of degradation of throughput efficiency.

### 5.1.6. Burst Switching Scheme

Burst switching (also known as wideband packet switching) is based on a technique in which routing and call information is carried along with the message rather than transferred to functions in the switch. This type of switching technique is referred as Burst switching. The general arrangement of a burst and a switch is shown in Figure 11. A burst consists of a bit stream of variable length divided into a header, a burst of voice or data, and a terminating flag. Voice bursts are variable in length and contain entire talkspurts. Data bursts are of packet length and are packed between the voice signals.

A user is connected to the burst switching network through a channel switch (CS) which provides access to a digital channel in a multiplexed digital stream. Switching is based on the burst header information, and the logical connection stays for the entire burst. In order to give priority to voice bursts, the CS buffers data bursts (delays them). The mixed voice and data burst streams are passed through several channel switches to a link switch (LS) where the bursts are prioritized and prepared for insertion in a ring of hub switches (HS). Each HS passes burst to one neighbor.

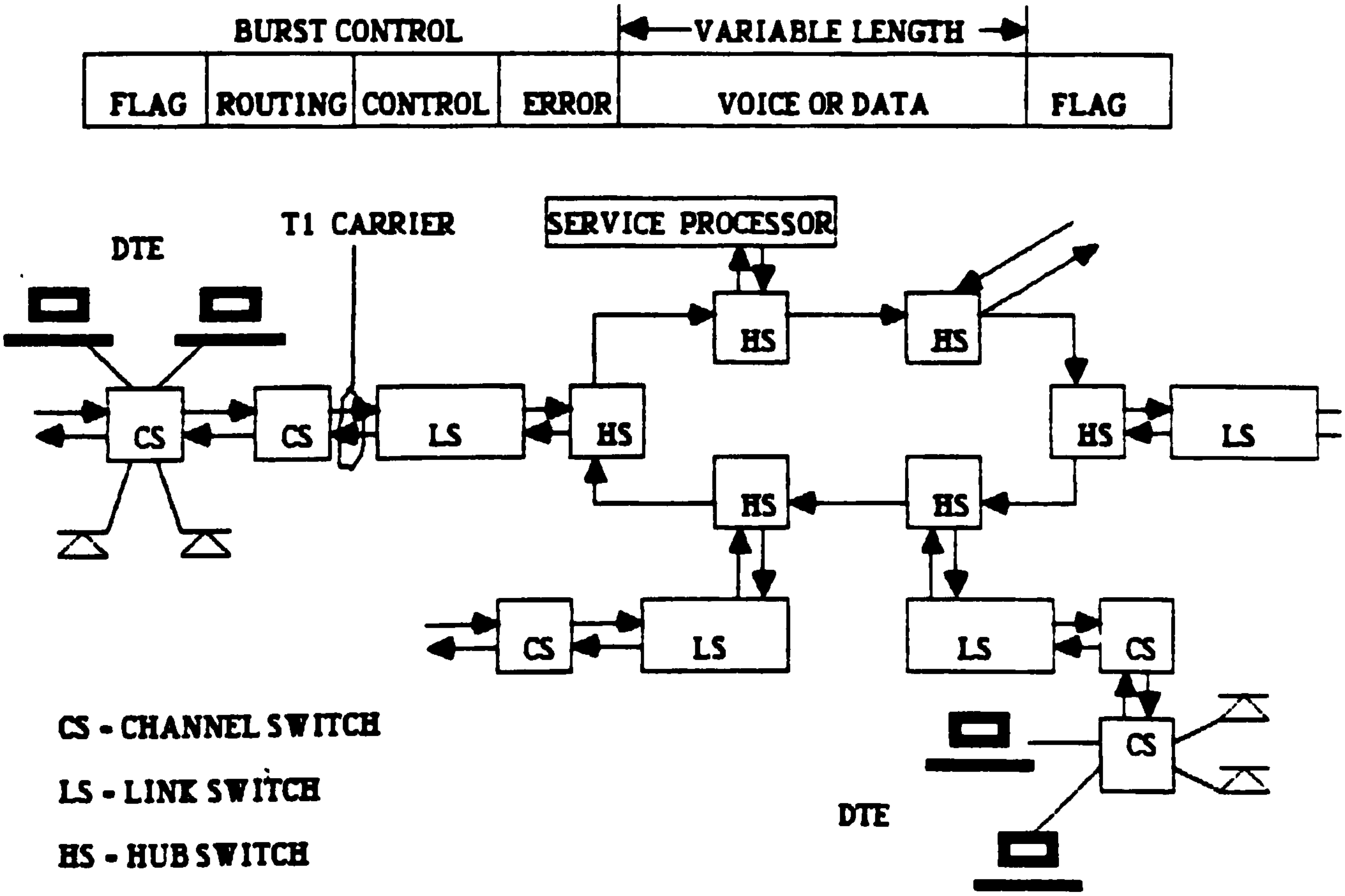



Figure 11. PRINCIPLE OF BURST SWITCH

A burst is circulated until it arrives at the HS, which connects, through a LS, the CS serving its destination. The link-and-ring structure is sized to be non-blocking on the basis of the capacities of the transmission facilities and the CSs. No common control or store-and-forward functions are needed in the intermediate switches to control path establishment and information transfer.

If the called party is connected to the same CS, or a CS in the same link, bursts are routed directly by the CS. If the called party is on another link, a routing request message is sent to the service processor, which returns the information for insertion in the header for call setup.

The principal difference between "burst switching" and other techniques is that the switching decisions necessary for the insertion of pieces of an individual user's data stream are made at the higher levels of the multiplexing/switching hierarchy based on the information obtained at lower levels. Channel utilization efficiency in excess of 99% has been mentioned. The only problem with this is that of contention -- more sources requesting services than there are channels available to serve them.

## 5.2. Integrated Network Architecture

Integrated Network Architecture scheme [EHD85] is based on the combination of the best features of the Public Branch Exchange (PBX) with those of the Local Area Network (LAN) in the same network. Table 2 shows the best features of PBX and LAN.

Table 2. Best Features

| PBX | LAN |
|---|---|
| Circuit Switched | Packet Switched |
| Star Topology | Bus, Ring or Star Topology |
| Central Processing | Distributed Processing |
| Low Data Rate 64 kb/s | High Data Rate > 1 Mb/s |

### 5.2.1. Network Structure

The distributed system (a small LAN in a ring configuration) connects the multi-functional terminals, providing circuit switching and packet switching on demand. Several end systems are connected to one ring access station, which operates as a cluster controller. This ring structured network is connected through circuit and packet switched links to the packet and circuit switched type of PBX. The multi-functional terminals are also connected to the PBX in the star-configuration. The layout of the structure is shown in Figure 12.

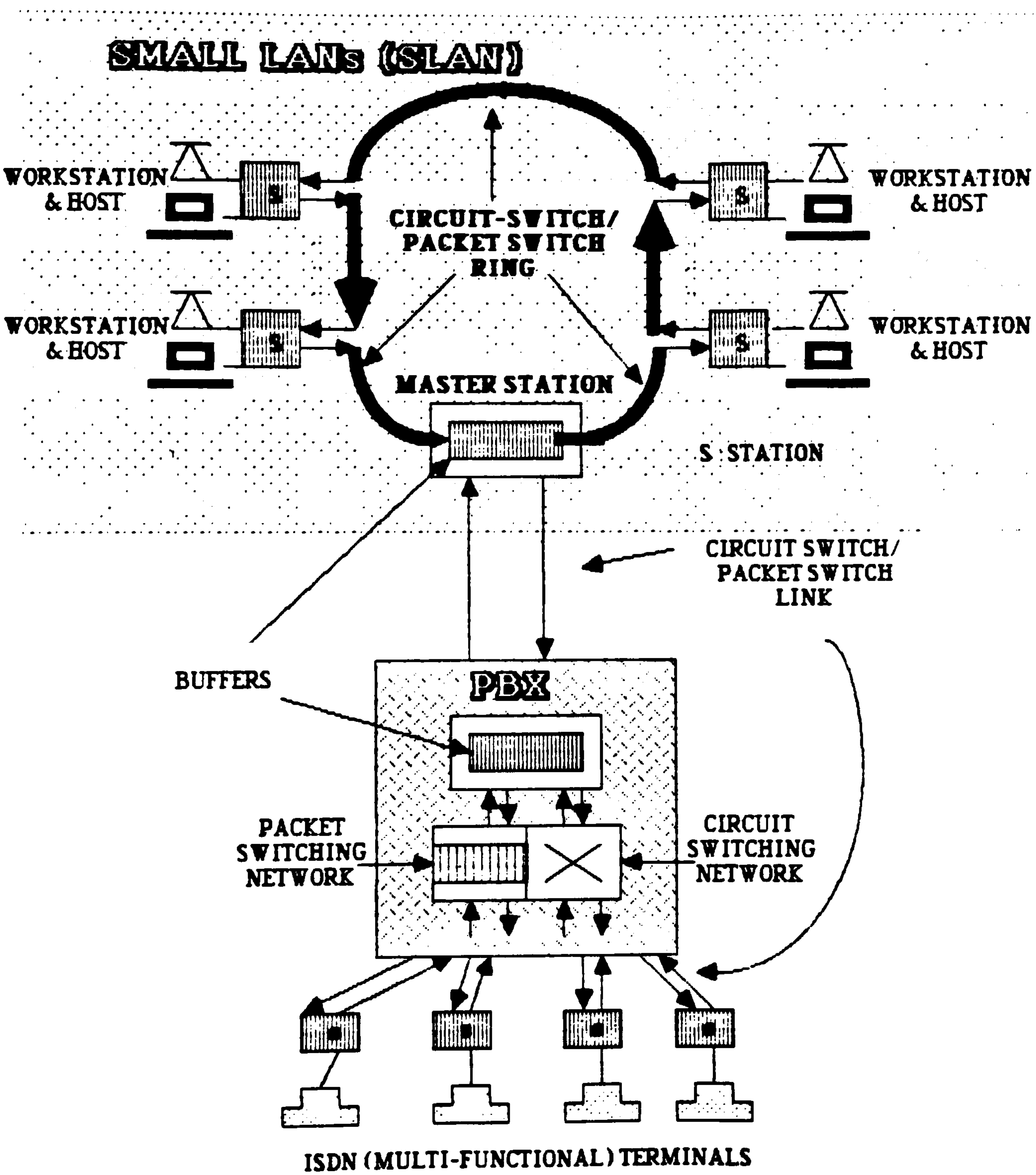



Figure 12. INTEGRATED NETWORK STRUCTURE

This network structure provides synchronous transmission for circuit switched voice and circuit switched data(mass data transfer) with a variable and adaptable bandwidth. It also provides asynchronous transmission of data packets similar to the LAN's with a high throughput rate.

The following advantages have been mentioned.

- Limitation of distributed functions to very small areas (SLAN - small LAN)
- Handling of mass traffic, operational and maintenance functions by centralized nodes
- Maintaining of the adequate service-specific switching principles(Circuit Switch or Packet Switch)
- Matching of specific grade of service criteria as throughput and delay for Packet Switched services and blocking for Circuit Switched services
- Imbedding of the new structure into the existing infrastructure

### 5.2.2. Integration on Ring Structure

The basis for the circuit switched/packet switched integrated ring and the circuit switched/packet switched link between the integrated systems is a synchronous pulse frame with fixed length. This frame is partitioned into equal sized time slots, similar to the PCM(pulse code

modulation) frame. One time slot is able to carry one circuit switched channel with 64 kbps transmission rate, where the same time slot can be used for both transmission directions providing full-duplex connectivity. Allocation of time slots to new circuit switched calls is done by means of signaling procedure at call establishment. To achieve short delays and independence from the packet switched traffic, a separate signaling channel consisting of one or more time slots is utilized. The Master Station (MS) in the ring generates the pulse frame and buffers the frame, compensating for different propagation delays. This station also provides gateway-functions, gaining access from the ring to the other part of the network.

Improvement of ring utilization can be achieved using the moving boundary method between the circuit switched and packet switched part. That is, packet switched data is carried in the second part of the pulse frame, beginning immediately after the circuit switched connection with the largest time slot number.

Ring utilization can also be improved using a slotted frame and interleaving circuit switched and packet switched data schemes. In this scheme, a circuit switched call may occupy any empty time slot within the whole pulse frame as long as it does not exceed the maximum allowable circuit switched channels. All other "idle" time slots

can be used for packet switched data.

Implementation problems of the moving boundary and slotted/interleaved scheme have been mentioned [EHG85], because of the uncontrolled access to the packet switched part of the pulse frame. The following scheme has been mentioned to rectify this problem. Messages at the ring-station are partitioned into equal-sized and individually addressed minipackets (MP). Each minipacket fits into one time slot. Two bits of every time slot are marked, indicating whether this time slot is available for packet switched data and whether this is empty. Each ring-station inserts these minipackets into unused slots, and also detects its own address in the header of a minipacket. The ring-station copies the contents into its buffer and sets the time slot empty or inserts its minipacket to be sent. This way empty slots are used on demand by the sending station and the receiving station is responsible for emptying the slot. The only drawback of this method is that of additional overhead for addressing of minipackets.

# CHAPTER 6

# RECENT DEVELOPMENTS

## 6. Recent Developments

More recently, there has been intense development activity in digital hybrid or integrated networks which combine elements of circuit and packet switching, like Voice, Video, Facsimile, and data file transfer, within the same network. An example is the Integrated Service Digital Network (ISDN). All this is feasible because of the recent technological developments in electronic/digital switches and transmission media used in fabricating integrated networks.

### 6.1. Integrated Service Digital Network

AN Integrated Service Digital Network is a network that evolved from the telephony IDN (Integrated Digital Network) which provides end-to-end connectivity to support a wide range of services, including voice and non-voice services. Users have access by means of a limited set of standard multipurpose customer interfaces [SIE84].

```
I S D N
--- ---
 |   |__ Digital End to End Connectivity
 |
 |__ Standard Interface for all services
```

Integrated Services Digital Network, generally referred to as ISDN, is a very diverse subject. At this early stage in the development of ISDN technologies, differences in experience and background have promoted different understandings of its nature and characteristics [DEC84].

As currently defined, ISDN can be regarded as a fully digital telecommunications network providing multipurpose services to end-users in a uniform way. Transmission within the ISDN may involve circuit-switched and packet-switched networks, that are transparent to the user. Switching may be done in two modes, Voice and data, from call to call and within a call. A virtual circuit might be required for voice when a packet-switched network is encountered within ISDN.

## 6.2. Parts of ISDN

The ideal ISDN would be made up of three network elements - integrated access, adaptive links, and integrated switches. Such an ISDN would provide a standard interface for all data speeds and services (Figure 13).

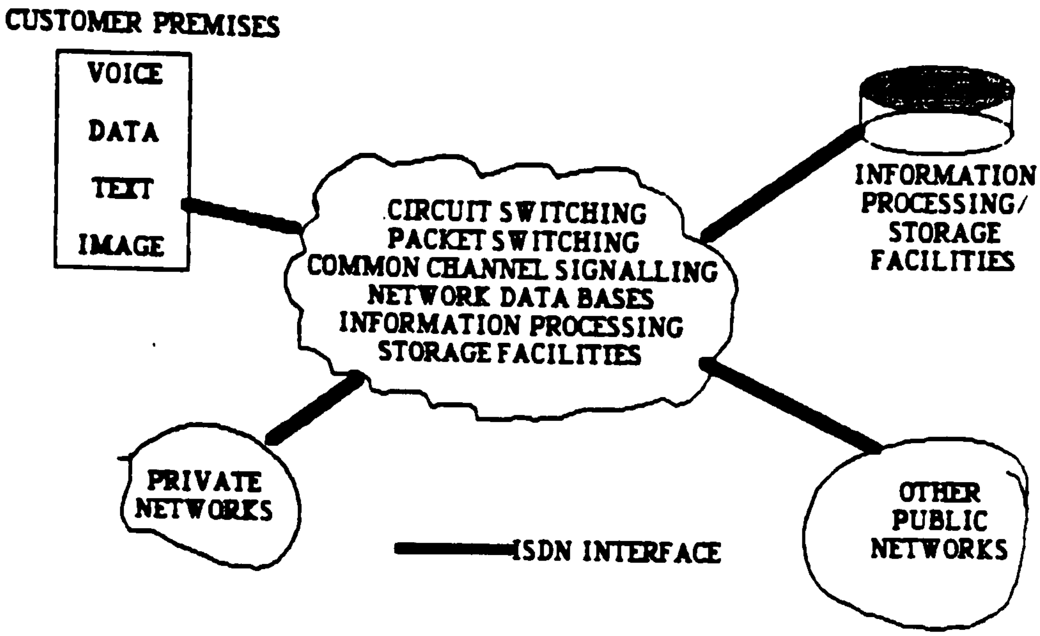



Figure 13. ISDN

### 6.2.1. Integrated access

Integrated access is the simultaneous transmission of voice and data transmission over the same loop. Simultaneous voice and data will provide the ability to gain access to independent services. For example, one can skim an electronic newspaper while talking to a friend. The addition of integrated signaling provides even more capability: the ability to gain access to interactive services. For example, one could establish both a voice and data connection to a travel agent, to allow terminal screen information to be displayed while discussing reservations with the agent.

### 6.2.2. Adaptive Link

The adaptive link, sometimes called a "digital pipe", consists of digital transmission facilities, that can handle any type of traffic with multiple bit rates and dynamically allocated channels.

### 6.2.3. Integrated Switches

An integrated switch is a single switch incorporating both circuit and packet switch capabilities. This single switch would determine the most effective method for each communication, and would simultaneously handle voice, data, and a variety of special features.

## 6.3. CCITT Recommendations

In 1984, the first CCITT recommendations for ISDN were established, laying the foundation for the development of ISDN products.

### 6.3.1. Physical Interface Structure

The Committee has selected the fundamental digital rate building block as 64 kilobits per second(kb/s). Two major physical interfaces have been defined: the basic-rate interface and the primary-rate interface. The basic-rate is intended to serve information sources or sinks of relatively small capacity, such as terminals. The primary-rate is for large-capacity sources such as PBXs.

Both have a simılar structure, that is, number of B-channels (for voice and bulk circuit-switched data) and one D-channel (for packet-switched data and signaling). The basic-rate arrangement is 2B + D. The primary-rate arrangement for North America and Japan is 23B + D, and for Europe it is 30B + D. Internetworking between the two is 23B + D. The B-channels are 64 kb/s each in both the basic and primary rates. The D channel is 16 kb/s for the basic rate and 64 kb/s for the primary rate.

The D-channel is primarily a common signaling channel for the B-channels, but may also be used as a limited-capacity communications channel.

### 6.3.2. Layer Interface Recommendations

ISDN layering will be in accordance with the Open System Interconnection (OSI) Reference Model.

Layers 1-3 will provide a description of the services characteristics provided by the network. The allocation of the network features to these layers will provide a means of organizing network features, to define the service characteristics to be provided by the network.

Layers 5-7 of the model will deal with the definition and description of communications between users (as opposed to communications between user and network). This approach allows the development of "Higher Layer Proto-

cols" which can be used to define user-to-user services in a manner which is independent of the network that may ultimately be used to convey these services.

Layer 4 will provide the bridge between the network independent protocols of the higher layers and the network services provided by Layers 1-3.

Key goals of the CCITT ISDN layer interface recommendations are user access via a limited set of multipurpose network connection types and user-network interfaces. Limiting the number of interfaces maximizes user flexibility, presents uniformity to the user of many different services, and reduces costs.

### 6.3.3. Networking Protocols

The Committee has specified the following networking protocols for the interface of the first three layers between customer premises equipment and the network. Layer 1 bit rates are mentioned above. For Layer 2 (Q.921) the local access protocol for the D-channel (LAPD) (since networking protocols are to be used on the D-channel for signaling and information transfer) has been mentioned. This is a link-layer protocol similar to the protocol defined for packet data in X.25 (LAPB), but provides the following functions: Multiple logical links in the D-channel, Detection and recovery of transmission errors, Flow control,

Sequence numbering and control, and Transparency. Two network layer (layer 3) protocols mentioned are Q.931 for the circuit signaling on the D-channel to support the B-channel, and X.25 for the data transfer mode with the D-channel. The layer 3 signaling protocol, Q.931, provides the following functions: signaling relationship for multiple calls between the user and the network; call establishment, maintenance, and clearing; and access protocol for a packet-mode service called user-to-user signaling.

The basic concept of ISDN as shown in Figure 14 is a channelized interface between the network and customer premises equipment. A single interface is divided into a discrete number of channels in multiples of 64 kb/s, with the exception of the signaling channel within the basic-rate interface, which is 16 kb/s. There are two board types of channels -- one for information transfer, the B-channel, and one for signaling, the D-channel.

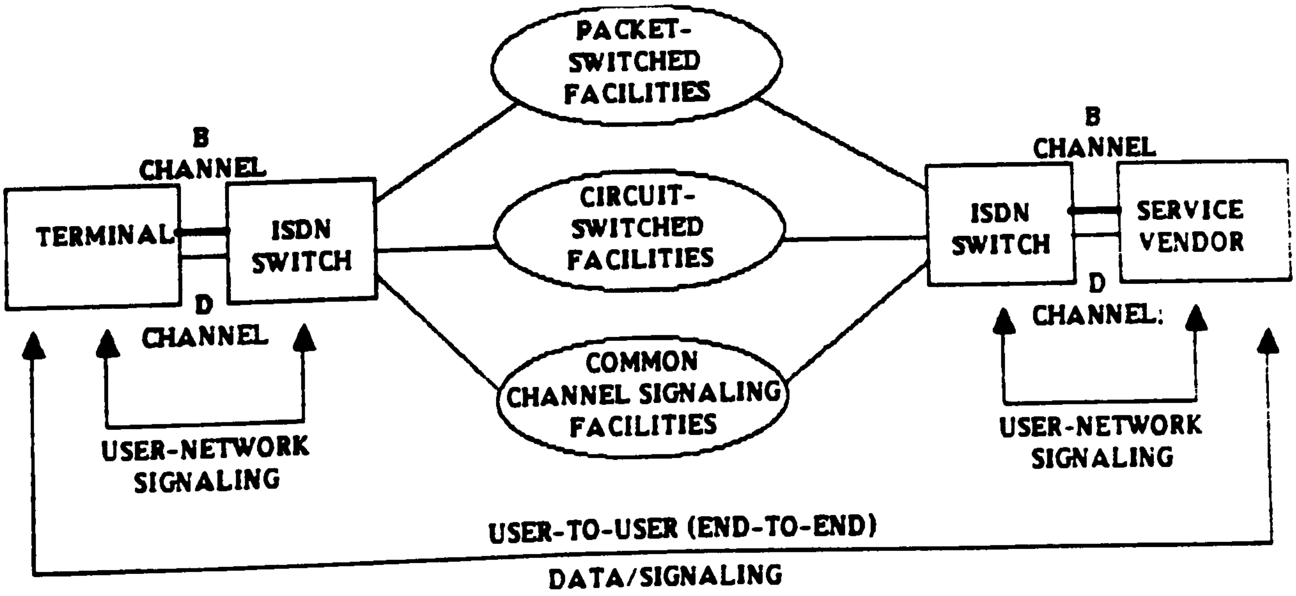



Figure 14. ISDN CHANNELIZED INTERFACE BETWEEN THE NETWORK AND CUSTOMER PREMISES EQUIPMENT

Customer premises equipment and the network communicate with each other over the D-channel in order to set up communications links through the B channels. The network uses this information in the signaling link along with its own signaling infrastructure to set up connections across the network. When the links are established, user information is sent over the ISDN access through the B-channels. The information (data/voice/video) is sent into the network in digital form. The generic network gateway architecture for the network side of the ISDN interface (Figure 15) is being proposed by AT&T.

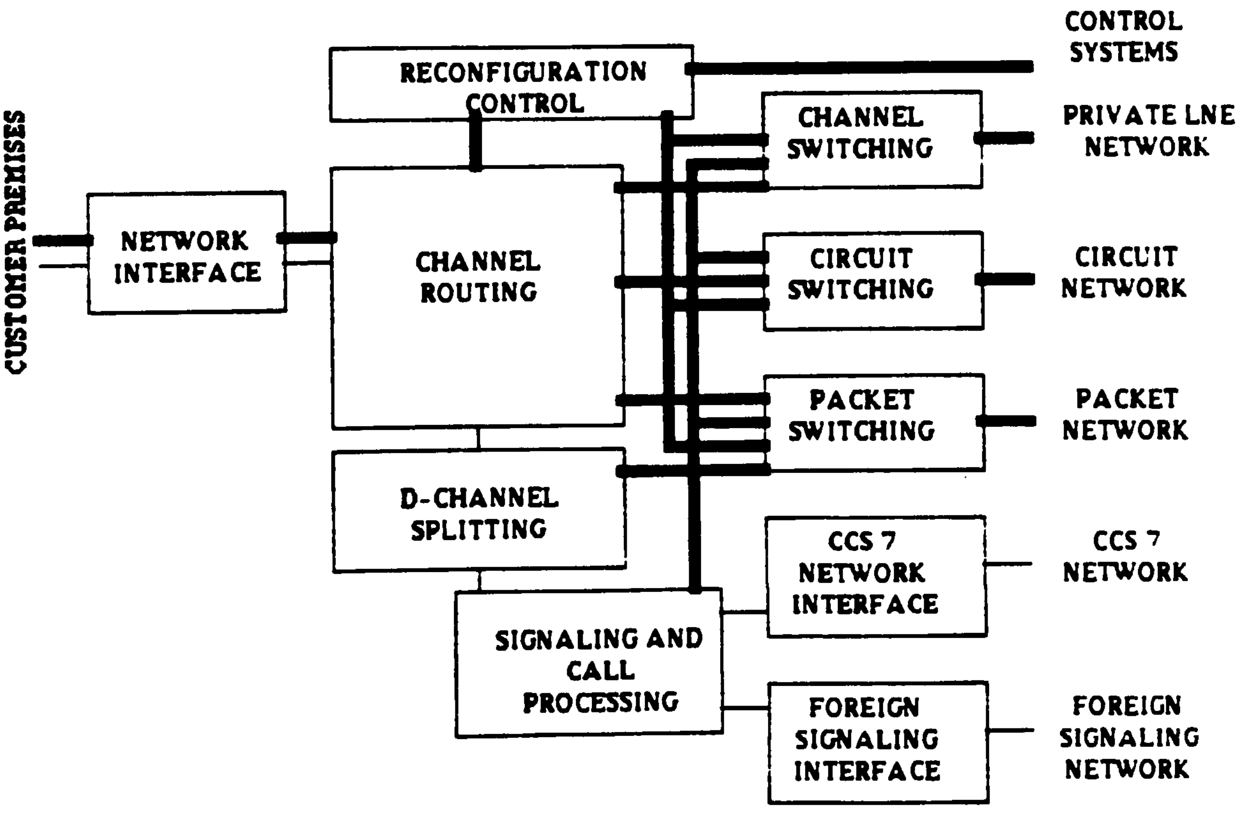



Figure 15. GENERIC NETWORK GATEWAY ARCHITECTURE FOR THE NETWORK SIDE OF THE ISDN INTERFACE

### 6.3.4. Information Transfer Protocol

The information transfer protocol spans all layers of the OSI model. The following protocol layers are mentioned for different applications.

- Circuit-switched data layer 1 provides for synchronous data transfer at 64, 384, and 1536 kb/s. The end user is free to use any protocol above layer 1.
- Packet-switched data layer 1 provides for synchronous data transfer at 16 kb/s and 64 kb/s. On the B-channel, layer 2 uses the LAPB X.25 protocol. On the D-channel, layer 2 uses the LAPD protocol. Layer 3 uses the standard X.25 layer 3 protocol.

### 6.3.5. Common Channel Signaling

The Common Channel Signaling will be used for customer-network access and interexchange signaling. This allows the control of multiple circuit-switched connections using a separate common signaling path. This separation of the signaling (call control) path from the user information path will help in supporting multiple services. It is the alternative to the signaling technique used in X.25 and X.21 access protocols where the signaling information is conveyed via the same path as the user

information.

### 6.4. User Aspects of ISDN

ISDN provides end-users with the following basic capabilities:

1. Digital end-to-end connectivity
2. Support of a wide range of services via a limited set of standard interfaces.

For the users in the business sector ISDN will provide:

1. Extended service choice beyond existing services. For example:
   - enhancement of telephone services
   - 64K bit/s leased circuit services
   - 64K bit/s circuit-switched data transmission services
   - packet-switched data transmission services
   - telematic services such as teletex and videotex
   - alarm services
2. Global connectivity between user's equipment, arising from the standard user interfaces.

## 6.5. Efforts for ISDN's

CCITT Study Group XVIII is currently active in establishing performance objectives and standards for ISDN at other higher rates in the digital hierarchy (such as 144 kb/s, 1.544 Mb/s, and 2.048 Mb/s).

Various telecommunications administrations and organizations in a number of countries have developed detailed plans to achieve compatibility with ISDN at varying degrees during the time frame 1985-1990. The pace at which progress is made in different countries to achieve the compatibility, and the range of services introduced, depends largely on the existing level of digital conversion of telephone plants and on economic considerations.

## 6.6. Digital Switching Systems -- The Basis for ISDN

The following are the most popular digital switching systems being used today in order to realize ISDN : the #5ESS developed by AT&T Technologies, the EWSD developed by Siemens Communication Systems, the DMS 100 manufactured by Northern Telecom, Inc., and the GTE-5 EAX manufactured by GTE Communication Systems Corporation. All these digital switches provide computer-controlled, time division switching or statistical multiplexed switching. A distributed architecture is provided by utilizing a host module in conjunction with many microprocessor-controlled units.

Most of their software is written in high-level language and is transportable and extendable. The major characteristics of these switches are as follows:

- They provide virtually non-blocking access between non-concentrated switch terminations.
- They provide integration of voice and digital data services into a single switch.
- They provide a direct digital interfacing with digital facility terminations.
- They provide host capability for remotely located modules.
- They provide signaling and transmission treatment by an interface unit.
- They use distributed control.
- They provide testing access to non digital (analog) facilities as an integral part of the interface units.

### 6.6.1. #5ESS

The #5ESS switching system is AT&T's first digital central office switch. The hardware architecture of the #5ESS is shown in Figure 16. It consists of three basic system modules: The Administrative Module, The Communication Module, and Switching Module. The Administrative Module (AM) performs the high-level global functions of

the switch, and is based on the AT&T 3B20D processor. The Communication Module (CM) performs the routing of control messages between processors and the transfer of calls between Switching Modules. The Switching Module (SM) performs most of the call processing functions, as well as circuit maintenance.

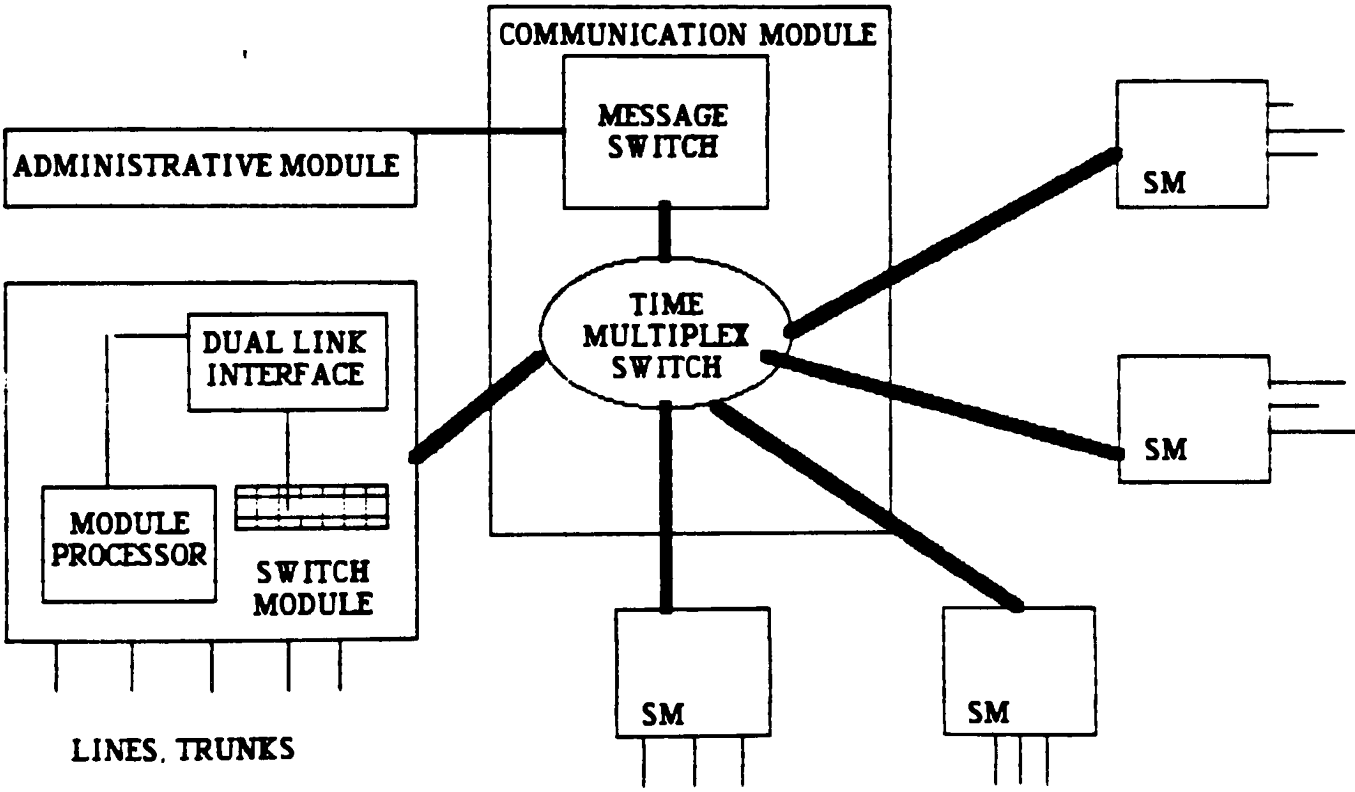



Figure 16. *5ESS SWITCH ARCHITECTURE

The switching center network employs a time-space-space-space-space-time structure.

The AM consists of the processor, disk storage, and tape backup units. It performs call processing which cannot be efficiently performed by the SM processors. The Master Control (MC) within the AM is the primary communications link between the 5ESS switch and personnel administering and maintaining the system.

The CM is the focal point for voice/data/message switching. Its major components are the Message Switch (MS) and the Time Multiplex Switch (TMS). The MS consists of three major units: the Message Interface Control Unit, the Message Switch Control Unit, and the Message Switch Peripheral Unit. The Message Interface Control Unit provides system synchronization and the interfacing for the transmission of the control message time slots between the TMS and the AM or SM processors. The Message Switch Control Unit performs message routing of all inter-processor control messages. The Message Switch Peripheral Unit provides for intelligent interfacing between the Message Switch Control Unit and the SMs for message transfer.

The TMS provides the physical paths for the digital signals transmitted between the SMs and the AM. The medium for transmission is the Network Control and Timing (NTC)

link. The TMS consists of a Time Multiplex Switch Unit (TMSU) and a Time Multiplex Control Unit(TMCU). The TMSU switches digital voice and data between the SMs and also passes control information to and from the Message Switch.

The SM provides the system interface with all external lines and trunks. It converts signals received from lines and trunks into the internal digital time division format of the switch. The SM also provides the interface to the #5ESS switch for the Remote Switching Modules (RSMs). The RSM provides the same services as a host SM, including intra-switching of local RSM calls.

The #5ESS switch software is divided into different software subsystems: real time operating system, call processing, maintenance, administration and office data. The operating system is UNIX-based. Call processing consists of three software subsystems: feature control, routing and terminal allocation, and peripheral control. Maintenance software provides for the following capabilities: human-machine interface, switch maintenance, terminal maintenance, system integrity, and program update. Administrative software provides the following administrative tasks: call record assembler, measurements, billing, network management, external data link communications package, and service evaluation. Office data is contained in a relational database under control of the database management

system. The system provides interfaces and access mechanisms between the different types of software system and data in the database. The proposed ISDN network architecture using #5ESS switch is shown in Figure 17.

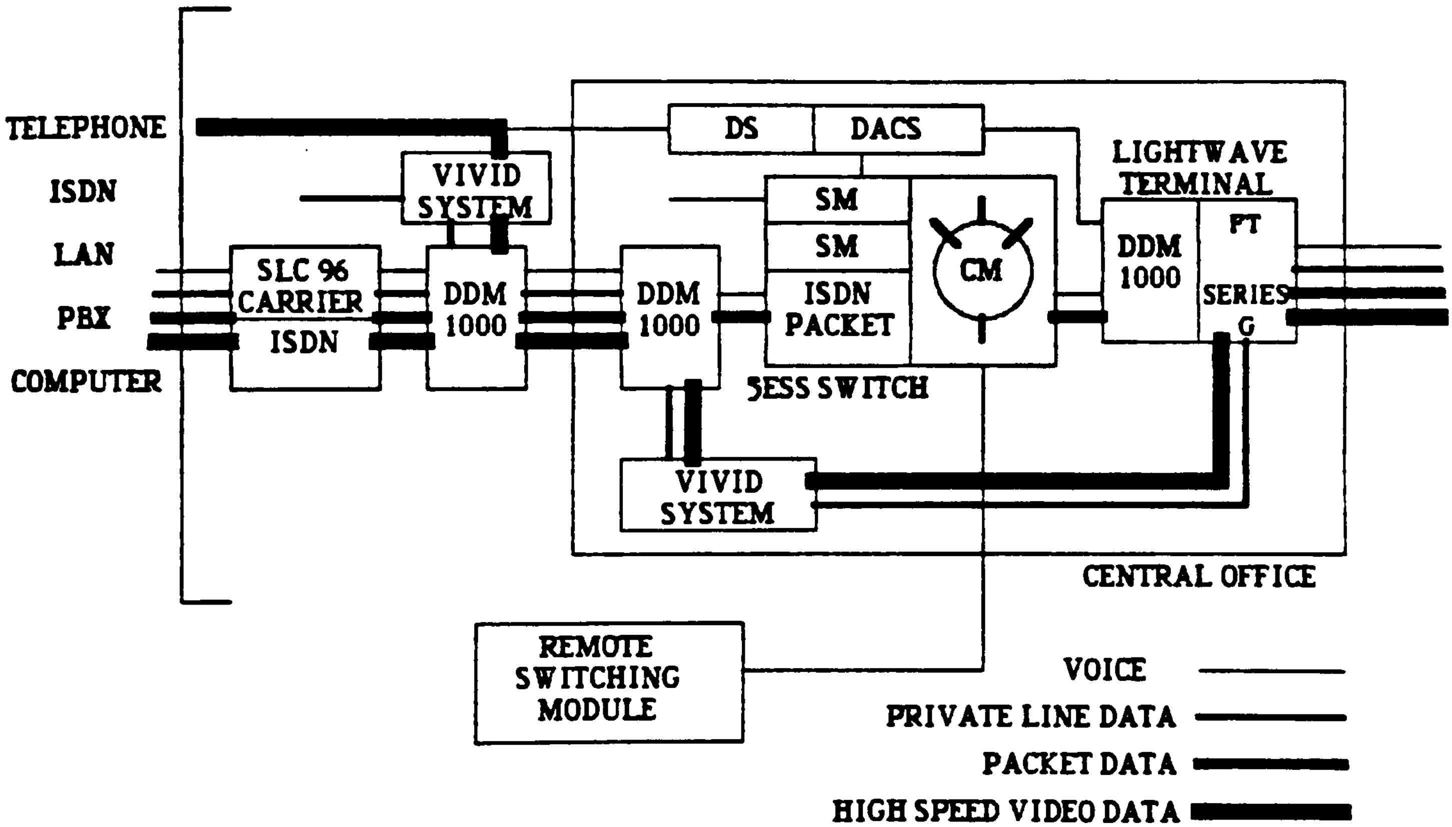
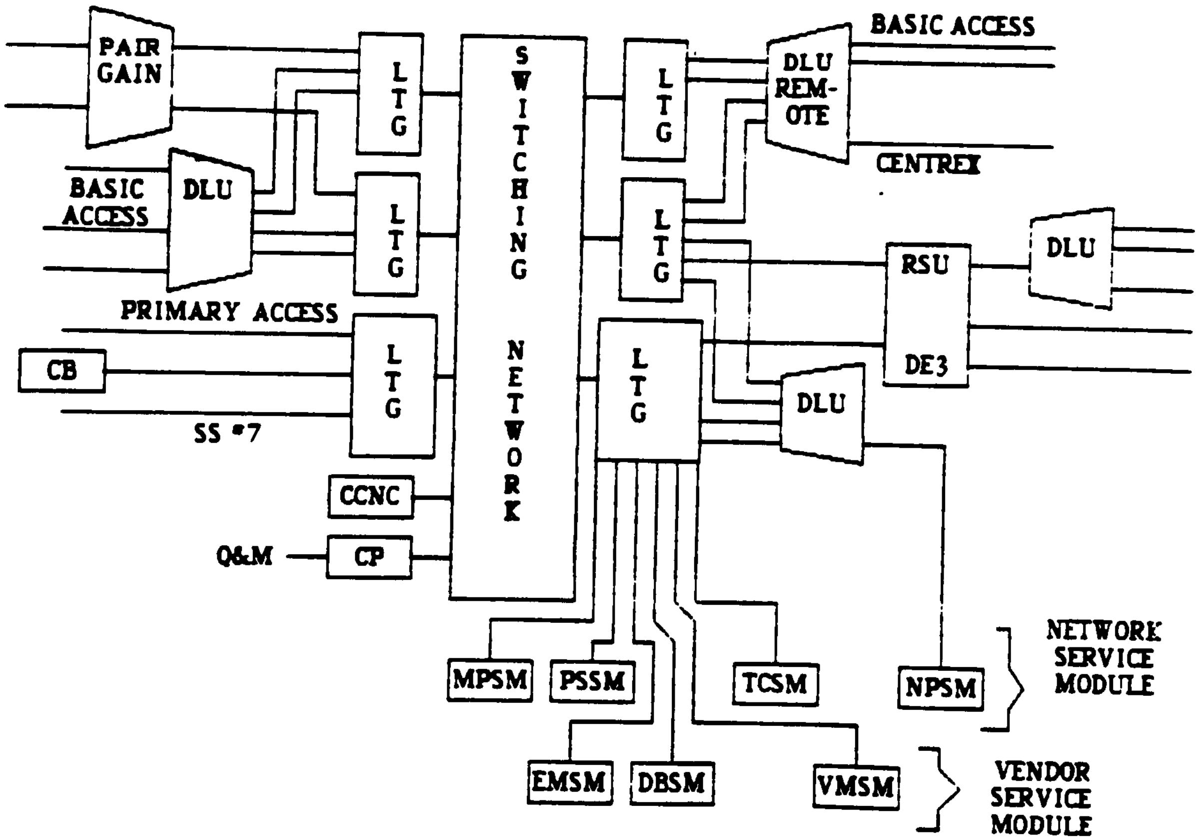



Figure 17. 1987 NETWORK ARCHITECTURE

## 6.6.2. EWSD

The EWSD switching system is Siemens's digital central office switch. The hardware architecture is shown in Figure 18. The Digital Line Unit (DLU) supports either analog carrier or integrated digital carrier. The DLU can be collocated with the EWSD, and it can be remote as well. The Line Trunk Group (LTG) provides for conventional trunking arrangement, including digital T1 carrier trunks.




| | | | |
|---|---|---|---|
| NPSM | NETWORK PAD SERVICE MODULE | RSU | REMOTE SWITCHING UNIT |
| TCSM | TELECONFERENCING SERVICE MODULE | CCNC | COMMON CHANNEL NETWORK CONTROLLER |
| PSSM | PACKET SWITCH SERVICE MODULE | DLU | DIGITAL LINE UNIT |
| MPSM | MODEM POOL SERVICE MODULE | CP | COORDINATING PROCESSOR |
| EMSM | ELECTRONIC MAIL SERVICE MODULE | LTG | LINE TRUNK GROUP |
| DBSM | DATA BASE SERVICE MODULE | | |
| VMSM | VOICE MAIL SERVICE MODULE | | |

Figure 18 EWSD SWITCH ARCHITECTURE

The Common Channel Network Controller (CCNC) provides the lower layer processing without burdening the central Coordination Processor (CP). The Remote Switching Unit (RSU) provides a flexible stand-alone switching capability with centralized Operations, Administration and Maintenance (OA&M) interface. The switching network is a time-space-time structure.

The EWSD architecture has distributed processing with the number of processors increasing approximately linearly with the number of lines. The CP performs the safeguarding, maintenance, digit analysis, and routing functions. The switching network is virtually non-blocking, and thus provides the throughput necessary for ISDN applications such as channel networks and special services.

The EWSD switch allows for simultaneous, and independent, voice or data calls to be carried on each B channel as well as packet data to be simultaneously carried on the D channel.

The EWSD handles packet data in a unique and efficient way. As shown in Figure 19, the packet data is separated from the signaling data on the D channel in the Subscriber Loop Module - Digital (SLMD). The packet data is then statistically multiplexed across the eight Basic access lines served by Peripheral Board Controller (PBC) and across lines of a DLU. Statistical multiplexing is accomplished by time slot assignment.

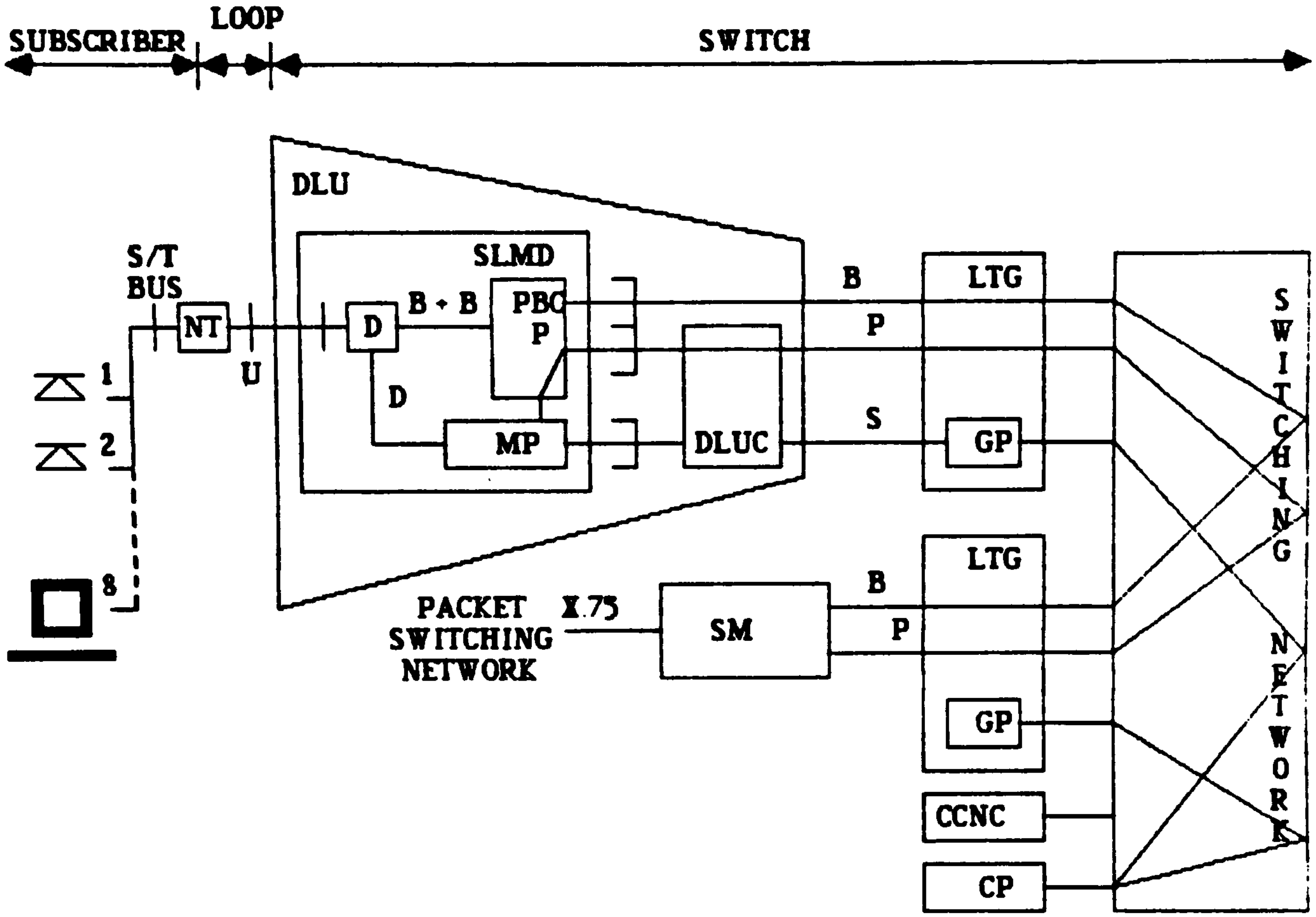



Figure 19. EWSD-ISDN PACKET CALL HANDLING

The proposed ISDN network architecture using EWSD switch is shown in Figure 20.

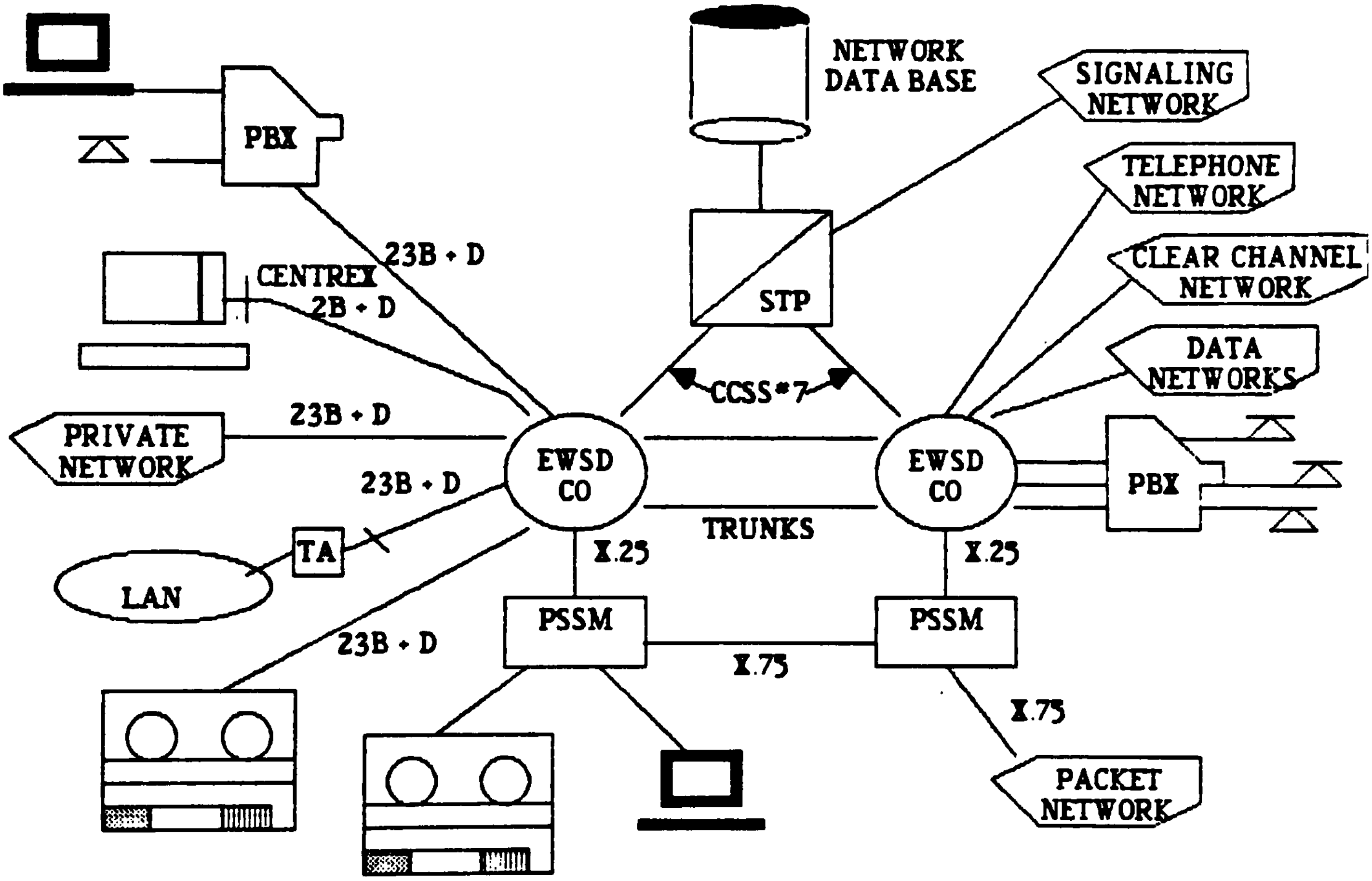



Figure 20. ESWD ISDN ARCHITECTURE

## 6.6.3. DMS 100

The DMS 100 switching system is Northern Telecom's' digital central Office switch. The system architecture is shown in Figure 21. A distributed processing architecture is used with call processing distributed between the Central Control (CC) and Peripheral Modules (PMs).

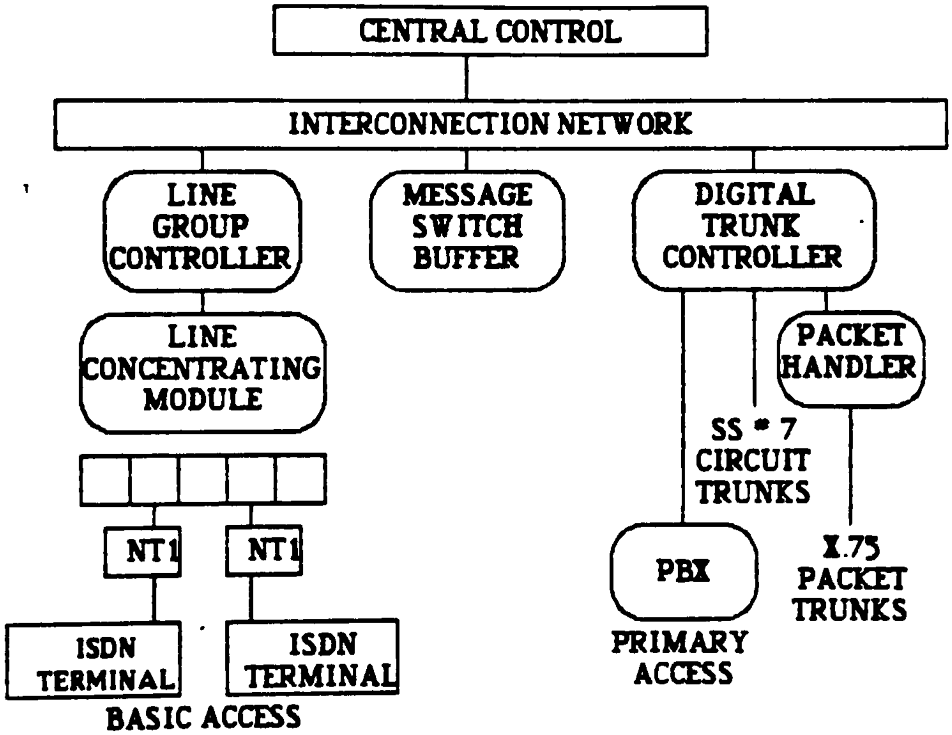



Figure 21. DMS-100 ISDN ARCHITECTURE

For example, logical analysis is done by the CC while digit collection and call supervision are handled by the PMs (such as the line group controller, the message switch buffer, and the digital trunk controller).

Digital Trunk Controller (DTCs) are the PMs that terminate trunks on DMS-100 systems. Lines are terminated on a Line Concentrating Module (LCM), which is connected to a Line Group Controller (LGC). The intelligence and service circuits for line call processing are provided by the LGC. The network is time-division multiplexed, blocking/nonblocking, and time-space switched.

### 6.6.4. GTD-5 EAX

The GTD-5 EAX switching system is GTE's digital central office. The system architecture is shown in Figure 22. It consists of peripheral, network, and control equipment in the Base Unit (BU), and remote units with individual microprocessors. The BU consists of a range of Facility Interface Units (FIUs) which present standard interfaces to the switching network and control equipment. The network is a time-division multiplexed, essentially nonblocking, time-space-time network, switching 12 parallel bits in each of the 64-kb/s channels. The Remote Line Unit (RLU) consists of a Remote Switch and Control Unit (RCU) module interconnecting analog line and/or digital trunk FIUs to digital trunks connecting to the BU. RCU contains the same network elements as the RSU.

The control architecture is modular, distributed, and contains multiprocessors. It consists of a central

control and three levels of peripheral processors. A peripheral processor unit is allocated to specific hardware and its program performs functions for the hardware it controls. The central control processor units perform the logical analysis and sequencing of calls. Each contains an administrative processor unit used for the administrative and maintenance functions. Central control processor units and base unit peripheral processor units communicate by way of a duplicated message distribution circuit. This provides a rotational priority and conflict resolver circuit to arbitrate contention between the processor units.

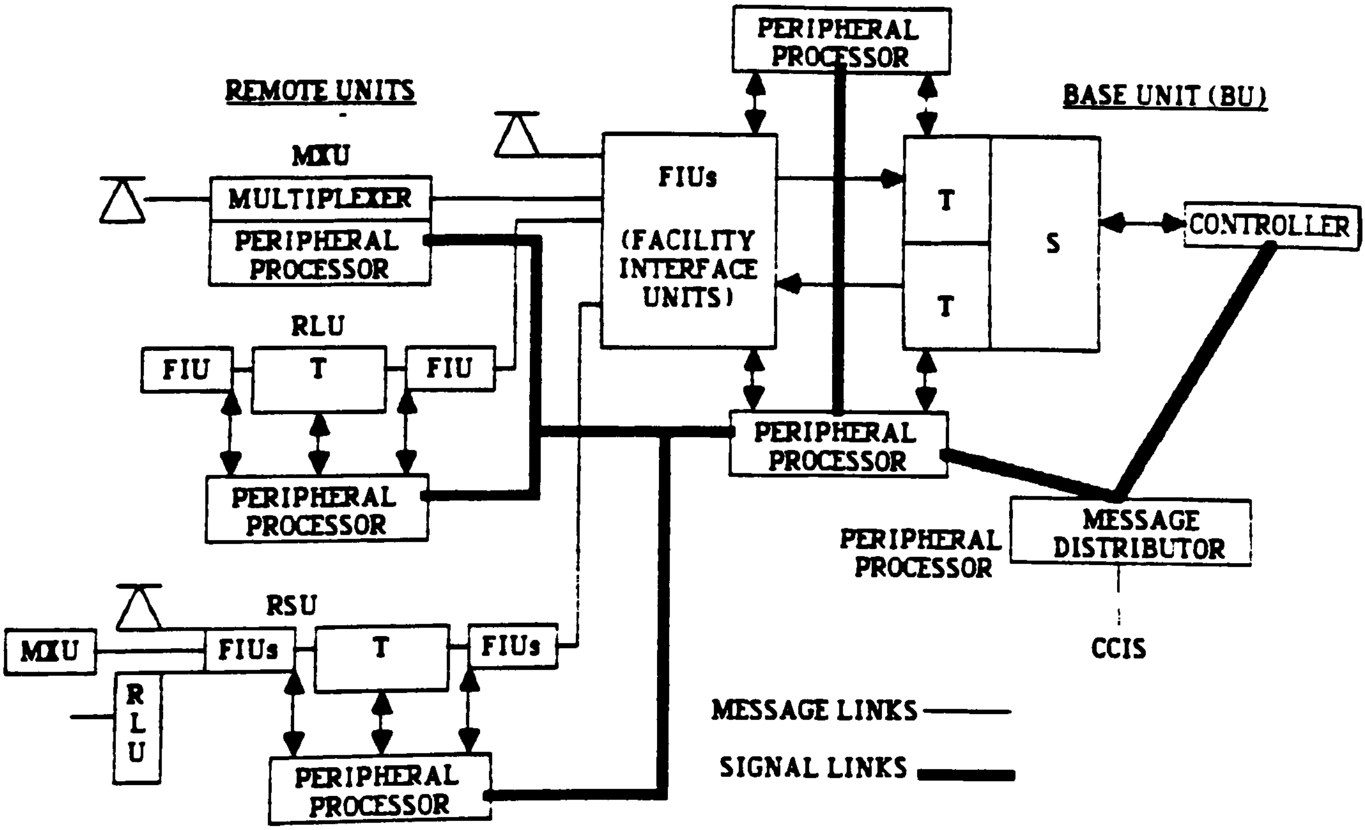



Figure 22. BLOCK DIAGRAM OF GTD-5EAX SYSTEM

CHAPTER 7

# PERFORMANCE ANALYSIS AND COMPARISON

## 7. Performance

### 7.1. Performance Analysis

The performance of the integrated circuit and data switched network is generally specified by the circuit traffic (voice call) blocking probability and the mean packet waiting time. The state of the integrated circuit and data switched network is given by the amount of circuit traffic (voice calls) and the number of data packets in the system. In order to evaluate these parameters, the following analytical techniques (mathematical approaches) have been proposed.

#### 7.1.1. For Circuit Type Traffic

The performance of the circuit type traffic can be easily obtained, either exactly or approximately, by the Erlang B formulation, depending on the modeling assumptions. The Erlang B Formulation is the probability of blocking versus the amount of traffic (Erlangs). The amount of traffic is the product of Call arrival rate (in calls per hour) and Average holding time (in hours).

### 7.1.2. For Packet/Data Type Traffic

For the performance of data type traffic in a hybrid network, the following approaches have been proposed.

#### 7.1.2.1. Quasi-Static approach

Quasi-Static approach [GKW82] consists of an analysis based on the behavior of the packet queueing process in the movable boundary SENET-concept. The packet queueing process in the SENET-concept exhibits behavior that is markedly different from that of a system in which the transmission capacity is fixed. (Each change in the size of the boundary results in a change in the transmission capacity of the packet system). In effect, the queueing system (considering the single-server queueing system with a randomly varying service rate) can enter different modes. If a given mode has a steady state, the queueing process will eventually settle into steady-state behavior if the size of the boundary, at a given time, remains fixed for a sufficiently long time. Using this quasi-static behavior in a Markov process model, the following approximate formula for the mean number of packets in the system has been mentioned.

$$E[L|n_c = k] = \rho(k) + \left(\frac{1 + C_p^2}{2}\right) \frac{\rho^2(k)}{1 - \rho(k)}$$

Where $E[L|n_c = k]$ is the steady-state mean number of packets in the system in mode k. $C_p^2$ is the coefficient of variation of packet lengths. The traffic utilization is

$$\rho = \frac{\lambda_p b}{\mu p[N - \bar{n}_c]} < 1$$

where
- $b$ is a time (sec.)
- $\lambda p$ is the packet arrival rate
- $1/\mu p$ is the mean packet length
- $N$ is the size of the boundary
- $\mu p(N - \bar{n}_c)/b$ is the average serving rate of the queueing system
- $\bar{n}_c$ is the average number of bits per frame

## 7.1.2.2. Fluid Approximation

The Fluid Approximation approach views the packet arrival and departure processes in the SENET approach as consisting of fluid flows. This is a good approximation when the utilization of a queueing exceeds unity. In this approximation, the accumulation of packets in the buffer during an overload period is represented as a growth of the fluid, and freeing of the packets from the buffer during stable period, as a shrinkage of the fluid. The overload period is a time period of the breakdown during processing of the packets, and the interbreakdown time periods are the stable periods.

The following steps are mentioned for using this fluid approximation method for estimating the average number of packets in the system.

1. Find the mean buffer growth and shrinkage rates.
2. Find the mean variance of the duration of overload and stable periods.
3. Use Hsu's result (see appendix) for finding average waiting time.
4. Determine the average waiting time using M/G/1 formula.
5. Use equations (see appendix) to obtain the average number of packets in the system.

### 7.1.2.3. Two-Dimensional Approaches

The first approach consists of an analysis that yields exact solutions of two-dimensional random process formulations. This approach involves the use of a determinant equation method for the solutions of mean packet waiting time, the number of packets in the system, and the probability of the packet buffer being full. The key difficulty with this approach is that in order to obtain numerical results, the roots of a determinant equation must be found. Some of these roots tend to cluster near 1. This limits the applicability of the approach to a small buffer sized hybrid network systems only.

The second approach also consists of an analysis that yields exact solutions of two-dimensional random process formulations. But this approach involves the use of iterative and recursive matrix methods. If the packet buffer is finite, direct recursion of the difference equations can be used to obtain the solutions, i.e., mean packet waiting time, the number of packets in the system, and the probability of the packet buffer being full. If the packet buffer is infinite, iterative recursion of the difference equations can be used to obtain the solutions. Though this method is suitable for both small and large Hybrid network systems, in practice, this method converges very slowly for typical parameter values that arise in voice and data systems.

In order to circumvent the numerical difficulties of the above methods, the matrix diagonalization approach has been proposed [WIL84] in order to obtain the solutions for the mean number of data packets in the system. Then, the mean delay of the packet can be found by Little's formula (the average number of customers in a queueing system is equal to the average arrival rate of customers to that system, times the average time spent in that system). The matrix diagonalization approach (see appendix) has been found more effective than either iterative or recursive techniques. In addition, the storage requirements of this

method are independent of the buffer size of the Hybrid network.

## 7.2. Comparison

The Integrated switches & networks, described in previous chapters, utilize different mechanisms and technologies in order to have packet and circuit switching within a single network, thereby making it difficult to compare them on the same basis. Therefore, performance criteria will be the basis for comparing the above mentioned switches and networks.

### 7.2.1. Integrated Switches

There are different opinions among investigators concerning which switching scheme is best for integrating circuit (voice) and packet (data) switching techniques. The comparison of SENET with fixed boundary, SENET with movable boundary, SENET with movable boundary and TASI, and flexible hybrid approaches based on the mean delay per interactive data packet versus throughput is shown in Figure 23. According to the author [ROM82] the farther a delay curve is from the Y-axis, the better the performance. The voice delay in the interated systems is somewhat greater than in the circuit switching, once the connection has been established. This is because of the 10 milisecond frame interval and the placement of the voice

bits in the output frame. The integrated systems shows less variation in voice delay with changes in throughput than do the packet-switching systems [ROM82].

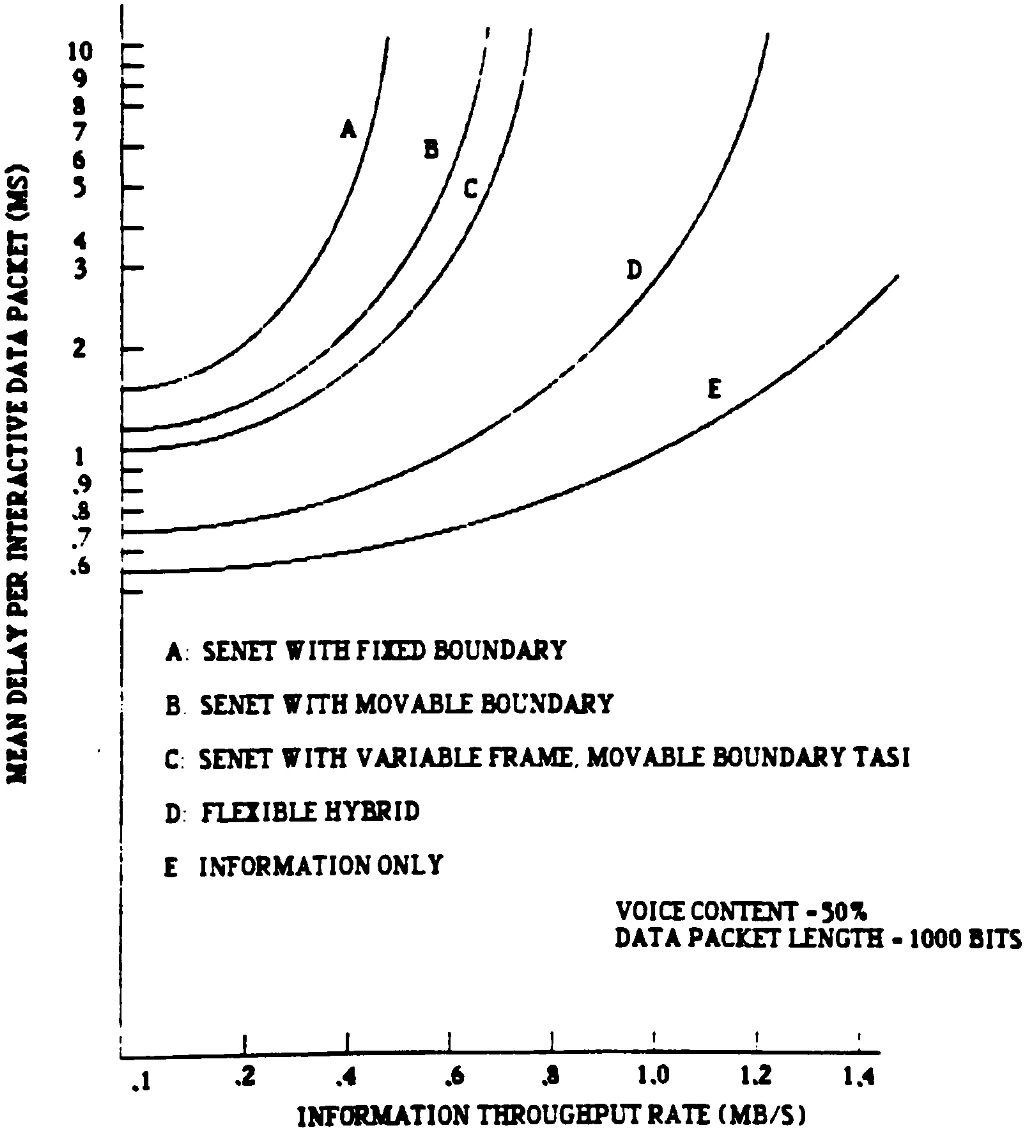



Figure 23 MEAN DELAY PER INTERACTIVE DATA PACKET VERSUS THROUGHPUT FOR VARIOUS HYBRID CONCEPTS

The comparison of no cuts, full cuts, and partial cuts-through approaches are shown in Figure 24. According to the author the farther a delay curve is to the Y-axis, the better the performance [LLM84].

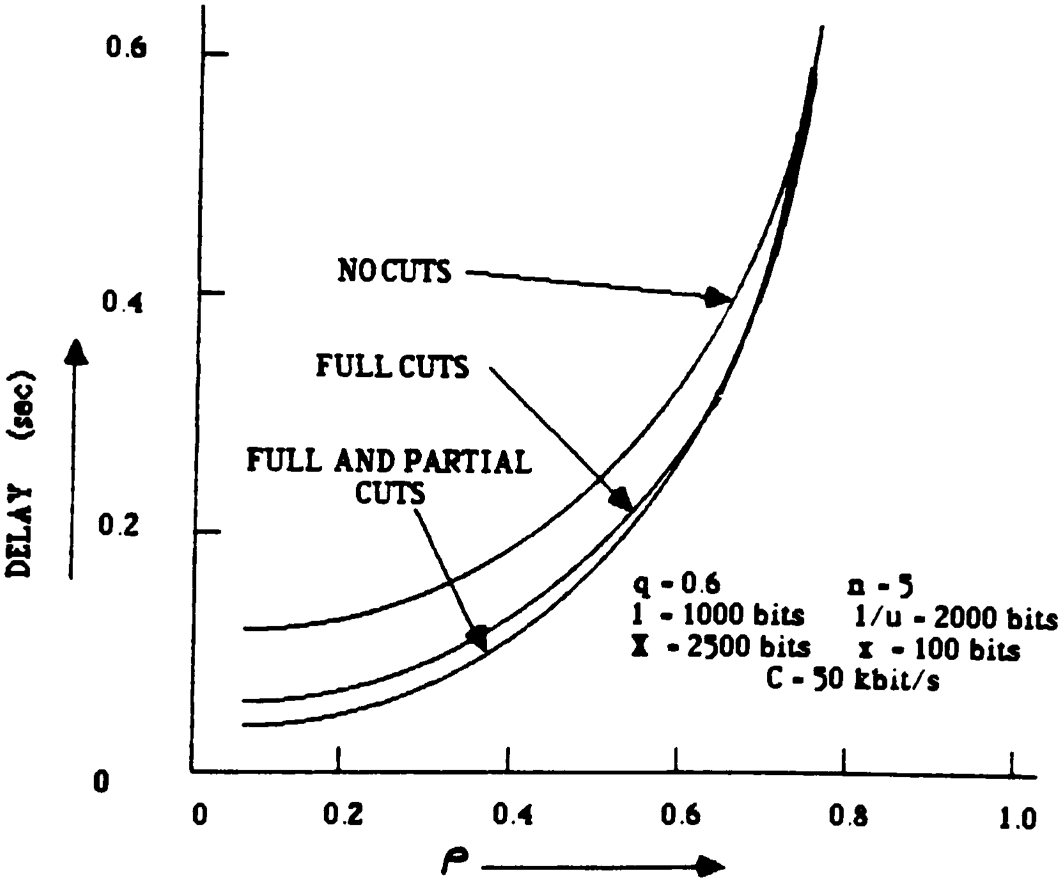



Figure 24. AVERAGE MESSAGE DELAY AGAINST TRAFFIC INTENSITY

## 7.2.2. Integrated Networks

Based on the simulation study of a ring system network [EHD85], minipacket protocol is compared with fixed boundary system for the waiting times for packet switched (PS) messages as shown in Figure 25. The waiting times are shown as a function of the total offered PS-load of constant message length of 1024 bits or 22 minipackets. The circuit switched (CS) loads are 30%, 50%, and 70%. The

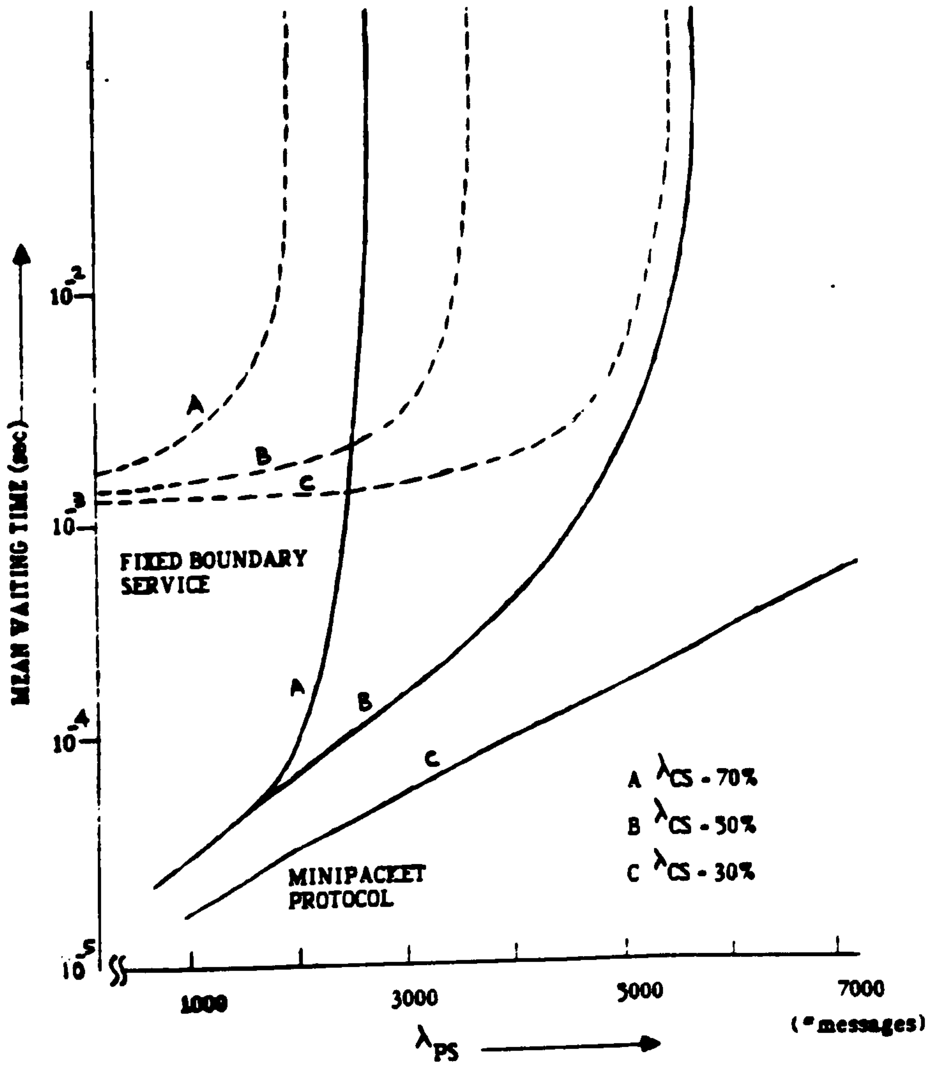



Figure 25. MEAN WAITING TIMES FOR PS-MESSAGES

According to the author, the throughput is better in the fixed boundary system because of a station holding the token sending all PS-messages which are in the buffer, but the waiting time is higher in a very low loaded system. The waiting times in a minipacket system are much lower and the throughput is maximum.

The performance of the integrated services digital network, as it is being realized by the the commercial digital switch makers, will be known at the end of the switch maker's field trials. Field trials are in progress in Europe and in the USA (for USA see Table 3). The principal objective of these trial activities are:

- to identify the real costs of ISDN
- to gain experience in the design and implementation of ISDN architectural principles
- to provide a test bed for new telecommunication services to determine the information needs of customers; and
- to develop new product and service concepts.

## Table 3. ISDN IMPLEMENTATION IN THE USA

| Region | Bell operating company | User location | Central Office switch | Architecture | Schedule |
|---|---|---|---|---|---|
| Ameritech | Illinois Bell | Dakbrook, Ill. — McDonald's Corp. | AT&T 5ESS | Island ISDN node | Dec. '86-mid-'88 |
| | Wisconsin Bell | Wisc. demonstration | Siemens AG EWSD | Stand-alone | July '85 - Dec. '86 |
| Bell Atlantic Corp. | Chesapeake and Potomac Telephone Co., Bell of Pennsylvania, New Jersey Bell. | Demonstration | NEC Corp. NEAX61E | ISDN adjunct | Sept. '85-3rd quarter '86 |
| | To be determined | To be determined | Siemens AG EWSD | To be determined | mid-'87 |
| BellSouth Corp. | Southern Bell Telephone and Telegraph Co. | Prime Computer, Inc. Trust Co. of Georgia | Northern Telecom, Inc. DMS-100 & AT&T 5ESS | Three interworking nodes | May '87-May '88 |
| Nynex Corp. | New York Telephone Co. | Manhattan — To be determined | Siemens AG EWSD | Basic & primary rate | 2nd quarter '87 |
| Pacific Telesis Group | Pacific Bell Telephone Co. | San Francisco | NEC Corp. NEAX61E (& AT&T 1AESS) Northern Telecom, Inc. DMS-100, AT&T 5ESS | Interworking nodes | 3rd quarter '87 - '89 |
| Southwestern Bell Corp. | Southwestern Bell Telephone Co. | St. Louis, Mo. | AT&T 5ESS, Northern Telecom, Inc. DMS-100 | Island nodes and interworking nodes | May '87-May '88 |
| US West Co. | Mountain Bell | Phoenix — State Government Honeywell, Inc. Telegroup | Northern Telecom Inc. DMS-100 | Single node, 3-remote host | Nov. '86-April '87 |
| | | Phoenix — GTE Corp. Mountain Bell | GTE Corp. GTD5EAX | Single node, 3-remote host, basic & primary rate, GTE ISDN-capable PBX | Dec. '86-April '87 |
| | | Phoenix, Chandler, Ariz. — To be determined | AT&T 5ESS | Optical-remote host | Feb. '87-Aug. '87 |
| | | Denver | NEC Corp. NEAX61E | 1AESS adjunct | Jan. '86-Nov. '86 |
| | Northwestern Bell | To be determined | NEC Corp. NEAX61E | Three ISDN adjuncts colocated with 1AESS | April-June '87 |
| | Pacific Northwest Bell | Portland, Ore. — US National Bank of Oregon | Northern Telecom, Inc. DMS-100 | Island ISDN node | Nov. '86-Nov. '87 |

# CHAPTER 8

# CONCLUSIONS

## 8. Conclusions

One of the factors contributing to the rapid growth experienced by the industrialized countries over the last half-century has been the development of effective data communication networks. Recently, much of the focus has been on the integration of data and voice, or of data, voice, and video within the same network. Integration of data, voice and video is being realized by integrating packet and circuit switching within the same switch or network, known as an integrated or hybrid switch network.

The purpose of this thesis was to present the key aspects of the integration of circuit and packet switching within the same switch/network in one place from several scattered publications which have appeared since 1978. Integration of circuit and packet switching characteristics into a single switch/network will enable the handling of three classes of communication traffic -- interruptible, bursty, and continuous -- in the most efficient way.

Approaches to the integration of circuit and packet switching range from circuit switched networks acting as a transport mechanism for packets to master framing. In

master framing time-division circuit switching and packets with self-contained routing and control information are carried in a common frame between switching centers. SENET and PACUIT are the approaches to master frame integration. With slight variations, each technique combines the three classes of traffic, with minimum overhead and high link efficiency.

Routing techniques for integrated circuit and packet switching range from distributed to centralized approaches.

Switch architectures for integrated circuit and packet switching range from a flexible-hybrid scheme, which is a combination on SENET and TASI, to the burst switching scheme, where routing and call information is carried along with the message rather than being transferred to functions in the switch. The burst switching is being implemented by GTE Corporation.

Network architecture for integrated circuit and packet switching is based on the combination of best features of PBX and LAN. This type of network architecture is recently introduced by Bell-Northern Research Ltd. known as "Meridian SL".

The recent development related to ISDN so far refers only to a set of standards for integrated access to a telephone central office from a customer's premises.

Access must be for circuit and packet switched data and voice. Under the ISDN concept, after reaching the central office, all traffic could be conveyed over separate networks to their exit points, where they must be integrated again. So, an ISDN could really be several backbone networks, and that is the way many vendors have depicted it.

The most popular digital switching systems (#5ESS, EWSD, GTE-5EAX and DMS-100) are being used in order to realize ISDN. All these digital switches provide computer-controlled, time division switching or statistical mutliplexed switching and have distributed architectures.

In an integrated circuit and packet switched network, the performance of the circuit type traffic can be easily obtained, either exactly or approximately, by the Erlang B formulae. Conversely, for packet type traffic, the performance analyses range from the quasi-static approach, where the behavior of the packet queueing process is based on the moving boundary SENET-concept, to two-dimensional approaches, where the numbers of voice calls and data packets represent the two dimensions.

The areas which need further investigation are the real-world implementations of some of the techniques mentioned in chapter 5, integration in the "Photon Switching" and the outcome of the ISDN field trials. The Photon

Switching is being considered for handling the traffic and signals in a fiber-optic transmission.

## GLOSSARY

Asynchronous transmission: A mode of communication characterizes by start/stop transmission with undefined time intervals between transmissions.

Central office: A switching system in the public network.

Common channel signalling: A signalling method using a link common to a number of channels for the transmission of signals necessary for the traffic by way of these channels.

Common control: A form of automatic control for a switching system that concentrates all control functions into one equipment shared by all connections.

Digital switching: A process in which connections are established by operations on digital signals without converting them to analogue signals.

Erlang: A measure of traffic intensity. Basically, a measure of the utilization of a resource(e.g, the average number of busy circuits in a trunk, or the ratio of time an individual circuit is busy).

Pulse code modulation: A process in which a signal is sampled, and the magnitude of each sample with respect to a fixed reference is quantized and converted by coding to a digital signal.

Space division: A technique for providing separate channels by assigning a physical path to each channel.

Statistical multiplex: A technique that combines voice or data signals in a higher bit rate stream by sharing the time available among the talkspurts or packets from the users.

Synchronous transmission: A mode of digital transmission in which discrete elements(symbols) are transmitted at a fixed and continuous rate.

TASI: Time Assignment Speech Interpolation, the practice of concentrating a group of voice signals onto a smaller group of channels by dynamically switching active voice signals to idle channels.

Time division: A technique for providing separate channels by assigning a unique succession of time slots to each channel.

Time-space:
Time-space-time:
Time-space-space-space-space-time:
All these represent a combination of Time division and Space division techniques.

Translation: In automatic telephony the retransmission of received trains of impulses after changing the number of impulses in each train and/or changing the number of trains.

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## APPENDIX

### MATRIX DIAGONALIZE APPROACH

1. The state of the hybrid system is given by the two-dimentional Markov process (see fig A.)

   $$(X(t),\ Y(t)),\ 0 \leq X(t) \leq N \text{ and } 0 \leq Y(t) \leq M$$

   where N, number of voice calls is always finite, and M, number of data packets may or may not be finite. X(t) corresponds to the voice process and Y(t) to data process.

2. The steady state marginal probabilities $\Pi$ for the voice (X) are given by

   $$\Pi = \sum_{j=0}^{\infty} P_j \qquad \text{------------ (1)}$$

   where $P_j$ is the N-dimensional column vector of the matrix.

   $$P_{ij} = \lim_{t \to \infty} Pr[X(t) = i,\ Y(t) = j] \text{ -- (2)}$$

   By taking the difference equations of the components of P ($1 \leq j \leq M - 1$) and solving for the components of P we obtain a second-order vector difference equation.

   $$P_{j+1} = A_{j+1} P_j + B_{j+1} P \quad , \quad 1 \leq j \leq M \text{ - (3)}$$

   with an appropriate set of boundary conditions for j=0 and j=M.

   Now since any second-order difference equation can be reducd to a first-order equation of twice the dimensionality. Therefore the 2(N + 1)-dimensional vector, define as

   $$x_j \triangleq \begin{bmatrix} P_{j+1} \\ P_j \end{bmatrix}$$

   Equation (3) then becomes

   $$x_{j+1} = K x_j \ , \qquad K = \begin{bmatrix} A & B \\ I & 0 \end{bmatrix} \text{ - (4)}$$

   The solution to (4) is of the form

   $$x_j = K^j x_0 \ ; \text{ thus,}$$

   $$\sum_{j=0}^{M-1} x_j = \sum_{j=0}^{M-1} K^j x_0 = F\ x \qquad \text{- (5)}$$

   where

   $$F = (I - K^M)^{-1} (I - K) \qquad \text{------ (6)}$$

$$\triangleq \begin{bmatrix} F_1 & F_2 \\ F_3 & F_4 \end{bmatrix}$$

and F is a $2(N+1) \times 2(N+1)$ matrix and $F_1$, $F_2$, $F_3$, $F_4$ are its $(N+1) \times (N+1)$ submatrices. $F^{-1}$ or F can be evaluated by diagonalizing K.

From the definition of x we obtain another expression for equation (5).

$$\sum_{j=0}^{M-1} x_j = \sum_{j=0}^{M-1} \begin{bmatrix} P_{j+1} \\ P_j \end{bmatrix} = \begin{bmatrix} \Pi - P_0 \\ \Pi - P_M \end{bmatrix} \quad \text{-- (7)}$$

by combining (5) and (7) we obtain

$$\begin{bmatrix} P_1 \\ P_0 \end{bmatrix} = x_0 = \begin{bmatrix} F_1 & F_2 \\ F_3 & F_4 \end{bmatrix} \begin{bmatrix} \Pi - P_0 \\ \Pi - P_M \end{bmatrix} \quad \text{------- (8)}$$

Since $P_i$ can be expressed in terms of $P_0$, multiplying (8) out and solving for $P_0$ and $P_M$. That is taking the difference equations for the components of $P_0$ and solve for the components of $P_1$, we obtain an equation of the form $P_1 = A'P_0$. Substitute for $P_1$, and multiply (8) out, we obtain

$$A'P_0 = F_1(\Pi - P_0) + F_2(\Pi - P_M)$$

$$P_0 = F_3(II - P\ ) + F_4(\Pi - P_M)$$

solving for $P_0$ and $P_M$ we obtain

$$P_0 = (C + F_4^{-1} - F_2^{-1}A'\ )^{-1} C\Pi, \quad \text{----- (9)}$$

$$\text{where } C = F_4^{-1}F_3 - F_2^{-1}F_1 \quad \text{and}$$

$$P_M = F_4^{-1}F\Pi + \Pi - F_4^{-1}F_3P_0 - F_4^{-1}P_0 - \quad (10)$$

(10) gives the probability of having the packet buffer full.

Similarly using the diagonalization approach, the mean number of data packets in the system can be found.

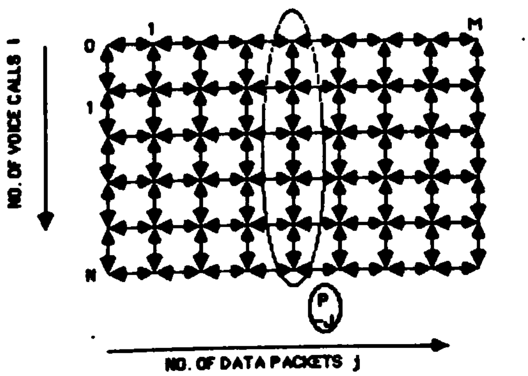


FIG. A.

FLUID APPROXIMATION

HSU's results:

Let $\bar{X}$ represents the amount of work, in units of packets, accumulated during an overload period of duration $\alpha$ and Y represents the amount of work that can be processed during a stable period of duration $\beta$. The onset of an overload period corresponds to a valley in the graph of Q(t) versus time, and the ending of an overload period corresponds to a peak in graph, see Fig. B. Hsu showed that if the breakdown periods and the stable periods consist of alternating renewal process, then the probability density functions of the peaks and valleys are exactly the same as the density functions for the response time and the waiting time, respectively, for a GI/G/1 queueing system for which the interarrival times are distributed as Y and the services times as X above.

If $E[Q_v]$ and $E[Q_\pi]$ denote the mean valley and the mean peak in the graph of Q(t), and $E[W^*]$ and $E[T^*]$ the mean waiting time and mean response times, respectively. Then Hsu's result implies the equalities

$$E[Q_v] = E[W^*]$$

and

$$E[Q_\pi] = E[T^*]$$

To estimate the average number of packets in the system, E[Q(t)], in terms of E[Q ] and E[Q ]. Define a nonempty period as a time segment begining when the buffer first becomes nonempty and ending when it first returns to 0. Subsegment of the nonempty periods in the graph are marked as *. The average value of Q(t) in the subsegment is given by the sum of a nonzero valley Q and adjacent peak Q = Q + X divided by 2. There for possible estimate for heavy traffic conditions is:

$$\bar{Q} = E\left[\frac{Q_v + (Q_v + X)}{2} \,\middle|\, Q_v > 0\right]$$

$$= E[Q_v \mid Q_v > 0] + \frac{1}{2} E[X]$$

$$= \frac{1}{\rho^*} E[Q\ ] + \frac{1}{2} E[X] \qquad \text{-- (1.1)}$$

where $\rho^* = E[X]/E[Y]$ is the utilization of the GI/G/1 system.

Estimate for light traffic conditions is:

$$\overline{Q} = E\left|\frac{Q_V + (Q_V + X}{2}\right|$$

$$= E[Q\ ] + \frac{1}{2}\,E[X] \qquad \text{-- (1.2)}$$

Estimates for the mean number of packets in the system, can be obtained by multiplying (1.1) and (1.2) by the porportion of time that fluid-approximation system is nonempty.

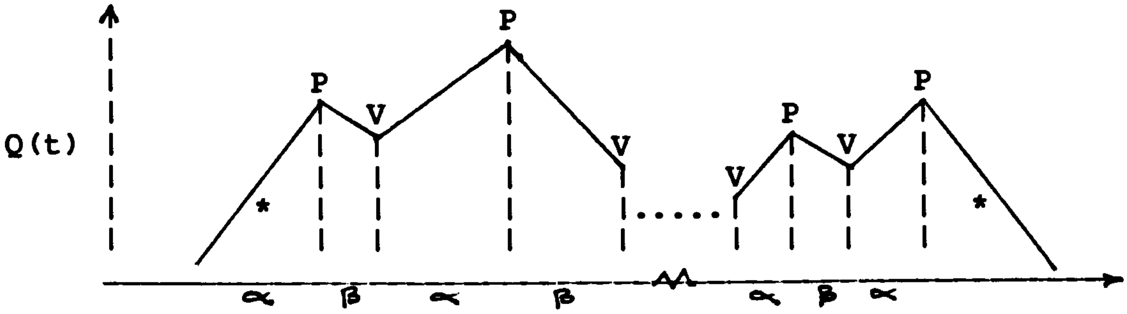


FIG. B. Fluid approximation behavior of queueing system subject to breakdowns.