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Control Signaling in a Military Switching Environment

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The views, opinions, and/or findings contained in this report are those of the author and should not be construed as an official Department of the Army position, policy or decision, unless designated by other official documentation.

PREFACE

The study reported here is one part of a continuing program being conducted by the Institute for Telecommunication Sciences (ITS) for the U.S. Army's Communication Systems Agency (CSA) in support of that Agency's Access Area Digital Switching System (AADSS) program on project orders 501-RD, 804-RD, and 809-RD.

Previous related reports have dealt with parametric cost alternatives for local digital distribution systems, as well as with several topics in access area switching. The latter has included preliminary switching hub evaluations for the base environment, digital PABX switching, and signaling technology in common use today.

Administration and technical monitoring of the study contract was performed by Messrs. L.H. Wagner and T. Michelli, both with CSA. Technical and management supervision of the program was provided by M. Nesenbergs and Dr. P. McManamon, both of ITS.

TABLE OF CONTENTS

LIST OF FIGURES

LIST OF TABLES

LIST OF ACRONYMS

AADSS	Access Area Digital Switching System
ADCCP	Advanced Data Communications Control Procedure
ACK	acknowledgement of correct reception
ACU	acknowledgement signal unit
ANSI	American National Standards Institute
AM	amplitude modulation
APC	adaptive predictive coding
ARPA	Advanced Research Projects Agency
ARPANET	packet switched network developed by ARPA
ARQ	automatic repeat request for error control
ASCII	American Standard Code for Information
AT&T	American Telephone and Telegraph Co.
AUTODIN	automatic digital network
AUTO-SEVOCOM	automatic secure voice communication network
AUTOVON	automatic voice network
BASCOP	base communications program
BDLC	Burrough's data link control
BER	bit error rate
BORSCHT	battery, overload, ringing, supervision, clock, hybrid and testing
BSC	binary synchronous control, also called bi-sync
CATV	community antenna television, also called cable TV
CCIS	common channel interoffice (or interswitch) signaling
CCITT	International Telegraph and Telephone Consultative Committee
CCTV	closed circuit television
CERCOM	Communications and Electronics Readiness Command (U.S. Army)
CO	central office
COMSEC	communications security
CONUS	continental United States (excluding Alaska, Hawaii, and various territories)
CPODA	contention-based priority oriented demand assignment

CPU	central processing unit
CRC	cyclic redundancy check
CRT	cathode ray tube
CSA	Communications Systems Agency (U.S. Army)
CVSD	continuously variable slope delta, voice digitization technique
DCA	Defense Communications Agency
DCE	data circuit-terminating equipment
DCS	Defense Communication System (terrestrial, strategic backbone)
DDCMP	digital data communications message protocol
DEC	Digital Equipment Corporation
DECNET	Digital Equipment Corporations Network Architecture
DLC	data link control
DoC	Department of Commerce
DoD	Department of Defense
DNA	Digital Network Architecture (NCR)
DPCM	differential pulse code modulation
DSCS	Defense Satellite Communications System
DSE	data switching equipment
DTE	data terminal equipment
EBCDIC	extended binary coded decimal interchange code
ECOM	Electronics Command (U.S. Army)
EFTO	encyrpted for transmission only
EIA	Electronic Industries Association
EO	end office
EPABX	electronic private automatic branch exchange
ESS	electronic switching system
ETS	European telephone service
FAX	facsimile
FDM	frequency division multiplex
FDMA	frequency division multiple access
FEC	forward error correction
FM	frequency modulation

FS 1033	proposed Federal Standard for user oriented performance parameters
GPIB	General Purpose Interface Bus
HDLC	high-level data link control
HP	Hewlett-Packard Company
Hz	hertz, or cycles per second
IAM	initial address message
IBM	International Business Machines Corp.
IEEE	Institute of Electrical and Electronics Engineers, Inc.
IMP	interface message processor
I/O	input and/or output
ISO	International Standards Organization
ISU	initial signaling unit
ITS	Institute for Telecommunication Sciences
ITU	International Telecommunications Union
LPC	linear predictive coding
HRC	horizontal redundancy check
LSI	large scale integration
LSU	lone signal unit
MBS	multiblock synchronization signaling unit
MF	multi-frequency
MFDT	multi-frequency dual tone
MTBF	mean time between failures
MTTR	mean time to repair
MUM	multi-unit message
MUX	multiplexer
NAK	negative acknowledgement
NCA	National Command Authority
NCR	National Cash Register Company
NMM	network management and maintenance signal
NPA	numbering plan area, also known as area code in telephony
NTIA	National Telecommunications and Information Administration
O&M	operation and maintenance
PABX	private automatic branch exchange

PCM	pulse code modulation
P2SV	phase two secure voice network
RS	recommended standard by EIA
RS-232C	interface between DTE & DCE employing serial binary data interchange
RS-449	general purpose interface for DTE & DCE employing serial binary data interchange
RS-422	electrical characteristics of balanced voltage digital interface circuits
RS 423	electrical characteristics of unbalanced voltage digital interface circuits
SAC	Strategic Air Command
SAM	subsequent address message
SATIN	SAC Automatic Total Information Network
SBC	sideband coder
SC	suppressed carrier
SCU	system control unit
SDLC	synchronous data link control (IBM)
SENET	slotted envelope network
SF	single frequency
SNA	Systems Network Architecture (IBM)
SSB	single sideband
SSTV	slow scan television
SSU	subsequent signaling unit
STP	signal transfer point in CCIS
SU	signaling unit
SYU	synchronizing signaling unit
T-CXR	T-carrier
TDM	time-division multiplex
TDMA	time-division multiple access
TRI-TAC	Joint Tactical Communications Program
TTY	teletypewriter
V.	CCITT designation for series of recommendations on data transmission over telephone networks
VDP	voice digitization processor

VRC	vertical redundancy check
V.10	electrical characteristics for unbalanced double-current circuits, also known as X.26
V.11	electrical characteristics for balanced double-current circuits, also known as X.27
WATS	wide area telephone service
WE	Western Electric Corporation
WIN	WWMCCS intercomputer network
WWMCCS	World-wide Military Command and Control System
X.	CCITT designation for series of recommendations on data transmission over public data networks
X3.28	American National Standard for use of communication control characters of ASCII for Information Interchange (1976)
X3.44	American National Standard for determination of the performance of data communication systems (1974)
X3.69	American National Standard for general purpose interface between DTE & DCE for synchronous operation on public data network (Proposed)
X.21	general purpose interface between DTE & DCE for synchronous operation on public data network
X.24	definitions for interchange circuits between DTE & DCE on public data networks
X.25	interface between DTE & DCE for terminals operating in packet mode on public data networks
X.60	common channel signaling for synchronous data applications - data user part

CONTROL SIGNALING IN A MILITARY SWITCHING ENVIRONMENT

R.F. Linfield*

A number of factors, including network topology, traffic characteristics, switching technology, and transmission media, affect the choice of control signaling techniques to be used in networks of the future. In this report these factors are considered from a military standpoint. It is projected that, over the next two decades, all of the elements of military networks will undergo a transition from essentially all-analog to all-digital systems. It is also possible that a unified integrated network capable of handling digital voice and data could evolve. During the forthcoming period of transition, as well as for the ultimate network configuration, the control signaling system will play a critical role in terms of both performance and cost. This report examines that role by evaluating several advanced signaling systems, particularly those using common control links. Digital systems that have potential application on both the line and trunk sides of the digital switch are examined.

1. SYNOPSIS

One of the major issues which arises in the design, development and deployment of a telecommunications network is the specification of network management and control. The design of the control signaling system is a prime example. Because military networks tend to evolve slowly, perhaps over a decade or two, resolution now of the key control signaling issues could influence, if not fully determine, the direction future networks will take in the 1980's and 1990's. This report provides an introduction to these issues by reviewing the more important advanced signaling concepts available today.

The report begins with a review of networks in general and the military switching environment in particular (Secs. 2 and 3). Next, the network control functions are defined, and ways and

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means for performing these functions are described (Sec. 4). Specific signaling systems which are either in use or are proposed for use are described in Section 5. Their applications to the military switching environment are compared and evaluated. Recommendations are made concerning choice of signaling schemes for future military network controls (Sec. 6). The remaining issues in the control area are summarized last. Ultimately, these issues can only be resolved by conclusive field testing and further study (Sec. 7). References are listed at the end (Sec. 8).

The synopsis summarizes the background, key results, and conclusions developed at in this report.

1.1. Background

Telecommunications means information transfer at a distance. A switched telecommunications network, such as provided by telephone systems, provides economical telecommunications to many users. This is because the many relatively low demand users can share the network resources. Network resources consist of three basic elements - terminals to provide transducer functions, nodes to provide queueing, routing, and redressing functions, and links to provide transmission functions (see Fig. 1-1).

Terminal functions permit coupling between users (human) and between users' machines (computers). Node functions are performed by concentrators, processors, and switches. Link functions are performed by a variety of facilities conveying information, in electromagnetic form; e.g., wires, coaxial cables, radio or optical fibers. Several users can share a terminal, terminals can share the switches and switches can share the transmission facilities to minimize information transfer costs.

Special signals (different from voice and data information bearing signals) are conveyed over the links to control this sharing process. The system devoted to these special control signals is known as the signaling system. The signaling system, like a nerve system in a human body, is pervasive throughout all

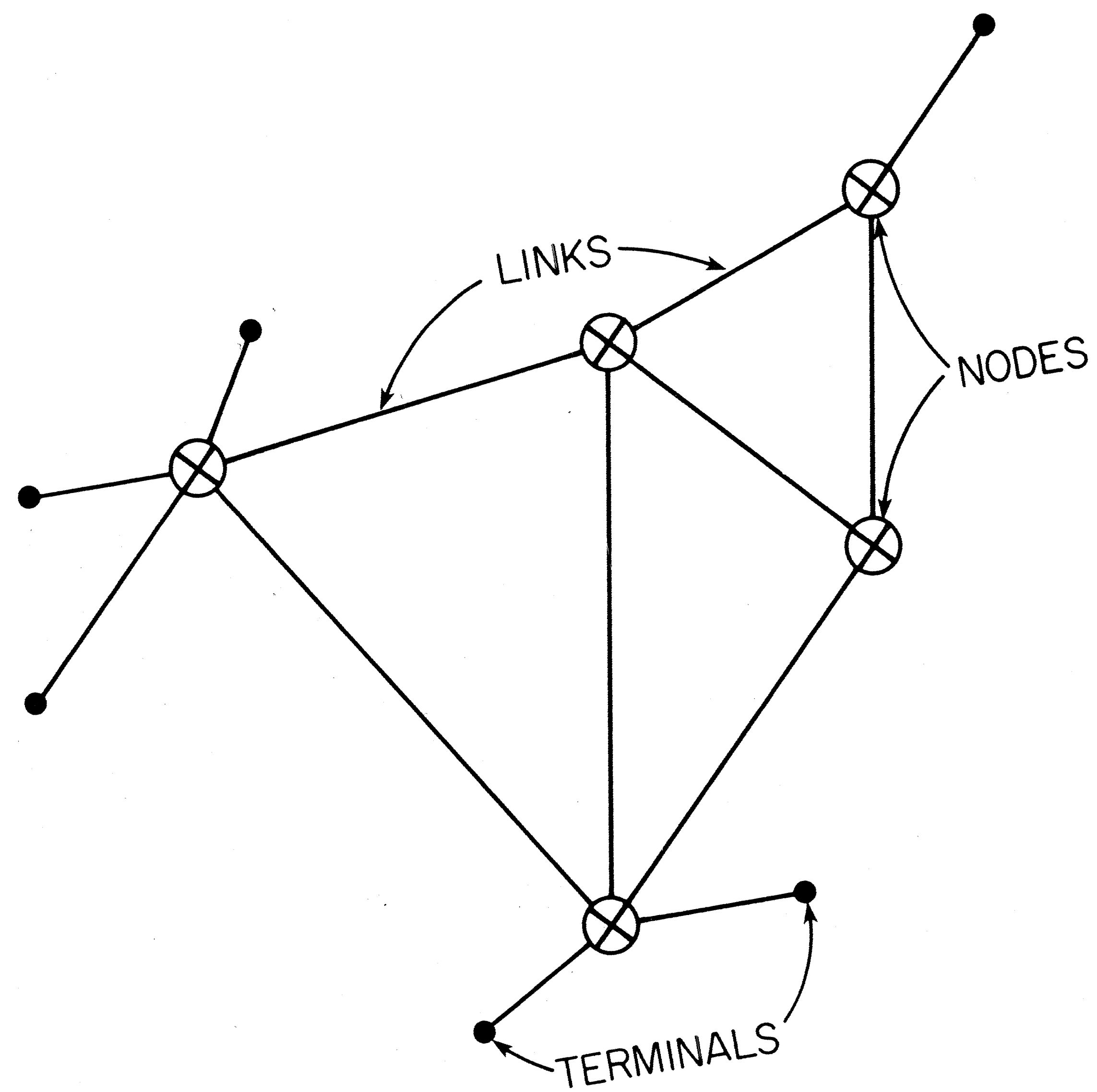


Figure 1-1. A telecommunications network.

the required network elements. Station signaling, over links called lines, remotely controls the terminals; interswitch signaling, over links called trunks, remotely controls the switches. Together, the station signaling systems and the interswitch signaling systems control and manage the entire network.

Like all technologies, the evolutions of telecommunications networks are shared by two driving forces: performance and cost. A given network, more often than not, is a compromise between the typically conflicting objectives of performance and cost. For instance, a common carrier network user desires efficiency, accuracy and reliability. He wants to minimize his costs by using the network as little as is necessary to meet his needs. At the same time, he may desire fast access with service always available. The common carrier network supplier, in contrast to the user, has economic reasons to sell as much service as possible; to produce more income by sharing the network resources among more revenue generating customers.

The military user of a network has many of the same needs as the civilian user, but requires a higher degree of reliability and availability. In addition, the military user has certain unique requirements such as mobility, security and survivability. The military user's service needs are different and the network he uses must be adaptable to more dynamic conditions. The military network operator may also be viewed as a paying consumer rather than a producer (collector) of revenue. Such operators must provide adequate service in a most economic fashion. This economy can only be realized by careful consideration of all the network elements, including cost to make, own, operate, and maintain the terminals, the switching nodes and the transmission links. The importance of minimizing costs cannot be overemphasized. Almost one billion dollars per year is expended on new installations, and on the operation and maintenance, of the Defense Communication System (DCS)* alone. This is more than one

*The acronyms of this report, including those used in figures and tables are on pp. xiii through xvii above.

fifth of the yearly total expenditures for communications by the Dept. of Defense.

Evolution of civilian and military telecommunication networks must take into account compatibility with existing systems and adaptability to future developments. Adaptability and compatibility involve a number of complex network control and management issues. The signaling strategy used for transferring control and management information impacts the total network. The strategy requires control signaling not only between all elements within a given network (intranet control), but also between the elements of independently operating networks (internet control). Intranet control is concerned with internal network interfaces and the remote control of the terminals, switches and transmission facilities within the network. Internet control is concerned with the gateways for interconnecting networks which may be separated functionally as well as geographically. This aspect is discussed in Section 2.1.

The Access Area Digital Switching System (AADSS) provides internet functional and geographic interfaces. The access interfaces provided by AADSS are indicated in Figure 1-2. The AADSS provides internet control and traffic flow between tactical, non-tactical, and strategic systems, plus commercial networks.

In the past, internetting has had severe limitations because individual networks were designed and dedicated for specific users. The engineering techniques and operating procedures employed were seldom standardized. In the future, the technological and procedural limitations may be overcome by using broadly applicable integrations of protocols, such as A/D conversion, hierarchal architectures, stored program controls, and common channel interswitch signaling.

The forecasted evolution of military networks from all-analog, to analog-digital hybrids, and ultimately to all-digital network configurations has already begun. Digitization, which started at the strategic backbone level, is expected to progress to the local military base tactical and non-tactical level within

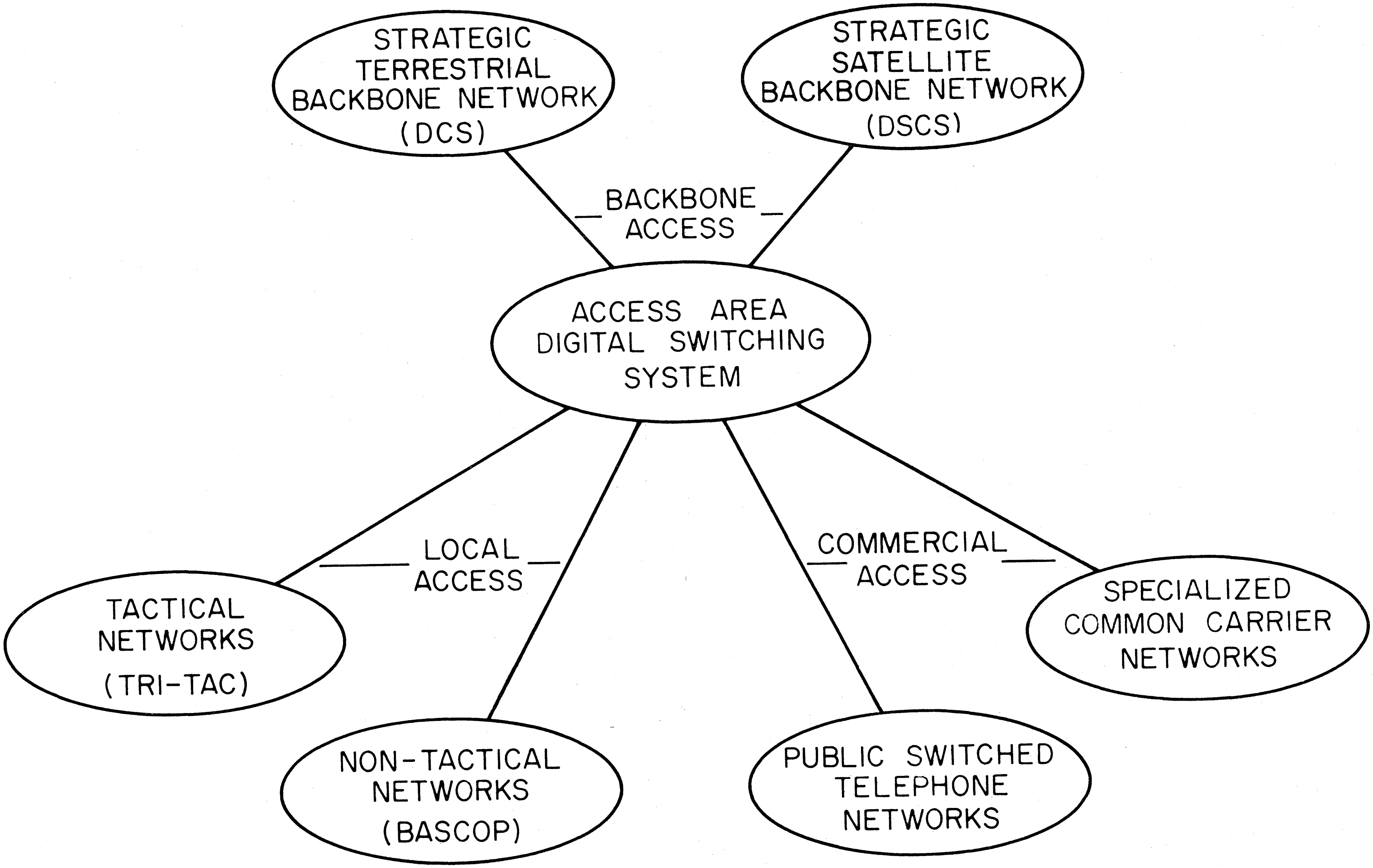


Figure 1-2. Access interfaces.

two decades. As this trend continues, it is entirely possible that network services may also be combined. Some projections describe a single network capable of handling all of the military voice and data communications needs. The justification for unification must be based on cost effectiveness of the total system.

This study begins by examining the present status of military networks, the projected traffic needs, potential switching and transmission technology and numerous cost factors. Since signaling systems must interface with both new and old network elements, this review is essential.

Future military telecommunications networks are expected to handle an increasing number of widely dispersed terminals with diverse traffic rates, transaction periods and acceptable delivery times. Traffic projections are given in Section 2.2.

Terminal types will vary from massive data management systems under distributed computer control, to traditional teletypewriters and telephones, to assorted sensors and monitoring systems. Although data traffic is expected to increase, it is predicted that over the next decade 90% of the military terminals will serve voice subscribers. Approximately 1 million telephone instruments will be connected to the DCS global network from 1500 separate access areas ranging from small clusters (<300) to medium clusters (300 to 2000), to large clusters (>2000) of terminals. Terminal densities will range from less than 10 per square kilometer to over 10,000 per square kilometer. The traffic generated by these terminals may vary in data rate from less than 100 bits per second, for low speed teletypewriters, to hundreds of kilobits per second for high resolution graphics and imaging. Information exchange transactions will range from a few hundred bits for interactive messages to several million bits for bulk data transfers. Delivery times will range from a few hundred milliseconds for voice and video to several hours for bulk data transfers will be either standard operating practice or system requirements of the future.

Tying these terminals together, so that each can be connected to any other terminal, is a massive networking task. Various sizes and types of switches are used at network nodes depending on the amount and kind of traffic to be used. Switch types generally fall into one of two categories: circuit switches or store and forward switches. Hybrid switches may also be used to handle both voice and data traffic. Advanced switching technology utilizes a common control system which consists of a computer processor with software programs stored in a memory to perform the control functions.

The switching in stored program controlled systems may use space division switching, time division switching, or combinations of these. In a space division switch, a matrix of crosspoints is formed using metallic reed relays or electronic semiconductors. In time division switches logic gates perform the time slot interchange functions and no matrix is required. Basic switching concepts including circuit, message and packet switching are discussed in Section 2.3.

The switching nodes of the network are linked together by various types of transmission facilities. The transmission facility employed may be wire pairs, coaxial cable, fiber optical waveguide, microwave radio or satellite, depending on such factors as node separation, traffic load, and terrain traversed. The links between terminals and nodes and between the nodes carry the user information. They also can be used to transfer the control information between the switch control systems (i.e., the processors) although a separate network may also be used for this purpose. A description of some of the transmission facilities in use today is given in Section 2.4.

The differences between the military switching environment and the commercial switching environment are noted in Section 3. Comparisons between the two environments are given in terms of the following characteristics:

- survivability
- availability
- security and privacy
- reliability
- traffic
- adaptability
- compatibility
- services offered
- performance
- cost.

In addition, some military networks which are currently being used are described. The effect of digitization, stored program control, common channel signaling and service integration are reviewed. Some projections are given regarding military networks in the future. Emphasis is on the control and management of these future networks.

This background material is essential to the understanding of the report sections which follow because the switching environment has a considerable impact on the selection of signal systems which control the network.

1.2. Network Control

The network control and management functions are considered in Section 4. Specific systems used to accomplish these functions for both telephony and digital data networks are described in Section 5.

Various types of signals may flow on a network to control the terminals, the nodes, the links, and the total network. Advanced signaling techniques are considered for both circuit-switched networks and store-and-forward switched networks. Since most of the network projections indicate a trend toward all-digital networks, the emphasis is on digital out-of-band signaling systems for the line side of the switch and digital common channel signal systems for the trunk side.

Signaling system requirements for military networks are reviewed in Section 4.1 in terms of the distinctive characteristics of these networks. The signaling issues which must be resolved in order to meet these requirements are also summarized. Basic concepts used by most of the advanced signaling techniques, including modes of operation, procedures, synchronization, and error control, are explained in Section 4.2. The user-oriented performance criteria used to evaluate and compare operating systems are defined in Section 4.3.

The control procedures or protocols for controlling voice networks using circuit switching are different from data networks using message or packet switching. In Section 5, these differences are indicated by describing typical examples of signaling systems and the protocols they use.

Signaling formats, codes, and protocols specified by various standards organizations for controlling several kinds of networks are described in detail in Section 5.1 for voice networks and in Section 5.2 for data networks. These include the common channel interoffice signaling (CCIS) system and the CCITT Signaling System No. 6 for telephone networks, and the X.25 protocol for packet switched data networks.

A new signaling system is described in Section 5.1, denoted CCITT No. 7. This system is being developed for use on networks which utilize both digital switching and digital transmission for either voice and data. CCITT No. 7 is designed for 64 kb/s operation and contains two parts, a message transfer part and a user part. The message transfer part is common to both voice and data circuits whereas the user part depends on the application. Other combination-type protocols are also discussed. These involve "nesting" various protocol levels together in a single format. The link, network, system, and user levels are distinguishable by position of bit sequences in the master frame.

1.3. Conclusions and Recommendations

The report describes how the signaling system permeates and impacts all the elements of a network, i.e., the terminal, the switching and concentrating nodes, and the transmission links. Thus, the ultimate choice of a signaling system for future integrated digital networks requires the resolution of issues which also involves all of the network elements.

Integrated digital networks for telephony (switching and transmission) are already evolving in the commercial world. However, this digitization process is being implemented in different ways by different carriers. Some begin with an overlay of digital facilities, others by replacement of obsolete transmission facilities or switching centers. Still others are installing digital islands of networks which ultimately will be connected by digital long-haul facilities. Just how this evolution will progress in the military world, and how fast is unknown. In the interim transition period, all of the user's information, whether it be voice or data is transferred to switching nodes over dedicated subscriber lines. A switching node connects these lines to other lines or to outgoing trunks. The interconnection process may take many forms and use various media for control purposes depending on local requirements for transmission signaling, numbering, etc. The need for compatibility between existing equipments and evolving systems is a challenge to designers of the military networks of the future. Some AADSS applications are presented in Section 6. The AADSS is viewed as an interconnecting gateway between military networks of various types (e.g., strategic, tactical, and non-tactical)* and at various levels (e.g., global, regional, and local).

Closely related to the integration of digital network elements is the integration of the services that an all-digital network could provide. New terminals and new switching systems capable of handling both voice and data are required to integrate these services and thereby provide flexibility in the options available to the military users. The extension of all-digital formats to the subscriber's terminal and the use of an all-digital network to interconnect such terminals is a possibility in the future. When and if such a network should evolve, some of the functions and features of the signaling system to be used to control this unified system can be defined. The signaling

*Non-tactical networks included those which support base operations, e.g., fire, military police, medical, etc.

system should (1) be capable of using any link including those normally used to handle data, (2) permit the addition of new network elements and new service features using a multilevel format, and (3) utilize a common discipline between data transmission and signaling so that a link of the control signaling system is basically the same as any other data link.

Areas where further studies are recommended before a final control system can be selected are outlined in Section 7. The recommendations include work on:

1. The impact of signaling systems on the control processor capabilities.
2. The optimal control system configuration for network management functions.
3. Control processor requirements as a function of routing strategy.
4. Multiple access protocols for application on integrated satellite networks.
5. Network performance standardization parameters and quantitative measures of these parameters.
6. Traffic engineering studies to optimize the total flow of intranetwork and internetwork telecommunications traffic.
7. Computer simulation studies to aid in items 3, 5, and 6 above.

2. TELECOMMUNICATIONS NETWORKS

A telecommunications network provides the paths needed between user terminals for information transfer. This information may be voice, data, narrative or imagery. The terminals are connected to nodes for concentration and switching so that wideband transmission facilities can be shared by many users (see Fig. 1-1).

In this section, the major network elements (terminals, switches, and transmission facilities) and the network topologies are reviewed. One should emphasize that it is the network topology that ties all of the elements together. The network control system utilizes its own, perhaps slightly different, topology to distribute signaling messages.

The AADSS signaling system must interface with terminals located in a military base environment. Interbase communications proceeds from the intrabase level via the local access network to a regional level via the regional access network and, at times, to a global level via a backbone access network. The control signaling system therefore is pervasive throughout all network elements at all access levels.

In order to evaluate given signaling systems, it is necessary to consider not only the network topology but the terminals and the traffic they generate. One must consider the basic switching concepts for both separate and integrated voice and data switching, as well as the transmission media to serve both analog and digital facilities.

Although ultimately all of the network elements may utilize digital technologies, it is useful to review analog implementations currently in use. One expects that the control signaling systems developed in the future will be required, in many instances, to interface with such existing network elements. This interface compatibility between the "old" and the "new" is particularly important during the expected transition from all-analog to all-digital networks.

The networking concepts presented in this section are not necessarily unique nor optimum. However, they do indicate potential configurations and general trends.

2.1. Topology

This section is concerned with the interconnecting links and the structural form of the networks. Studies have been conducted on network topology to show how the network can be configured to maximize traffic flow and minimize cost. (See, for example, Coviello and Rosner, 1974.) The topographic structure of a network also affects survivability. Redundancy is incorporated in all network elements to insure that certain critical users can maintain communication under a range of adverse conditions.

2.1.1. Types of Networks

There are numerous network topologies which can be envisioned for connecting n nodes of a network together. In fact, if n is large, it can be shown (Nesenbergs and Linfield, 1976) that the total number of interconnecting possibilities approaches $2^{n^2/2}$. For n=1000 (not an unreasonable number of nodes) the number of possible network topologies is approximately $10^{150,000}$, an astronomical number. Since each network must have at least one route between any two nodes, the total number of possible routes could be of the order of $n^2 2^{n^2/2}$.

Figure 2-1 depicts several general network types with their commonly used names. Real networks usually differ from these ideal models. However, real networks can usually be viewed as constructed from such simple subnetworks.

2.1.2. Examples for Access Networks

Figure 2-2 illustrates the topology for two network structures which might be used to provide access from a local base environment to a global "backbone" network using a three level hierarchal structure.

In Figure 2-2a the terminals in a local access area are served by a local access switch. A group of such local access switches are shown connected by a star network. This is typical of the local loop plant for the telephone network on a military base. Lines to telephone subscriber instruments radiate out from a central office (CO) on the base and from private automatic branch exchanges (PABX) in office complexes. The CO is connected via trunks to an end office (EO) of the commercial common carrier network, to tactical and non-tactical networks via cable or radio links and to the AUTOVON, AUTODIN, and AUTOSEVOCOM backbone switching centers. These Defense Communication System (DCS) networks are described in Section 3.1.

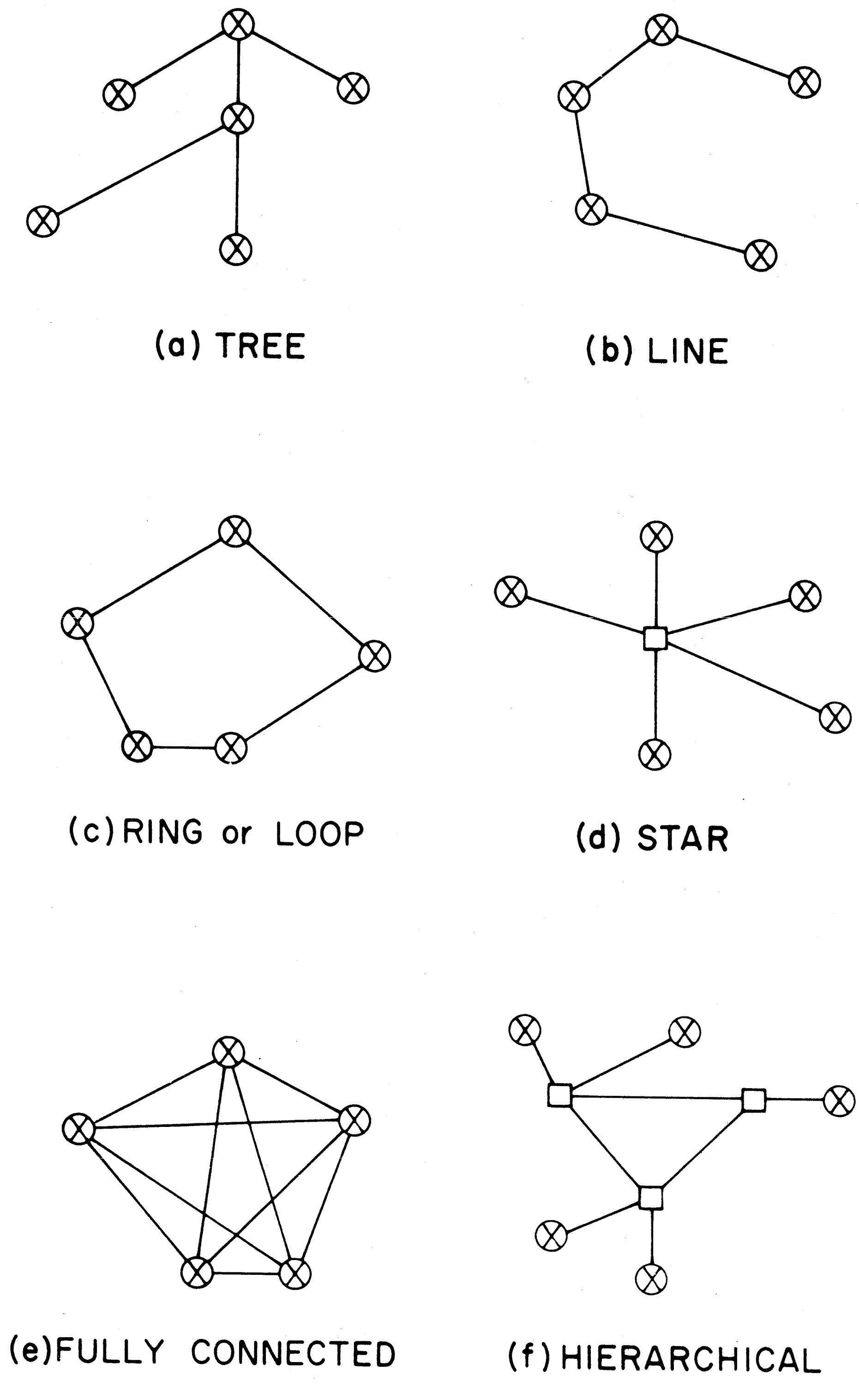
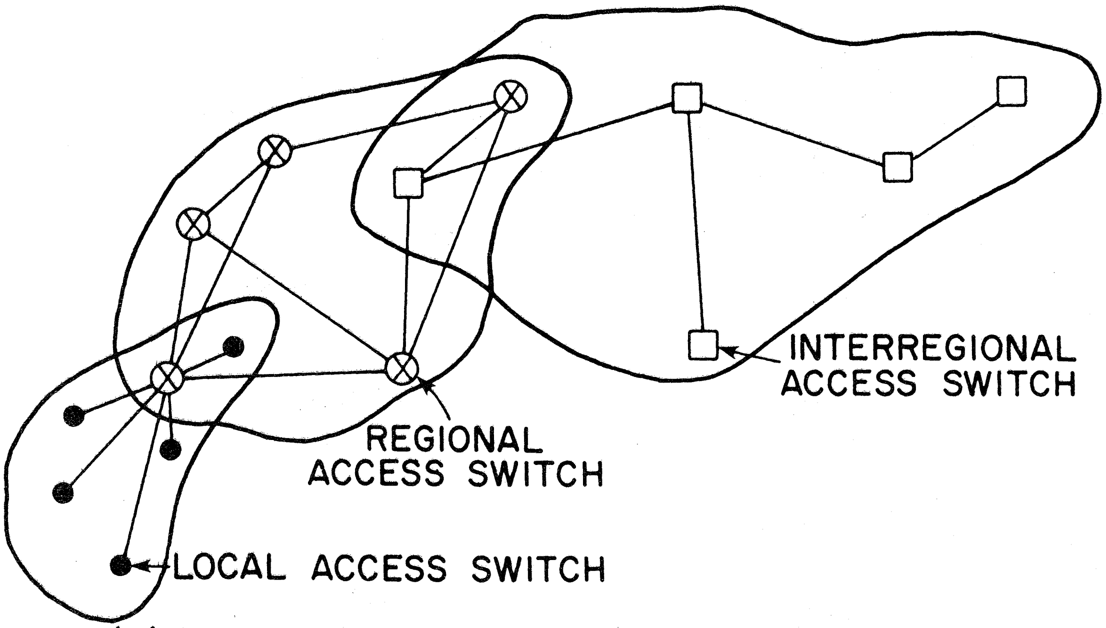
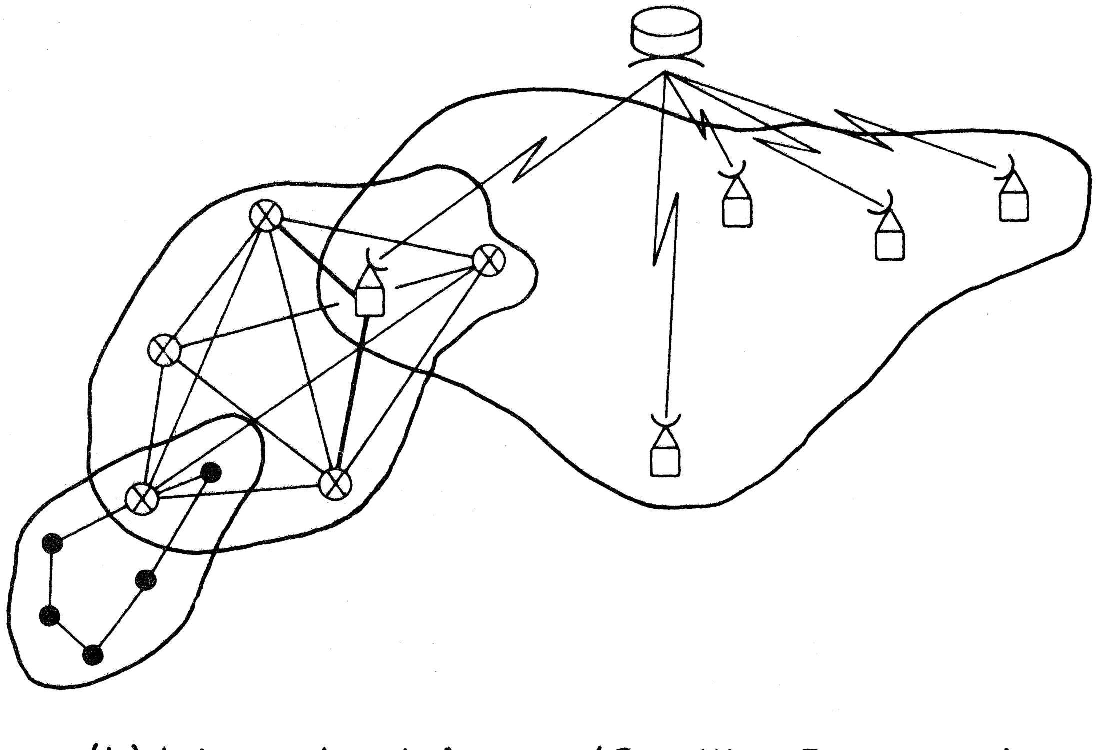


Figure 2-1. Network topologies.



(a) Interregional Access (Terrestrial Backbone)



(b) Interregional Access (Satellite Backbone)

Figure 2-2. Hierarchical network configurations for interregional access.

The regional access network shown in Figure 2-2a uses a partially connected network. This grid-like structure is typical of that used by AUTOVON. The AUTOVON switches and transmission facilities are leased from the common carrier industry for private military use. Generally there are at least two indirect paths connecting any switch pair in the network.

The interregional access nodes are connected by a tree network. This terrestrial backbone configuration is exemplified by the digital European backbone (DEB) network for the DCS.

An alternative hierarchy of networks is shown in Figure 2-2b. Here a loop configuration interconnects terminals or PABX's in the local access area, a fully connected grid connects the regional access nodes, and the interregional access is via a star network that has a satellite for the central node. This satellite backbone is typified by the Defense Satellite Communications System (DSCS).

It is obvious that numerous other topologies can be envisioned. Combinations of several ideal topologies are also used. For example, the terrestrial and satellite backbone networks can operate in conjunction with each other to enhance survivability and to carry the traffic load jointly.

The hierarchy may extend to even lower levels on a base as indicated in Figure 2-3. The central office or the main post switch serves not only individual terminals, but also remote concentrators and private automatic branch exchanges (PABX's). The PABX may be located in an office complex or other cluster of terminals. Substantial parts of the PABX such as concentration and multiplexing may be remoted from the central hub of the PABX. Note that in Figures 2-2 and 2-3 only one of several subhierarchical networks are indicated.

2.2. Terminals and Traffic

The ultimate source and destination of nearly all information traversing the network is the user terminal. One exception to this is control information which may terminate in a node.

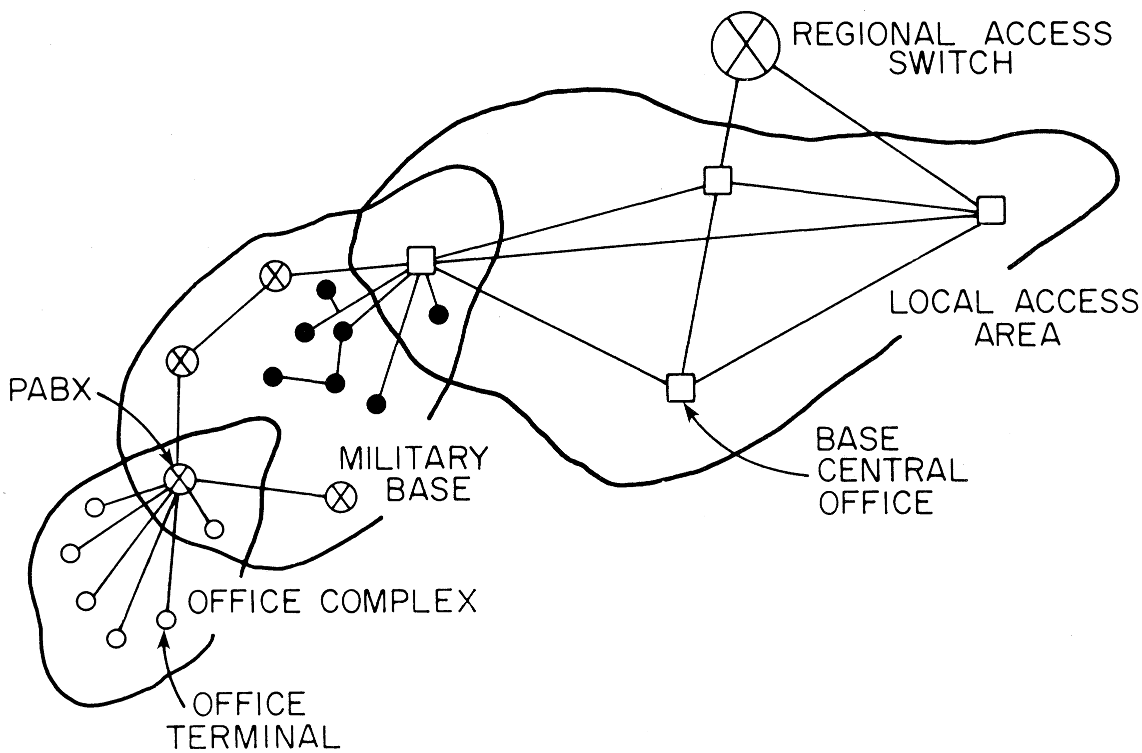


Figure 2-3. Hierarchical network configuration for local access.

The user terminal is the input/output device which generates and accepts the traffic handled by the network.

The types of terminals, their geographic distributions, and the volume and kind of traffic generated by each terminal are important factors in any network design. Terminals are discussed in the following section.

2.2.1. Types of Terminals

There is no foreseeable limit to the types and variations of terminals. Furthermore, new types are continually being introduced. They range from the traditional voice terminals consisting of a telephone handset with rotary or pushbutton dial; to keyboard and printer terminals for teletypewriting and computer access; to visual display terminals with cathode ray tube or other optical readout. Many terminals incorporate microprocessors and memory. Software programs add useful intelligence and processing power to such "smart" terminals. A program addition can change the character of the terminal to meet changing communications requirements or to adapt to new applications.

The telephone handset is the most common terminal in use today. There are over 300,000 handsets in use by the military now and the number is not expected to decline. Over 500,000 telephone terminals with about 1 million telephone instruments (including extension telephones) are projected to be in worldwide use by all US military services in the mid 1980's (Wagner, 1977). Data terminals are experiencing their own relatively rapid growth and could, over the same period, approach 50,000 terminals worldwide. Note that this still is only 10% of the telephone terminal population.

The DCA has projected the number of terminals of different types for the continental United States (CONUS) and overseas as shown in Table 2-1. The average duration of the different messages generated by these terminals ranges from several hours for bulk data transfers, to less than a minute for voice calls, and to a few seconds for interactive data transfers.

Table 2-1. Projected Number of Military Terminals for Various Types of Traffic

Traffic Type	Number of Terminals	
	CONUS	Overseas
Voice	400,000	100,000
Interactive data	13,500	2,500
Computer data	2,250	250
Narrative	6,000	4,000
Facsimile	3,300	1,200

2.2.2. Terminal Distributions and Densities

The number of terminals in the military base environment varies with the size and type of base. Projections made by Nesenbergs and Linfield (1976) for the types and numbers of terminals at Ft. Huachuca, Arizona, and Ft. Monmouth, New Jersey, (and its environs) are given in Table 2-2. These projections are for the 1980 to 1985 time frame.

Ft. Huachuca is a U.S. Army base located within a 15-square kilometer service area. The total population of military and civilian personnel is 10,000. The base is headquarters for the U.S. Army's Communications Systems Agency. The population density, which reflects terminal density is not uniform over the 15 km^2 service area. In fact, 80% of the base residents occupy a core area of about 4 km^2.

Ft. Monmouth and the surrounding access area (including Camp Woods, Camp Evans, the CERCOM building and other nearby installations) is the headquarters of the U.S. Army Communications and Electronics Readiness Command and the Communications Research and Development Command (CORADCOM). Other tenants include CSA, the Satellite Communications Agency and elements of the Electronics Research and Development Command and the Aviation Research and Development Command (AVRADCOM). The military, civilian and dependent population of this access area is approximately 15,300 with about 10% being military personnel.

Table 2-2. Number and Types of Terminals Projected for Two Military Base Environments

Terminal Type (nominal rate)	Estimated No. of Terminals (1980 to 1985)	
	Ft. Huachuca	Ft. Monmouth
<u>Telephones</u>		
•Analog (4 kHz)	5000	8000
•Digital (64 kb/s)	500	1000
•Video (6 Mb/s)	5	5
<u>Computer Access</u>		
•Low speed (450 b/s)	34	50
•Med speed (3.6 kb/s)	16	20
•High speed (48 kb/s)	5	10
<u>Teletype</u>		
•Low speed (300 b/s)	50	100
•High speed (4.8 kb/s)	20	20
<u>Facsimile</u>		
•Low speed (9.6 kb/s)	6	10
•High speed (56 kb/s)	2	5
•Color (1.5 Mb/s)	2	5
<u>Television</u>		
•CATV (1 way channels)	12	8
•CCTV (2 way channels)	2	4
•SSTV (2 way channels)	2	6

There are projected to be some 1500 access areas worldwide in the early 1980's. These areas will range in size from small (50 to 300 terminals) to medium (300 to 2000 terminals) to large (over 2000 terminals) (see Wagner, 1977). The terminal density in these areas may vary from less than 10 per square kilometer to over 10,000 per square kilometer. These densities have been classified by subregions in the local access area as low suburban, high suburban, low residential, high residential, low office complex, high office complex, and so forth, using average subscriber density profiles for each subregion (Nesenbergs and Linfield, 1976). This classification is useful for estimating local access network costs. Some resulting estimates are given in Section 2.4.4.

2.2.3. Traffic Generated

The volume, data rate, duration and delivery delay for traffic generated by various types of terminals is indicated in Table 2-3. Only nominal values are shown in this table for most of the parameters. Specific actual values may depart considerably from these nominal values. To be used, data rates must conform to the DoD and FTSC standard rates for synchronous transmission (FTSC, 1973). The standards basically establish rates at increments of 75×2^k bit per second for k less than 7, or 8000xk bits per second for k greater than 6. Thus the data rate generated by any given terminal may vary over a fairly wide range. Examples of this range for various types of terminals are plotted on Figure 2-4. A wide rate variation is indicated for digital voice terminals. The voice digitization rate depends on the process employed. At the higher rates (e.g., PCM and DPCM) the voice quality is generally higher for a given bit error rate over the channel. Gold (1977) describes how a network could be designed to adapt to these different digital rates and thereby alleviate congestion. This is accomplished by slowly reducing the voice digitization rate, with a corresponding slow degradation in quality, as the traffic load on the network increases. A disadvantage is the increased complexity of network control.

Table 2-3. Characteristics of Traffic Generated by Various Types of Terminals

	Volume	Digital Rate (one-way)	Call Duration	Delivery Delay
<u>Voice</u>				
PCM	continuous bits	64 kb/s	min	<200 ms
LPC	continuous bits	2.4 kb/s	min	<200 ms
<u>Data</u>				
Data Base Update	10^2 b/message	2.4-16 kb/s	s	secs to min
Interactive	10^3 b/message	150 b/s-9.6 kb/s	hrs (Bursts in s)	s
Query/Response	10^4 bits/ transaction	150 b/s-9.6 kb/s	s to min	<1 s
Bulk	10^5-10^8 bits/ transaction	100 kb/s	min to hrs	min to hrs
<u>Narrative</u>				
Alpha Coded Text	$3x10^4$ bits/page	75 b/s-9.6 kb/s	s to min	min to hrs
Text Editing	10^3 bits/page	75 b/s-9.6 kb/s	s to min	s
<u>Facsimile</u>				
No Gray Scale	$3x10^5$ bits/page	4.8 kb/s	min	min to hrs
Half Tone Photo	$3x10^6$ bits/page	9.6 kb/s	min	min to hrs
Color	10^7 bits/page	1.5 Mb/s	min	min to hrs
<u>Video</u>				
Picture Phone	continuous	6.3 Mb/s	min	<200 ms
Color TV	continuous	30 Mb/s	hrs	s
Slow Scan TV	continuous	100 kb/s	min	min

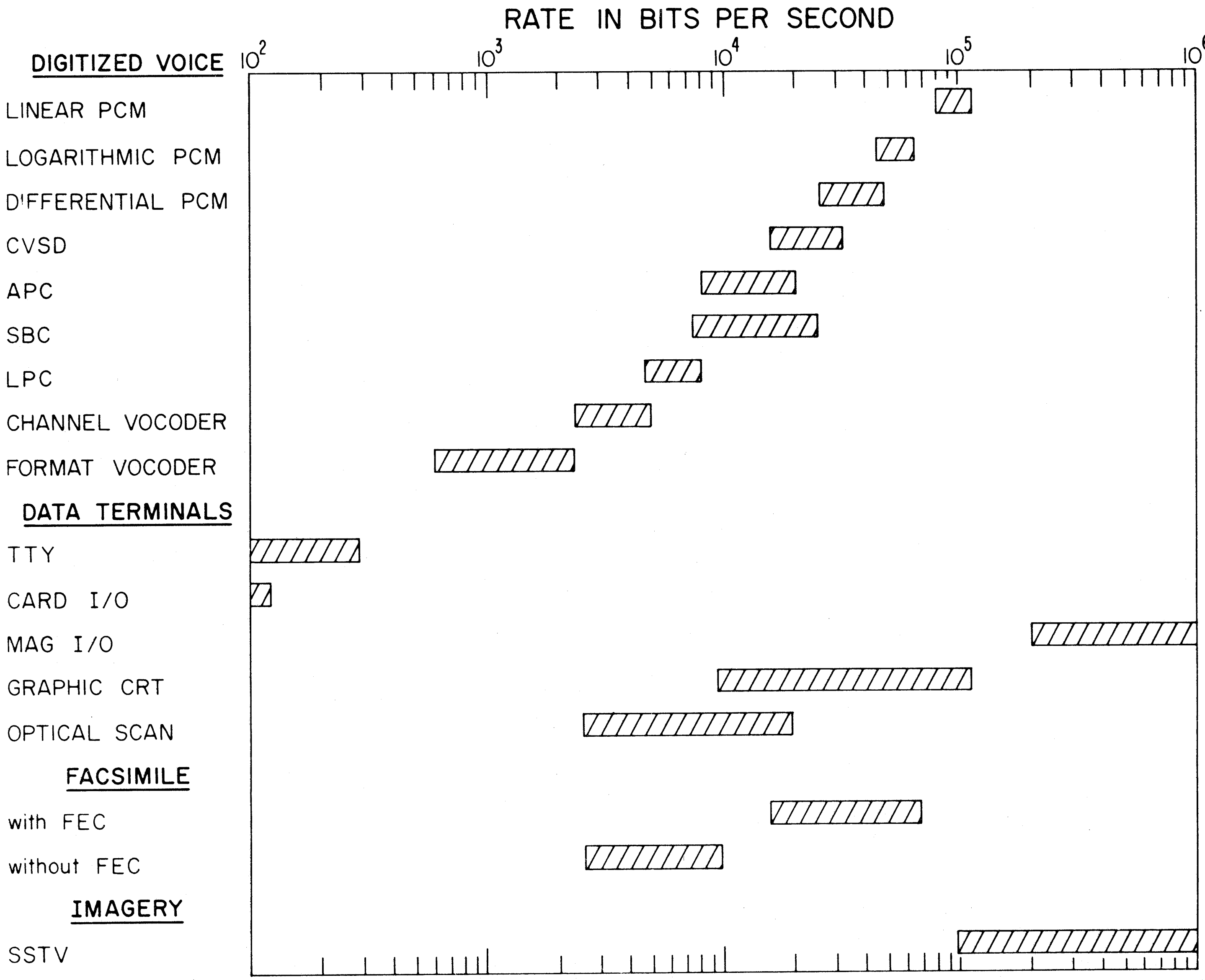


Figure 2-4. Traffic rates generated by various types of terminals.

Similar approaches with similar advantages and disadvantages could be used with other types of terminals and traffic.

2.2.4. Traffic Classifications

It is apparent from Table 2-3 that traffic can be classified according to various characteristics. Three classifications are generally used; Class I for continuous traffic, Class II for burst traffic, and Class III for interruptable traffic. Caviello and Vena (1975) characterized each class by examining a variety of terminals including man/man, man/machine and machine/machine operation. Their results are summarized in Table 2-4. They note that the major distinction between Class III and Classes I and II is one of priority and service level, rather than real technical considerations. Thus, although Class I and Class II traffic are handled differently in a network using circuit and packet switching, the Class III traffic can normally be handled by either type of switch depending on specific situations. Different switching networks and their traffic handling capabilities are discussed in the next subsection.

2.3. Nodes

The nodes of a network may serve both switched and dedicated subscribers. Nodal points may contain multiplexers, concentrators, and switches in various combinations, as well as link interface and signaling equipment.

A multiplexer combines the traffic from many channels onto a few high-speed channels. Each of the few channels is required to carry no more traffic than the individual input channels. Fixed rules determine how channels are multiplexed.

A concentrator, in contrast, is called an adaptive, a programmable, or an intelligent multiplexer. It observes the intermittent nature of traffic on several input channels to provide a more constant flow of traffic on a single output channel. Unlike multiplexers, the single output channel cannot carry the sum of traffic of the concentrator input channels if all are busy simultaneously. Thus blocking can occur.

Table 2-4. Characteristics of Traffic Classes

Class I	Class II	Class III
Continuous	Burst	Interruptable
Real Time Information	Discrete Messages	Not Real Time
Connection Delay	Near Real Time	Long Delivery Delay
Fixed Delivery Delay	Short Delivery Delay	Error Controlled
No Error Control	Potential Error Controlled	Indefinite Lengths
Long Holding Time	Short Total Lengths	Non-Blocking
Blockable	Non-Blocking	Arbitrary Users
Compatible Users	Arbitrary Users	

Analog multiplexers and concentrators use frequency division multiplexing (FDM) whereas digital multiplexers and concentrators combine digital traffic in time slots by time division multiplexing (TDM). This TDM technique is the only practical method to use in an all-digital network, since D/A and A/D conversions can thus be avoided.

2.3.1. Switching

Two switching concepts may be employed at a node; a circuit switch and a store-and-forward switch. The basic differences are illustrated in Figure 2-5. In a circuit-switched network the switch establishes a direct connection between terminals. This connection is maintained for the duration of the information transfer phase. When the phase is finished, the connection is terminated. The information transfer takes place essentially in real time and is completely transparent to the user. Interconnecting cross points are either metallic or electronic for the space division circuit switch. A "virtual" circuit switch employs time division switching. Logic gates and a high speed memory provide the time division switching in essentially real time.

The control of a circuit-switched network for both time and space division switching requires a signaling system to establish, maintain and terminate the continuous communications path between the terminals. Circuit switching is ideally suited for the transfer of user information having a relatively long duration and requiring real-time delivery, such as for voice, video and facsimile (i.e., Class I traffic).

2.3.2. Store-and-Forward Switching

There are two forms of store-and-forward switching: message switching and packet switching. Message switching involves formatting the information to be transferred into long blocks of data, called messages. Control information is added to each block by the terminal before forwarding the block to the switch. Each block is stored at the switch until an end-to-end transmission path is available for use. In contrast to circuit

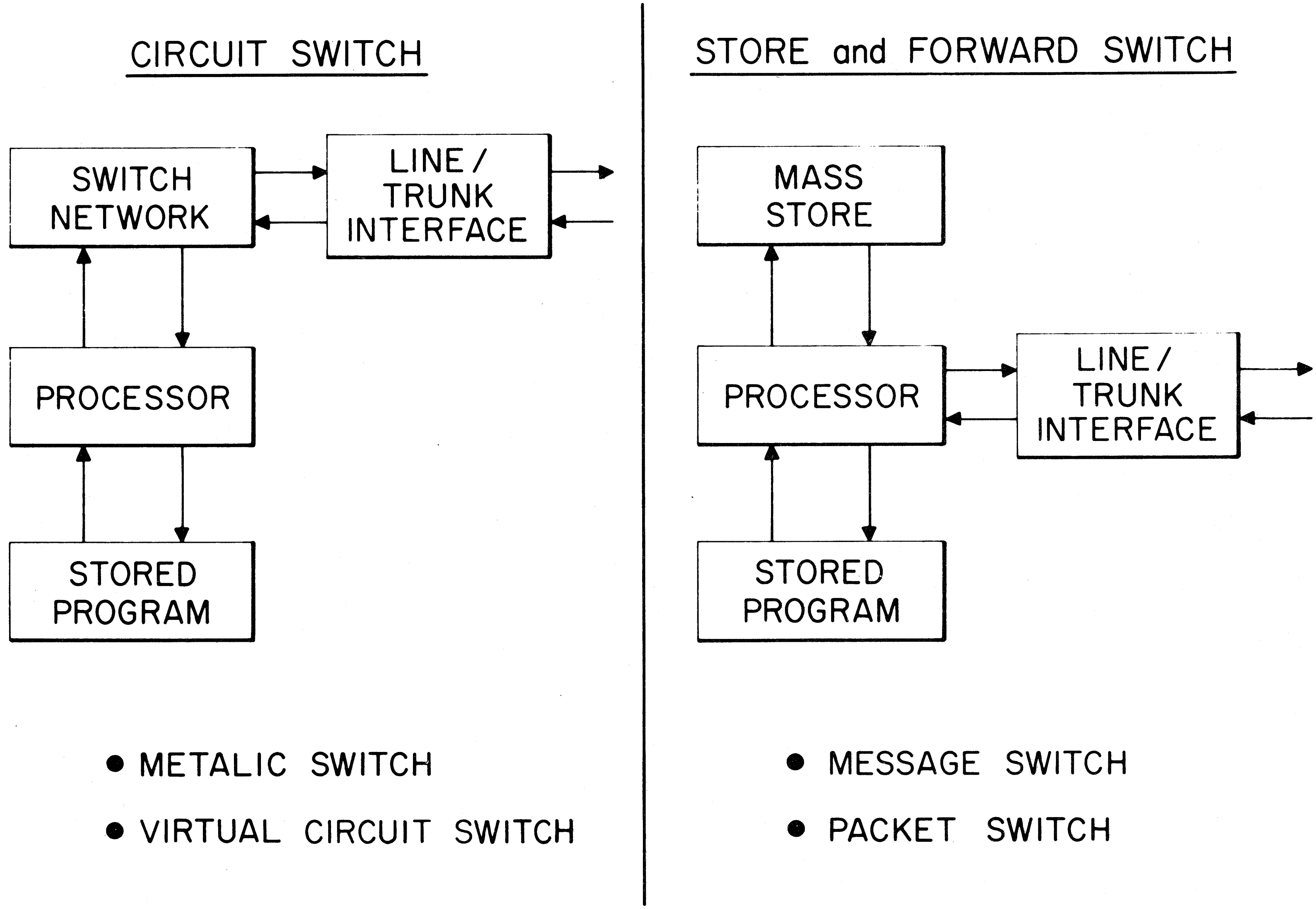


Figure 2-5. Basic switching concepts.

switching, network resources are not dedicated to specific users but are shared by all on an "as available" basis. A message switch must store and forward several relatively long blocks of data from many users. The maximum block length and number of blocks is limited by the storage capacity of the switch. Large processors are required to keep track of all the blocks flowing in the network. High capacity disks and tapes are used for storage. Message blocks are often delayed for long times while waiting to be forwarded. These random delays can be decreased by restricting the block length, redistributing the storage and controlling the storage access times. This leads to packet switching.

Packet switches handle shorter blocks of data than message switches. The maximum block length is usually limited to a few hundred or thousand bits. These blocks are called packets. The packet switch accepts and holds the packet only until a link is available. Packets may be forwarded through the network via varying routes. Smaller processors are required because each switch does not have to keep track of all the packets en route. The holding process is accomplished in core storage and buffers rather than on disks and tapes. Thus packet switching holds the information for a short time before forwarding, rather than storing the information for a long time.

A packet switched network can provide two kinds of service to the users - datagram service or virtual service. The basic difference is the order in which packets are received. Datagram service is more fundamental since in it individual packets are delivered independently and may have only an approximate relationship to the order of entry. Each packet must be fully addressed when entered into the network. Virtual service, in contrast, resembles a virtual circuit switch. At the destination, data packets are delivered in the same order they are entered into the network, even though they may be forwarded via different routes. A finite delay is involved depending on the length of the packets, the number of packets per message, and the number of links traversed (Roberts, 1976).

Various procedures called protocols are used to control terminals and data networks for store-and-forward switching. Several levels of protocol are used to control the terminals, the links and the switches. Control information called overhead is contained in each message block or packet. The amount of overhead required reduces the network efficiency. Some protocols at various levels are described in Section 5.

2.3.3. Switching Summary and Comparisons

Figure 2-6 summarizes the basic switching concepts and relates them to specific military applications. The AUTODIN II network (see Sec. 3) employs packet switching protocols which support both datagram and virtual service to the users. The AN/TCC-39 switch is being developed by the U.S. Army under the TRI-TAC program. This switch will incorporate both virtual circuit switching and message switching capabilities for the phase two secure voice (P2SV) network.

Table 2-5 compares the advantages and disadvantages of the circuit, message, and packet type of switches when used for data communications. The characteristics noted for each type of switch in Table 2-5 can also be compared with the characteristics of traffic classes in Table 2-4. It is apparent from this comparison that the circuit switch is ideally suited for some information transfers whereas store-and-forward switching is ideally suited for others. The circuit switch is inherently a two-way transmission switch with little or no delay. It is most suited for handling narrowband, lower speed transmissions which occur continuously and for long periods of time, i.e., Class I information.

The message and packet switches are inherently unidirectional. Data encounter relatively long delays when passing through message switches and relatively short delays through packet switches. The message switch is most suited for interruptable transmissions with long acceptable delivery times, i.e., Class

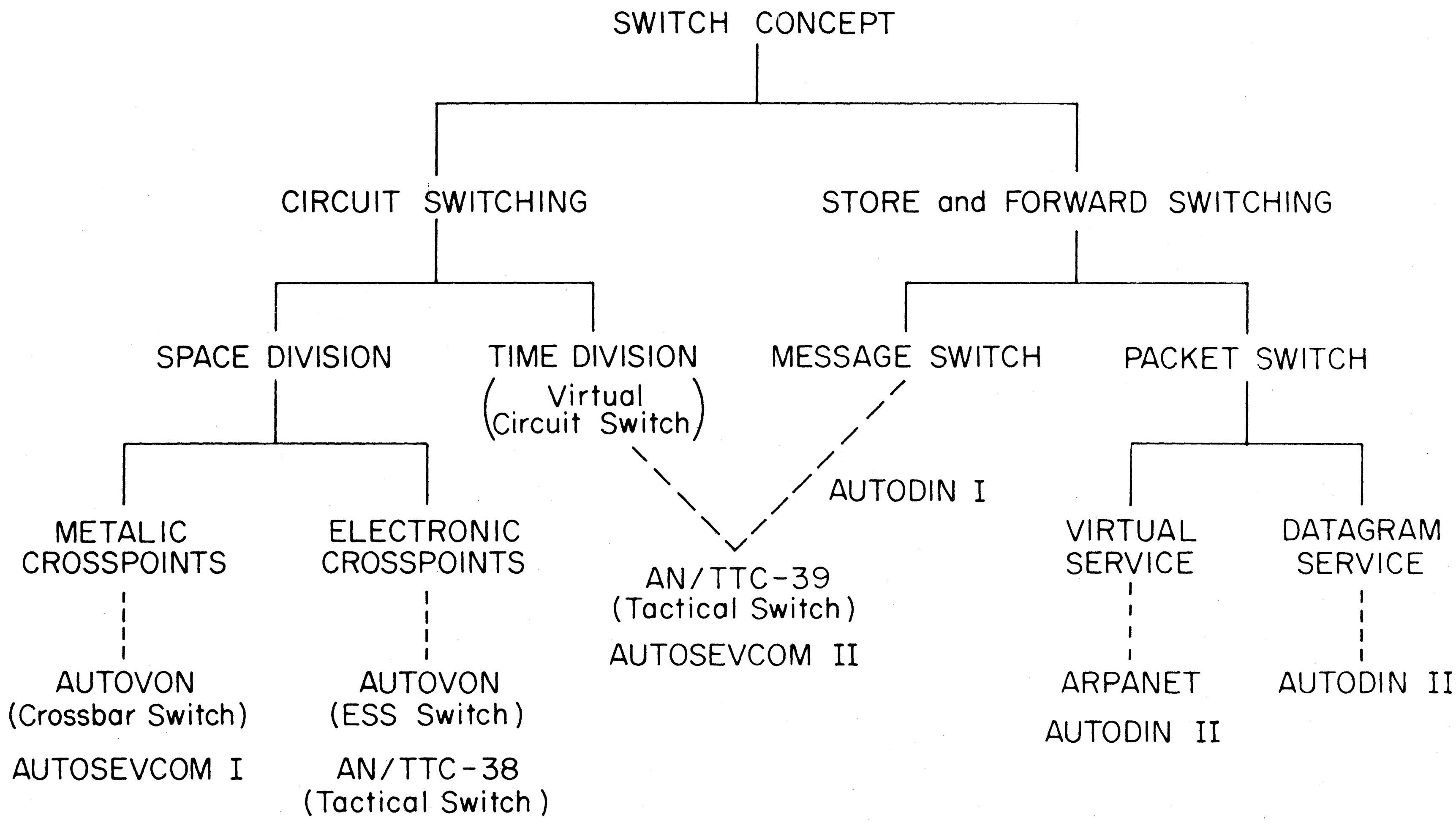


Figure 2-6. The military switching environment.

Table 2-5. Comparison of Three Basic Switching Techniques for Data Communication Networks

Circuit	Message	Packet
Advantages	Advantages	Advantages
Voice Compatible	Code and Speed Conversion	Short Delay
Common Procedures	Nonblocking	Rapid Exchange of Interactive Traffic
Fully Transparent	High Efficiency and Channel Utilization	Code and Speed Conversion
Disadvantages	Disadvantages	Disadvantages
Large Processing Burden	Large Processors	Many Small Processors
Requires Compatible Terminals	High Capacity Storage using Disks on Tapes	Core Storage/Buffers
Subject to Blocking	Variable Delay	Complex Routing and Control
	Poor Response to Interactive Traffic	May be Blocked with Short Delay

III data. The large storage capacity and processing capability of the message switch are substitutes for more elaborate and costly transmission facilities.

The packet switch is more suited to high data rate transmissions of information with burst characteristics, i.e., Class II traffic. Less storage capacity and processing capability is required in the packet switch. The holding delay is shared by all users and therefore the buffer storage at each switch need not be large.

Although different switching techniques provide better performance for different classes of traffic, this does not necessarily imply that different transmission and switching networks are required.

The three basic switching concepts are not mutually exclusive. A single network capable of handling voice and data could be comprised of both circuit and store-and-forward switching elements. The evolution of an integrated (voice/data) switch for potential military applications is described in Section 3.3.

2.3.4. Switching Control Systems

Since this report is concerned with network control, it is useful to examine the circuit switch and the store-and-forward switch from the control standpoint. The signaling system associated with the switch must perform three basic functions:

1. Interrogation which involves the transmission and the reception of control information.
2. Interpretation by a control unit to determine what, if any, action is to be taken by the switch.
3. Interconnection to complete the data transfer (either by operating the switch matrix or by forwarding data from storage, for example).

The interrelationship between these three functions is shown in Figure 2-7 for a circuit switch. In a modern electronic switching system, the control unit is part of the central processing unit (CPU). Associated with the CPU are several memories containing, among other things, the stored programs.

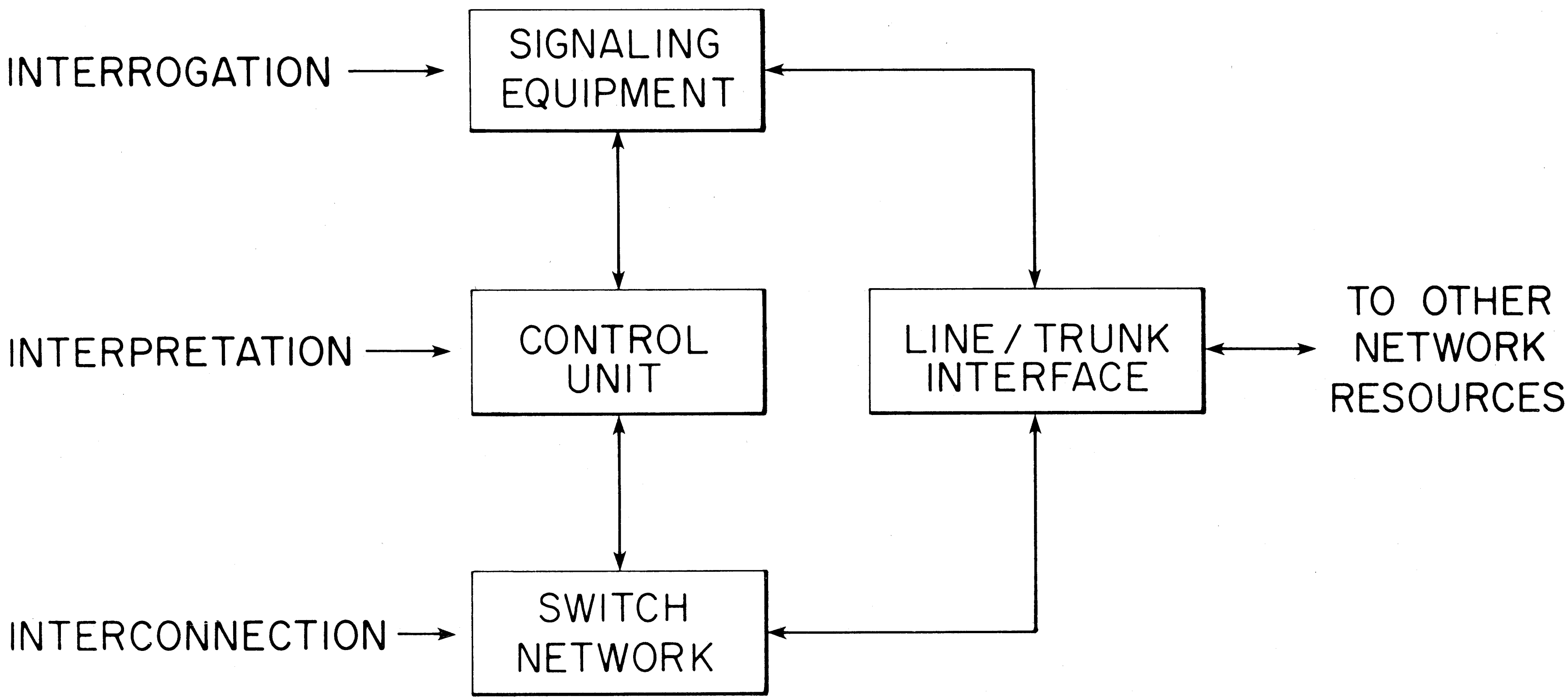


Figure 2-7. Interrelation between circuit switching functions.

Control information received by the signaling equipment is evaluated in the processor and, according to the program instructions, the switch matrix is activated. The switch matrix may use metallic reed relays or electronic semiconductors for switching cross points.

In a store-and-forward switch the processor itself makes the connection and no switching matrix is required. Data blocks with their associated control overhead are stored in memory or queued in buffers until the processor can interpret their destination, find an open route, and forward them.

The protocols for controlling data terminal equipment and link terminating equipment on a packet-switch network have recently been standardized. The virtual service protocol denoted X.25 by the CCITT is recommended as the international standard for operating in the packet mode on public switched networks. New standards are being proposed for datagram service. These protocols and others used to control store-and-forward switches are described by Folts and Cotton (1977) and reviewed in Section 5.2.

2.3.5. Switching Technology Developments

The circuit-switching technology for telephony has undergone considerable change since its inception. Table 2-6 indicates some of the major developments which have occurred at approximate 20-year intervals beginning around 1880. The control unit has progressed from completely manual to fully automatic, the signaling from handcranked magnetos and aural addressing to push button dialers and digital signals over common signaling channels; and the circuit switch control from operators with plugs and jacks to microprocessors with stored programs.

Packet switched networks are a more recent introduction. For example, ARPANET, conceived by the Advanced Research Projects Agency, has undergone only a few changes since its operation began in 1969. The network is designed to interconnect dissimilar computers throughout the United States, allowing users and programs at a computer center to access and interactively use

Table 2-6. Network Switching, Control and Signaling Development

Circa	Interconnection		Control		Signaling	
	Concept	Switching Technology	Operation	Type	Supervision	Addressing
1880	Analog	Plug & Jack	Manual	Direct	Magneto	Aural
1900	Analog	Step-by-step	Mechanical	Direct	DC	Pulse DC & Reverting
1920	Analog	Panel	Motor Driven	Indirect	DC	PCI
1940	Analog	Cross Bar	Relay	Common	DC+SF	MF
1960	PAM	Electronic	Wired Logic	Common	SF	MF
1970	PCM	Micro-processor	Software	Stored Program	Digital	Digital

facilities at another center. Feinler (1976) and Feinler and Postel (1976) provide a description of ARPANET resources and protocols, respectively. Most of the original development work on packet switched networks has resulted from research activities on the ARPANET. The AUTODIN II network follows many of the same concepts and is essentially a second generation packet switched network (BBN, 1978).

2.4. Links

Military and commercial networks currently use various combinations of landlines, radio, and satellites for the links between nodes. These links in large transcontinental networks constitute nearly half of the investment cost of the network. Major cost elements are the wires and cables in the landlines, radio equipment associated repeaters, and multiplexing equipment. About one-third of this investment cost for transmission facilities is for long-haul facilities and two-thirds for local access, primarily the lines to the terminals (or local loops) (Members of the Technical Staff, 1978).

It is largely due to this high percentage cost of transmission facilities that networks are designed to share their transmission resources efficiently in order to become economically attractive.

2.4.1. Transmission Categories

Transmission facilities may be categorized into 1) local loops for terminal connections, 2) short-haul facilities for regional access, and 3) long-haul facilities for interregional access.

A typical local loop for a telephone terminal consists of a twisted pair of No. 26 gauge insulated copper wire. The average length of a typical loop is about 2 km and the maximum length is about 8 km. For analog voice transmissions over such a loop, loading coils are installed approximately every 1.5 km to improve the frequency response.

Short-haul facilities typically employ many pairs of twisted wire in shielded cables. Frequency division multiplexing allows several analog voice channels to be carried on each pair. For example the type N carrier uses double sideband amplitude modulation and frequency multiplexing to yield a capacity of twelve voice channels. The frequency of these 4 kHz multiplexed channels spans the spectrum from 148 kHz to 196 kHz. Short-haul facilities operate over distances of a few hundred kilometers.

A digital short-haul facility capable of handling 24 voice channels was introduced by the industry in 1962. Called the T-carrier, the facility uses pulse code modulation and time division multiplexing to provide a true digital transmission system. Since in this report we are concerned with digital network elements and the control of these elements, the T-carrier system will be described in more detail in a subsequent paragraph.

Long-haul facilities are designed for extremely efficient transmission over intercontinental distances. Unlike the local loop, or short-haul systems, multiplexing hierarchies permit the transmission of thousands of voice channels via coaxial cable, line-of-sight microwave channels and via satellites. The satellite links employ multiple transponders each capable of handling several thousand voice channels. Troposcatter radio links are also used for global access.

Pertinent characteristics of several short-haul and long-haul transmission facilities commonly used by the telephone industry for voice and data transmission are listed in Table 2-7.

2.4.2. Full-duplex and Half-duplex Transmission

It is common practice to denote a channel capable of unidirectional, either-way transmission as a half-duplex or 2-wire channel. Two such one-way channels are required for full-duplex or 4-wire operation. The actual channel separation may not be a physical separation. For example, two channels, one for each direction, may be carried in the same wire pair using frequency division multiplex. A digital transmission facility operating at

Table 2-7. Transmission Facilities in Common Use by U.S. Communications Industry

	Carrier Designator	Modulation Scheme	Transmission Media	Carrier Channel Capacity (one-way voice ckts)	Total System Capacity (two-way voice ckts)	Repeater Spacing in km	Total System Length in km
Short-Haul Facilities	O	AM	Open wire	16	16	13	300
	N	AM	Wire cable	12	12	13	300
	ON	AM	Wire cable	20	20	80	250
	T-1	PCM	Wire cable	24	24	1.5	80
	T-2	PCM	Wire cable	96	96	1.5	800
	T-4	PCM	Coaxial cable	4,032	4,032	1.5	800
Long-Haul Facilities	K	AM	Wire cable	12	48	25	2,400
	TD	FDM/FM	LOS microwave	1,200	2,000	50	6,000
	TH	FDM/FM	LOS microwave	1,860	11,600	50	6,000
	L-1	SSB/SC	Coaxial cable	600	600	13	6,000
	L-3	SSB/SC	Coaxial cable	1,860	16,740	6	6,000
	L-4	SSB/SC	Coaxial cable	3,600	32,400	3	6,000
	L-5	SSB/SC	Coaxial cable	10,800	97,200	1.5	6,000
	EMT	FDM/FM	Troposcatter	120	120	300	5,000
	INTELSAT IV	FDM/FM	Satellite	300 per transponder	6,000	N/A	13,000

baseband is normally half duplex. Two digital channels are usually required to operate full-duplex. Note however that an analog voice system operating at voice frequency is full duplex and operates over two wires. Simultaneous two-way data transmission over this two-wire loop may require an additional pair of wires for full-duplex operation. The voice network used for AUTODIN is 4-wire.

Any given transmission facility may employ several full- or half-duplex channels. Individual coaxial cables, for example, are normally operated as one-way systems. The number of coaxial cables in a bundle of cables may vary from 2 to 20, each cable capable of handling many frequency multiplexed voice channels. The L-5 system, for example, contains 20 coaxial cables in a single large bundle. Since each coaxial cable can carry 10,800 voice channels, the total two-way system capacity is 108,000 channels. One pair of cables, however, is reserved for a spare and the total system capacity shown in Table 2-7 is therefore 97,200.

The satellite system INTELSAT IV listed in Table 2-7 contains multiple transponders in the satellite each capable of handling 300 one-way voice channels. Since there are 40 transponders, the total number of two-way voice channels carried by the satellite is 6000.

The channel capacity in terms of one-way voice channels and the total system capacity in terms of two-way voice channels are indicated in Table 2-7 for long-haul facilities.

2.4.3. T-carrier Systems

As mentioned previously, digital T-carrier systems capable of handling 24 voice channels were introduced in 1962 using pulse code modulation (PCM) for voice channels (James and Muench, 1972). The initial system, denoted as the T-1 carrier* used D-1* channel banks at the facility terminations to digitize

*The T-carrier designations T-1, T-2, etc. and channel bank designations D-1, D-2, etc. are Western Electric Co. designations that have become generic terms for almost the entire industry.

and multiplex analog voice channels. Initially the digitization process was accomplished in a D-1 channel bank. Each voice frequency was sampled 8000 times per second and quantized to 7 bit accuracy. An eighth bit was added for signaling purposes. Later channel banks (e.g., D-2 and D-3) use 8 bit quantization and signaling bits were borrowed periodically. This 8-bit "encoding" is used today in D-1 channel banks for T-1 transmission where there are several links in tandem. A non-linear quantization process is normally used to maintain a relatively constant signal-to-quantizing distortion ratio over a wide range of talker levels. Compression techniques quantize the low-level samples with small steps and high-level samples with larger steps.

Quantizing 8000 samples per second with 8 bit accuracy yields a 64 kb/s bit rate per voice channel. In the T-1 carrier system, 24 of these channels are multiplexed in 8-bit groups into a frame. One bit is added for synchronizing purposes so the total frame contains 193 bits. The T-1 line format is shown in Figure 2-8. A bipolar pulse train is used, where zero bits are denoted by an absence of pulses and one bits are alternately positive and negative pulses. Since there are 8000 frames per second, the total bit rate is 1.544 Mb/s. The multiplexing process can be accomplished either before or after the 8-bit encoding process as illustrated in Figure 2-9. Most channel banks multiplex before so that common encoder/decoder can serve all channels. This process, however, precludes the possibility of remoting the process to a single terminal, i.e., a digitized telephone.

The T-carrier can be multiplexed to higher levels, to carry more voice channels more effectively over greater distances. The hierarchal levels common to the industry are shown in Table 2-8. The T-3 carrier is only used internally at a switch center.

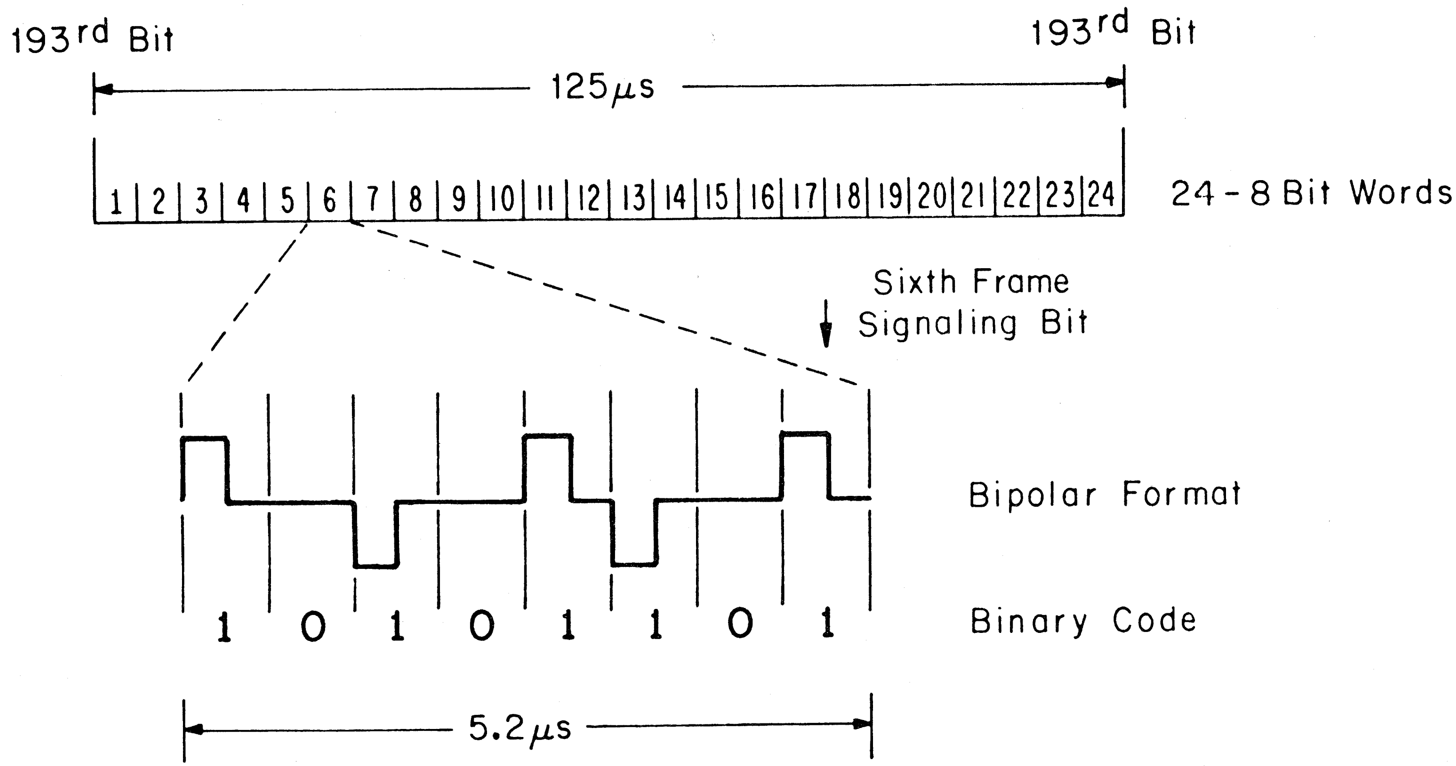
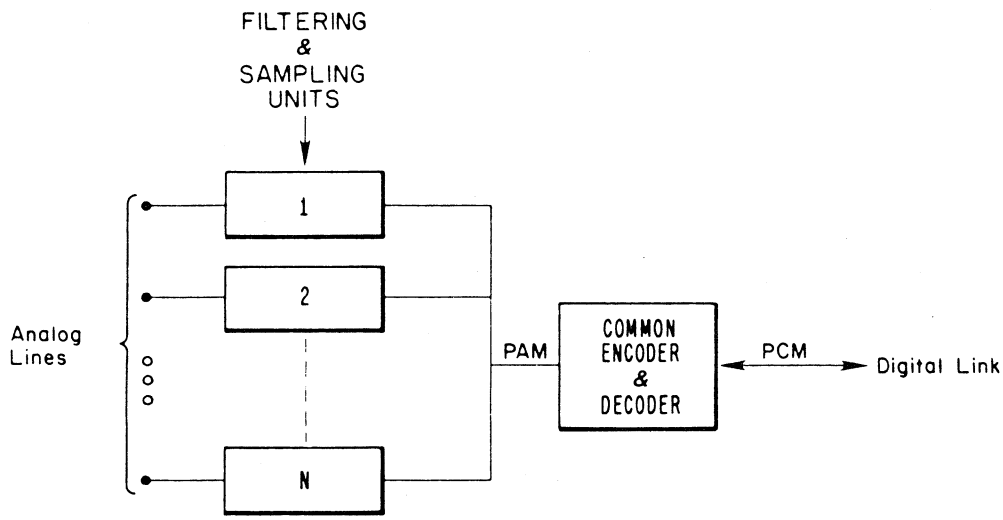
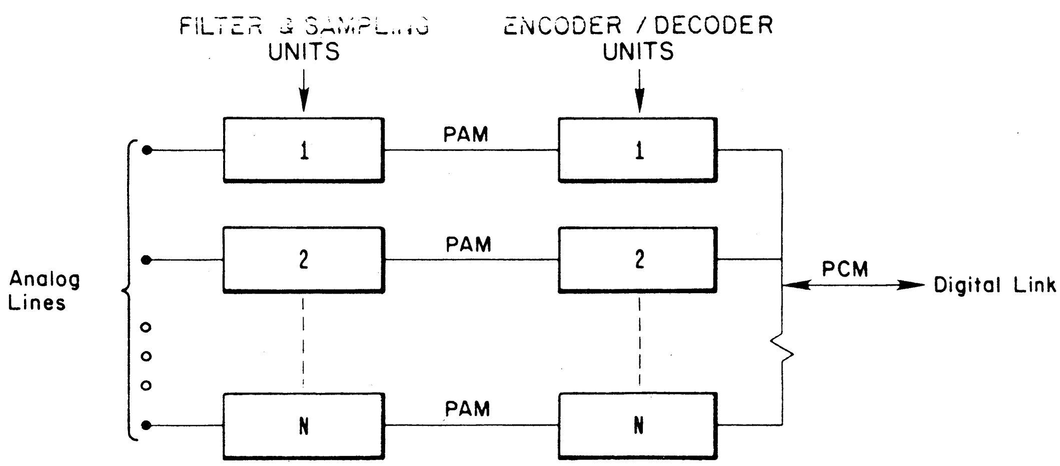


Figure 2-8. Multiplexed PCM framing and signaling format for T-carrier system.



a) Before Coding



b) After Coding

Figure 2-9. PCM multiplexing techniques.

Table 2-8. Digital Multiplexing Hierarchies Used in North America

Carrier	T1 Hierarchy	Voice Channels	Rate (Mb/s)
T1	1	24	1.544
T1C	2	48	3.152
T2	4	96	6.312
T3	28	672	44.736
T4	168	4032	274.176

The specifications for PCM multiplexors for the Digital Radio and Multiplex Acquisition (DRAMA) system has a somewhat different hierarchy. The first-level multiplexor performs the analog-to-digital conversion and time-division multiplexing for three, six, twelve and twenty-four voice channels. The digitization process for the 24 channel increment is essentially the same as T1, i.e., 1.544 kb/s. The three, six and twelve channel modes are used for DSCS and the twenty-four channel mode for DCS. The output parts from several first-level multiplexors can be combined in a second-level multiplexor as follows:

Number of 1.544 Mb/s Ports	Number of Voice Channels	Data Rate (Mb/s)
2	48	3.168
4	96	6.336
6	144	9.504
8	192	12.672

Digital data can also be interleaved by replacing voice channel cards in the first-level multiplexor. Data bit rates specified are 0 to 20 kb/s and 50 kb/s asynchronous and 56 kb/s, 64 kb/s and 128 kb/s synchronous.

The DCS system in central Europe is being upgraded to form a Digital European Backbone (DEB). Terrestrial microwave links operating in the frequency band from 7,125 MHz to 8,400 MHz are phase-shift modulated to convey the 12.672 Mb/s rate corresponding to two T2 lines or 192 digital voice channels.

In Europe and in many foreign countries, a different format is used for commercial digital transmission facilities although the basic PCM concept is the same. The European system uses a 32 channel system and format adopted by the Consultative Committee for International Telephone and Telegraph (CCITT). The thirty-two 8-bit channels occupy a frame 256 bits long. No framing bit is added for synchronization, so the total bit rate is 2.048 Mb/s. A 16-frame cycle makes up a multiframe. Multiframes occur at a rate of 500 per second. Two of the 32 channels are reserved for signaling and synchronization purposes. Therefore, there are 30 digital voice channels available. The signaling system uses channel 16 for control information by allocating 4 bits per multiframe for each of the 30 voice channels. Since multiframes occur at a rate of 500 per second, each of the 30 voice channels has a 2000 b/s signaling rate. If common channel signaling is used, channel 16 provides a 64-kb/s signaling rate just like any other voice channel. The standard CCITT format is shown in Figure 2-10.

Comparisons between the CCITT standard system and the T-1 carrier used in North America are given in Table 2-9.

A tutorial discussion of these basic digital transmission concepts is given in CEEIA (1977). A detailed description of T-carrier systems is given in Members of Technical Staff (1971). The international recommended standard is found in CCITT (1977e).

2.4.4. Upgrading Local Loops

The outside loop plant on military bases is comprised mostly of thousands of pairs of pulp-insulated copper wires bundled into pressurized lead sheath cable. Replacement of these outside facilities is an expensive undertaking (CEEIA, 1977). The strategy for upgrading and digitizing this local access to terminals is important in determining overall future costs of the military networks.

Whenever possible, one can utilize the existing loop plant rather than replace it. A twisted pair of No. 26 copper wire is capable of handling unidirectional digital traffic. However, if

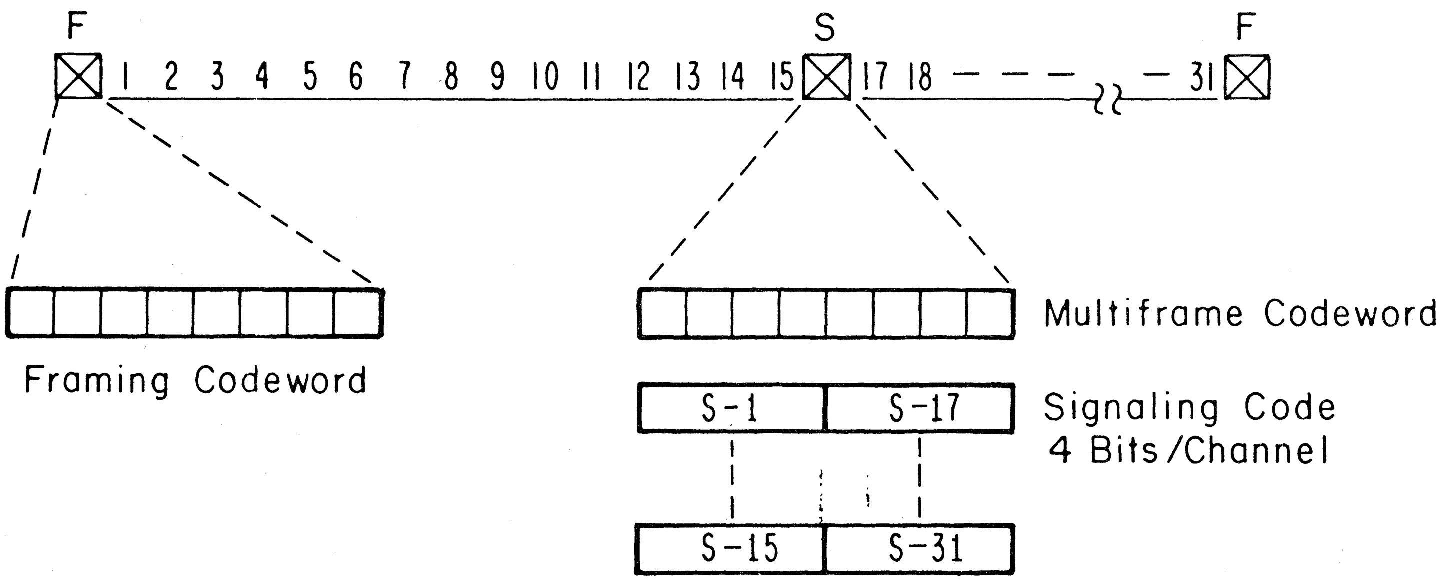


Figure 2-10. Multiplexed PCM framing and signaling format for CCITT recommended digital carrier.

Table 2-9. Comparisons Between T-1 Carrier and CCITT Recommended Digital Carrier

Comparison Item	T-1 Carrier System	CCITT Recommended Carrier System
Sampling rate	8000/s	8000/s
Modulation	8 bit PCM	8 bit PCM
Companding	μ 255 law	A-law
Bits/time slot	8	8
Bits/frame	193	256
Time slots/frame	24	32
Frames/multiframe	12	16
Bit duration	0.6479 μs	0.4882 μs
Time slot duration	5.181 μs	3.9056 μs
Frame duration	125 μs	125 μs
Bit rate	1.544 Mb/s	2.048 Mb/s
Voice channels/frame	24	30
Bit rate/channel	64 kb/s	64 kb/s
Signaling scheme	1 bit borrowed in 6th frame	16th channel dedicated
Signaling rate	1.300 b/s/per channel	2000 b/s per channel
Synchronizing scheme	193rd bit	First character dedicated

the loading coils are spaced at 1.5 km intervals, and if the length of the loop exceeds 5 kilometers, then the data rate is limited to a few hundred bits per second. The data rate can be increased to 50 kb/s, and possibly more, depending on the distance, by removing the loading coils. This could permit a half-duplex digital voice channel to be transmitted over existing pairs of wires. When the loading coils are replaced with digital repeaters, it is possible for each wire pair to handle T-carrier at 1.544 Mb/s and in some cases T-1C carrier at 3.152 Mb/s.

Since many digital systems require full-duplex transmissions, two pairs would normally be required for the T-carrier type of link.

Another loop plant strategy involves the gradual replacement of wire pairs with new technology. Suggested replacement schemes include packet radio, packet satellite, lasers, fiber optics, and coaxial cables.

A recent study by Nesenbergs and Linfield (1976) has examined the costs of coaxial cable local distribution systems in some detail. Cost estimates were made for a replacement of the outside plant at Ft. Monmouth and Ft. Huachuca with either buried or overhead installations, using either star or loop topologies. The cost was found to vary as a function of system components, topology and method of control, as well as the method used to install the network.

The results for one type of installation, using buried coaxial cable with distributed control, at these two sites are shown in Table 2-10. Inside plant costs include local switching centers, concentration hubs, modems, multiplexers, and the link, but exclude installation and terminal costs. The outside plant costs include the coaxial cable for trunks, feeders, drops, amplifiers, and bridges as well as installation costs.

Table 2-11 compares the estimated total cost for a 6000 terminal, local-access network at Ft. Huachuca, assuming different types of installation (buried vs. overhead), different

Table 2-10. Cost Comparisons on a Per Drop Basis (Buried Network with Distributed Control)

	Ft. Huachuca	Ft. Monmouth	
	Projected	Current	Projected
No. of Terminals	6,000	5,300	10,470
Area Served	9 km^2	10 km^2	10 km^2
Terminal Density	667/km^2	530/km^2	1,047/km^2
Bandwidth Required	43 MHz	45 MHz	156 MHz
Bandwidth/Terminal	7 kHz	8.5 kHz	14.8 kHz
Plant Cost Estimates			
•Outside Plant	$1,480K	$1,304K	$2,247K
•Inside Plant	1,934K	2,033K	5,505K
•Total Cost	$3,414K	$3,337K	$7,752K
Cost Per Drop			
•Outside Plant	$ 245	$ 246	$ 215
•Inside Plant	322	383	525
•Total Cost/Drop	$ 567	$ 629	$ 740

Table 2-11. Estimated Cost for 6000 Terminal Local Access Network at Ft. Huachuca (in 1976 dollars)

	Underground Installation				Overhead Installation			
	Star		Loop		Star		Loop	
	Centr.	Distr.	Centr.	Distr.	Centr.	Distr.	Centr.	Distr.
Total Cost ($M)	5.4	4.0	4.4	3.4	4.6	3.2	3.5	2.5
Cost/Terminal ($K)	0.90	0.68	0.73	0.57	0.76	0.52	0.59	0.42
Cost/KBPS/Term. (¢)	0.78	0.58	0.63	0.49	0.66	0.45	0.50	0.36

topologies (star to loop), and different control strategies (centralized vs. distributed).

The results of the Nesenbergs and Linfield study indicated that overhead cable installations are less expensive than underground cable installations. The star configuration appears more expensive than the loop configuration and centralized control more expensive than distributed control.

The cost figures given in Table 2-11 do not include any terminal costs. Addition of terminal costs is expected to add another 20% to 25% to the total cost (excluding any large and expensive computer terminals).

In general, it has been found that the local access network cost per terminal drop varies also as a function of terminal density and channel capacity or bandwidth per terminal. The general cost relationship as a function of these parameters is shown by the curves in Figure 2-11. The channel capacity indicated is the number of Hertz required per terminal and assumes 32 kb/s and 2 b/Hz (i.e., 16 kHz) is required for a single voice channel. These curves are used in Section 3.2 below to estimate the replacement costs for local access networks.

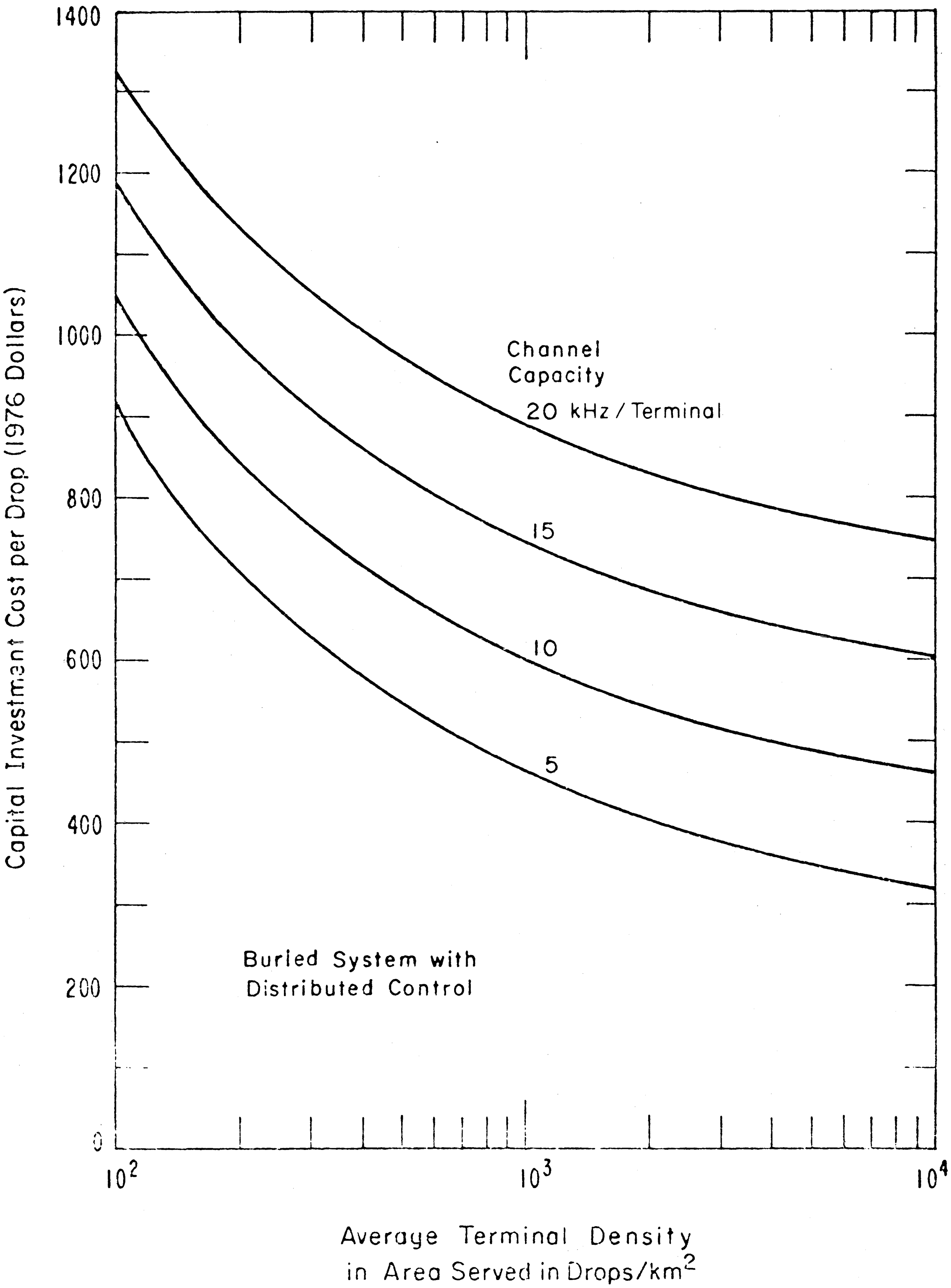


Figure 2-11. Generalized cost curves for local distribution networks.

3. THE MILITARY SWITCHING ENVIRONMENT

Telecommunications networks used by the DoD have special requirements and must meet specifications which are often more stringent than those required of commercial networks.

In this section, the present differences are noted so that the unique switching and signaling aspects of future military networks can be predicted. This is necessary for understanding the issues involved in network control. These issues and some advanced signaling systems which may be used to resolve them are discussed in Section 5. In this section we describe pertinent military networks, their distinctive characteristics, and projections of what future networks may be in the 1985 to 1990 time frame.

3.1. Examples of Military Networks

Military networks can be divided into three major categories, global-strategic, tactical, and non-tactical. Each of the three has unique features to meet different requirements. Examples of networks used by the military are given below. Unique characteristics are defined in Section 3.2.

3.1.1. Strategic Networks

The global-strategic Defense Communications System (DCS) is managed by the Defense Communications Agency (DCA) for the combined military services, Army, Navy and Air Force. Major components of the current network, DCS I, include the Automatic Voice Network (AUTOVON I), the Automatic Digital Network (AUTODIN I) and the Automatic Secure Voice Network (AUTOSEVOCOM I). These networks sometimes share common transmission facilities which include satellite links, high frequency and troposcatter radio links, line-of-sight microwave links, and terrestrial wire and coaxial cable links.

In addition, the DCS includes of several special purpose networks such as the packet-switched computer network, ARPANET, developed by the Advanced Research Projects Agency (ARPA). The once experimental ARPANET was transferred to DCA for operational use in 1975. Another DCA responsibility is the European Telephone System (ETS) which provides telephone service for U.S. Forces in Central Europe.

A two-way data network called SATIN IV will provide secure record communications between the National Command Authority (NCA) and the commander-in-chief of SAC (CINCSAC). The dialed AUTOVON network is the prime candidate for SATIN IV backbone because it has a greater survivability than AUTODIN, since the number of nodes is limited in AUTODIN.

A network for connecting medium and large automatic data processors is planned for the World-wide Military Command and Control System (WWMCCS). It is expected that this WWMCCS Intercomputer Network (WIN) will utilize the AUTODIN II network to connect 35 large- and medium-scale computer systems and remote terminals at 26 locations around the world (Kuo, 1977).

Some characteristics of current and projected AUTOVON, AUTODIN and AUTOSEVOCOM systems are given in Table 3-1. This information was obtained from a number of sources including Rosner (1973), Ochiogrosso et al. (1977) and Levine (1976).

3.1.2. Tactical

Tactical communications for all military services are being combined under a joint tactical communications program called TRI-TAC. Under TRI-TAC, the Army, Navy and Air Force are developing a new series of mobile, interoperable communications equipment for use by the tactical commanders in the 1980's. Each military service is assigned responsibility for certain system components. For example, the Army is responsible for developing the AN/TTC-39 switch along with other components. The switch incorporates both circuit- and message-switching capabilities and is designed to meet not only the tactical requirements but the switching needs for portions of the DCS strategic network.

3.1.3. Non-tactical

Non-tactical communications networks serve local military bases and are the responsibility of each military department and the base commander. Facilities provided include telephone, record

Table 3-1. Current and Projected Characteristics of DCS Backbone Networks

Network	Characteristic	Current (1978)	Projected (1985/1990)
AUTOVON	No. of Subscribers	5.0×10^5	1.0×10^6
	Call attempts/day	7.5×10^5	Unknown
	Switch type	Circuit	Hybrid
	No. of nodes		
	CONUS	65	43
	Overseas	16	20
	Access areas	1,000	1,500
	Traffic (Erlangs/hr)		
	Backbone (CONUS)	2,200	2,500
	Node	36	42
AUTODIN	Computers	250	2,500
	Remote Terminal	1,400	25,000
	Messages/day	3.5×10^5	1.2×10^7
	Switch type	Message	Packet
	No. of nodes		
	CONUS	9	24
	Overseas	8	7
	Backbone traffic		
	Average (b/h)	7×10^9	1.4×10^{10}
	Peak (b/s)	8×10^6	1.0×10^8
	Nodal traffic		
	Average (b/s)	2.5×10^5	5.0×10^5
	Peak (b/s)	1.0×10^6	1.25×10^6
AUTOSEVOCOM	Subscribers	1,400	10,000
	Terminal rates		
	Wideband	50 kb/s	9.6 kb/s
	Narrowband	2.4 to 9.6 kb/s	
	Nodes	AUTOVON	20
	Traffic	Unknown	Unknown

and computer data, video imagery such as facsimile and television (MITRE, 1976). Many of these services are presently provided with separate networks. Ultimately, a fully integrated multi-service system might be used, but only if found to be cost effective.

The U.S. Army Communications Command (ACC) is responsible for the communications on some 400 Army bases throughout the world. A Base Communications plan (BASCOP) is being developed which provides the system architecture for future upgrading of the base plant (see CEEIA, 1977). Networks for integrating multimode communications and electronic facilities to satisfy the Army's base-level information processing and transfer needs is also being evaluated under the Army Base Information Transfer System (ARBITS) program (see MITRE, 1976).

3.2. Distinctive Characteristics

The special requirements of military networks and certain characteristics which distinguish them from commercial networks can have a considerable impact on the network management and control. These distinctive characteristics are defined and described in this section. Table 3-2 compares the behavior of commercial and military networks in terms of pertinent characteristic parameters, e.g., survivability, availability, security, etc. Major distinctions arise also between strategic and tactical networks. Non-tactical systems often use commercial equipment, although the traffic loads may differ.

3.2.1. Survivability

Military networks are subject to unusual hazards like jamming and enemy action which commercial networks are not designed to contend with. The ability of a network to combat jamming and to survive destructive forces determines survivability. In the military environment it is essential for the network to provide communications to critical users at all times. A responsive control system, capable of real-time fault isolation and dynamic network reconfiguration and anti-jamming capability,

Table 3-2. Comparison of Commercial and Military Networks

Characteristics	Commercial Networks	Military Networks
Survivability	Limited by economics	Extremely high Redeployment Adaptive routing
Availability	Priorities usually not assigned	Preemption priorities Require precedence Input and output alerts Low blocking probability
Security and Privacy	Low level privacy for certain transactions only	Multi-level security required on all transactions
Reliability	Moderate MTTR	Low MTTR
Dynamics	Relatively stable traffic statistics Limited spare capacity	Large uncertain traffic statistics Spare capacity required for peaks
Adaptability	High inertia system, slow to adapt to changing environment and technology	High flexibility to suit military operations
Compatibility	Interoperability dictated by economics	Interfacing required between tactical, non-tactical and strategic networks
Special Services	Limited to specific users	Unique military Required by many users
Access Area	Depends on loop and switch costs and terminal density	Depends on geographical layout
Cost	Networks implemented to produce revenue	Network cost to be minimized
Performance	Depends on markets	Depends on special military requirements

enhances survivability. Reconfiguration of the network topology is accomplished at the switch nodes and is ultimately limited by the availability of hardware, such as links between nodes. A critical factor in network design is the network topology and the control strategy for that topology. A mesh or grid network for example provides more alternate routes. Its control program must be more versatile to take advantage of these alternatives. Distributed control, rather than centralized control, aids survivability. Thus, star networks and hierarchal structures with centralized control may not be desirable.

3.2.2. Security and Privacy

Network security protects the addresses and information being transferred from disclosure either accidentally or intentionally. Privacy refers to the prevention of unauthorized use of the network, either for sending or receiving information. Both security and privacy require 1) limiting access to authorized user terminals, 2) providing physical security to network elements, 3) shielding elements to prevent acoustic, visual or electromagnetic interception of information, and 4) address and text encryption for protection during transmission. The encryption process has considerable impact on network control.

Military networks utilize multilevel security. The security classifications used in AUTODIN I are listed in Table 3-3.

Table 3-3. Security Levels for AUTODIN I

Level	Designator
Top Secret	T
Secret	S
Confidential	C
Restricted	R
Encrypt for Transmission Only (EFTO)	E
Classified (clear transmission)	M
Unclassified	U

The letter designations shown in the table are used in message headers, they are part of overhead control signals to identify the security level. When the designator is received at a node it must be checked against the security classification of the destination terminal. A secure route must be found to that terminal. And finally, if the above two conditions are met, the appropriate action can be taken by the switch.

In some instances the control signals themselves may require encryption. For example, signaling encryption may be used to prevent disclosure of traffic statistics, volumes and addresses. However, when signaling encryption is used, these signals must be decrypted at each switch to be interpreted. Control signals are then "clear" internally at a node only and electromagnetic security measures must be taken. If non-secure signaling is acceptable, secure information can be circuit switched in encrypted form. This may not be feasible in a store-and-forward switch where the control information is added as overhead to each information block.

3.2.3. Reliability and Availability

Reliability in the context used here is concerned with the mean time between failure (MTBF) and mean time to repair (MTTR) of the network elements and the subsystems which comprise these elements. Thus the structural properties including hardware and software components of the network determine reliability. The network management and control system monitors each element and subsystem. It measures performance degradations and aids in preventive maintenance. When likely failures are indicated, the system isolates the fault and takes corrective action in order to reduce the MTBF.

Availability refers to the probability that a user will have access to the network. The system must meet the demands of the user when and if he wants to use it. Users of military networks often must contend with the limited resources available. When contention occurs, priorities are established using multiple precedence levels. In AUTODIN I, for example, the precedence levels established for CONUS are given in Table 3-4. The handling time or speed of service from receipt at the origination node until delivery at the destination node is also given in the table. It is estimated that 50% of the messages handled by AUTODIN I are processed at the routine level and approximately 1% are processed at the flash level. A 16-level precedence scheme is planned for AUTODIN II (Shah, 1977).

AUTOVON I employs a flash override precedence level, in addition to the four levels shown in Table 3-4 for AUTODIN I. When lines or trunks are busy the AUTOVON circuit switch automatically preempts lower precedence calls in progress.

These precedence procedures add considerable complexity to the control of military networks and significant overhead must be added to the control signaling system.

The preemption requirements can have considerable impact on the signaling strategy employed. In-band signaling systems currently in use on many telephone networks require the existence of an idle channel before signaling can even begin. If all trunks are busy one must be preempted for a higher precedence call. If the higher precedence call cannot be completed, the channel is tied up during the attempted access, previous call disconnect and signaling disconnect times. The preempted caller may reaccess. With out-of-band signaling on a per-trunk or common-channel basis, preemptable calls need not be interrupted until the completion of a precedence call is verified.

Table 3-4. Precedence Levels for AUTODIN I

Level	Precedence	Designator	Handling Time
I	Flash	Z	10 minutes
II	Immediate	O	30 minutes
III	Priority	P	3 hours
IV	Routine	R	6 hours

3.2.4. Adaptability

The nodes, and sometimes the links of tactical networks are often highly transportable and must be able to adapt to a mobile environment. Switches require rugged construction to withstand shock, vibration and severe temperature changes. The weight, size, and number of such switch nodes may be quite limited.

Strategic networks usually are not transportable but they are subject to changing military or worldwide conditions. These conditions could necessitate large changes in traffic loads and geographic coverage.

Non-tactical networks must also adapt to a changing environment. Sudden growth and declines in the base population are common. Operation and maintenance personnel have a large turnover. Computers, processors and terminals are being moved as required.

These changing environments affect network control in a number of ways. The terminal registration or numbering plans must adapt to contingencies, including the possibility of assigning identification numbers independently of geographical location. The changing traffic loads require network reconfiguration, which in turn requires revisions of routing plans and route control.

3.2.5. Compatibility

A military network, be it strategic, tactical, or non-tactical, usually must interface at one or more points with other networks of the commercial type. Because the interconnecting networks are often dissimilar, the network control requires conversion at the interface point. This conversion process includes transmission media and mode, code, speed, and format for not only control signals, but user information as well.

As the digitization and possible integration of military networks progresses, the signaling compatibility during the transition period is an important factor to consider. The signaling system interface compatibility is discussed in detail in Section 3.3.

3.2.6. Special Services

There are numerous functions and features available with the newer switching systems which incorporate stored program control. Advanced signaling techniques are required to bring these functions and features to the user. Commerical equipments have many desirable functions and features available in a variety of combinations. Specific military needs are not yet well established, but will undoubtedly evolve into special combinations of services and additional function and feature packages. This unique aspect will entail modification to equipment hardware and software, and particularly to the network control system.

Table 3-5 lists some of the more important functions and features the military user may require. Certain of these functions and features are unique to the military; others are already available in the commercial market. They are included here because they either expedite communication services or increase survivability, reliability, and availability.

In general, special services have an impact on signaling systems because they increase the number of signal types used for control. For store-and-forward switches, the signaling overhead and the switch complexity are both increased to handle the new functions.

3.2.7. Access Area

For commercial networks, such as the public switched telephone network, the size of a local access area handled by a central office (i.e., the exchange area) is limited by the cost of the loop facilities, especially in sparsely populated areas, and the cost of the switching facilities in densely populated areas.

The size of the local access area for a military network may involve other considerations. The access area is described as a geographic area comprising one or more post, camp, station or base entities. The actual area may be determined by factors other than loop and switching costs. The location of the

Table 3-5. Functions and Features Required in Military Signaling Systems

Switching Functions	Service Features
Network Management - traffic measurements - performance trends - equipment monitoring - failure detection - reconfiguration	Response Phase - waiting - transfer - pickup - forwarding - conferencing - camp-on
Routing Plan - adaptable - saturable	Direct Access Phase - preset dialing - automatic dialing - abbreviated dialing - night answer - pickup
Numbering Plan - flexible - code restriction - translation	Information Transfer Phase - security - privacy - accuracy - class of service
Internetting - access limiting - signaling conversion	Priority Access Phase - preemption - precedence - intercept

recognized access nodes is one such factor. Potential service boundaries constitute another factor. In the CONUS, the local access network may service an area whose diameter ranges from 1 km to possibly 50 km. The entities served within this area would include all military traffic from Army, Navy and Air Force installations.

Different military users may have different communication requirements and therefore different signaling requirements. Thus, the military access adds to the signaling system complexity.

Table 3-6 illustrates, as an example, the number of lines of different service types as functions of area size. Area size is presented in terms of total number of lines. The table also indicates the number of digital trunks required to serve such an area (Linfield and Nesenbergs, 1978).

3.2.8. Performance

Network users are interested in performance from their standpoint. This standpoint is depicted in such performance parameters as user-to-user delays, information throughput (including overhead), grade of service (blocking probability), and accuracy (error probability). The proposed Federal Standard 1033 (FTSC, 1977) lists some 26 primary and secondary user-oriented performance standards. Although these standards are being developed for digital data communications, many of them are also applicable to digital voice systems. The performance parameters in FS 1033 are defined for three phases of the telecommunications process: the access phase, the information transfer phase and the disengagement phase. Performance during each phase is judged in terms of efficiency, accuracy, and reliability. Numerical values are not specified, only the criteria which describes the performance during each phase.

The performance of a network is largely determined by the network control, strategy and signaling system used, particularly performance during access and termination phases. Although the signaling system itself affects neither information transfer nor

Table 3-6. Number of Various Type Lines Served in Different Size Areas

Services	Area Size in Number of Lines Served									
	50	100	200	300	600	1,000	2,000	4,000	5,000	10,000
Analog - Phones	44	88	175	262	525	880	1,760	3,520	4,400	8,800
Analog - Slow Data	1	2	4	6	12	18	36	72	90	180
Digital - Phones	4	9	18	27	54	88	176	352	440	880
Digital - Slow Data	1	1	2	3	6	9	18	36	45	90
Digital - Fast Data	-	-	1	2	3	5	10	20	25	50
Digital - Trunks	15	30	60	90	180	300	600	1,200	1,500	3,000

control signal transfer, accuracy and reliability are important. These signaling aspects are discussed in Section 4.

3.2.9. Cost

The DoD views the DCS as a revenue consumer as distinct from common carrier networks which are revenue producers (Rosner, 1973). Thus, whereas a common carrier desires to sell as much service as possible to users of the network, the DCS is designed to provide only the service required to meet military needs and at a minimum cost. The importance of cost cannot be overemphasized. The DCS costs about $500 million per year to operate. Another $400 million per year is invested in new systems. This $900 million expended per year represents about 22% of the DoD expenditures for communications, command and control (Babcock, 1977).

The current DCS contains three differently switched systems, the circuit-switched AUTOVON network, a message switched AUTODIN and the packet-switched ARPANET. As new generations of networks emerge, the direction they take will depend largely on cost. Digital networks, when fully implemented, are expected greatly to reduce the operation and maintenance (O&M) costs. However, during the transitional phase from all-analog to all-digital elements, the O&M costs may temporarily increase.

As noted previously, the local loop plant costs can dominate the network cost. Unfortunately, upgrading existing loop facilities can be an expensive proposition. Using the curves in Figure 2-11 and the projected total of 1 million telephone instruments and 500,000 terminals, it is possible to estimate the total cost required to replace the wire pairs in the loop plants with coaxial cable. An average terminal density of 1000 terminals per square kilometer has been projected for Ft. Monmouth in Table 2-10. Using this value and assuming a 4 kHz bandwidth for analog phones and 16 kHz bandwidth for digital phones*, the costs per terminal are $450 and $780, respectively. The total cost of the

*The 16 kHz digital bandwidth assumes a DPCM digitized voice system operating at 32 kb/s and 2 b/Hz signals.

local access network replacement is $225 million for analog telephones and $390 million for digital telephones. Telephone terminals amount to 95% of all terminals. The other 5% consists of data terminals and computers which require additional digital capacity. These results are summarized in Table 3-7.

Another major cost item for digital switches in a military environment is the software development costs. These costs have been growing rapidly, due largely to the specialized labor required. The unique military features could be expected to foster this trend unless special steps are taken to standardize software packages and requirements. The use of higher level languages for programming switching software could be used to increase programmer productivity. This, however, adds to the overhead of assembly-language instructions produced.

3.3. Future Trends

Future changes in military communications networks will result from innovations in technology. These changes are driven by the market demand for new services and better performance at lower cost. Several recent technologies have found applications in network elements. These include digitization, stored program control, common channel signaling, traffic integration, and satellite systems.

It is important to review these innovations, because the sections that follow are concerned with the network control of future telecommunication systems. The selection of advanced signaling systems must be based on this emerging technology, to insure that these systems best meet the user's needs.

Table 3-8 summarizes the impact new technologies have on major network elements. One technology that is not listed is large-scale integrated circuitry (LSI) and very large-scale integrated circuitry (VLSI). This manufacturing process is, of course, a major factor in reducing cost, size, weight and power consumption in all network elements. At the time when these devices are incorporated into large networks of the future, the

Table 3-7. Total Investment Costs to Replace Local Access for Buried Outside Plant and Distributed Control

Total No. of Terminals	95% Analog Voice	95% Digital Voice
No. of voice terminals	500,000	500,000
No. of voice subscribers (2 ext per terminal)	1,000,000	1,000,000
Bandwidth/terminal	4 kHz	16 kHz
Bit Rate (2 bits/Hz)	N.A.	32 kb/s
Average density	$1000/km^2$	$1000/km^2$
4/terminal drop	\$450	\$780
Total cost (excluding terminal costs)	\$225 million	\$390 million

Table 3-8. The Impact of New Technologies on Network Elements

Network Elements / Technology	User Terminals	Concentrators and Switches	Transmission Facilities
Digitization	Provides easier end-to-end security. Simplifies interface with transmission facilities	More effectively managed Reduced size, weight, and power consumption Modularity	Greater noise immunity Eliminates marginal operations Improves bandwidth utilization Eliminates modems
Switching with Stored Program Control	More features to user Permits common-channel signaling on trunk side	Adds flexibility Simplifies hardware Performs more functions	Permits common-channel inter-switch signaling
Common Channel Signaling	More signals for more services Reduces access time	Greater reliability Reduced cross-switch delays Increases traffic handling capacity	Faster operations Requires special link testing Resistent to interference
Voice/Data Traffic Integration	Permits interconnecting broader community of terminals	Adds complexity Adapts to best traffic handling procedure	Single facility handles all classes of traffic Pooling excess capacity increases efficiency

economies realized could provide order of magnitude reductions in costs for manufacture, operation, and maintenance.

3.3.1. Digitization

Although it appears that future digital technology will permit reductions in space, power and eventually the cost of telecommunications equipment, there are both advantages and disadvantages to the digitization of future networks. These are summarized in the following paragraphs for each network element.

As noted previously, the dominant terminals in the network will probably continue to be the voice terminals; i.e., the telephone instruments. Digitizing voice terminals has certain advantages, not the least of which is security. A block diagram of such a terminal is shown in Figure 3-1. Digital encryption devices generate signals which are less susceptible to deciphering then encrypted analog signals by unauthorized persons. At the same time, such devices are becoming less expensive when based on LSI circuitry. In the future, digital voice terminals may reduce the bandwidth required for voice transmission. For example, a digitized voice channel using linear predictive coding (LPC) requires approximately 2.4 kb/s for transmission. Four such voice channels can be carried over an analog voice-grade line at 9.6 kb/s which normally handles one full-duplex voice link. Digital voice terminals may ultimately be capable of interfacing with other digital computers, e.g., a voice-operated computer.

The digital transmission of signals over both analog voice band and data links offers several advantages. Digital signals are amenable to regeneration and are therefore more resistant to noise over long distances. The reliability of a digital link is greater. No modems are required. Accuracy is improved using simply implemented error control schemes. LSI interfacing equipment is less expensive to build. The nodes, digital switches, concentrators, and multiplexors are simpler and easier to design. Mixers, filters, and converters are no longer required. The LSI microprocessor can be used to control not only the switches, but to provide intelligence in the terminals as well.

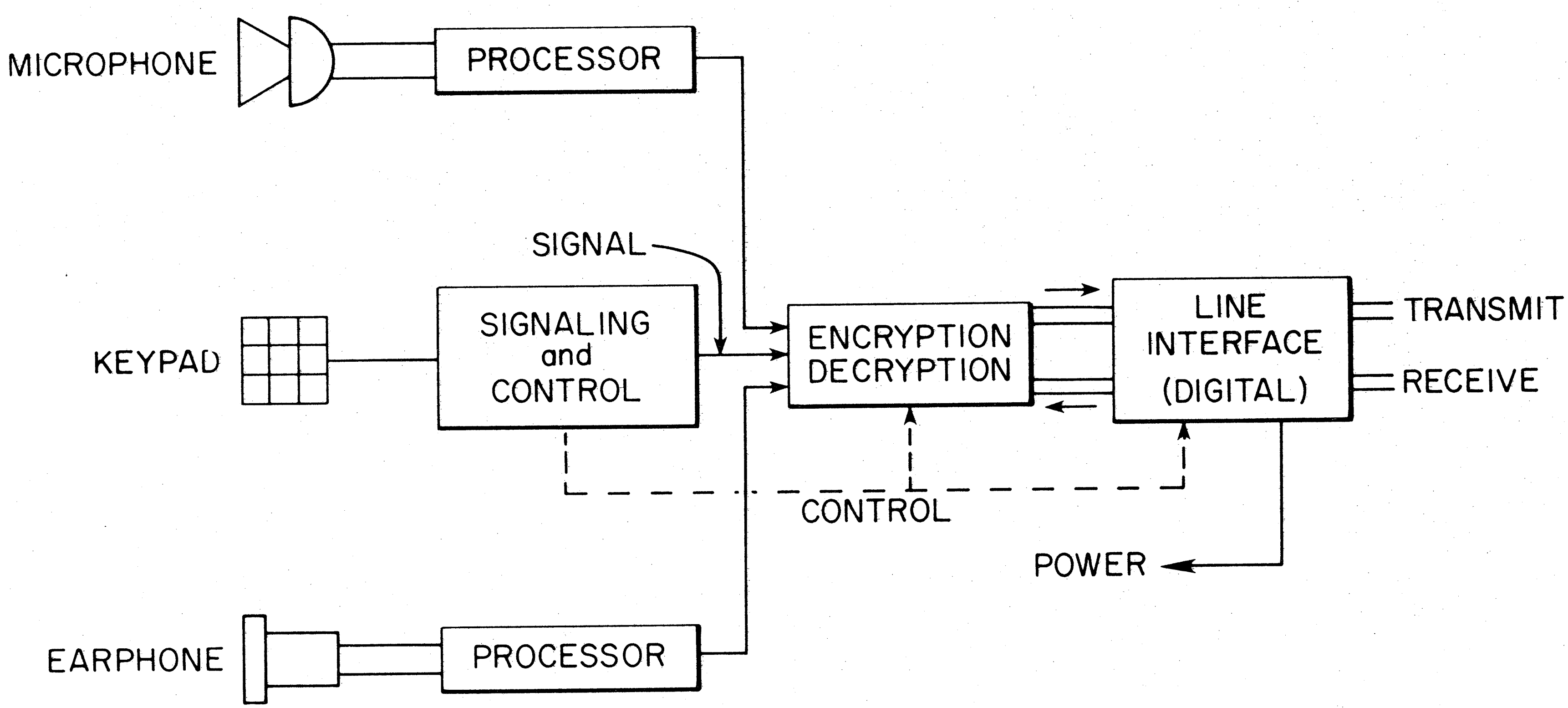


Figure 3-1. Digital secure voice terminal block diagram.

Digital signaling systems provide larger signaling alphabets for control of more functions and providing more services to the user. Robust signaling systems (which are less susceptible to environmental degradations) use digital techniques for the same reasons given above.

There are also some disadvantages including the cost of digital telephones and the problems associated with full duplex operation over 2-wire loops using digital transmission. Software development costs for the switch is another disadvantage.

Some digital transmission is being implemented with the DCS Digital European Backbone (DEB) in central Europe. This trend may continue and could encompass the local base networks in the future according to the Base Communication Plan, BASCOP (CEEIA, 1977) if proven cost effective.

3.3.2. Stored Program Control and Common-Channel Signaling

The use of software, stored in the memory of a processor, is expedient to control space- and time division switches. The stored program control (SPC) adds flexibility and operating speed to the switching process. The practical advantages of centralized SPC were demonstrated in 1965 with the introduction of the No. 1 ESS by Bell Laboratories. More recently, decentralized control has emerged. Decentralization uses microprocessors and LSI technology. Decentralization distributes the processing functions of network control, but to do so, it places a greater burden on the signaling systems. More signaling is needed to allow the network processors to communicate with each other. The interprocessor communication links may take many forms and these are described in Section 4.2. One form of particular interest is common channel signaling. It provides the SPC with new capabilities by virtue of its increased signaling capacity relative to conventional signaling. The SPC switching, in conjunction with common-channel signaling, may provide the adaptability required by military networks in the future. Simple software additions permit accommodation to new services as needed. However, the cost effectiveness must still be demonstrated.

3.3.3. Integration

As military networks for strategic, tactical and non-tactical communications evolve to meet the needs of the future, it is possible that a unified common user network could result with some sacrifice in performance. The advantages and disadvantages of a single integrated network capable of handling voice and data have been discussed by many workers including Rosner (1973), Coviello and Vena (1975), Esterling and Haku (1975), Occhiogrosso et al. (1977), Gitman et al. (1977), Ross et al. (1977), and Schutzer and Ricci (1976). The cost effectiveness of such a unified network has been evaluated under the ARBITS and AFBITS programs (MITRE, 1976) for base communications.

Integrated networks are designed to serve communities of users with different traffic types by sharing switching equipment as well as transmission capacity. Figure 3-2 illustrates how integrated networks could evolve from completely separate voice and data networks, to networks which share only the transmission facilities, and finally to a completely integrated network. Many current networks share transmission facilities as shown in Figure 3-2b. Digital data from data terminals are converted to analog form using modulators and demodulators (modems) for transmission on analog facilities. In an alternate form, the analog (voice) information may be digitized using various coding and decoding (codec) schemes for transmission on digital facilities. In either case the switches at the nodes are separate circuit and store-and-forward (S&F) type switches.

It is also feasible to combine these nodal functions into a single entity, so that a completely integrated network is obtained (see Fig. 3-2c). The principal advantage of this is that the integrated network can serve a greater number of terminal types. The total cost of transmission facilities can be reduced by optimizing the sharing possibilities. There may also be some disadvantages. The integrated network may only be economically achieved by sacrificing performance to certain classes of users.

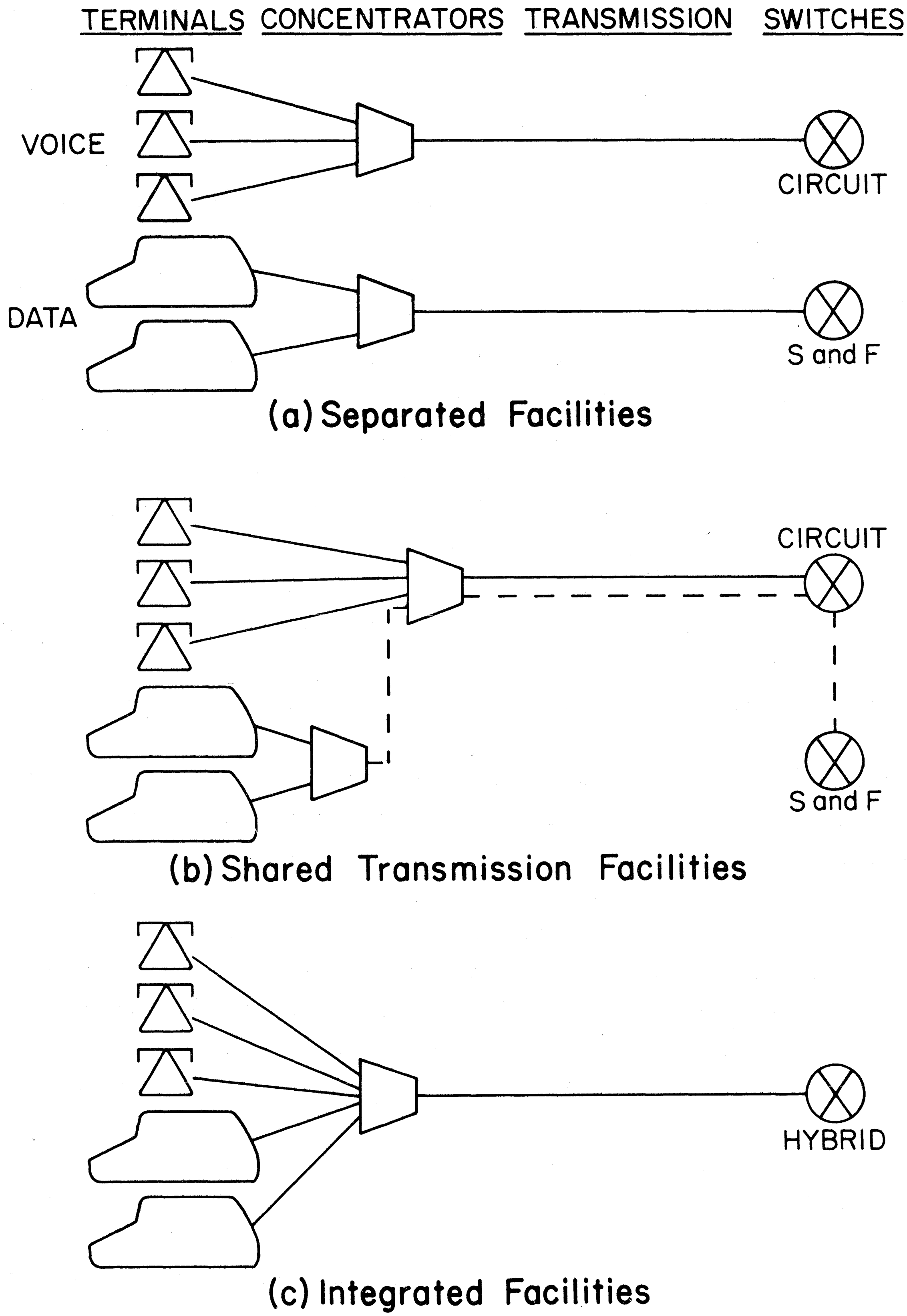


Figure 3-2. Integrated network evolution.

Increased complexity may increase vulnerability to catastrophic failures.

An integrated network may take different forms. One form can use digitized voice terminals and digital transmission facilities, including digital (or virtual) circuit switches. However, the advantages of store-and-forward switching are lost for this network.

Another alternative is to packet switch all of the traffic. Packet switching of digitized voice signals, however, is still in the experimental stages. Gold (1977) describes ARPANET experiments conducted with packetized voice. Two problems arise for packet voice. One is related to overhead and the other to quality degradation due to variable delays through the network. A packetized virtual circuit proposed by Forgie and Nemeth (1977) whereby network links are preassigned to the user during dial-up could reduce these problems. Packet overhead is reduced by distributing destination data to the nodes during dial-up and the packets merely carry an identifier. The reduced overhead increases efficiency and the fixed routing eliminates the variable delay.

Circuit and packet switching networks can be combined using a "hybrid" switch, where the network adapts to the traffic type by selecting the appropriate switching technique. An example, described by Coviello and Vena (1975), and GTE (1975) is the slotted envelope network (SENET). A constant, self-synchronizing master frame is divided into time slots, which can be dynamically allocated to different traffic classes, and switched accordingly. One packet-switched channel is assigned to control virtual connections for class I traffic. Other classes of traffic carry control signals along with the messages.

The evolution of a hybrid switch, from an all-analog system to an all-digital system, is depicted in Figure 3-3. In Figure 3-3a, the control signaling for the switch is furnished on a per channel basis using an in-band signaling system on the lines and trunks. In Figure 3-3b, the common-channel signaling is used.

The signaling information for several lines and trunks is transferred over a separate data link using modems to interface with the link. A digital time-division switch establishes virtual connections for both voice and data. Since analog facilities are used for transmission, the digital conversion occurs in the interface to the lines and trunks. In Figure 3-3c the processor and memory are expanded and additional lines and trunks are added, so that data can be packet switched and voice can be circuit switched. The control signaling for voice circuits is handled by one of the packet switched data channels. Figure 3-3d is the hybrid switch. Digital data and voice are both packet switched and circuit switched using the slotted envelope transmission facilities. The control signals are handled as if they were digital data.

The evolution of military networks from all-analog to combined analog and digital and ultimately to all-digital configurations is already progressing. Voice and data integration, on the other hand, is still in exploratory R&D stages, as noted previously.

3.3.4. Satellites and Packet Radio Systems

It is expected that satellite communications will continue to play a role in future military network operations. Satellite intermediate message processors (IMP's) have been developed for use on the ARPANET. These satellite IMP's use random access packet broadcast techniques pioneered by the ALOHA system. A large number of geographically distributed users are connected to a central computer via satellite radio links (Abramson and Kuo, 1973; Schwartz, 1977). Satellites could, in the future, be exploited to provide access on the regional and local levels as well as in backbone structures if costs are more attractive than terrestrial alternatives.

Packet radio system concepts are also being developed for use in terrestrial tactical networks and local distribution of data to offset the high cost of local distributions (Kahn, 1977).

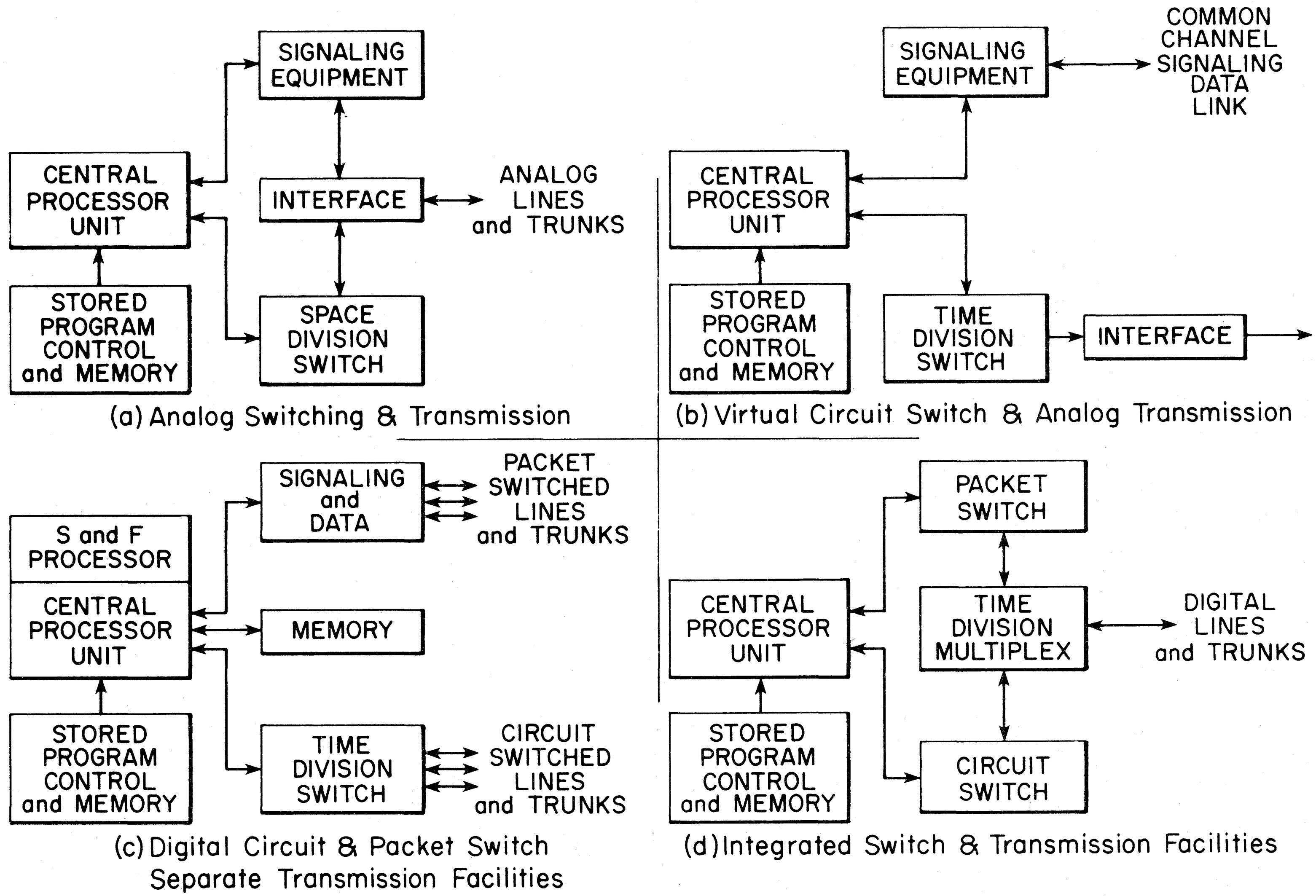


Figure 3-3. Evolution of hybrid switch.

Control of the packet switched systems, which statistically share a service entity (e.g., a satellite or terrestrial subsystem), is beyond the scope of this study. However, it is important not to overlook such controls when selecting the advanced signaling systems for future military communication networks.

3.4. Network Forecasts

Predicting the future is always risky, particularly in the telecommunications area which is continually undergoing both innovative technological and service-requirement changes. Near term predictions over the next 5 years appear fairly reliable because changes can be expected to occur slowly due to the large investments in existing facilities. The network configurations projected 20 years in the future are, of course, far less reliable.

Some recent forecasts have been made for the strategic DCS network. See, for example, Krevsky et al. (1972), Rosner (1973), Paschall, et al. (1976), and Levine (1976). The BASCOP program includes plans to digitize local facilities (CEEIA, 1977). The ARBITS program could lead to integrated services in the local non-tactical environment (MITRE, 1976). Tactical networks are being developed for use by all military services in the 1980's under the TRI-TAC program (Hoover, 1977).

Using these plans and programs, plus information from other sources, the general characteristics of current, transitional and long range military network configurations are estimated in Table 3-9. These projections indicate only the broadest expected network characteristics. They are useful in defining network control and management procedures that are, (a) compatible with the imminent transitional networks, and (b) adaptable to networks expected in the more distant future. Several advanced signaling system alternatives are discussed in Sections 4 and 5. Some aspects which pertain to the transitional period are covered in the following subsection.

Table 3-9. Military Network Forecasts

	Current (1970's)	Transitional (1980 to 1990)	Long Range (1990's)
Networks	Separated by technology type and by service type	Combined technology types, some integrated by service	Combined strategic, tactical and non-tactical with integrated services
•Strategic	Mostly analog, some digital	Many digital	All digital
•Tactical	Mostly analog with separate subnets	Analog/digital hybrids, some internetting with strategic networks	All digital
•Non-tactical	Separated classes, mostly analog	Integrated data, separate voice and video, analog/digital hybrids	Integrated voice, data and slow speed imagery, separate video (TV)
Terminals and Traffic Voice	Over 95% analog	90% analog, 10% digital PCM	10% analog, 90% digital, with lower digitization rates
Data	Some intelligent	Mostly intelligent	Added intelligence with microprocessors
Nodes Switches	Mostly analog, step-by-step, crossbar, some electronic ckt. sw. and digital message switch	Analog and digital, electronic ckt. sw. with time division multiplexing, separate packet switch	Mostly digital, hybrid packet and circuit, time division
Concentrators	FDM	FDM and TDM	Statistical TDM
Multiplexors	FDM	FDM, Statistical TDM	Mostly STDM

Table 3.9 (cont.)

	Current (1970's)	Transitional (1980 to 1990)	Long Range (1990's)
Links			
Long Haul	Mostly analog with linear repeaters, LOS microwave, some satellite, coaxial cable, troposcatter	Analog/digital, linear and regenerative repeaters	Digital with regenerative repeaters, satellite, LOS, coaxial cables
Short Haul	Analog and T-carrier	T1, T2, T4 carrier	Satellite, packet radio, fiber optics, millimeter waveguides
Local Loops	Mostly analog, quasi-analog, 2- and 4-wire pairs	Digital and quasi-analog, 2- and 4-wire pairs, T1 carrier and coaxial cable	Quasi-analog and digital coaxial cable, fiber optic cables, T-carriers
Network Control and Management	Deterministic routing, manual and semi-automatic control	Deterministic routing, semi-automatic control	Adaptive routing, automatic control
Signaling System			
Lines	dc and ac, in-band and out-band	dc and ac (in-band)	Digital, in-slot, channel
Trunks	MFDT addressing, SF supervision	ac (in-band) and common channel	associated common channel some non-associated

3.5. Transitional Interfacing

Interfacing new digital networks with the existing analog networks raises several issues of signaling compatibility. Several of these issues were considered by Linfield and Nesenbergs (1978) and will only be summarized here.

The details of future network configurations are unknown. Therefore, the selection of advanced signaling systems for future networks cannot be divorced from issues of adaptability. Integration of voice and data on a single network is one such issue. Should such integration occur, the network controls would have to differ from those existing now on separated networks.

Adaptability and compatibility during the transition period and beyond is a key factor in the development of access area digital switching systems (AADSS). The AADSS program must be concerned with control signaling between elements of a given network (intranet control). During the transitional period, the intranet signaling system must interface with both analog and digital facilities. The interfaces for intranet and internet control are described in the following paragraphs.

Intranet Control Interfacing

An example of the transitional interfacing problems is the intranet control of the telephone network. As the digitization process progresses from the highest hierarchy backbone switches to regional, local, and PABX switch levels, and ultimately to the terminals themselves, more interface gateways arise. Although their unit costs may be somewhat reduced, the total expense increases. Overall, the interfacing issues become more and more important. Figure 3-4 indicates one approach to the problem at the PABX level (Linfield and Nesenbergs, 1978). Exchangeable modules are used to facilitate analog-to-digital transition. An electronic switch with SPC and time division switching is used to perform the switching functions. The interface modules shown provide analog to digital conversion for up to four analog lines or digital multiplexing for digital lines. Digital T-carrier

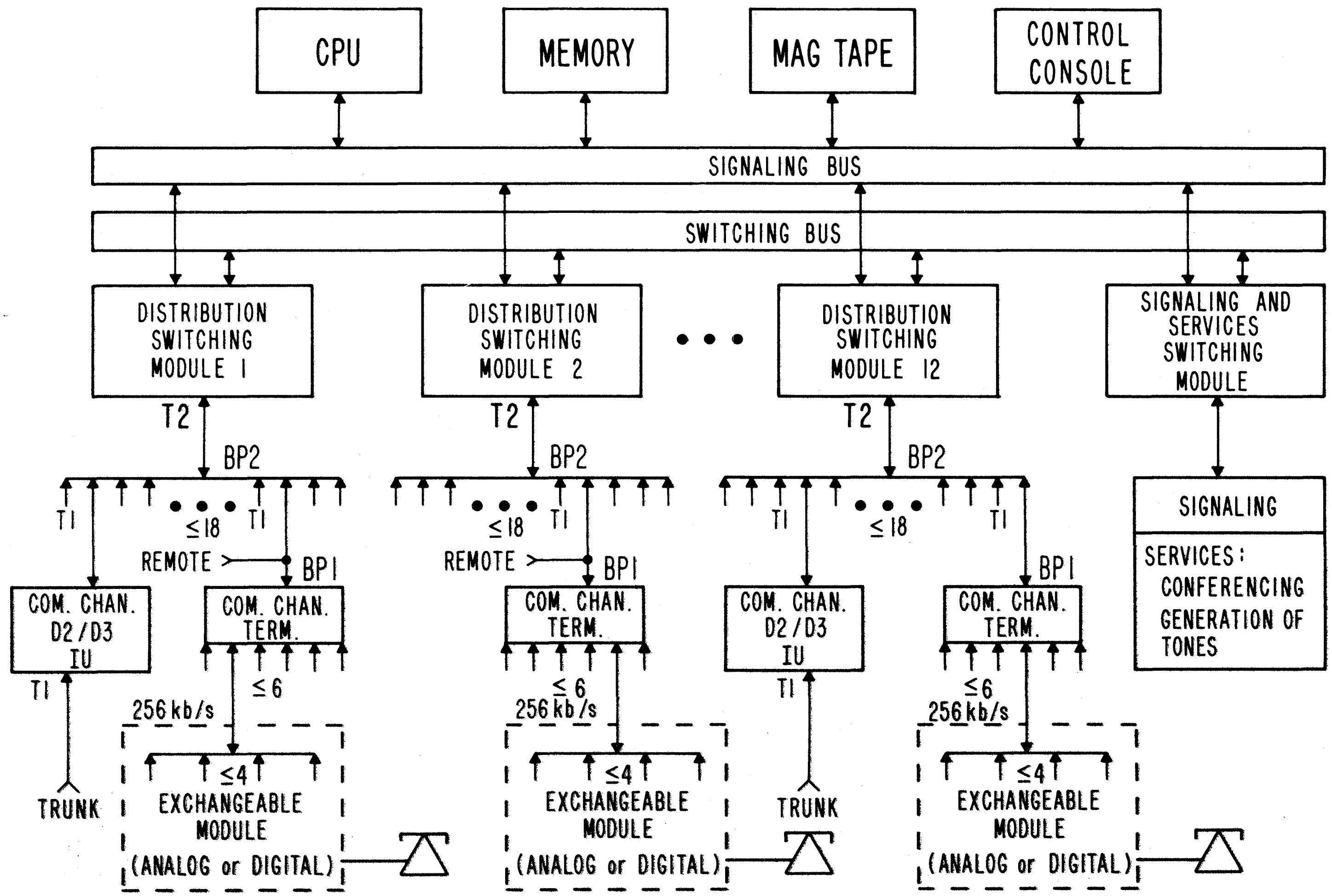


Figure 3-4. Exchangeable four-line modules to facilitate analog-to-digital transition on a PABX.

trunk transmission facilities are assumed, although a combination of digital and analog trunks could be used.

A major consideration in any terminal loop or line interface is the BORSCHT. This is an acronym used by the telephone industry to cover all the functions required in a line interface circuit. Most of these functions are also required on trunk side, but for somewhat different purposes. The basic functions of BORSCHT are

B Battery feed
O Overload protection
R Ringing
S Supervision and signaling
C Codec and clock
H Hybrid (2-wire to 4-wire)
T Test.

A block diagram shown in Figure 3-5 indicates the hardware relationships between elements of BORSCHT. It assumes digital switching for both analog lines and trunks, and digital lines and trunks. Other examples of BORSCHT for the digital line interface to a telephone are given by Melvin (1978).

It is apparent that nearly all the elements of BORSCHT appear in the interfaces of Figure 3-5. The battery supplies the dc current for dialing and for the microphone on the line side. The battery voltage is typically 48 volts. The dc circuit sees a low line impedance of about 400 ohms. The ac terminating impedance matches the telephone cable of either 600 or 900 ohms. A conventional carbon microphone requires 20 to 80 mA of line current.

Overload protection is required on all incoming lines and trunks. This protects the more sensitive circuitry from damage due to high voltage transients induced by lightning or power lines. Typical protection levels are for 1000 volts and 1 ampere surges lasting about 1 millisecond. Greater protection is required for sensitive solid state switching devices.

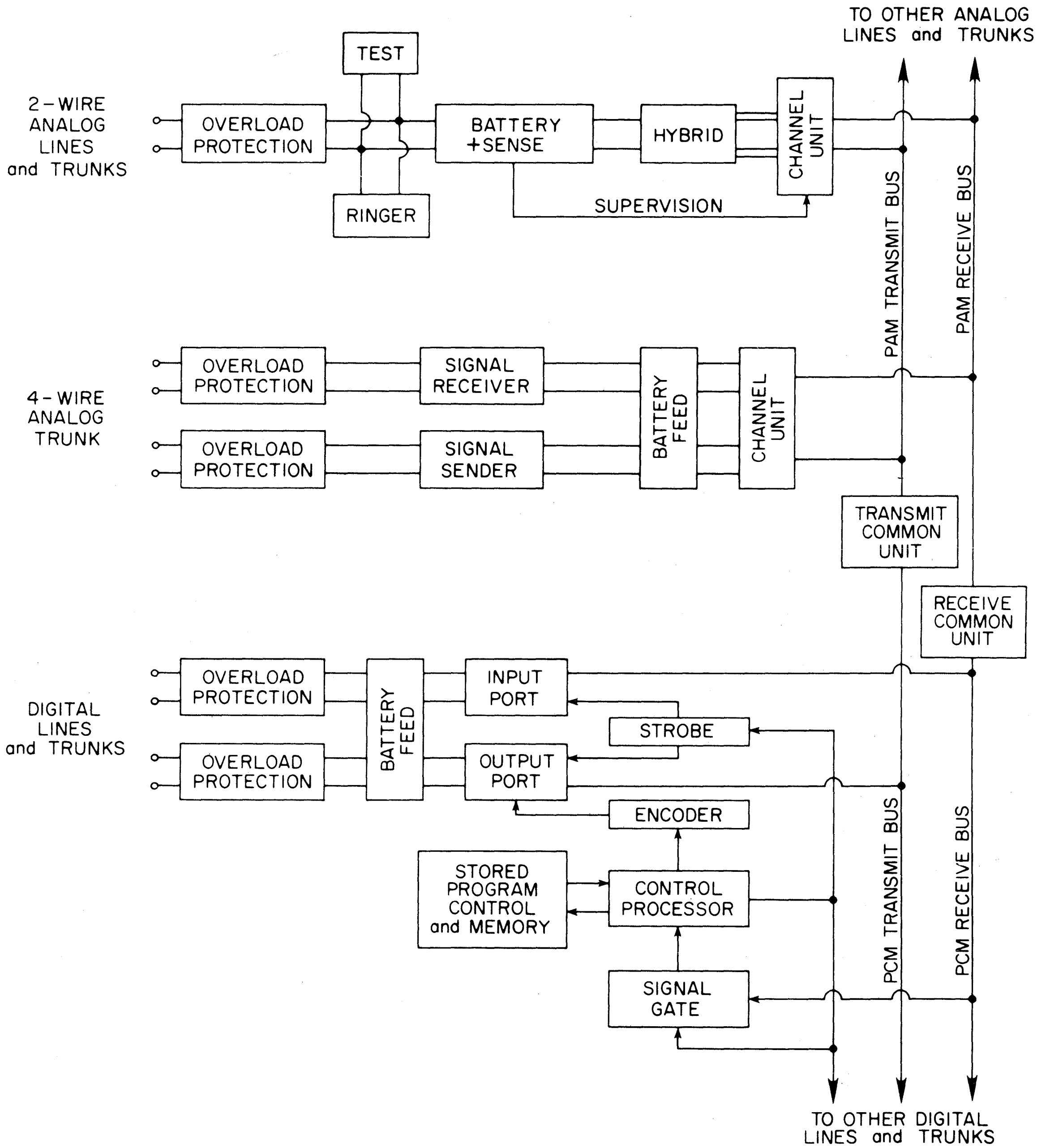


Figure 3-5. Analog and digital interfacing function block diagram.

Ringing power is required on the line side. A conventional telephone instrument requires a nominal 20 Hz tone at 86 volts rms to ring the bell.

Supervision and address signaling from a conventional telephone requires 15 mA for dc hook status control and rotary dialing of dc pulses.

The clocking, codec, and channel filters are required to match the analog loop to the digital switch. The channel filter is nominally 200 Hz to 3400 Hz bandpass.

The hybrid converts 2-wire analog line to a 4-wire system. The hybrid is on the terminal side of the codec since digital carriers are inherently unidirectional. The line hybrid must be designed to prevent voice degradation due to coupling between the transmit and receive paths. Echo cancelling may also be used in the hybrid. The hybrid is not required on 4-wire lines and trunks.

The test circuitry provides a means to switch the line to a test facility. On the trunk side, test signals are injected to insure proper operation of the trunk. This is particularly important when common channel signaling is used, since the control signals are not conveyed by the trunk. On 4-wire trunks, a typical 2010 Hz test signal is looped back to the source to indicate a continuity check. On 2-wire trunks, a 1780 Hz signal is used in the outbound direction and 2010 Hz signal is returned.

During the transition from an analog voice network to a digital voice network, conventional telephones may be attached to some form of voice digitization processor (VDP). If a secure link is required, the processor may be followed by a communications encryption device (COMSEC unit) as shown in Figure 3-6a. All the elements of BORSCHT must get by the VDP and the COMSEC units, if a conventional telephone instrument is used.

Ultimately, a low-cost digital voice terminal may be developed using LSI circuitry. A simplified block diagram of such a unit is indicated in Figure 3-6b. Certain time slots are allocated for inbound and outbound digital control signals. No

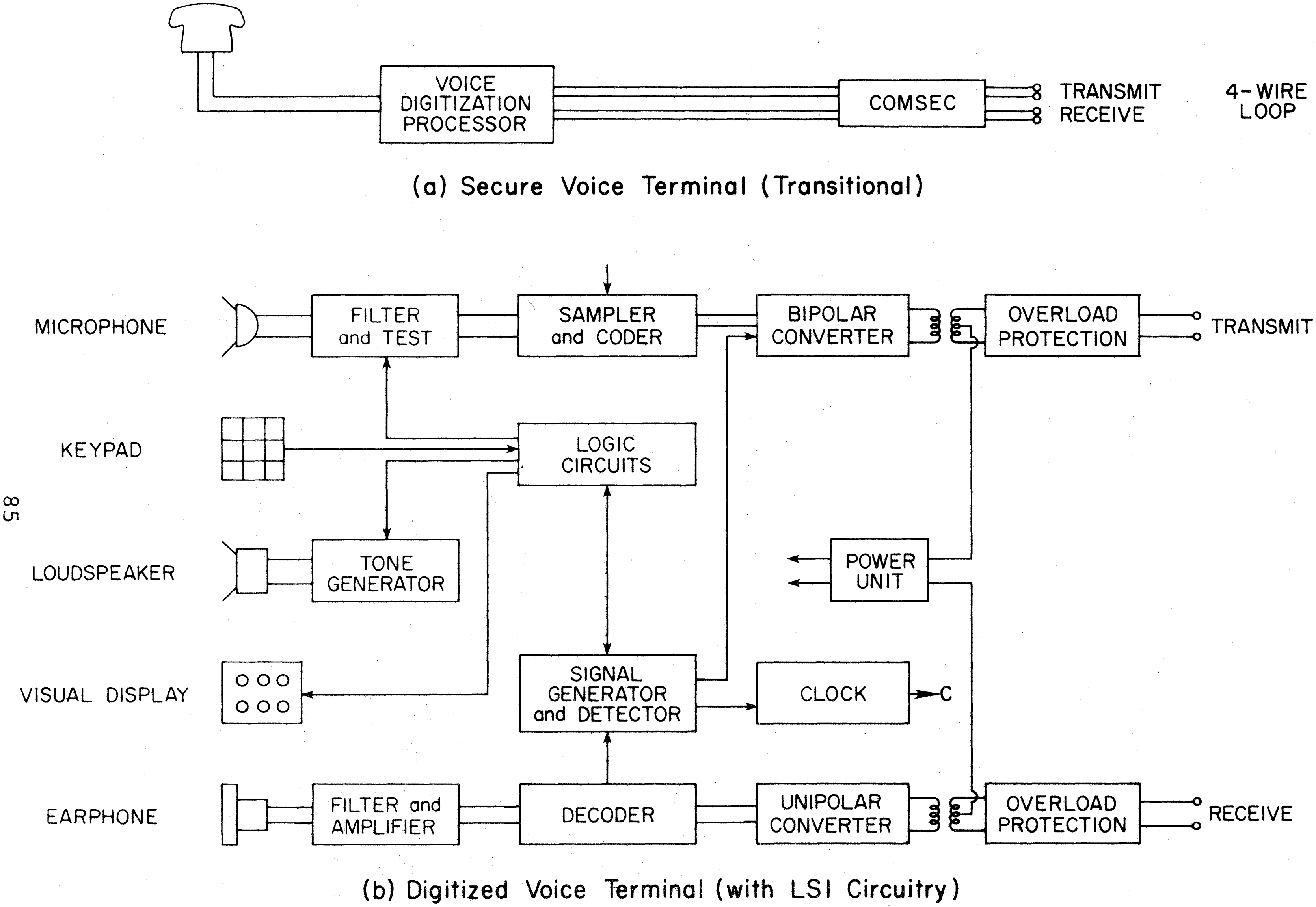


Figure 3-6. Digitized voice terminals.

hybrid is required. The control signals operate a signal generator and speaker for both alerting and call progress tone generation. Power is furnished from the switching node via a phantom circuit on the 4-wire line. Clocking is derived from the incoming line. All incoming and outgoing control operations are performed by the logic circuitry. Except for the power, the remaining elements of BORSCHT are passed digitally.

Internet Control Interfacing

The future AADSS may be required to interface with and to provide internet controls to strategic, tactical and non-tactical networks, as well as to commercial networks. During the transitional phase of digitizing and integrating, the required control interfaces may be extremely complex. Figure 3-7 indicates most of the potential interfaces which must be required in AADSS for switching voice circuits, whenever both analog and digital transmission facilities are interconnected.

In addition, data circuits may have to be passed through the same switching node. Existing data networks use a variety of signaling schemes. The AADSS must act as a gateway between these networks by performing mode, code, speed, and format conversions on the control signals. Possible gateway concepts are described in Section 6.

Figure 3-7 does not imply that every AADSS should be implemented with every known interface capability. Many of the interfaces indicated are used selectively by the commercial telephone networks in their end offices around the country. The AADSS for a given region needs only to incorporate the interfaces required to match the nearest end office.

The following sections 4 and 5 specify those advanced signaling systems which have potential network control applications at the AADSS. The emphasis is on digital networks which may be either integrated or not. Digital signaling techniques for both the line side and trunk side are considered. The signaling systems used for circuit-switched voice differ from

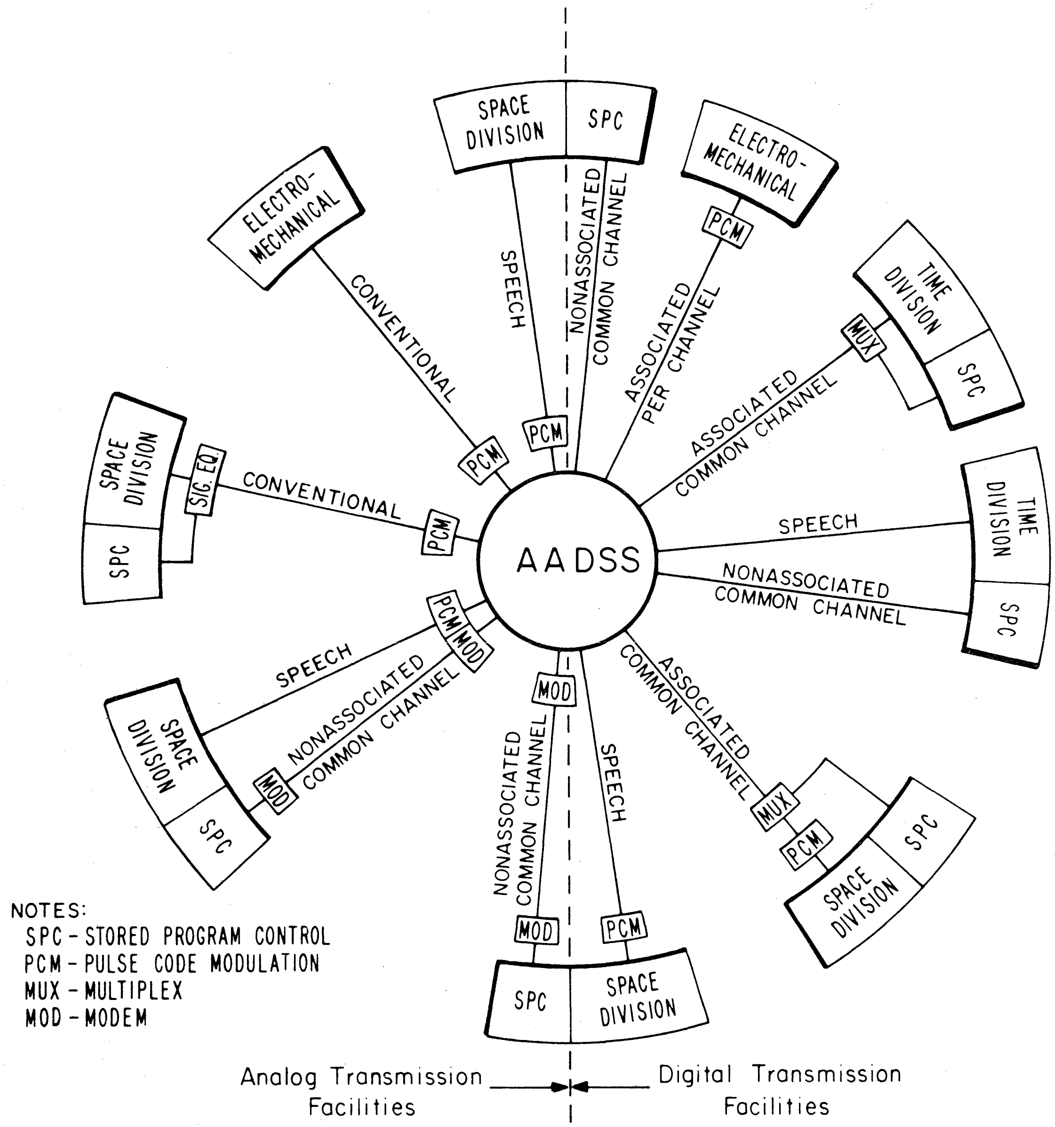


Figure 3-7. Potential signaling interfaces to access area digital switch for transitional network environment.

those used for packet-switched data, but the technique can be combined for controlling hybrid (circuit and packet) switching.

4. NETWORK CONTROL

In Section 2 the basic elements of a telecommunications network were described. In Section 3, the unique characteristics of the military switching environment were presented and future forecasts were made for the military network. In this section the network control and management aspects are considered.

Two types of signals flow on a network: user information and control information. The user information or traffic (e.g., voice, data, video), which flows between terminals, represents a useful service to the customers. It justifies the existence of the network. Control information is necessary to maintain this flow.

Control information can be classified into three basic signal categories.

1. Signals that operate switches and terminals to establish, maintain, and terminate the transfer of user information. These control operations are usually on a real-time or short-time basis.
2. Signals that control the links to ensure efficient, accurate and reliable network transmissions.
3. Signals that manage the overall network, its elements, and its operations on a longer term basis.

Table 4-1 lists the major network elements and gives examples of their basic functions that require the transfer of control information. This transfer or exchange of information for network control requires a special signaling system. Since all network elements are involved, the signaling system permeates the entire network. Communication processors at the nodes must exchange information with the terminals and with each other. This exchange may be accomplished over the same links used for transferring user information or over separate links. The

Table 4-1. Functions Performed by Various Types of Control Signals

Terminal Control Signals
•Attending
•Addressing
•Alerting
•Supervision
Node Control Signals
•Registration
•Translation
•Path Searching and Selecting
•Routing
Link Control Signals
•Synchronizing
•Error Protection
•Testing
Network Management Signals
•Flow Control
•Status Monitoring
•Performance Assessment
•Reconfiguring
•Recording

control facilities which perform all these control functions are therefore equivalent to the nerve system of the network. They sense user needs and act accordingly. This nerve system which remotely controls the terminals, the nodes, the links, and the entire network is the subject of this section. The field of network control is called "signaling", or more appropriately "control signaling."

The Consultative Committee of International Telegraphy and Telephone defines signaling as (CCITT, 1973a):

> "Signaling: The exchange of electrical information (other than speech) specifically concerned with the establishment and control of connections, and management in a communications network."

This definition applies primarily to circuit switched telephone circuits. However, if the parenthetical expression were amplified to be "other than user information, e.g., speech, data, and video" the definition would also apply to all signaling networks. Data, of course, can be sent over voice networks using unmodified telephony signaling to establish, maintain and terminate the connection. However, the transfer of data may be expedited by adding special control signals. Error control signals are a good example. The parity check, repeat request, as well as other additional signals are called "overhead" since no user information is involved. Overhead signals are required to provide the orderly exchange of user information over the network. The procedures or arrangements established by overhead are called protocols. Certain standard protocols are commonly used on voice circuits - e.g. saying "hello" to start a conversation and goodbye to terminate one. On data links various levels of protocols are used to exchange user data and to ensure the proper flow of information over the network. These protocol levels are described in a subsequent section.

Figure 4-1 shows separate network control procedures for voice and data networks. Examples of specific signaling systems used under each breakdown are indicated on this figure.

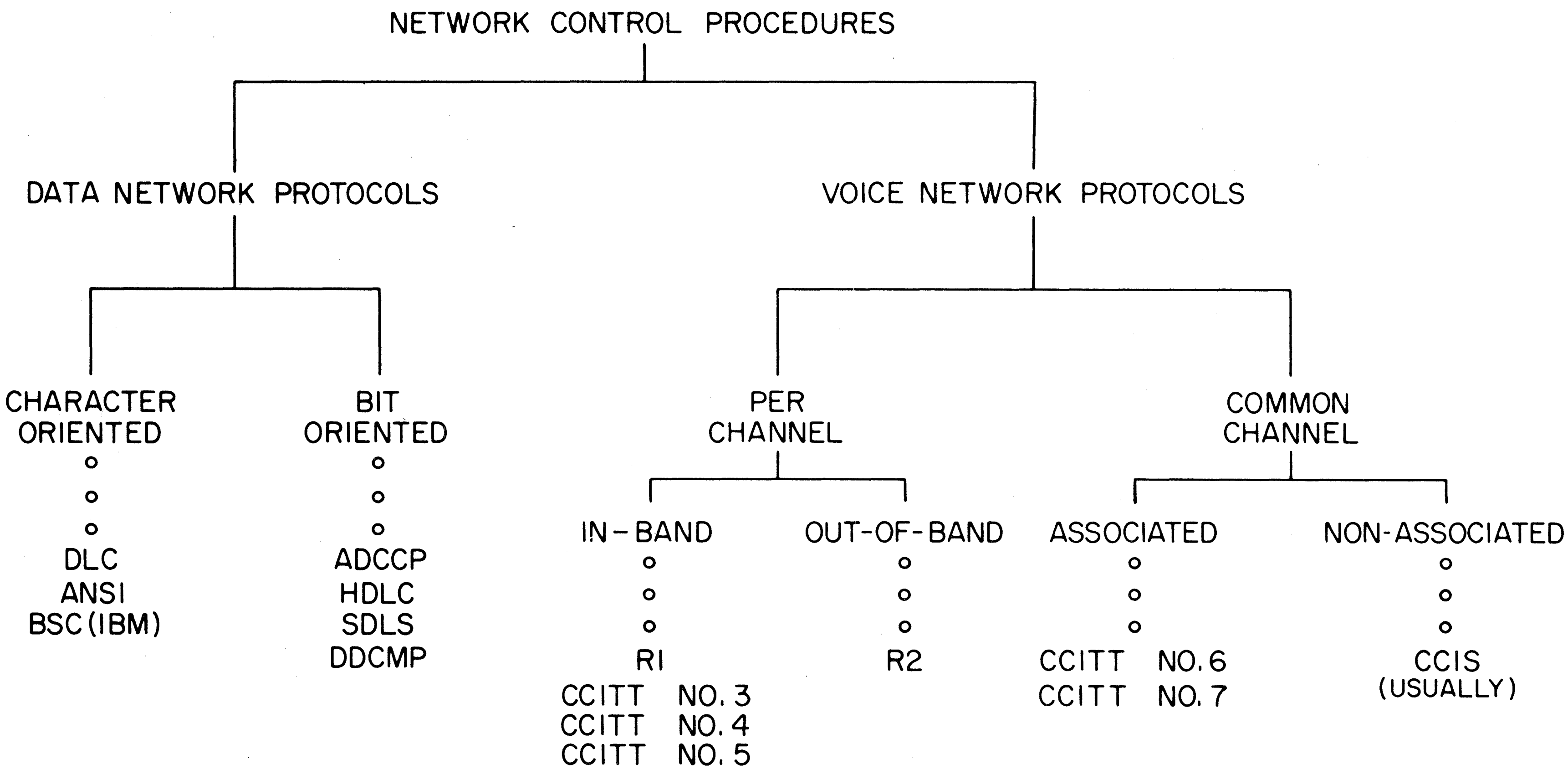


Figure 4-1. Network control procedures for voice and data networks.

Traditional signaling systems incorporating in-band and out-of-band signaling techniques were reviewed by Linfield and Nesenbergs (1978) for telephone networks. In this section, "advanced signaling" techniques are considered for both circuit switched networks and store-and-forward switched networks. Since most network projections indicate a trend toward all-digital networks, digital control signaling systems are emphasized. Both terminals and switches are involved with signaling. Signaling systems are considered separately for the line side and the trunk side of the switch. The emphasis here is on out-of-band digital signaling for digital terminals and common channel interswitch signaling for the trunks. These two signaling techniques are essential for accommodating both voice and data on an integrated network.

In the sections which follow it is assumed that the nodes of the future military telecommunication networks will be automatically run by processors with stored program control. The interprocessor control information will be transferred digitally using some form of signaling system as described in Section 4.2. Specific systems are described in Section 5 for voice and data networks.

4.1. Requirements and Issues to be Resolved

Initially, the major requirements of the signaling system used to control a military telecommunications network can be listed in a qualitative way. This is done in Table 4-2 where the unique characteristics of the military environment given in Section 3.2 are listed and the corresponding signaling system requirements are summarized. The justification for these requirements can be found in Section 3.2. In practice, such a list of requirements is seldom complete. Special, tailor-made requirements do arise often. Nevertheless, the list indicates the kinds of factors imposed on a network by the signaling system that must be considered.

Table 4-2. Signaling System Requirements for Military Networks

Characteristic	Signaling Requirement
Survivability	Furnish redressing functions (e.g., adaptive routing, network reconfigurations) to restore critical links during catastrophic failures.
Security/Privacy	Control encryption/decryption devices and identify source, destination and routing.
Availability	Administer priority controls and reroute/accept overflows.
Adaptability	Handle wide range of data rates and traffic classes with signaling blocks independent of message lengths. System should be open-ended to add network elements and have flexibility to introduce new functions.
Compatibility	Applicable to half- and full-duplex operations with mode/code/format and speed conversion.
Special Services	Spare signaling alphabet or extendable formats and spare link capacity to provide new features.
Accuracy	Variable redundancy for error detection with automatic correction (see Sec. 4.2).
Efficiency	Monitor and vary traffic flow for optimum throughput, locally or overall.
Reliability	Able to reconfigure network elements to avoid faults and minimize downtime.
Performance	Signaling must be error free and transparent. User information and control information must be readily distinguishable (also see Sec. 4.3).
Cost	Simplicity for numerous terminals. Complexity permitted for more intelligent terminals. Common discipline for integrated networks. Accounting of services supplied.

In order to meet these requirements, there are a number of signaling issues and tradeoffs that must be resolved. These issues are summarized and listed in Table 4-3. Two categories of issues, technical and operational, are shown in the table.

4.2. Advanced Signaling Techniques

Past and present day signaling systems use direct current (dc), multifrequency (MF) or single frequency (SF) techniques. They are known to suffer from slow speed limitations, insufficient number of unique signals, false operations, fraudulent manipulations, and mass trunk seizures (Dahlbom, 1977).

All of these limitations can be eliminated with advanced signaling techniques. These are possible for modern electronic switches that are controlled by processors and stored programs. The related common channel signaling systems have been introduced and are becoming more widespread. Common channel signaling is defined by the CCITT as follows (CCITT, 1977d):

> "A signaling method using a signaling link common to a number of channels for the transmission of all signals necessary for traffic via these channels."

Since future military networks are expected to evolve into configurations that employ digital terminals, digital switches and digital transmission facilities, network control must also undergo a transition into the digital world. This means that the processor-controlled switches must communicate with each other using digital signaling systems. For economic reasons, several information channels can share a common signaling channel. When the voice digitization process extends to the subscribers instrument, digital signaling on a per-terminal basis must realistically follow.

The advanced signaling systems considered here are digital systems using common channel signaling between circuit switches (on the trunk side) and some form of digital signaling to and from the terminal (on the line side), on a per-terminal basis.

Table 4-3. Signaling Issues to be Resolved

<u>Operational Issues</u>

- clocked vs. non-clocked links
- circuit switching vs. store-and-forward switching vs. hybrid switching
- common channel vs. per channel signaling for circuit switching
- end-to-end vs. link-by-link signaling plan
- compelled vs. non-compelled signal sequences
- one-way vs. two-way trunks
- centralized vs. decentralized control strategy
- location dependent vs. independent numbering plan

<u>Technical Issues</u>

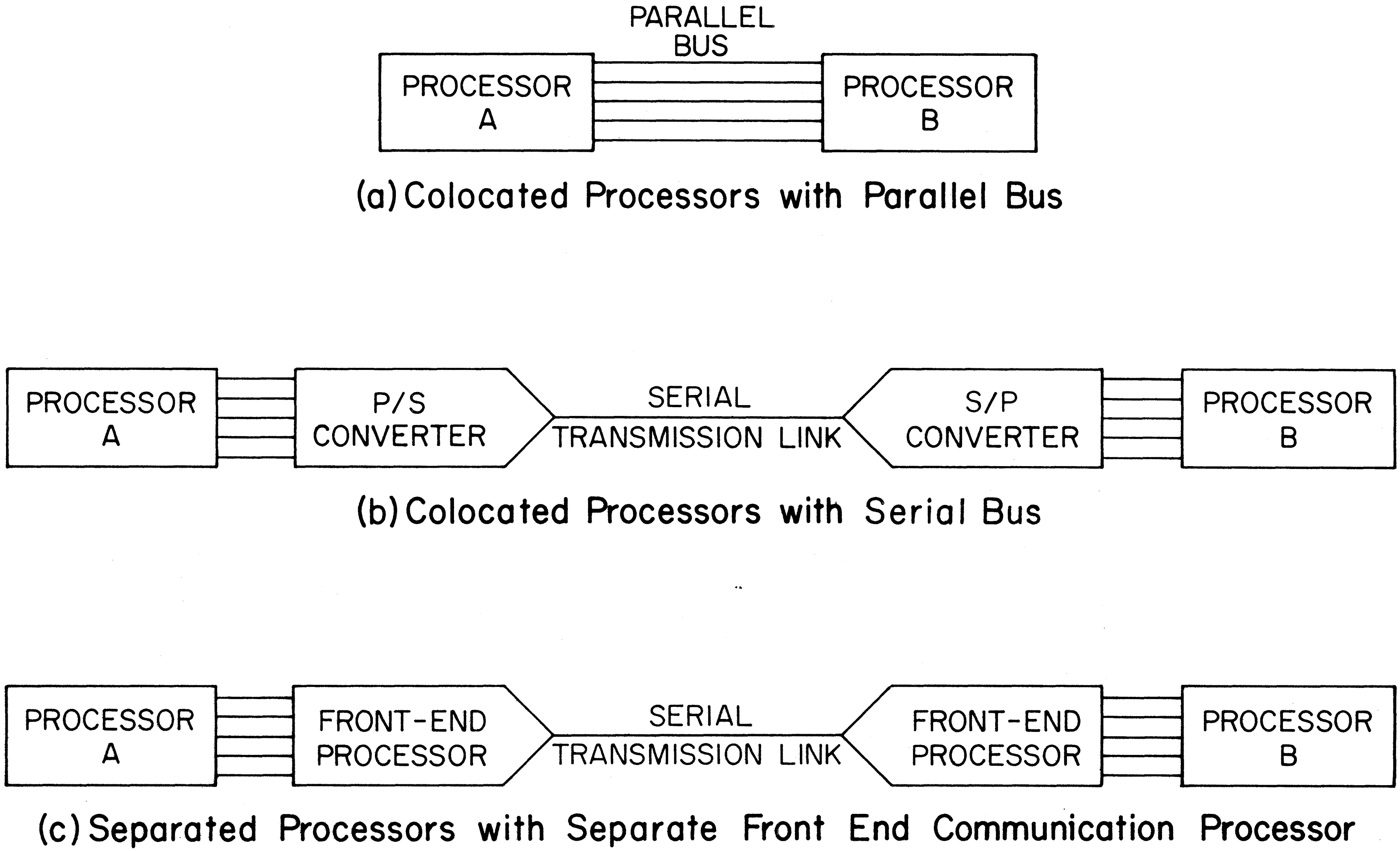
- parallel vs. serial transmission
- synchronous vs. non-synchronous formats
- bit-oriented vs. character-oriented codes
- forward error correction vs. correction by retransmission

Digital data transfers via store-and-forward type (e.g., packet) switches will undoubtedly continue to use overhead control information added to blocks of data. Both cases require inter-processor communications on the trunk side of the switch.

Internally, computer processing systems are logically organized for parallel interaction. The internal codes are usually some form of binary coded decimal, where four-bit characters are sufficient to represent one decimal digit. Two 4-bit characters may form a one 8-bit byte which is the typical internal code, although 16-bit, 26-bit and 32-bit bytes are also used. Each 8-bit byte represents one of 256 possible commands, instructions, symbols or numbers needed within a computer. One such set of 256 codes is the extended binary-coded-decimal information code (EBCDIC) used in IBM System 360 computers.

Although the internal transfer of information within a computer is often accomplished in parallel with the above 8-bit bytes, the external transfers either to other computers or to external peripheral devices are usually accomplished serially. The requirement for parallel to serial conversion depends on the separation distance of the communicating entities. Serial transmission becomes cost effective at larger separations. When processors are collocated, parallel transfer may be used to increase the speed and to avoid the need for parallel to serial converters.

Figure 4-2 depicts three methods for interprocessor communications. Since the internal codes are in parallel, a parallel bus can be used when processors are collocated or are a short distance apart. When they are separated as in Figure 4-2b, the interconnection over serial transmission facilities requires parallel-to-serial conversion at the transmit end and serial-to-parallel conversion at the receive end. These parallel-to-serial and serial-to-parallel conversions are accomplished with a communication interface. The control functions to be converted may be an integral part of the processor and use a dedicated data link for this purpose. Often a separate front-end processor



(a) Colocated Processors with Parallel Bus

(b) Colocated Processors with Serial Bus

(c) Separated Processors with Separate Front End Communication Processor

Figure 4-2. Interprocessor communications techniques.

is used as shown in Figure 4-2c. Other functions required for serial transmission are added in the front end, such as synchronization characters or bits, and error control information. Both processors exchange information serially over the link whenever serial is more efficient and less costly. The serial link, however, can be viewed as a means for transferring parallel data but usually at a reduced speed.

Serial transmissions may be synchronous or non-synchronous. With non-synchronous systems, each code word, or byte, or character may occur independently of the others. Start and stop bits are added to distinguish them. This added overhead reduces the link efficiency. Synchronous code formats utilize the line more efficiently but require bit, word, and frame clocking between the communication interfaces.

In the following subsections it is assumed that the signaling communications between switch processors is basically the same as between any separated general purpose processors. The signaling system includes a front-end processor for the communications interface and the control information is transferred over some channel using serial transmission of synchronous data. Digital signals convey the control information. The transmission facilities operate digitally at baseband or in a quasi-analog form using modulation and demodulation (modem) techniques.

4.2.1. Modes of Operation

User information and control information proceed through the nodes of a network as electrical signals. It is necessary to distinguish between these two types of electrical signals. When all the control information is transferred on a separate channel dedicated for this purpose, the signaling is known as common channel signaling. When the control information is transferred on the same channel as the user information, the separation process becomes more difficult. The three most common current methods for separating the two types of information are:

a) To use a unique sequence of bits (characters) which are recognizable as control signals.

b) To allocate certain time slots in a time division multiplexed channel for control characters only.

c) To specify fixed positions in a sequence of bits strictly for control information, perhaps of distinguishable length.

These three methods and the separate channel method are illustrated in Figure 4-3. Each method is described in more detail in the following paragraphs.

Unique Code Words

Word or byte discrimination can be achieved by adding a qualifier bit to each word or byte. The qualifier bit could always be a 1 for user information and a 0 for control information, for example.

Alternative forms are the EBCDIC 8-bit code and the American Standard Code for Information Exchange (ASCII). Both of these codes allocate certain sequences of bits for control characters and other sequences for alpha-numeric characters. The ASCII code, for example, is a 7-bit code with a possibility of 128 distinguishable characters. Wherever bits 6 and 7 are both zero, the remaining five bits indicate one of 32 possible control characters.

One or more redundant parity bits are often added to a character-oriented code for error control purposes. The American National Standard Institute's data link control (DLC) code is defined as a 7-bit per character code with an odd parity bit added for synchronous transmission and an even parity bit added for non-synchronous operation (Gray, 1972). When these control characters are transmitted, the corruption of even one bit upon reception can convert control information to user information and vice versa. These false indications cannot be entirely avoided, when all possible code combinations are available to some users on the same link, or when transmission errors occur. A

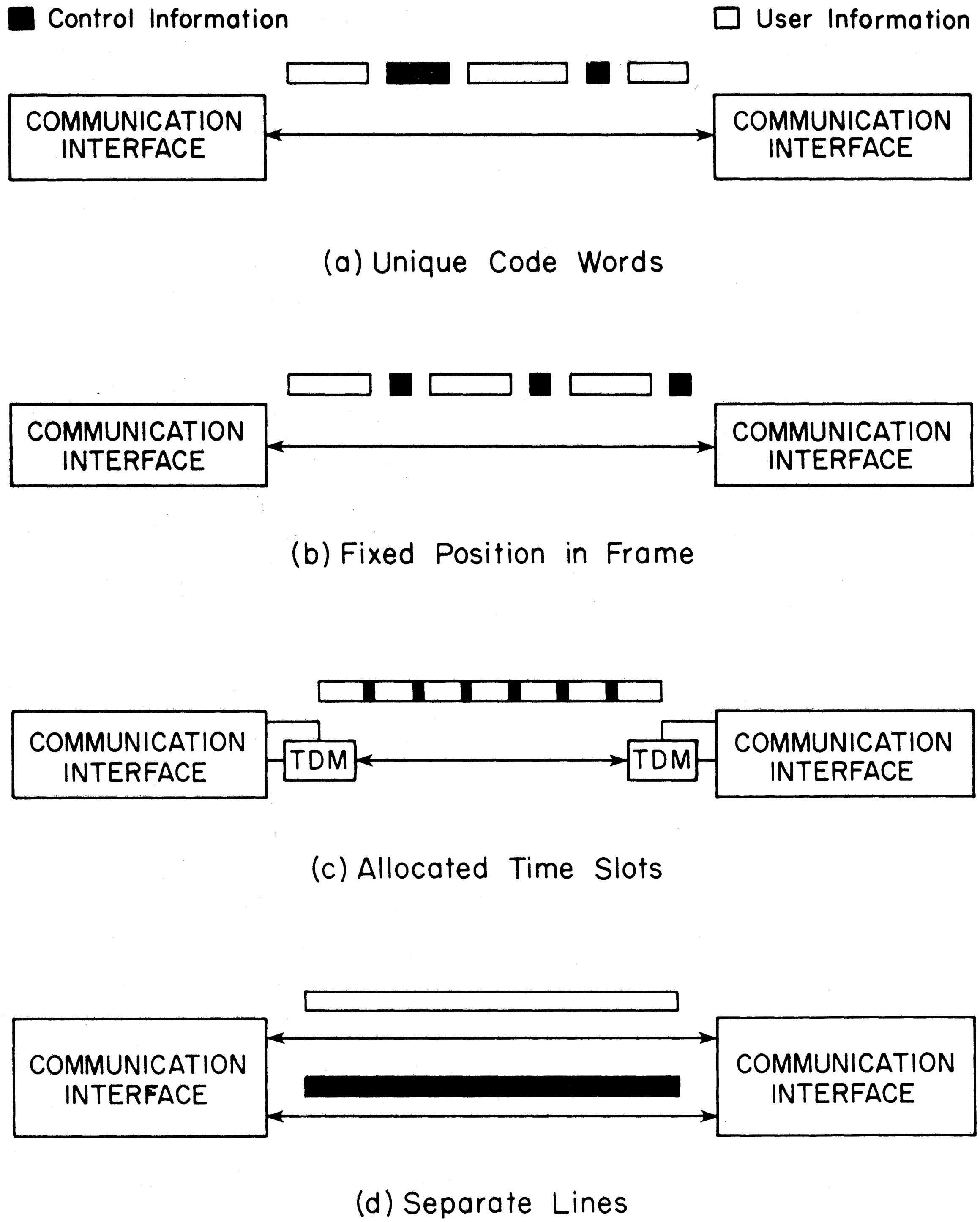


Figure 4-3. Methods for distinguishing user and control information.

similar character confusion problem occurs for traditional in-band signaling on a telephone link.

Fixed Position in a Frame

The AUTODIN I message switch assigns control characters to fixed position format within frames. Idle line characters are transmitted continuously to establish and to maintain bit synchronization and character framing between the terminals and the switch. Whenever a message is to be sent, the idle characters are interrupted and a header is transmitted to indicate the start of a block. The switch recognizes this start character, as well as subsequent control characters by their prescribed order in the block. The control character codes are the same as other message characters. Because of their unique positions in the block the two character types can only be confused when timing errors occur. Note that a 7-bit character code permits 128 control characters in this format.

This technique has several disadvantages. Due to buffer storage a delay is required to detect control characters and to correct errors. Throughput is reduced by overhead and coordination procedures, and there is considerable terminal complexity (Hamsher, 1967).

Allocated Time Slots

At least three variations of digital signaling systems are used to allocate time slots in the transmission channel. Time division multiplexing interleaves both the user and control information on the same channel, and generates a continuous bit stream. The two types of information are distinguishable by their time occurrence and synchronization is essential.

The slot allocation method defines the signaling technique. Three such methods are defined by the CCITT (1973a):

a) Speech digit signaling - Certain digit time slots of digitized speech are periodically used for signaling.

b) In-slot signaling - Signaling associated with a channel and transmitted in a digit time slot permanently (or periodically) allocated in the channel time slot.

c) Out-slot signaling - Signaling associated with a channel but transmitted in one or more separate digit time slots not within the channel time slot.

Speech digit signaling is similar to traditional signaling systems, except that digital transmission facilities are used. For example, a typical telephone terminal currently uses dc signaling on the subscriber loop for line status control, and either dc pulses generated by a rotary dial or dual frequency tones generated by push buttons for addressing. Traditionally, the interswitch signaling over analog facilities uses an in-band frequency tone (2600 Hz) for trunk status information and multi-frequency signaling for addressing. The status tone is applied during idle conditions and removed during a busy condition. Associated with a switch there may be a common pool of senders. The function of a sender is to pulse 2 out of 5 tones for each address digit over a trunk. A detector-requestor combination (also from a common pool) at the receive end detects these tones and makes the appropriate next circuit connection. Over digital facilities (allocation method (a) above), the idle and busy conditions are distinguished by transmitting a 0 or 1 state during a designated time slot in the digitized speech channel. The in-band, multifrequency address tones are coded the same way as voice signals.

In-slot signaling (method (b) above) is used with T-carrier systems as described in Section 2.4.

Out-slot signaling is used with the international digital carrier systems recommended by CCITT. It was also described in Section 2.4. Other examples of in-slot and out-slot signaling systems are discussed in Section 5.

4.2.2. Signaling on Separate Links

When a separate channel is used to transfer the signaling information for several user channels, the technique is known as common channel signaling or common channel interswitch (interoffice) signaling (CCIS).

Common channel signaling systems eliminate most of the limitations of speech digit and in-slot signaling systems. The control information is transmitted in digital form using baseband transmission facilities or quasi-analog facilities. A high-speed bidirectional link is used to transfer many forward and backward signals between processors. In a network with many nodes, the signals may be transferred on a link-by-link basis and processed at each node. This mode of operation is called the associated mode. The associated mode is cost effective when there are a large number of trunks between nodes which also terminate the common channel signaling system. When the number of trunks is reduced in parts of the network, the non-associated mode may be used.

There are two forms of non-associated signaling, fully disassociated and quasi-associated. The disassociated mode employs a completely separate network for signaling purposes. The nodes of this network, called signaling transfer points (STP), relay data from one signaling link to another. The quasi-associated mode combines the associated and disassociated modes. It is used in locations where direct trunk groups between switches are not large enough to support economically an associated extra CCIS link.

The three modes of operation: associated, disassociated, and quasi-associated, are illustrated in Figure 4-4. In the associated mode of Figure 4-4a the common channel signaling link along its entire length, including end points, closely tracks the interswitch trunk groups served. In the quasi-associated mode the common channel signals are not necessarily conterminal, with all trunk groups served as shown in Figure 4-4b. In the disassociated mode there is no close or simple association between the common channel signaling links and the trunk groups served. The disassociated mode permits nodes to communicate via signaling links even though there are no functioning connecting trunks. This capability is useful for performing network management

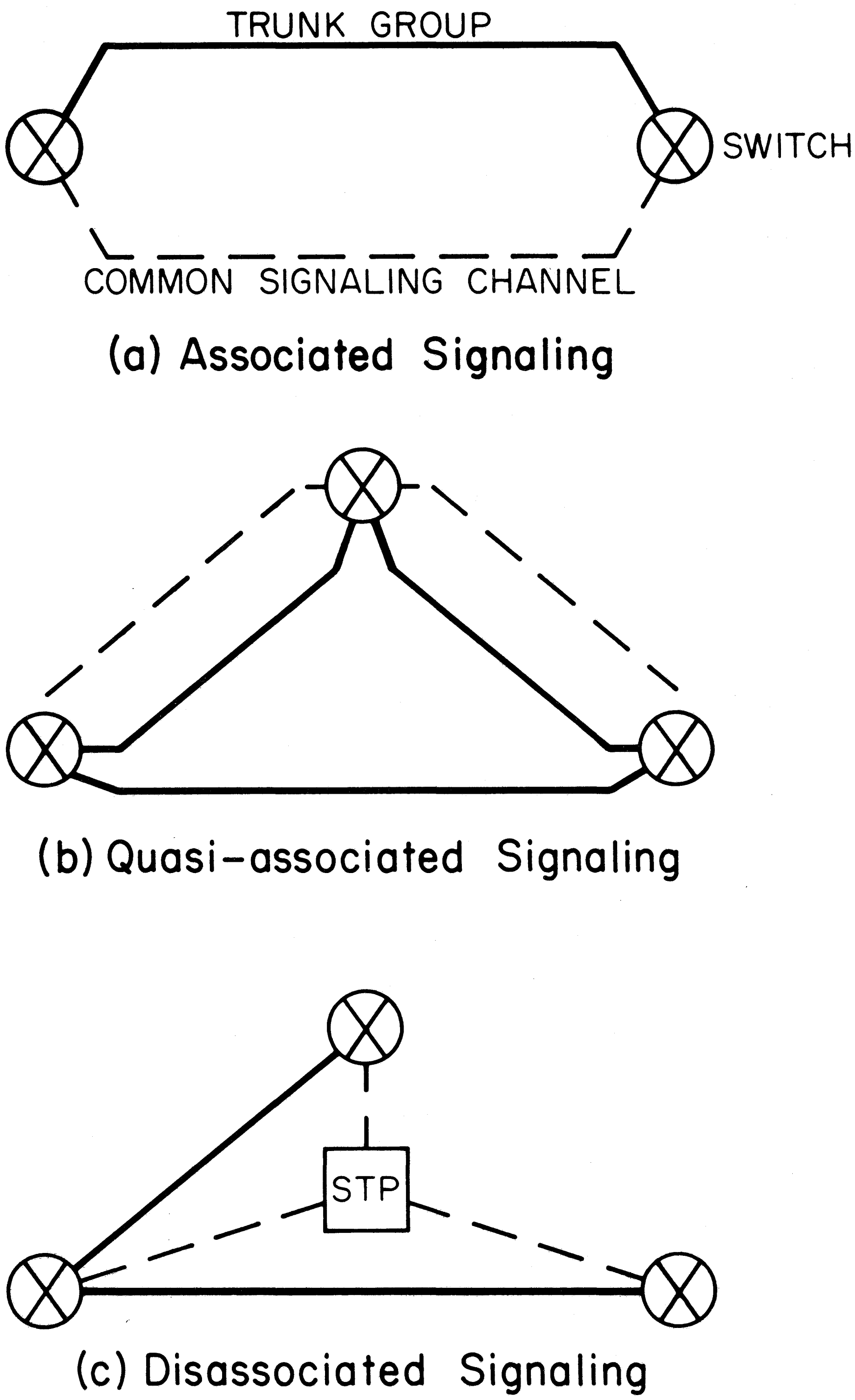


Figure 4-4. Modes of operation for common channel signaling systems.

functions as well as control functions. Examples of CCIS systems are described in Section 5.1.

4.2.3. Control Procedures

The procedures for controlling the flow of data between two terminals, as well as between two switching nodes, can be divided into discernible groupings or control levels called protocols. Five levels have been distinguished as follows (see Folts (1977) and des Jardines and Brosi (1977)):

Level 1. The physical level which provides mechanical and electrical specifications for communicating across a physical medium.

Level 2. The link logical control level which enables message sequences to be transferred across the physical link.

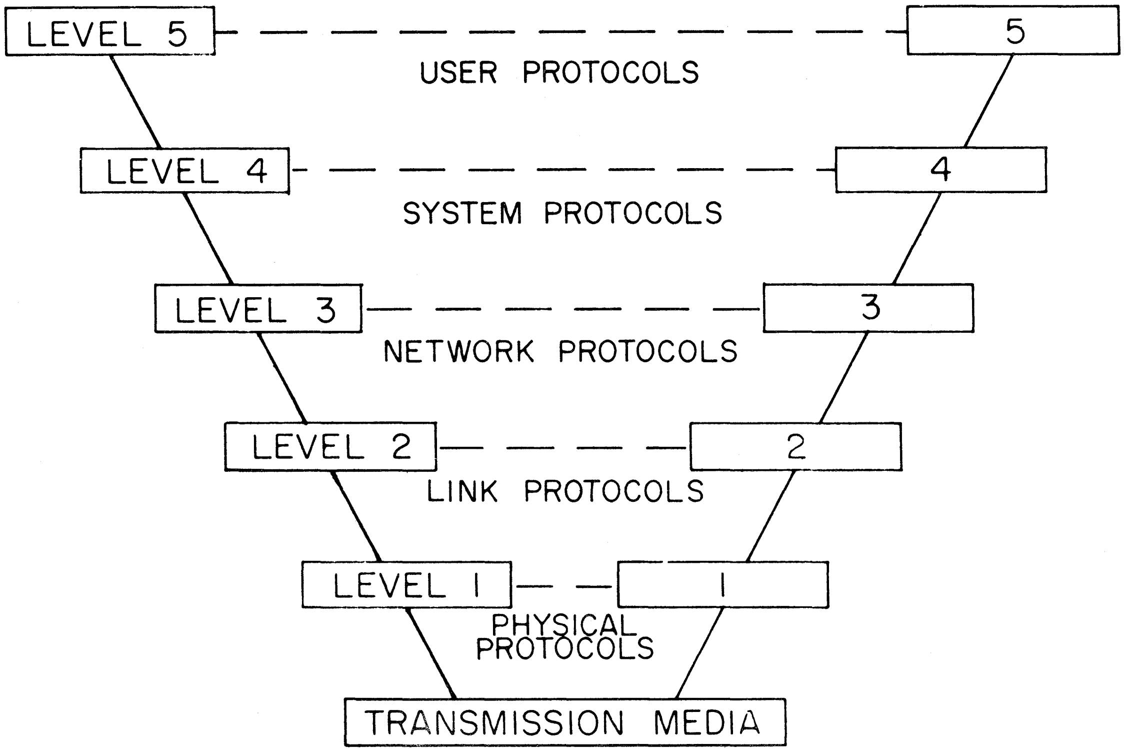
Level 3. The communication logical control level for establishing end-to-end connections through the network.

Level 4. The system logic control level to manipulate terminal resources.

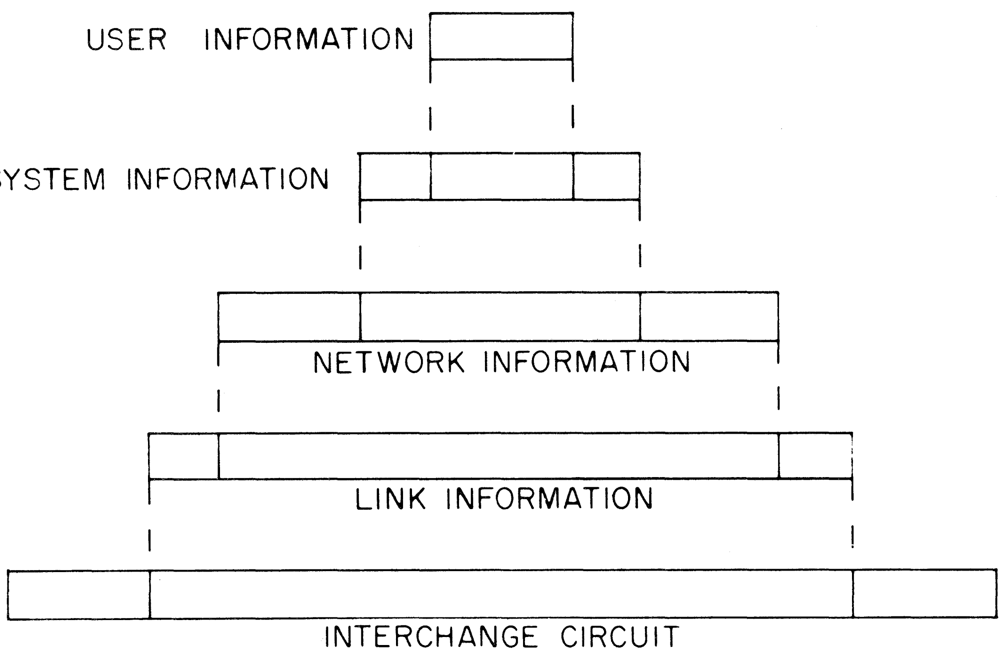
Level 5. The user level for exchanging user resources and networkwide resources.

It is possible to define interface protocols for all five levels at a single point in the network. The interface point is traditionally selected between approximately defined data terminal equipment (DTE) and the data circuit terminating equipment (DCE). Since each level depends only on lower levels and is independent of the higher levels, it is possible to standardize control procedures on a level-by-level basis.

The interface structure and protocol levels are indicated in Figure 4-5a. Messages containing user information or control information may be transferred across the interface layers by segmenting the message as shown in Figure 4-5b. The control overhead required for each protocol level is added starting at the highest level and then progressing outward to the lowest level. This message structure maintains the transparency required by the layered control functions.



(a) Interface Structure and Levels of Protocol



(b) Segmented Message Structure for Transfering Information Between Levels

Figure 4-5. Interfacing protocol structures.

A number of standard protocols have been defined by national and international standards organizations for the lower levels. These are discussed in Section 5.2.

The protocols may be implemented in hardware or software, or both. The protocols transfer user information and control information between node processors and between host computers, as well as among various terminals.

4.2.4. Synchronization

Synchronous transmission is usually employed in signaling systems because it permits more speed and reliability in control signal transfers. No transmission time is wasted, for instance, on the extra start and stop bits required in nonsynchronous systems. Various levels of synchronization must be recognized. They include bit, byte, word, character, block, entire message, or frame and multiframe synchronization.

Bit synchronization is required to aid the detection process, character sync to define the code, and block or frame sync to separate multiplexed codewords.

Various schemes are employed to maintain bit integrity over a network. The schemes fall into two categories: clocked and nonclocked. The clocked approach provides synchronization throughout the network by controlling the timing at each node from the same master clock. In a unclocked network the timing at certain nodes may be independent of other nodes. Buffers and bit stuffers are used to retime incoming bits to match a local clock.

The advantages and disadvantages of various network synchronization schemes are not considered here. Some synchronization techniques used in control signaling systems are included in specific system descriptions. A recent paper by Harrington (1978) examines the network time and synchronization requirements for various network topologies. The emphasis of Harrington's review is on circuit switched networks, however, he does consider message, packet and integrated voice/data networks synchronization.

Network timing can be derived by either synchronous or asynchronous methods. The synchronous approaches employ clocks slaved to a common master clock, with frequency averaging and external time references. Asynchronous approaches use atomic clocks with buffer compensation or less precise crystal controlled clocks and bit stuffing.

The synchronization technique to be employed on a given network depends not only on the topology, but on other factors such as reliability, survivability, and maintainability. The complexity of the synchronization technique can also have a considerable impact on cost (GTE, 1975).

<u>4.2.5. Error Control</u>

Transmission errors are known to occur due to noise and signal distortion on the link. Noise results from natural causes (e.g., lightning) and man-made causes (e.g., switching noise, interference from crosstalk and jamming). Signal amplitude and delay distortions are caused by link imperfections. Binary errors occur when noise distortion causes the information receiver to interpret a 1 as a 0 or vice versa. The resulting inaccuracies impact the network and its use in different ways. The accuracy required by a user depends on the type of traffic generated and its subsequent use. For example, accuracy is extremely important when transferring software programs, critical figures, such as targeting information and coordinates, and for signaling information such as destination addresses and priority information. Accuracy is less important with other types of information such as narrative, record and sensor data. When the latter information is transmitted in the clear there is typically sufficient inherent redundancy for a user to interpret the message correctly.

Digital voice transmissions pose different accuracy requirements. Acceptable error rates depend on the voice digitizatiion process used. For instance, a typical pulse code modulation

(PCM) system operating at 64 kb/s is at the intelligible threshold with end-to-end error rates of one in 100 bits. The same intelligibility can be achieved with 1 in 10^4 bit error rates using an LPC system operating at 2.4 kb/s.

The error rate on data links and signaling links can be reduced in a number of ways. Transmission error rates of 1 in 10^6 bits can be reduced to 1 in 10^9 bits with fairly simple error detecting and correcting schemes. More complicated error control schemes permit almost any accuracy level. All of these schemes involve redundant coding which increases overhead and reduces throughput.

Figure 4-6 shows a simplified breakdown of error control schemes. There are two basic schemes which have been used separately or together. Error correcting codes may be used in forward error correcting schemes (FEC) to recognize and to correct individual bit errors. Either block or convolutional FEC codes are applicable to blocks of data. Error detecting codes only indicate the existence of some unknown number of errors in a given block. Corrections can then be made by requesting a repeat (ARQ) transmission of that block. There is a small probability that some errors may pass undetected.

The simplest error detection scheme is a duplicate copy. A single parity check bit on a group of bits only provides protection against a single, or any odd number of bit errors in that group. More parity bits, as in cyclic redundancy checking, can protect against more extensive bursts of errors.

The FEC schemes require no reverse channel but are more complex to implement, especially at the decoder. The ARQ schemes require either a half-duplex or full-duplex channel.

In detection and retransmission schemes, the network retransmission procedures must be coordinated for all communicating stations. Once errors are detected in a block, the entire block is retransmitted until the error detecting code indicates no errors. Blocks must be numbered so that the transmitter and the receiver can tell which block is being acknowledged (ACK) via the

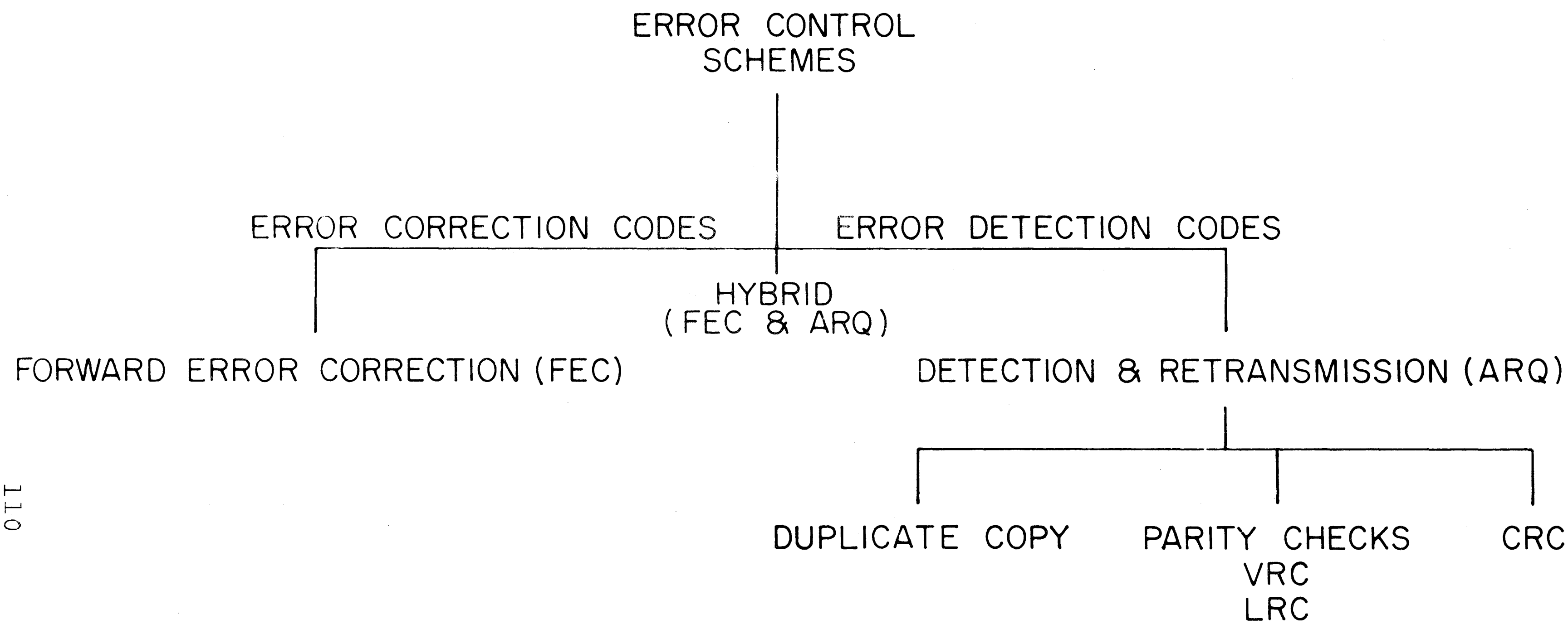


Figure 4-6. Error control schemes used on data links.

reverse channel. A negative acknowledgement (NAK) is returned for blocks received in error. When several blocks are outstanding on different links of the network the ACKs and NAKs are numbered corresponding to the appropriate block and pair of nodes.

Acknowledgement blocks may be tagged with the transmission block numbers. The ACKs and NAKs are returned using 0 or 1 bits in the appropriate position. The relatively short acknowledgement block may be part of a longer reverse information block. It may contain a joint or separate error check field.

The transmitting station must store outstanding blocks until a positive ACK is received.

Currently used control schemes are described in the following paragraphs.

Forward Error Correction

When a reverse channel is not available or its cost is impractical, forward error correction can be used to reduce the error rate over the channel. Even with the sophisticated codes required for FEC, however, the protection against errors is not great, especially during extended periods of channel disturbances. Thus, FEC has not found significant application for telephone network signaling systems over the terrestrial data networks (Burton and Sullivan, 1972).

With the FEC scheme, error presence is detected and the likely bits are corrected at the receive terminal or node. There is no need for retransmission. Several FEC codes have been developed to correct single and multiple errors in a block or in a convolutional stream.

One of the simplest FEC block codes is the Hamming code (Hamming, 1950). It can detect all double errors, or it can correct any single bit error. This code requires m check bits for every 2^m-m-1 data bits. Typical values for m are 3,4,5,....

Parity Checks

Processors often use internal parity checks. This checking can extend to the links as well. A single parity check on a byte, character, word, or any sequence of bits, is used to indicate whether the number of 1 bits in the sequence is an odd or even number. When used on a column of a data field, such a parity check is called a vertical redundancy check (VRC). The VRC fails if an even number of bits is in error in that column. Since multiple errors, including even numbers of errors, can occur fairly often the VRC parity is supplemented with a horizontal redundancy check (HRC). When HRC and VRC are used together as shown in Figure 4-7a, then an even number of bits greater than three must be in error in order to go undetected. A set of N 7-bit ASCII codewords is shown in Figure 4-7a. The ASCII code is similar to the ISO 7-bit code and the CCITT international alphabet No. 5 (see Davies and Barber, 1976).

The overhead or level of redundancy required in a combined horizontal and vertical code is fairly high. The ratio of check bits to information bits is given by:

$$\frac{8+N}{7N}$$

where N is the number of characters (see Fig. 4-7a).

For the ASCII code with 7 bits/character and a block length of 10 characters this ratio is

$$\frac{10+8}{7(10)} \cong 0.26.$$

For long blocks (i.e., large N) this ratio approaches 1/7.

The combined VRC and HRC schemes with retransmission is capable of reducing the error rate over a link by two or three orders of magnitude. Thus a data link with an error rate of 10^{-5} might have an undetected error rate of 10^{-7} to 10^{-8} (Martin, 1973).

Cyclic Redundancy Checks

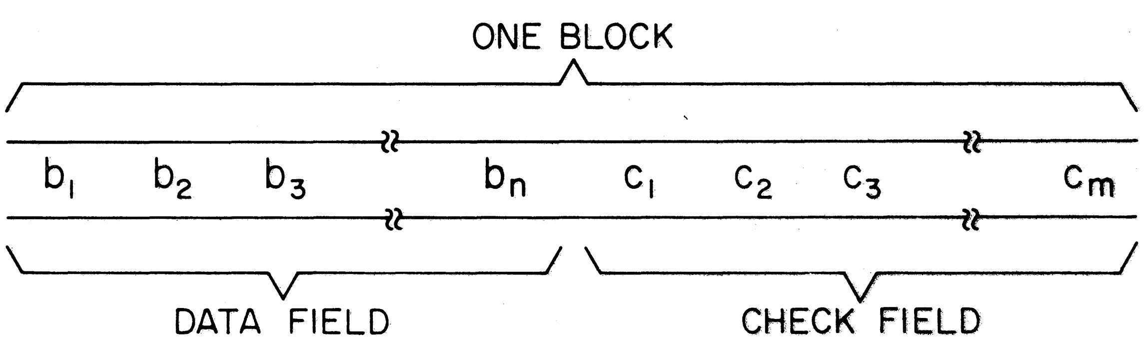
The cyclic redundancy check (CRC) scheme for block error detection is based on a redundant parity check field, as shown in Figure 4-7b. The data or information field $b_1, b_2, \ldots b_n$ is considered as one long binary polynomial

$$P(x) = b_n X^{n-1} + b_{n-1} X^{n-2} + \ldots + b_3 X^2 + b_2 X^1 + b_1$$

ONE BLOCK or RECORD

CHAR. POS. / BIT POS.	1	2	3	4	5		N	LRC PARITY
1	b_1							P_1
2	b_2							P_2
3	b_3							P_3
4	b_4							P_4
5	b_5							P_5
6	b_6							P_6
7	b_7							P_7
8	P	P	P	P	P			P_0

(a) Vertical and Longitudinal Parity Check Scheme



(b) Cyclic Redundancy Check Scheme

Figure 4-7. ARC, VRC and CRC error detecting schemes.

where b_i represents binary numbers and n is the total number of data bits in the block. The length of the check field c_i is m.

Encoding is accomplished by adding to P(x) a remainder R(x) such that the combination P(x) and R(x) is divisible by a selected generating polynomial G(x).

At the receiver, the entire received block is tested whether it is or is not a multiple of the generating polynomial G(x). If the remainder is 0, the received polynomial is divisible by G(x), and the message is assumed correct. If the remainder is not 0, some kind of error has occurred (Peterson and Brown, 1961).

Although the polynomial coding circuitry is considered to be more complex then the VRC or HRC scheme, the transmission efficiency can be higher. For a 3% redundancy (24 bit CRC remainder added to 800 bit block) the undetected error probability is always below 10^{-8}. For a 10% redundancy, on the same 800 bit block, the probability of undetected errors is at least 10^{-27} (Martin, 1973).

The length of the check field can vary with the number of data bits in a block. Common channel signaling systems CCITT No. 6 and CCIS use signaling blocks containing a 20-bit signaling field with an 8-bit CRC field. The proposed CCITT No. 7 system is expected to employ a 64-bit signaling field with a 16-bit CRC field.

Error control procedures for data transmission itself also can include some form of CRC. The Advanced Data Communications Control Procedure (ADCCP) for example uses a 16-bit CRC for a variable length data field. The AUTODIN II binary-oriented procedure Mode IV is similar to ADCCP, except a 32-bit CRC is used.

Error detection and correction schemes can be implemented using existing processor hardware or software when a computer is part of the system. A separate hardware implementation is relatively complex and expensive, but it reduces the load on the processor. A combined implementation simplifies the process by using software for check arithmetic. Whiting (1975) describes two software implementations, one emulating the hardware method and the other using a table of partial remainders for CRC schemes.

A hardware implementation of CRC using shift registers and an 8-bit check code of the form

$$G(x) = X^8 + X^2 + X + 1$$

is shown in the block diagram of Figure 4-8. This is the method recommended by CCITT (1977).

The shift registers of the encoder (Fig. 4-8a) are initially set to zero, gates A and B are enabled, and gate C is inhibited. All of the information bits and control bits in the block to be checked are clocked into the input. These bits generate the remainder within the shift register, while simultaneously appearing at the output. After the desired number of bits per block has been clocked in, the A and B gates are inhibited (the source is stopped!) and the gate C is enabled. The register is then shifted for eight additional counts. This appends the remainder, i.e., the required check bits, to the output sequence.

The corresponding decoder is shown in Figure 4-8b. The storage registers are initially set to zero, block synchronizing, and gates A, B and E are enabled while gate D is inhibited. The information bits of the word to be checked are clocked in. Gates A and B are inhibited, gate D is enabled. The eight check bits are now clocked in. If the block contains no errors all inputs to the error detector will be zero. Otherwise, some 1's appear at the error detector and retransmission is requested.

The generating polynomial recommended by the CCITT No. 6 system detects all 1, 2, and 3 bit errors in a 28-bit signaling word, because the minimum distance between code words is 4.

4.3. Performance Aspects

Section 3.2 noted several distinctive characteristics of the military switching environment and the importance of the network control system for that environment. Here, the relevant performance aspects are explored. There are several ways in which

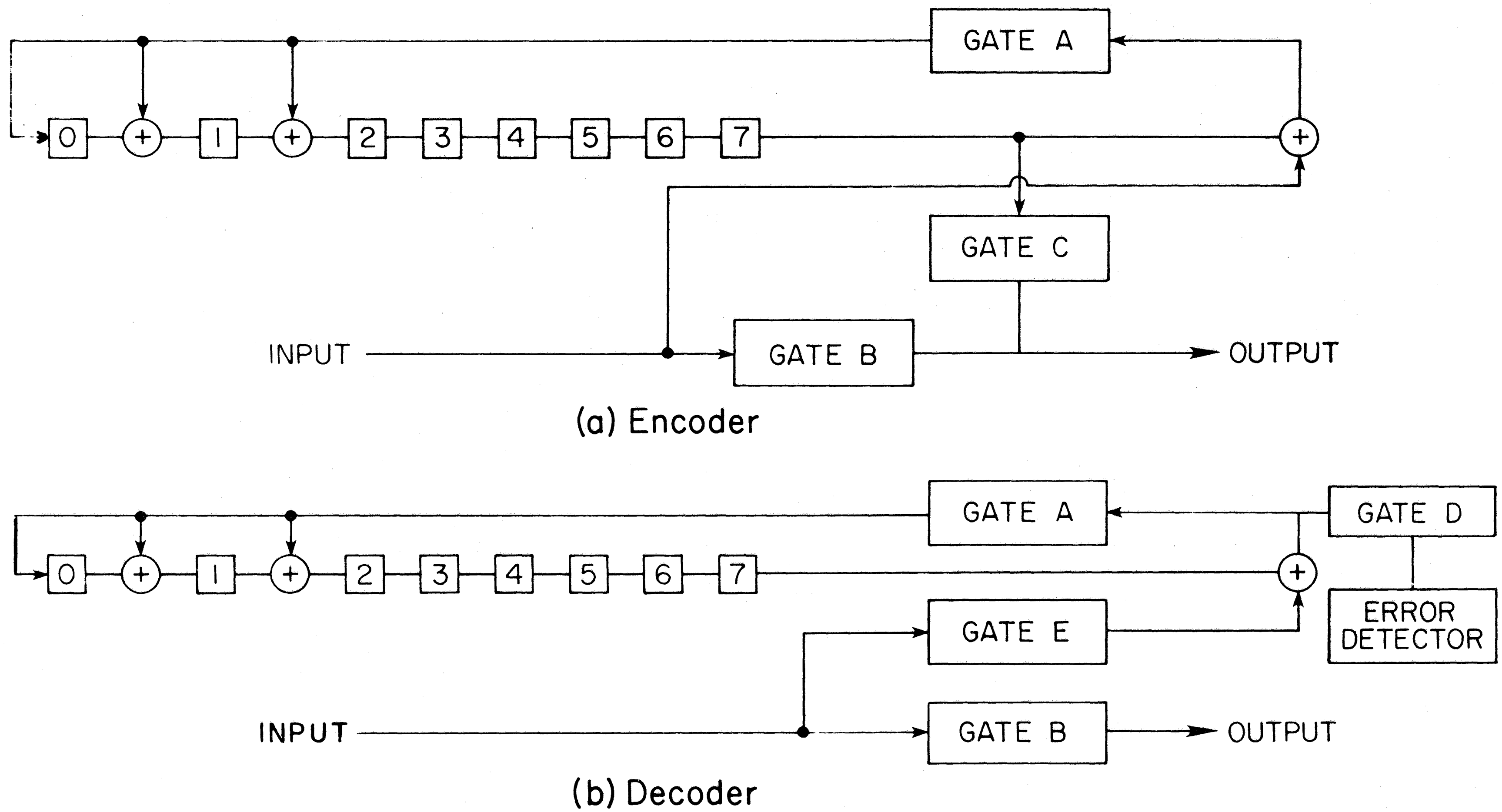


Figure 4-8. Cyclic redundancy check system.

the control signaling system impacts on user performance. These ways are reviewed here, together with some quantitative numbers for the state of the performance of different types of signaling systems.

The user performance parameters are specified at the user/system interface. This interface may differ from the points at which the system performance measurements are made. The block diagram of Figure 4-9 clarifies the distinction between these interface points. The network information transfer interface is between the data terminal equipment (DTE) and the data circuit terminating equipment (DCE). Physical, electrical, and protocol standards are usually defined for this point.

In contrast, the user performance criteria are defined at the user/system functional interface. As an example of this distinction it is useful to consider a simple teletypewriter-to-teletypewriter communication service between two operators over a dedicated transmission line. The user interface is between the operator and the teletypewriter and the information transfer interface is between the teletypewriter and the line interface unit. This line interface may be a modem for analog transmission facilities, or a time division multiplexer for digital facilities.

In the teletypewriter example, where a dedicated facility is used, control signaling is required. The physical and electrical interface and some form of link control (levels 1 and 2 of Sec. 4.1.3) are both needed, although there are no switch nodes present. The higher control protocols (levels 3, 4 and 5) are characteristic of the switched network environment.

Examples of level protocols, which either exist or have been proposed as standards, are described in Section 5.

The following subsections summarize the performance parameters for the user/system functional interface. The impact of the signaling system on these parameters should be determined at least on a qualitative basis.

4.3.1. User Oriented Performance Parameters

The selection of control strategies, as well as the signaling system itself, has an important bearing on the performance of a

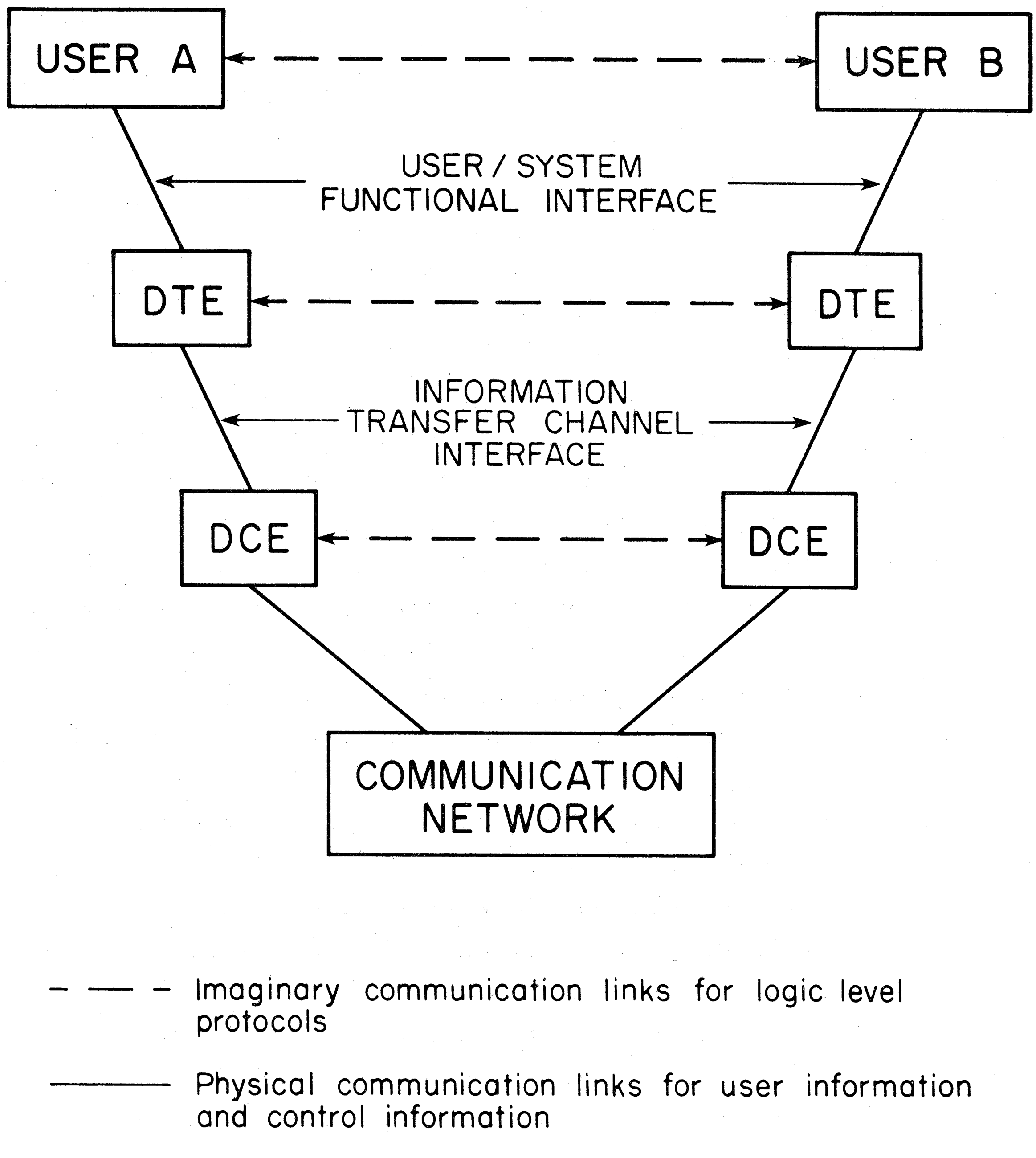


Figure 4-9. Interface points for user-oriented and technically oriented performance parameters.

telecommunication network from a user standpoint. Just how important the effects are depends on the criteria used to judge the performance.

User-oriented performance standards have been proposed by the Federal Telecommunications Standards Committee (FTSC, 1977) and by the American National Standards Institute (ANSI, 1974). These standards provide methods for describing the performance of digital communication systems in terms of the service they provide to the user. When specific numerical values are obtained by measurement or by calculation, the standards permit meaningful comparisons to be made between systems offering equivalent services, while using similar or different facilities and procedures.

It is important to examine at least one of these proposed standards in detail, so that the effect of the signaling system on specific performance parameters can be evaluated.

The development of proposed Federal Standard 1033 is described in detail by Seitz and McManamon (1978).

Table 4-4 lists twenty-six proposed digital communications performance parameters. Kimmett and Seitz (1978) have calculated values for these performance parameters for three different types of communication services. Their estimates are also shown in the table. The three services used in these examples are: 1) a non-switched private line, teletypewriter-to-teletypewriter communication service that connects only two operators, 2) a circuit switched teletypewriter-to-teletypewriter service connecting selected pairs of operators through the public switched network, and 3) a message-switched teletypewriter-to-teletypewriter service providing communication between pairs of operators through an automated store-and-forward switching center.

The performance values shown in Table 4-4 are calculated parameters based on assumed transaction profiles and performance outcomes. They are not intended to be used in characterizing or comparing actual offered services. They do, however, demonstrate how the three basic systems differ in performance.

Table 4-4. Compilation of Parameter Values for Proposed Federal Standard 1033

PERFORMANCE PARAMETER	NONSWITCHED PRIVATE LINE Example	CIRCUIT-SWITCHED EXAMPLE	MESSAGE-SWITCHED EXAMPLE
Access Time	5.2 Seconds	37.5 Seconds	15.4 Minutes
Incorrect Access Probability	0	1.1×10^{-4}	0
Access Denial Probability	1.7×10^{-3}	1.1×10^{-2}	6.8×10^{-4}
Block Transfer Time	68 milliseconds	68 milliseconds	16.2 minutes
Bit Transfer Time	68 milliseconds	68 milliseconds	16.2 minutes
Block Error Probability	1.5×10^{-4}	1.5×10^{-4}	2.5×10^{-4}
Bit Error Probability	$1.5x10^{-5} \leq P(bl_e) \leq 1.5x10^{-4}$	$1.5x10^{-5} \leq P(bl_e) \leq 1.5x10^{-4}$	$2.5x10^{-5} \leq P(bl_e) 2.5x10^{-4}$
Block Misdelivery Probability	0	1.1×10^{-4}	6×10^{-7}
Bit Misdelivery Probability	0	1.1×10^{-4}	6×10^{-7}
Block Loss Probability	4.7×10^{-5}	6.8×10^{-4}	1.7×10^{-3}
Bit Loss Probability	4.7×10^{-5}	6.8×10^{-4}	1.7×10^{-3}
Extra Block Probability	0	0	7×10^{-7}
Extra Bit Probability	0	0	7×10^{-7}

Table 4-4 (continued)

Extra Bit Probability	0	0	7×10^{-7}
Block Transfer Rate	5 blocks/second	14.6 blocks/second	36.2 blocks/minute
Bit Transfer Rate	35 bits/second	102.4 bits/second	4.2 bits/second
Block Rate Efficiency	23 percent	68 percent	2.8 percent
Bit Rate Efficiency	25 percent	68 percent	2.8 percent
Disengagement Time	0.5 seconds	2.25 seconds	1.5 seconds
Disengagement Denial Probability	4×10^{-4}	5.7×10^{-5}	1.1×10^{-4}
Outage Probability	8×10^{-4}	1.4×10^{-3}	3×10^{-3}
Service Time Between Outages	43 UIT hours	8.2 UIT hours	93 UIT hours
Outage Duration	1 hour	38 minutes	42 minutes
User Access Time Fraction	0.19	0.4	0
User Block Transfer Time Fraction	0	0	0
User Message Transfer Time Fraction	0.67	0.02	0
User Disengagement Time Fraction	0.8	0	0

This enables each to be designed to meet different end-to-end service requirements.

Payne (1978) has measured some of the user performance parameters on the ARPANET. These measurements include access time, incorrect access probability and access denial probability during the access phase, as well as disengagement time and disengagement failure probability during the disengagement phase. This measurement program is continuing, with emphasis on the more pertinent performance parameters during the information transfer phase.

Network control plays a major role in determining many of the performance parameters, but not necessarily all of them. The network topology, switch design and transmission state also affect performance.

Coviello and Vena (1975), in their justification for network integration, indicate that the performance, in terms of blocking probability for circuit-switched networks and in terms of queueing delay for store-and-forward switching networks, depends on the number of trunk facilities between nodes. In a military environment, the spare capacity required to handle the peak loads could be considerable. For example, although five voice channels may be sufficient to handle the average number of calls between two nodes, ten may be needed to achieve a 0.02 blocking probability for the actual random statistics of call arrivals. Thus, on the average, such trunks are utilized only 50% of the time. Due to the uncertainty of call statistics, many of the performance parameters can only be specified as design goals, not requirements. Some of these design goals are given in Section 5.

4.3.2. Impact of Signaling Systems on Performance

It is difficult to determine quantitatively the impact of the signaling system on all of the 26 performance parameters listed in Table 4-4. Yet, signaling must have some impact on nearly all of them, because without controls the user information cannot be transferred.

The signaling system has a major impact on the access phase and disengagement phase of a circuit switched network. However, these impacts need not be similar. For example, once (with the aid of CCIS) a path has been established through the circuit-switched network, the error control of user data no longer involves the common signaling channel. Error control is added to the data as overhead. In a store-and-forward switched network the path-establishment and error-control overhead do not follow separate paths but are appended to each block of information bits.

The affect of different dialing and signaling systems on the access and disengagement times of the public, circuit-switched, telephone network can be derived from the results of a recent survey by Duffy and Mercer (1978). The mean access time for direct-distance dialed (DDD) attempts is indicated in Figure 4-10 for a user with a dual tone multifrequency (DTMF) dialer. This dialing time is approximately doubled when a rotary dialer generates dc pulses instead. The connect time from end of dialing to ring return is almost 11 seconds for conventional signaling on a per channel basis (e.g., System R1) as shown in Figure 4-10. When all the switch nodes are controlled with SPC processors and common channel interoffice signaling (CCIS) is used, this connect time is expected to be reduced to 2-3 seconds. (Note: Signaling systems R1 and CCIS are described in Sec. 5.1.)

Table 4-5 summarizes the total mean access times and disengagement times for plain old-fashioned telephone service (POTS).

The message transfer time for these telephone networks depends on the distance. Most long-haul carrier systems have a propagation time of 0.004 ms/km. The average propagation delay as a function of connection length is given below (AT&T, 1970).

Connection Length	Distance in km	Average Propagation Delay in Milliseconds
Short	0 to 290	2.5
Medium	290 to 1170	4.0
Long	1170 to 4700	15.0

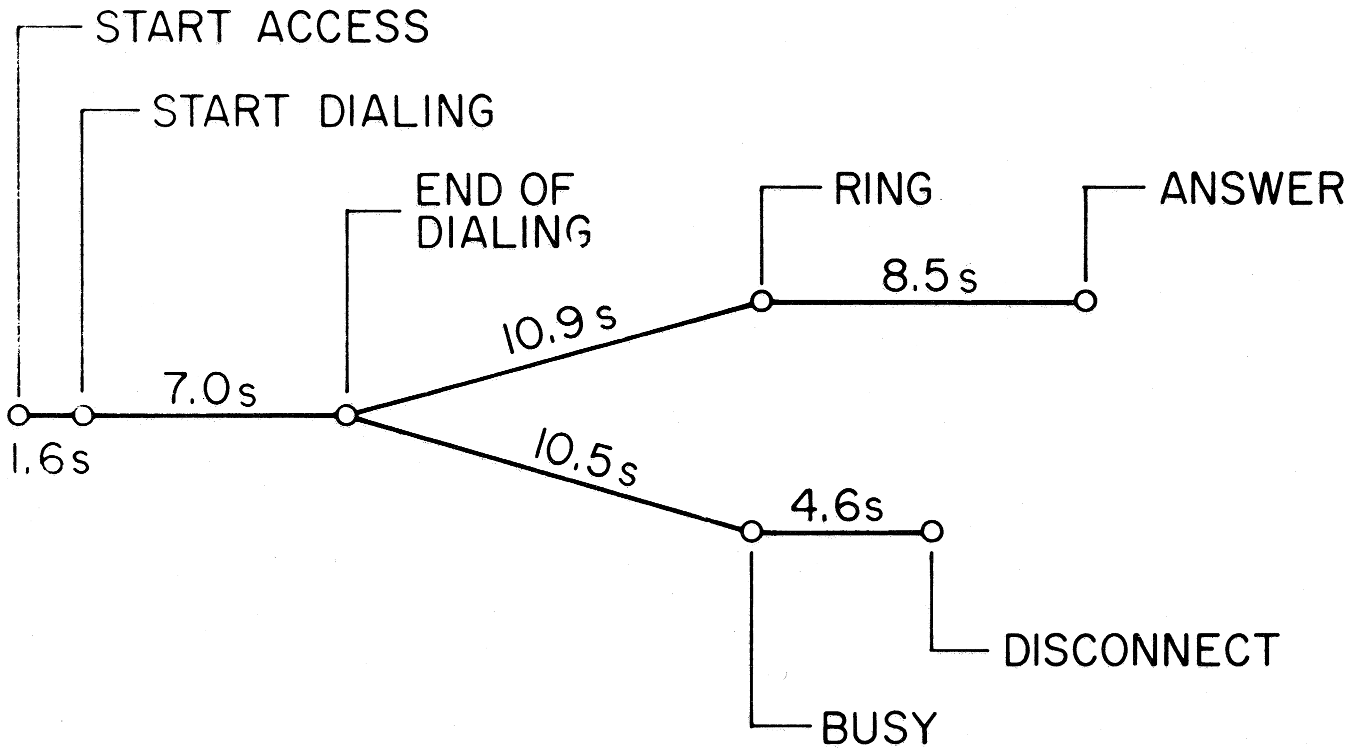


Figure 4-10. Mean access time on DDD network using pushbutton dialer and conventional signaling.

Table 4-5. Some Performance Parameter Estimates for Various Communication Networks

	Network	Procedure	Access Time	Transfer Time	Disengagement Time	Ref.
Voice	POTS	Rotary dial Conv. Sig	35 s	2.5-15 ms	4 s	(1) (6)
Voice	POTS	MF dial Conv. Sig	28 s	2.5-15 ms	4 s	(1) (6)
Voice	POTS	MF dial CCIS	20 s	2.5-15 ms	3 s	(1) (6)
Data	Nonswitched Private Line	No dial 150 baud-BSC	5 s	68 ms	0.5 s	(2)
Data	Circuit Sw. TTY	Rotary dial 150 baud-BSC	40 s	68 ms	4 s	(2)
Data	Circuit Sw. Comp. Term	MF dial 2400 baud-BSC	33 s	5 ms	3 s	(2)
Data	Message Sw. Off-line	150 baud Polled	15 min.	16 min.	1.5 s	(2)
Data	Message Sw. On-line	150 baud Datagram	0	16 min.	1.5 s	(3)
Data	Packet Sw. (ARPANET)	50 k baud Virtual Ckt. Serv.	9 s	93 ms	5 s	(4), (5)

(1) Duffy & Mercer (1978)
(2) Kimmett & Seitz (1978)
(3) Estimated
(4) Payne (1978)
(5) Kleinrock & Naylor (1974)
(6) AT&T (1970)

These average ranges of propagation delays are indicated in Table 4-5 for POTS.

Values of access, transfer, and disengagement times for several data networks are also shown in Table 4-5 for comparison purposes.

The transmission rates in b/s (bits per second) are related to baud speeds for each data network link. The baud is a measure of the carrier modulation rate or signaling speed for analog facilities. With a true binary system, 1 baud=1 bit/sec. The conversion from words/min to bits/sec is obtained as follows

$$b/s = w/m \times ch/w \times b/ch \times m/s.$$

Assuming 6 characters per word and 10 bits per character (including overhead bits for error detection), a 150 word per minute teletypewriter operates at 150 w/m×6 ch/w×10 b/ch×1/60 m/s or 150 b/s.

Three of the data networks use binary synchronous control (BSC) protocol for signaling the control information. BSC is a character-oriented link control procedure. The other data networks utilize bit-oriented link control procedures. These character and bit-oriented procedures are described in Section 5.2.

Table 4-6 summarizes the signaling system impacts on certain users performance parameters in a qualitative manner. It is assumed that entirely digital networks are used, and that the digital parameters apply to voice as well as to data communications. Although this table seems fairly subjective, it is useful in evaluating specific signaling systems for potential application to the AADSS. Those items that are denoted "major impact" should carry more weight in selection of suitable systems. Note, for example that the signaling system has little effect on the information transfer rate on a circuit switched network because signaling impacts only the access and disengagement phase. On a packet switched network providing datagram service the destination information is contained in each packet and thus added control overhead reduces the user's information rate.

Table 4-6. Signaling System Impacts on Performance of Digital Networks

Performance Parameter	Signaling System Impacts: For Circuit Switched Network			Signaling System Impacts: For Packet Switched Network		
	Major	Moderate	Minor	Major	Moderate	Minor
Access Time	X				X	
Incorrect Access Probability		X		X		
Access Denial Probability		X		X		
Block and Bit Transfer Time			X			X
Error Probabilities			X		X	
Information Transfer Rate			X	X		
Disengagement Time	X			X		
Disengagement Denial		X			X	
Outage Probability	X				X	
Service Time Between Outages	X				X	
Outage Duration	X				X	

5. EXISTING AND PROPOSED SIGNALING SYSTEMS

It was indicated previously in Figure 4-1 that there are different control procedures for different digital networks. In particular, the controls for store-and-forward switching differ from circuit switching networks. In this section these differences are examined in detail. Typical examples are given of signaling systems and their protocols. Numerous protocols have been standardized either commercially, nationally or internationally. Because many protocols are very similar to each other, we will select and emphasize one or two control procedures from each category.

5.1. Voice Network Protocols

The CCITT (1977a) has standardized a number of signaling systems for general use in international automatic and semi-automatic circuit switching networks. These are designated by serial numbers 3, 4, 5, and 6. Systems No. 3, 4, and 5 are designed to operate over analog trunks on a per-channel basis. All require in-band signaling. System No. 6 is a common channel signaling system which operates digitally over quasi-analog or digital facilities. The CCITT is also developing a common channel signaling system for synchronous data applications on public data networks. This new signaling system is expected to be officially designated as System No. 7 (CCITT, 1977b).

In addition, two regional signaling systems have been standardized by the CCITT (1977c). System R-1 is commonly used in North America and System R-2 is used in Europe.

All seven of these systems are briefly described below.

<u>System No. 3</u> - It is an international in-band system adapted for one-way operation. A single frequency 2280 Hz is used for both on-hook supervision and pulse type addressing.

<u>System No. 4</u> - This system is similar to No. 3, except that two in-band frequencies, 2040 Hz and 2400 Hz, are used for end-to-end operation over one-way links. System No. 4 is used exclusively in Europe.

System No. 5 - System No. 5 is the international in-band system for two-way operation. It uses two frequencies, 2400 Hz and 2600 Hz, for link-by-link supervision; and two-out-of-six frequencies (700, 900, 1100, 1300, 1500, and 1700 Hz) for addressing.

System No. 5 bis - The "No. 5 bis" system uses the same supervisory signals as No. 5, i.e., 2400 and 2600 Hz. The six in-band address frequencies however have now an expanded function. They provide forward and backward signaling of control information during the call establishment phase.

System No. 6 - This is the international system designed for digital two-way signaling over a common channel. The digital signals may be transmitted over a quasi-analog channel at 2400 b/s or over digital channels derived from the PCM multiplexer at 4 kb/s. The North American CCIS system is similar to System No. 6. The differences between the two will be noted in subsequent detailed descriptions.

System No. 7 - This system is being developed for general application in public synchronous digital networks for voice, data, integrated voice/data services. Proposed No. 7 characteristics are given in Recommendation X.60 by the CCITT (1977b), but many specifics are still under consideration. The functional structure of System No. 7 is divided into a user part and a message transfer part. This makes No. 7 useful for other communication needs besides control signaling. It is designed to operate over 64 kb/s digital channels but it can be used over lower rate channels.

System R1 - This is the standard system currently used on large portions of the U.S. telephone network. In many areas the system is being replaced with the common channel interswitch signaling (CCIS) system, that is similar to System No. 6. System R1 operates over analog or digital transmission facilities. The supervisory or line status signals use a continuous idle tone of 2600 Hz over analog links and associated in-slot signaling over T-1 carriers. The latter is done by borrowing the eighth bit of each channel, once per every 6 frames (see Sec. 2.4.3). The address digits are transmitted in-band, link-by-link, using two out of six frequencies. The six frequencies are 700, 900, 1100, 1300, 1500 and 1700 Hz. The system is also suitable for signaling on satellite links. No digital version of address signaling is specified in R1. However, the tones can be digitized like voice signals and handled as the so called speech digit signaling. Certain periodic time slots, normally used for digital speech, are then used for signaling.

System R2 - System R2 is another analog/digital signaling system. Its analog version uses out-of-band supervisory signals at a frequency of 3825 Hz. The digital version uses one of the 32 channels which make up the 2.048 Mb/s European digital carrier system. The address digits are in-band and continuously compelled. Forward signaling frequencies are 1380, 1500, 1620, 1740, 1860, and 1980 Hz. Backward signaling frequencies are 1140, 1020, 900, 780, 660, and 540 Hz. The out-of-band supervisory signals cannot be transmitted via an analog input to a PCM channel bank. However, a digital version of R2 has been developed. In this version, in-band address signals are applied to the analog inputs of the digital speech circuits.

Table 5-1 summarizes the characteristics and capabilities of the signaling systems recommended by the CCITT. It is apparent that Systems 6, 7, R1 and R2 are all suitable for digital networks, however, only 6 and 7 are designed for common channel signaling between digital switch nodes with SPC.

System No. 6, CCIS, and System No. 7 are described in the following subsections. The basic protocols for CCIS and System No. 6 are the same. The differences noted are largely due to the

Table 5-1. Summary of Telephone Signaling Systems Recommended by the CCITT

System	Supervision		Addressing	Suitable For				Potential Uses				Reference
	In Band	Out of Band	In Band	Analog Transmission	Digital Transmission	Common Channel	Time Division Switch Nodes	Operation	Satellite Link	Direction		
No. 3	2280 Hz	None	SF Pulsing	X				End to End		1 way		CCITT, 1960
No. 4	2040 Hz and 2400 Hz	None	SF Pulsing	X				End to End		1 way	Europe	CCITT, 1973a
No. 5	2400 Hz and 2600 Hz	None	MFDT	X				Link by Link	X	2 way	US and Europe	CCITT, 1973b
No. 6 (CCIS)	Binary Code	Binary Code	Binary Code	X (Quasi)	X	X	X	Link by Link	X	2 way	Europe (US)	CCITT, 1977a
No. 7	Binary Code	Binary Code	Binary Code		X	X	X	Link by Link	X	2 way	Proposes US & Europe	CCITT, 1977b
R1	2600 Hz	None	MFDT	X	X		One Only	Link by Link	X	2 way	US	CCITT, 1977c
R2		3825 Hz	MFDT	X	X		One Only	Link by Link		2 way	Europe	CCITT, 1977c

fact that System No. 6 was developed for use internationally, whereas CCIS is being implemented for domestic use in the U.S. on toll networks. System No. 7 is a newer system still under development. Its fundamental concepts are radically different from System No. 6 and CCIS.

Only the basic concepts, code formats, and operating procedures of these three common-channel signaling systems are defined in the following paragraphs. More details can be found in CCITT (1977a) for System No. 6, and in Ritchie et al. (1978) for CCIS. Some of the basic concepts of System No. 7 can be found in CCITT (1976), CCITT (1977b) and Appenzeller (1977).

5.1.1. Common Channel Interoffice Signaling (CCIS)

The CCIS system was introduced in the North American toll network in 1976. By 1980, CCIS is expected to control over 300,000 intertoll trunks between electronic switching centers (Cann, 1977).

The CCIS network is usually operated in a disassociated mode over quasi-analog links. The network is arranged as shown in Figure 5-1 which shows how CCIS controls interconnecting links between two separate regions. There are 10 such regions in CONUS, and two signal transfer points (STP's) within each region. The two STP's within a region are each connected to two switching centers, as well as to three other STP's, to provide redundancy in case of signal link or STP failures.

A typical signaling link is designed to operate at 2.4 kb/s. It can carry the signaling information for up to 3000 trunks. Normally, however, each CCIS link is assigned 1500 trunks. This means that in a case of failure, the load of the inoperative link can be added to another operating link. Signaling link operation at 4.8 kb/s has been contemplated to minimize delays and to increase capacity over busy links (Dahlbom, 1972). Usually, only two STP's are involved in establishing, maintaining and disengaging a call. This is necessary so that answer signal delays are acceptable during the transition to an all CCIS network.

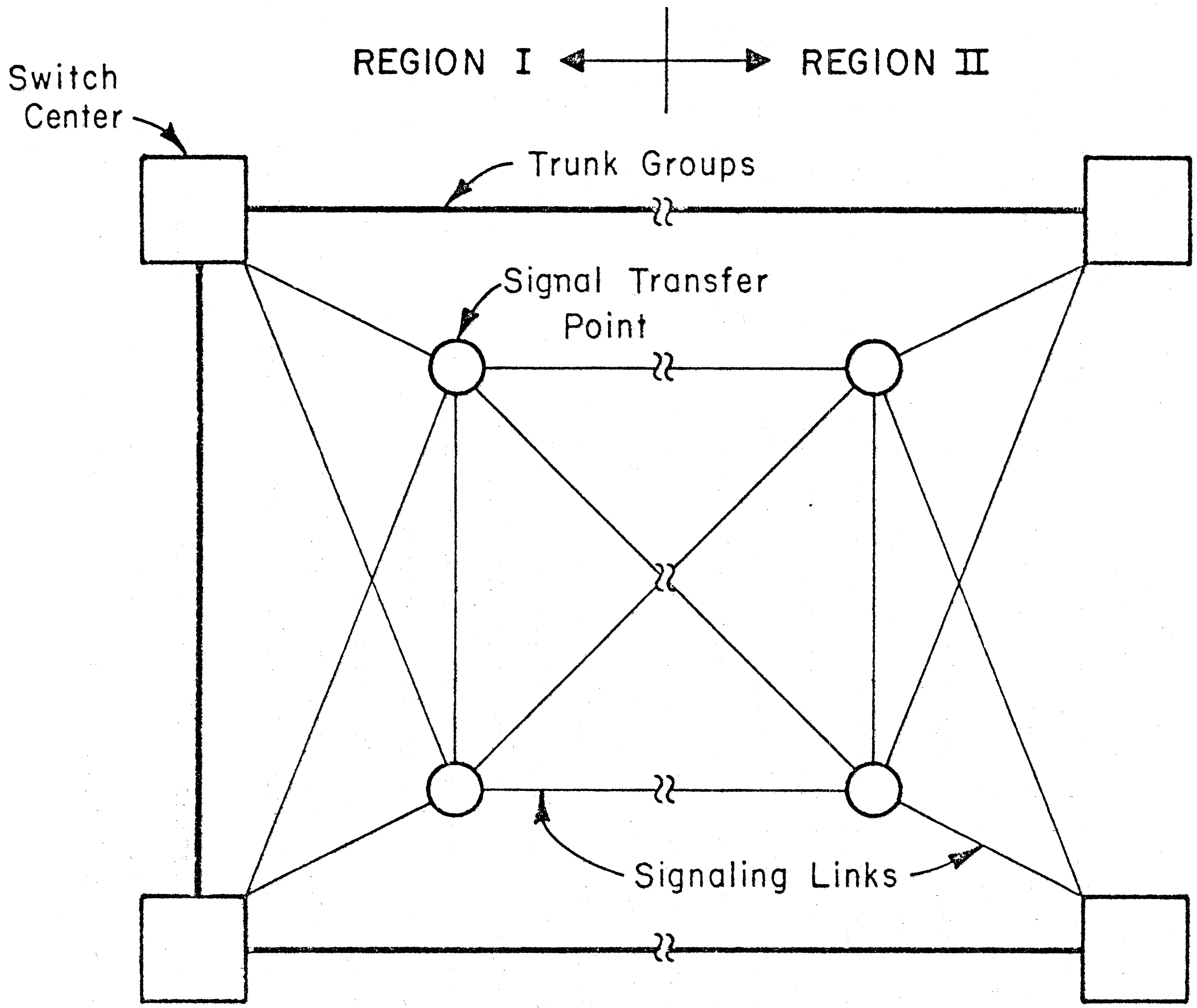


Figure 5-1. Network structure for disassociated common channel interoffice signaling (CCIS).

A block diagram of the CCIS system is shown in Figure 5-2. Separate links are identified as information transfer links, data links, and signaling links. Control information, generated in the processor, is in accordance with the CCIS signaling unit format. It is delivered in parallel, using either 8 or 16 bit bytes, to the block denoted signaling terminal. This terminal stores the outgoing information in an output buffer. Redundant error control coding is added to the outgoing signaling units. This error control is based on the previously described CRC, with error detection and retransmission. Each signaling unit (SU) contains 20 control information bits and 8 error check bits. A sequence of eleven signaling units and one acknowledgement unit (ACU) define one block.

The analog link interface equipment shown in Figure 5-2 modulates outgoing signals and demodulates incoming signals. The modem operates in the synchronous mode. Special synchronizing signaling units (SSU) are transmitted continuously, whenever other types of signaling units are not being sent. The SSU's maintain bit synchronization and SU synchronization, whereas the ACU is used to maintain block synchronization.

Upon reception and demodulation at a distant processor, the output data are delivered to a CRC decoder and the error check is performed. Error free signaling units are transferred to an input buffer of the signaling terminal where they are stored until they can be processed. Acknowledgement signals are returned to the originating terminal either to confirm reception of error-free SU's or to request retransmission of SU's received in error. Thus, all unacknowledged SU's must be kept stored at the originating terminal until a correct acknowledgement has been received.

The CCIS system, like any common channel system, requires a special continuity check feature over the trunks being connected. This feature is indicated in Figure 5-2 by the continuity check

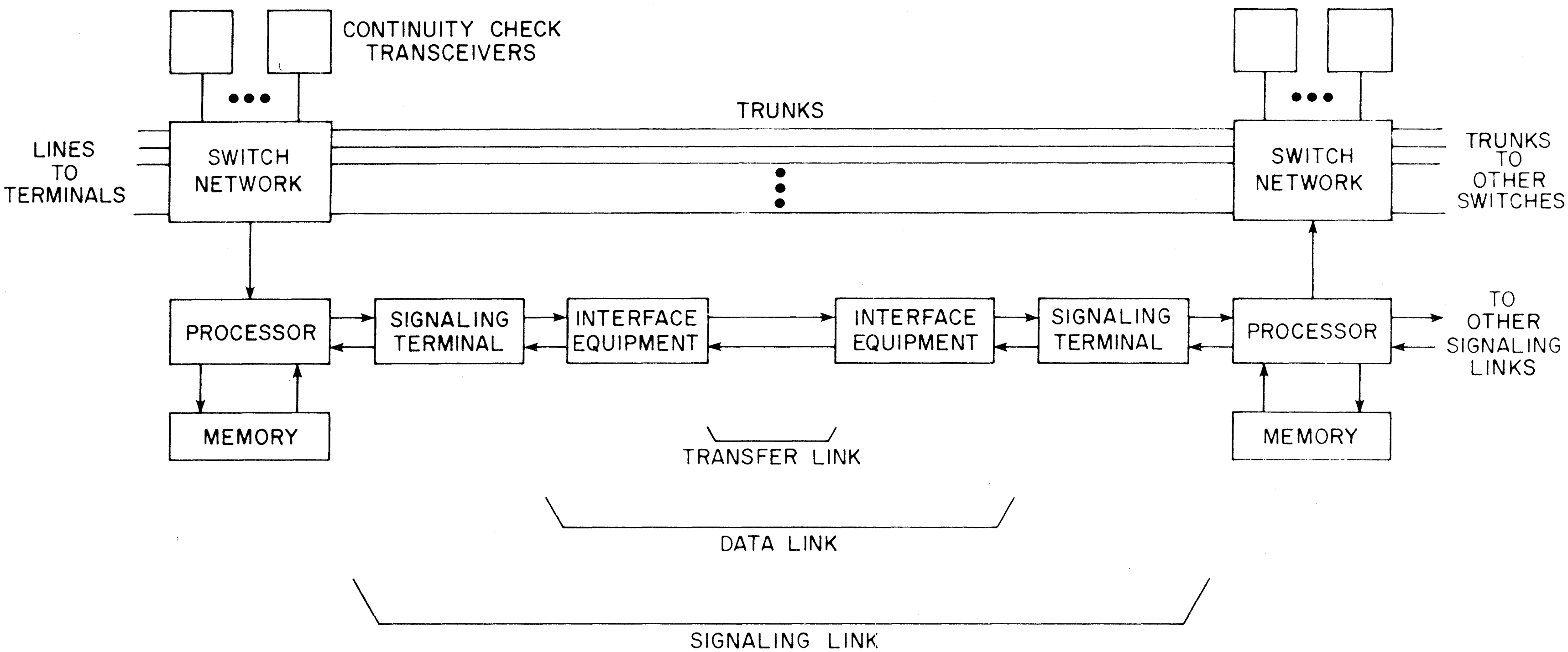


Figure 5-2. Control signaling via common channel signaling link.

transceivers. During the call set-up phase for a 4-wire trunk, one transceiver is selected from a common pool. It is briefly connected to the chosen trunk at the originating switch. A loop-back connection is made via the destination switch and a 2010 Hz signal is transmitted over the loop. A proper level return signal verifies that the trunk is operable. Over 2-wire trunks the continuity check signal is 1780 Hz outbound and 2010 Hz on the return. Transceivers are needed at both ends of the trunk for the 2-wire continuity check.

Signaling Codes and Formats for CCIS

The format used for the various types of signaling units in the CCIS system are indicated in Figure 5-3. As noted previously, each signaling unit consists of 28 bits, eight of which are parity-check bits. The other 20 always contain a three-bit header, which defines the type of unit being sent. This leaves 17 bits for various types of control information.

The three-bit headers occupy positions 1 through 3 of the SU. They are transmitted first. The heading code indicates eight different types of signaling units as follows.

Type	Acronym	Heading Code
Telephone Signals	LSU	000, 001 010, 100
System Control, Management or Maintenance Signals	LSU	111
Acknowledgement Signal	ACU	011
Initial Signaling Units	ISU	101
Subsequent Signal Units	SSU	110

The four LSU's used for telephone signals are further distinguished by signal information bits 4 to 7, that immediately follow the heading code. This is shown in Table 5-2. The remaining bits 8 to 20 contain the signaling information.

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28

BIT NUMBERING SCHEME

HEADING	SIGNAL INFO.	BAND NUMBER	CKT. NO.	CHECK

LONE SIGNAL UNIT (LSU) or INITIAL SIGNAL UNIT (ISU)

1 1 0	SIGNAL INFORMATION	CHECK

SUBSEQUENT SIGNAL UNIT (SSU)

0 1 1	ACKNOWLEDGEMENT INDICATOR	ACK. NO.	BLK. NO.	CHECK

ACKNOWLEDGEMENT SIGNAL UNIT (ACU)

1 1 1	0 1 1 1	0 1 1 1 0 0 0 1	DIV.	SEQ. NO.	CHECK

SYNCHRONIZATION SIGNAL UNIT (SYU)

1 1 1	0 1 1 1	CONTROL INFORMATION	CHECK

SYSTEM CONTROL UNIT (SCU)

1 1 1	SIGNAL INFO.	BAND NO. or MISC. INFO.	MGMT. INFO.	CHECK

ONE-UNIT MANAGEMENT SIGNAL

Figure 5-3. Basic format of signal units used with common channel interoffice signaling system.

Table 5-2. Signal Information Codes for LSU's

Header Codes Bits 1,2&3 / Sig. Info. Codes Bits 4,5,6&7	000	001	010	100	111
0000	X	Failure	Blocking Ack	X	
0001	Continuity	Refused	Unblocking Ack	X	
0010	X	Confusion	X	X	
0011	X	X	X		
0100	X	Trunk Congestion	X	Busy	Network Management (International)
0101	X	Switch Congestion	X	X	Signaling Management (International)
0110	X	X	Release Guard	X	
0111	X	X	Answer (no charge)	X	Signaling Management (Unbanded)
1000	Clear for	Reset	Answer (charge)	X	Network Management (National)
1001	Block	X	X	X	Signaling Management (Banded)
1010	Unblock	X	Clear Back #1	Address Complete No charge	
1011	X	X	Reanswer	Address Complete coin	
1100	X	X	Clear Back #2	Address Complete charge	
1101	X	X	Clear Back #3	X	
1111	X	X	Reanswer	X	

X indicates spare

If the header identifies an ISU (code 101), then bit 4 indicates the type of ISU, while bits 5 to 7 indicate the number of SSU's which follow. This multi-unit message (MUM) may contain up to eight subsequent signal units in tandem. It provides an efficient means for transmitting large groups of related information. Each SSU requires only a 3-bit identification header. The 17 remaining bits can all contain control information, such as address, line coordination, etc.

Signaling a Call

The CCIS steps involved in a successful completion of a 10-digit call are shown in Figure 5-4. Only the first 20 bits of each signaling unit are indicated in the figure. The other 8 bits are parity check bits. In North America, a 10-digit long distance call normally requires three decimal digits for the numbering plan area (NPA), three decimal digits for the central office code, and four decimal digits for the terminal as follows.

NPA	OFFICE CODE	TERMINAL
N B X	NNX	XXXX

where N is any digit between 2 and 9, B is 0 or 1, and X is any digit between 0 and 9.

At least 30 bits are required for the 10-digit address. However, it is simpler to assign 4 bits to each of the N, B, and X. Then, the above address consumes 40 bits. When a caller goes off hook and dials the destination address, the address is stored in a register by the processor. An initial address message is formed consisting of an ISU and an appropriate number of SSU's. The signaling information bits in the ISU indicate that four SSU's are to follow. This ISU contains 13 other bits (8 through 20), which are to designate the trunk. Thus, there are 8192 different trunk labels available.

The first SSU following the ISU contains information concerning the service desired, such as the satellite links

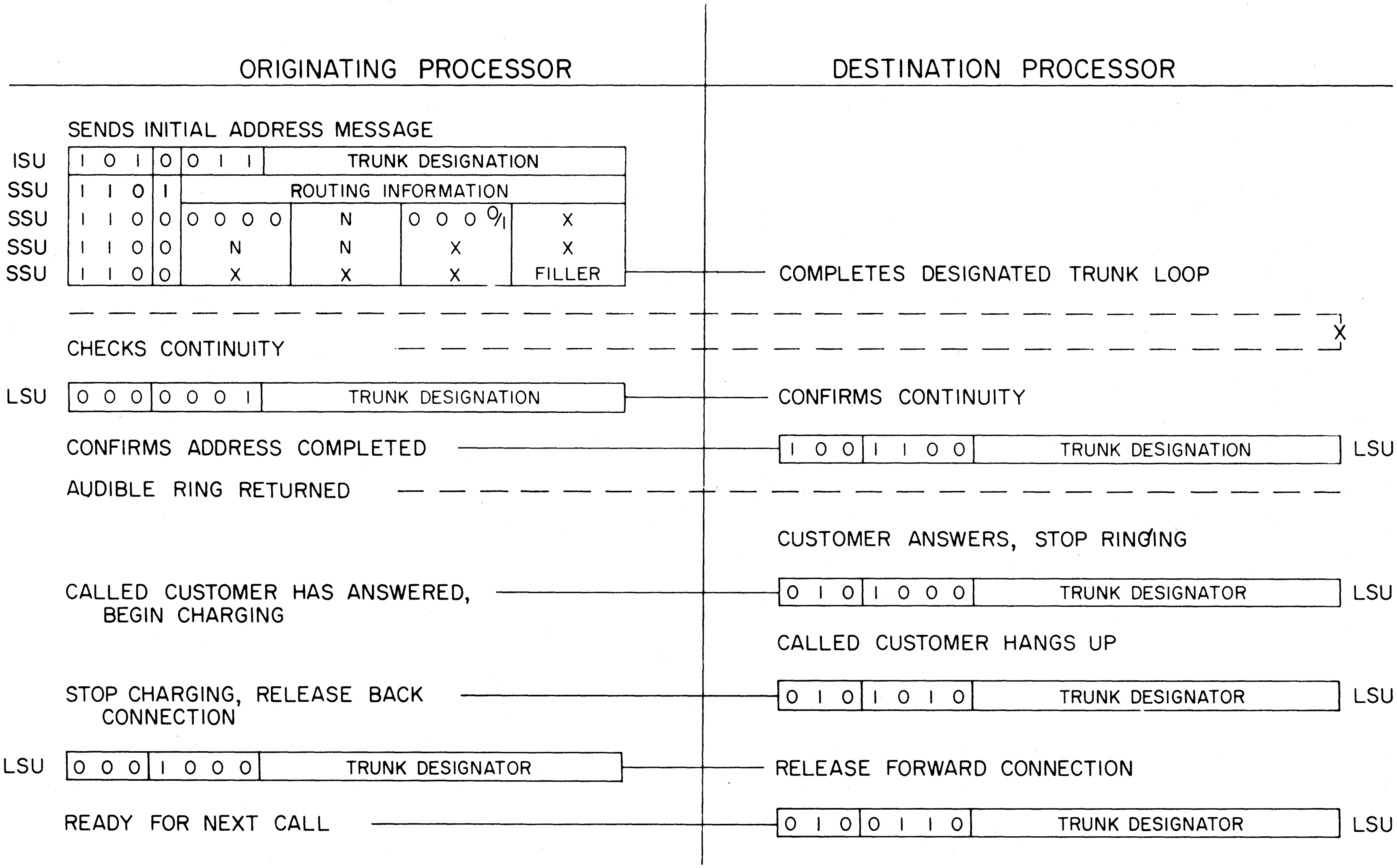


Figure 5-4. Steps required for 10-digit call using CCIS.

included, echo suppressors involved, or special line features required.

The next three SSU's contain the dialed 10-digit address. Upon receipt of this address, the destination processor closes the loop on the designated trunk to check continuity. An LSU from the originating office verifies the continuity check. Another LSU at the destination office confirms that the address is complete and rings the called station. An audible ring is returned to the caller. When the called customer goes off hook, an LSU is returned so that charging can begin. Either the calling or the called customer may hang up first. In either case, the nearest processor sends an LSU to indicate release, stop charging, and clear all the involved line and trunk units.

Release forward and release backward signals indicate that both customers are on-hook. A final LSU is transmitted to confirm that the trunk is ready for another call.

There occur, of course, many other situations which are not shown in Figure 5-4, such as customer busy, trunk congestion, inoperative number, etc. When these situations occur, an appropriate LSU is returned to the originating processor. It takes appropriate action and, if necessary, alerts the calling customer on the progress of the call.

The priorities for various types of signals are as follows.

Priority	Signal Type
1	ACU (Error ACK or NAK)
2	Fault Information
3	Retransmit Answer Signals
4	Answer Signals
5	Retransmit Telephone Signals
6	Telephone Signals
7	Retransmit Management Signal
8	Management Signals
9	SYU (Synchronization signaling unit)

These priorities establish the signaling sequence when conflicts arise at the transmitting terminal.

Signaling Block Structure

As noted previously the SU's are combined in groups to form a 12-unit block. The block structure containing the initial address message from Figure 5-4 is shown in Figure 5-5. Direction of transmission is to the right. Therefore, the blocks are numbered in reverse. Synchronizing signal units, SYU's, are sent periodically and whenever there is a need for fillers. Every twelfth SU is an acknowledgement unit (ACU). The ACU format contains the identifying header (011) and eleven acknowledgement bits, one for each SU in a block. A zero bit indicates positive acknowledgement, and a one bit indicates negative acknowledgement. These eleven bits are followed by three bits, which identify the number of the previous block being acknowledged, and three other bits indicating the sequential number of the block just completed by this ACU. Since the three bits can be coded for up to eight blocks, there may be this number of blocks in doubt, and outstanding on the link.

A typical CCIS link is operated at 2.4 kb/s data rate. Since there are 28 bits per SU and 12 SU's per block the acknowledgement total time required to receive 8 blocks is

$$\frac{28 \times 12 \times 8}{2.4 \times 10^3} = 1.12 \text{ seconds.}$$

Of this time (96 SU's), the time for 64 SU's is available between emission of an SU and subsequent reception of the ACU containing its acknowledgement. The number, 64 SU's is based on a CCITT recommendation (CCITT, 1977d). The remaining time for the other 32 SU's is required for processing. Thus the maximum loop propagation delay is about 750 ms. When negative acknowledgements occur, multiple loop delays must be allowed for.

Transmission

The CCIS signals are normally transmitted full duplex over a 4-wire link. An 1800 Hz carrier is modulated by the binary

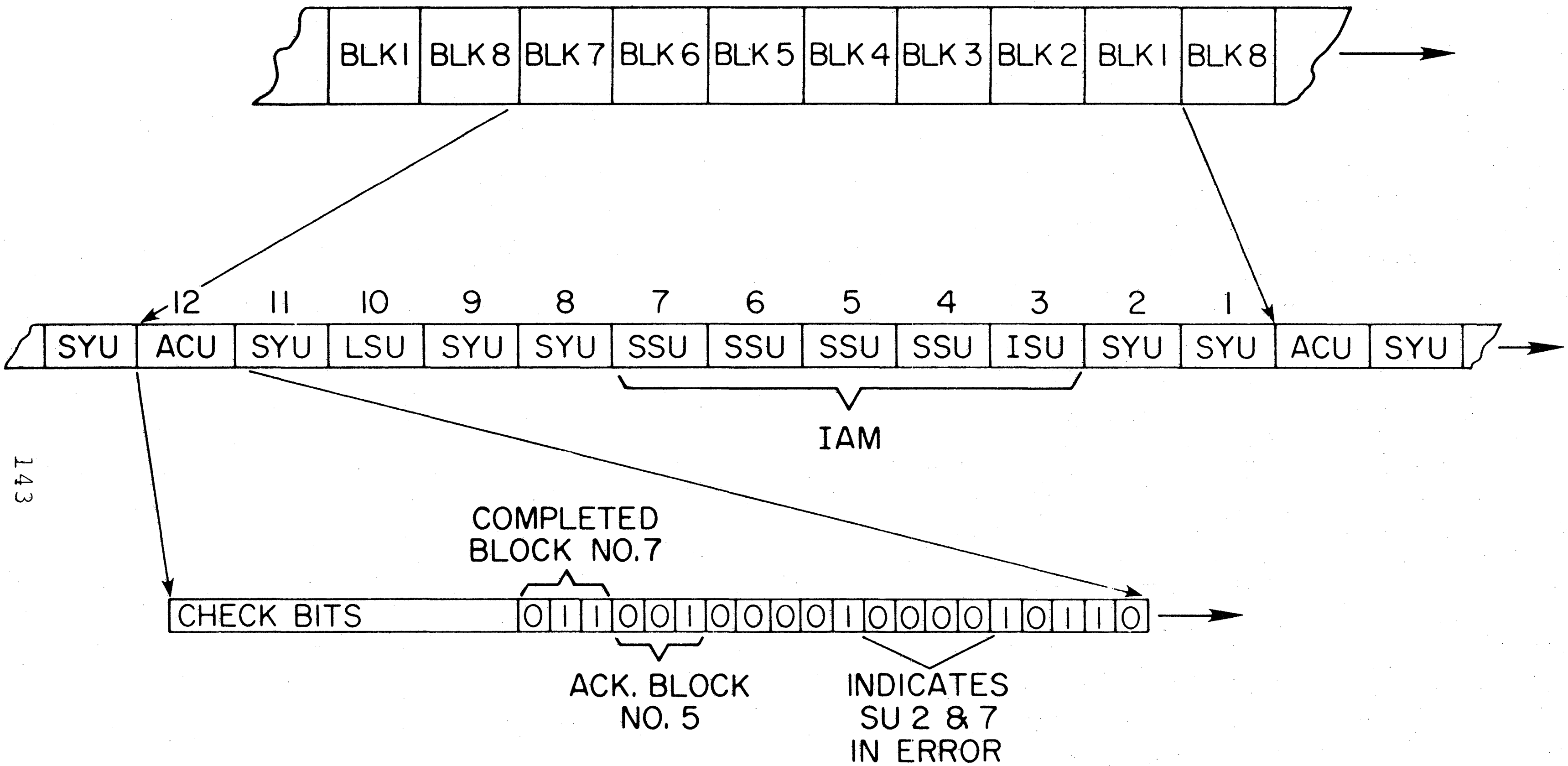


Figure 5-5. CCIS block structure.

signaling bits using a quaternary or four-phase shift keying (QPSK) at 1200 baud. Each differential phase increment contains two information bits as follows:

Bits	Phase Differential
00	45°
01	135°
11	225°
10	315°

The actual bit rate is 2400 b/s.

The modem characteristics are summarized in Table 5-3.

Table 5-3. CCIS Modem Characteristics

Modulation:	differential four-phase
Modulation Rate:	1200 baud
Bit Rate:	2400 b/s
Demodulation:	differential coherent four-phase
Operation:	full duplex
Link:	4-wire
Carrier Frequency:	1800 Hz
Carrier Envelope Frequency:	600 Hz
Timing Frequency:	2400 Hz (one cycle/bit)

5.1.2. CCITT System No. 6

System No. 6 is the international common channel signaling system recommended by the CCITT. It was initially standardized in 1968. After extensive field testing, it was recommended for in-service application in 1972 (Bernard, 1974).

The basic No. 6 protocol is very similar to CCIS, although the present specifications for System No. 6 provide for operation over digital data channels at bit rates of 4 kb/s and 56 kb/s, and at 2.4 kb/s over quasi-analog channels.

System No. 6 is oriented toward the associated mode of operation rather than the non-associated mode used by CCIS.

This is so, perhaps, because System No. 6 was designed to handle fewer and relatively shorter trunks than CCIS. These differences will become apparent in the following paragraphs. The operating procedures for signaling are the same as given for CCIS.

Signaling Codes and Format for System No. 6

The signal unit formats for System No. 6 are shown in Figure 5-6. A five-bit header is used. This reduces the trunk designator field to 11 bits. The five-bit header provides 32 distinct headings. Currently, only about 25 of these are used. The others are being retained as spares for regional and national needs.

When the first two bits of the header are zero, subsequent SSU's are indicated. The next two bits then give the number of SSU's in the multi-unit message.

A three-bit header 011 is used for ACU's. This leaves sufficient bits for unique identification of all blocks in the sequence, for the identity of a block being acknowledged, and for the eleven signal units per block.

The heading code 11101 is used for synchronizing signal units (SYU's), system control units (SCU's) and the multiblock synchronizing unit (MBS). The distinction between these units is obtained from the immediately following four-bit field (i.e., bits 6 to 7).

Table 5-4 summarizes the heading codes used in System No. 6. The various codes used in System No. 6 are given in the Appendix.

Signaling Block Structure

The block structure for System No. 6 is similar to the CCIS block structure, except that in No. 6, eight consecutive blocks can be synchronized using the MBS signal unit. Such eight consecutive blocks are called a multi-block. The MBS format provides a five-bit sequence number in bit positions 13 through 17 to index 32 multiblocks. Thus, the maximum number of blocks in the error control loop is 256 rather than the 8 blocks used by CCIS. See Figure 5-7. This multiblock structure permits speeds of operation up to 4 kb/s over both 1.544 Mb/s and 2.048 Mb/s primary multiplex channels, as well as at 56 kb/s. The maximum

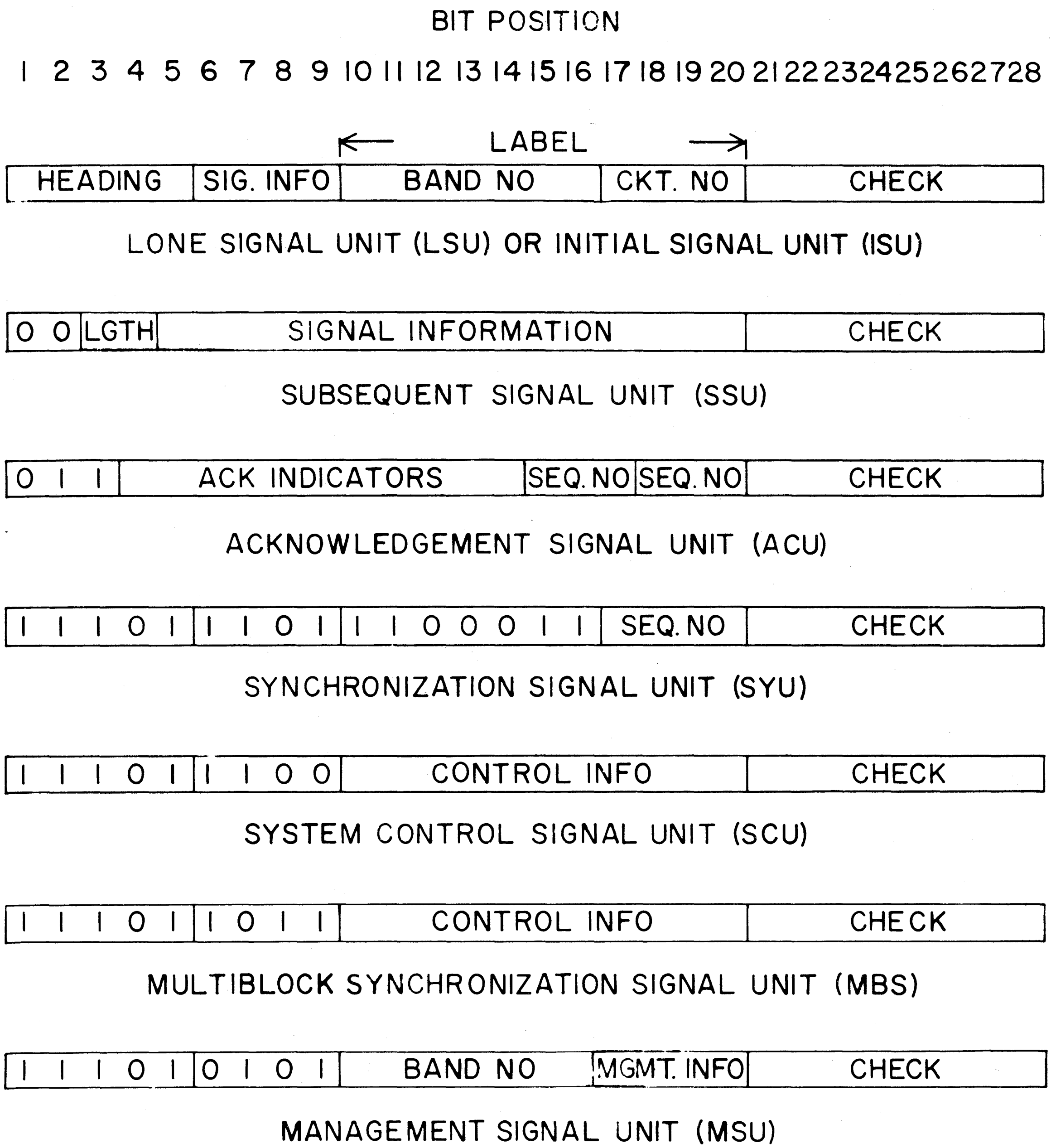


Figure 5-6. Basic format of signal units used with International Signaling System No. 6.

Table 5-4. Heading Codes Allocated for System No. 6

Code	Signal Unit
000	SSU
011	ACU
10000	ISU
10001 10010 10011 10100 10101 10110 10111	Subsequent address message (SAM)
11000 11001 11010 11011	International Telephone Signals
11101	SCU, MBS, and NMM

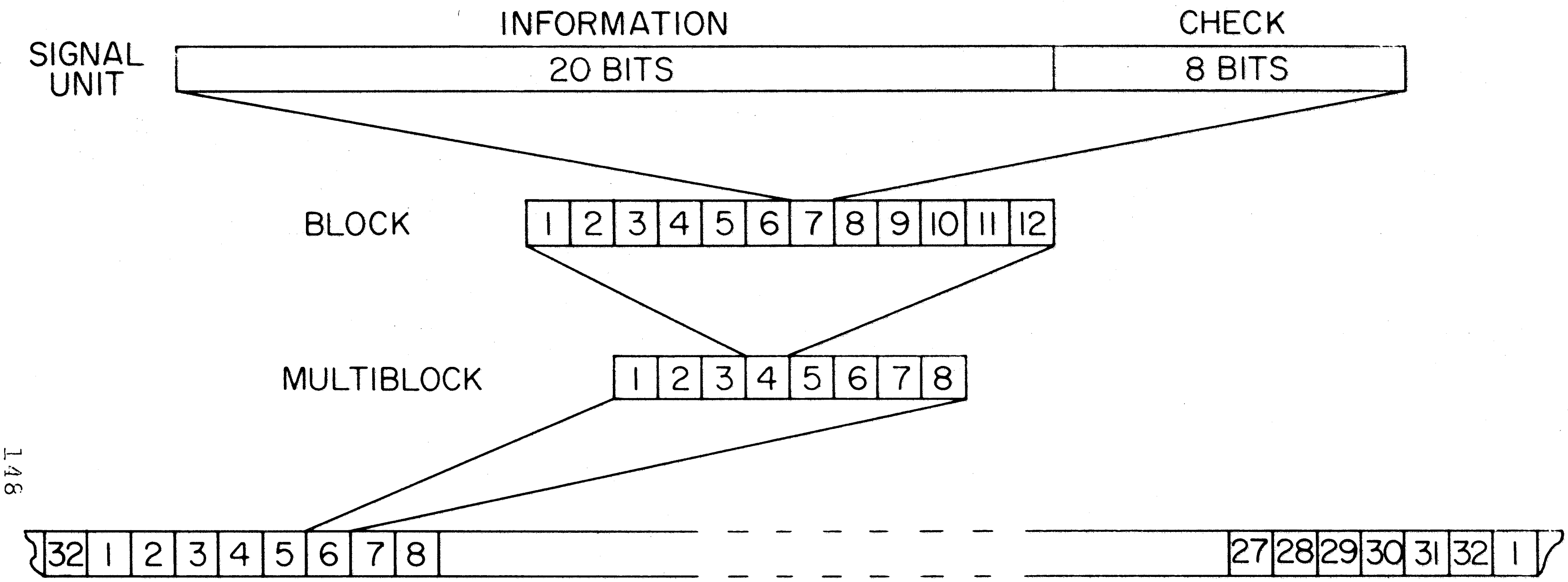


Figure 5-7. Block structure for International Signaling System No. 6.

permissable loop propagation time with and without the multiblock counter is given below. The MBS unit is required at high data rates over long paths.

Link Data Rate	Loop Delay in ms	MBS
2.4 kb/s	740	no
4 kb/s	448	no
56 kb/s	32	no
56 kb/s	1520	yes

Transmission

Transmission of System No. 6 signal data over quasi-analog channels is essentially identical to that for the CCIS system. In addition, System No. 6 can be used on digital signaling links with 2 kb/s, 4 kb/s and 56 kb/s data rates. The basic transmission schemes are shown in Figure 5-8. Over these digital facilities, the 2 or 4 kb/s signaling data are transferred serially to the interface equipment.

At the interface the bit stream is modulated onto the 64 kb/s voice channel, with 16 bits of the voice channel assigned to each bit of the 4 kb/s signaling channel. The entire 16 bits are inserted into the time slots allocated for signaling in each frame of the multiplex equipment.

Data transmission at 56 kb/s is also feasible over a 2.048 Mb/s primary multiplex channel. The 28 bits of the SU are placed in bit positions 1 through 7 of four 8-bit bytes and transferred serially at 64 kb/s. The four bytes are inserted into designated time slots of four successive frames.

5.1.3. CCITT System No. 7

Study Group XI of the CCITT is studying a new common channel signaling system for use on networks which utilize digital switching and digital transmission for integrated voice and data. The design emphasis is on simplicity and flexibility. The basic concept differs considerably from CCIS and System No. 6, although signaling units of fixed length are still envisioned. The System will be optimized for 4-wire operation over digital links at

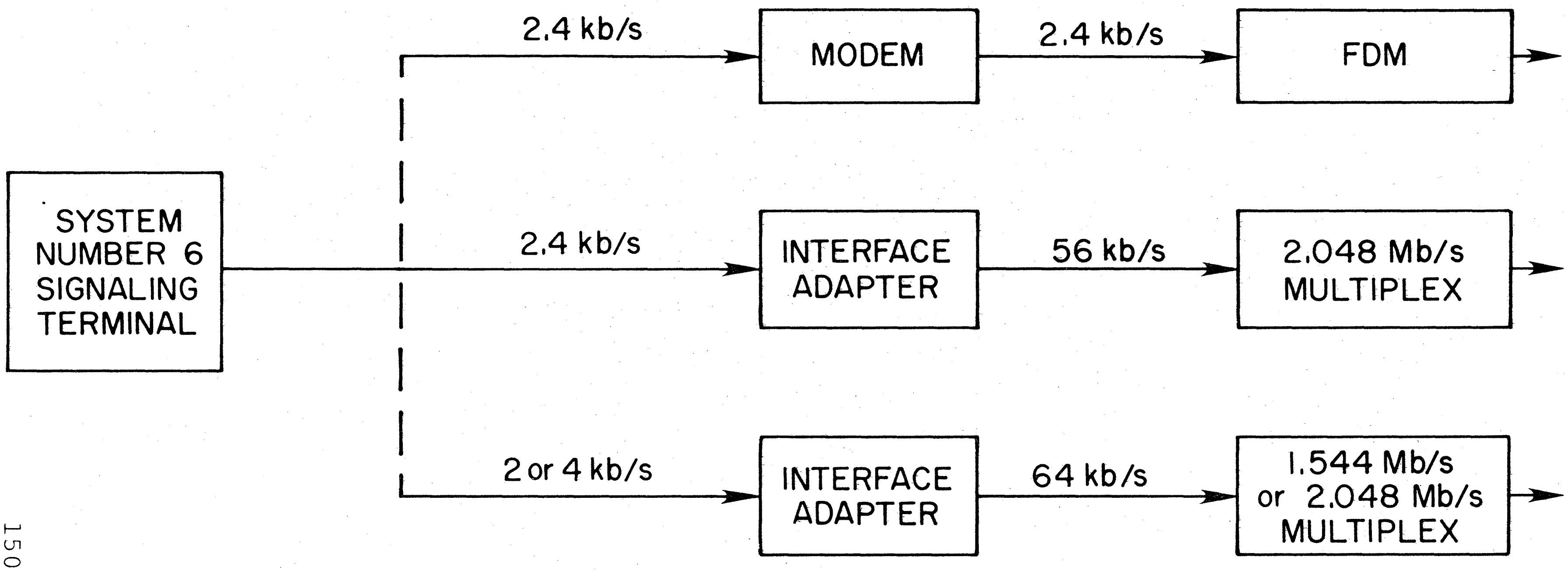


Figure 5-8. Transmission schemes defined for system No. 6.

64 kb/s. It will also be capable of working efficiently at lower bit rates and over analog links. Recommendations for a complete and detailed specification of this new common channel system, denoted Signaling System No. 7, are expected by 1980.

Only certain general aspects can be given at this time. The known features concern the message transfer part of the system. Study Group XI has indicated that the basic structure of Signaling System No. 7 will be divided into two clearly defined functional parts: a user part and a common message transfer part, as shown in Figure 5-9. The user part resides at both ends of a message link. It may utilize several different message processors, each making use of the message transfer part, e.g., call control, management, action, etc. The message transfer part includes the signaling data link and performs the link control functions, such as error control.

One signal unit format being considered is exemplified in Figure 5-9b. The signal unit length will be a multiple of 8 bits. In the example shown in the figure, the total unit length is 80 bits, half of which are user data. The other 40 bits are divided into link control fields, as indicated on the figure. The forward sequence number identifies the signal unit. A backward sequence number identifies some previously received signal unit which is being acknowledged either positively or negatively by the acknowledgement indicator. The signal unit indicator field identifies the type of signal unit being sent. Four types of signal units are provided to permit expansion of the user data field. They are 1) lone signal unit, 2) initial signal unit or multi-unit message, 3) subsequent signal unit, and 4) final signal unit. Multiple unit messages cannot be interrupted, because interruption may cause excessive queueing delays. The maximum number of signal units in a multi-unit message may have to be limited to approximately six.

The service indicator field provides 16 code combinations to direct the user data to an appropriate user part, e.g., telephone call control, data call control, network management, maintenance management, signaling system control, etc.

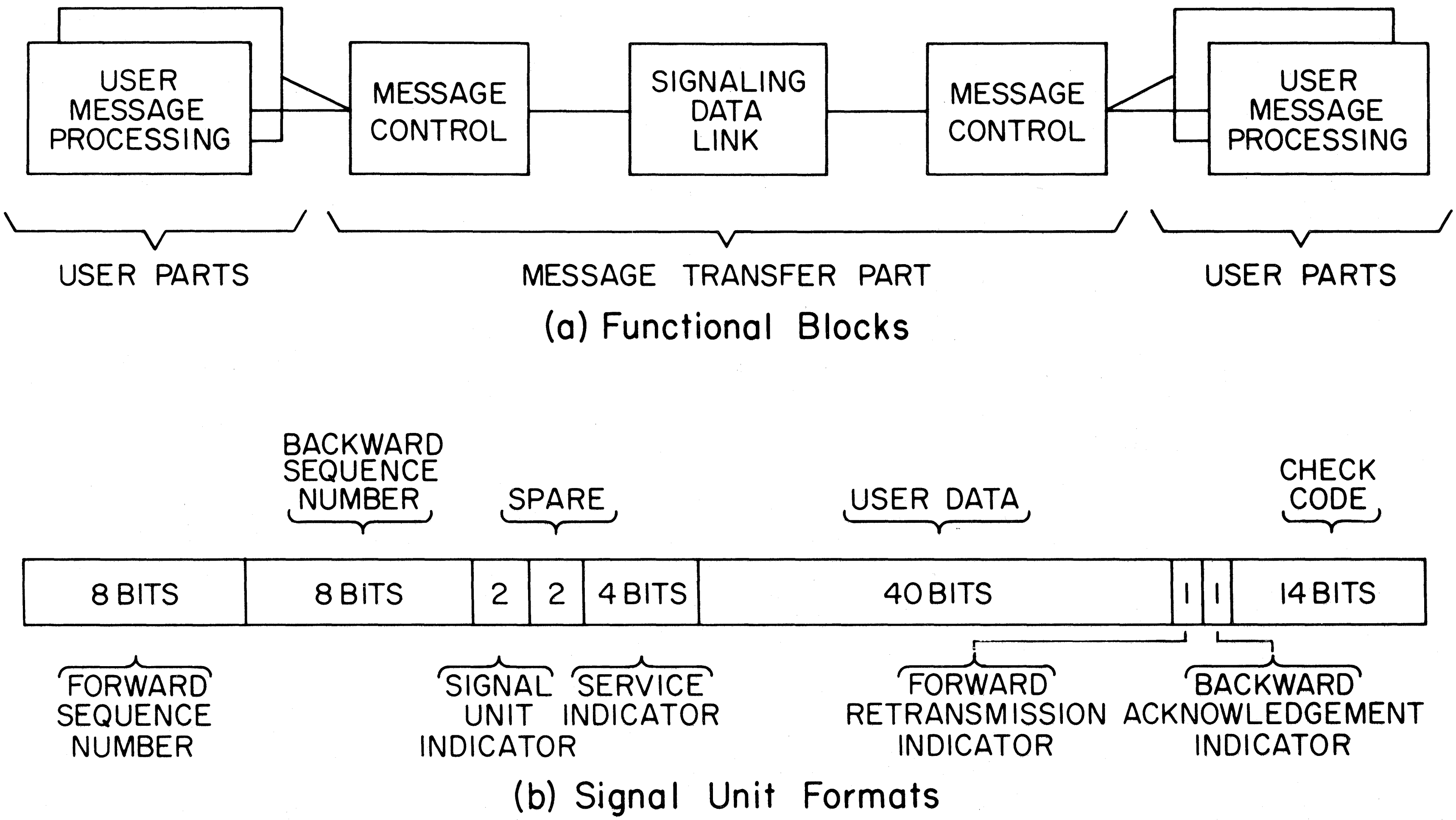


Figure 5-9. Function and format of proposed system No. 7.

The service indicator field identifies the user data for the different user parts for message processing. If the signal unit is being retransmitted because of a previous negative acknowledgement, this is indicated by the bit state in the retransmission indicator field.

Error detection is provided by including check bits in each signal unit. A 14-bit check sequence is shown in Figure 5-9b. An alternative format uses 16 check bits and the retransmission and acknowledgement indicator bits are borrowed from the forward and backward sequence numbers. Error correction is normally performed by retransmission. However, forward error correction may be provided as an option. Alternative methods for reduction of retransmission requirements are being considered. This involves repetative retransmissions during the idle periods of either the last signal unit sent, or of all unacknowledged signal units stored in the transmitter buffer.

The user data field for a lone signal unit will probably contain a four-bit header, to identify groups of signals, and a four-bit information field to identify the type of signal within each group. The remaining 32 bits are then available for labeling, addressing, and other information, depending on the signal unit type.

There are, as mentioned above, many detailed specifications and aspects of the system which require further study. Unresolved issues include the exact form of the polynomial to be used in generation of check bits, synchronization procedures, arrangements for insuring signal service availability and continuity, encoding of telephony signals in the user data field, performance characteristics of the signaling data link, and service requirements under normal and fault conditions.

The ultimate objective of System No. 7 is to provide a high-speed signaling system which is more suited for the digital environment. It should be capable of serving both dedicated and common user networks. The System No. 7 capabilities would include signaling over point-to-point terrestrial and satellite links including two satellite links in tandem. The latter requirements may be included to maintain essential voice communications. Point-to-multipoint operation may also be feasible.

5.2. Data Network Protocols

In Section 5.1 three common-channel signaling systems were described, which are now either used or proposed for telephone networks. The CCIS and CCITT No. 6 systems have been implemented for use nationally and internationally on circuit-switched voice networks. System No. 7 is being developed for international and national use to control circuit-switched, synchronous data networks. One of its main goals is to shorten the access time for data calls. The system may also be used to control voice networks and for maintenance and administration purposes. In this section the protocols for data networks are considered, including protocols from Level 1 through Level 5 as defined in Section 4.2.3.

Level 1 standards define the interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE). These standards or combination of level 1 standards provide the physical, electrical and functional interchange to establish, maintain, and disconnect the physical link between the DTE and DCE or between two DTE's. Thus they provide a common means of connecting a computer directly to peripheral equipment or remotely via communication equipment. This connection may be made using a serial or parallel interface. The serial interface transmits the data one bit at a time either synchronously or non-synchronously. It is commonly used when long communication links are required between terminals. The parallel interface transmits data in groups of bits (usually 8-bit bytes) and is used over short links because it is more efficient.

Figure 5-10 indicates the various standards which have been recommended by standards organizations for use in serial or parallel interfaces. The breakdown includes parallel and serial interfaces. Some serial interface standards define functional

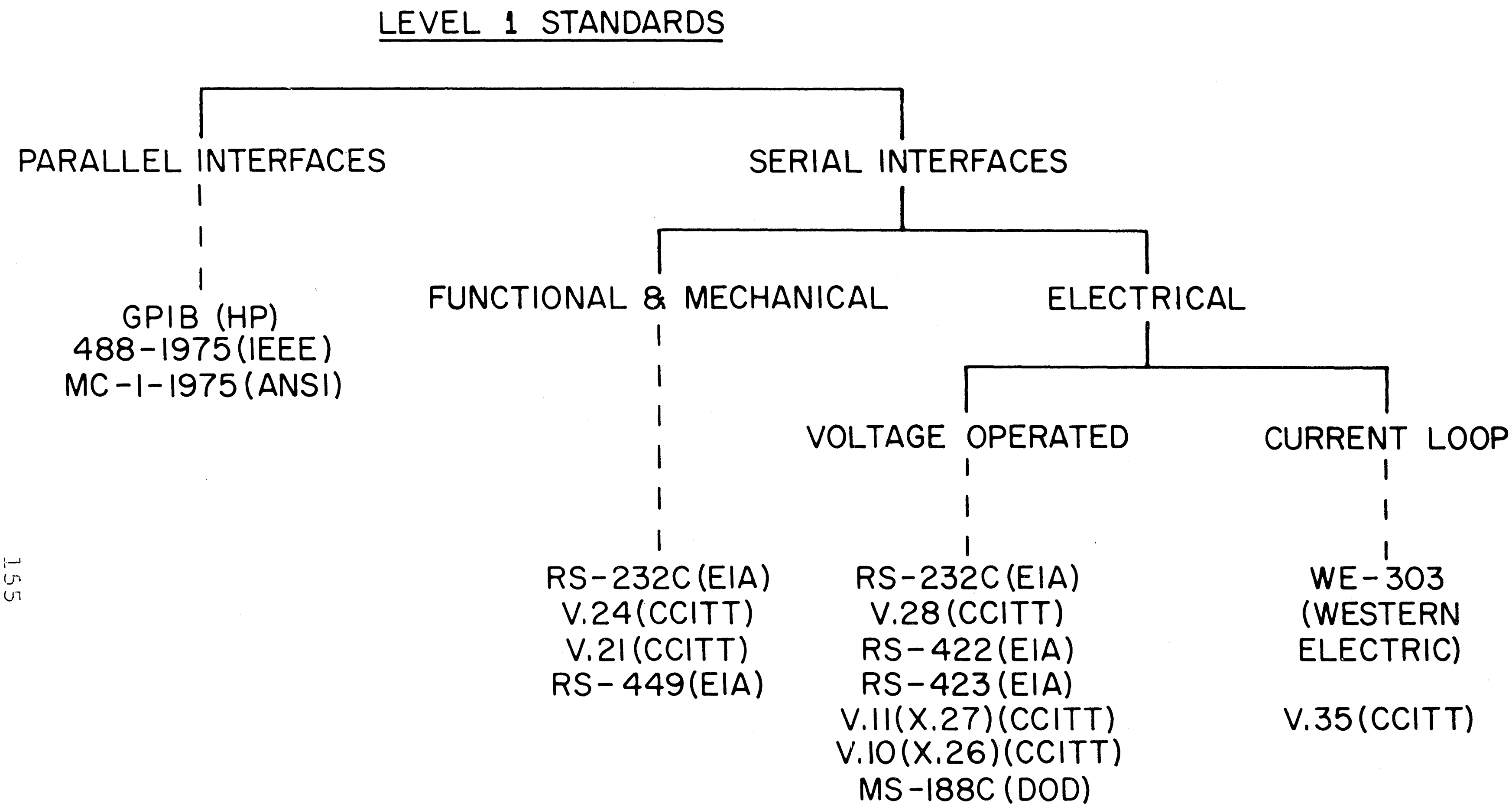


Figure 5-10. Interface standards promulgated by various organizations.

and physical characteristics, others define only the electrical characteristics while still others combine all characteristics.

5.2.1. Parallel Interfaces for Level 1

The General Purpose Interface Bus (GPIB) was originally developed by a commercial industry for interfacing programmable electronic test equipment. It has been adopted by both the IEEE and ANSI and is widely used for transmitting data bits in parallel between test equipment and computer peripherals. The parallel bus utilizes a 24-wire cable to carry eight parallel data lines and 8 parallel control lines. The data are transferred in 8-bit bytes at rates up to 1 Mb/s for short (1 meter) cable lengths. Up to 15 devices separated by 15 meters can share the bus at lower rates.

5.2.2. Serial Interfaces for Level 1

The serial interfaces indicated in Figure 5-10 are divided into voltage operated and current loop categories. The current loop interface WE 303 is used by AT&T to interconnect high speed terminals to wideband modems operating over private lines. Various versions of the interface provide data rates from 19.2 kb/s to 230 kb/s over short cable lengths. A binary "1" is signified by a current less than 5 ma and a binary "0" by currents between 5 mA and 23 mA for a 100-ohms circuit.

The CCITT V.35 standard is a high-speed interface similar to WE 303.

Voltage-operated standards typically operate at lower speeds. The international standards recommended by CCITT are in many instances similar to the recommended standards (RS series) promulgated by the EIA. Table 5-5 compares the pin numbers and functions of the RS-232C and CCITT V.24 standards.

The RS-232C interface provides an unbalanced connection between the DTE and the DCE. Two voltages of opposite polarity in the 5 to 25 volt range are required. Cable distances are denoted to be about 15 meters and data rates to 20 kb/s. This standard is gradually being replaced with RS-422 for balanced connections and RS-423 for unbalanced connections. These new standards alter the electrical characteristics of the interface

Table 5-5. Standard Interface for Serial Binary Data Interchange

Pin #	RS-232C (EIA) Designation	CCITT V.24 Equivalent	Description
1	AA	101	Protective ground
2	BA	103	Transmit data
3	BD	104	Receive data
4	CA	105	Request to send
5	CB	106	Clear to send
6	CC	107	Data set ready
7	AB	102	Signal ground common return
8	CF	109	Received line signal detector
9			
10			
11			
12	SCF	122	Secondary receive line detector
13	SGB	121	Secondary clear to send
14	SBA	118	Secondary transmitted data
15	DB	114	Transmit signal element timing (DCE)
16	SBB	119	Secondary receive data
17	DD	115	Receive signal element timing (DCE)
18			
19	SCA	120	Secondary request to send
20	CD	108	Data terminal ready
21	CG	110	Signal quality detector
22	CE	125	Ring indicator
23	CH/CI	111/112	Data signal rate selector
24	DA	113	Transmit signal element timing (DTE)
25			

signals and specify voltage levels which are more compatible with digital logic levels currently used in TTL circuitry. The span of RS-422 is 1200 meters at 100 kb/s and up to 12 meters at 10 Mb/s. Using RS-423 for unbalanced connections the span is either 60 meters at 10 kb/s or 12 meters at 100 kb/s.

RS-449 specifies the mechanical pin assignments and connector types used with both RS-422 and RS-423.

5.2.3. Character Oriented Protocols for Level 2

Level 2 protocols can have an important impact on overall network costs. This is because the transmission facilities and the software to control these facilities are a major cost factor. Therefore, the link control procedure must be simple to implement and should provide efficient and reliable communications.

There are two basic types of level 2 protocols. The common type is a character-oriented protocol such as the Binary Synchronous Control (BSC) protocol used by IBM or the data link control (DLC) recommended by ANSI. These character-oriented protocols use a code set whereby certain codewords are allocated for control purposes and other code words for user information. A header field specifies the control parameters and an information field contains the control characters. The character codes used in ASCII and EBCDIC are indicated in Table 5-6. The 7-bit ASCII code is the American National Standard Institute version of the ISO code R646. The same code is also recommended by the CCITT as International Telegraph Alphabet No. 5 (ITA-5). The 8-bit EBCDIC code is used in IBM 360 computers.

A 7-bit code (e.g., the R646 ISO code) gives 2^7=128 code word combinations. These provide 95 printing symbols, a space, and 32 control characters. Bits 6 and 7 are both zero when indicating a control character as shown in Table 5-6. The 8-bit EBCDIC code gives 256 combinations. About one quarter of these are control characters.

Other codes use a shift character to allow more symbols to be represented than is possible with a one-to-one correspondence between symbols and code patterns.

Table 5-6. Character Oriented Codes

Symbol	Meaning	Binary Code R646 (ISO) ITA-5 (CCITT) ASCII (ANSI)	Binary Code EBCDIC (IBM)
NUL	Null	000 0000	0000 0000
SOH	Start of heading	000 0001	0000 0001
STX	Start of text	000 0010	0000 0010
ETX	End of text	000 0011	0000 0011
EOT	End of transmission	000 0100	0011 0111
ENQ	Enquiry	000 0101	0010 1101
ACK	Acknowledgement	000 0110	0010 1110
BEL	Bell alarm	000 0111	0010 1111
BS	Backspace	000 1000	0001 0110
HT	Horizontal tabulation	000 1001	0000 0101
LF	Line feed	000 1010	0010 0101
VT	Vertical tabulation	000 1011	0000 1011
FF	Form feed	000 1100	0000 1100
CR	Carriage return	000 1101	0000 1101
SO	Shift out	000 1110	0000 1110
SI	Shift in	000 1111	0000 1111
DLE	Data link escape	001 0000	0001 0000
NAK	Negative acknowledgement	001 0101	0011 1101
SYN	Synchronization	001 0110	0011 0010
ETB	End of transmission mode	001 0111	0010 0110
ESC	Escape	001 1011	0010 0111

In general, if M combinations of an n-unit code are used as shift commands, the total number of commands will be $M(2^n-M)$ which is maximum when $M=2^{n-1}$. This gives $2^{2(n-1)}$ possible symbols compared with 2^n symbols without shifting (Davies and Barber, 1976).

5.2.4. Bit-Oriented Protocols for Level 2

Bit-oriented protocols are procedures used for arranging messages in frames. A frame is a sequence of control and information bits which is delimited by a flag at the beginning and end of a sequence. The flag is a unique code word. Transparency is achieved by insuring that none of the bit sequences between the flags corresponds to a flag during transmission.

The American National Standards Institute (ANSI) has adopted a bit-oriented level 2 protocol called the Advanced Data Communications Control Procedure (ADCCP).

The International Standards Organization (ISO) protocol is called high-level data link control (HDLC) and the International Business Machines (IBM) protocol is denoted synchronous data link control (SDLC). All are bit oriented, data link control protocols (i.e., level 2) and all are similar procedures. The HDLC protocol is described in more detail in the next section since it is used to support the level 3 protocol, X.25, used on packet-switch networks.

The AUTODIN II network will use a protocol similar to HDLC and ADCCP in one of its operating modes (Sevcik, 1977).

5.2.5. Packet-Switched Protocols for Level 3

The CCITT, supported by ISO and ANSI, recently adopted the X.25 protocol for interfacing with public packet-switched networks (CCITT, 1977b). Level 1 of X.25 specifies that the physical, electrical, and functional characteristics shall be in accordance with level 1 of X.21. The CCITT X.21 and the U.S. version ANSI X.21 is a simple interface implementing a minimum number of interchange circuits to connect DTE's to any telecommunication network. The standard provides full transparency (bit sequence independence) for data transfer.

Level 2 of X.25 employs the HDLC protocol specified by ISO for control between two nodes on the packet-switched network.

Level 3 of X.25 defines the formatting of packets and control procedures for end-to-end communications over several links in tandem. Each of these X.25 levels is described in more detail below.

<u>Level 1</u>

Figure 5-11 illustrates the X.21 functional interface and circuit designations. Call establishments and terminations are provided by the control circuit C. This control circuit also provides transparency by separating the data and control functions. The indicator circuit I is used to signify when the establishment phase is complete and data transfer can begin. Since the interface is transparent, the user can employ any higher level protocol. The transmit circuit T and receive circuit R are used for transferring information, containing both user data and logic level control information.

The X.21 interface uses a 15-pin connector between the DTE and the DCE. The electrical interface characteristics correspond to Recommendation X.27 or RS-422 for the DCE. Either X.26 or X.27 may be used for the DTE.

The X.21 protocol uses the character oriented International Alphabet No. 5 (IA5). It is similar to ASCII for the higher levels, such as the establishment phase of a call (Folts and Cotton, 1977). Basically, X.21 is a bit-oriented protocol for the data transfer phase. The user is free to use his own format protocol and his own frame synchronization for data transfer (see Folts, 1977). Only the physical-electrical level 1 characteristics of X.21 are applied to packet switching. Packet switched networks use X.25 at levels 2 and 3.

<u>Level 2</u>

The HDLC protocol is specified for X.25 to provide error and flow controls over the access link between the DTE and the network. The HDLC format is shown in Figure 5-12. The number across the top of the figure indicates the order of bit transmission.

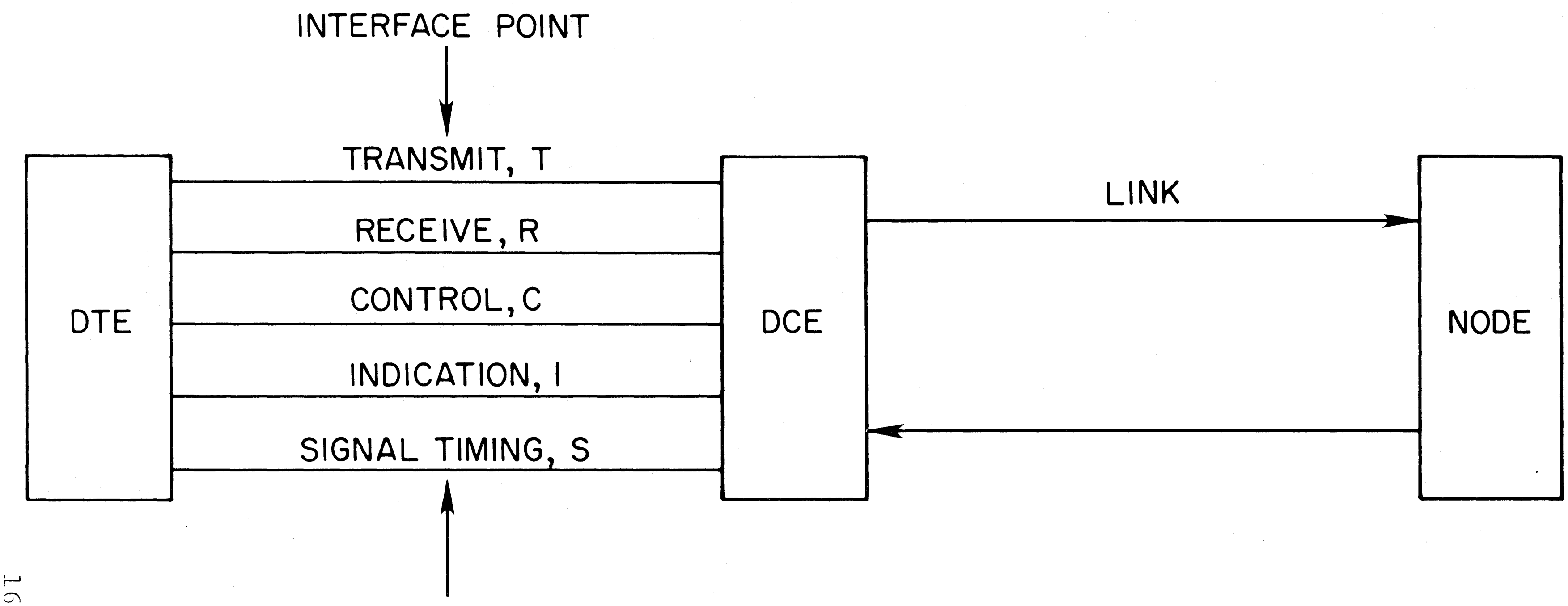


Figure 5-11. Level 1 interface (X.21) for packet switched networks.

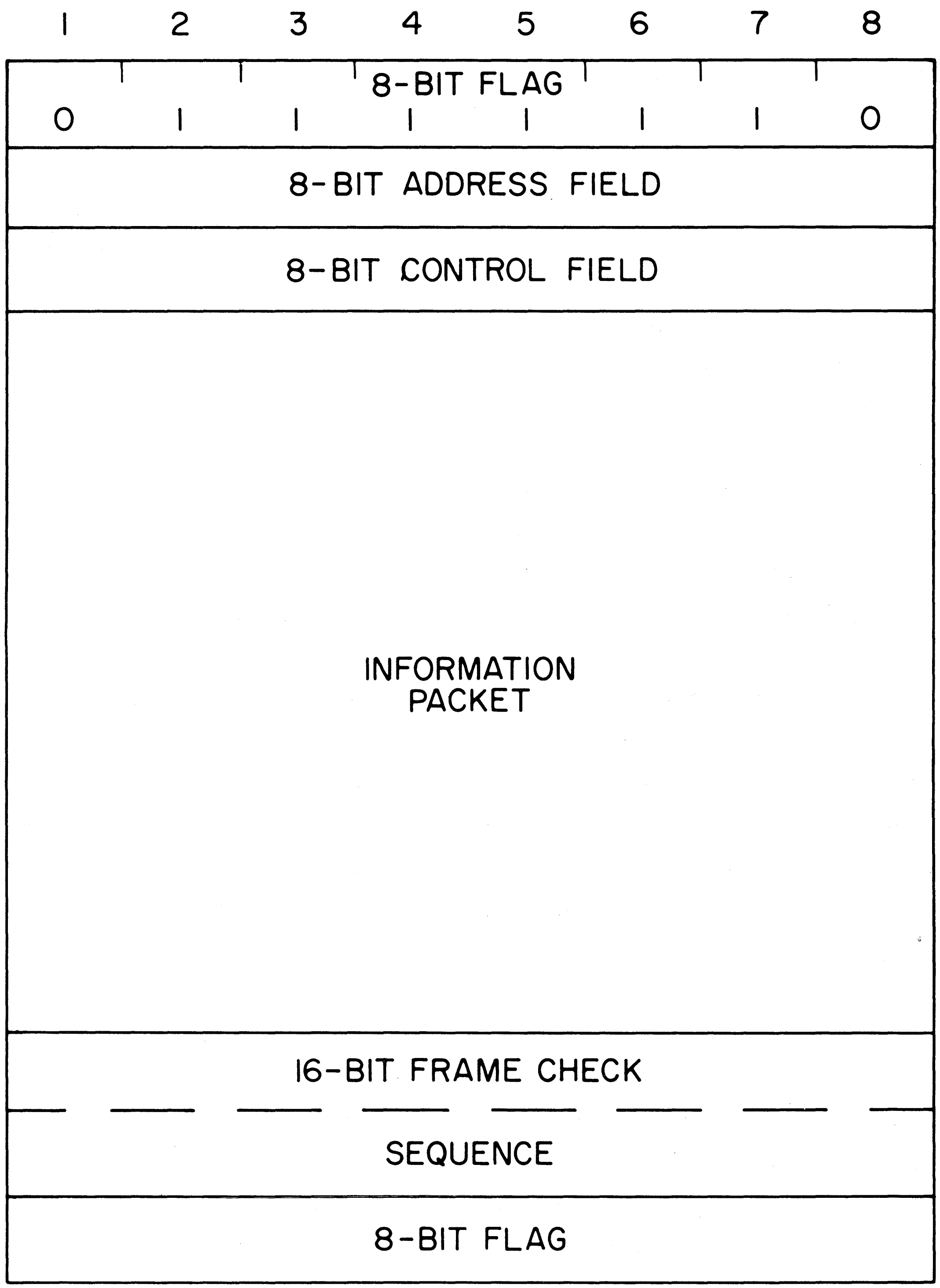


Figure 5-12. Frame format for level 2 (HDLC) of X.25 protocol.

The check sequence contains 16 bits, derived from the generator polynomial $X^{16}+X^{12}+X^5+1$. Transparency is achieved by using 0-bit insertion when six or more 1's appear in sequence, except for the flag itself.

All HDLC frames start and end with an eight-bit (octet) flag consisting of the bit sequence 01111110. A single flag may be used to signify both the start and the end of a frame.

The eight-bit address field indicates whether the direction of commands and responses is from DCE to DTE, or from DTE to DCE.

The control field contains a command or a response, and sequence numbers where applicable. Three types of formats are used to distinguish between information transfer, supervisory functions and control functions.

The information field contains a packet of information whose octet length and makeup is described under level 3.

Level 3

The X.25 interface recommends a maximum data field length of 128 octets, where an octet is one 8-bit byte. The network may disassemble these 128 octets into smaller packets for internal switching. The recommendation also allows for other data fields, but they must be of fixed length. Normally, if more user data are required, a full data packet is marked by the DTE with a "to-be-continued tag". This tag indicates that more data follow in the subsequent packet. Packets generated by a source DTE are transmitted in an ordered sequence to the destination.

Different types of packets are used from DCE to DTE and from DTE to DCE. Both types include packets for call setup, clearing, data, interrupts, flow control, reset, and restart.

Figure 5-13 illustrates the call request packet format. This format is used during the call establishment phase to set up a virtual circuit between origination and destination terminals.

The 4-bit format identifier indicates the general format of the rest of the header and the sequence numbering scheme employed.

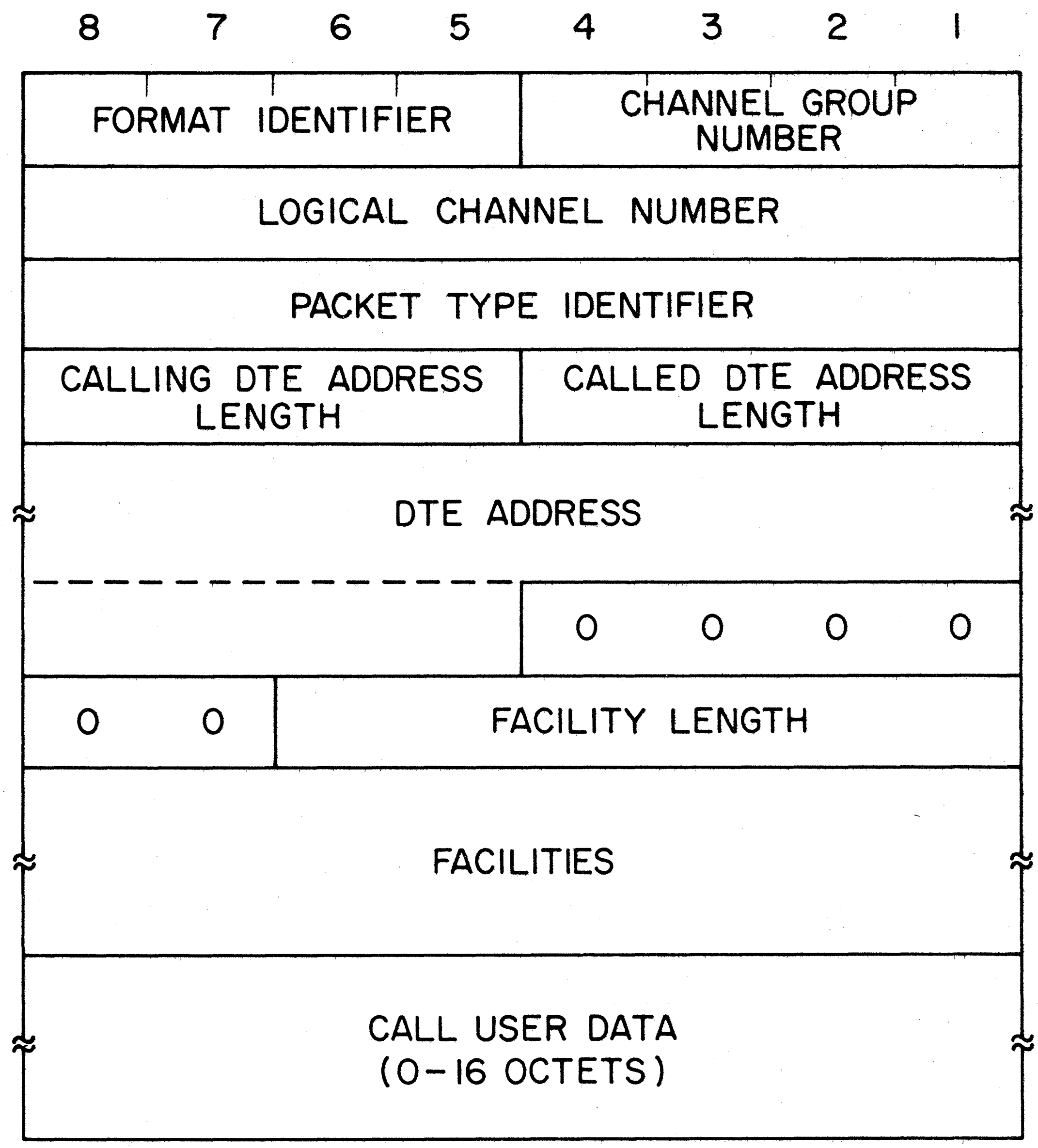


Figure 5-13. Call request packet used with the X.25 protocol.

The logical channel group number and the actual channel number identify the virtual circuit. These two numbers appear in subsequent packets, except for restart packets.

The packet identifier in octet 3 distinguishes the particular packet type being transmitted, i.e., request, data, interrupt, clear, etc.

There are address fields that occupy one half of an octet. These semi-octets indicate the length of the address in semi-octets, for the called DTE and the calling DTE. The actual addresses are contained in octet 5 and subsequent octets.

The fields provide a means for communicating with optional user facilities or to realize other special features via the DTE. The field is not required in applications where peripheral equipment is not used.

The user data are sent in the octets following the field. In the call-request packet the maximum length of this data field is 16 octets.

If more user data are to be transmitted, then additional packets are formed using the data packet format of Figure 5-14. Since a virtual circuit has already been established with the call-request packet, these additional data are identified only by their logical or packet number. The third octet contains transmit and receive sequence numbers and one indicator bit, to show when more data follow.

The X.25 protocol provides the interface to a packet-switched data network to provide virtual service to the user. Additional studies are being conducted by CCITT and ANSI to extend the interface to provide datagram service, where all the appropriate control and address information is contained in each packet. Thus, in contrast to virtual service, there are no call establishment or disengagement packets involved in datagram service.

The recommended X.25 protocol for packet-switched networks is a good example of how the various protocol levels interact. When communicating digital information, levels may appear

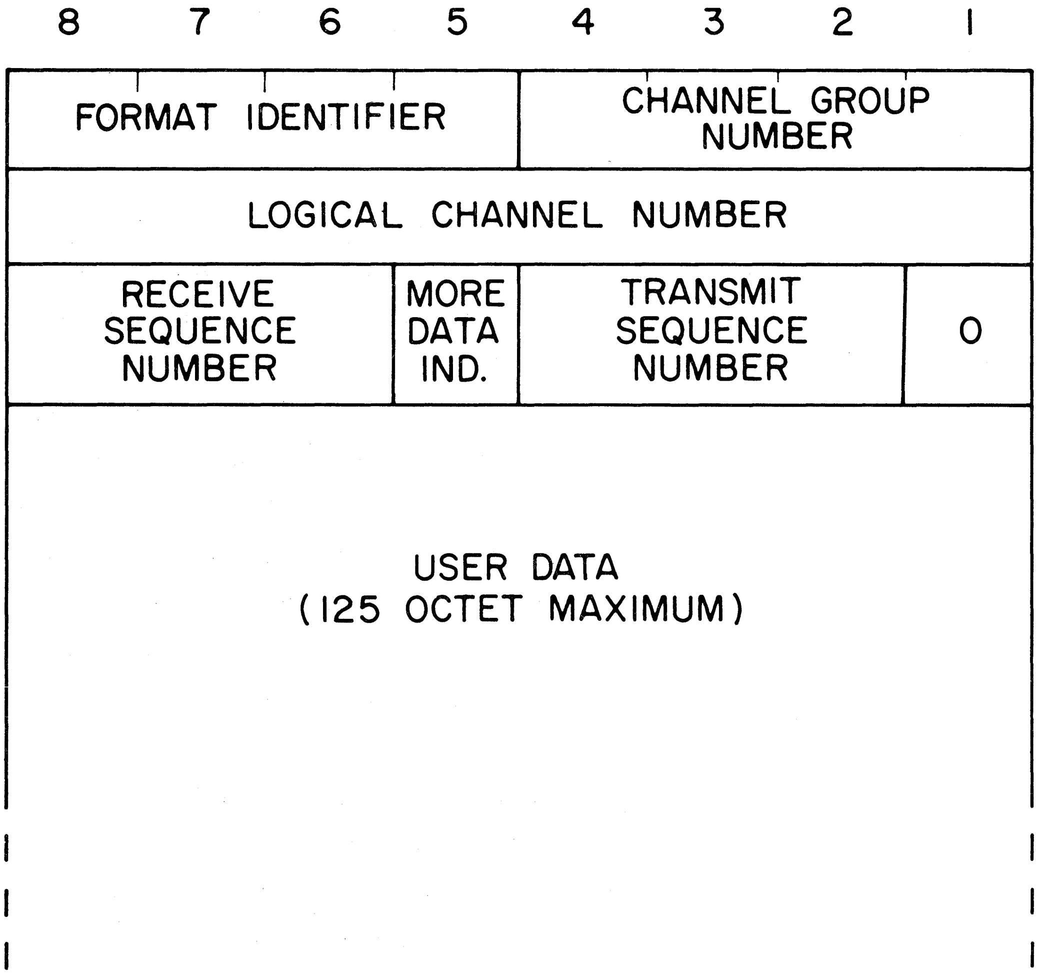


Figure 5-14. Data packet format used with the X.25 protocol.

independent of each other. At the same time, levels can be formatted together by nesting one protocol within another. Failure of a higher level protocol does not affect the lower level. Also, changing the format or some other characteristics of a high-level protocol does not change the lower level protocols.

This nesting concept leads to combination protocols, whereby different higher level protocols are used with common lower level protocols. Such nesting extends the functions and services realizable by a variety of terminals.

The concept of a general-purpose protocol to integrate different networks has been used before. One example is reported by Nichols (1976). The information field portion of an HDLC type frame, as shown in Figure 5-12, is used in the example to transfer control information for terminal classes ranging from low-speed, five-level-code, teletypewriter terminals, to medium-speed, eight-level-code, intelligent terminals, and to high-speed computer terminals.

The SENET concept described by Coviello and Vena (1975) and GTE (1975) uses a high-level control protocol imbedded in the information field of the ADCCP protocol. It dynamically controls the virtual circuit-switched and packet-switched channels. The protocol includes a CCIS type signaling scheme for the circuit-switched and packet-switched channels. The SENET/DAX format, which is compatible with ADCCP protocols for link control is illustrated in Figure 5-15.

System No. 7, itself, appears to be evolving as a combined protocol due to the separation of functions between the user part and the message transfer part. The user part does not have to apply just to subsystems of the processor at a given node. The user part could contain data for any other remote user terminal.

Several U.S. manufacturers are offering inexpensive data-link control chips which support more than one protocol. Protocol conversions are handled by software and can be changed

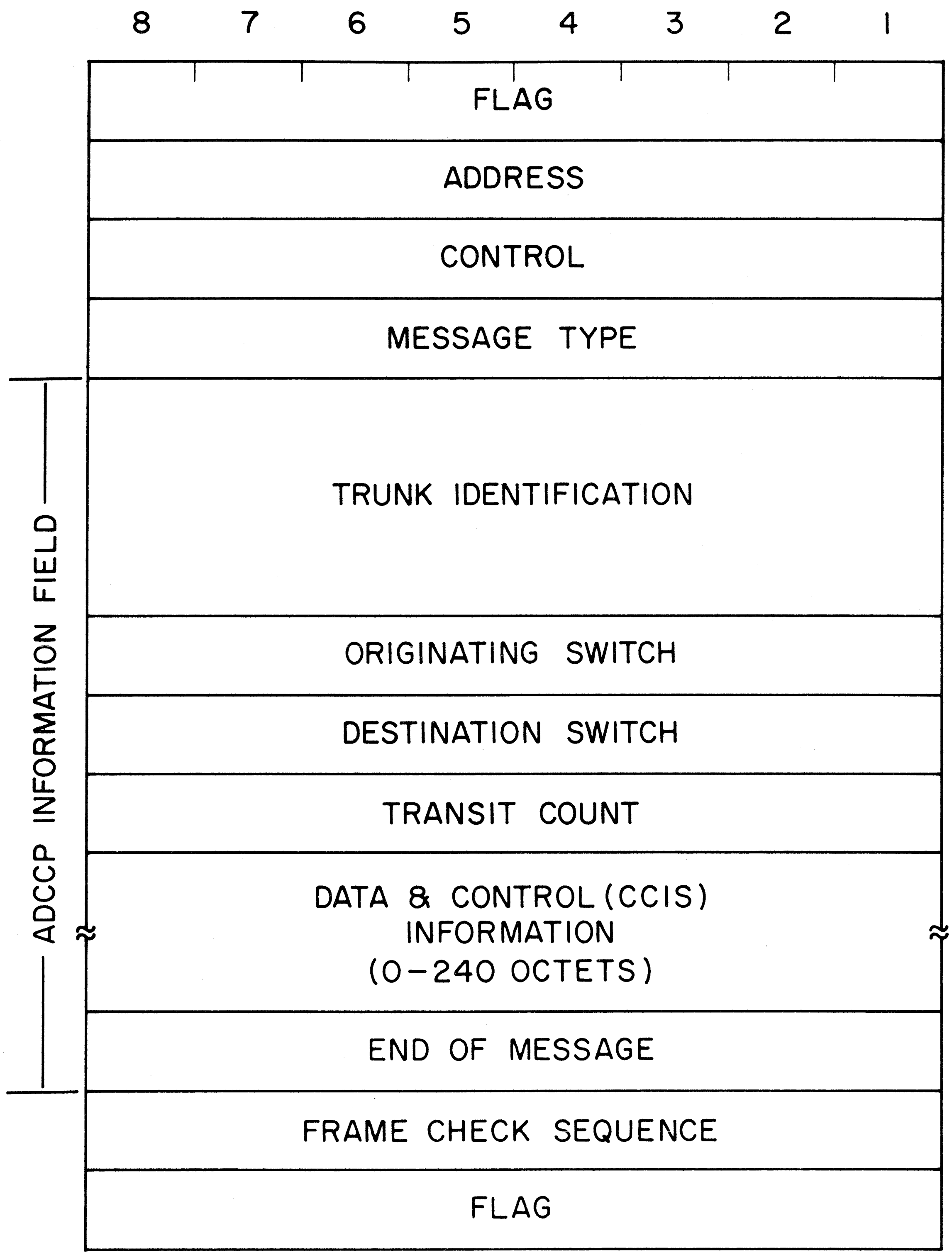


Figure 5-15. SENET/DAX packet format.

using the same hardware. One multi-protocol LSI chip can support character-oriented protocols from a terminal and bit-oriented protocols for node-to-node communications (Weissberger, 1978).

As network complexity and user diversity increases, protocol conversion may be handled with distributed microprocessors. Such microprocessors can be placed at the nodes, as part of the signaling terminal or in conjunction with the switching processor. A microprocessor implementation of the X.25 protocol is described by Shearin and King (1977) for levels 1 and 2, to provide full duplex operation of data links at rates up to 3 Mb/s.

5.2.6. Network Control Architectures for Level 4

Level four establishes the system control procedures. This could be processor subsystem control or peripheral-device control. Level four protocols coordinate the transfer of data between a users application program and the operating system.

Currently no standards have been recommended although several computer manufacturers are developing protocols for this level. Since these system control architectures are still under extensive development and test, a description is beyond the scope of this report. Information on some of these protocols is given by McFayden (1976), Wecker (1976), and Conant and Wecker (1976).

5.2.7. Performance Standards for Level 5

ANSI (1974) describes the American National Standard X3.44-1974 for the determination of the performance of data communication systems. This standard is applicable to synchronous and non-synchronous data communication links employing the ASCII code. Performance criteria are established from a user viewpoint and include system capacity, delay factors, accuracy and availability.

A similar standard has been proposed by the Federal Telecommunications Standards Committee (FTSC). Denoted FS-1033 this Federal Standard defines user-oriented performance parameters in terms of efficiency, accuracy and reliability for three communication functions, namely access, information transfer, and disengagement. These FS-1033 parameters with sample parameter values for different types of networks were discussed in Section 4.3.

The X3.44-1974 and FS-1033 standards are considered level 5 since they involve performance evaluation at interface between the terminal and the user.

Table 5-7 summarizes the known standards for interface levels 1 through 5. Definitions for the standard designations used and the promulgating agencies' acronyms are given in the list of acronyms at the beginning of this report.

5.2.8. AUTODIN II Access Protocols

AUTODIN II is a packet switched network being developed to meet the data transmission requirements of DoD users primarily in CONUS. Network applications include interactive, query/response, bulk data transfer and narrative. The four basic protocols under network control (i.e., level 1 through level 4) are being implemented for AUTODIN II. Different modes of access at certain levels provide support to a variety of terminals and services. For example, at level 3 the network supports either datagram or virtual service. The user access protocols specified for AUTODIN II are summarized below*.

Level 1

Defined by MIL STD 188C or EIA RS 232C.

Convertible to MIL STD 188-114 and EIA RS 449.

Level 2

Mode I: Full duplex, character oriented, binary synchronous (currently used by message terminals to access AUTODIN I).

Mode IB: Full duplex, character oriented, binary synchronous (ANSI X3.28 and BSC (IBM)).

Mode IIA: Full duplex, character oriented, asynchronous.

Mode IIAH: Half duplex, character oriented, asynchronous.

*Because the system is under design and development some changes are expected. The information given was obtained from DCA "System Performance Specification (Type A) for AUTDON II Phase I dated February, 1977.

Table 5-7. Recommended Standards for Data Networks

	International		United States	
Level	Current	New	Current	New
1	V.24/V.28 (CCITT)	X.21 (CCITT) V.10 (X.26) (CCITT) V.11 (X.27) (CCITT)	RS-232C (EIA)	RS-449 (EIA) RS-423 (EIA) RS-422 (EIA) BSR-X3.69 (ANSI)
2	1745 (ISO)	HDLC (ISO)	DLC (ANSI) BSC (IBM) X3.28 (ANSI)	ADCCP (ANSI) BDLC (Burrough) SDLC (IBM) DDCMP (DEC)
3		X.25 (CCITT)	X3.281 (ANSI)	X.25 (CCITT)
4				SNA (IBM)* DNA (NCR)* DECNET (DEC)* DCA (UNIVAC)*
5				FS 1033 (FTSC) (Proposed) X3.44-1974 (ANSI)

Blanks indicate no applicable known standard.

*Examples under development by industry (not necessarily complete).

Mode VI: Full duplex, bit oriented, binary synchronous (compliant with ANSI advanced Data Communication Control Procedure, ADCCP).

Level 3

3A: Datagram Service

3B: Virtual Connection Service (analogous to CCITT X.25)

Level 4

Terminal handler protocol provides interprocess control via operator key commands to network virtual terminal.

5.3. Satellite and Packet Radio Protocols

A variety of protocols have been developed for use on radio and satellite channels. These are particularly useful when a multiplicities of data sources transmits randomly to a central destination (e.g., a number of access terminals using the same computer). Usually such traffic comes in bursts and has a very low duty cycle. Table 5-8 gives some examples of the peak and average data rates for different types of user terminals.

When the peak to average ratio is high, as shown in the table, then it is possible to connect a large number of geographically distributed users to a central computer.

5.3.1. Pure Aloha

A number of protocols have been developed to provide random access using the same radio frequency channel for packet radio,

Table 5-8. Traffic Characteristics of Typical User of Data Communications Satellite

User	Average Data Rate (b/s)	Peak Data Rate (b/s)
Input Terminal (e.g., TTY)	1	10^2
Output Terminal (e.g., CRT)	10	10^4
Remote Job Entry (RJE)	10^2	10^4
Computer to Computer	10^4	10^6

and satellite communications. Most of these random access techniques have evolved from the pure Aloha system (see Abramson and Kuo, 1973). This pure Aloha system uses a protocol whereby all terminals transmit randomly. Because of the bursts of traffic and because each user transmits at will, there is a non-zero probability that two or more sources may transmit simultaneously and the message (or packet) could not be delivered. Actually this probability is quite low and even with a large number of users the channel utilization (i.e., the maximum throughput) is 1/2e* or 18% of the channel rate. Thus, for a typical channel rate of 50 kb/s, a large number of users (~250) can transmit peak bursts of data at 50 kb/s with an average rate of $0.18 \times 50 \times 10^3 = 9$ kb/s.

5.3.2. Slotted Aloha

The average rate can be increased to 1/e or 37% of the channel rate using a slotted Aloha protocol. With slotted Aloha, all users are synchronized to a common clock and transmit only in specified time slots. The probability of simultaneous transmissions occuring is reduced by a factor of two. This improvement however is obtained with the added complexity of synchronizing all user terminals (Kleinrock and Lam, 1973).

An additional increase in throughput can be obtained if the slotted Aloha is augmented with the additional feature of "capture". When two packets are transmitted simultaneously the packet with the higher signal level "captures" the receiver and is not lost. Therefore, the throughput is increased. The amount of this increase can be shown to be 57%. Again this increased throughput is obtained at the cost of the added complexity.

5.3.3. Reservation and Carrier Sense Protocols

There are still other schemes for increasing the throughput of an Aloha type of network. The reservation Aloha protocol

*e=2.718.

is based on the user's requesting a transmission slot in advance. The carrier-sense protocol permits the user to transmit a packet only when no carrier exists. These carrier sensing schemes should theoretically be capable of an average throughput rate which exceeds 80% of the channel rate. However, such benefits are achievable only when the users are not widely distributed geographically. The delay in sensing the carrier is only a small fraction of the slot interval.

5.3.4. Protocols for Mixed Traffic on Radio Networks

Typical channel traffic may not be comprised entirely of bursts, as it often includes data from continuous sources, such as voice and facsimile. When such a mixture occurs, one talks of integrated voice and data traffic. The slotted envelope network (SENET) concept described by Coviello and Vena (1975) appears useful for integrated radio networks. A special protocol which is related to reservation Aloha is required in this application. Such a protocol is being developed by ARPA and DCA.

This broadcast satellite channel transmission protocol currently is being implemented for experimental evaluation on INTELSAT-IV and is described by Jacobs et al. (1977). Called CPODA, for Contention-based Priority Oriented Demand Assignment, the protocol is designed to handle packetized voice and data traffic in a multiple access satellite broadcast channel. The channel can be shared by hundreds of earth stations while generating an arbitrary mix of bursty and non-bursty traffic. Jacobs and his colleagues define these traffic types as block traffic and stream traffic. For both types it is assumed that the messages have differing precedence levels and delivery delay requirements, differing load distributions among the earth stations, and either multiple or single destination addresses.

The CPODA protocol is based on the reservation system which has been shown to be more cost effective than TDMA, FDMA and simple ALOHA techniques (Roberts, 1973). The reservation system employed with CPODA utilizes a centralized controller to inform all other, presently idle, stations of their scheduled transmission times. Several CPODA protocol concepts are currently being evaluated using four earth stations in the Atlantic region. The data rate from these stations is either 64 kb/s or 16 kb/s for both transmitting and receiving. Three of the stations use satellite IMP's which are part of the ARPANET. Signals are broadcast from the INTELSAT-IV-A. The fourth station uses an unattended COMSAT earth terminal.

6. APPLICATIONS TO THE ACCESS AREA DIGITAL SWITCHING SYSTEM

The access area digital switching system (AADSS) may be viewed as an interconnecting gateway for local networks. The gateway provides intracommunications at the base and tactical level. It interconnects with global-strategic networks, which may be terrestrial (DCS), satellite (DSCS), or combinations of these networks. Finally, the AADSS must provide the interface between the military networks of various types and the public switched networks on a world-wide basis.

The interfacing structure is depicted by the diagram in Figure 6-1. The AADSS performs duplex protocol transformations by matching the transmission facilities of the various networks indicated in the figure. This AADSS role must be accomplished during the military transition from analog to part digital, and all-digital network configurations. Ultimately, it may be used to interface with - and even to be a part of - a fully integrated voice and data network.

A major dissimilarity between networks is reflected in the means by which they are controlled, i.e., in their protocols. Most protocols in use, or under present development, have not been standardized. This includes protocols for random access satellite links and packet radio links.

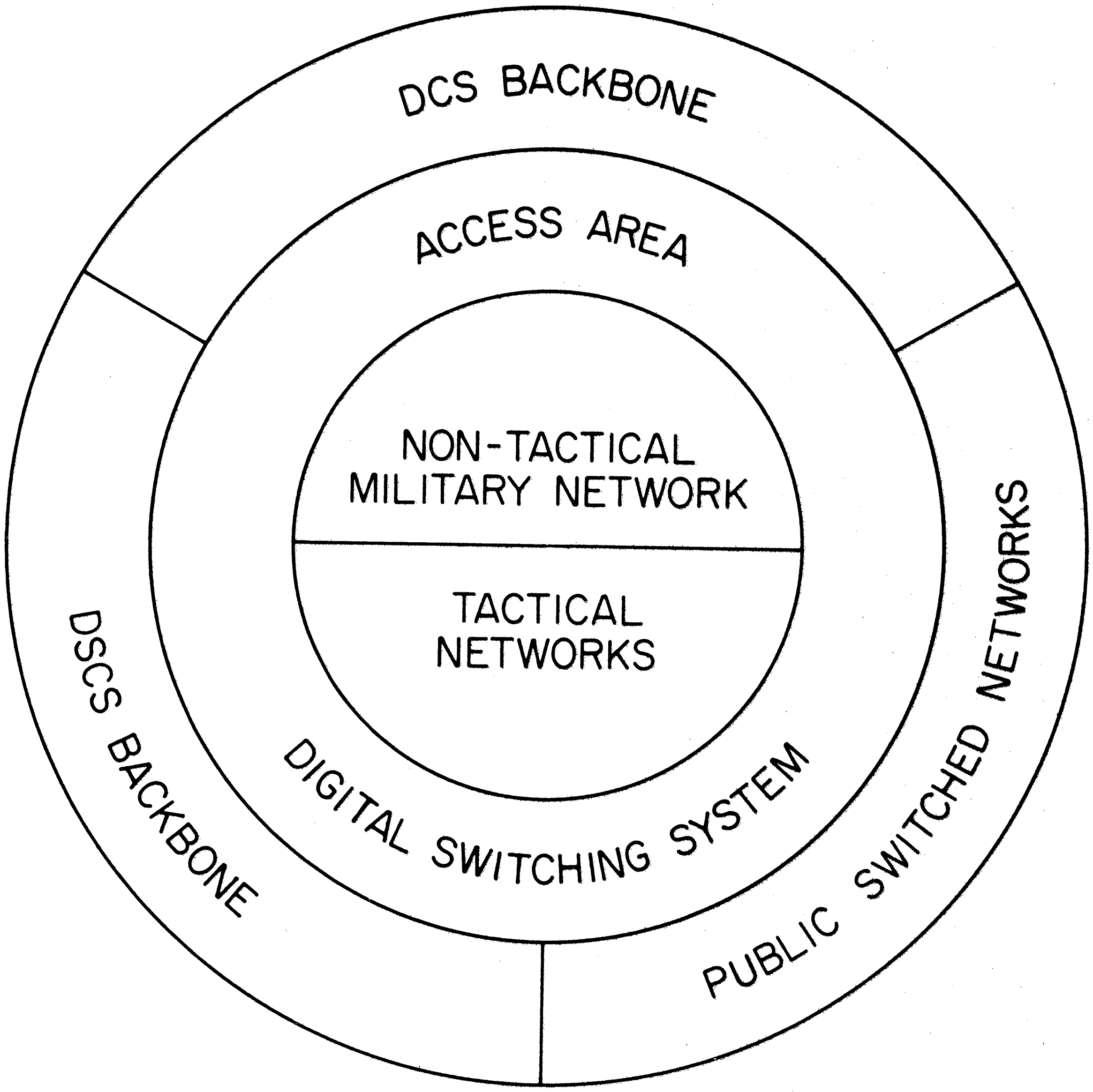


Figure 6-1. Access area digital switching system interfaces.

In the following subsections, we compare the various circuit switch and packet switch protocols that have been described for use on digital facilities. We also review the role of the AADSS for protocol transformations at various inter- and intra-network interface levels.

6.1. Signaling System Comparisons

The signal formats for common channel signaling using digital techniques are compared in Figure 6-2. The CCIS, as well as CCITT System Nos. 6 and No. 7, provide formats for common channel signaling over telephone networks. The ADCCP, SDLC, HDLC formats are used primarily for controlling data links.

Table 6-1 compares the various signaling techniques in terms of transmission facilities, signaling rate and operating mode.

Table 6-2 provides a qualitative comparison between conventional analog signaling systems and three types of digital systems. The digital signaling mixtures are those on a per channel basis, and the common channel signaling system operating in either the associated or non-associated mode. The comparison parameters used represent the unique military environment characteristics described in Section 3. Although Table 6-2 is largely subjective, it appears that both common channel inter-switch systems come close to meeting most of the AADSS objectives.

6.2. Intranet and Internet Gateways

Individual AADSS installations may face different requirements. There may be a variety of regional applications and services. At the local level, the AADSS may be viewed as a protocol transformation system that interfaces dissimilar terminals (see Fig. 2-3 in Sec. 2.1.2). In this intranet application, the AADSS performs protocol, mode, format and speed conversions for various interacting terminals.

At the regional level, the AADSS need not interface with any terminals directly, but it does provide gateways between switch

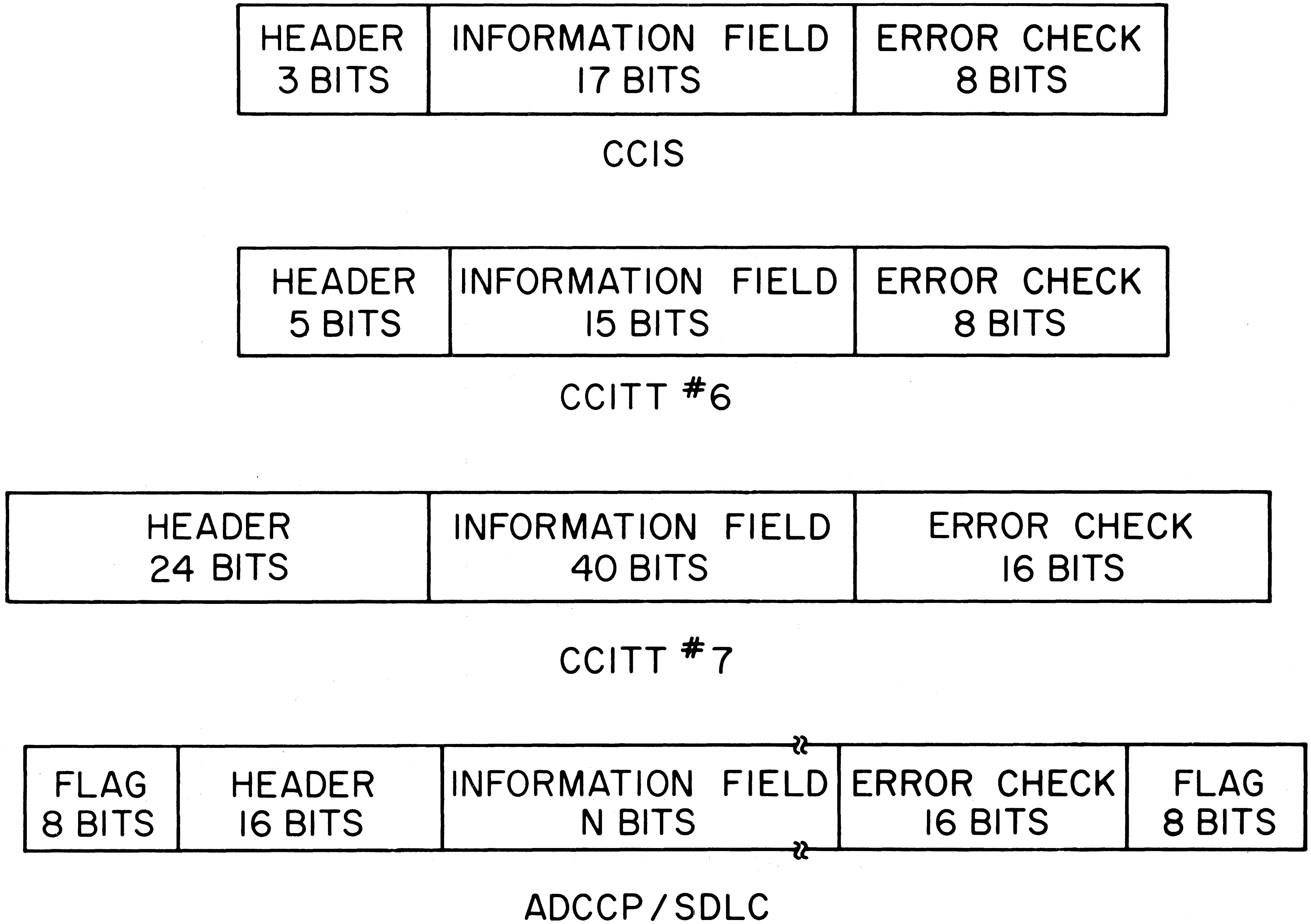


Figure 6-2. Signal format comparisons.

Table 6-1. Common Channel Signaling Technique Comparisons

Type	Transmission	Rate	Operating Mode
CCIS	Quasi-Analog	2.4 kb/s	Nonassociated
CCITT No. 6	Quasi-Analog	2.4 kb/s	Associated and Quasi-Associated
CCITT No. 6	Digital	4 kb/s to 64 kb/s	Associated and Quasi-Associated
CCITT No. 7	Digital	64 kb/s (max)	Associated and Nonassociated
ADCCP	Digital or Quasi-Analog	Unspecified	Associated

Table 6-2. Qualitative Signaling System Comparisons

Item \ Method	Conventional (Analog)	Per Channel (Digital)	Common Ch. (Associated)	Common Ch. (Non-associated)
Compatibility	High	High	Med	Med
Survivability	High	High	Med	Low (with single route)
Reliability	High	Med	Med	Med
Adaptability	Low	Med	High	High
Mobility	Med	High	High	High
Service Choices	Low	Med	High	High
Security/Privacy	Low	Med	High	High
Complexity	Low	Med	Med	High
Relative Cost	Low	Med	Med	High

nodes of different types for local and backbone networks. This AADSS aspect was depicted previously in Figure 2-2.

The AADSS characteristics must vary as the applications change from local, to regional, and possibly to inter-regional levels.

A summary of the basic characteristics of each level is given in Table 6-3.

Figure 6-3 illustrates a common channel signaling system operating between two terminals with different protocols, but connected to the same network. Here the protocol transformation occurs on the line side of the switch. The common channel signaling system between the switches may be operated in the associated or non-associated mode, using a common protocol for communicating between the control processors. The common protocols may differ from those used by either terminal.

Figure 6-4 depicts the signaling interface transformations between networks. Here the AADSS serves as an internet gateway. Network A may be a local digital distribution system on a military base. From network A, data are transferred to the backbone network via the AADSS where a protocol transformation takes place. Both data and converted signaling information are forwarded to another network via digital or analog trunks. Network B may be the DCS backbone and network C may be a common carrier, for example.

A time division multiplexer is shown as the trunk interface for network A. The digital trunk may be a T-carrier with Tl, T2, or even T4 trunks, depending on the traffic loads.

A block diagram of the AADSS for regional internetting is shown in Figure 6-5. Both analog and digital trunk interfaces can be added or deleted in modular form as the interconnecting networks undergo future transitions. The basic switch is assumed to be a hybrid type, capable of handling both circuit switched voice and packet switched data. The control signals are transferred via the packet switched networks, just like any other data.

Table 6-3. Summary of AADSS Characteristics

	Application		
	Local	Regional	Interregional
Network Topology	Star Loop Tree	Loop Grid Sparsely Connected	Tree Grid Nearly Fully Connected
Terminals	Voice Data	None (Internodal)	None (Internodal)
Traffic	Analog and Digital	Primarily Digital (Analog to DCO)	All Digital
Links	Wire Pairs T-Carrier Packet Radio Fiber Optic Cable Coaxial Cable	T-Carrier FDM	Microwave Radio Coaxial Cable Satellite Radio
Switching (all SPC)	Separated Space & Time Div.	Hybrid Space & Time Division	Hybrid Space & Time Division
Signaling System			
• Protocols	Special Purpose	Transformable	General Purpose Transparent Protocol
• Mode	Per Channel & Associated	Associated & Quasi-associated	Associated & Quasi- associated
• Transmission	Analog, Quasi- analog & Digital	Quasi-analog & Digital	Quasi-analog & Digital Undetermined
• Synchronization	Central Clock	Undetermined	ARQ on Terrestrial
• Error Control	CRC with ARQ	CRC with ARQ	FEC on Satellite

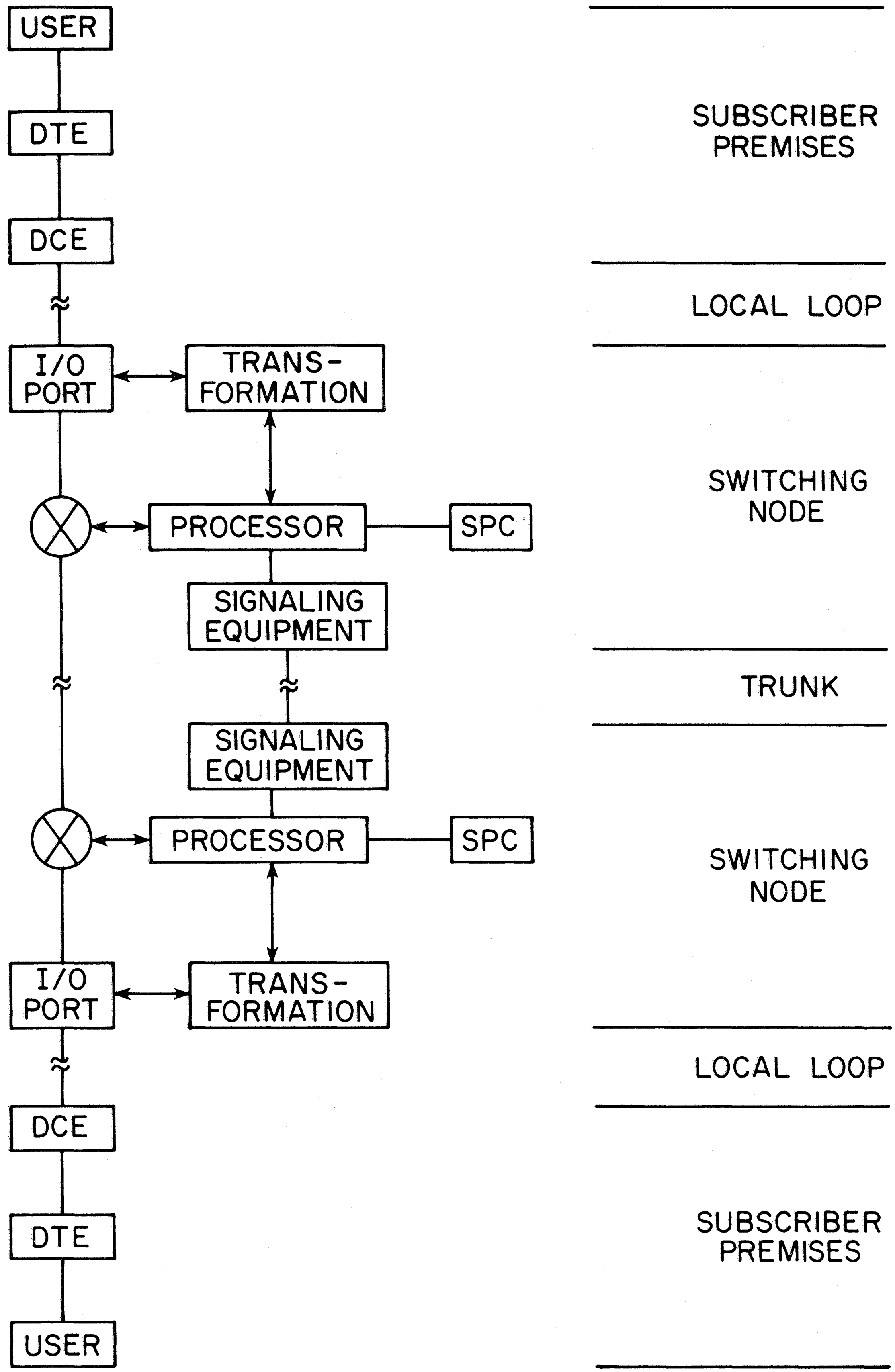


Figure 6-3. Intranet gateway.

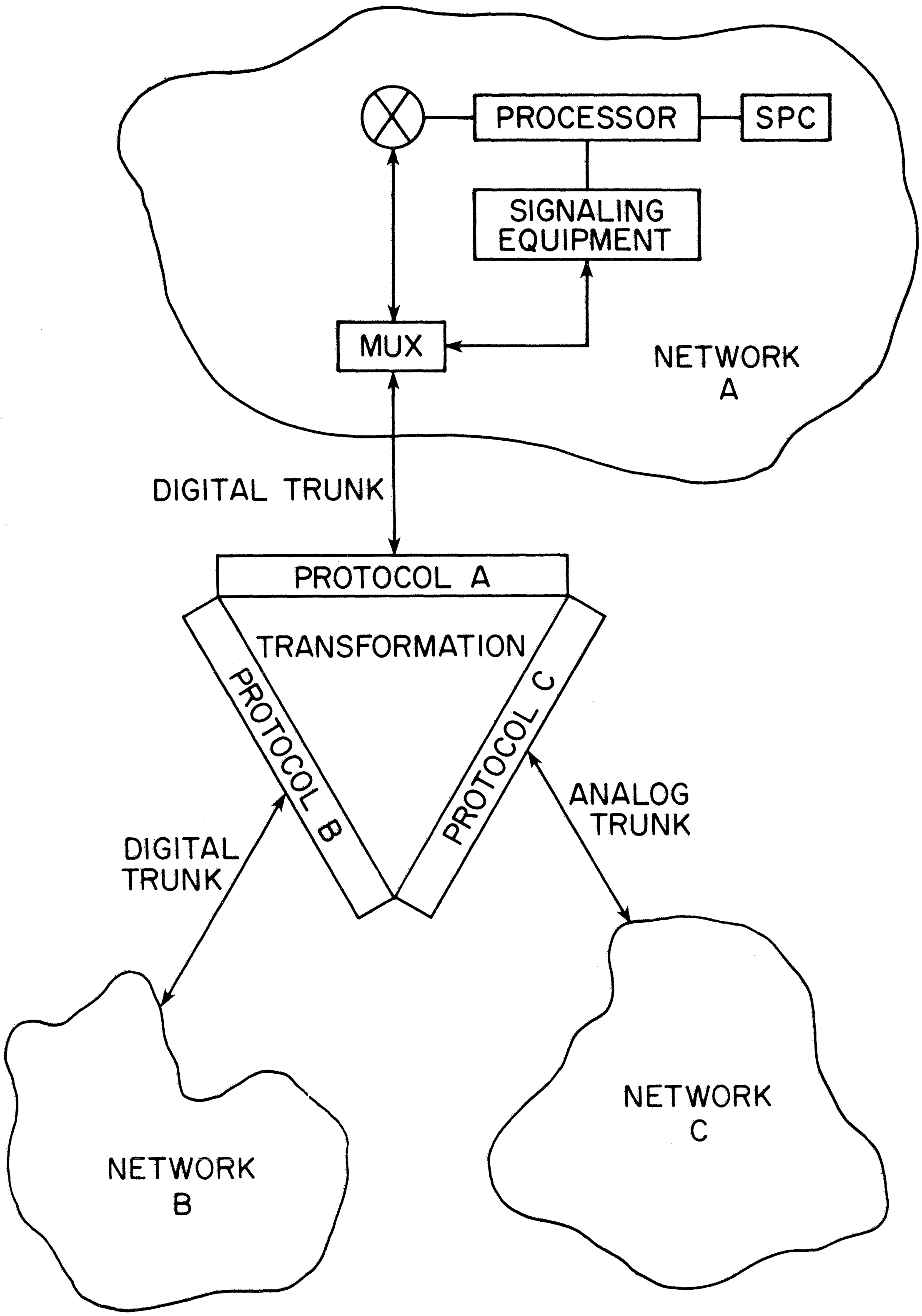


Figure 6-4. Internet gateway.

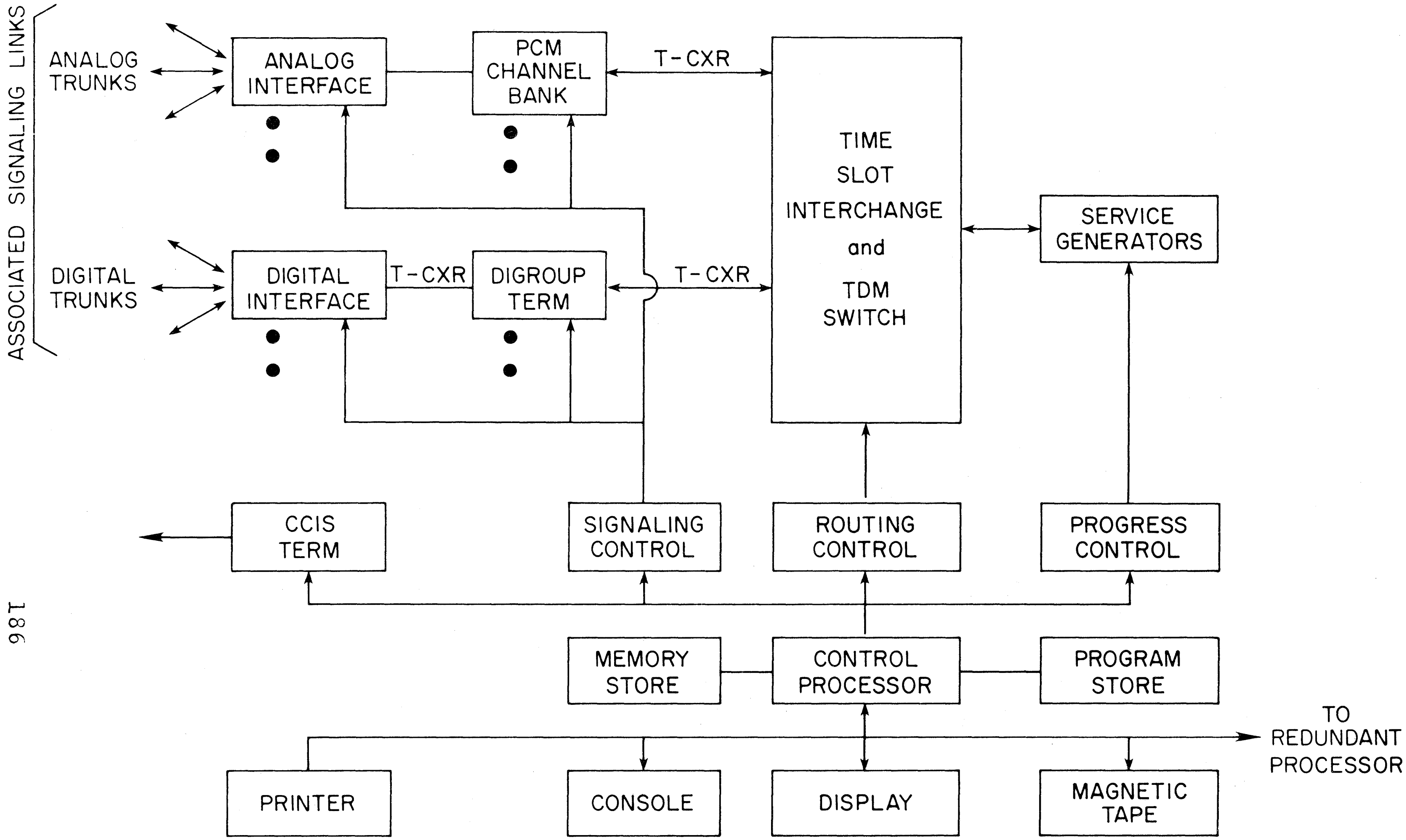


Figure 6-5. Transitional AADSS for regional internetting.

The signaling for circuit switched networks may be multiplexed onto the T-carrier. Therefore, the interswitch signaling is normally operated in the associated mode, although in some cases a quasi-associated mode could be used. A general-purpose protocol provides the means for interprocessor communications and for transferring user data.

For signaling, the information field conveys the control data. For data transfer the same field conveys user data. The transmission channels are therefore interchangeable.

As an example, it is possible to envision a packet-switched network protocol which utilizes data packets for transferring user data, as X.25 does. At the same time it could utilize control packets containing the CCITT System No. 6 format or CCIS format to handle circuit switched calls.

6.3. Example of AADSS Application

A local digital distribution system for Ft. Monmouth, New Jersey, using coaxial cable has been evaluated in terms of cost parameters by Nesenbergs and Linfield (1976). This access area, with a population of approximately 15,000 has one terminal for every three persons. Over 95% of these terminals are mainline telephones. Current traffic estimates, assuming all terminals were operated simultaneously, could probably be accommodated with about 45 MHz of channel bandwidth. One network configuration envisioned to handle this traffic is shown in Figure 6-6. A coaxial cable, essentially identical to that used for cable television, is used to connect a series of switching and concentration hubs around the base. Primary control to these access hubs is located at a centralized hub, in this case, Vail Hall. The network configuration may be either a star or a loop, the latter being used in Figure 6-6. Connection between subscriber terminals and the hubs (not shown on the figure) may be two-wire pairs, or additional coaxial cables using a tree, loop or star subnet topology.

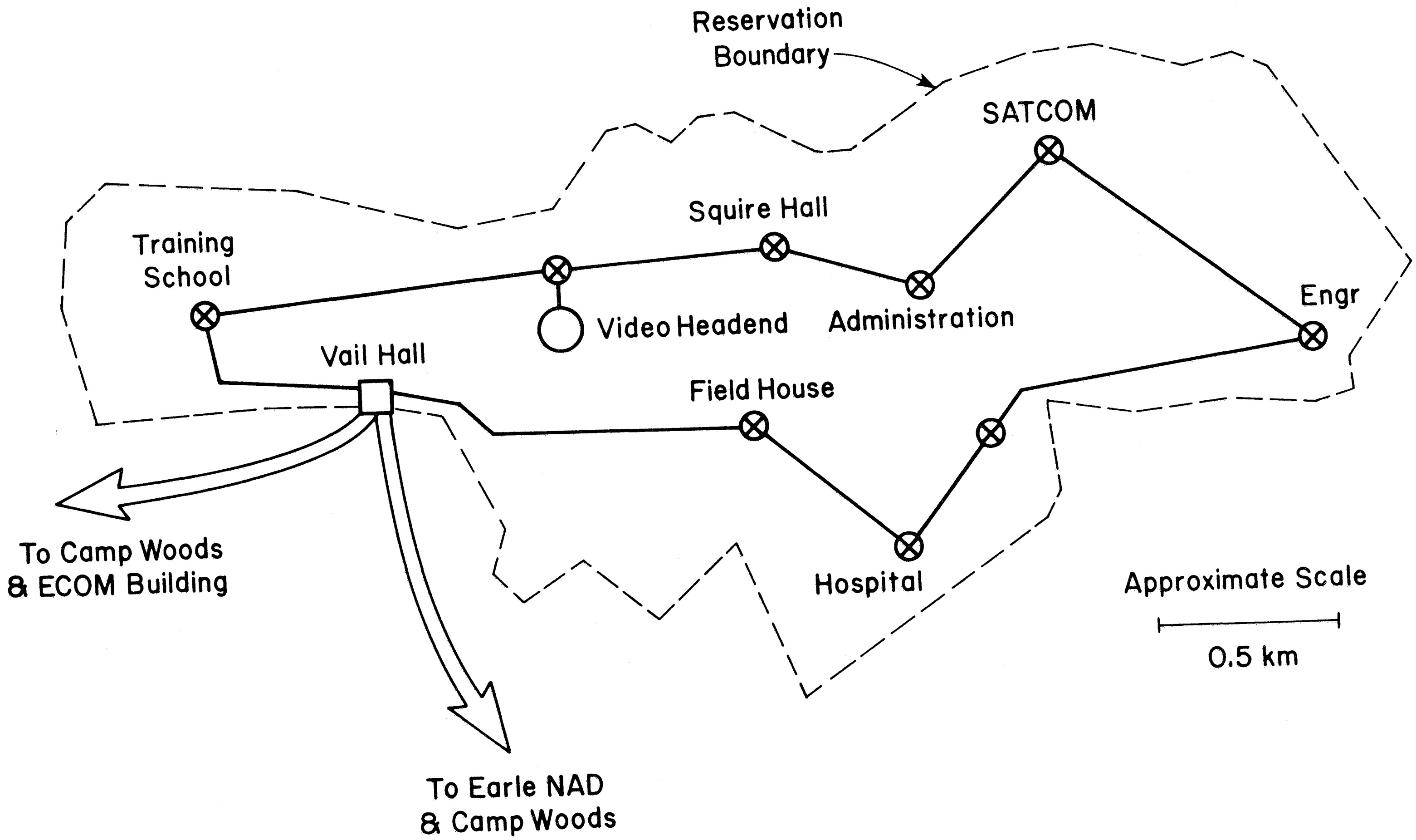


Figure 6-6. Local access at Ft. Monmouth, NJ via coaxial loop configuration.

The central hub, shown as a square in Figure 6-6, is also connected to other military facilities in the area using T-carrier lines. Ultimately, these lines connect to the strategic backbone network. This is depicted in Figure 6-7. All of the military bases, including Army, Navy, and Air Force facilities in the New Jersey area, are interconnected using a line or tree network. Access to terrestrial backbone switches is accomplished at both ends of the line to enhance the survivability. In addition, access to a Defense Communications Satellite System (DSCS) terminal provides connectivity to the Digital European backbone (DEB) and hence to the military bases in Western Europe (see Fig. 6-8). The DEB consists primarily of line-of-sight microwave facilities linking the backbone switches. Troposcatter facilities link across non-accessible regions to switching centers in Spain, Turkey, Africa, etc. This is exemplified in Figure 6-8 by the link from the southern end of the DEB in Italy to a backbone switch in Spain.

Figure 6-9 is a block diagram of the nodes and links required to connect voice or computer terminals in Ft. Monmouth to voice or computer terminals somewhere in Europe, e.g., Spain in this example.

Access to public networks in CONUS may be made either from the dial central office (DCO) on the base as shown in Figure 6-9 or from the AADSS which serves as the DCO for several dispersed clusters of terminals.

Access to tactical networks and terrestrial or satellite backbone networks is also via the AADSS. Figure 6-9 depicts access to the DEB via DSCS terminals. In Europe the digital backbone may include LOS microwave links, troposcatter links and coaxial cable links. Ultimately, connections are made to local terminals through another AADSS which also provides the interface to the European telephone system.

A computer terminal operator in Western Europe who desires the use of a system resource (e.g., files and data, processors, virtual terminals) in Ft. Monmouth is concerned with exchanging

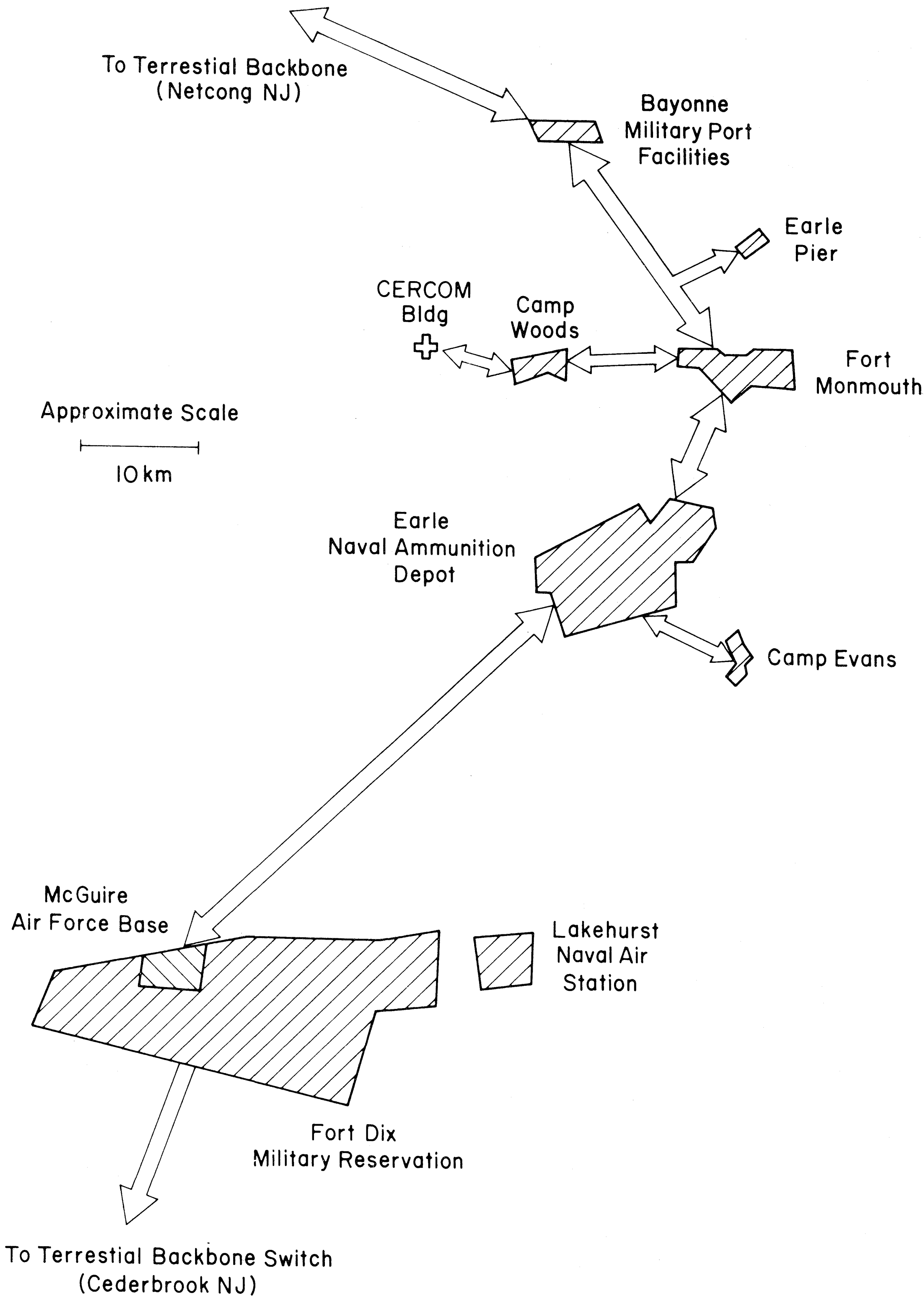


Figure 6-7. Regional access to military reservations in New Jersey area via T-carrier.

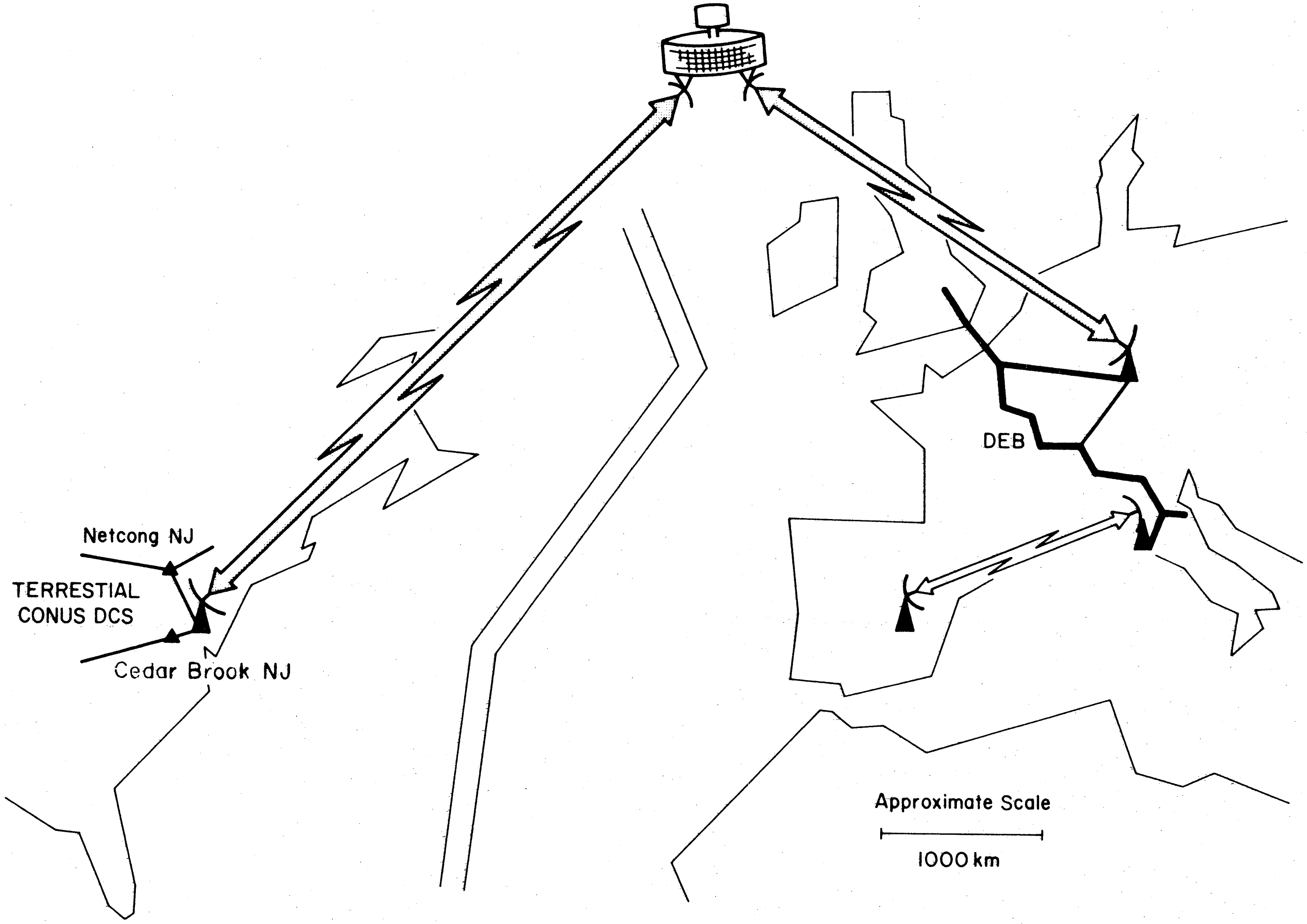


Figure 6-8. Strategic access to digital European backbone via DSCS.

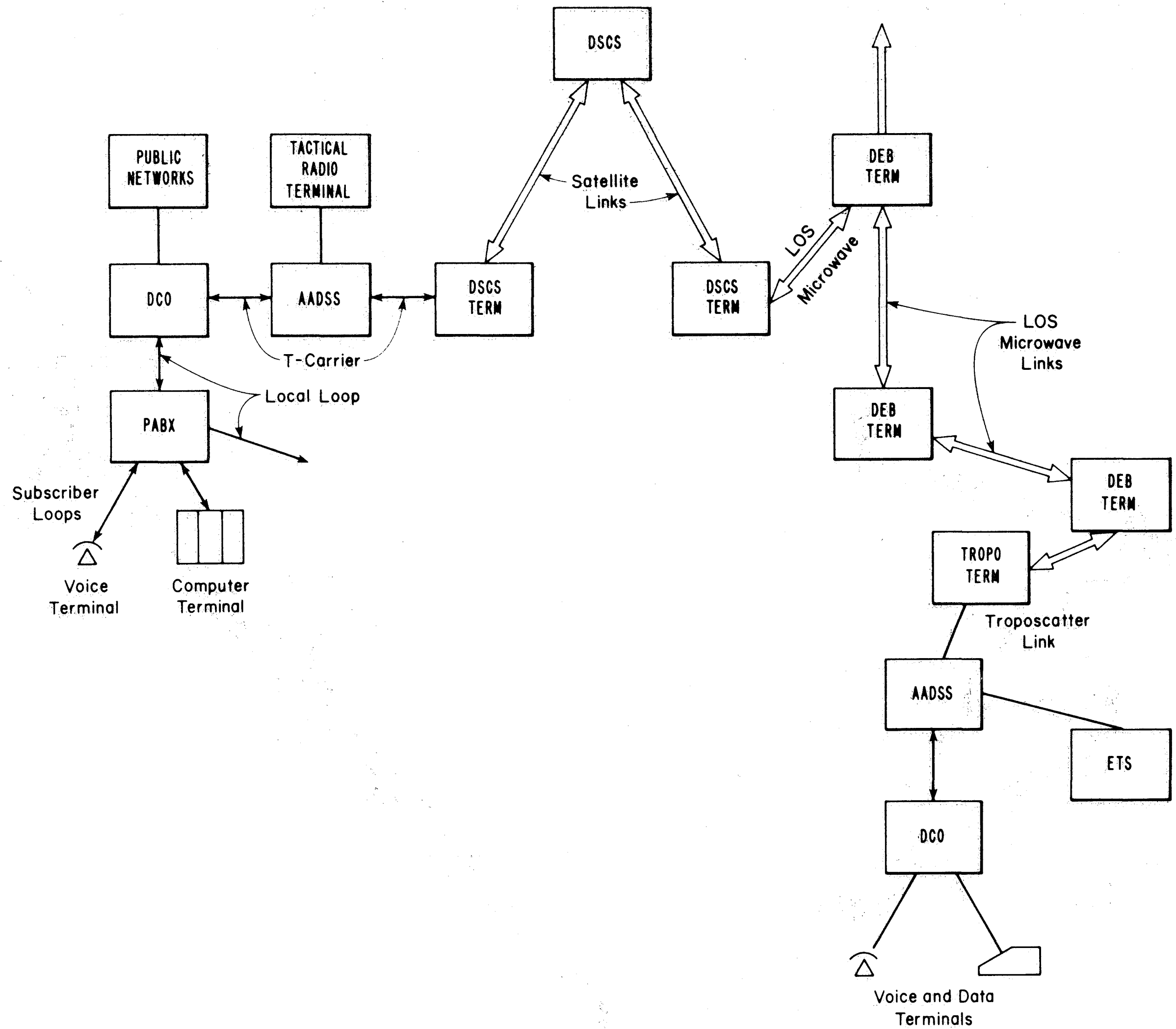


Figure 6-9. Terminal access via the AADSS.

information between these resources and with the content of the processor languages emitted and recognized by these resources. This user-oriented control function is the level 5 protocol discussed previously. It depends on all the previous levels 1 through 4 for proper operation. Thus the terminal operators at each end of the link view the network from level 5 and the network elements perform the other functions leading to this level.

In Figure 6-9 the level 4 functions may consist of several separate protocols for particular applications such as file transfers and access to virtual terminals, e.g., CRT terminals, printers, and other displays. These protocols may reside in the DTE itself. Level 3 protocols for communicating over an arbitrary network reside at the DCO or more likely in the AADSS level 2 protocols reside at every node so that control is maintained between each node. Level 1 is the lowest level required to establish, maintain and disengage communications between the network resources. The purpose of the function and application for these five levels are summarized in Table 6-4.

7. RECOMMENDED STUDY AREAS

This report has shown that the signaling system affects the operation and the performance of the network. Thus, the ultimate choice of a signaling system requires the resolution of issues which involve the entire network and all of its elements. Basic issues concerning the network characteristics and the switching node capabilities must be resolved before the final signaling system is selected. These basic issues are listed in Table 7-1. A regional AADSS is concerned with all of these issues, but rather selectively in different areas.

7.1. The Impact of the Signaling System on Processor Capabilities

This report has reviewed how user and control information may proceed through several nodes of a network, and how it may

Table 6-4. Control Functions or Protocol Levels Involved Between Terminals in Figure 6-9

Level	Function of Protocol	Purpose	Application Area Examples
1. Interface Control	Establish, maintain and disconnect physical link	Mechanical, electrical and functional interchange between DTE and DCE	Between terminal to line circuit on base
2. Link Control	Ensure reliable data transfer over single link	Provides control between two physical nodes	Between DCO and AADSS
3. Communications Control	Ensure reliable transfer via multiple links and nodes, i.e., Network Transport	Formatting, routing, network management	Between AADSS's
4. System Control	Defines system resources to be virtually interconnected	Formatting data information fields and characterizing information transfer	Remote terminal control to peripheral devices
5. User Control	Actual information exchange between resources	User level coordination of operations	Users of remote terminal and interaction between network resources

Table 7-1. AADSS Issue Summary

Network	Switching	Signaling
o Topology	o Voice/Data Integration	o Type of System
o Routing Strategy	o Hub/Concentration Point Locations	o Operating Mode
o Numbering Plan	o Hardware and Software Roles	o Transmission
o Centralized vs. Distributed Control	o Features and Services	o Security
o Interworking	o A/D Conversion Points	o Interfacing
o Link Capacities	o Traffic Requirements	
o Error Control	o Blocking Requirements	
o Security	o Delays Requirements	

be transferred between networks using different control procedures. It is assumed that the nodes of future military networks will be automatically operated by processors with stored program control. The interprocessor communications may take different forms, such as CCIS, CCITT System No. 6, CCITT System No. 7, HDLC, ADCCP, etc. Each form affects the traffic management capability of the processor in a different way. A study is recommended to evaluate these processor capabilities. This study should establish the speed and memory requirements of the control processor as a function of traffic statistics, signaling system, scheduling and routing strategies and, to the extent possible, the service features furnished to the users. These factors are clearly interrelated. They also depend on other system issues, such as the degree of centralization or decentralization permitted for the network control. Service definitions, service requirements, and performance objectives also play a role. The degree of integration of various traffic types and nodal capacities for such traffic substreams, are likely to be critical parameters.

7.2 Network Management

Long-term status monitoring, failure detection, redressing, and maintenance functions, as well as other aspects of network management, require further study. Different control system configurations appear suited for performing different network management functions. For example, Rosner (1978) has indicated that the maximum traffic throughput is achieved using decentralized control, whereas the assurance of network integrity tends toward a more centralized control implementation. This and other aspects, such as active-versus passive management control schemes, must be resolved before an optimum network control system can be defined, designed, and finally implemented.

7.3. Routing Strategy

The number of network control steps required for routing random traffic through networks with generally unknown topologies is another pertinent study area. Some estimates developed earlier by Nesenbergs and Linfield (1976) showed that the number of control steps is proportional to the logarithm of the number of possible routes through a network. This number of possible routes is reduced as the network structure is progressively better defined. Table 7-2 shows the number of routing control steps, or binary decisions, required for several network topologies as a function of the number of nodes, n. These relative task proportions provide a basis for estimating processor requirements, such as amount of logic circuitry, the magnitude of software, and the storage capacity required to store, forward, and to route calls.

Table 7-2. Number of Routing Control Steps for n-node Networks

Configuration	Proportionality Factor
Arbitrary Network	n^2
Arbitrary Tree	$n\log_2 n$
Arbitrary Ring	n
Simplest Star or Ring	$\log_2 n$
Constrained Tree	n to $n\log_2 n$
Constrained Ring	$\log_2 n$ to n
Practical Tree	$\sim n$
Practical Ring	$\sim \sqrt{n}$

7.4. Multiple Access Protocols for Integrated Satellite Networks

The multiple-access protocols for packet radio and satellite broadcast systems are still in the experimental evaluation stage. One advanced protocol (CPODA) is being tested on ARPANET satellite IMP's. The system utilizes the slotted envelope network (SENET) concept in conjunction with a reservation ALOHA type of protocol. The CPODA protocol was discussed briefly in Section 5.2. Although the general approach appears to be sound and cost effective, the detail specifications for this protocol have not been finalized. Further studies are expected to result in a number of modifications to the protocol.

7.5. Network Performance Standardization

In AADSS, just as in other tactical, non-tactical, and DCS backbone networks, the traffic handling capability has the ultimate effect on usefulness and cost-effectiveness of military systems. Straightforward traffic increase and forced overloading of facilities results in deterioration of network performance. While qualitatively this appears simple enough, the causes and effects are by no means clear quantitatively.

It appears important to agree first what parameters should be used for AADSS network performance. The previously mentioned delays, blocking probabilities, throughput efficiencies, and perhaps other characteristics, may be good initial candidates for network performance description.

At the second level, network performance standardization can only be completed by assigning numerical values to the above parameters. Thus, for instance, the standard blocking probability for circuit switched telephone lines could be specified as 1% at the end offices. For trunks, depending on various factors, one could specify a lower standard blocking probability, such as perhaps 0.2%. For message and packet switching, standard delay numbers may be quite important. End-to-end delays impact on the quality of digitized packet voice. Link and node delays relate to buffer size, processor speed, and channel speed requirements.

7.6. Traffic Capabilities

The basic requirement for AADSS is to carry telecommunications traffic. Three technical descriptors are needed to specify that traffic:

(a) The total traffic volume (in b/s or Hz) through given gateways.

(b) Individual traffic substreams, generated by different user scenarios, their service modes, speeds and access statistics.

(c) The service requirements of the constituent users.

The network performance standards, outlined earlier under Section 7.5, must be sufficient to provide for the demands of (b) and (c). And in the interests of economic viability, it must do that at the highest overall traffic volume (a).

It appears, therefore, that AADSS intranetwork and inter-network telecommunications should be traffic engineered to carry the total flow in the most efficient, cost effective, manner-while satisfying the DoD network performance requirements.

7.7. Computer Simulation

Computer simulation is an ideal tool for resolving some of the problems associated with the studies recommended above. Tests have already been conducted at ITS using a PDP-11/40 computer and the ARPANET to measure user performance parameters (see Payne, 1978). This facility could also be used to simulate certain AADSS functions on the digital computer and to evaluate signaling protocols on a packet-switched network including, for example, routing strategies, performance standardization and traffic capabilities.

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APPENDIX. SIGNAL UNIT FORMATS AND CODES FOR CCITT SIGNALING SYSTEM NO. 6

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SECTION 3

SIGNAL UNIT FORMATS AND CODES

Recommendation Q.257

3.1 GENERAL

3.1.1 *Types of message and signal unit (SU)*

Signalling and other information carried by the common signalling link is transferred by means of messages consisting of one or more signal units.

A signal unit (SU) is the smallest defined group of bits on the signalling channel and contains 28 bits.

Dependent upon the number of signal units necessary to transmit one message, the message is called a one-unit message or a multi-unit message.

3.1.1.1 *One-unit message, lone signal unit (LSU)*

A one-unit message is a message which is transmitted entirely within one signal unit. Such a signal unit is called a lone signal unit (LSU). It is designed to transmit either:

a) a single telephone signal,

b) a signalling-system-control signal, or

c) a management signal.

3.1.1.2 *Multi-unit message (MUM)*

A multi-unit message (MUM) consists of 2, 3, 4, 5 or 6 signal units in tandem. It is designed to transmit a number of related signals (e.g. address signals) in an efficient way. A special case of the mulit-unit messages is the initial address message, which is the only one which can have six signal units in tandem and has a minimum of three signal units.

3.1.1.3 *Initial signal unit (ISU)*

The first signal unit of a multi-unit message is called the initial signal unit (ISU).

3.1.1.4 *Subsequent signal unit (SSU)*

The second and any following signal unit of a multi-unit message are called subsequent signal units (SSU).

3.1.2 *Basic formats*

3.1.2.1 *Basic format of a lone signal unit*

The basic format of a lone signal unit is shown in Figure 5/Q.257.

X X X X X /	X X X X /	X X X X X X X /	X X X X /	X X X X X X X X
1-5	6-9	10-16	17-20	21-28
Heading	Signal information	Band number	Circuit number	Check
		$\underbrace{\text{Band number} \quad \text{Circuit number}}_{\text{Label}}$		

FIGURE 5/Q.257 – Basic format of: – a lone signal unit
– an initial signal unit of a multi-unit message

The basic format of a lone signal unit is not used in all cases. Where a different format is used it is shown in the sections relating to individual signal units.

3.1.2.2 *Basic format of a multi-unit message*

The format of the initial signal unit of a multi-unit message is shown in Figure 5/Q.257. The use of a special code in the signal information field (bits 6-9) distinguishes an initial signal unit from a lone signal unit. See 3.1.2.1 above.

The format of a subsequent signal unit of a multi-unit message is shown in Figure 6/Q.257.

0 0 /	X X /	X X X X X X X X X X X X X X X X /	X X X X X X X X
1-2	3-4	5-20	21-28
*	**	Signal information	Check

* Subsequent signal unit heading code
** Length indicator

FIGURE 6/Q.257 – Format of a subsequent signal unit of a multi-unit message

For some messages, the signal information field of a subsequent signal unit (bits 5-20) can be sub-divided, notably in address messages where the field is divided into four 4-bit parts.

3.1.3 *Codes for the general parts of signal units*

The interpretation of a message depends upon a system of codes in various parts of the message.

3.1.3.1 *Heading*

The heading is used to identify the type of:

a) group of signals,

b) message, or

c) signal.

The heading generally consists of the first five bits of the signal units (bits 1-5). There are two exceptions to this rule, viz.:

– all subsequent signal units are identified by the same 2-bit heading code **0 0** (bits 1-2);

– the acknowledgement signal unit is identified by a 3-bit heading code **0 1 1** (bits 1-3).

The heading codes are allocated as follows:

0 0	Subsequent signal unit
0 1 0 0 0 **0 1 0 0 1** **0 1 0 1 0** **0 1 0 1 1**	Spare (reserved for regional and/or national use)
0 1 1	Acknowledgement signal unit
1 0 0 0 0	Initial signal unit of an initial address message (or of a multi-unit message)
1 0 0 0 1 **1 0 0 1 0** **1 0 0 1 1** **1 0 1 0 0** **1 0 1 0 1** **1 0 1 1 0** **1 0 1 1 1**	Subsequent address message (one-unit message or multi-unit message)
1 1 0 0 0 **1 1 0 0 1** **1 1 0 1 0** **1 1 0 1 1**	International telephone signals
1 1 1 0 0	Spare (reserved for regional and/or national use)
1 1 1 0 1	Signalling-system-control signals (except acknowledgement signal unit) and management signals
1 1 1 1 0 **1 1 1 1 1**	Spare (reserved for regional and/or national use)

The heading code allocation is also shown in Table 1 at the end of this Section.

3.1.3.2 *Signal information*

Signal units with a 5-bit heading code have a signal information field of four bits (bits 6-9). The signal information field is used:

a) to define a particular signal within a group of signals being defined by the heading code.

b) to define a sub-group within a group of signals, or

c) to indicate that the signal unit is an initial signal unit and that the subsequent signal unit(s) contain(s) a number of signals belonging to the group of signals defined by the heading code.

For case *c)*, the signal information code **0 0 0 0** is used except with heading code **1 0 0 0 0** which alone is sufficient to identify the signal unit as an initial signal unit.

The allocation of signal information codes is shown in Table 1 at the end of this Section.

3.1.3.3 *Label*

Messages which relate to a speech circuit (or a group or sub-group of speech circuits) must carry a label to identify that circuit (or group of circuits). Only one label per message is used.

To identify a group of up to sixteen speech circuits, a 7-bit *band number* is used (bits 10-16).

To identify a circuit within a group of up to sixteen speech circuits, an additional 4-bit code (circuit number) is used (bits 17-20). See Figure 5/Q.257.

This provides a total of 11 bits which can be used to identify 2048 speech circuits.

Label codes will be assigned by the Administration concerned.

The label field position is in bits 10-20 of either a lone signal unit or an initial signal unit of a multi-unit message. Subsequent signal units of multi-unit messages do not require a label. Where a 7-bit band number alone is sufficient to identify the destination of a signal (e.g. some management signals), bits 17-20 can contain some further signalling information.

3.1.3.4 *Length indicator*

Subsequent signal units have a length indicator field of two bits (bits 3-4) to indicate the number of subsequent signal units contained in a multi-unit message. Each subsequent signal unit of a multi-unit message carries the same length indicator. The codes used are shown below.

Number of subsequent signal units	Length indicator	
	Initial address message	Other multi-unit messages
1	–	00
2	01	01
3	10	10
4	11	11
5	00	–

The length indicator **0 0** has a different, but unambiguous meaning in the initial address message because the initial address message has a minimum requirement of two subsequent signal units.

3.1.3.5 *Check*

Every signal unit has a check field of eight bits (bits 21-28) for error detection purposes (see Recommendation Q.277).

Recommendation Q.258

3.2 TELEPHONE SIGNALS

3.2.1 *Initial address message (IAM)*

The initial address message (IAM) is the first message of a call. It is a special case of the multi-unit message as it consists of a minimum of three signal units and a maximum of six signal units. It can contain different types of information – address signals (including ST), other routing information, and the filler code – under the same heading code.

3.2.1.1 *Format of the initial address message*

The format of the initial signal unit is shown in Figure 5/Q.257.

The format of the subsequent signal units is shown in Figure 6/Q.257 except for the subsequent signal units numbers 2-5 in which the signal information field (bits 5-20) is sub-divided into four 4-bit parts so that four address signals can be carried in each of these subsequent signal units.

The subsequent signal units of an initial address message do not require the 5-bit heading or 11-bit label as this information is already contained in the initial signal unit.

The number of address signals available for transmission determines the length of the initial address message.

3.2.1.2 *Codes used in the initial address message*

a) *Initial signal unit*

– The 5-bit heading code **1 0 0 0 0** is used.

– The signal information code **0 0 0 0** is used.

– The assigned label code is used.

b) *Subsequent signal unit (number 1)*

– The heading code **0 0** is used.

– The length indicator is coded as appropriate (see Recommendation Q.257, 3.1.3.4).

– Bit 5: country code indicator:
  - **0** country code not included
  - **1** country code included

– Bit 6: nature of circuit indicator:
  - **0** no satellite circuit in the connection
  - **1** one satellite circuit in the connection

– Bit 7: echo-suppressor indicator:
  - **0** outgoing half-echo suppressor not included
  - **1** outgoing half-echo suppressor included

– Bit 8: spare (reserved for international use) [1)]

[1)] These bits are coded as **0** at present.

– Bits 9-12: spare (reserved for regional and/or national use) [1)]

– Bits 13-16: calling-party's-category indicator

0 0 0 0	spare
0 0 0 1	operator, language French
0 0 1 0	operator, language English
0 0 1 1	operator, language German
0 1 0 0	operator, language Russian
0 1 0 1	operator, language Spanish
0 1 1 0	available to Administration for selecting a particular language provided by mutual agreement
0 1 1 1	
1 0 0 0	
1 0 0 1	reserved (see Recommendation Q.104)
1 0 1 0	ordinary calling subscriber
1 0 1 1	calling subscriber with priority
1 1 0 0	data call
1 1 0 1	test call
1 1 1 0	spare
1 1 1 1	spare (reserved for regional and/or national use)

– Bits 17-20: spare (reserved for regional and/or national use) [1)]

c) *Subsequent signal units (numbers 2-5) – telephone call*

– The heading code **0 0** is used.

– The length indicator is coded as appropriate (see Recommendation Q.257, 3.1.3.4).

– The four 4-bit parts of the signal information field contain address signals in sequence, bits 5-8, bits 9-12, etc., and coded as follows:

0 0 0 0	filler (no information)
0 0 0 1	digit 1
0 0 1 0	digit 2
0 0 1 1	digit 3
0 1 0 0	digit 4
0 1 0 1	digit 5
0 1 1 0	digit 6
0 1 1 1	digit 7
1 0 0 0	digit 8
1 0 0 1	digit 9
1 0 1 0	digit 0
1 0 1 1	code 11
1 1 0 0	code 12
1 1 0 1	spare
1 1 1 0	spare
1 1 1 1	ST

The filler code **0 0 0 0** is used where needed to complete the signal information field of the last subsequent signal unit of the initial address message.

d) *Subsequent signal unit (number 2) – test call*

– The heading code **0 0** is used.

– The length indicator is coded as appropriate (see Recommendation Q.257, 3.1.3.4).

– The first 4-bit part (bits 5-8) of the signal information field contains an address signal coded as follows:

0 0 0 0	system No. 6 continuity check
0 0 0 1	ATME 2 – signalling check and transmission test
0 0 1 0	ATME 2 – signalling check only
0 0 1 1	spare
0 1 0 0	spare
0 1 0 1	spare
0 1 1 0	spare
0 1 1 1	spare
1 0 0 0	spare
1 0 0 1	spare
1 0 1 0	spare
1 0 1 1	spare
1 1 0 0	spare
1 1 0 1	spare
1 1 1 0	spare
1 1 1 1	spare

The codes used to complete the signal information field of the subsequent signal unit (number 2) test call are the end-of-pulsing (ST) and fillers.

3.2.1.3 *Example of an initial address message*

An example of a three-unit initial address message is shown in Figure 7/Q.258.

ISU	1 0 0 0 0 /	0 0 0 0 /	X X X X X X X X X X X X /				X X X X X X X X
	Heading code	Signal information code	Label				Check
1st SSU	0 0 /	0 1 /	X X X 0 0 0 0 0 X X X X 0 0 0 0 /				X X X X X X X X
	*	**	Other routing information				Check
2nd SSU	0 0 /	0 1 /	X X X X	X X X X	X X X X	X X X X /	X X X X X X X X
	*	**	1st	2nd	3rd	4th	Check
			Address signals				

* Subsequent signal unit heading code
** Length indicator

FIGURE 7/Q.258 – Example of a three-unit initial address message

3.2.2 *Subsequent address message (SAM)*

A subsequent address message (SAM) is used to transmit additional address signals not available when the initial address message is formed.

A subsequent address message may be either a one-unit message or a multi-unit message.

3.2.2.1 *Formats of subsequent address messages*

a) *Lone signal unit*

The format of the lone signal unit is shown in Figure 5/Q.257.

b) *Multi-unit message*

The format of the initial signal unit is shown in Figure 5/Q.257.

The format of the subsequent signal units is shown in Figure 6/Q.257. In this case, however, the signal information fields of every subsequent signal unit are sub-divided into four 4-bit parts.

3.2.2.2 *Codes used in subsequent address messages*

a) *Heading*

Heading codes in the range **1 0 0 0 1** - **1 0 1 1 1** are used in the lone signal unit or initial signal unit depending on the sequence number of the subsequent address message concerned. The first subsequent address message of a call uses heading **1 0 0 0 1**, the second **1 0 0 1 0**, the third **1 0 0 1 1**, etc. While it is preferred to limit the number of subsequent address messages, if more than seven are sent, the sequence is recycled so that the eighth uses heading code **1 0 0 0 1**.

Subsequent signal units of subsequent address messages use the heading code **0 0**.

b) *Signal information*

– *Lone signal unit*

In the case of a one-unit subsequent address message, the signal information field (bits 6-9) contains one of the address signals which are coded as follows:

0 0 0 1	digit 1
0 0 1 0	digit 2
0 0 1 1	digit 3
0 1 0 0	digit 4
0 1 0 1	digit 5
0 1 1 0	digit 6
0 1 1 1	digit 7
1 0 0 0	digit 8
1 0 0 1	digit 9
1 0 1 0	digit 0
1 1 1 1	ST

Codes **1 0 1 1**, **1 1 0 0**, **1 1 0 1**, **1 1 1 0** and **0 0 0 0** are not used in the signal information field of a one-unit subsequent address message.

– *Multi-unit message*

The signal information field of the initial signal unit is coded as **0 0 0 0**.

The signal information field of the subsequent signal units contains the address signals which are coded as follows:

0 0 0 0	filler (no information)
0 0 0 1	digit 1
0 0 1 0	digit 2
0 0 1 1	digit 3
0 1 0 0	digit 4
0 1 0 1	digit 5
0 1 1 0	digit 6
0 1 1 1	digit 7
1 0 0 0	digit 8
1 0 0 1	digit 9
1 0 1 0	digit 0
1 1 1 1	ST

Signal information codes **1 0 1 1**, **1 1 0 0**, **1 1 0 1** and **1 1 1 0** are not used in multi-unit subsequent address messages.

The filler code **0 0 0 0** is used, where needed, to complete the signal information field of the last subsequent signal unit of the subsequent address message.

c) *Label*

The assigned label code is used.

3.2.3 *Other telephone signals*

3.2.3.1 *Telephone signals with heading code* **1 0 0 0 0**.

The following signal information codes, in conjunction with heading code **1 0 0 0 0**, are allocated:

0 0 0 0	initial signal unit of an initial address message (see Recommendation Q.258, 3.2.1.2)	
0 0 0 1	spare	(reserved for international use)
0 0 1 0	spare	(reserved for regional and/or national use)
0 0 1 1	spare	
0 1 0 0	spare	
0 1 0 1	spare	
0 1 1 0	spare	
0 1 1 1	spare	
1 0 0 0	spare	
1 0 0 1	spare	
1 0 1 0	spare	
1 0 1 1	spare	
1 1 0 0	spare	
1 1 0 1	spare	
1 1 1 0	spare	
1 1 1 1	spare	

The formats for messages using signal information code **0 0 0 1** have not yet been decided. The formats for messages using signal information codes in the range **0 0 1 0** - **1 1 1 1** will be determined by regional organizations and/or national Administrations.

3.2.3.2 *Telephone signals with heading code* **1 1 0 0 0**

The format of one-unit telephone signals using heading code **1 1 0 0 0** is shown in Figure 5/Q.257.

Signals, sent in the backward direction, in lone signal units using heading code **1 1 0 0 0**, are allocated signal information codes as follows:

0 0 0 1	release-guard
0 0 1 0	answer, charge (priority)
0 0 1 1	answer, no charge (priority)
0 1 0 0	clear-back No. 1
0 1 0 1	reanswer No. 1
0 1 1 0	clear-back No. 2
0 1 1 1	reanswer No. 2
1 0 0 0	clear-back No. 3
1 0 0 1	reanswer No. 3
1 0 1 0	spare
1 0 1 1	spare
1 1 0 0	spare
1 1 0 1	spare
1 1 1 0	spare
1 1 1 1	spare

Signal information code **0 0 0 0** indicates that the signal unit is the initial signal unit of a multi-unit message. This facility is reserved for possible future expansion.

3.2.3.3 *Telephone signals with heading code* **1 1 0 0 1**

The format of one-unit telephone signals using heading code **1 1 0 0 1** is shown in Figure 5/Q.257.

Signals, sent in the backward direction, in lone signal units using heading code **1 1 0 0 1**, are allocated signal information codes as follows:

0 0 0 1	spare
0 0 1 0	spare
0 0 1 1	switching-equipment-congestion
0 1 0 0	circuit-group-congestion
0 1 0 1	national-network-congestion
0 1 1 0	spare
0 1 1 1	spare
1 0 0 0	call-failure
1 0 0 1	spare
1 0 1 0	spare
1 0 1 1	spare
1 1 0 0	spare
1 1 0 1	spare
1 1 1 0	confusion
1 1 1 1	spare

Signal information code **0 0 0 0** indicates that the signal unit is the initial signal unit of a multi-unit message. This facility is reserved for possible future expansion.

3.2.3.4 *Telephone signals with heading code* **1 1 0 1 0**

The format of one-unit telephone signals using heading code **1 1 0 1 0** is shown in Figure 5/Q.257.

Signals, in lone signal units using heading code **1 1 0 1 0**, are allocated signal information codes as follows:

0 0 0 1	continuity	sent in the forward direction
0 0 1 0	clear-forward	
0 0 1 1	forward-transfer	
0 1 0 0	spare	
0 1 0 1	spare	
0 1 1 0	spare	
0 1 1 1	spare	
1 0 0 0	spare	
1 0 0 1	spare	
1 0 1 0	spare	
1 0 1 1	blocking	sent in either direction
1 1 0 0	unblocking	
1 1 0 1	blocking-acknowledgement	
1 1 1 0	unblocking-acknowledgement	
1 1 1 1	message refusal	

Signal information code **0 0 0 0** indicates that the signal unit is the initial signal unit of a multi-unit message. This facility is reserved for possible future expansion.

3.2.3.5 *Telephone signals with heading code* **1 1 0 1 1**

The format of one-unit telephone signals using heading code **1 1 0 1 1** is shown in Figure 5/Q.257.

Signals, sent in the backward direction, in lone signal units using heading code **1 1 0 1 1**, are allocated signal information codes as follows:

0 0 0 1	address-complete, subscriber-free, charge
0 0 1 0	address-complete, subscriber-free, no charge
0 0 1 1	address-complete, subscriber-free, coin-box
0 1 0 0	subscriber-busy (electrical)
0 1 0 1	vacant-national number
0 1 1 0	line-out-of-service
0 1 1 1	subscriber-transferred (changed number)
1 0 0 0	spare
1 0 0 1	spare
1 0 1 0	address-complete, charge
1 0 1 1	address-complete, no charge
1 1 0 0	address-complete, coin-box
1 1 0 1	address-incomplete
1 1 1 0	spare
1 1 1 1	spare

Signal information code **0 0 0 0** indicates that the signal unit is the initial signal unit of a multi-unit message. This facility is reserved for possible future expansion.

3.2.3.6 *Reserved heading codes*

The signal information codes under the heading codes **0 1 0 0 0**, **0 1 0 0 1**, **0 1 0 1 0**, **0 1 0 1 1**, **1 1 1 0 0**, **1 1 1 1 0** and **1 1 1 1 1** are reserved for regional and/or national use.

Signal information code **0 0 0 0** indicates that the signal unit is the initial signal unit of a multi-unit message. This facility is reserved for possible future expansion.

3.2.4 *Examples of address messages*

Examples of address messages are given below to elucidate the formats and codes adopted for address messages. As there is no telephone signal information contained in the check fields of the signal units, these fields are not shown in the examples.

3.2.4.1 *Transit call* from USA (international exchange New York) to the Netherlands (international exchange Amsterdam) via the United Kingdom (transit exchange London).

Assumptions:
- Semi-automatic traffic, English language.
- The signalling links New York-London and London-Amsterdam are both associated with their respective speech circuit groups.
- Speech path New York-London is a satellite circuit equipped with echo suppressors, speech path London-Amsterdam is a cable circuit not equipped with echo suppressors (due to bilateral agreement between the Administrations concerned).
- Dialled information: 31 2150 43551.
- *En bloc* operation.

a) *Address message New York-London*

10000/0000/ 000 0101/0011
00/11/1110 0000 0010 0000
00/11/0011/0001/0010/0001
00/11/0101/1010/0100/0011
00/11/0101/0101/0001/1111

b) *Address message London-Amsterdam*

10000/0000/ 000 0000/1010
00/11/0100 0000 0010 0000
00/11/0010/0001/0101/1010
00/11/0100/0011/0101/0101
00/11/0001/1111/0000/0000

The intermediate CT London serves as a transit exchange.

3.2.4.2 *Direct call* from the Netherlands (international exchange Amsterdam) to USA (international exchange New York).

Assumptions:
- Automatic traffic, ordinary subscriber.
- Speech path Amsterdam-New York is a cable circuit equipped with echo suppressors.
- Speech circuit group Amsterdam-New York has no associated signalling link. Signal information will be transferred via the two signalling links Amsterdam-London and London-New York in tandem, thus using a quasi-associated mode of operation.
- Dialled information: 1 201 949 5813.
- Overlap with subscribers' dialling operation.

a) *Address messages Amsterdam-London*

10000/0000/ 001 0000/1001	Initial address message
00/10/0010 0000 1010 0000	
00/10/0010/1010/0001/1001	
00/10/0100/1001/0000/0000	
10001/0101/ 001 0000/1001	– First subsequent address message
10010/1000/ 001 0000/1001	– Second subsequent address message
10011/0001/ 001 0000/1001	– Third subsequent address message
10100/0011/ 001 0000/1001	– Fourth subsequent address message
10101/1111/ 001 0000/1001*	– Fifth subsequent address message

* ST-signal, sent if the end of the address has been recognized.

b) *Address messages London-New York*

Exactly the same messages are sent as under a).

The London exchange serves as signal transfer point only. It is assumed that by agreement between the Administrations concerned there is no need for a change of label at this signal transfer point.

Recommendation Q.259

3.3 SIGNALLING-SYSTEM-CONTROL SIGNALS

3.3.1 *General*

The signalling-system-control signals are not related to telephone signal information. They are necessary for the proper functioning of the signalling system.

All signalling-system-control signals specified (see Recommendation Q.255) are transferred by means of lone signal units:

– acknowledgement signal unit,

– synchronization signal unit, and

– system-control signal unit.

3.3.2 *Acknowledgement signal unit (ACU)*

The function of the acknowledgement signal unit (ACU) is described in Recommendation Q.251.

3.3.2.1 *Format of the ACU*

The format of the ACU is given in Figure 8/Q.259.

0 1 1 /	X X X X X X X X X X X /	X X X /	X X X /	X X X X X X X X
1-3	4-14	15-17	18-20	21-28
Heading code	Acknowledgement indicators	*	**	Check

* Sequence number of block being acknowledged
** Sequence number of block completed by this ACU

FIGURE 8/Q.259 – Format of the acknowledgement signal unit

3.3.2.2 *Codes for the ACU parts*

a) *Heading*

The heading code **0 1 1** is used.

b) *Acknowledgement indicators*

The ACU contains 11 acknowledgement indicators to acknowledge sequentially the corresponding eleven signal units of a block received. That is, bit 4 refers to the first signal unit in the block being acknowledged, bit 5 refers to the second, etc. Each indicator will be coded in the following way:

0 no error detected,

1 error detected

The *error detected* condition includes signals rejected by the terminal as covered in Recommendations Q.277, Q.278 and Q.293, 8.6.1.

c) *Block sequence numbers*

Both the block being acknowledged and the block completed by the ACU are indicated by cyclic sequence numbers from the series **0 0 0, 0 0 1, 0 1 0, 0 1 1, 1 0 0, 1 0 1, 1 1 0, 1 1 1, 0 0 0**...

3.3.3 *Synchronization signal unit (SYU)*

The function of the synchronization signal unit (SYU) is described in Recommendation Q.251.

3.3.3.1 *Format of the SYU*

The format of the SYU is given in Figure 9/Q.259.

1 1 1 0 1 / 1 1 0 1 / 1 1 0 0 0 1 1	/ X X X X	/ X X X X X X X X
1-16	17-20	21-28
Synchronization pattern	*	Check

* Sequence number of signal unit in the block

FIGURE 9/Q.259 – **Format of the synchronization signal unit**

3.3.3.2 *Codes for the SYU parts*

a) *Synchronization pattern*

This pattern is coded as: **1 1 1 0 1 1 1 0 1 1 1 0 0 0 1 1.**

The first nine bits of the synchronization pattern may be considered to contain the heading and signal information fields which are coded **1 1 1 0 1** and **1 1 0 1** respectively.

The heading code **1 1 1 0 1** is used for signalling-system-control signals (except ACU) as well as for management signals. The spare signal information codes can be allocated either to system-control signals or to management signals.

b) *Signal unit sequence number*

The sequence number may have any code of the 4-bit binary code **0 0 0 0, 0 0 0 1, 0 0 1 0** up to **1 0 1 0** inclusive. The number chosen for a synchronization signal unit is determined by the position of that synchronization signal unit in the block of signal units.

The remaining codes **1 0 1 1** to **1 1 1 1** are not assigned.

3.3.4 *System-control signal units (SCU)*

The function of the system control signal units is described in Recommendation Q.255.

3.3.4.1 *Format of an SCU*

The format of an SCU is given in Figure 10/Q.259.

1 1 1 0 1 /	1 1 0 0 /	X X X /	X X X X /	X X X X /	X X X X X X X X
1-5	6-9	10-12	13-16	17-20	21-28
Heading code	Signal information code	Control information			Check

FIGURE 10/Q.259 – **Format of a system-control signal unit**

3.3.4.2 *Codes for the SCU parts*

a) *Heading*

The heading code **1 1 1 0 1** is used.

The heading code **1 1 1 0 1** is used for signalling-system-control signals (except ACU) as well as management signals. The spare signal information codes can be allocated either to system-control signals or to management signals.

b) *Signal information*

The signal information code **1 1 0 0** is used.

c) *Control information*

– bits 10-12 are coded as **0 0 1**. The other codes are spare.

– bits 13-16 are coded as **0 0 0 1**. The other codes are spare.

– bits 17-20 system-control signals, defined in Recommendation Q.255, are coded as follows:

0 0 0 0	spare
0 0 0 1	changeover
0 0 1 0	manual-changeover
0 0 1 1	spare
0 1 0 0	standby-ready
0 1 0 1	spare
0 1 1 0	load-transfer
0 1 1 1	emergency-load-transfer
1 0 0 0	spare
1 0 0 1	spare
1 0 1 0	manual-changeover-acknowledgement
1 0 1 1	spare
1 1 0 0	standby-ready-acknowledgement
1 1 0 1	spare
1 1 1 0	load-transfer-acknowledgement
1 1 1 1	spare

3.3.5 *Multi-block-synchronization signal units (MBS)*

The function of the multi-block-synchronization signal units is described in Recommendation Q.255.

3.3.5.1 *Format of an MBS*

The format of an MBS is given in Figure 11/Q.259.

1 1 1 0 1 /	1 0 1 1 /	X X X /	X X X X X /	X X X /	X X X X X X X X
1-5	6-9	10-12	13-17	18-20	21-28
Heading code	Signal information code	Control information			Check

FIGURE 11/Q.259 – **Format of a multi-block-synchronization signal unit**

3.3.5.2 *Codes for the multi-block-synchronization signal unit parts*

a) *Heading*

The heading code **1 1 1 0 1** is used.

The heading code **1 1 1 0 1** is used for signalling system control signals (except ACU) as well as management signals. See 3.3.4.2.

b) *Signal information*

The signal information code **1 0 1 1** is used.

c) *Control information*

– bits 10-12 are coded as follows:

0 0 0 multi-block monitoring signal

1 0 0 multi-block acknowledgement signal

The other codes are spare.

– bits 13-17 indicate the sequence number of the multi-block in which the multi-block monitoring signal is sent by a 5-bit binary code from the series **0 0 0 0 0**, **0 0 0 0 1**, **0 0 0 1 0**, ..., **1 1 1 1 1**, **0 0 0 0 0**.

– bits 18-20 indicate the sequence number of the block in which the multi-block monitoring signal is sent (or placed into the output buffer) [see 3.3.2.2 c) above].

Recommendation Q.260

3.4 MANAGEMENT SIGNALS

3.4.1 *General*

Management signals may include:

– network-management signals,

– network-maintenance signals,

– signalling-network-management signals,

i.e. signals concerned with the management of the signalling network and of the speech circuit network.

These signals may be transferred by means of one-unit messages or multi-unit messages.

3.4.1.1 *Basic format of management signals*

The basic format of a one-unit management message is shown in Figure 12/Q.260.

1 1 1 0 1 /	X X X X /	X X X X X X X X X X X /	X X X X X X X X
1-5	6-9	10-20	21-28
Heading code	Signal information	Management information	Check

FIGURE 12/Q.260 – **Basic format of a one-unit management message**

The management information field, bits 10-20, may be subdivided as required. When a band number is included in the management signal unit, it is placed in bits 10-16.

The format of multi-unit management signals has not yet been decided.

3.4.1.2 *Codes for management signals*

a) *Heading*

The heading code **1 1 1 0 1** is used. This code is also assigned to signalling-system-control signals (except ACU). See Recommendation Q.259.

b) *Signal information*

Signal information codes are assigned as follows:

0 0 0 1 network-management and network-maintenance signal units

0 0 1 0 spare

0 0 1 1 spare

0 1 0 0 spare

0 1 0 1 signalling-network-management signal unit

0 1 1 0 spare (reserved for regional and/or national use)

0 1 1 1 spare (reserved for regional and/or national use)

1 0 0 0 spare

1 0 0 1 spare

1 0 1 0 spare

1 0 1 1 spare

1 1 0 0 SCU (see Recommendation Q.259)

1 1 0 1 SYU (see Recommendation Q.259)

1 1 1 0 spare (reserved for regional and/or national use)

1 1 1 1 spare (reserved for regional and/or national use)

Signal information code **0 0 0 0** indicates that the signal unit is the initial signal unit of a multi-unit message. This facility is reserved for possible future expansion.

The spare international signal information codes may be assigned to either management signals or signalling-system-control signals.

c) *Management information*

The codes used in the management information field are shown later in this Recommendation.

3.4.2 *Network-management signals*

As network-management signals have not yet been defined, no detailed format can be given, except for the heading and signal information fields, which are coded as **1 1 1 0 1** and **0 0 0 1** respectively.

3.4.3 *Network-maintenance signals*

As network-maintenance signals have not yet been defined, no detailed format can be given, except for the heading and signal information fields, which are coded as **1 1 1 0 1** and **0 0 0 1** respectively.

3.4.4 *Signalling-network-management signals*

3.4.4.1 *Format of a signalling-network-management signal*

The format of a one-unit signalling-network-management message is given in Figure 13/Q.260.

1 1 1 0 1 /	0 1 0 1 /	X X X X X X X /	X X X X /	X X X X X X X X
1-5	6-9	10-16	17-20	21-28
Heading code	Signal information code	Band number	Management information	Check

FIGURE 13/Q.260 – **Format of a one-unit signalling-network-management message**

3.4.4.2 *Codes for the signalling-network-management signal unit parts*

a) *Heading*

The heading code **1 1 1 0 1** is used.

b) *Signal information*

The signal information code **0 1 0 1** is used.

c) *Band number*

The band number (bits 10-16) indicates the group or sub-group of circuits to which the signal refers (see Recommendation Q.257).

d) *Management information*

The codes used in the management information field are allocated as follows:

0 0 0 0	spare
0 0 0 1	spare
0 0 1 0	spare
0 0 1 1	spare
0 1 0 0	spare
0 1 0 1	transfer-prohibited
0 1 1 0	transfer-allowed
0 1 1 1	spare
1 0 0 0	transfer-allowed-acknowledgement
1 0 0 1	spare
1 0 1 0	spare
1 0 1 1	spare
1 1 0 0	spare
1 1 0 1	spare
1 1 1 0	spare
1 1 1 1	spare

TABLE 1 – Allocation of heading and signal information codes

Bits 1-5 / Bits 6-9	0000X	0001X	0010X	0011X	01000	01001	01010	01011	011XX	10000	10001	10010	10011	10100	10101	10110	10111	11000	11001	11010	11011	11100	11101	11110	11111	Bits 1-5 / Bits 6-9
0000	SSU				ISU of MUM	ISU of MUM	ISU of MUM	ISU of MUM		ISU of IAM	ISU of SAM 1	ISU of SAM 2	ISU of SAM 3	ISU of SAM 4	ISU of SAM 5	ISU of SAM 6	ISU of SAM 7	ISU of MUM	ISU of MUM	ISU of MUM	ISU of MUM	ISU of MUM	ISU of MUM	ISU of MUM	ISU of MUM	0000
NOT 0000					LSU	LSU	LSU	LSU		ISU of MUM	Lone SAM 1	Lone SAM 2	Lone SAM 3	Lone SAM 4	Lone SAM 5	Lone SAM 6	Lone SAM 7	LSU	LSU	LSU	LSU	LSU	LSU	LSU	LSU	NOT 0000
0000	ONE SSU or FIVE SSUs (IAM only)	TWO SSUs	THREE SSUs	FOUR SSUs	RESERVED FOR REGIONAL and/or NATIONAL USE	RESERVED FOR REGIONAL and/or NATIONAL USE	RESERVED FOR REGIONAL and/or NATIONAL USE	RESERVED FOR REGIONAL and/or NATIONAL USE	ACU													RESERVED FOR REGIONAL and/or NATIONAL USE		RESERVED FOR REGIONAL and/or NATIONAL USE	RESERVED FOR REGIONAL and/or NATIONAL USE	0000
0001											1	1	1	1	1	1	1	RLG		COT	AFC		NMM			0001
0010										RESERVED FOR REGIONAL and/or NATIONAL USE	2	2	2	2	2	2	2	ANC		CLF	AFN					0010
0011											3	3	3	3	3	3	3	ANN	SEC	FOT	AFX					0011
0100											4	4	4	4	4	4	4	CB 1	CGC		SSB					0100
0101											5	5	5	5	5	5	5	RA 1	NNC		VNN		SNM			0101
0110											6	6	6	6	6	6	6	CB 2			LOS		REGIONAL and/or NATIONAL			0110
0111											7	7	7	7	7	7	7	RA 2			SST					0111
1000											8	8	8	8	8	8	8	CB 3	CFL							1000
1001											9	9	9	9	9	9	9	RA 3								1001
1010											0	0	0	0	0	0	0				ADC					1010
1011																				BLO	ADN		MBS			1011
1100																				UBL	ADX		SCU			1100
1101																				BLA	ADI		SYU			1101
1110																			COF	UBA			REGIONAL and/or NATIONAL			1110
1111											ST	ST	ST	ST	ST	ST	ST			MRF						1111

CCITT-54418

Note. – All unassigned codes are reserved for international use. The interpretation of the abbreviations for signals is given in the List of abbreviations on the inside of the flap of the cover at the end of the Volume.

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15. ABSTRACT *(A 200-word or less factual summary of most significant information. If document includes a significant bibliography or literature survey, mention it here.)*

A number of factors, including network topology, traffic characteristics, switching technology, and transmission media, affect the choice of control signaling techniques to be used in networks of the future. In this report these factors are considered from a military standpoint. It is projected that, over the next two decades, all of the elements of military networks will undergo a transition from essentially all-analog to all-digital systems. It is also possible that a unified integrated network capable of handling digital voice and data could evolve. During the forthcoming period of transition, as well as for the ultimate network configuration, the control signaling system will play a critical role in terms of both performance and cost. This report examines that role by evaluating several advanced signaling systems, particularly those using common control links. Digital systems that have potential application on both the line and trunk sides of the digital switch are examined.

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