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Access Area Switching and Signaling: Concepts, Issues, and Alternatives

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PREFACE

The study, whose results are presented in this report, was supported by the U.S. Army Communications Systems Agency (CSA), Ft. Monmouth, New Jersey, under project order no. 501-RD. Administration and technical monitoring of the program was performed for the agency by L.H. Wagner (U.S. Army, CSA). Technical and management supervision at the National Telecommunications and Information Administration/Institute for Telecommunication Sciences was provided by Dr. P. McManamon.

This report is part of a series prepared for CSA in support of the Access Area Digital Switching System (AADSS) program. Previous related reports have dealt with parametric cost alternatives for local digital distribution systems, as well as with preliminary switching hub evaluations, for the military base environment. This report covers two topics. It describes an example of a stored program controlled digital PABX switch suitable for the access area. It reviews signaling functions and methods that are currently in common use for the remote control of switches. Subsequent reports will address more recent, as well as more complex, signaling issues and their resolution. Such complex issues arise in the planned integrated - voice plus data - communications networks. They involve such advanced signaling techniques as the common channel interoffice signaling (CCIS), and others having foreseeable impact on military switching in access areas.

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ACCESS AREA SWITCHING AND SIGNALING: CONCEPTS, ISSUES, AND ALTERNATIVES

R.F. Linfield and M. Nesenbergs*

This report covers two key tasks of the Access Area Digital Switch (AADSS) program being conducted by NTIA/ITS for the U.S. Army Communications Systems Agency.

First, a brief introduction to digital electronic private automatic branch exchanges (PABX or EPABX) with stored program control is given, followed by some examples of system design. These examples offer a background, against which AADSS switching and signaling concepts, issues and alternatives can be reviewed. Furthermore, these systems provide integrated interfaces and digital switching to local access areas of the Defense Communications System (DCS). System functions and service features are discussed and initial cost projections given for installation sizes of interest.

Second, digital and analog signaling techniques of all existing types are reviewed. The main concepts in establishing and maintaining circuit connections and other message transactions are outlined in present day and near future technology. Interface issues during the foreseeable DCS transition from analog to digital integrated systems, as well as other signaling problems in the access area, are summarized.

1. DIGITAL PABX EXAMPLE

1.1. Introduction

This discussion deals with Private Automatic Branch Exchanges, otherwise known as PABX's. The field is further restricted not only to the electronic PABX (sometimes called EPABX), but also to digital connectivity which takes advantage of the time division switching technology. By definition, a PABX provides an exchange of calls among local users, and for calls to

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and from nearby central offices and the long-distance telephone networks. In DoD communications, the local user community is typically the military base, while the telephone networks are those of DCS and the U.S. common carriers. In many foreign countries one must deal with the appropriate postal telegraph and telephone agencies.

Figure 1 illustrates the basic concept of PABX deployment to access area hubs, both remote (local) and central (head end). The hubs shown provide access area local (turn-around) telecommunications services, plus interconnections to DCS and to the end offices of the telephone companies. As shown, the hub PABX's can be directly linked by trunks.

What exactly constitutes a hub or a PABX, either physically or functionally, may be an issue for deliberation. The physical main frame at the central location may be only part of the PABX installation. Substantial hardware parts may be remoted (see Fig. 2) by distances ranging from fractions of a mile to several miles. As will be discussed later, remoting of PABX modules into terminal clusters may offer economic advantages. The individual PABX may or may not reside at any one particular private domain. Because of variously ordered processing functions along the signal path, it is unclear where a line or trunk enters or exits the PABX. Perhaps a formal designation of a line card or an interface unit as the "boundary" would be acceptable.

In the user community the central hub PABX retains more than the majority of the end-office features. Thus, at certain locations and for certain users, the new PABX offers a host of additional services, such as call forwarding, camp-on, conferencing, etc. Most of the features and functions of the older manual switches are still provided and at a higher speed. Restrictions may affect toll dialing, individual billing, and priority preempts on the PABX. Broadly, new PABX's tend to offer more commercial user features and yet remain compatible with all older telephone office systems as well as with all existing handsets. The new PABX is also said to be cost effective. The

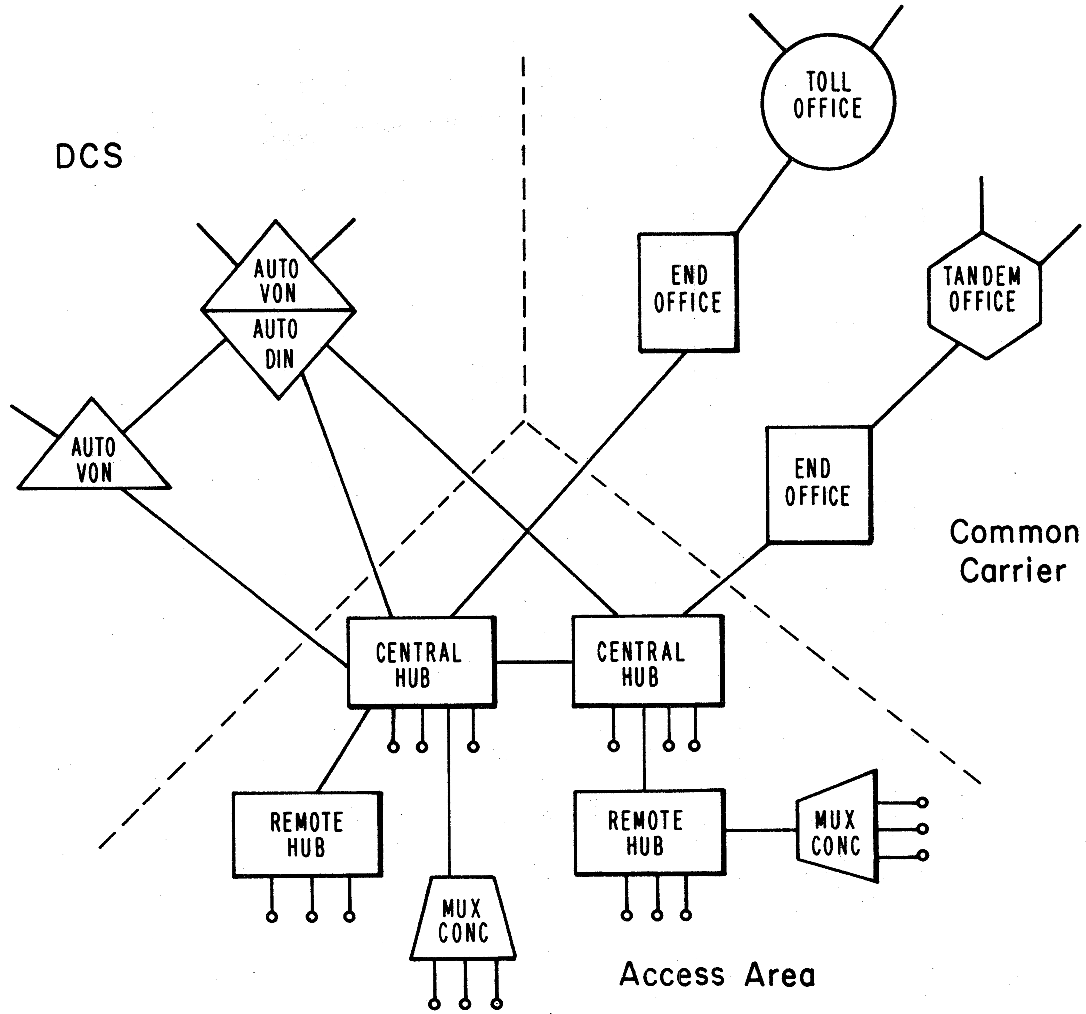

Figure 1. Basic deployment concept of access area hubs and PABX's with DCS and common carrier networks.

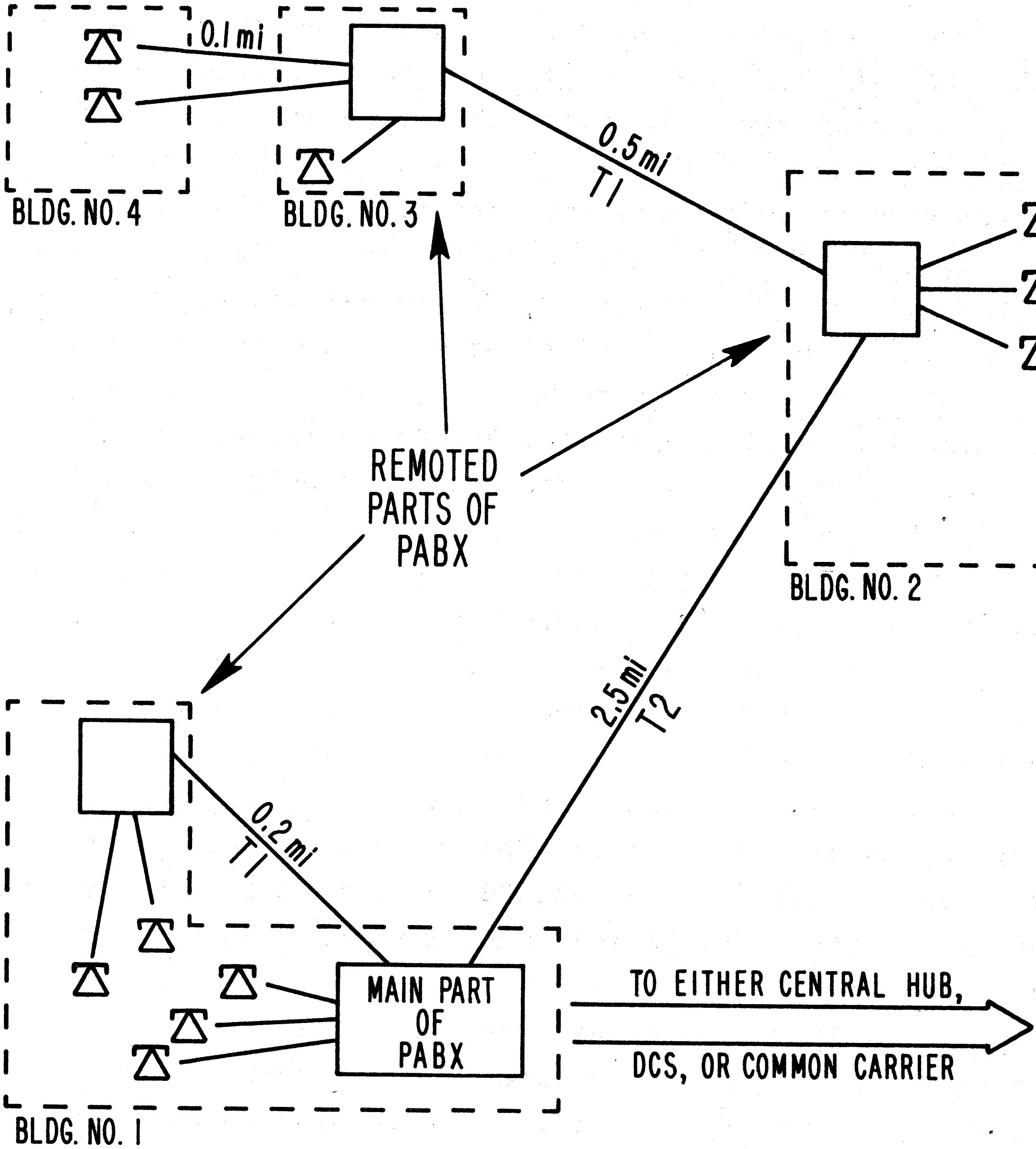

Figure 2. Dispersal of the physical PABX plant at hub level.

basis for this fortunate trend is the claimed application of more powerful and sophisticated software to the state-of-the-art LSI hardware.

Although it appears that digital technology will, in the future, permit reductions in space, in power, and eventually in the cost of integrated voice and data telecommunications equipment, it is incorrect to infer that all the technical and non-technical problems have been solved. One of the objectives of this report is to raise issues, indicate alternatives, and recommend areas where future work might be beneficially concentrated. This is done in Section 3. The issues and recommendations summarized in Section 3 relate primarily to the switching and signaling aspects of the system.

Switching issues which impact PABX design invoke questions like: What traffic and blocking assumptions should be used? Where should the analog-to-digital conversion points be located? What are the service requirements? How are the roles of hardware and software separated in the control processing units?

Signaling strategies involve the total network design and the interface structure. Signaling issues raise questions like: Should a common channel be used or should signaling be accomplished on a per-channel basis? What protocol, procedure, and mode of operation should be used? What routing strategy enhances reliability and survivability? Should network fault monitoring and controls be both centralized? Should routing and address information be secure?

In addition, there are other issues which are not directly related to switching or signaling and as such are not emphasized in this report. These issues include: transmission problems (e.g., loss, crosstalk, echoing, etc.); equipment protection problems (e.g., lightning, power line, electromagnetic transients, etc.); survivability problems (e.g., routing strategy, network topology, etc.) catastropic failure problems (e.g., hardware, software, clocking, etc.) and the network management and maintenance problems.

Throughout the design, development and installation of the AADS system a limiting factor must be the cost. A major cost item for digital switches using stored program control appears to be the software development costs. These costs tend to grow rapidly when specialized programs are required, as for the military environment. Digitizing the loop plant in the access area will have a considerable impact on cost. Operating and maintenance costs may be substantially reduced when the digital network is fully implemented. However these O&M costs may actually increase during the transition period while a mixture of old and new systems exists.

Many issues raised in this report can only be resolved by extensive field trials with operating systems. They all can benefit from further study, but to different degrees.

The contents of this first major section will address the main structure and key features of an example PABX. First, in Section 1.2 the prototype PABX, its larger components, and terms used, are introduced. In Section 1.3, the design is tailored to the present DCS trend from all-analog systems of the past to future all-digital lines and, of course, to all-digital trunks. Section 1.4 outlines a PABX example, where 10% of all telephone lines, and 100% of the trunks, are digital. The example serves, with minor modular variation, service areas that range in size from 50 to 10,000 lines, subject to an assumed grade of service level (different blocking probabilities for lines and trunks). System cost is the topic of Section 1.5.

Various aspects of signaling used to remotely control the switching centers are presented in Section 2. Switching and signaling issues are summarized in Section 3.

1.2. Prototype PABX

1.2.1. Stored Program Control

To define a prototype PABX, we first assume a stored program control (SPC) type of system. Stored program has been the way of doing switching control for both space and time division switching networks for over twenty years. There seems to be no technical, operational, or economic reasons to depart from SPC. It is anticipated that this control program is centrally stored in the smaller size PABX's, handling perhaps up to 1000 lines. However, there may be sufficient reasons of survivability or remote controls to decentralize and to provide backup storage for larger size PABX's.

1.2.2. Time-Division Switching

By assumption, we confine the applicable switching technology further to time-division multiplexing (TDM) or some version thereof. The TDM switch or pulse code modulation (PCM) switch leads to PABX's that are variously known as digital electronic PABX's (EPABX). We exclude pulse amplitude modulation due to transmission problems involved.

It has been nearly twenty years since the first digital switch, the ESSEX, was demonstrated (Vaughan, 1959). In that time, TDM switching has made great inroads in the commercial PABX world, as well as in the switching world in general, that previously had used space-division switching of metallic circuits. There are many practical reasons for the advent of TDM. Time-switching uses the best known technology that, of course, closely tracks the hardware and software trends in digital computers. Impressive developments, especially in memories and LSI techniques, have made high-speed control processes not only technically possible but also economically attractive. Also, TDM has increased the efficient usage ranges of constituent network elements through concentration of messages. Despite its numerous storing, gating, switching, synchronization, and transmission steps, time-division switching maintains the high transmission quality associated with digital processors and PCM voice signals. The system reliability of TDM remains to be proven relative to the older electro-mechanical continuous path switches, but appears quite promising.

The lists of offered user services and service features have grown as the PABX system softwares have matured in the commercial environment. Individual softwares have become more sophisticated and more powerful. These and other TDM advantages are documented in industry publications of both existing and planned PABX systems (Epstein et al., 1972; Bird, 1973; Gueldenpfennig, 1976; Kasson, 1976; Shiff, 1976; Wegner, 1976).

To meet the military requirements for local access-area communications, as well as to interconnect distant bases, the TDM switching approach is assumed. Time divisions permit voice, data, and other services to be offered by a common integrated system, where switching and transmission are jointly designed to place modems and A/D converters at the very far ends of all circuit paths, or nearly so. A review and analysis of the broad technical aspects of TDM switching, from the early ESSEX to the latest data-bus implementations, is given by Inose and Saito (1974).

1.2.3. General Prototype

A prototype PABX with stored program and time-division circuit switching is shown in Figure 3. This illustration depicts a more or less typical PABX with a folded-duplex configuration. A folded network allows for connections between any of the terminals, whereas a non-folded network only allows for connections between opposite sides of the network. The PABX serves anywhere from 50 to 10,000 lines, and perhaps 10 to 1000 trunks. Both lines and trunks can be either analog or digital. The modems associated with the analog circuits are assumed to be part of the line/trunk (interface) facility. They are not considered to be a part of the PABX discussed here. The analog lines and trunks also require A/D and D/A conversion devices and these are considered part of the PABX. The A/D converters are shown in Figure 3 to be part of a network that performs a multitude of tasks: A/D, D/A, remoting, concentration, expansion, and various interface functions. To abbreviate, let us refer to this network as the "concentration network."

1.2.4. Concentration Network

The set of functions performed by a PABX concentration network varies among different manufacturers and PABX models. So

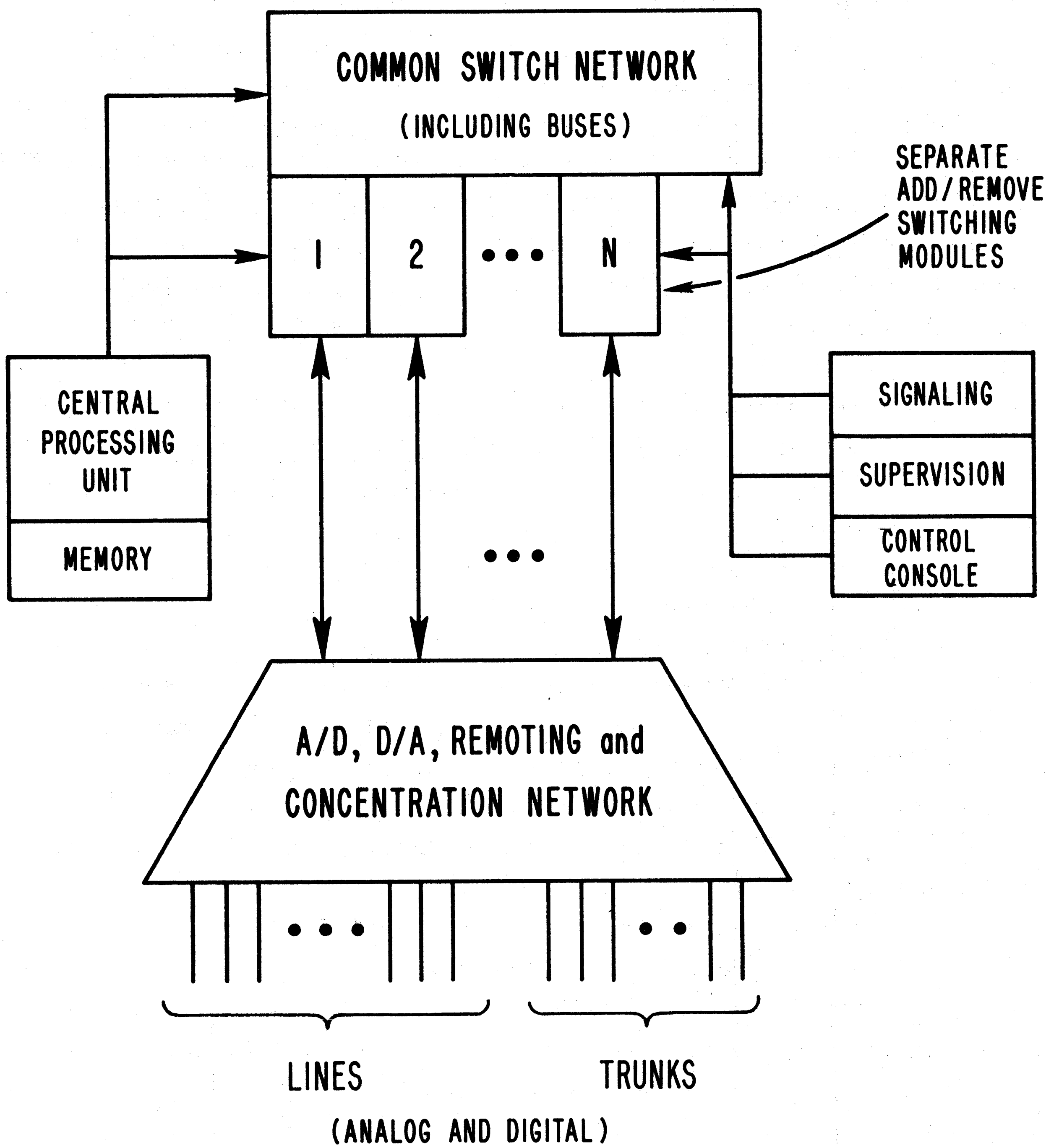

Figure 3. Prototype PABX with stored program control and TDM circuit switching.

does the specific nature of functions, the number of process elements and the order in which the elements are arranged. To explain the issues involved, consider Figure 4. In a simplified way, this figure shows how some of the concentration network functions can be remoted from the main PABX (see also Figure 2). There may be cost advantages in locating certain elements at the central PABX site and in remotely distributing other elements. The A/D conversion can be per line (at individual handsets), per line cluster (perhaps remote) or per group of lines (at the PABX channel bank). The concentration of digital circuits may be more economical and effective in handling traffic if the circuits are distributed over a different number of concentration stages than is customary for analog multiplexing. The preferred numbers of each device class separately, or in some compound arrangement, are far from clear for different size and type service areas.

It is perhaps beyond the limits of this discussion to detail, much less to analyze, the best A/D, remoting, and concentration network alternatives here. Cost, of course, is important. And, because there may be thousands of lines and individual line functions, the concentration network may amount to a noteworthy, if not dominant, cost item in Figure 3.

1.2.5. Grade of Service

To reduce the cost of the total switching network, the concentration network has the role of minimizing the number and cost of individual line (and trunk) facilities. As shown in Figure 4, this is done through advantageous siting (remoting) of concentration modules. In this fashion, for example, 64 voice lines could be PCM-digitized and concentrated (multiplexed) on a single 24-channel T1 line. If this is done, the service suffers somewhat as individual calls may be blocked at the concentrator.

In accepted telecommunications practice, one chooses a grade of service that corresponds to the probability of blocking being P=0.01 per line, given that during the busy hour the lines (user stations) carry a specified average load of traffic. These

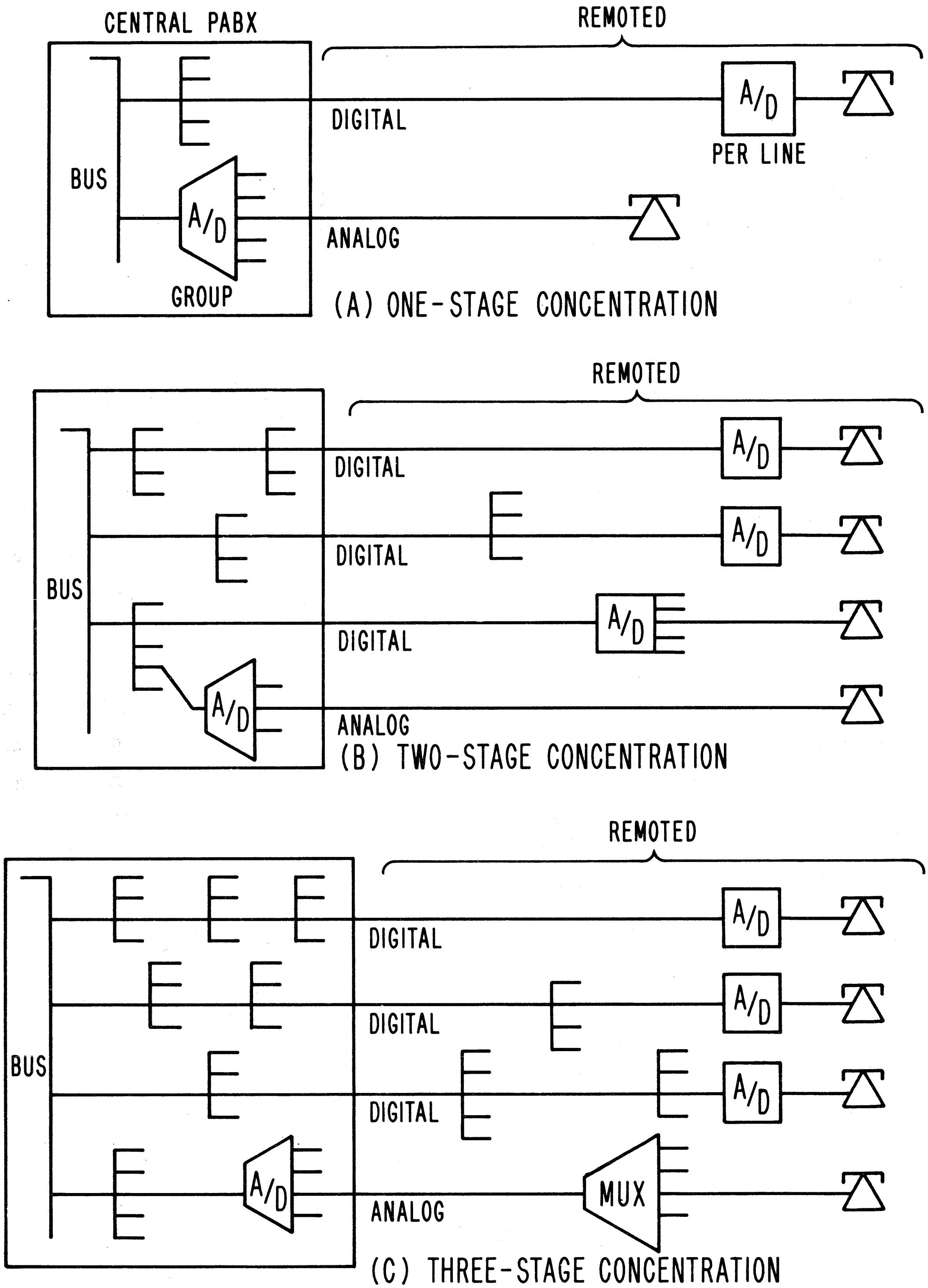

Figure 4. Several central- and remote-location options for the concentration network.

specified hourly loads have been noted to vary from 2.5 to 12 ccs/line[1], for general PABX design objectives.

To have a standard benchmark in this document, we define the following blocking criterion. An average per-line traffic load of 5 ccs/line shall lead a to blocking probability not exceeding P=0.01 for all lines of the PABX. It is desirable to have this criterion apply under realistic user statistics encountered in military base communications during the busy hour. When reliable statistics are not available, recourse to accepted models, such as Poisson or Erlang B, C formulas (or their refinements), may be warranted (AT&T, 1961; Kleinrock, 1975). The relative applicability of such models should, whenever possible, be taken into account.

A possible approach to load-blocking estimation is shown in Figure 5. These curves are based on the Erlang C formula (AT&T, 1961), which depicts an overly conservative caller behavior and leads to system over-design. The abscissa of Figure 5 is L, the number of lines being concentrated. The ordinate, denoted as C, is the number of identical-capacity server channels. Each line carries an average load of 5 ccs. Four curves are shown. A pair of curves, called "one-way blocking," is computed for P=0.002 and P=0.01 and corresponds to a single pass through the blocking network. The second pair, called "two-way blocking," is computed for P=0.001 and P=0.005, per each of two passes, and corresponds to two passes through assumed statistically independent (same L and C) networks. As an example, Figure 5 shows that P=0.01 loss (one-way) results when L=40 lines are concentrated into C=12 channels. For large L asymptotic straight lines are shown in Figure 5. These lines were extrapolated graphically from available tables and charts (AT&T, 1961; Siemens, 1970).

In the PABX grade-of-service definition, as a sound design principle, it is assumed in this report that switching buses

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[1]A ccs is a measure of traffic intensity in hundred-call seconds per hour. Many engineering tables also use the Erlang where 1 Erlang=36 ccs.

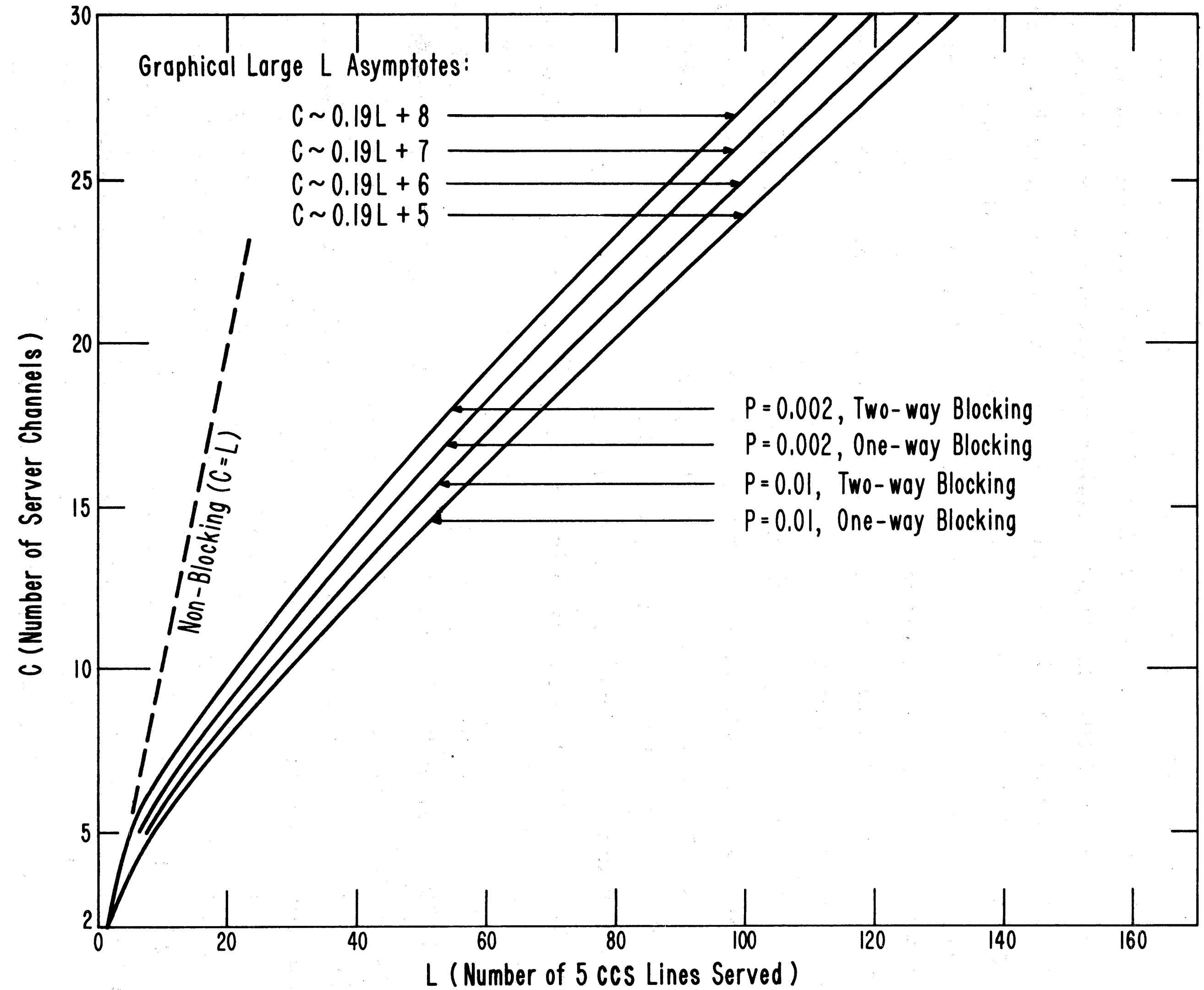

Figure 5. The number of server channels C required for L lines at 5 ccs/line, for one-way and two-way blocking probability P. Based on the Erlang C formula.

suffer insignificant blocking. Likewise, the blocking encountered in the CPU and memory related processes is assumed to be so low as to be negligible when compared with blocking at the concentration stages of the network. The CPU and memory provisions must therefore be selected to be essentially non-blocking under foreseeable call and feature traffic loads, even under emergency conditions. The blocking of trunks, if any, is usually specified at a negligible level relative to line blocking. We assume a trunk blocking of P=0.002.

1.2.6. Switching Network

The actual switching from the incoming line to the appropriate time slot(s), and back to the outgoing line, takes place in the non-blocking main switching network. As shown in Figure 3, it consists of two elements: the common switch network and a set of switching modules. The common switch network may include several high-speed buses and/or highways, whose contents range from message traffic, to control, to signaling. They are accessible to the CPU, the control supervision, and to the switching modules. The nearly identical switching modules can be deployed in varied numbers to match, together with the concentration network, the needs of the service area. It may be advantageous to remote one or more modules from the main PABX site to a distant building.

Both the switching network modules and the common part of the switching network can be variously implemented, but by no means independently of each other. The modules and buses work together, their functions are closely interrelated, and their time-division switching speeds are in the Mb/s range. This cooperative and yet independent action of the modules is called "distributed" switching. Distributed switching can employ besides the previously mentioned data buses, such high-speed elements as time-delay switches (pulse shifters), data and address memory registers, and their assemblies into perhaps a dozen varieties of space-time-space and time-space-time switches (Marcus, 1970; Inose and Saito, 1974; Kobylar, 1974; Pitroda, 1974).

1.2.7. Signaling and Supervision

The functions of signaling and supervision have been used in telephony with various overlapping meanings (Members of Technical Staff, 1971; AT&T, 1975). The acronym BORSCHT (battery, overload protection, ringing, supervision, clock, hybrid, and testing) has been coined to cover everything without being too specific. Here we associate "signaling" with the transfer of electrical information (other than speech) which is specifically concerned with establishing, maintaining and disengaging a connection between subscribers on the network. Thus supervision is a subclass of signaling and is concerned with line and equipment status. Signaling also involves transferring address and routing information, sending tones indicative of call progress, and exchanging information relating to network management and control.

For example, customer lines (loops) are subjected to the following signaling functions:

Seize	-	the calling party is recognized as going off-hook by the PABX. Dial tone is provided.
Dialing	-	The addressing data is accepted and stored by the PABX.
Ringing	-	Ringing signal is supplied to the called line.
Ring return	-	Audible ringing tone is supplied to the calling line.
Busy	-	A busy (off-hook) condition is recognized on the called line. Audible busy signal is returned to the calling line.
Connect	-	PABX establishes and maintains a time/space switched message path between the calling and called lines.
Disconnect	-	Either party is recognized as going on-hook by the PABX.

In conventional two-wire local loops different signaling techniques are employed for different functions. For example, DC current flowing in the loop indicates an off-hook condition; a 20 Hz signal connected to the loop is used for ringing; dial pulsing provides the address code. In the future, digital signaling formats or codes may be used on digital lines to provide a greater variety of functions. Standard self synchronizing signaling codes are recommended (AT&T, 1975).

On digital trunks an expanded signaling format is required. In addition to the supervisory and address information, trunk conditions and routing information is required. Error detection codes and acknowledgment signals are used to increase reliability.

Once detected, the signaling information is usually stored by the PABX and forwarded to the appropriate action elements. Under CPU control audible signaling waveforms can be fetched from a special read only memory (ROM) that provides common storage for busy tones, dial tones, ringing, etc. in digital form. These too may be switched through the network similar to digital voice or data messages.

The usual transmission of digital voice signals using pulse code modulation (PCM) requires a 64 kb/s channel. Several channels may be time multiplexed together for digital transmission. A common multiplexer combines 24 such channels for T1 carrier transmission at 1.544 Mb/s. In one multiplexing format (D1) 8 kb/s are used for synchronization and signaling, leaving 56 kb/s for carrying each voice channel (James and Muench, 1972). Another format (D2) allocates fewer bits per second for synchronizing and signaling and thereby provides better voice quality. In a multiplexed hierarchy one common 64 kb/s channel may be devoted entirely to the signaling for many other channels. When this is done one has common channel signaling (CCIS).

A detailed discussion of various signaling concepts and techniques is given in Section 2 of this report.

1.2.8. CPU and Software

The entire operation of the PABX is under CPU control. A major element of this task is the management of the time-division network. The CPU program consists of several parts that may be stored differently. The basic initialization program may be in a permanent ROM. System programs that require updating once a month, or twice a year can be stored in a pseudo-permanent memory. Other operational routines may be inserted through a read/write RAM, and kept for convenient periods of time. The nature and size of these programs vary considerably among manufacturers, relative software vs. hardware exchanges, PABX size, and the service features offered by the PABX.

The CPU not only assigns time-slot addresses for sending/receiving lines and trunks on the data-bus, it also oversees the detection, generation, and routing of the previously described signaling and supervision signals through the PABX. Depending on the PABX configuration and parallel/serial functions performed, the signaling and supervision tasks may amount to a CPU activity rivaling the time-division switching itself (Kataoka et al., 1974).

Last, but not least, is the PABX software involvement in the service features offered. Recent PABX trends show more and more service features tailored both to the commercial users and their interactions with the common carriers. Several of these business features can be useful to the DoD access area communications (see Sec. 1.2.9, to follow). All of these features are a consequence of improved software, perhaps with specially devised programming languages. The size of the software depends on the service function and feature lists offered, as well as on the built-in freedom to tailor services to individual service area and individual customer requirements.

1.2.9. Features: Basic and Enhanced

Present day PABX's offer a great variety of operational and service variants. It is not uncommon to divide these characteristics into two groups. The first group, called system functions, pertain more to PABX design and operation. The second group, called service features, are traits more visible to the PABX user. While there are elements that seem to straddle such a simplified division, it is also important to recognize that certain features are far more essential than others.

To illustrate the issues involved we present Table 1. This table consists of nearly 50 functions or features, which is about a third or a half of the features listed in a given sales brochure. After a brief description of its nature, each characteristic is assessed as to its access area basic need, hence an argument for incorporation in the prototype PABX.

1.3. PABX Design for Analog-to-Digital Transition

1.3.1. Current and Projected Transmission Facilities

Present military base transmission facilities consist mostly of analog lines and trunks. Ordinary non-secure voice, FAX, and low-speed teletypes modulated onto the voiceband, are all examples of analog traffic. To pass this traffic through a modern PABX with a TDM switching system as described in Section 1.2, A/D and D/A conversion is required. When the signal path involves many switching points, many A/D conversions occur. To pass digital signals on analog lines modems are also needed. All of this, of course, costs a considerable amount of money, degrades the quality of signals on certain links, and is not necessary for future installations.

The projected future all-digital integrated DCS system will augment the time-division switching centers with digital lines and digital trunks. As stated earlier, this will reduce the number of modems and A/D converters by restricting them to the very ends of circuit paths. Unfortunately, the entire communications plant is too large to be economically replacable in one bold stroke. It is necessary to plan for a transition period, where analog and digital lines/trunks can coexist and where expedient, gradual transitions from analog to digital can take place. The purpose of this section is to illustrate how this transition may be facilitated in PABX design.

1.3.2. Example PABX for the Transition Period

Figure 6 shows an example PABX that can work with nearly arbitrary mixes of digital and analog lines and digital and analog trunks. This example PABX is upgraded by a simple re-

Table 1. Selected PABX Functions and Features

#	Name	Description or Comment	System Function Needed	System Function Desirable	Service Feature Basic	Service Feature Enhanced
1	Attendant	At least one, and a single console	X		X	
2	BORSCHT	Battery, overload protection, ringing, signaling, clock, hybrid, testing-requirement	X			
3	Call Forward	All incoming calls to one of several designated stations			X	
4	Call Pickup	Answer calls in a given group only			X	
5	Call Transfer		X		X	
6	Call Waiting	Indicates that another call is waiting			X	
7	Camp-on	Reminds original caller when the other party has gone on-hook. Sets up call if so indicated by original caller			X	
8	Classes of Service	Restrictions on various groups of users, for toll calls, priority, security, etc.			X	
9	Conferencing Few	Conferencing for 3 or 4 parties			X	
10	Conferencing Many	Conferencing for larger groups, 5 or more				X
11	Diagnostics	Automatic, mainly for maintenance	X			
12	DID	Direct Inward Dial			X	
13	Digital Lines	64 kb/s PCM, 16 kb/s CVSD, A/D at sets	X		X	
14	Digital Repeaters	Inward and/or outward	X			
15	Digital Trunks	64 kb/s T1,T2,..., compatibility	X			
16	DOD	Direct Outward Dial			X	

Table 1 (cont.)

#	Name	Description or Comment	System Function Needed	System Function Desirable	Service Feature Basic	Service Feature Enhanced
17	DTMF Conversion	Dialing compatibility	X			
18	Ext-to-Ext	Extension to extension dialing in PABX local area			X	
19	Extra Consoles	Backup consoles, more attendants		X		
20	Flexible Numbering	Sta. No's vs. Frame No.		X		X
21	Foreign Exchange	Translation and control for remoted operation	X			
22	Hold	For additional in-calls without aid			X	
23	Intercept	By attendant	X		X	
24	Line Lockout	If station is off-hook too long		X		
25	Line/Trunk Card Density	Assume 75% packing density on cards	X			
26	Line P=0.01	Assume probability of blocking for lines when 5 ccs/line	X		X	
27	Message Accounting	For billing, statistics		X		X
28	MODEM Compatibility	With a list of standard modems, their interfaces	X			
29	Modularity	For growth: distribution and concentration modules. For operation: line and trunk card interchange	X			
30	Multiple Listing	Main No., residence, extension(s)		X		X
31	Night Answer	When nobody at consoles, call preassigned station(s)		X	X	
32	Override	Ability to break into certain existing calls, with constraints				X
33	Privacy	Optional lockout of break-ins			X	
34	Private Lines	Dedicated lines through PABX to CO		X		X
35	Power Fail Transfer	Automatic routing of certain CO trunks to designated lines	X			

Table 1 (cont.)

#	Name	Description or Comment	System Function		Service Feature	
			Needed	Desirable	Basic	Enhanced
36	Public Address	To broadcast input		X		
37	Secure Voice	A/D and encryption at certain stations. Protected facilities			X	
38	Speed Calling	Abbreviated dialing to frequently called stations				X
39	Standby CPU	Duplicated CPU for reliability, periodically exercised	X			
40	Standby Power	Standby power plant, periodically exercised	X			
41	Stored Program Control	Best CPU control with present technology	X			
42	Tandem	PABX to interconnect other PABX's, CO's		X		
43	TDM	Time-division switching	X			
44	Toll Restriction	On outgoing toll calls for certain station categories, perhaps all			X	
45	Tone Buzz	To alert station when phone is off-hook too long		X		X
46	Traffic Measurement	To gather statistics for upgrading or modification of PABX		X		
47	Trunk P=0.002	Assumed probability of blocking for trunks, when 5 ccs/line and 30% trunk traffic	X		X	
48	Trunk Types	Include the following trunks: 2W, pulse, indial and outdial; 2W, DTMF, indial and outdial; 2W, pulse, tie trunks; 2W, ringdown; 4W, pulse, tie trunks; 4W, AUTOVON; 4W, digital.	X			
49	Voice Recorders	Several recording channels to record urgent messages				X

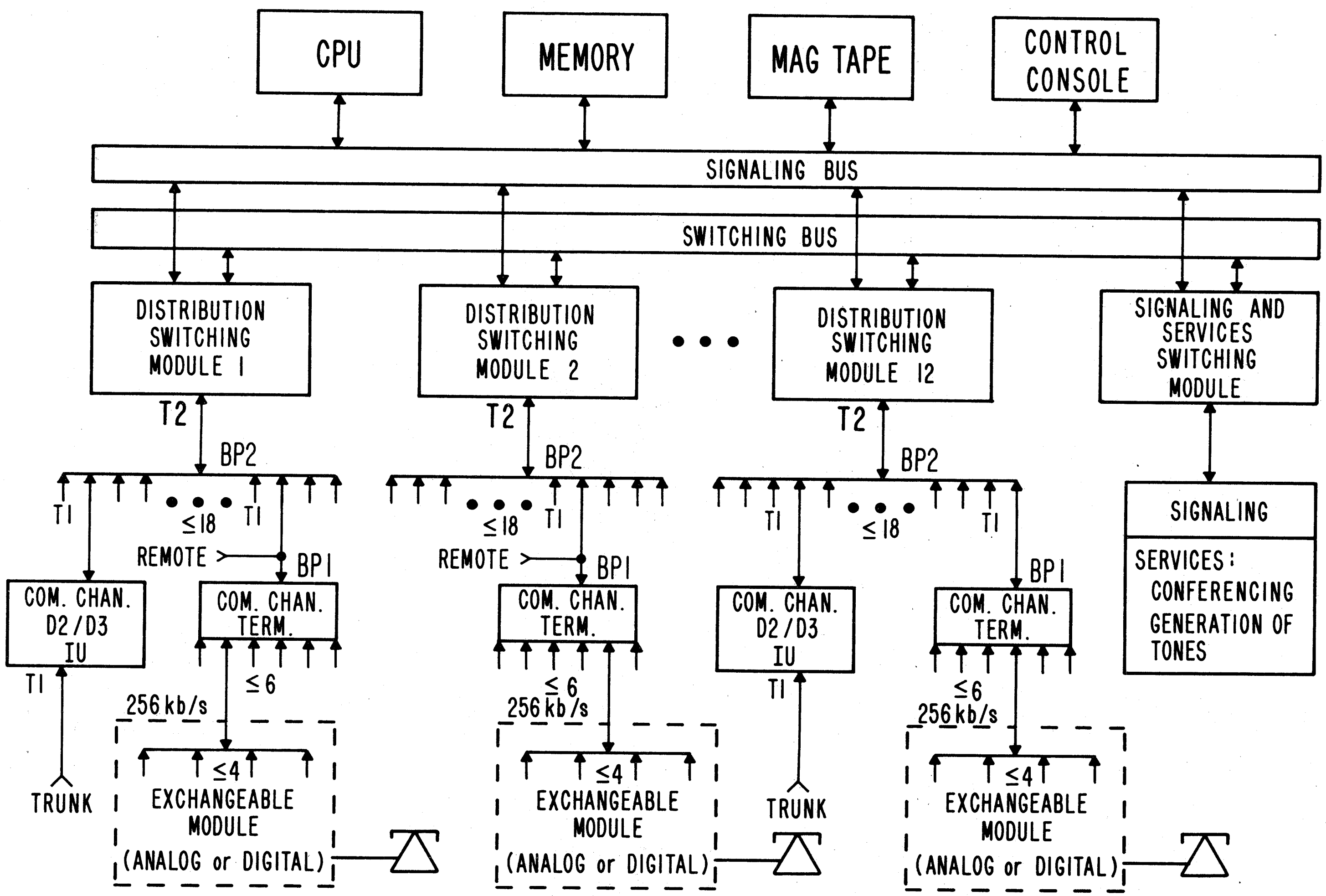

Figure 6. Exchangeable four-line modules to facilitate analog-to-digital transition on a PABX.

placement of fully exchangeable modules. The modules, of which three are shown in the lower part of Figure 6, are assumed to serve up to four analog telephones, or up to four digitized phones, or (perhaps up to four also) digital trunks of compatible data rates.

The PABX shown in Figure 6 is of the same general type described in Figure 3. The lines and trunks enter through a suitable concentration network which is assumed modular. After that, the actual switching takes place in one or two distribution switching modules (also called switching network, or simply network modules) and on a common switching bus. Figure 6 shows 12 switching modules, plus one module for signaling and services.

Each switching module in Figure 6 accommodates up to 4x6x18=432 lines, trunks, or a mixture of the two. The twelve modules shown have a total line capacity of above 5000, which may be called a large PABX system.

Individual T2 feeds to distribution switching modules may be replaced with entire digitized 432 line plus trunk concentration modules. Likewise, the T1 lines can be variously replaced, remoted, and used for trunking. Note that Figure 6 emphasizes the module interchangability at a relatively low level, where no more than four lines are multiplexed.

A more graphic illustration of the module exchange is given in Figure 7. Here the three exchangeable modules from the lower part of Figure 6 are repeated three times. In (A) all the modules serve analog two-wire (2W) and four-wire (4W) telephone lines. In (B) a part analog and part digital mix of telephone lines is shown. Finally, (C) of Figure 7 contains all-digital phone lines and trunks. One must stress that Figure 7 is a mere overview. A detailed specification of the individual line and trunk status is required in PABX concentration network design. Part of such a specification, as shown in Figure 8, involves all 4x6x18=432 line and trunk terminals per each of the 12 switching modules. Of course, uniform concentrator groupings into fours (first level), sixes (second level), and eighteens (third level)

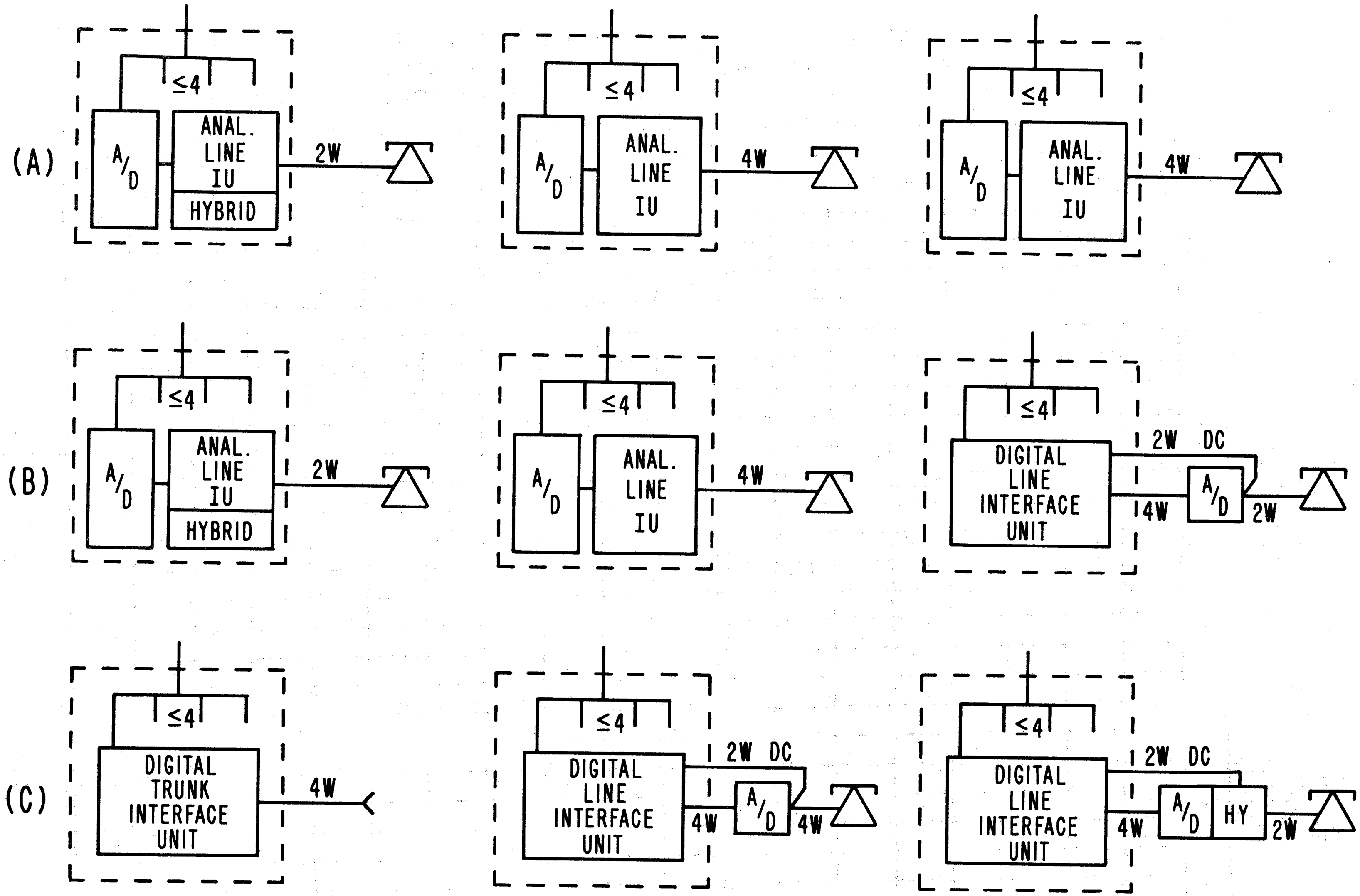

Figure 7. Transition to digital lines with some examples of exchangeable modules: (A) All analog lines; (B) Part analog, part digital lines; and (C) All digital lines.

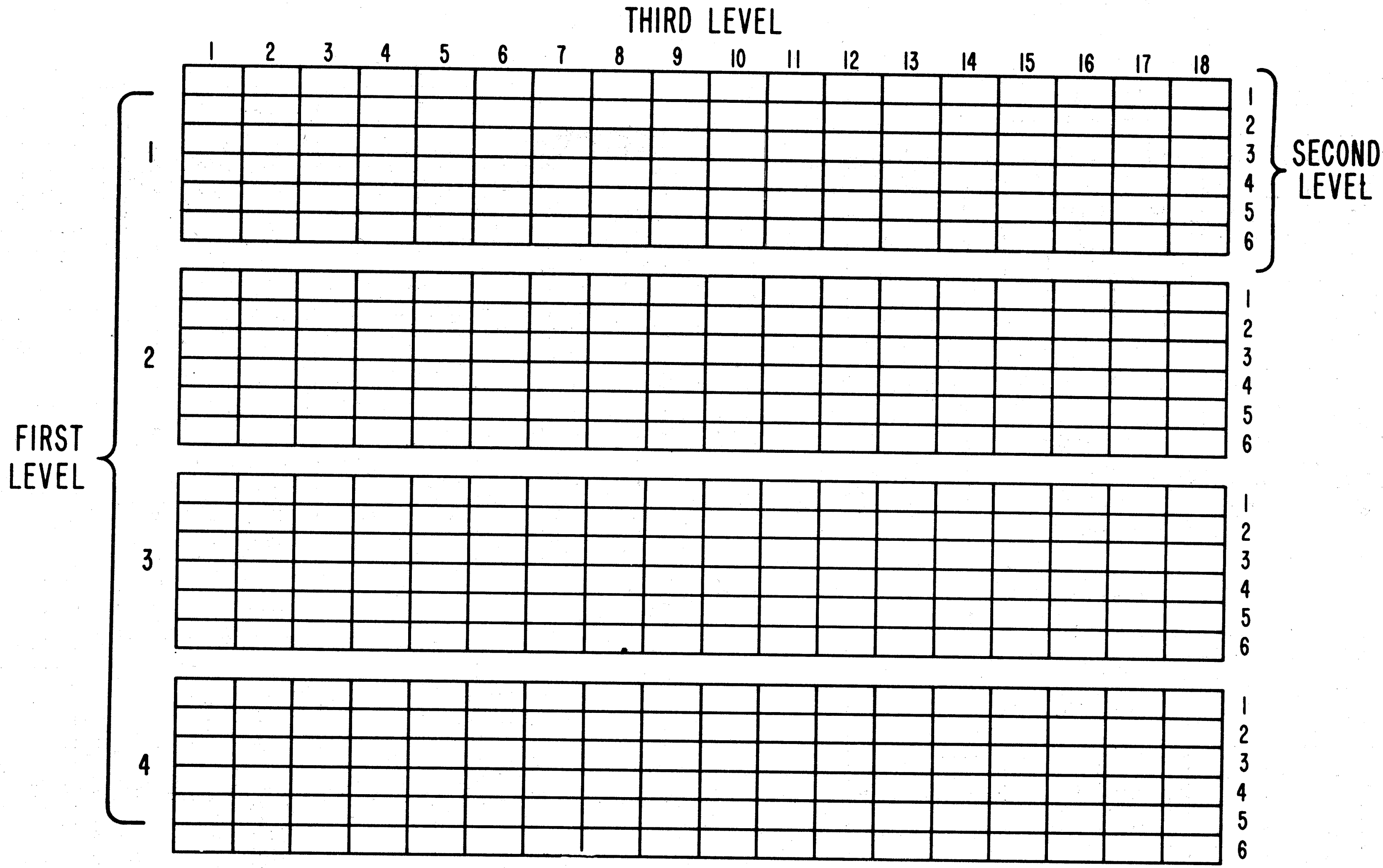

Figure 8. Tabular summary of the three-level concentration with 4x6x18=432 exchangeable ports, where each port can be either analog or digital, line or trunk, or idle.

would expedite the writing of A's for analog, D's for digital, or other designation, into the hundreds of boxes of Figure 8. Note that concentrator ports may be left idle to alleviate, equalize, or otherwise manage traffic to-and-from switching modules.

1.3.3. Design Aspects of the Example PABX

The design of the example PABX (see Fig. 6) contains three levels of concentration. At the first level, immediately after the exchangeable modules, there is no blocking and the total data rate for ordinary PCM can be as high as 256 kb/s. At the second level of concentration, where up to six 256 kb/s streams are concentrated into one T1 line, several things happen. First, the 24 channels of the 1.544 Mb/s T1 line are used as follows:

22 channels carry messages,
1 channel is for common channel signaling,
1 channel is kept as a spare.

Next, since 24 lines are compressed into 22 server channels, some blocking of calls can occur. This is denoted in Figure 6 as Blocking Point 1 (BP1). Figure 5 shows that the blocking probability at BP1 for L=24, C=22, is negligible for 5 ccs/line traffic. From BP1 into the T1/T2 hierarchy the system employs a separate 64 kb/s channel as a common channel for signaling.

At the next higher concentration level, eighteen T1 lines are concentrated into a single T2 line. The 96 channels of the 6.312 Mb/s T2 line are assigned as follows:

92 channels carry messages,
2 channels are common for signaling,
2 channels are spares.

There is blocking at this T1/T2 juncture. It is denoted as Blocking Point 2 (BP2) in Figure 6, and its blocking probability (see Fig. 5) can be estimated as near $P \cong 0.001$ for L=432 and C=92 (using the large L asymptotes for 5 ccs/line traffic density).

1.3.4. Design of Individual Exchangeable Interface Units

Figure 7 indicated that the interface units may come in several forms. Various required card functions are outlined here.

Analog cards may interface with either two-wire (2W) or four-wire (4W) analog lines. Figure 9 depicts a 2W-to-4W hybrid followed by an A/D converter. A standard 64 kb/s PCM is assumed. Finally, four digitized signals are TDM multiplexed on a single 4W outgoing circuit. In the reverse direction, demultiplexing and D/A conversion takes place. The signaling interface and ringing control circuit controls signaling and ringing. It inserts signaling bits into the TDM stream to identify on-hook and off-hook conditions. While Figure 9 shows an upper bound of 2.4 kb/s signaling rate, a lesser rate may suffice for most installations. For dial phones, addressing and hook-switch signaling is done by altering the loop current. For DTMF phones, hook-switch signaling is likewise accomplished by changes in loop current, but addressing tones are sent over the VF circuits to the DTMF receiver. Ring return is accomplished by transmitting tones stored in a read-only memory (ROM) in digital format.

A slightly modified interface unit may be needed for analog VF trunks. As shown in Figure 10, the E&M signaling warrants a special DC loop-to-E&M converter to be added to this interface card. Otherwise, trunks with loop signaling can use the same line card shown in Figure 10. E and M signaling is usually converted to single frequency (SF) for better transmission over long distances. This E&M to SF converter is not shown in Figure 10, as it is frequently associated with the trunk transmission plant. For E&M signaling, the DC state of the M lead at the originating station determines the state of the E lead at the destination, and vice versa (AT&T, 1975) (also see Sec. 2).

The digital line interface unit faces the problem of how to provide DC and loop signaling to the telephone set. The A/D is assumed to be either inside or in the vicinity of the telephone itself. To obtain loop signaling it appears necessary to run a 2W signaling line from the interface card, past the A/D, to the handset. If the telephone set is of older vintage with a built-in or attached hybrid, then the signaling wires can be brought in on the 2W side of the hybrid. This is illustrated in Figure 11.

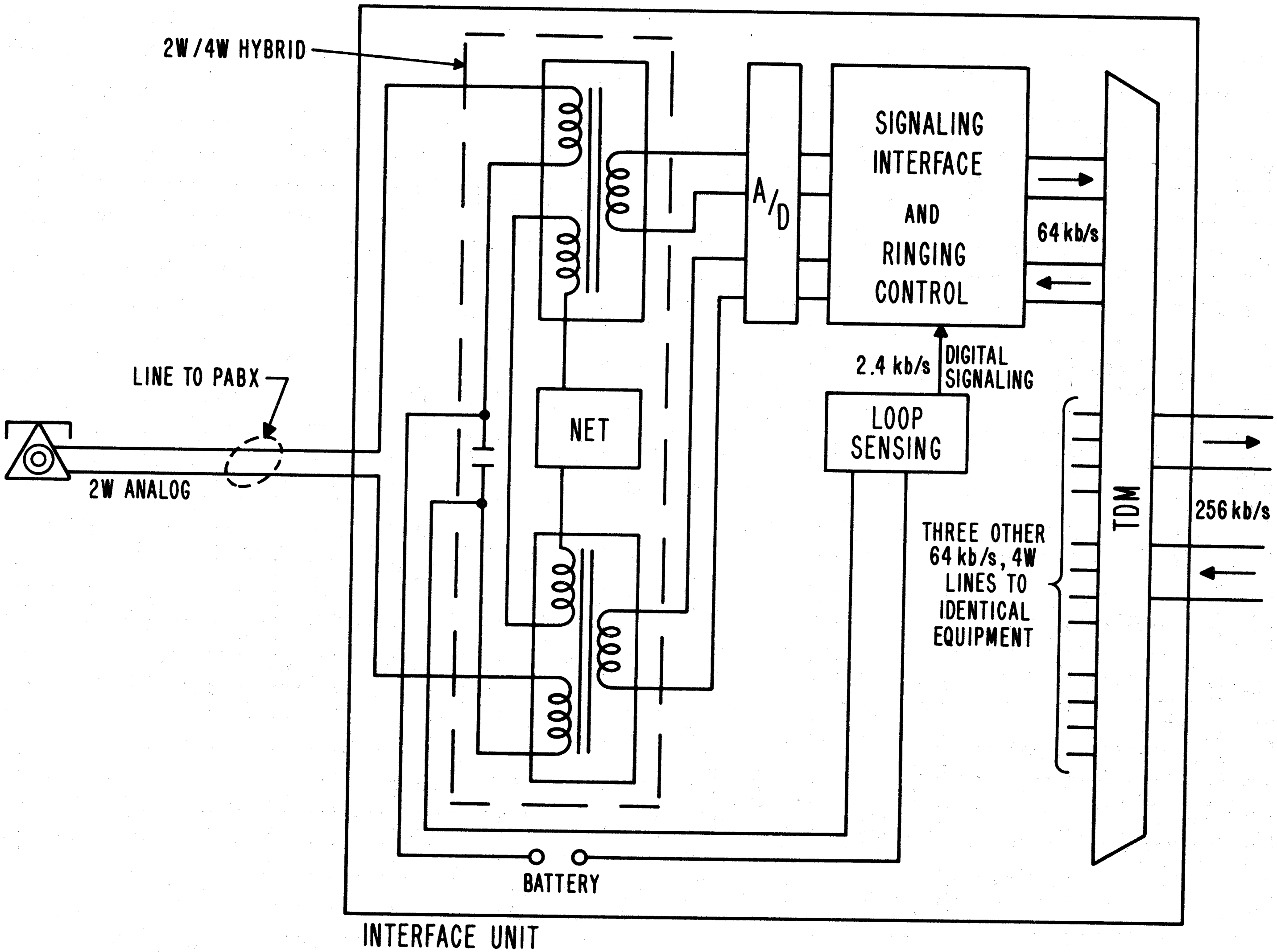

Figure 9. Analog line interface unit.

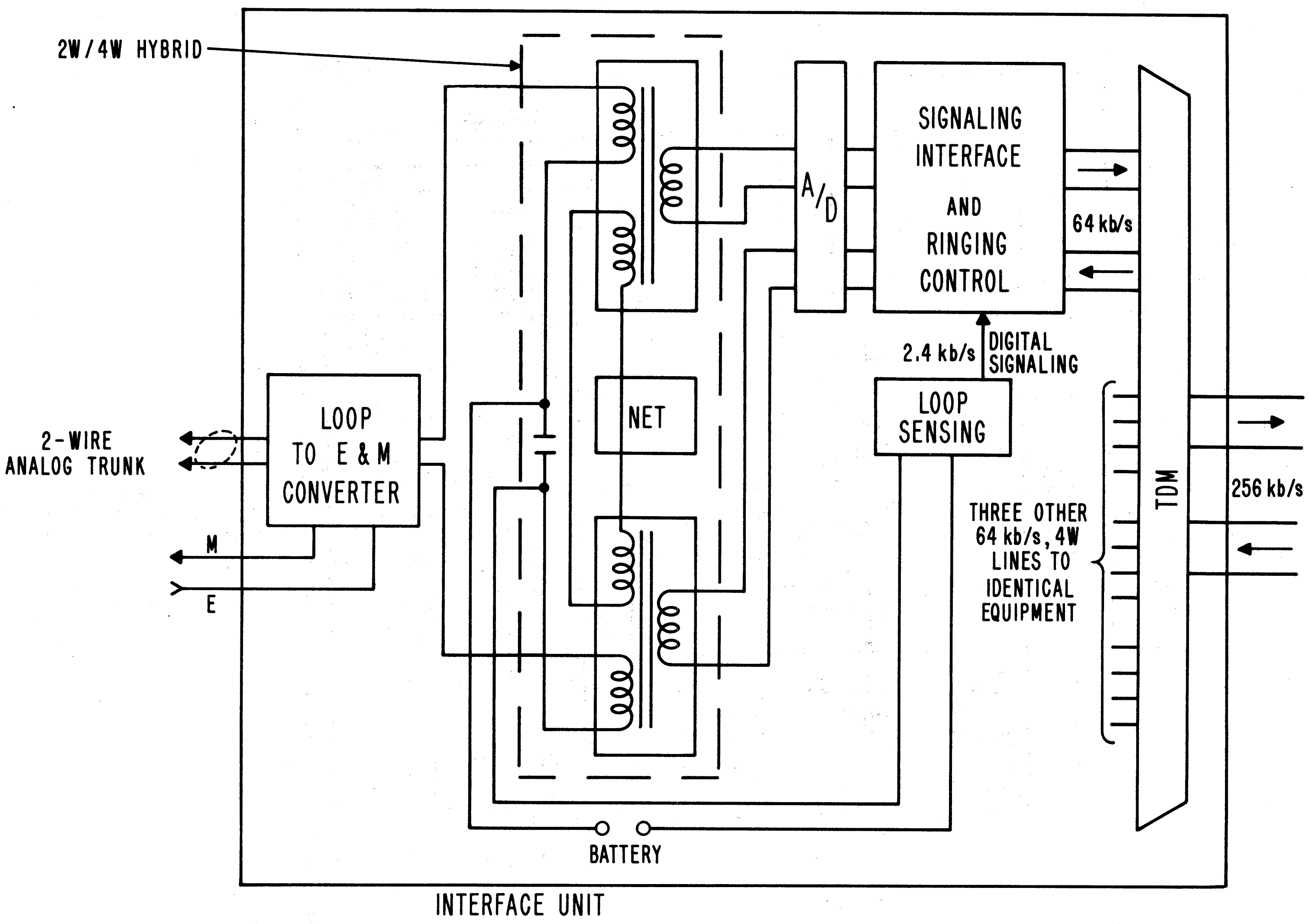

Figure 10. Interface unit for a two-wire analog trunk with E&M signaling.

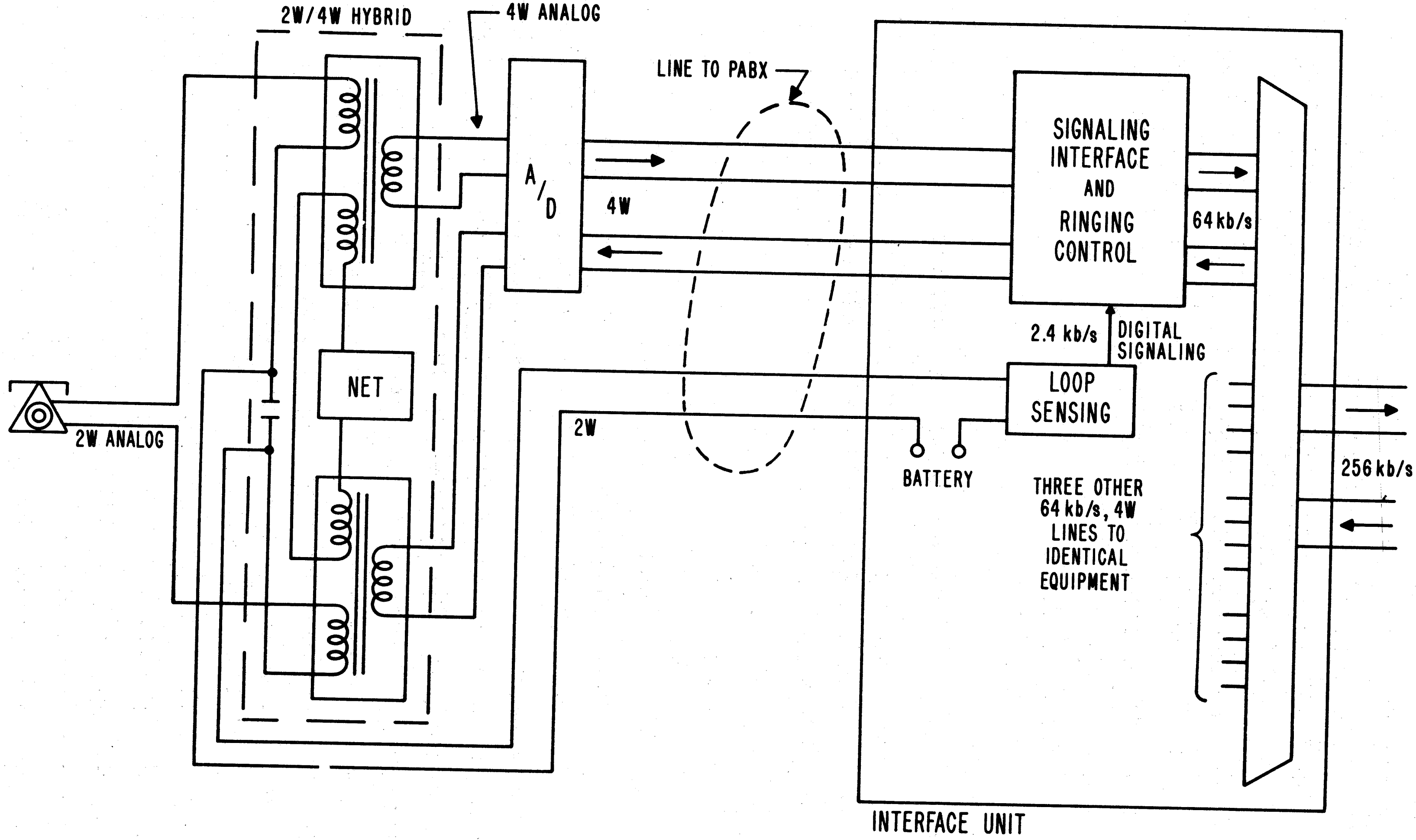

Figure 11. Digital line interface unit for an existing hybrid.

If on the other hand, the telephone set has no hybrid -- as may be the case in the future -- then the extra insertion of a hybrid appears unjustified and unsound with regard to design and cost. Some transducer circuit, whether capacitivelly or otherwise coupled, should be able to bring the DC loop past the A/D and into a 4W handset. Although we are not cognizant of such a coupler on the market, we indicate its 4W-vs.-6W funtional role in Figure 12.

The digital trunk interface requires little comment. It is essentially the same interface card type used throughout the PABX to route the standard 64 Kb/s PCM circuits.

Some additional line and trunk interface terminals are described in Section 2. With a properly designed telephone set it appears feasible to interface other analog or digital transmission lines with a four-wire termination.

1.4. The 10% Digital Telephone System

1.4.1. Base Service Requirements

In this section, an illustrative PABX system is defined. Initially, the PABX serves the communications needs of two conceptual military base sizes. The first, a relatively large base, is defined to contain 5000 lines (user stations). The second is a smaller base of 1000 lines. In both cases we assume that 10% of all telephone sets are digital, with A/D and D/A conversion either incorporated in the handset or, equivalently, provided as a station attachment in close proximity to the handset.

The data rates of these digitized phone lines are assumed to be 64 kb/s, as in existing standard PCM (Hartley et al., 1967; Boxall, 1969). To accommodate future lower rate options, such as 16, 8, or 4 kb/s digitized voice, compatible network modifications will be briefly indicated.

The services to be provided, and the number of terminals each, are specified in Table 2. These counts of different types of services are scaled scenarios from the earlier LDDS study

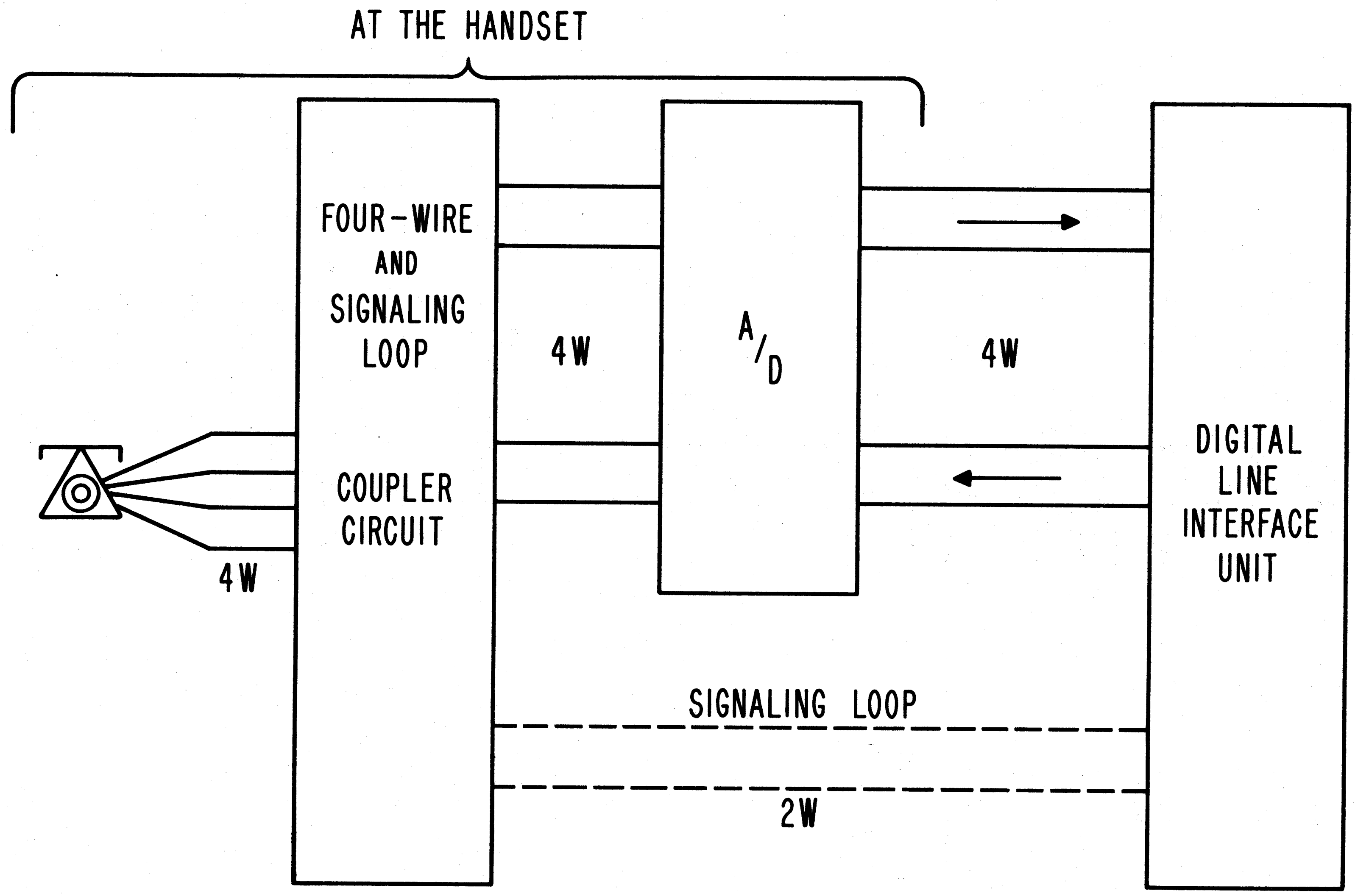

Figure 12. The four-wire and signaling loop coupler circuit configuration.

Table 2. Number of Service Terminals for Two Selected Base Sizes

Service	Terminals for 5000 Line Base	Terminals for 1000 Line Base
Telephones:		
Analog (4kHz)	4410	882
Digital (64 kb/s)	490	98
Computer Terminals:		
Low Speed (1.2 kb/s)	25	5
Medium Speed (2.4 kb/s)	10	2
High Speed (48 kb/s)	5	1
Teletypes:		
Low Speed (300 b/s)	50	10
High Speed (4.8 kb/s)	10	2
Total	5000	1000

(Nesenbergs and Linfield, 1976). Note that the number of digitized telephones is 9.8% of the total number of base terminals.

We assume that only the high-speed (i.e., the 48 kb/s) computer terminals employ digital lines. All the other lower speed computer terminals, as well as teletypes, are carried on analog lines via appropriate modems. The PABX views such lines as serving any other voice frequency station, and does not demodulate to the baseband for switching.

The future goal of an integrated digital DCS calls for the eventual realization of all digital trunks. The necessary transition stages were described in Section 1.3. The actual transition period must progress through a mix of both analog and digital trunks. The relative percentage of analog vs. digital trunk facilities at the base level may thus vary from 0% to 100%, depending on age and type of existing base facilities, as well as on the progress of the digital deployment. In the present PABX example, we assume that all of the trunks are digital.

1.4.2. Outline for the Example PABX

The overall design of the 10% digital phone, both 5000 and 1000 line, PABX example starts with the general prototype of Section 1.2.3 (see Fig. 3). In consecutive order, the concentration network (see Secs. 1.2.4 and 1.2.5), the main switching network (see Sec. 1.2.6), signaling and supervision (Sec. 1.2.7), the CPU control (Sec. 1.2.8), and other elements will be outlined. Design issues and assumptions made will be described as they arise for the PABX.

1.4.3. Outline of the Concentration Network

The concentration network interconnects the user lines and interoffice trunks (to CO's, PABX's, etc.) with the switching modules of the PABX. For system-wide ease of interconnection and remoting, the concentration network is assumed to be compatible with the so called U.S. telecommunications network of the Bell System (James and Muench, 1972) and its DCS worldwide counterpart. This includes the established analog hierarchy and, more significantly, the T1, T2, etc., digital hierarchy at the rates above one Mb/s. Standardization of rates is assumed mandatory at or near the higher rate switching module levels in the concentration network.

The issues involved are illustrated in Figure 13. This figure shows a fraction of the concentration network that corresponds to a single switching module. For brevity, we call the network of Figure 13 a "concentration network module." This particular concentration network module has three "blocking points," denoted as BP1, BP2, and BP3. These blocking points are arranged to satisfy our first assumed criterion of concentration (compare with Sec. 1.2.5):

I. The $P \leq 0.01$ grade of service for 5 ccs/line is to be met by all lines on all one-way and two-way blocking paths, even if every concentrator/multiplexer carries its maximum permitted loading.

A second more stringent criterion is assumed for trunks:

II. For trunks the blocking probability shall not exceed $P=0.002$ under the 5 ccs/line load, and subject to 30% of all traffic passing either once or twice through trunk concentration networks.

Consider blocking point BP1. It terminates the analog three-stage concentration string on the right side of Figure 13. By permitting no more than 10 of the 12 channels per basic channel group, no more than 4 groups per supergroup (standard definition permits 5 groups), and finally up to 2 supergroups per digitizer/multiplexer to a T1 line, the blocking probability at BP1 can be assessed. In the notation of Figure 13, T1[22+1+1] means that there are 22 PCM message channels, 1 common channel for signaling, and 1 spare in the T1 "D" bank. Therefore, in Figure 5, blocking point BP1 falls at $L=80$, $C=22$, which is adequate for the $P=0.01$ line loss objective (criterion I).

Blocking point BP2 occurs after two stages of digital concentration (left side of Fig. 13). It provides 22 server chan-

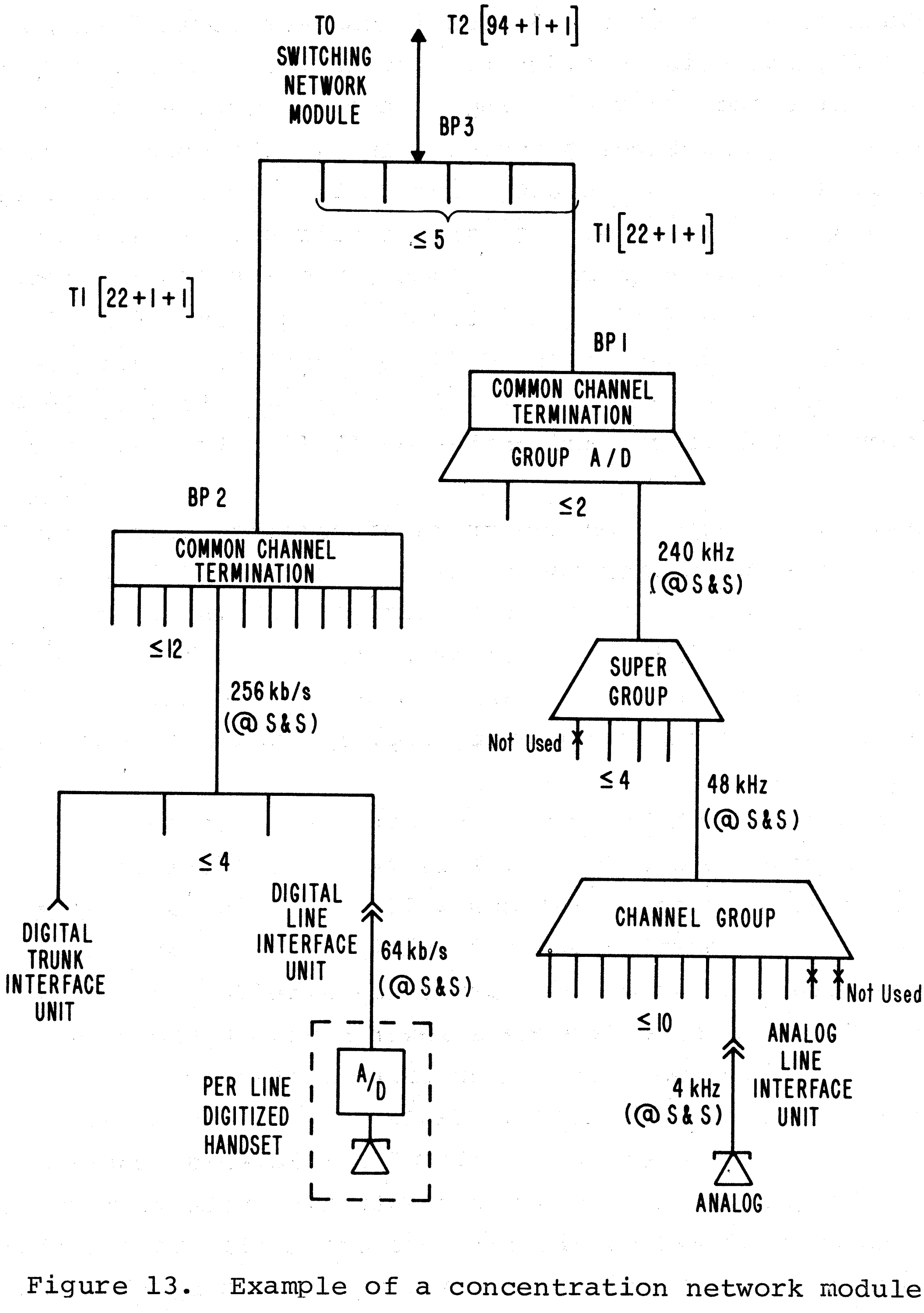

Figure 13. Example of a concentration network module.

nels on a similar T1[22+1+1] line, for no more than 48 digital telephone sets. In Figure 5, BP2 is characterized by a point L=48, C=22, which is considerably better than the P=0.01 and P=0.002, line and trunk blocking objectives (criteria I and II).

The third and final blocking point in the concentration module is BP3. It concentrates one T1 line from the digital sets and up to 5 T1 lines from analog sets unto a single T2 [94+1+1] line. As will be seen later, when trunks are introduced, this 1:5 rule need not be strictly enforced. In Figure 5, BP3 corresponds to a point L=448, C=94 (off scale). Using the large L asymptote, one finds that C=94 is sufficient to satisfy both criterion I (for lines) and criterion II (for trunks), i.e., $P<0.002$.

As a side comment on the grade of service, recall the Erlang C formula that underlies the plots of Figure 5. If this formula leads to over-conservative system designs, as is claimed, then a closer traffic analysis just might reveal whether the disabling of two inputs on the basic channel bank and one input to the supergroup (lower right corner of Fig. 13) was actually necessary.

The maximum number of user terminals available per concentration network module (Fig. 13) is 448. Realistically, it is unlikely that many installations will result in full (100%) loading of all concentrators and multiplexers. For growth, cost, and other reasons, some input cards may expediently be left vacant. To proceed, we assume a lower bound on the card occupancy figure and denote it as our third criterion:

III. All user terminals of Table 2 are to be accessed even if the concentrator/multiplexer low-side packing density is as low as 75% per concentration network module.

Since 75% of 448 equals 336, criterion III implies that no more than 336 user stations can be accommodated by the module shown in Figure 13. To serve the 5000 line service area mentioned in Section 1.4.1, at least 15 such modules are required. The lines of a 1000 line base can be accommodated on 3 concentration modules. Trunks must be added to other CO's, PABX's, and so forth. From criteria II and III one concludes that around 336(0.30)/0.75=135 trunk ports should be added for each module. This means that a 5000 line installation requires as many as 2025 trunks (equivalent to 5 extra modules), and a 1000 line PABX requires 405 trunks (one module). More than 6 T1 lines per T2 trunk must be used in Figure 13 to do this.

The trunks need not be segregated into special digital-trunks-only modules. The trunks, like ordinary 64 kb/s digitized phones, can be distributed anywhere in the digital part (left side of Fig. 13) of the concentration network. Line and trunk exchange is executed by interchange of line and trunk circuit cards. As noted earlier, the blocking probabilities at points BP1 and BP2 are sufficiently low to justify this (criterion II).

The entire switching networks for 5000 and 1000 line installations are summarized in Figure 14. The concentration modules shown contain mixes of 90% analog lines, 10% digital lines, plus an extra 30% digital trunks. The individual modules, and thence the entire network of Figure 14, have been sized to meet the grade of service criteria (I, II), as well as the card occupancy criterion (III). By necessity now, the ratio of "digital to analog" traffic T1 lines has been altered from the previously cited 1:5 to about 3:5.

So far, Figures 13 and 14 have not explicitly shown the disposition of computer terminals and teletypes (Table 2). Only the highest rate computer terminal requires a digital channel plus, of course, appropriate line interfaces and modems for distant operation. These 48 kb/s computer terminals enter on the left side of Figure 13. The remaining lower speed computer terminals and teletypes have a choice of analog or digital paths in Figure 13. Because of their small relative number (2%), these data terminals do not materially alter the number of constituent elements, trunks, modules, or their grade of service estimates.

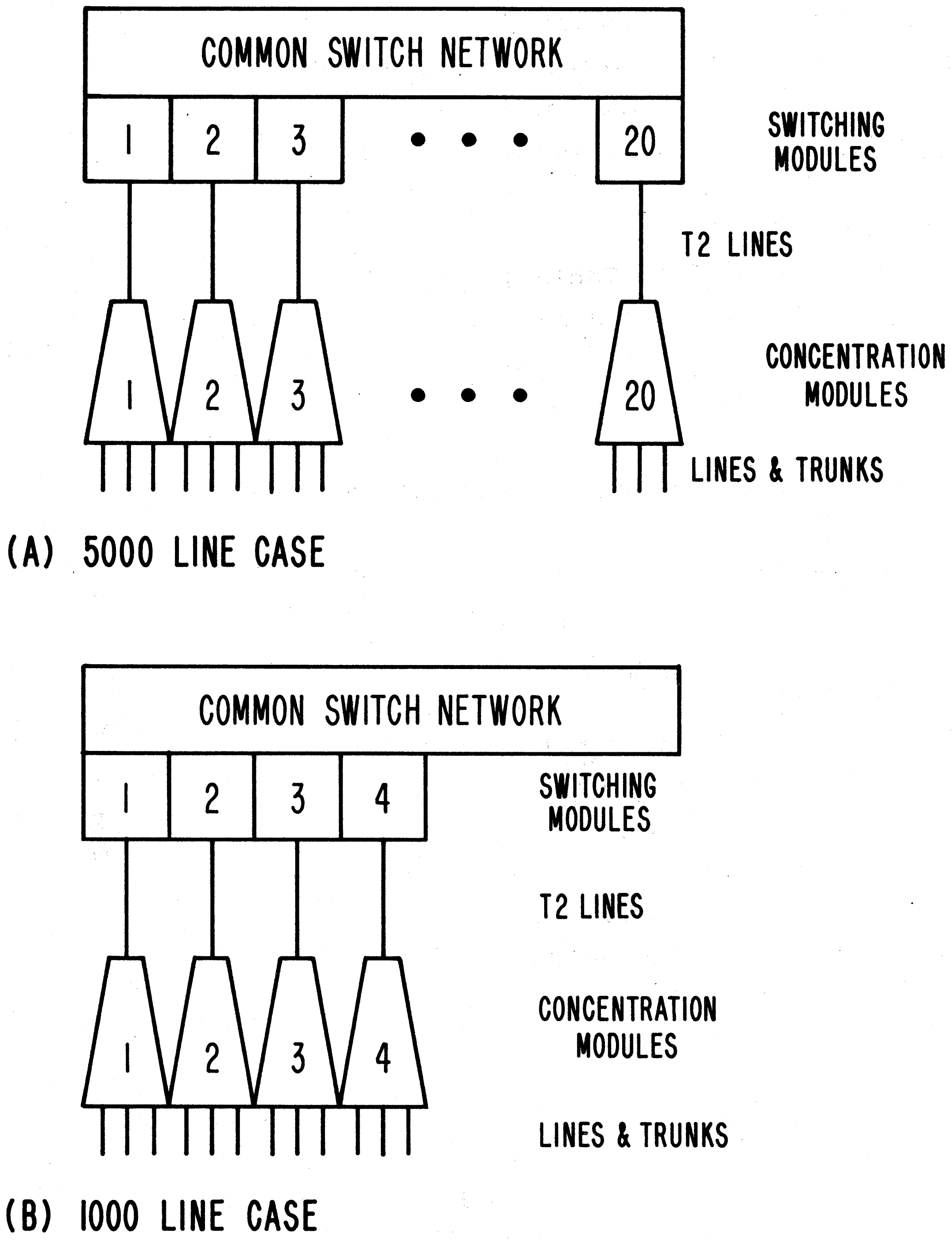

Figure 14. Switching network examples for 10% digital phone system with mixed line and trunk concentration modules.

The utility of this material is summarized in Tables 3, 4, and 5. First, Table 3 illustrates the number of lines of different service types as envisioned for different size service areas. One may refer to an area with 50-300 lines as small; to an area with 2000-10,000 as large; and anything in between as medium. Next, Table 4 shows that the number of ports provided for all services has been increased by about 33% over the actual lines served. Finally, Table 5 combines the implementation of Tables 3 and 4 with the concentration network structure of Figure 13, and concludes with estimates of network element numbers. One notes that most numbers in Table 5 grow more-or-less linearly with the number of lines/ports provided. An exception occurs for smaller area sizes, where for instance a single switching module appears adequate for all areas with less than 200 lines.

A look into the future of digitized voice suggests possibilities of 16 kb/s CVSD, or perhaps other options of data compression to 8, 4, etc. kb/s. Such services can be offered through our PABX example by replacing the 64 kb/s PCM line cards (Fig. 13) with an appropriate lower rate card. This possibility is illustrated in Figure 15, which shows two of many 16 kb/s and 4 kb/s variants.

1.4.4. Common Switch Network and Modules

The overall switching network is outlined in Figure 14. In addition to the concentration modules, the figure shows the common switch network and a variable number of switching (distribution) modules. These switching modules are assumed to utilize distributed TDM switching as discussed earlier in Section 1.2.6. To be more specific, Figure 16 illustrates two typical concepts, the pure data-bus and the data-bus memory switching hybrid. For our PABX discussion we assume the pure bus system.

The typical PCM bit frame is the same as one-eighth of the PAM frame. The PCM frame lasts 1/64,000 of a second, or 15.625 μs. In each frame the active duplex lines and trunks each write and read a single bit on the bus. The addition of a few simplex

Table 3. 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

Table 4. Number of Port Assignments Provided for Different Service Areas

Services	Area Size in Number of Line Ports Provided									
	70	138	272	400	800	1,335	2,670	5,340	6,680	13,350
Analog - Phones	60	120	235	350	700	1,170	2,340	4,680	5,870	11,740
Analog - Slow Data	2	3	5	8	16	25	50	100	120	240
Digital - Phones	6	12	26	35	72	120	240	480	590	1,174
Digital - Slow Data	2	3	4	4	8	12	24	48	60	120
Digital - Fast Data	-	-	2	3	4	8	16	32	40	76
Digital - Trunks	20	40	80	120	240	400	800	1,600	2,000	4,000

Table 5. Numbers of Concentration Network Elements for Different Size Areas

Network Elements	Area Size in Number of Lines/Ports									
	50/ 70	100/ 138	200/ 272	300/ 400	600/ 800	1000/ 1335	2000/ 2670	4000/ 5340	5000/ 6680	10000/ 13350
Analog Line Interface Units	62	123	240	358	716	1195	2390	4780	5990	11980
Channel Groups (10 ports @)	7	13	24	36	72	120	239	478	599	1198
Super Groups (4 ports @)	2	4	6	9	18	30	60	120	150	300
Groups A/D & Tl (2 @)	1	2	3	5	9	15	30	60	75	150
Digital Line Interface Units	6	12	28	38	76	128	256	512	630	1250
Slow Data IU's & MUX (4 ports @)	1	1	1	1	2	3	6	12	15	30
Digital Trunk IU	20	40	80	120	240	400	800	1600	2000	4000
Data MUX (4 ports @)	7	13	28	40	80	133	266	531	662	1320
Data Conc. Tl (12 ports @)	1	2	3	4	7	12	23	45	56	110
T2 Modules (6 Tl @)	1	1	1	2	3	5	9	18	22	44

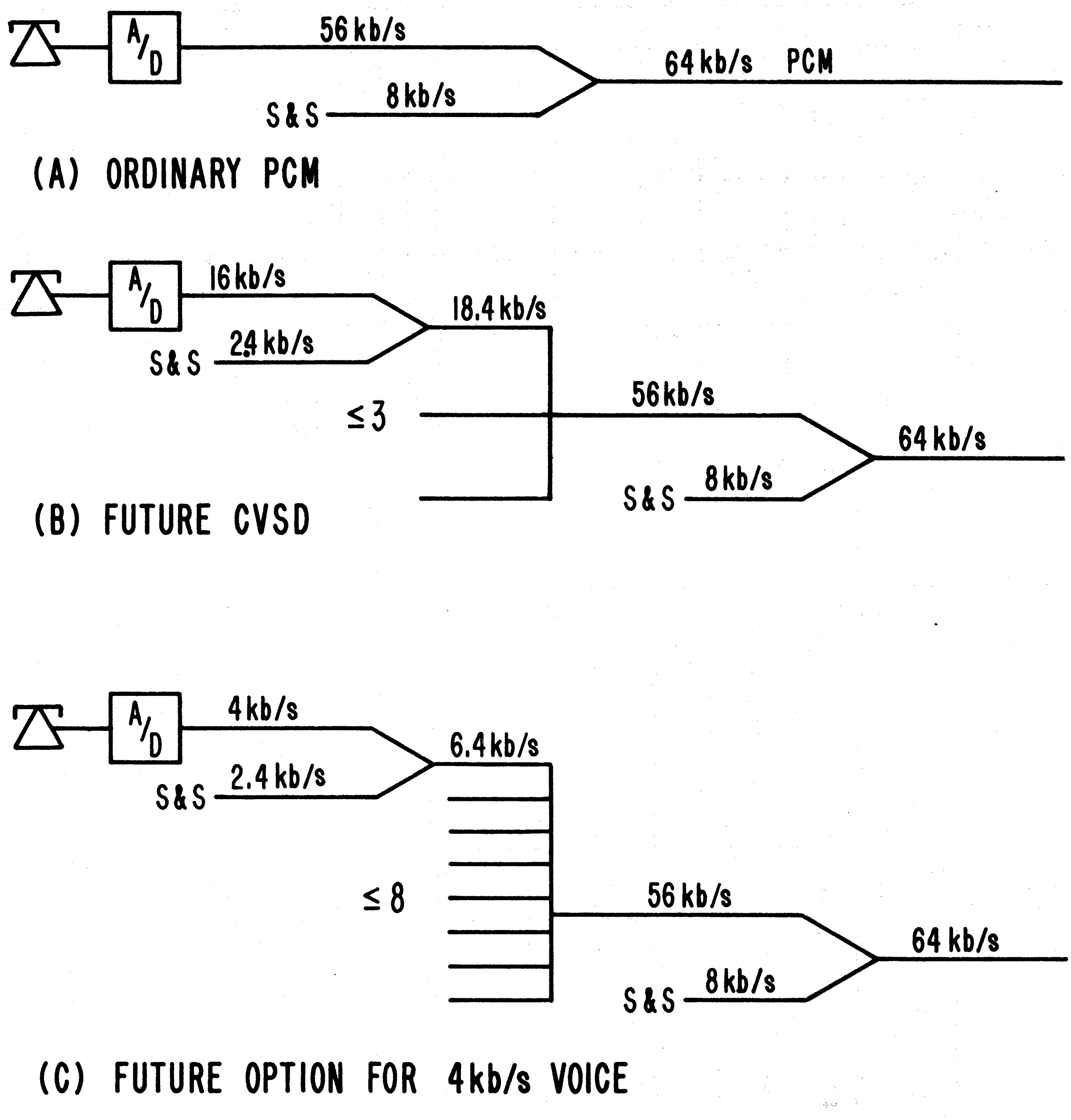

Figure 15. Accommodation of future 16 kb/s and 4 kb/s voice data rates into ordinary PCM.

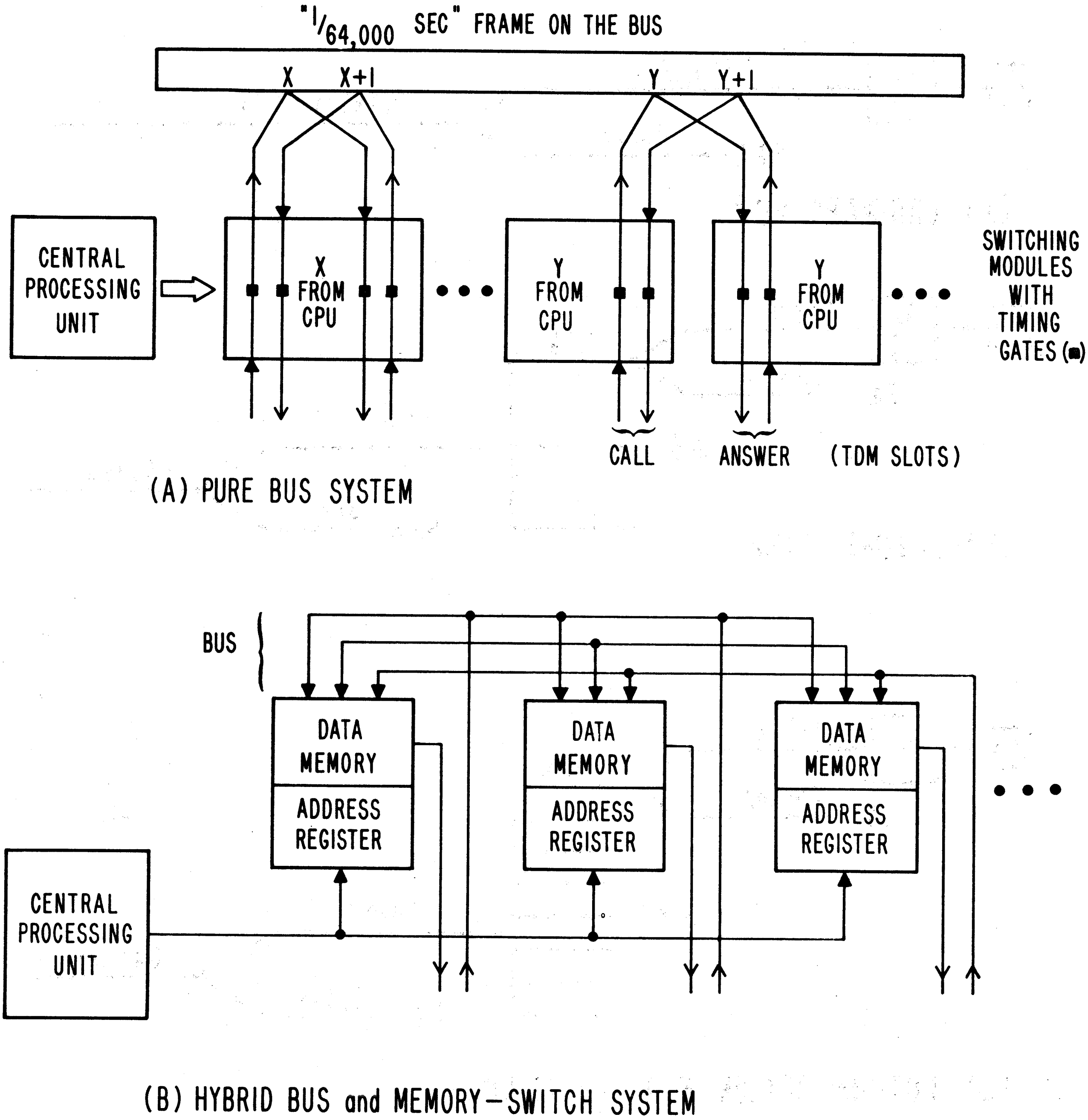

Figure 16. Two typical PABX time division switching concepts.

terminals or conference calls appears insignificant in the total data-rate estimation. One concludes that the message data-bus may have to carry around 6.312 Mb/s for each T2 line used in the PABX. Part of the total flow deals with signaling and supervision (see Sec. 2). This part involves a considerably lower number of bits, perhaps of the order of 4 kb/s. For reasons of simplicity, these overhead bits may be assigned to a separate bus, or buses.

The 5000 line system (see Fig. 14, (A)) must accordingly amass a total of around 126 Mb/s on its message data-bus to be non-blocking in the common switch network. The 1000 line system (Fig. 14 (B)) requires a total of about 24 Mb/s on its bus. Both of these can be implemented with parallel conductors. If one assumes that a single high-speed circuit with between 4 and 20 module junctions can carry 25 Mb/s rates, then the 5000 line PABX requires a bus of 5 parallel conductors, while the more typical 1000 line system can manage with a single conductor.

As shown in Figure 16 (A), the incoming and outgoing message bits for each two-way conversation occupy well defined time slots, x,y,..., in each frame on the message data-bus. Under master clock synchronization and CPU instruction, individual modules are instructed when to gate the bus (i.e., when to write and read the x,y,...), and to which of their constituent 336 lines the x,y,... bits should be routed on the serial TDM streams. The presence of a central clock, perhaps derived by mutual synchronization techniques from DCS or Bell toll network, lessens the scope of time-delay slot switching. But, some time-delay switching may still be needed.

While the details of bus read-and-write arrangements can vary considerably, their essential feature is the nonblocking distribution of one parallel set of T2 streams into another. A further outline is given in Figure 17. Perhaps the most significant issue here is the function of input and output buffers. Depending on synchronization formats imposed, these

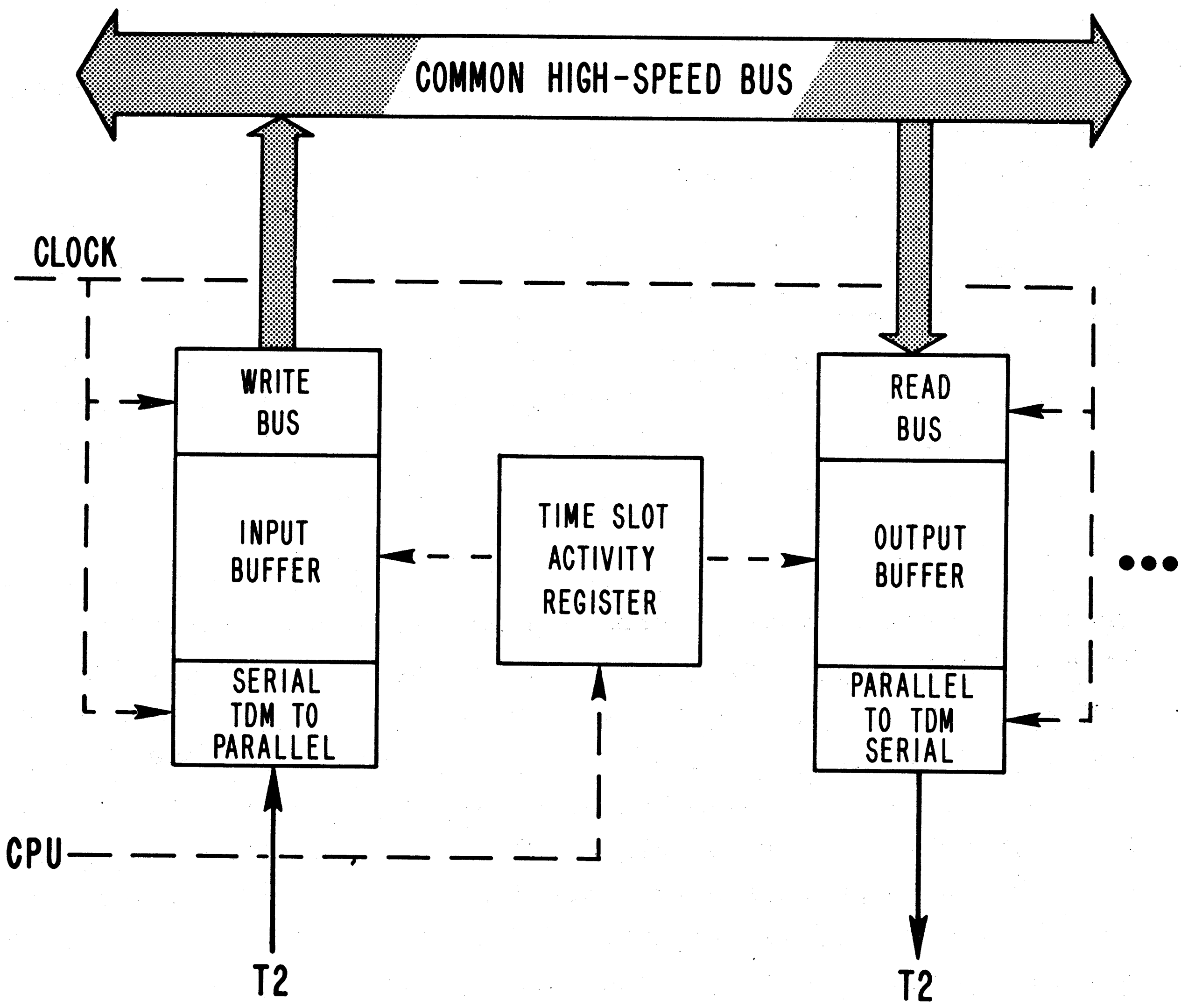

Figure 17. An outline of a single distribution module on a common high-speed bus.

buffers can consist of several (2,3,4,...) stages of space/time switches, each having time-slot interchange junctors either without delay, with fixed delay, or with variable delay.

1.4.5. Signaling and Supervision Outline

The PABX must incorporate all required signaling and supervision functions mentioned in Section 1.2.7. These loop functions are matters of standard practice in existing PABX installations and their details are discussed in Section 2 of this report. There are, however, two noteworthy issues to be resolved.

The first issue concerns the digital lines to digitized telephone sets. Even with 4-wire installation, some aspects of BORSCHT (see Sec. 1.2.7), such as economics of dc power, may have to be resolved. The addition of extra wire pairs is one solution, but this adds to the cost of the loop plant. Other alternatives may involve a redesigned customer loop plant, again at considerable cost. Some of the possible alternatives are given in Section 2.

The second issue is the common channel for signaling and how close to the user stations, in particular the digitized ones, this common channel should go. In Figure 13 one solution is indicated. It consists of a simple rule. The common signaling channel is initiated and terminated at the same channel banks where T1 lines are assembled and disassembled.

Below T1, signaling and supervision (S&S) occupies part of the 64 kb/s PCM channel. On the analog side (see Fig. 13), standard channel group S&S arrangements can be employed.

1.4.6. CPU and Software Needs

The central processing unit (CPU) supplies the high-speed processing and control necessary to operate the PABX. This includes all service functions and features for the local 50 to 10,000 lines, and their interaction with other PABX's, CO's, etc. It is assumed that the CPU deals through a special common control bus that, on the systems side, accesses a read/write RAM, tape storage and console TTY; and on the operations side reaches into all switching network modules. The topology of this control is depicted in Figure 18.

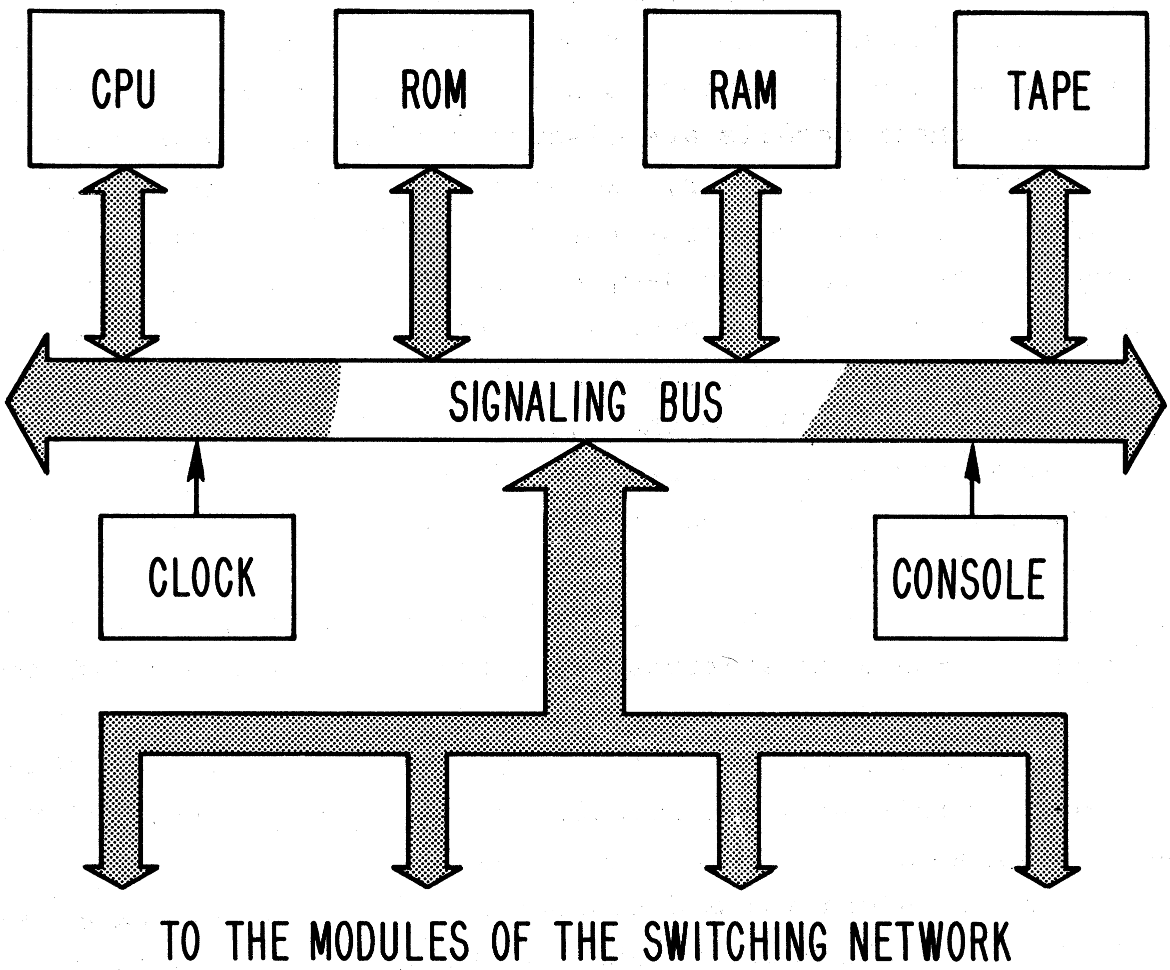

Figure 18. The CPU, its software memories, and the control distribution buses.

We assume that the CPU is equivalent to a special purpose minicomputer, with enough capacity to handle the largest access area system. When used for the 1000 line system, such a CPU may be underutilized and the PABX has considerable growth potential. Even then, an additional standby CPU unit is advisable for reliability.

The CPU has several memories. The ROM is a permanent store for the start and recovery steps, plus some basic system rules. The RAM, a read/write memory is used to store all operating software system functions, features, and restrictions. The magnetic tape unit is primarily a backup for the RAM, and it also provides entry for loading, program modifications, and data. Finally, the operator has a console to access the control system manually and to provide status display.

Whether it is synchronous or asynchronous, the common control bus can be used to distribute the system clock through the PABX.

During the life of the system certain functions and features are apt to be added, modified, or deleted. When these functions are executed from software, the software itself is subject to potential changes. These function and feature programs are stored in the RAM. Like subroutines, these programs are called by the CPU to execute the arising functions and features of services requested.

1.4.7. Service Features and Functions

From the nearly 50 features and functions listed in Table 1, we assume that 34 are essential enough to be incorporated in the PABX example. These 34 requested features and functions are listed in Table 6.

1.5. Prefatory PABX Cost Estimate

There is considerable freedom of choice and industry-wide uncertainty about the realization of various PABX function and feature combinations. The roles of hardware and software can complement each other in numerous ways. The programming languages may differ. Since the manufacturers tend to incorporate

Table 6. Features Selected for the Example

#	Feature	#	Feature
1.	Attendant	18.	Foreign Exchange
2.	BORSCHT	19.	Hold
3.	Call Forward	20.	Intercept
4.	Call Pickup	21.	Line/Trunk Card Density
5.	Call Transfer	22.	Line P=0.01
6.	Call Waiting	23.	MODEM Compatibility
7.	Camp-on	24.	Modularity
8.	Classes of Service	25.	Night Answer
9.	Conferencing Few	26.	Power Fail Transfer
10.	DID	27.	Secure Voice
11.	Digital Lines	28.	Standby CPU
12.	Digital Repeaters	29.	Standby Power
13.	Digital Trunks	30.	Stored Program Control
14.	DoD	31.	TDM
15.	DTMF Conversion	32.	Toll Restriction
16.	Ext-to-Ext	33.	Trunk P=0.002
17.	Flexible Numbering	34.	Trunk Types

proprietary aspects in software which provides features, the limited example discussed here cannot assess the detailed aspects of cost and feature relationships. One may conjecture that one or more of such elements, as

software,
cabinets,
console (TTY, display)
memory (RAM, tape),
clocks,
function generators,
interfaces,
special service modules,

and others, must be involved in providing system functions and service feature enhancements. Their aggregate costs must, of course, increase in some relation to the feature enhancements offered.

Partial price lists of some manufacturers have been scrutinized to the extent that interpretable specifics were available. The summarized results of this scrutiny are presented in Table 7. The numbers have been extrapolated and projected to the 10% digital phone PABX example introduced in Section 1.4. The table contains 18 major component categories, plus a final category called "others." The cost estimate applies to the 5000 and 1000 line installations.

It is seen that the eight largest items, namely CPU, software, cabinets, trunk card, line cards, per group A/D, channel groups, and individually digitized phones, account for 90% of the grand total. If the number of digital phone lines were increased from 10% to 50%, or even to 100%, the PABX costs would grow because the per-line equipment for digitized phones costs more than $350, and for analog phones, around $160. Note that the $350 figure can be explained immediately by adding up the line card, BORSCHT, and digitized handset numbers.

Software costs are more difficult to estimate then hardware costs. Software costs include all the effort involved in produc-

Table 7. Cost Projections for the 10% Digital Telephone PABX's

Major Cost Items	Cost Per Item ($)	F&F Cost Add.	5000 Lines		1000 Lines	
			# Items	Base Cost ($)	# Items	Base Cost ($)
CPU (Base, Backup)	30,000	*	1	30,000	1	30,000
Software (see Text)	20,000	*	1	20,000	1	20,000
Cabinets, etc.	5,000	*	5	25,000	3	15,000
Console (TTY, Panel)	2,500	*	2	5,000	1	2,500
Power (Main, Backup)	1,000		1	1,000	1	1,000
ROM/RAM (4K Model)	800	*	4	3,200	2	1,600
Magnetic Tape	3,000	*	2	6,000	1	3,000
Switch. Distribution Module	1,000	*	20	20,000	4	4,000
Switch. Digital Common Channel Termination	100		60	6,000	12	1,200
Switch. Digital Sub - Tl Concentrator	100		60	6,000	12	1,200
Switch. Digital Trunk Card	400		400	160,000	80	32,000
Switch. Digital Line Card	600		125	75,000	25	15,000
Switch. Digital Line BORSCHT	50		500	25,000	100	5,000
Switch. Analog Common Channel Terminal	120		75	9,000	15	1,800
Switch. Analog Per Group A/D	1,000		75	75,000	15	15,000
Switch. Analog Supergroup	100		150	15,000	30	3,000
Switch. Analog Channel Group	1,000		600	600,000	120	120,000
Digitized Handsets	150		500	75,000	100	15,000
Others (Clock, Bus.,...)		*		10,000		5,000
Totals ($)				1,166,200		289,300

ing and maintaining the necessary executive, support, and application programs and their documentation. The program development involves several phases including establishing performance and design requirements, creating detailed specifications, coding and debugging, installation, and final checkout. The \$20,000 estimate for software given in Table 7 is based on developing the software package for \$20M and amortizing this cost by the sale of 1000 units. The \$20M assumes that 70% of the software cost is attributed to various categories of personnel (e.g., engineers, programmers, managers, market specialists, etc.) and that the development of the software package requires 200 man years at \$70K per man year. This cost includes functions and features indicated in Table 6. Additional costs would occur if additional functions and features were required. The stars in the column identified as "F&F Cost Add" in Table 7 indicate those items where substantial cost add ons would occur due to additional functions and features (see Table 1). One notes from Table 7, that the smaller size PABX costs only slightly more per individual line. If one keeps the lines separate from trunks, then a 1000 line system costs \$289 per line, and a 5000 line system costs \$233 per line. If one combines the total counts of lines and trunks, then a 1000 line system costs around \$220 per line and trunk, and a 5000 line system costs around \$180 per line and trunk. These numbers appear reasonable when compared to estimates obtained from the PABX trade community. A simple model to depict the number of lines (L) dependence is the total cost/line = A + B/L, where A is the unit incremental cost for each added line and B is the initial cost common to all lines. Based on cost numbers of Table 7, two data points suffice to determine A and B. Counting only lines as L, not trunks, one obtains A $\cong$ \$220/line, and B $\cong$ \$70,000. For a small PABX installation of L=100 lines, this estimate yields a per line cost of \$920 and a total PABX cost of nearly \$100 K. One should emphasize here that this estimate appears more accurate for larger L. For small size PABX's, such as for L≤100, the model used can yield significant cost errors.

2. SIGNALING, SUPERVISION, PROTOCOLS AND INTERFACE ISSUES

2.1. Background and Scope

The previously discussed access area concepts of Figure 1 are further illustrated in Figure 19. Figure 19 is a simplified diagram of the access area digital switching system (AADSS) network. Users' voice and data terminals or stations are located in areas of terminal concentration, such as an office complex, and are connected via subscriber lines to switching hubs. These hubs are basically private automatic branch exchanges (PABX's). They may be interconnected to each other directly or via trunks to a central hub which acts as the central office. The central hub in turn connects to other central hubs, the central office of common carriers and to DCS offices over external trunks.

The local access area served by such a network may encompass a small number of terminals (<300), a medium number of terminals (300 to 2000), or a large number of terminals (>2000) (Wagner, 1977). The number, size and type of hubs in a given access area and the transmission network requirements depend on the total area, the terminal distribution, expected traffic and certain performance objectives.

At the present time (1977) most military bases utilize analog switching hubs and analog transmission facilities. External facilities are also primarily analog although the DCS European backbone is in the process of being changed to an all digital system as are many common carrier facilities (LaVean, 1977). This digitization process is expected to occur at the local access level in the future (1980 to 1990 time frame). It includes digital processor controlled switches, wideband digital transmission trunks and ultimately digital subscribers lines to voice terminals. As this evolution from all analog to all digital systems occurs, it is important to examine various aspects of the system to insure compatibility of the "new" with the "old" during the transition period and to plan the transition itself.

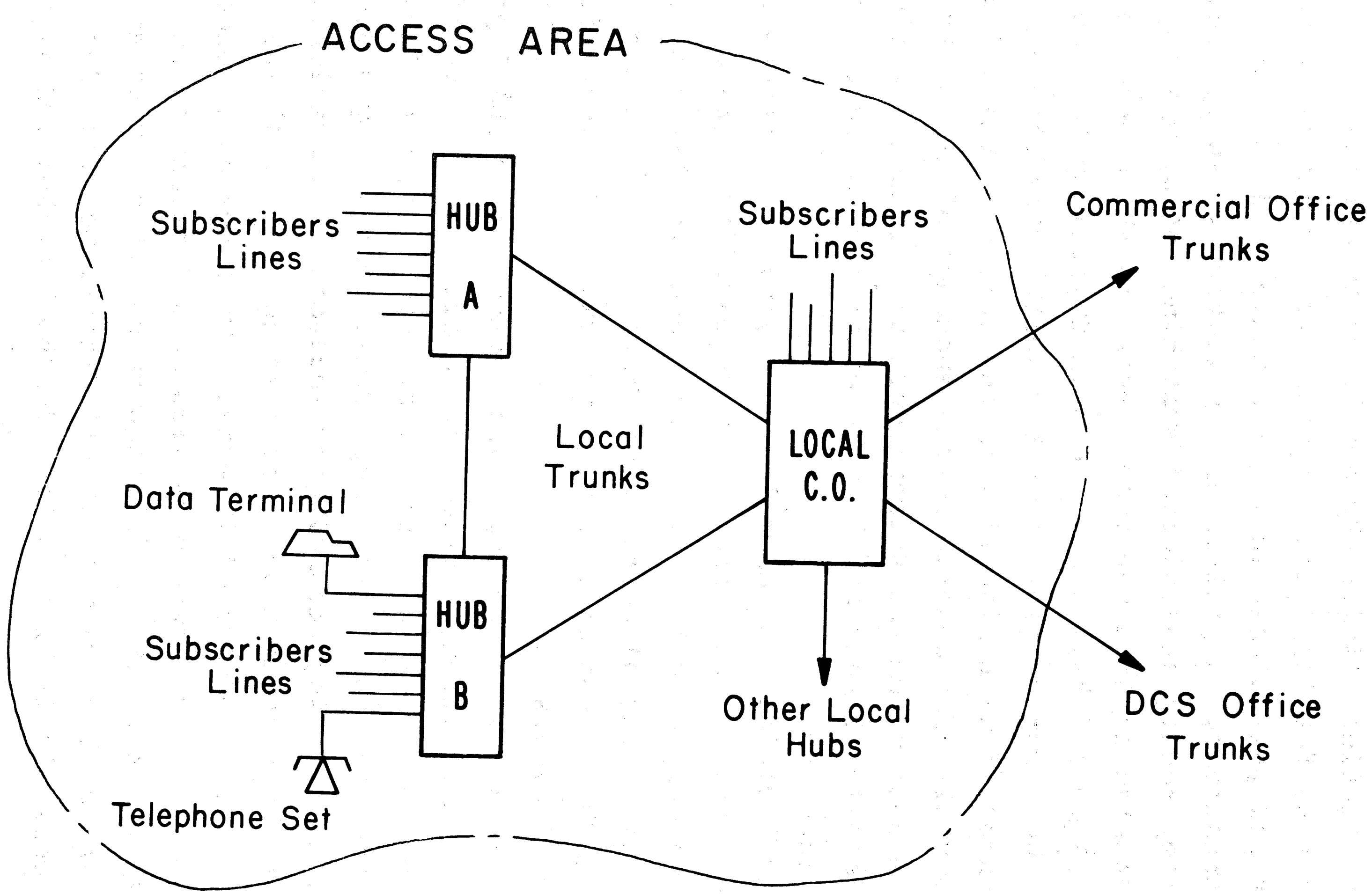

Figure 19. AADSS network.

The local access communications network of Figure 19 is a complex arrangement of transmission, switching, signaling, and terminal equipment. Transmission aspects were discussed by Nesenbergs and Linfield (1976) who developed functional and service parameters for local distribution systems and performed a preliminary analysis of service, performance, and cost of possible alternatives. A preliminary evaluation of switch hub alternatives for access area digital switching (AADS) was conducted and reported in an unpublished report to the sponsor by J.C. Blair[2]. A more detailed example of a digital switch hub is described in the first section of this report. This second section is concerned with signaling in the network. Signaling is the terminology used for describing the exchange of all control information, except voice and subscribers data, between terminals and hub, between local hubs and between central hubs and external switching centers.

Signaling includes, for example, supplying and interpreting the supervisory and address information required to operate the switches and thereby establishing a connection link between subscriber terminals. Signaling therefore involves the total network including the transmission links, the switch configuration and the users terminals.

Because signaling has an important impact both on AADS performance and on transitional issues, it is described in considerable detail in the following subsections. Signaling functions and concepts are discussed first (Sec. 2.2), followed by a description of various signaling technologies including supervision (Sec. 2.3). Past and present methods of addressing (Sec. 2.4) and audibles (Sec. 2.5) are also of interest. This review of older systems coupled with the newer ones aids in understanding not only the technology but also the interface problems that may be encountered during a transition to an all digital network (Sec. 2.6). Although the emphasis is on signaling for both analog and digital

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[2] J.C. Blair (1977), Preliminary Evaluation of Hub Alternatives for Access Area Digital Switching, ITS report to CSA, Project Order No. 501-RD, ITS, Boulder, Colorado 80303 (October, 1977).

voice terminals, signaling functions are also essential for data terminals. Strict procedures or protocols are required to initiate and maintain the data exchange (Sec. 2.7). Signaling has an important bearing on the design and performance of hubs capable of handling integrated voice and data. Signaling performance criteria must be defined to enable the comparison of different system configurations (Sec. 2.8). Access area network planners should consider and take advantage of these performance factors and their impact on cost before implementing significant parts of the AADSS. Some of the important issues relating to signaling which should be resolved by further study are summarized in Section 3.

2.2. Signaling Functions and Concepts

Plans for the future upgrading of communications for U.S. Army bases initially have concentrated on telephone facilities - the largest element in the present post communications environment. It is expected that ultimately the overall architectures of the upgraded facilities will provide a single integrated, multi-mode user, general purpose network for communicating narrative/record, computer, and other data traffic as well as voice traffic. The base communication plan (BASCOP) specified by the U.S. Army Communications Command (CEEIA, 1977) incorporates distributed digital switch hubs (PABX's) and remote switching units (RSU's), controlled by computer processing units (CPU's), all interconnected by digitally multiplexed transmission arteries. Subscriber lines or loops may be either analog or digital. Digital voice signals are generated either at the hub or at the terminal itself, using suitable compression and digitizing schemes such as linear prediction coding (LPC) or pulse code modulation (PCM). BASCOP currently does not provide guidance on the signaling and interfacing aspects of the local access network nor does it provide details concerning terminal interoperations with the switches. These signaling and interface aspects are the principal subject here.

Signaling is the means whereby the CPU at the hub and at the central office exchanges information with various user terminals (station signaling) and with other hubs and offices via trunks (interoffice signaling). This information exchange is required in order to establish, maintain, and clear connections engaged by a call through the switches, and to manage the information flow in the network. This type of signaling differs from the normal voice and data signals and therefore is sometimes referred to as control signaling (Davies and Barber, 1976). In the following paragraphs the control functions are first defined in terms of the tasks to be performed. Later the prevalent methods of exchanging the required information are classified and discussed.

2.2.1. Control Functions

Signaling control functions are defined differently in the switching literature. Leaky (1977), for example, divides the functions to be performed into three categories of communication tasks, short term actions, and subsidiary requirements. All three involve the basic operations of interrogation, interpretation, and action. Joel (1977) also uses three categories of basic functions, namely signaling, interconnecting, and controlling. The classification to be used here, however, is more detailed and tends to follow Dahlbom (1977). The functions are divided into three major areas of supervision, addressing and audibles, each with subclasses as depicted in Figure 20. Some authors include an additional set of management signal functions for maintenance and administrative purposes (Bellman et al., 1977). These management signals are not included here because they are similar to the supervisory signals, although they may originate in different parts of the network. Maintenance and administration signals are usually an essential part of the network and must be considered in the design of switching centers and the control strategies to be used.

2.2.2. Signaling Information Transfer Concepts

The signaling functions given in Figure 20 involve a transfer of information between user terminals or stations and

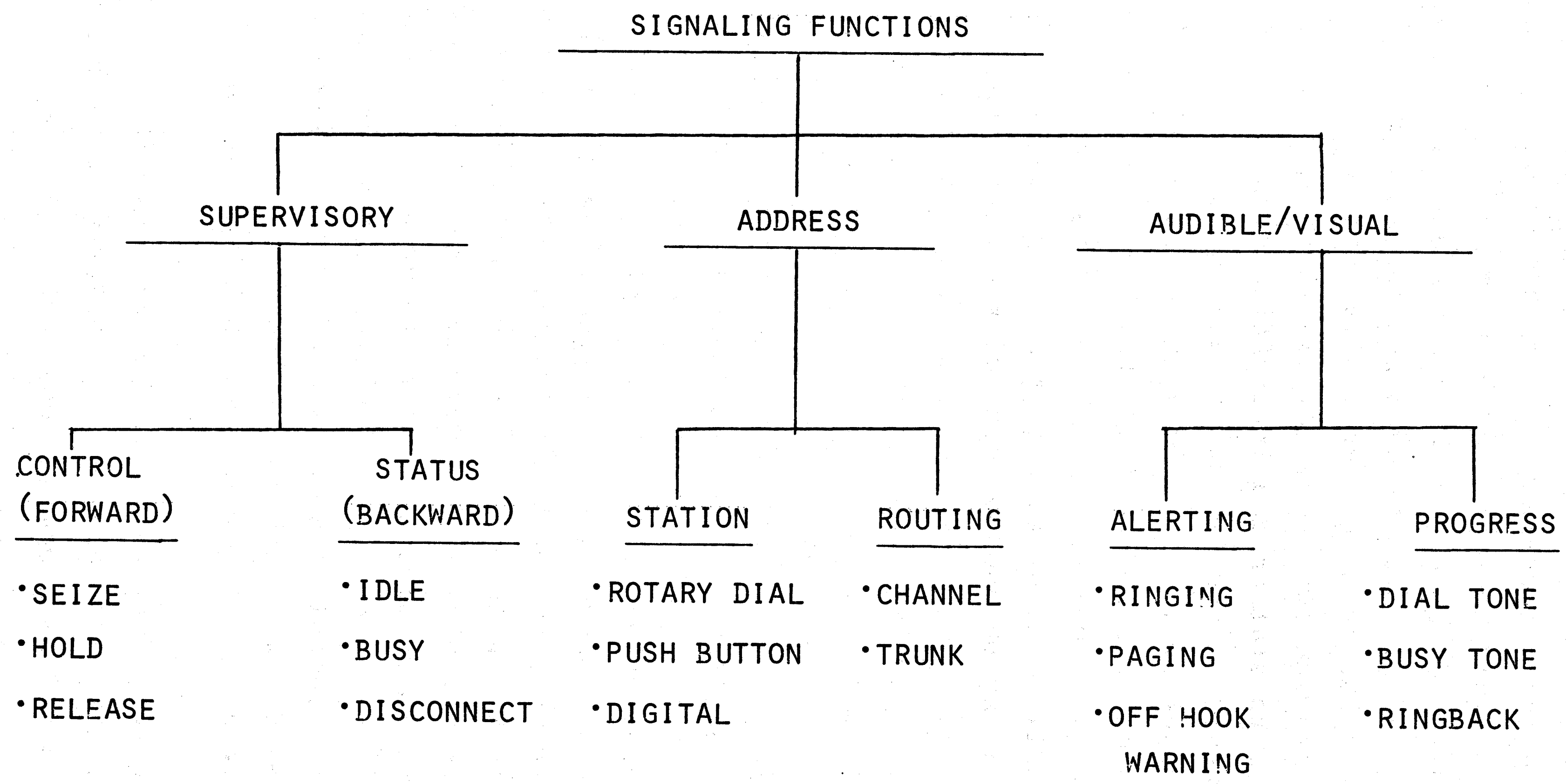


Figure 20. Signaling function breakdown.

the switch, and between switches in both directions. This transfer may take place alternately in each direction (half duplex), by time sharing the signaling link, or simultaneously in both directions (full duplex) by an appropriate separation on the link. Various concepts are used and they may differ depending on whether the transfer takes place between the station and the switch or between switches. Figure 21 shows some of the concepts used for both station and interoffice signaling. These concepts are discussed in the following paragraphs.

Interoffice signaling, i.e., signaling between switch hubs, is conventionally accomplished on a per trunk basis. Special signaling equipment is required at each end of the trunk. The actual signals usually consist of one or more frequencies although dc signaling may be used on some of the older trunks. Trunk signals may be compelled or noncompelled. Compelled signaling involves acknowledgement. The signal is applied continuously until reception is acknowledged by a return signal. Noncompelled signals do not require a return signal.

More recently, common channel signaling has been employed between the newer switching centers which employ stored program control. Common channel signaling uses a relatively high speed, dedicated data link (typically 2400 b/s) to carry the signaling information for many trunks. This data link is independent of the communication path used for voice transmission. The common channel may operate in an associated or nonassociated mode. In the associated mode the signals for several trunks are transferred between hubs which also terminate the trunks. In the nonassociated mode the signals may be processed forward through several signal transfer points which are not necessarily termination points for the trunks being controlled.

These interoffice signaling concepts are illustrated in Figure 22.

In Figure 22a separate signaling equipment is shown interfacing with each trunk with multifrequency (MF) address units switched to the trunk from a common pool. This concept is still

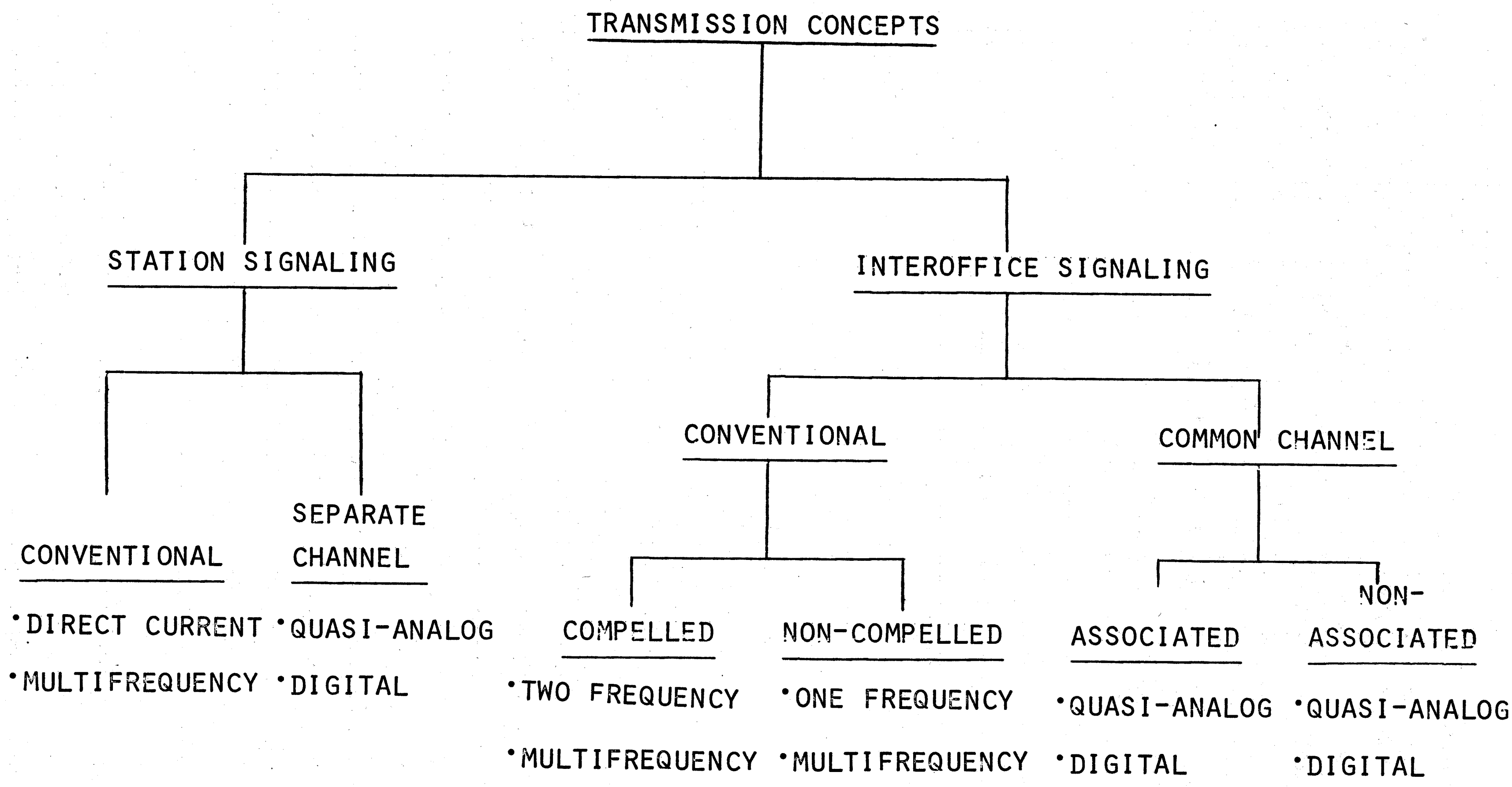


Figure 21. Signaling information transfer concepts.

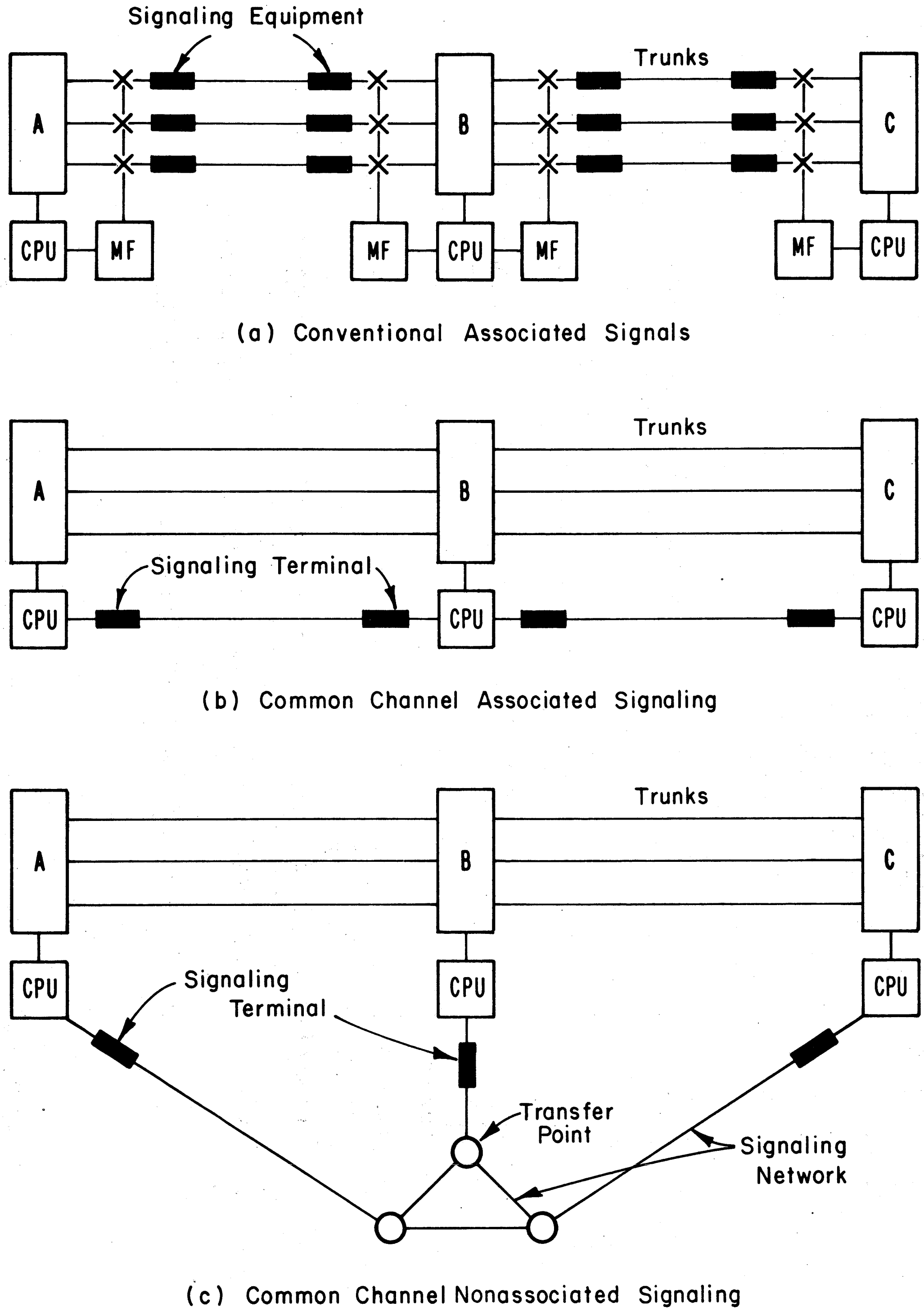


Figure 22. Interoffice signaling concepts.

used in switch centers employing older electro-mechanical switches. The signaling equipment, which includes registers and senders for each trunk, occupies a considerable space and accounts for a large part of the cost of a switch center (Davies and Barber, 1976).

It is feasible to have a direct common channel signaling link handling the signaling for several channels between the newer electronic switch centers. This link may be either associated as in Figure 22b or nonassociated as in Figure 22c. The advantage of the common channel signaling in terms of the physical size of the equipment required has been indicated by Leaky (1977). An electromechanical switch center with signaling equipment for each trunk might, for example, require 100 racks of equipment. If this mechanical switch was replaced by a digitally switched center with stored program control, the center might occupy 25 racks, 10 of which would be the electronic switch and 15 racks would contain signaling interface equipment for analog trunks. Furthermore, if these 15 racks of interface equipment were replaced with direct interprocessor signaling, i.e., common channel equipment, and digital trunks were used for transmission, then the interface racks required would be virtually eliminated.

Conventional station signaling is usually transmitted over the same two-wire pair as the voice signals by opening and closing the DC current path. A time sequenced interruption of this path by dial pulsing provides address information. Pairs of frequencies may also be transmitted for the address code. With a four-wire telephone set the signaling may use one pair of wires for voice and one pair for signaling in a quasi-analog mode whereby the digital signaling information is transmitted over the analog circuit using modulation and demodulation (modems) to match the signals to the channel. Over a four-wire digital link certain time slots are reserved for signaling purposes or are time shared with the digitized voice.

The main concepts for signaling on subscriber lines are illustrated in Figure 23. In Figure 23a the analog station signaling information is exchanged over the subscribers loop consisting of a wire pair and follows the same channel as the voice signals. This method is in common use today with conventional telephone sets. Figure 23b depicts another concept for signaling over a subscriber's line consisting of two wire pairs. Analog voice information is transferred over one pair and the signaling information over the other both using full duplex operation. In Figure 23c the station terminal is assumed to be a digital telephone operating over two wire pairs. Present day full duplex operation requires that both voice and signaling information be transmitted in each direction over separate wire pairs. In the future, technology could well be developed to accommodate digital full duplex operation over a single pair of wires.

In the following section, some traditional signaling techniques are discussed, followed by some advanced digital techniques. Both are included because the AADS centers may be required to interface with different types of switches and transmission facilities as implementation progresses. The technologies described are not necessarily all inclusive. Others are described in the literature. Pertinent sources include Breen and Dahlbom (1960), Hamsher (1967), and AT&T (1975) for historic systems. Information concerning some of the newer common channel technologies and digital signaling was obtained from Martin (1976), AT&T (1975), Crawford (1974), deLeon (1973) and CCITT (1973a).

2.3. Signaling Techniques

The technique used for signaling may be either analog or digital. Analog systems include both direct current (dc) and alternating current (ac) methods as shown in Figure 24. The dc techniques evolved first. They are simple to operate and inexpensive. These techniques are described in Section 2.3.1. Some of the methods are in common use today particularly for sub-

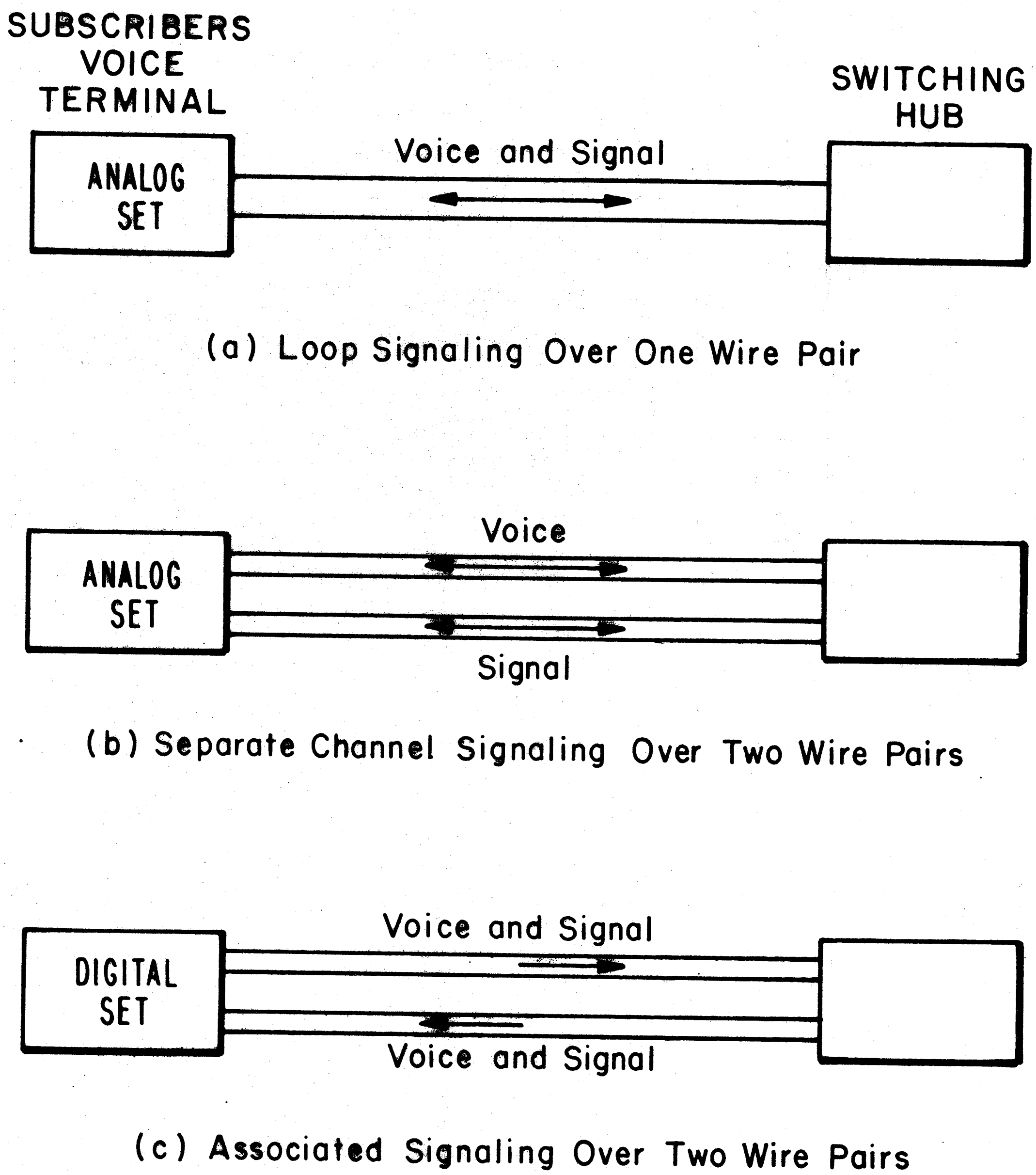


Figure 23. Station signaling concepts.

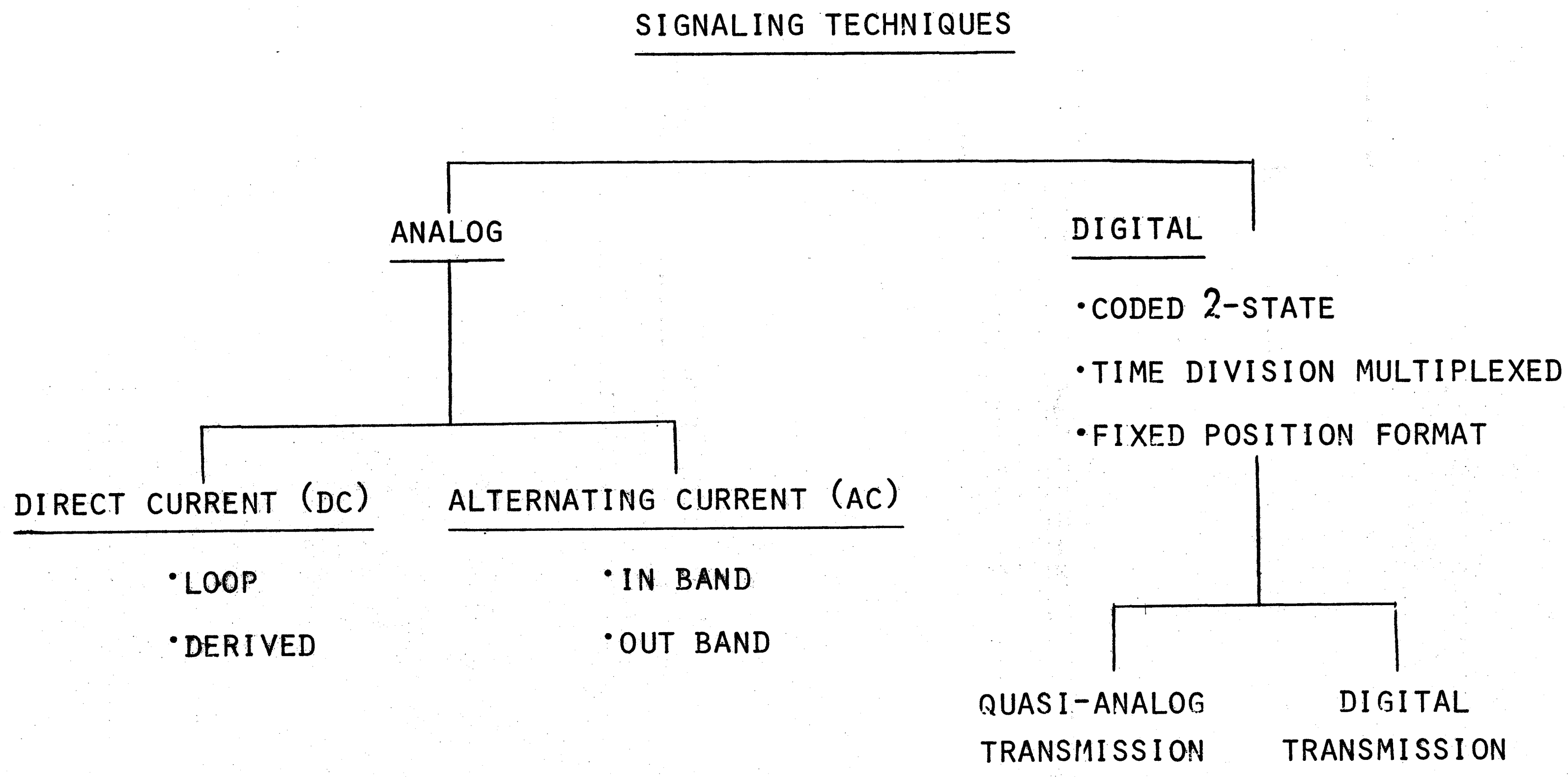


Figure 24. Basic technologies used for signaling.

scriber loops. The ac technique uses two state ac frequencies for supervisory signals and multifrequency signals for addressing. A single frequency (2600 Hz) is commonly used to indicate on-hook conditions for a trunk circuit. This and other ac methods of supervision are described in Section 2.3.2. Digital signaling is becoming more popular because it can provide more services and features. Unfortunately, it is also more complex, requires more bandwidth and is more costly. Digital supervisory signals may be transmitted as distinct codes equivalent to the two state dc or ac signals; they may be time division multiplexed with voice or data signals; or they may be sent in a prescribed format. Basic methods are described in Section 2.3.3. Although the emphasis in these sections is on supervisory signals for lines and trunks, many of these same techniques also apply for addressing. Addressing aspects are discussed in Section 2.4.

2.3.1. DC Supervision

Two basic techniques are used for dc supervision: loop signaling and derived signaling. Loop signaling schemes are used where a dc path is available. Usually this occurs on subscriber lines, but it is also feasible on short trunks. Derived signaling schemes are used on trunks, when a dc path is not available, or where extended dc ranges are needed. The two schemes are described below.

Loop Signaling

The basic circuit for loop signaling on a subscriber's 2-wire loop is a series circuit which provides one signaling state when the loop is open (on-hook) and another signaling state when it is closed (off-hook). Such a circuit is illustrated in Figure 25 for a 2-wire type 500D telephone set. It is apparent from the circuit in this figure that closing the hook switch, which is operated by lifting the handset, permits current to flow from a common battery supply at the switch hub. This current flow operates a relay at the hub whose contact(s) provide supervisory information to the hubs. Address information is provided by

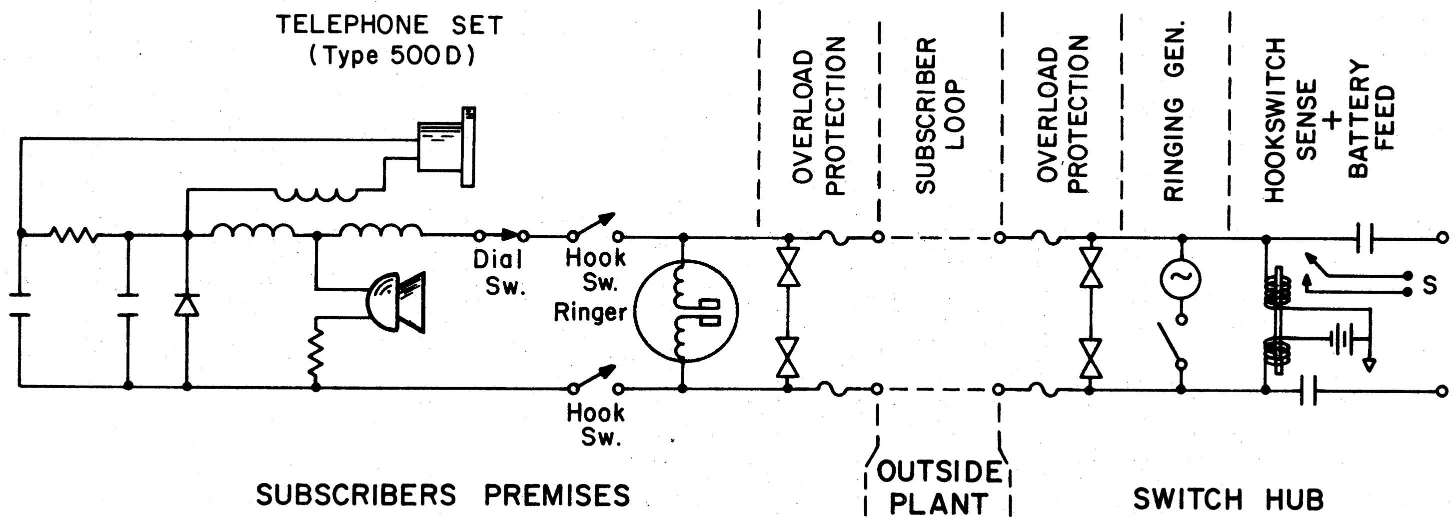


Figure 25. Conventional 2-wire telephone set and subscriber loop connections to switch hub.

interrupting the current with the dial switch. At the same time the battery provides the power to operate the transmitter (mouth piece) in the handset. Ringing current is provided to the set by a separate hub contact closure. The interface to the subscriber's premises and to the hub includes overload protection circuitry from the outside loop.

Methods used to accomplish two-state dc signaling on trunks using loop signaling include:

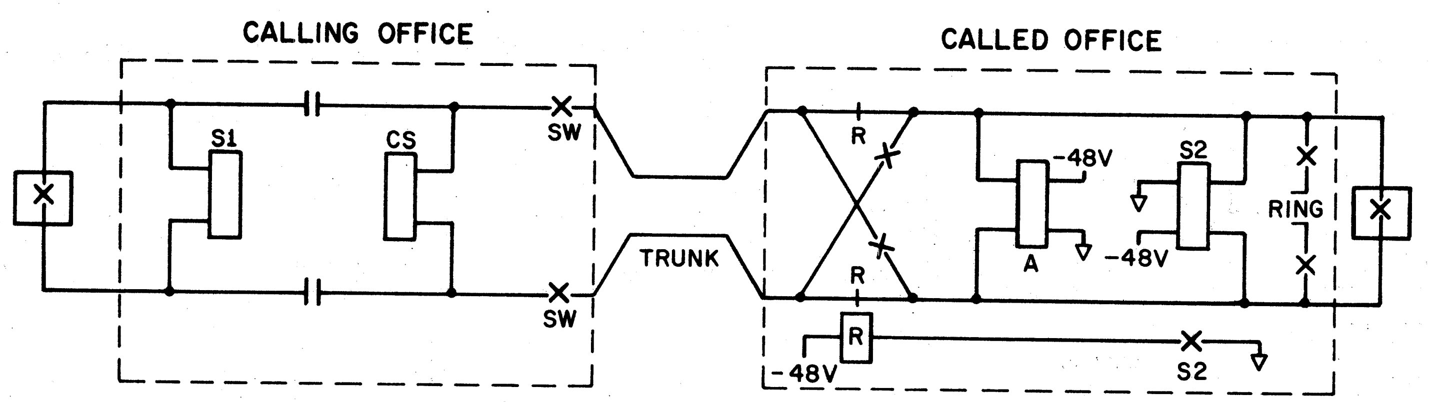
1. Interrupting the dc path.
2. Reversing polarity of the battery voltage.
3. Changing current magnitude by varying resistance in the circuit.
4. Combinations of the above.

Reverse battery signaling is a commonly used dc signaling scheme and the circuitry is diagrammed in Figure 26. Relays with delay coil windings are used in this circuit to sense the presence (operated) or absence (released) of current in the circuit. A polarized relay (CS) is used to sense the direction of the current flow. The CS relay also has a mechanical bias which causes it to release on an open circuit. Thus it detects three states - normal, operated and released. The condition of the various relays during the progress of a call is indicated in Figure 26.

Derived Signaling

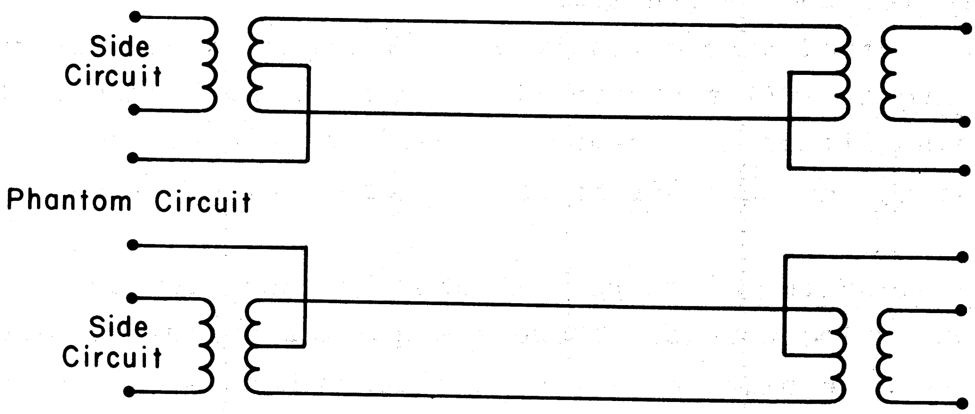
For trunks that use loop type signaling, the dc currents flow in a series circuit, whereas trunks with derived signaling superimpose a separate circuit on the voice transmission circuit. A schematic of one such superimposed circuit is drawn in Figure 27a. Using the phantom or derived circuit and the two side circuits as shown, it is possible actually to transmit three telephone conversations over two pairs of wires with little or no interference.

A similar scheme, once used for dc signaling over trunks, is the simplex circuit (SX) shown in Figure 27b. Here one wire pair is used for voice transmission. Signaling is conducted by

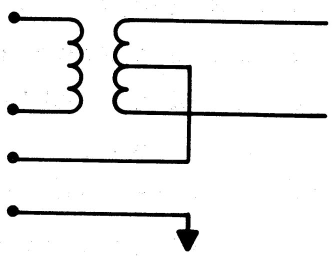


HOOK SWITCH	S1	CS	SW	R	A	S2	HOOK SWITCH
OFF	OPERATED	NORMAL	CLOSED	NORMAL	OPERATED	NORMAL	ON
OFF	OPERATED	OPERATED	CLOSED	REVERSED	OPERATED	OPERATED	OFF
ON	RELEASED	RELEASED	OPEN	REVERSED	RELEASED	OPERATED	OFF
ON	RELEASED	RELEASED	OPEN	NORMAL	RELEASED	RELEASED	ON

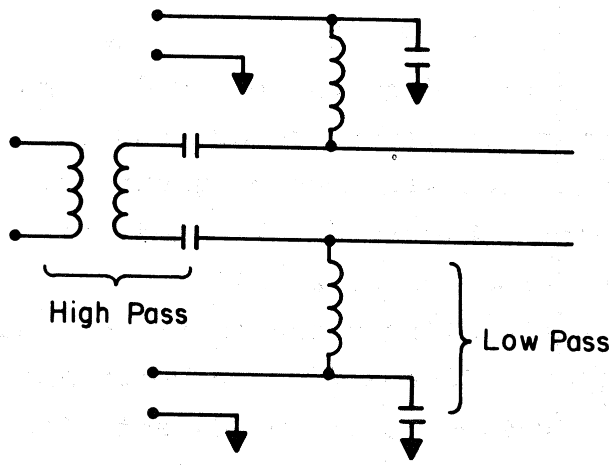
Figure 26. DC trunk signaling over wire pair with reversed battery.



(a) Three Transmission Circuits
on Two Wire Pair



(b) Simplex Circuit



(c) Composite Circuit

Figure 27. Derived signaling circuits.

paralleling the wire pair for one side of the circuit and by using ground for the return side.

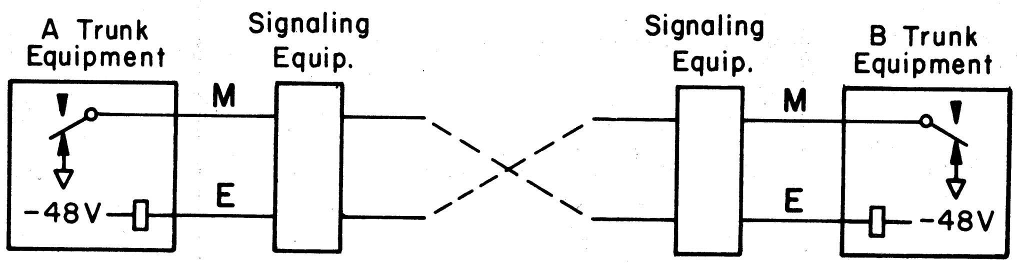
Another method more commonly used on analog trunks is the composite circuit (CX) shown in Figure 27c. A high-pass filter and a low-pass filter arrangement is used to separate dc and low frequency signaling currents from the higher voice frequency currents.

Duplex signaling circuits (DX) are also used for dc signaling on analog trunks. These are similar to the CX circuit but do not require high and low pass filters because the circuit is balanced and symmetrical at both ends.

For trunk signaling, some means for transferring the signal information between subscribers' lines and the derived signaling circuits must be provided. The signal exchange is obtained with trunk equipment, which transfers on-hook and off-hook status conditions from the subscriber's line to special leads historically designated E and M leads at the switching center. The E lead carries signals from the switching equipment to the signaling equipment and the M lead does the reverse. Thus signals between switch offices are transmitted via the M lead and received via the E lead. Note that the signaling equipment for the trunk may be dc, ac, or even digital. The basic E and M signaling concept is depicted by the block diagram in Figure 28. The M lead sends battery or ground conditions to the signaling equipment. The E lead receives open or ground conditions from the signaling equipment. Figure 28 also indicates the E and M conditions during the progress of a call.

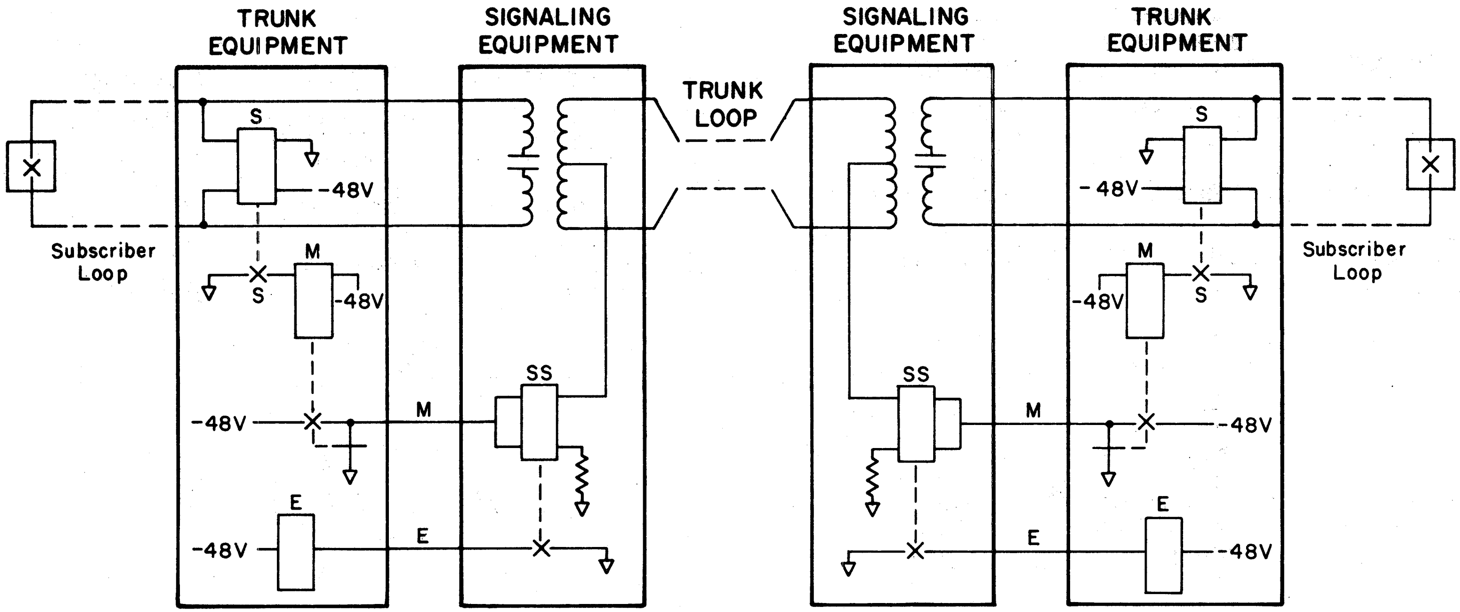
A circuit for dc signaling using an SX derived circuit and E and M control is drawn in Figure 29. In North America SX sets have been largely superceded by DX and CX sets for trunk signaling between older electromechanical switching centers.

Although the historic two-wire E and M control interface operated reliably between electromechanical switch centers, it was found to be unreliable in the electronic switch environment due to excessive noise pickup. A four-wire E and M lead



CONDITION AT A	M LEAD	E LEAD	M LEAD	E LEAD	CONDITION AT B
OFF HOOK	BATTERY	OPEN	GROUND	GROUND	ON HOOK
ON HOOK	GROUND	GROUND	BATTERY	OPEN	OFF HOOK
OFF HOOK	BATTERY	GROUND	BATTERY	GROUND	OFF HOOK
ON HOOK	GROUND	OPEN	GROUND	OPEN	ON HOOK

Figure 28. E and M signaling over trunk circuit.



NOTE: Battery On M Relay Operates Distant SS Relay
But Not Nearby SS Relay

Figure 29. DC trunk signaling using simplex circuit with E and M control.

interface incorporating ground return leads is used with electronic switch systems to reduce noise pickup problems. The schematic of one type of four-wire interface is shown in Figure 30. With this interface it is possible to also connect two trunk circuits or two signaling circuits together using cross connections between the E and M leads.

2.3.2. AC Supervision

Signaling with dc may not always be feasible over analog trunks. For example, dc signaling is impossible when carrier transmission facilities are used between distant offices. In such cases ac signals may be employed. A continuous frequency tone, usually 2600 Hz in North America is transmitted in each direction over four-wire trunks to indicate an idle or on-hook condition. When the trunk is seized, the tone is disconnected. On a two-wire trunk two continuous frequencies (2600 Hz and 2400 Hz) are used for forward and backward transmission.

Since these frequencies fall within the voice band and go anywhere the voice signal goes, this technique is known as in-band signaling. A less commonly used ac signaling scheme employs continuous frequency tones outside the voice bandwidth such as 3700 Hz in North America. This out-of-band signal is found on certain types of short-haul carrier systems.

Like dc signaling equipment, ac signaling equipment is associated with each trunk. It is operated by E and M leads to and from the switching equipment. An example of the circuit used for ac supervisory signaling on a four-wire trunk is shown in simplified form in Figure 31. This method is still in common use today for many toll telephone calls over analog trunks in North America. The tabulation associated with Figure 31 indicates the operations involved and the E and M lead conditions for these operations.

The one or two frequency signals corresponding to two or one wire pairs represent the supervisory states as in dc supervision. Multifrequency (MF) signals are also used for addressing. These MF signals are discussed in Section 2.4.5.

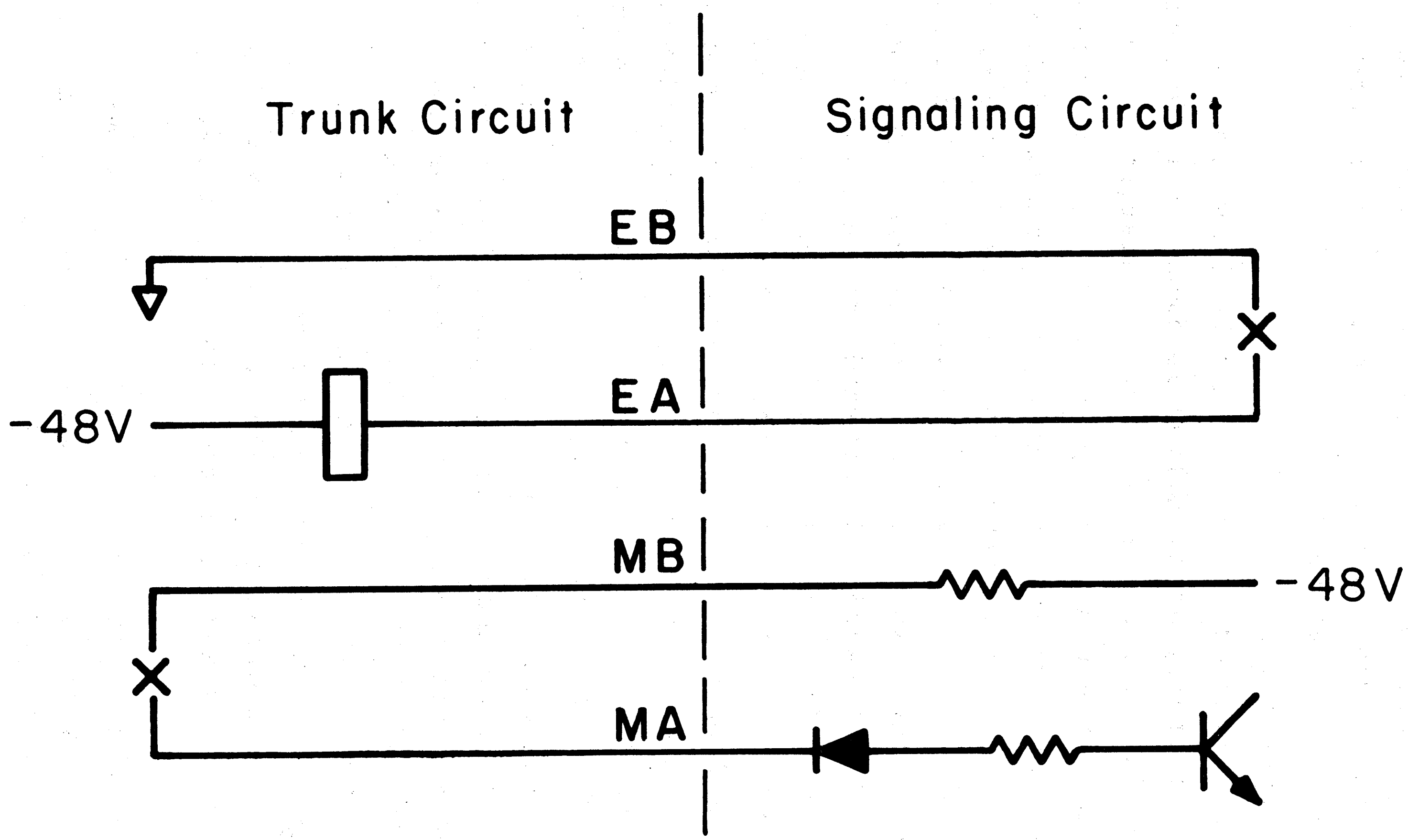
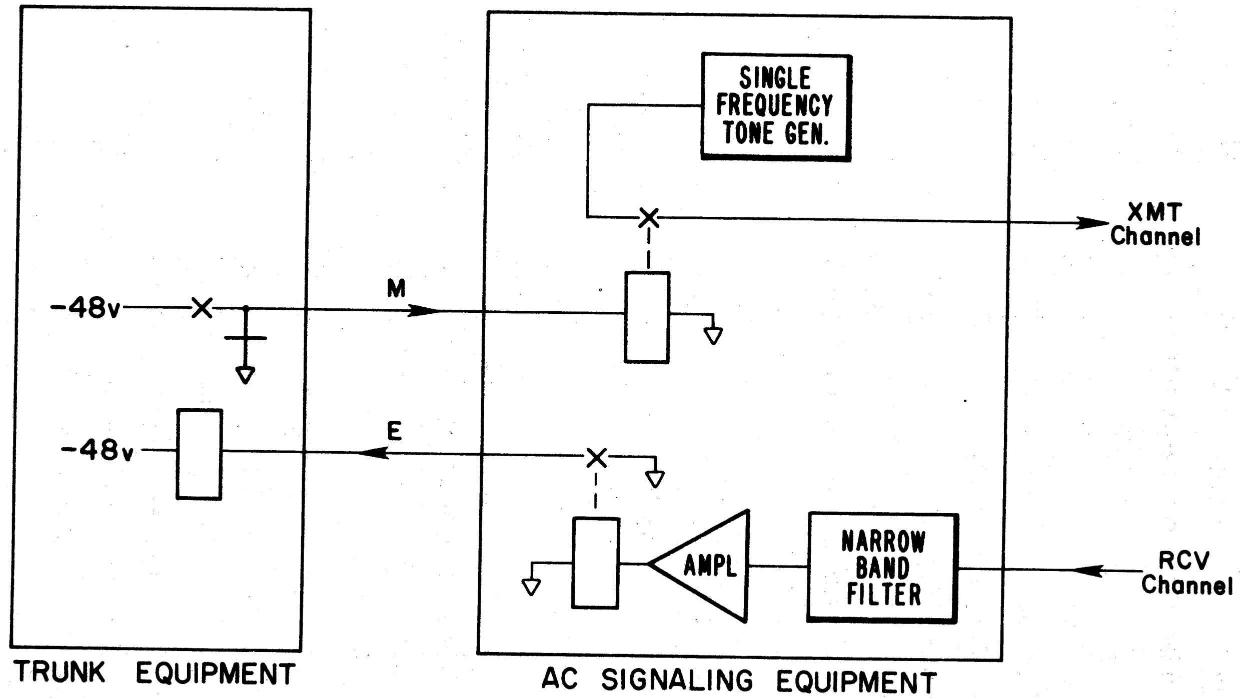


Figure 30. E and M interface with electronic switching centers.



SIGNAL	TONE	OPERATION	LEAD	CONDITION
On Hook	On	Transmitting	M	Ground
		Receiving	E	Open
Off Hook	Off	Transmitting	M	Battery
		Receiving	E	Ground

Figure 31. Single frequency signaling circuit and operations for trunk with two wire pairs.

2.3.3. Digital Supervision

Digital signals may be transferred via analog or digital transmission facilities. The facilities may be customer lines or interswitch trunks. For analog lines and trunks the digital signals modulate a carrier for transmission and demodulate after reception. This so called quasi-analog modulation and demodulation (modem) equipment is required at each end of the analog transmission facility. Given digital transmission facilities, the digital signals do not require a modem to interface with the line or trunk but can be transmitted directly using simple line interface equipment.

Digital signals may be derived from the dc states by a sampling and encoding process in the same manner as voice signals are digitized. Digital signals may also be generated directly by special logic circuitry. Once the appropriate digitization process has been accomplished, the signals may be transferred along with data or voice signals using a variety of formats. The signaling information may be interleaved with the voice and data information by time division multiplexing (TDM) the digital signals into a continuous bit stream. This may be considered out-of-slot signaling since separate time slots are allocated for signaling purposes. An alternative method is to send the signaling information in block format with a header or qualifier bits used to distinguish between signaling and information bits. This may be considered in-slot signaling since signaling information occupies the same channel as the voice information.

The TDM scheme has an advantage in that the signals are transmitted in real time and therefore can be used for both voice and data networks. Signaling in block format requires block processing storage and certain delays. These real-time delays cannot be tolerated in many applications. Block signaling schemes can, however, be used for message switching and for common channel signaling.

Digital signaling may be either associated or nonassociated. With associated signaling the digital transmissions occur either on a per channel basis or on a common channel. In both cases the signaling path parallels or is associated with the information path. With nonassociated signaling a completely separate network is used. Three basic digital signal techniques are described in the following paragraphs - namely associated signaling per channel, associated signaling over common channel and nonassociated signaling over common channel. Both line (station to hub) and trunk (hub to hub) signaling is considered.

Associated Signaling Per Channel

It is expected that a certain percentage of the future AADS customers will be capable of digitizing and encoding voice signals using some form of voice processor. Regardless of the processor employed, the output will probably consist of a binary coded bit stream with certain bit slots reserved for supervisory, address, and other types of information. Pulse code modulation (PCM) is one form of process which is in common use today and will be used as an illustration here (Members of Technical Staff, 1971).

A typical PCM system samples the 4 kHz analog voice signal at 8 k samples per second and encodes each sample with either 7 or 8 binary bits representing 128 and 256 amplitude steps respectively. When 7 bit encoding is used an additional bit is added for signaling. When 8 bit encoding is used the eighth bit of every sixth encoded sample is borrowed for signaling. A breakdown between voice and signaling rates for each scheme is summarized below.

Sampling Rate Samples/s	Quantization bits/sample	Voice Rate b/s	Signaling Rate b/s	Total Rate b/s
8k	7	56k	8k	64k
8k	8	62.7k	1.3k	64k

A block diagram of one concept envisioned for a digitized voice terminal with associated signaling per channel is shown in Figure 32. The voice processor portion, which includes the low

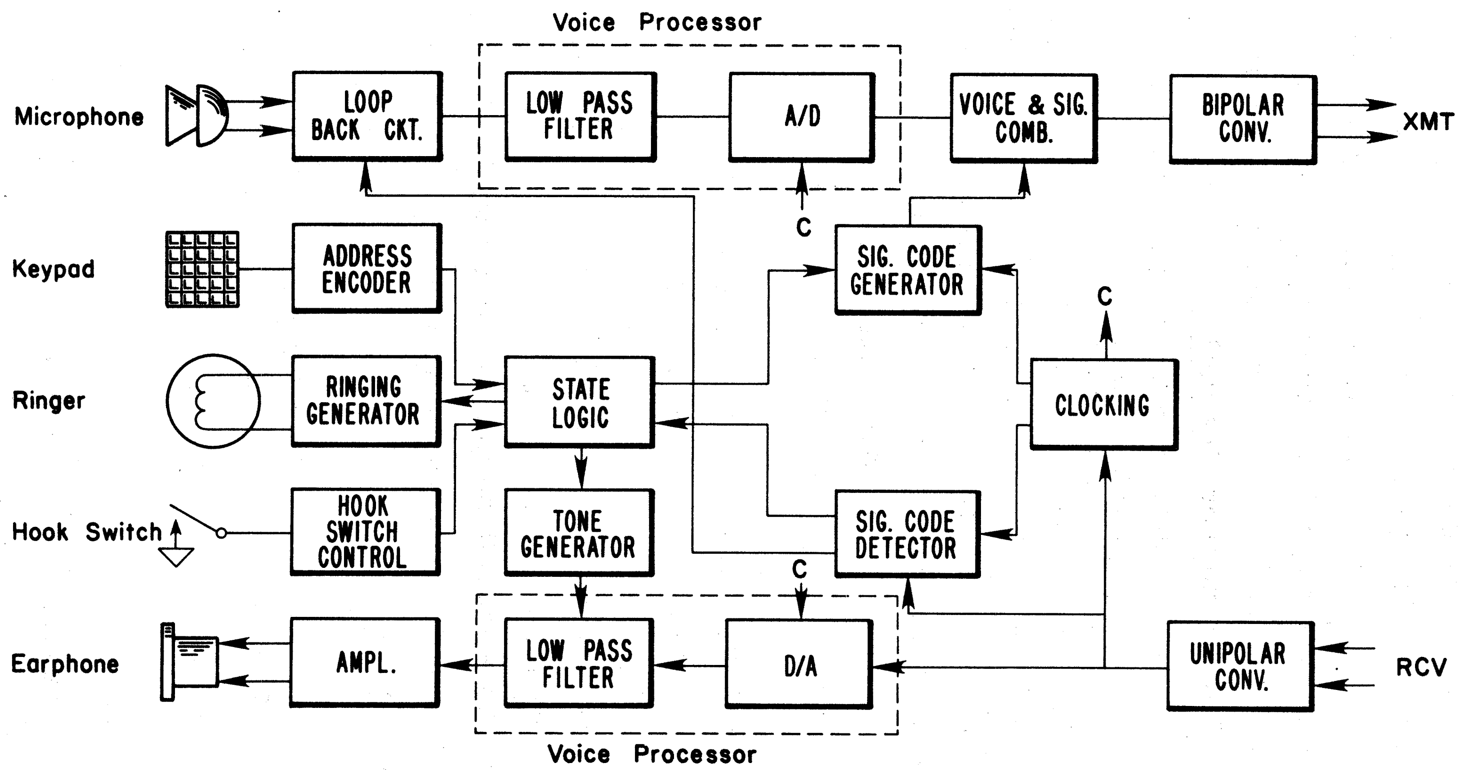


Figure 32. Digital voice terminal block diagram.

pass filter and the analog-to-digital converter (A/D), can assume various forms including PCM, adaptive delta modulation (ADM), continuously variable slope delta (CVSD), and linear predictive coding (LPC). A block diagram of a PCM processor is shown in Figure 33.

The processor incorporates a compression technique which quantizes low-level samples with small steps and high-level samples with larger steps. This nonlinear quantization maintains a relatively constant signal-to-quantizing distortion ratio over a wide range of talker levels by using a compression characteristic defined by the equation

$$y = \operatorname{sgn}(x)\frac{\ell n(1+\mu|x|)}{\ell n(1+\mu)}, \quad -1 \leq x \leq 1.$$

When μ=255 the signal-to-quantizing distortion ratio is nearly constant over a 40 dB range of input signals. Signaling information is inserted after and detected prior to the voice processing.

It is common practice to multiplex several analog voice channels into one digital channel using time division multiplexing for transmission over digital trunks and for digital switching. This can be accomplished either before or after the encoding as indicated in Figure 34. The most common multiplexer is shown in Figure 34a where multiplexing is accomplished after sampling and prior to encoding. The pulse amplitude modulated samples (PAM) are interleaved on a common bus using high speed clock driven gates. Common equipment can then be used to encode all the multiplexed samples on the bus. The bit code for each sample consisting of 7 or 8 bits appears consecutively, either bursted or equally spaced. This technique is normally used at the switch to combine several analog voice channels into a digital trunk.

It is also feasible to time division multiplex several channels after encoding as indicated in Figure 34b. Separate encoders and decoders are required for each channel. The coded bits corresponding to each sample are interleaved with equal spacing by taking one bit (rather than seven or eight as before)

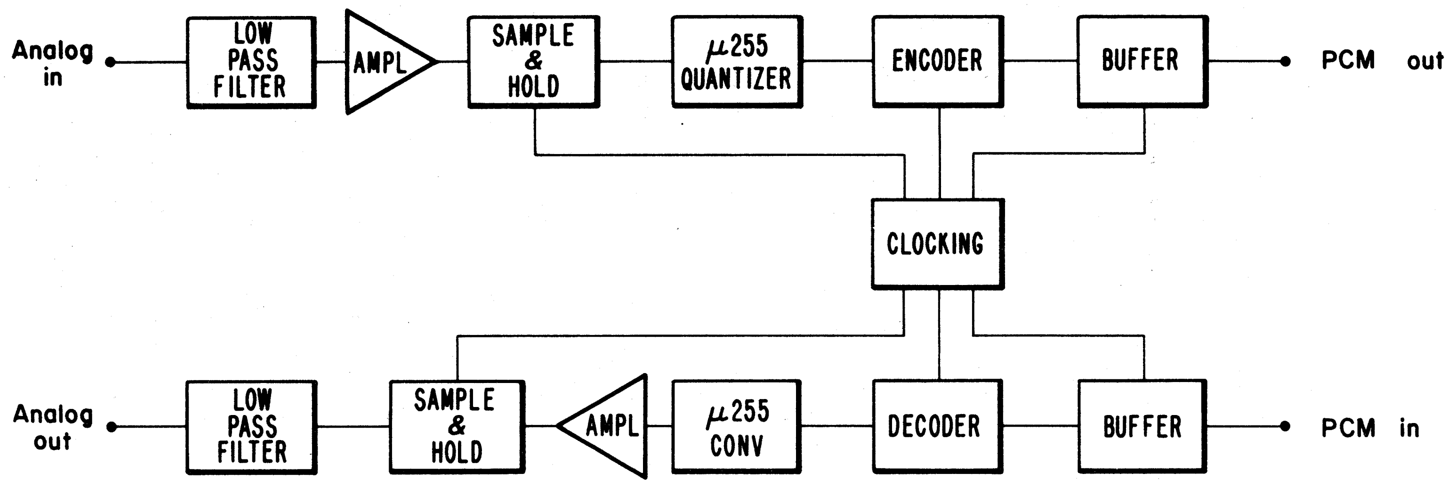


Figure 33. PCM voice processor.

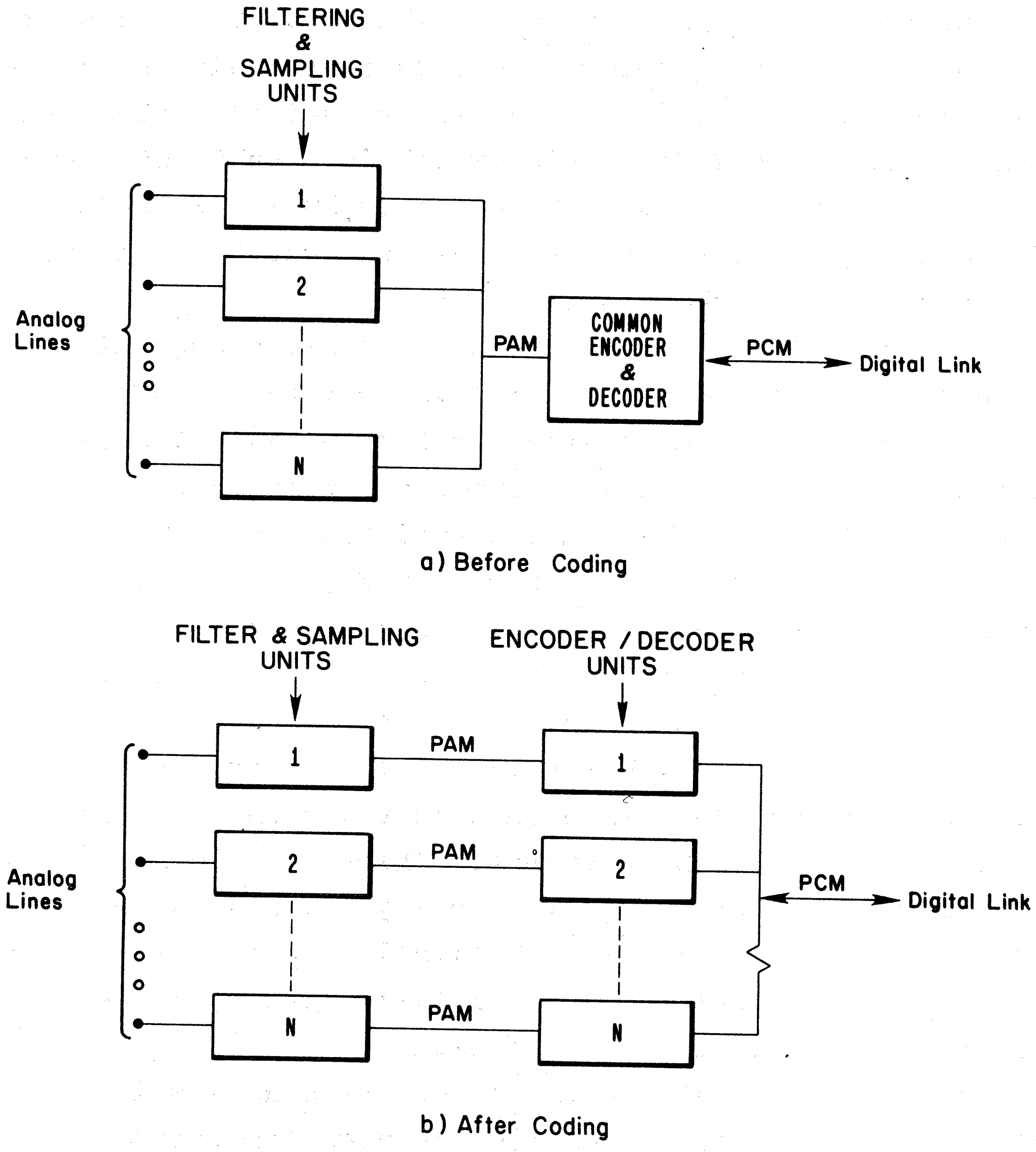


Figure 34. PCM multiplexing techniques.

from each channel and transmitting them sequentially. This method may be preferable where digital lines are used to link the switch to a digital station. Coding before TDM also appears to be simpler to implement when both digital voice and data are mixed to various degrees for transmission over digital carriers.

The number of voice channels, which can be multiplexed together using either method, depends on the digital transmission facilities to be employed. One widely used digital carrier in North America is the T1 carrier which is capable of transferring 24 voice channels over short hauls (<80 km.). In Europe a different digital carrier system is used which is capable of transferring 32 voice channels.

With the T1 system the twenty-four 8-bit channels are combined into a frame of 192 bits and one bit is added for frame identification. Therefore the total frame is 193 bits long and, since there are 8000 frames per second, the signaling rate is 193x8000=1.544 Mb/s. The multiplexed PCM framing and signaling format for the T1 carrier is shown in Figure 35. The 193rd bit is formed by alternating odd and even bits: the odd bits for channel framing and the even bits for signal frame identification. The actual channel framing code consists of alternate 1's and 0's, while the signal identification code consists of three 1's followed by three 0's. When interleaved these two codes produce a periodic sequence in 12 frames. The signal identification code provides a means for identifying every sixth frame where the borrowed eighth bit is used for signaling purposes.

The European system with thirty-two 8-bit channels results in a frame 256 bits long. No framing bit is added, so the total rate is 256x8000=2.048 Mb/s. One of the channels is used for frame synchronization and another is used for control signaling, leaving 30 channels for voice transmission. Each voice channel is allocated one 8-bit slot per frame and since there are 8000 frames per second there are 64 kb/s available for voice transmission. A 16 frame cycle makes up a multiframe. Multiframes occur at a rate of 500 per second and contain 4 bits of signaling

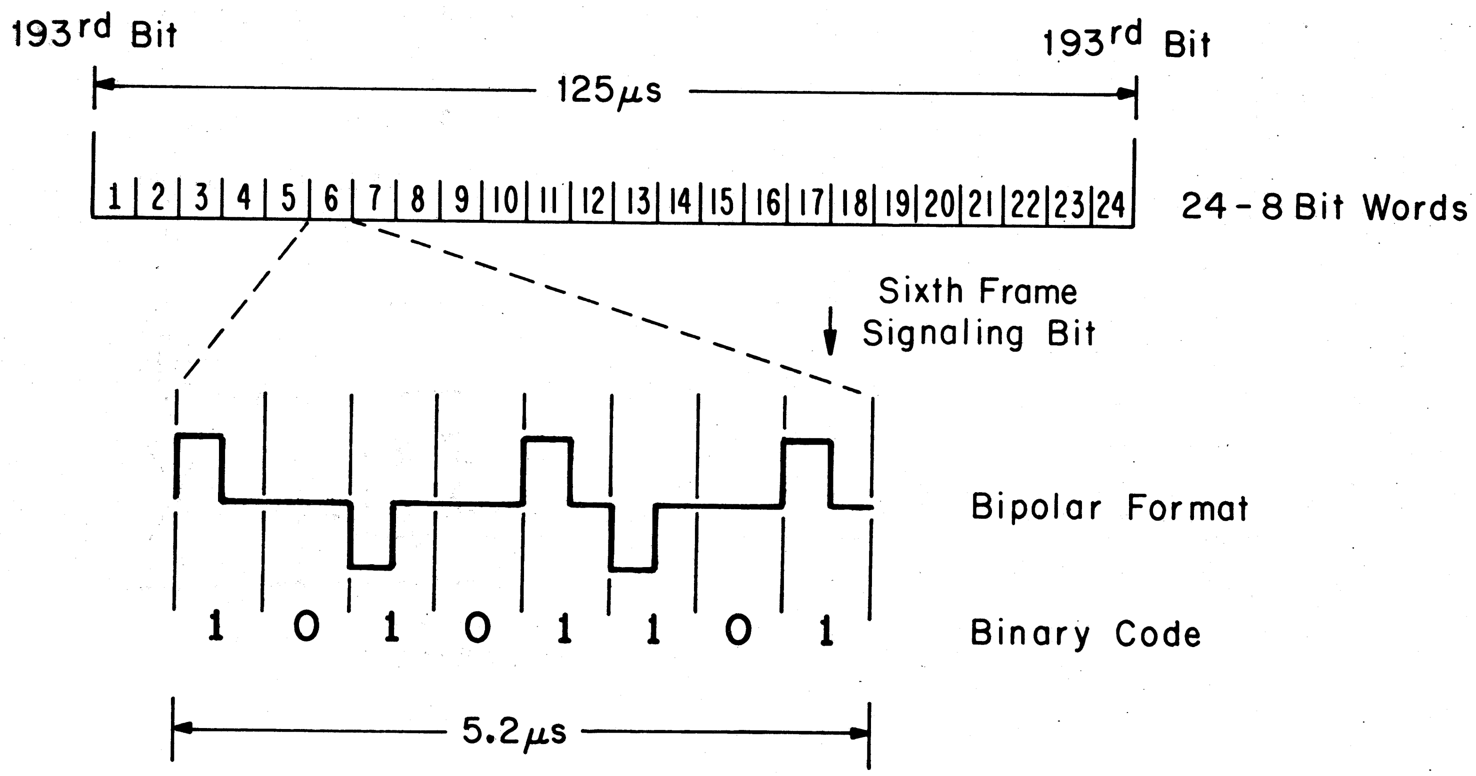


Figure 35. Multiplexed PCM framing and signaling format (North American system).

information per each of 32 channels. Therefore each of the 30 voice channels has a 4x500=2000 b/s signaling rate. The European format is shown in Figure 36.

It is possible to multiplex voice channels at higher levels than indicated above. This is desirable for longer trunks and on radio links. The T-carrier hierarchy, which has become an industry standard in North America, is listed in Table 8.

Table 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

Associated Signaling Over Common Channel

It is feasible to use associated common channel signaling with either the North American or European digital formats described in the previous section. In the European system this is accomplished by block encoding one of the 32 channels. Such a channel provides a 64 kb/s signaling rate. In the North American system each 193rd bit occurring in odd frames is still used for frame alignment and each 193rd bit occurring in even frames is used for signaling. Since this 193rd bit occurs at a rate of 8000 b/s, the signaling rate contained in the even frames is 4000 b/s. This is more than adequate for the 24 voice channels. These techniques apply to multiplexed T type carriers such as the digital trunks used for interconnecting switch centers.

It is also possible to have associated signaling over a separate channel between subscriber terminals and the switch. One example of such a terminal is shown in Figure 37. This

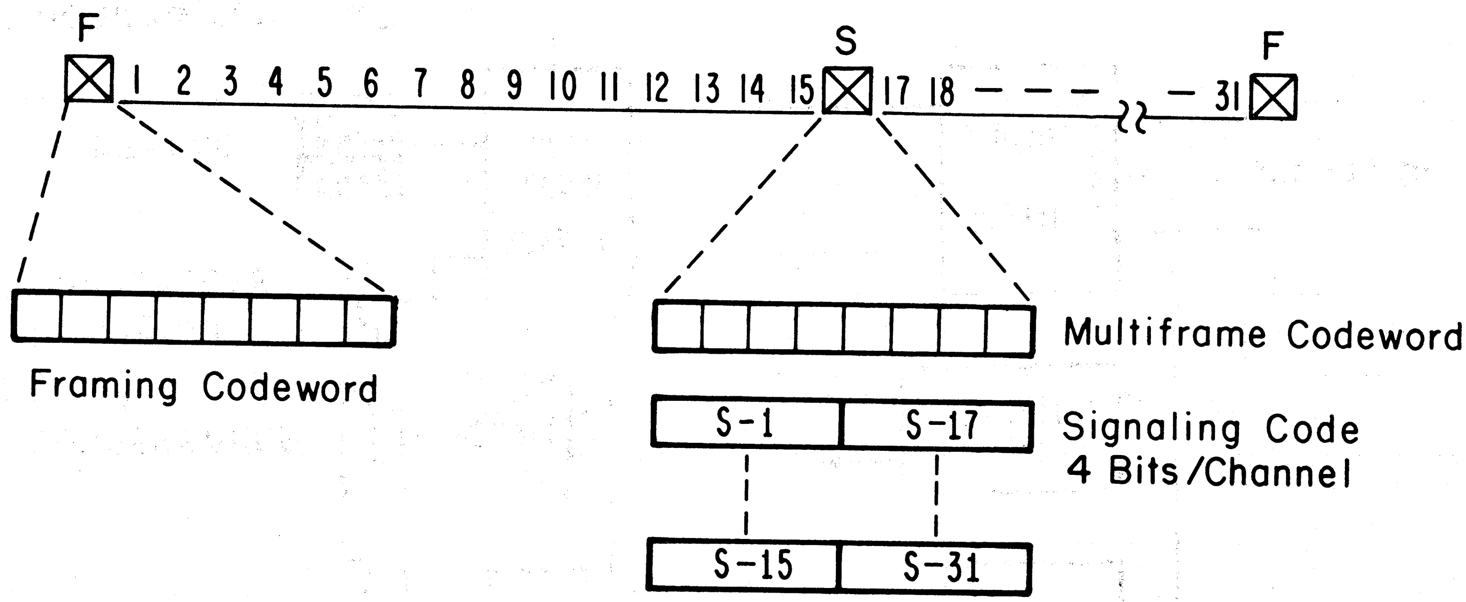


Figure 36. Multiplexed PCM framing and signaling format (European system).

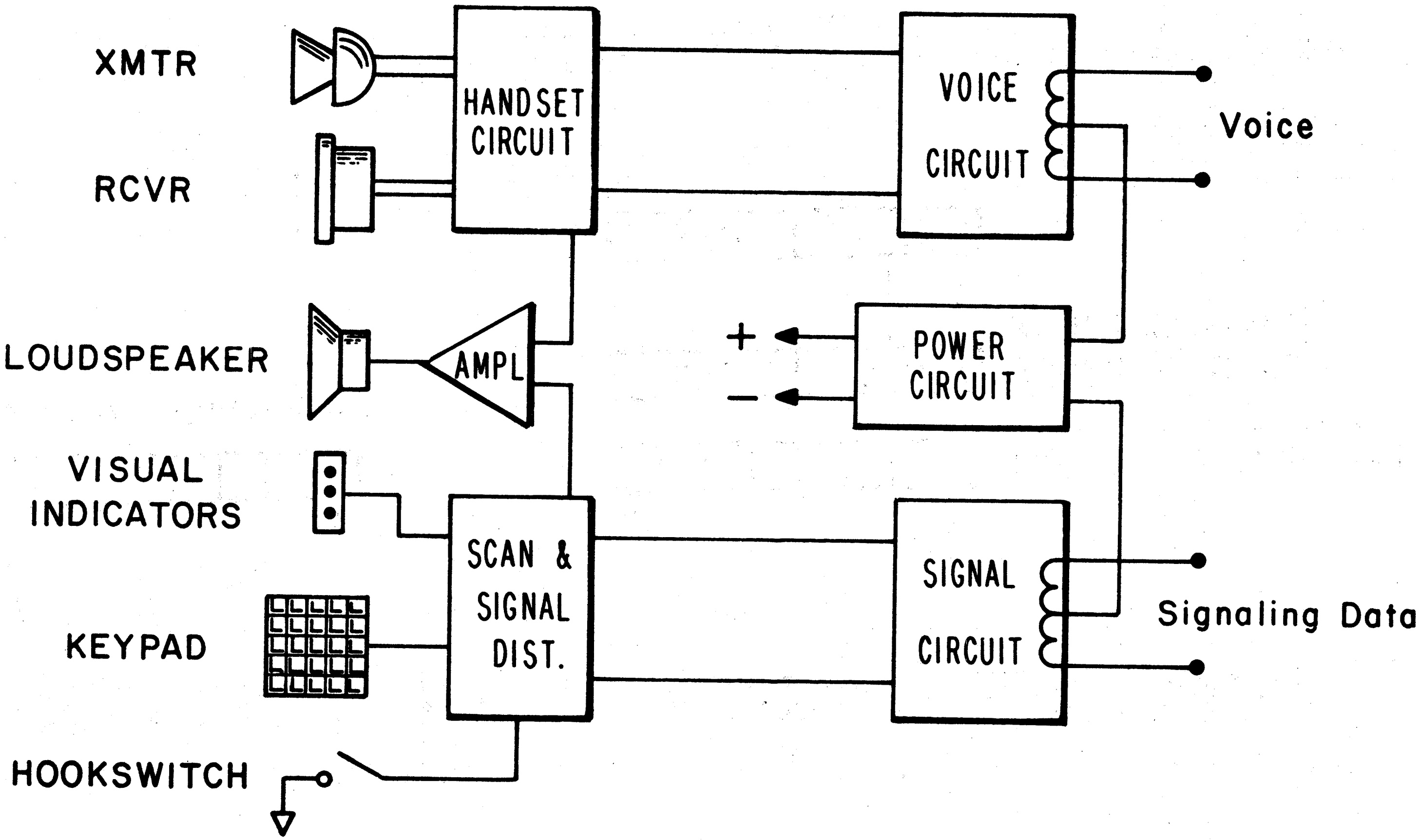


Figure 37. Analog station with quasi-analog signaling.

terminal was developed for use with the SL-1 business communications system (Audette et al., 1975). Voice signals are carried on one analog pair in the same way as for a conventional telephone set. Digital signaling information, including supervision, addressing, and audible control, is carried on a separate analog pair. Such digital signaling operates full duplex with 2400 b/s modems in a quasi-analog mode. Power for the set is derived from the voice and signaling pairs. The scan and signal distribution block in Figure 37 provides the control as follows. This unit sequentially scans the input keys and the hook switch, and passes the resulting digital control signals to the switch via the signaling circuit modem. Return signals, digitized by the modem, are passed to the distributor which activates the correct output device on the set.

<u>Nonassociated Signaling Over Common Channel</u>

This concept resembles a distributed computer network carrying short messages between processor controlled switching centers. Blocks of signaling information called signaling units have been specified internationally by CCITT as Signaling System No. 6 (CCITT, 1973a). In North America one has a similar version, called common channel interoffice signaling (CCIS) (AT&T, 1975). Each signaling unit consists of 28 bits: 20 bits of information and 8 bits for error detection (Ritchie et al., 1977). Several signal units of different types and identified by different headings are required to transmit supervisory, address and circuit routing information in order to establish a long distance call. Eleven such signal units are sent and these are all verified by an acknowledge unit sent in return. The link interface operates full duplex, usually at 2400 bits/s, over 4-kHz analog facilities. Modems are used to match the digital data to the analog facility. It is also possible to transmit the common channel signals over digital facilities. An example of the basic network structure is shown in Figure 38. Redundant links to signal transfer points insure reliability. Digital addressing aspects of CCIS are described in the following section.

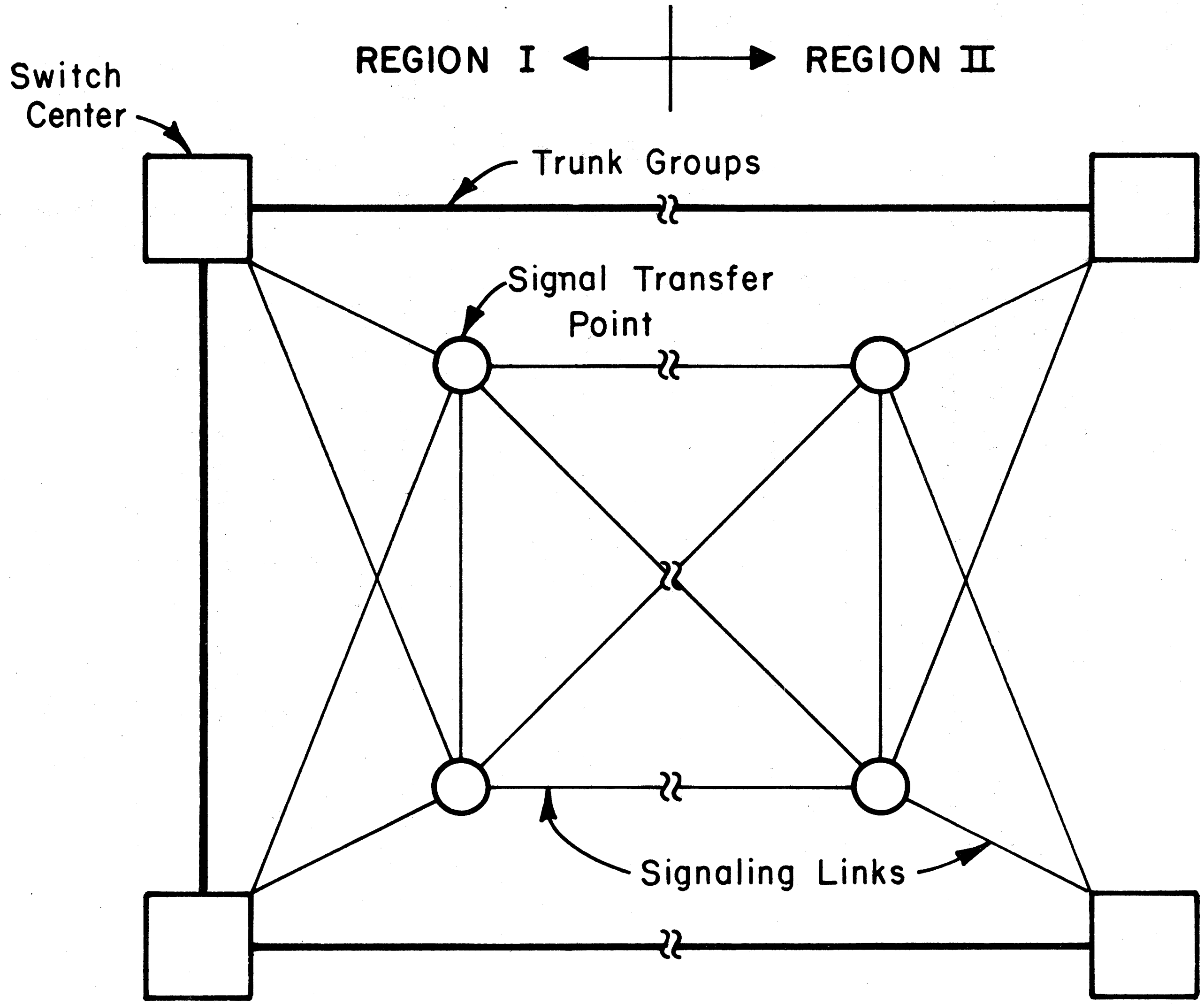


Figure 38. Network structure for nonassociated common channel signaling.

2.4. Addressing

The switch requires address information from the subscriber in order to establish a desired connection. This information consists of a sequence of decimal digits indicating the number (or address) of the called terminal (or station). Various forms of electrical codes and signaling techniques have been developed as automatic switching systems evolved. In the past, major changes have occurred at approximately 20-year intervals (Pitroda, 1977a; Joel, 1977). These major developments are outlined in Table 9 and described below.

Initially dc pulsing was used for address signaling between subscribers' terminals and the switch and between switching centers. These dc pulsing techniques interrupted the current flowing in the loop or derived circuit used for supervision. In the older step-by-step switches, the dc address pulses controlled the selectors on the switch directly.

Later, when motor driven rotary switches and panel switches were developed, indirect control was achieved. A commonly used technique was revertive pulsing, whereby the switch itself generated pulses and returned them to the sender for correlation with the desired position.

The indirect control of the motor driven switch evolved into a common control system when coordinate (crossbar) switches and electronic switches were developed. The common control system quickly completed one connection and then was available to complete another. Common control, with its associated registers and senders, provided the flexibility and speed needed to adapt to, or convert from, any of the earlier pulse type systems. Higher signaling speeds between switching centers were attained using multifrequency (MF) pulsing techniques. In MF pulsing each decimal digit is represented by a pair of tones within the voice bandwidth.

In the 1970's digital switching systems using microprocessors and core memories were been introduced. These systems permit direct switching of digitized (PCM) voice signals without

Table 9. 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

modems. Address signals are transmitted digitally using a common channel to control the connections for many lines and trunks.

What the future entails is difficult to project, but some of the trends are already apparent. The 64 kb/s PCM systems will probably be replaced with new voice digitizing methods operating at lower rates, possibly at 16 kb/s or less. Data traffic will undoubtedly increase and switches handling both voice and data will be developed. Stored program control of time division switches will probably continue, but the controls may be distributed to lower levels in the network using lower cost microprocessors. New services and features will be added as the software evolves. Transmission facilities will become digitized to a much larger extent and will operate at higher rates. For short hauls, the transmission facilities will include fiber optical cables, particularly in local areas of high terminal concentration. The digitization process will ultimately extend to the subscriber's terminal and digital signaling will predominate. Common channel signaling will probably extend throughout the network using both associated and nonassociated links.

The following paragraphs summarize many of the address techniques which have been used. The older systems are gradually being replaced as electronic switch centers are installed, however, many older systems are still in use today. Thus there is a need for compatibility with existing equipment. This compatibility restricts innovations to the switching center architecture (Pearce, 1977). The conventional pulsing codes of today are summarized in Table 10 and explained in the following subsections.

2.4.1. Dial Pulsing

Early step-by-step switches were operated directly using dc pulses generated by a rotary dial. The station dial, operated manually, rotates a cam that opens and closes a set of contacts in series with the hook switch, as shown earlier in Figure 25. A rotary dial opens and closes the loop at approximately 10 times per second. The break period (contact open) is normally 60% of the pulsing period. The time between one dialed digit and the

Table 10. Pulsing Codes Used for Addressing

Digit	Dial Pulses (breaks)	Revertive Pulses (shorts)	PCI (current)	MF (interoffice tones)	MF (push button tones)
0	10	1	O,n,O,n	1300, 1500 Hz	941, 1336 Hz
1	1	2	P,n,O,n	700, 900 Hz	697, 1209 Hz
2	2	3	O,N,O,n	700, 1100 Hz	697, 1336 Hz
3	3	4	P,N,O,n	900, 1100 Hz	697, 1474 Hz
4	4	5	O,n,P,n	700, 1300 Hz	770, 1209 Hz
5	5	6	P,n,O,N	900, 1300 Hz	770, 1336 Hz
6	6	7	O,n,O,N	1100, 1300 Hz	770, 1477 Hz
7	7	8	O,N,O,N	700, 1500 Hz	852, 1209 Hz
8	8	9	P,N,O,N	900, 1500 Hz	852, 1336 Hz
9	9	10	O,n,P,N	1100, 1500 Hz	852, 1477 Hz

next cannot be less than about 600 ms. Dialing time normally averages between 1 and 2 seconds per digit depending on the system and the user's dexterity.

Dial pulsing is still commonly used on many station sets. At the switch the dialed pulses are counted and transferred to registers, or stored in an electronic memory for subsequent control operations, or for signaling via other means on the trunks.

2.4.2. Revertive Pulsing

The revertive pulsing addressing technique was used in older offices for address signaling to other offices. The transmitting (originating) office sends a start pulse which causes a pulse generator in the receiving (terminating) office to send pulses back. These return pulses are counted at the transmit end and, when the desired digit is reached, a stop signal is transmitted.

2.4.3. Panel Call Indicator (PCI)

This is another one of the older addressing techniques. The pulse code is given in Table 7. A positive (P) current along with small (n) and large (N) negative currents are used, in addition to an open (O) line, to transmit the digits. Each digit requires four symbols of about 70 ms duration and therefore the signaling rate is about 4 digits per second.

2.4.4. Direct Current Key Pulsing (DCKP)

This technique was used between manual switch boards and automatic switching centers. A four-state coding technique similar to PCI was generated manually by an operator using one button per digit.

2.4.5. Multifrequency Key Pulsing (MFKP)

This technique is used today in North America on many trunks to transmit address information in the originating-to-terminating direction only. Although two-out-of-five frequencies are used to code digits, a sixth frequency is included for special purpose signals such as start and stop indicators. The six frequencies used are between 700 and 1700 Hz for interoffice signaling and can therefore be carried over any voice channel. The MF signals are usually transmitted in 1 s to 1.5 s bursts and the maximum rate is about 10 digits per second.

MF addressing may also be used for customer dialing with push button sets, although that warrants a different set of frequencies (see Table 7).

The use of SF supervision and MF addressing presently entails a connection time of about 10 seconds, on the average, to connect a long-distance call across the U.S.

2.4.6. Digital Addressing

The basic concept for digital addressing is the same as for digital supervision (see Sec. 2.3.3). Decimal digits for the subscribers number and routing information, which may be stored at the switch, are converted to a binary code and formatted into blocks of signaling units with error-detecting check bits. The transmission line may be either analog or digital. The CCIS system, for example, was developed to transmit the address and routing information digitally over a 4-kHz voice circuit using a 2400 b/s PSK modem. When fully implemented in North America, the CCIS network is expected to be capable of establishing a connection across the U.S. in about 3 seconds on the average. Under normal operating conditions, the nonassociated North American configuration of common channel interoffice signaling can carry approximately 1500 trunks per CCIS link, at the quoted 2400 b/s data rate. The same link can carry double the number of trunks (3000) under failure conditions, i.e., the load of a failed link plus its own normal load (Dalhbom, 1972).

It is possible to send the common channel information over digital links (CCITT, 1973b). One way is to digitize the analog output of the 2400 b/s modem with a PCM voice channel operating at 64 kb/s. Connections from the modem to the PCM transmission channel are made on a four-wire basis to both analog input and output terminals. This indirect method is both inefficient and costly.

More direct methods under consideration include:

1) Asynchronous operation via a 64 kb/s port to a PCM multiplexer. The 2400 b/s data link is derived by sampling the 2400 b/s common channel data stream asynchronously at 64 kb/s. No modem is required.

2) Synchronous operation via a 2 kb/s or 4 kb/s port of the PCM transmission system. These channels, which are submultiples of 8 kb/s, are available using alternate bits of the 193rd bit stream.

2.5. Audibles and Visuals

Specific signals are required on the network to generate audible tones or to operate visual indicators, in order to alert a station, as well as to inform the station on the progress of a call. A common audible alert consists of ringing the bell in the telephone set. (See Fig. 25.) This ringing process is accomplished at the switch by connecting a 20 Hz signal, of about 86 volts rms, across the subscriber's line for intermittent periods, usually 2 seconds on and 4 seconds off. On carrier facilities a 1000 Hz carrier is interrupted at 20 Hz for ringing purposes. When the telephone is answered, the off-hook condition is detected and the ringing signal is disconnected. Similarly, signals may be used to operate visual indicators on loudspeakers on the set.

Call progress tones are returned to a subscriber via the earphone or a loudspeaker in the set to indicate dial ready, line busy, ringback and various other conditions. Commonly used frequencies for these progress tones are indicated in Table 11.

In addition to these tones, it is sometimes necessary to send verbal messages to the subscriber. These messages may be stored in a recorder or voiced by an assistance operator.

Digital tones may be transmitted from a switch hub via T1 transmission lines by injecting the tone in PCM format onto the transmit bus, as shown in Figure 39. The tone generator may actually be a binary data stream stored in memory. It is also feasible to generate PCM analog tone representations by storing only the PCM samples for a single cycle of the waveform, representing either a single frequency tone or a mixed pair. The continuous tone is constructed by repetitively reading the store (Pitroda, 1977b). These tones can also be detected in digital form without digital filter conversion to analog waveforms.

Table 11. Audible Tones Commonly Used in North America

Tone	Frequencies (Hz)	Cadence
Dial	350+440	Continuous
Busy (station)	480+620	0.5s on, 0.5s off
Busy (network congestion)	480+620	0.2s on, 0.3s off
Ring return	440+480	2s on, 4s off
Off hook alert	Multifreq. howl	1s on, 1s off
Recording warning	1400	0.5s on, 15s off
Call waiting	440	0.3s on, 9.7s off

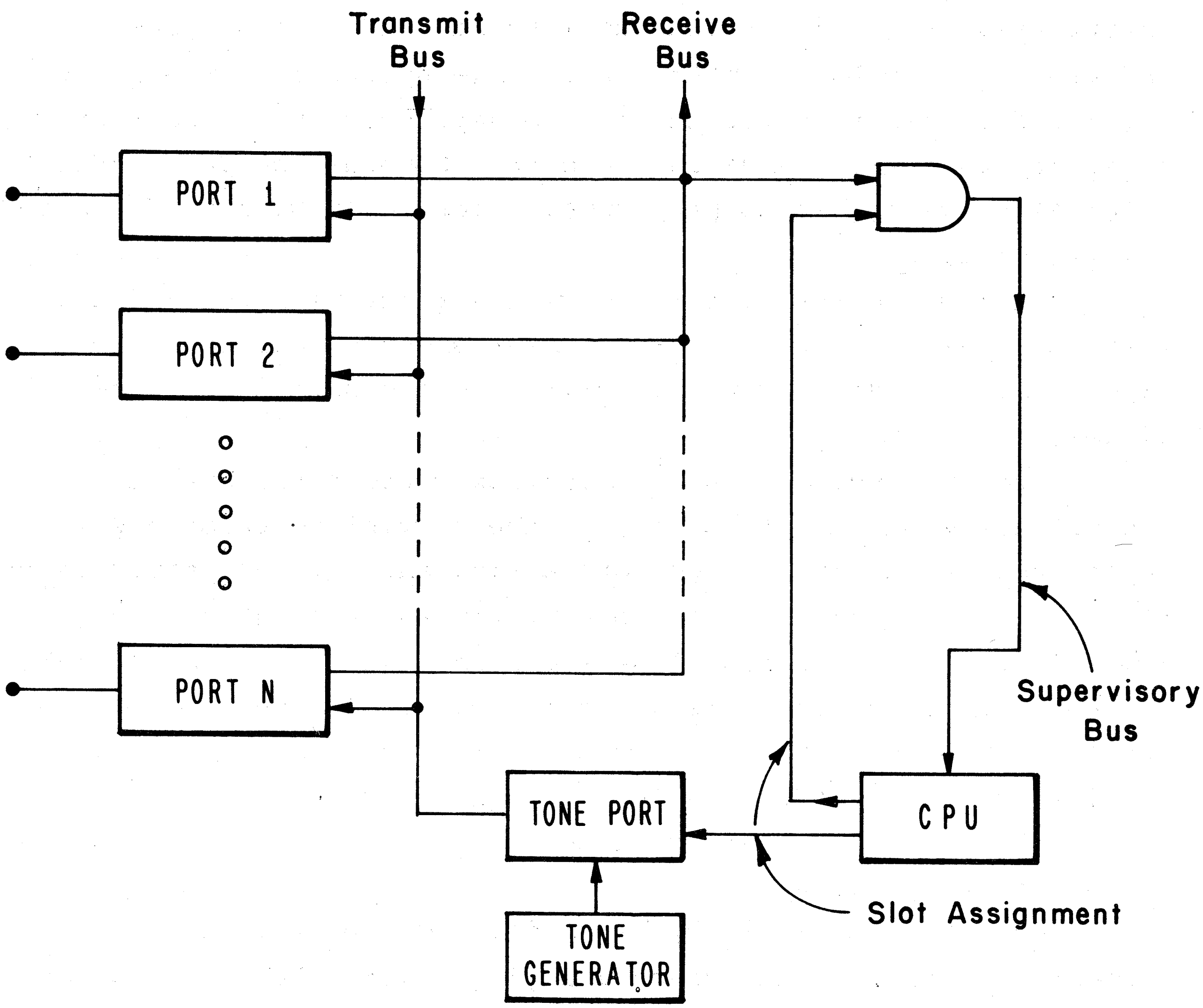


Figure 39. Digital interface for returning audible tones.

2.6. Interface Considerations

In AADSS, as elsewhere, there are two types of signaling interfaces, the device-dependent interface and the device-independent interface. There is a large number of hard- wired terminals requiring different device-dependent interfaces; only a few key ones will be discussed here. Some device-independent interfaces applicable to programmable terminals will be discussed in Section 2.7.

A switch hub must interface with various user terminals, with other switch hubs, and with external hubs, all of which may utilize different types of switching and signaling techniques.

As the DCS network is upgraded, user organizations will undoubtedly wish to retain many of their existing terminals. During the transition period both analog and digital transmission facilities may connect to a hub. Therefore each hub should be compatible with conventional sets and facilities and be adaptable to new electronic terminals which may evolve in the future. This can be accomplished by using separate line and trunk interface modules to serve different kinds of users. Modular interface units can be exchanged and replaced as requirements change and as the upgrading progresses.

Figure 40 indicates, by block diagram, the principal elements required to interface a digital switch hub with analog lines and trunks. In addition to the signaling aspects, the interface must meet other requirements such as battery feed, over voltage protection, and test access. These requirements along with ringing, supervision, and hybrid make up what has been known as BORSHT, or BORSCHT when clocking is required as in a digital interface (McCabe, 1977).

A digital subscriber's line interface block diagram is shown in Figure 41. A similar system is required for digital

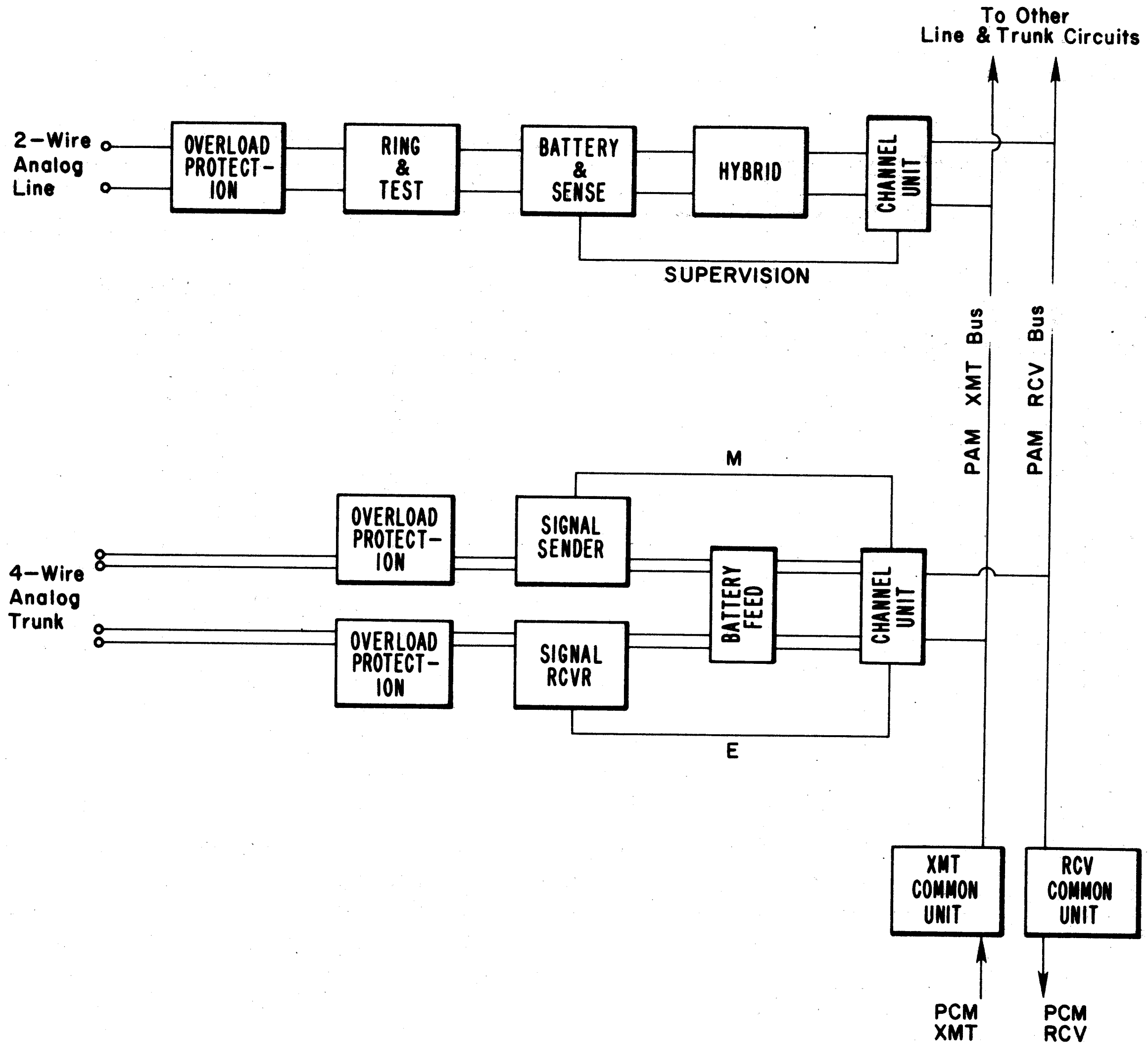


Figure 40. Interfacing analog lines and trunks to digital switch hub.

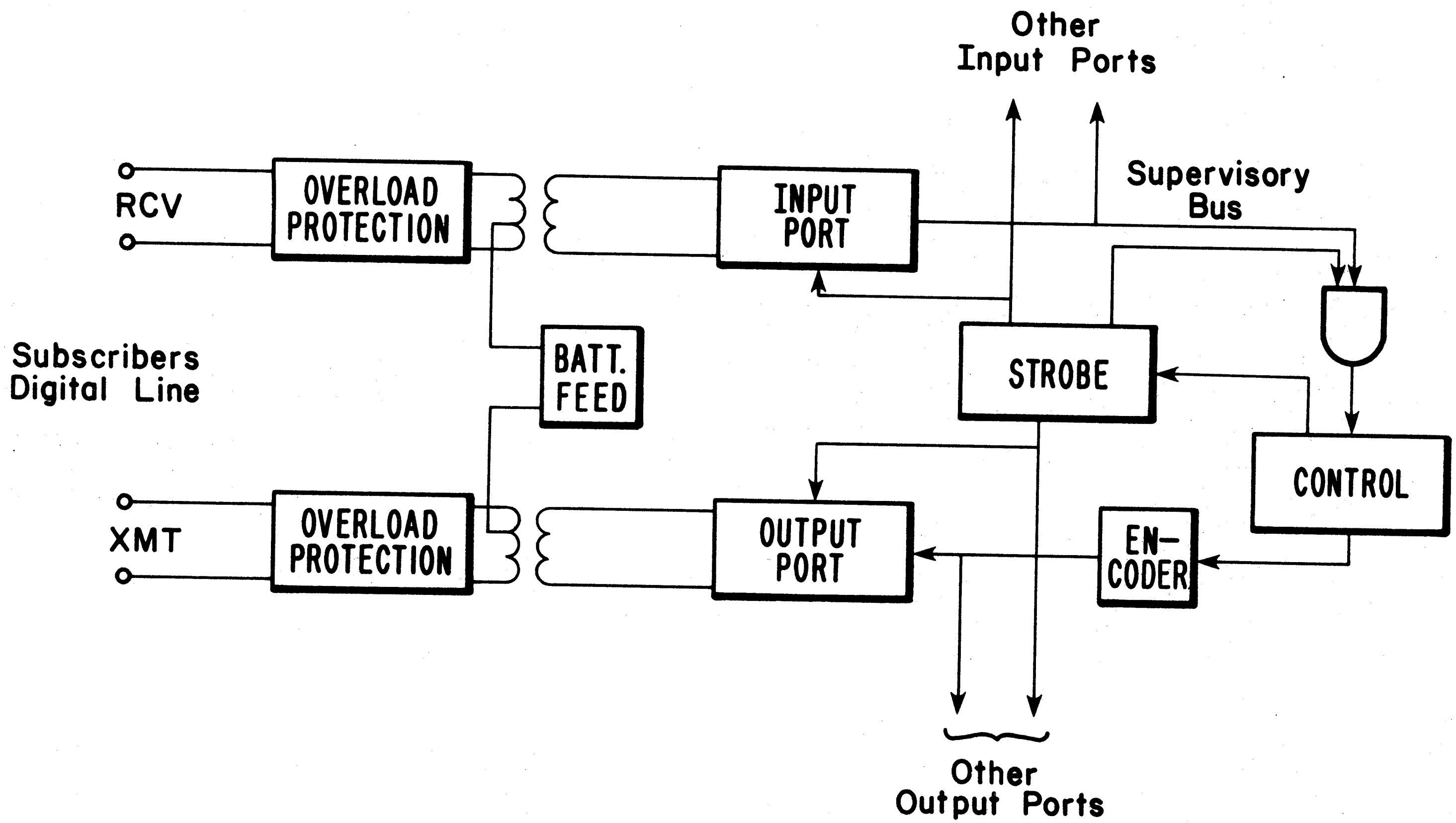


Figure 41. Digital subscribers line interface.

trunks where the battery feed is utilized for furnishing power to digital repeaters along the line.

2.7. Data Terminals and Protocols

The previous sections have primarily been concerned with signaling systems used for supervising and switching voice circuits, with emphasis on techniques used by the common carriers. These facilities can also be adapted to handle data terminals and data transmission. In the case of analog facilities designed for voice transmission, data transmission involves the use of modulators and demodulators (modems) in order to convert digital information into voice band analog signals. This data traffic is then handled just like voice traffic by circuit switching.

Establishing a connection between data modems may involve a telephone handset and the same procedure as for a telephone call. Once the call has been established the telephone set is disconnected and the modem connected to the line. A sequence of tones is sent between the transmit and receive terminals to establish synchronization and to disable echo suppressors. This process constitutes "handshaking" between the terminals and is required before data can be transferred. Some terminals are equipped with automatic dialers for establishing the connection.

The interface between the modem and the terminal requires a multiwire connection whose lead functions and electrical specifications have been fairly well standardized. The standard interface is sufficiently flexible to deal with a variety of modems, terminal designs, and operations. In North America the interface for the synchronous exchange of binary serial data over analog circuits operating up to 20 kb/s is the American Electronic Industry Association (EIA) Standard RS-232. Its international equivalents are CCITT Standards V.24 and V.28. The RS-232C standard specifies the control functions and signal pattern of 25 signal lines including spares, as indicated in Table 12. The table also lists the equivalent V.24 signals and circuit

Table 12. Standard Interface for Serial Binary Data Interchange

Pin #	EIA 232C 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			

designators which are similar but which use a 34 pin connector to include some special assignments, in addition to those in RS-232C.

Recently new standards, RS-449, RS-422, and RS-423 have been introduced by EIA to permit analog operation up to 60 kb/s over unbalanced circuits and 10 Mb/s over balanced circuits (Folts and Cotton, 1977).

The physical interface between data terminal equipment and a digital network is given by a proposed American National Standards Institute (ANSI) standard X.21.

In addition to these physical interface standards, there are logical interface standards which specify the way in which the data are interpreted across the interface to control the link. These logic level control procedures may be either character oriented or bit oriented. The procedures establish frame formats, specify modes of operation, and indicate the commands used in header codes for various operations. The physical level and link control level procedures are required whether the network is circuit switched or not.

Over switched networks, the flow of information via the network may involve a higher logic level protocol which specifies multiplexing logic software, interfacing, and message structures. Thus there are essentially three levels of interfacing protocols which may call for AADS standardization. Higher levels that involve system control and user-to-user communications on an end-to-end basis have not yet been standardized by formal standards organizations.

Table 13 summarizes some of the pertinent protocol levels and lists standards which are recommended for use in interchanging data.

The American National Standards Institute's (ANSI) advanced data communications control procedure (ADCCP), the high-level data-link control (HDLC) and the synchronous data-link control (SDLC) are all similar-level procedures. The ADCCP is divided into three parts: the frame format, the modes of operation, and the operational procedures. The ADCCP frame format is

Table 13. Interfacing Protocols (Data Terminal Equipment)

Level	I		II	III	IV+
Type	Linking		Link Control	Comm. Control	System Control
Purpose	Physical/Elect Interface		Data Exchange with Network	Information Flow via Network	User-User Communications
	(EIA)	(CCITT)			
Standards	RS-232C (20 kb/s)	V.24/V.28	ADCCP (ANSI)	X.25 (CCITT)	
	RS-449/ RS-423 (60 kb/s)	V.26[V.10]*	HDLC (ISO)	(Packet Sw. Network)	
(Existing and Proposed)	RS-449/ RS-422 (10 Mb/s)	V.27[V.11]*	SDLC (IBM)		

*Public Network (Modem)

shown in Figure 42. Autodin II uses this ADCCP format in one operating mode, except that 32 check bits are employed rather than the standard 16 check bits (Sevcik, 1977).

The X.25 interface has recently been adopted as the standard for enabling computers and intelligent terminals to gain access to public packet networks throughout the world. It utilizes the RS-232C standard as well as the HDLC standard for levels I and II and includes a packet level logical interface for level III (Rybczynski et al., 1976).

2.8. Performance Parameters

The system performance seen by a user is dependent on control signaling processes. The purpose here is to define the basic performance parameters from the user's standpoint and to give some numerical values for those parameters which are affected by system signaling techniques.

Information transfer between two users of a network is normally accomplished in three phases, the establishment or access phase, the information transfer or service phase and the disengagement or termination phase[3].

Performance parameters for each phase can be defined in terms of efficiency, accuracy and reliability.

User oriented performance standards for digital communication systems have been proposed by the Federal Telecommunications Standards Committee (FTSC, 1977) and by the American National Standards Institute (ANSI, 1971). Such standards provide a method of describing system performance independent of the network design characteristics. Numerical values are not given for the defined parameters but the standards establish the general criteria which describe the performance of digital telecommunication systems and services. When specific numerical values are

[3]Sometimes the access and termination phases are subdivided into physical and logical phases yielding a total of five phases for the total transfer interval (Schwartz, 1977).

FLAG	ADDRESS	CONTROL	USER DATA	FRAME CHECK	FLAG
01111110	8 BITS	8 BITS	UNSPECIFIED NO. OF BITS	16 BITS	01111110

Figure 42. ADCCP frame format.

realized, comparisons can be made between systems offering equivalent services and yet using different facilities and procedures.

Table 14 lists the performance parameters specified by the proposed Federal Standard 1033. Although these parameters apply directly to digital data transfer, many of them also apply between the digital interfaces on voice systems that use digital transmission and switching facilities. For analog information the parameters defined for the establishment and disengagement phase could apply. However, the parameters for the transfer phase would not.

In the following subsections representative numerical values are given to those performance parameters which are related to signaling. Data were derived from a connection survey conducted on the commercial telephone network in 1969 and 1970 (Bell Laboratories Staff, 1971).

2.8.1. Establishment Phase

The system efficiency during the establishment phase is defined by the access time which includes the conventional speed of service. Over the telephone network, speed of service is defined as the time between an off-hook condition and ringing the called party. This time can be subdivided into dial-tone delay, dialing time, and connect time. The connect time, in turn, is the period between end of dialing and start of ringing. It depends on the number of links in tandem and the signaling technique employed over each link. Access time, as defined in the standard includes speed of service plus the time required to answer a call and begin transferring information.

When a data link is to be established over the circuit, the additional time required for a handshaking routine between data terminals is included in the access time.

Table 15 indicates some typical values of the contributors to access time for establishing connections over the commercial telephone networks. The table does not include the average time required to answer a call.

A user may be denied access to the network for a number of reasons, the main one usually being peak traffic congestion that

Table 14. Performance Table from Proposed Federal Standard 1033

Criterion	Functional Phase		
	Access	User Information Transfer*	Disengagement
Efficiency	Access Time	Block Transfer Rate End-to-End Block Transfer Delay Block Efficiency Block Rate Efficiency	Disengagement Time
Accuracy	Incorrect Access Probability	Block Loss Probability Block Misdelivery Probability Added Block Probability Block Error Probability	-----
Reliability	Access Denial Probability Access Denial Time	Service Time Between Outages Outage Duration (Downtime)	Disengagement Failure Probability

*All block-oriented parameters are to be specified both on a block basis and on a bit basis. The block length used shall be an operator-defined average block length.

Table 15. Representative Values of Contributors to Access Time for Telephone Networks

Dial tone delay	
Light Traffic:	0.1s to 0.5s
Heavy Traffic:	1.0s to 100s
Users' dialing time	
Rotary dial:	10s manual, 8.8s automatic
Push button:	4s manual, 0.7s automatic
Connect time	
Local calls:	1s to 5s
Trunk calls:	10s to 15s
Handshaking time	
	~4s (depends on modem and protocol)

causes blocking. On a telephone network this probability of blocking is referred to as the grade of service. Typical values range from 0.001 to 0.05.

2.8.2. Information Transfer Phase

On voice links the quality of service during the service phase may be measured by subjective tests which indicate intelligibility, recognizability and other subjective measures. For digital voice links the error rate is a contributing factor to the intelligibility. Typical values of error probability for PCM voice signals may range from 10^{-2} to 10^{-5} depending on the line conditioning and the modem used.

On digital links the transfer rate of users information may be considerably less than the actual total data rate because communication control procedures, propagation delay, line turnaround time, and retransmission of erroneous data, all reduce the user's information rate. Grubb and Cotton (1975) derived some transfer rates from connection survey data. For a modem specified at 4800 b/s and operating over the dialed telephone network, the mean transfer rate was about 3600 b/s for a typical block of 10,000 bits. This, of course, varies considerably with the type of modem used.

The propagation delay varies widely depending on the distance and number of links traversed but is typically less than 50 ms on domestic terrestrial (non-satellite) connections (AT&T, 1970).

2.8.3. Disengagement Phase

The time required to disconnect completely a line is important because, until this time element is completed, the facilities cannot be used by others. The disconnection time is typically about 2 seconds on a switched telephone network (Martin, 1976). The probability of unsuccessful attempts to disengage is unknown.

3. SUMMARY, CONCLUSIONS AND RECOMMENDATIONS

3.1. Summary

Telecommunications networks contain several basic elements: terminal equipment to match the user to the network, signaling equipment to control the network, transmission equipment to convey information, and switching equipment to route the information to the desired recipient. The driving force behind the design of these equipments is performance and cost. Performance characteristics include efficiency, accuracy and reliability of information transfer, and other factors unique to the military environment such as survivability, security, and interoperability. Cost factors include cost to purchase, to manufacture, to own, to operate and to maintain.

This report covers two elements of the telecommunications network: switching and signaling. It extends a previous study of documenting seven AADSS system alternatives and goes beyond recommending a selected few. Illustrative descriptions of switching hubs or PABX's are presented first. These hubs employ advanced TDM technology with stored program control suitable for future upgrading towards an all-digital integrated DCS, including regional and local access areas. The combination of digital techniques, stored program control and large-scale integration of electronic circuits provides more features and services to the user and, at the same time, promises a greater potential for future cost savings.

Since the AADSS switching systems must readily adapt to a variety of network environments, the control signaling aspects are emphasized in the second section of this report. The principal concepts employed in establishing and maintaining circuit connections and other message transactions are reviewed, based on known present-day and predictable near-future technology.

Rapid changes in both switching and signaling technologies have occurred in the past and more changes may be expected in the future. Precautions should be taken as the system evolves from all analog to all digital, to insure compatibility of the "new" with the "old" during the transition period and to plan the transition itself.

In the following subsections we conclude by raising several issues which should be resolved before implementing significant parts of the AADSS. Further studies are recommended in the area of traffic engineering and common channel signaling.

3.2. PABX Issues

Section 1 describes various facets of digital PABX principles, design, implementation, operation, and eventual modification. We have neither designed nor identified a single preferred PABX design. Rather, the section outlines several designs that differ in several ways. The resolution of these differences involves a number of issues. We raise these issues next, knowing that to answer these questions may be trivial in certain instances and difficult in others. Some issues may be resolved by future studies, others by industrial developments and military affairs.

We have postulated stored program controlled PABX with time-division switching. Moreover (and this may be challenged) we have outlined a certain attractive switching network structure described as distributed switching with a folded duplex configuration. This configuration is repeated in Figures 3, 6, 13, and 14. It shows the transmission media entering from below, passing through concentration and distribution stages, a common message bus, and returning, while simultaneously their duplex twins traverse the opposite paths. The folded network allows almost unlimited connectivity.

In the common part of the switching network and in the distribution modules, the time-space-time (TST), the space-time-space (STS), and other options, such as STSSTS, or TSSSST, etc., remain largely unexplored. The nonblocking assumption for the common data-bus and the distribution switching modules looks like a reasonable design principle.

Those PABX components that occur on a one-per-line basis are expensive, especially for large installations. Such is the case of concentration stages that eventually interface all line and trunk ports. To be cost effective, these concentrating networks compress the number of signal paths. There are economic benefits for making these networks blocking with a specified grade of service, i.e., probability of blocking. In this study, a blocking probability P=0.01 is assumed for lines (two-way) and P=0.002 for trunks (one-way). There is some doubt whether these numbers are realistic for military bases at the assumed 5 ccs/ line average traffic load. It is also uncertain to what extent the Erlang C formula, which is used here, should or should not be the most appropriate projection for access area PABX services.

On the subject of blocking, it should be established roughly how many series blocking points are preferable for digital and analog concentration hierarchies. Is it better to do the most blocking early (for economic advantage)? Or perhaps late (to take advantage of the law of large numbers). Or to distribute the blocking uniformly (to benefit partly from both)? On a related topic of interface unit occupancy, where a 75% card density is postulated for both lines and trunks, some basis should be deduced for more expedient and perhaps diverse line and trunk packing strategies. Finally, the addition of digital trunks can be done in assorted mixes of individual PCM (64 kb/s) lines or their T1 (1.544 Mb/s) aggregates. The average proportions of these trunk types or their variances between access areas may be beneficial for network designs.

The issue of so called analog-to-digital transition (see Sec. 1.3) appears paramount to the DCS future plans. To emphasize this, we have shown various modular inplementations with different size and type modules (see Figs. 6, 13). There is a question of what size modules (per line, per T1, per T2) offer the most exchangeability advantages. How should sizing of modules be coordinated with remoting of PABX subsystems and what should their relationship be to complete compatibility with industry? Finally, the modularity issue is apt to get involved in the future digital voice development of the disputed 16 kb/s CVSD, other 8 kb/s, or even 4 kb/s alternatives, besides the existing

64 kb/s PCM practice.

Signaling and supervision (S&S) is more than a local matter. Trunks to other PABX's and CO's require interfaces and standard formats, as for CCIS. Common channel signaling may be used within the PABX switching networks as well. There is however a question whether such common signaling channels should extend all the way down to user loops, or only as far as the T1 lines, or just what should their span be?

The last issues pertain to the CPU, its software, and their combined impact on the offered services. For reliable operation, standby CPU units and standby basic memories appear as essential as backup power supplies. Computer hardware and software continue to develop rapidly, raising many questions about their specialized application to PABX control. What, for instance, are suitable separate roles for hardware and software? What areas of overlap are desirable? Software modifications, feature additions, and debugging may appear preferable to hardware overhaul, yet it may still involve considerable effort, time, and expense during the life of a system. Finally, the software is assumed to carry the burden of a long list of offered service features (see Tables 1 and 6). Clearly, some features must be more essential to access area missions, just as some features will cost more than others. At present, information concerning such benefit vs cost tradeoffs for PABX service features appears to be lacking.

3.3. AADSS Signaling Issues

Section 2 describes various signaling functions and techniques available for performing these functions in a switched telecommunications network. A preferred technique for the AADSS has not been identified. In the DCS and common carrier environment, one tends to lean toward some form of common channel signaling between switch hubs with stored program control. Nonassociated common channel signaling (CCIS) is currently being introduced by the common carriers in North America. If common channel signaling is used in the access area, several issues must be resolved; e.g., signal format, associated or nonassociated CCIS network, transfer rates, etc. Whatever scheme is selected, it should be recognized that the military requirements can be distinctly different from non-military networks, due to the nature of the user population and their services. The traffic statistics appear different, special features such as pre-emption and precedence are required, security and survivability are both essential, and terminal changes to new locations are common and frequent.

In-depth questions concerning network signaling can only be answered after further study. For example, issues of network topology affect survivability and routing strategies. Still other issues are concerned with new technology and its impact on AADSS implementation.

Some of these issues are outlined in the following paragraphs. They all relate to signaling aspects of the network and are concerned with voice/data integration, interface compatibility for interoperating systems, network reliability and survivability, signaling security, and future adaptability.

3.3.1. Voice and Data Integration

Ultimately it is expected that common digital facilities could be used to meet most needs for clear and secure voice transmissions, interactive and bulk data transfers, and narrative/record communications that use facsimile-like terminals and graphic displays. In this digital environment it is feasible that different switching functions will be combined. The best that an AADSS network could do would be to adopt automatically the best switching procedure for a given traffic scenario. At any given time the network might use circuit switching to transmit computer files, packet switching for interactive query and response data, and both circuit and packet switching for voice. This integrated system places a high burden on signaling. More functions with more complex formats are apt to require higher signaling rates.

Packet switching requires signaling on a per packet basis. This adds a considerable overhead compared to circuit switching.

Both associated and nonassociated common channel signaling are inherently circuit switching concepts. CCIS is not required for a packet switched network except for setup of circuits, continuity checks, and such secondary tasks as update of routing tables. On the packet level, CCIS does not appear necessary for future packetized digital voice systems. A packet switched circuit, however, can be used for common-channel signaling of circuit switched voice.

It is not clear how digitized voice and data should be mixed, multiplexed or concentrated for transmission and switching. The format for PCM voice (8 bits/sample) is not compatible with normal data transmissions. The PCM signals must either be retimed to alternate with data bits or the data bits retimed into 8 bit words. In either case, the signaling format must be capable of resolving the two types of traffic for switching purposes.

3.3.2. Interface Compatibility

Regardless of how the ultimate AADSS network evolves, it will be required to interoperate with other types of networks such as tactical radio, specialized common carrier and foreign networks. The switch hubs will have to interface with different types of hubs on the trunk side and with different types of terminals on the line side. This is particularly true during the transition period, perhaps 1980-1990, when existing networks with various conventional and upgraded elements will have to face totally new elements in the military base environment and elsewhere.

During this transition period the number of interface variations required can be extremely large and they must be economically and quickly adaptable to change. Standard and special purpose interface devices including buffering units, concentrators, multiplexers, code and speed converters will be needed. They all are affected by signaling formats and procedures.

Table 16 indicates the signaling used in different network environments which may interface a hub (Schreiner et al., 1977).

Table 16. Signaling Used in Different Network Environments

Exchange Type	Transmission System	Signaling
Electro-mechanical • Space Division	Analog Digital	Conventional Channel Associated
Stored Program Control • Space Division	Analog Digital	Conventional Common Channel Associated per Channel Disassociated common channel
Stored Program Control • Time Division	Analog Digital	Associated & Disassociated Common Channel Associated & Disassociated Common Channel
Line Concentrators • Time Division	Digital	Digital Remote Control Channels

One example of the interface compatibility problem concerns digital trunks that carry data to and from a common carrier. At the present time digital voice signals arriving at a space division switch of a common carrier via a T1 or T2 line are converted back to analog prior to switching and reconverted back to PCM for retransmission. Data traffic and signaling information arriving at such a switch could be destroyed by this conversion process if remedial steps are not taken. The external network interfaces must be compatible with the external network processes. Quasi-analog transmission (digital signals on analog circuits) could be used. Figure 43 illustrates potential signaling interface possibilities for the AADSS. It postulates that both analog and digital transmission facilities must exist during the transition period.

Decisions regarding signaling format, procedures and techniques must be made to insure compatibility.

3.3.3. Signaling Network Reliability and Survivability

The performance of the network depends in part on obtaining reliable signaling. For a circuit-switched network this is particularly true during the establishment and disengagement phases. On a packet-switched network signaling affects all phases.

Intentional or unintentional disruption of the signaling channel can incapacitate service over all communication channels involved. If a common channel signaling circuit is used, several tens or hundreds of circuits may be affected. Redundant signal links provide protection but add to the complexity of the signaling process.

Network topology, numbering plans, and routing strategies appear essential in developing sound signaling concepts.

A closely related subject is the monitoring, test and fault location aspects of network control, which also involve signaling formats and procedures. Control strategies may be active or passive, and centralized or decentralized, depending on the role of the control. Some hybrid schemes may be justified for the AADSS.

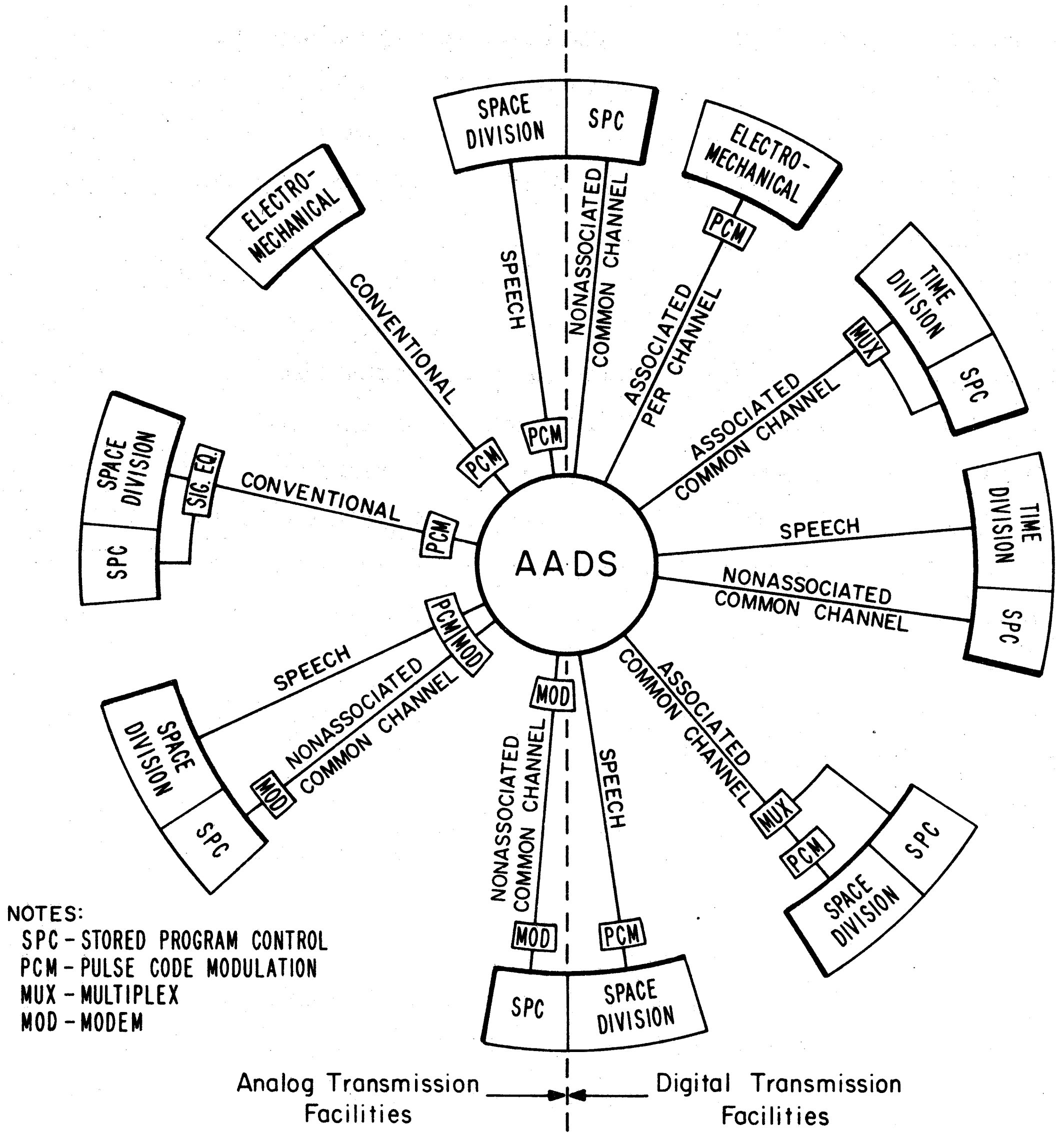


Figure 43. Potential signaling interfaces to access area digital switch for transitional network environment.

3.3.4. Signal Security

The desirability of encrypting signaling information (in addition to the conventional encryption process) is an issue which must be resolved. Signaling encryption adds a considerable amount to the complexity of the network and therefore increases costs. However, it aids the overall network security by preventing disclosure of traffic destinations and other statistics.

3.3.5. Adaptability

The signaling technique should provide a flexibility so it can be adapted to the future needs of the network. For example, it is expected that the ratio of data traffic to voice traffic will increase substantially in the future. New voice digitizing techniques, operating at substantially reduced bit rates, will undoubtedly become more attractive. More switches, PABX's and other elements of the access area will be digitized. Intelligent terminals of advanced design will become available. New features and services will be required.

All of these factors can affect the network signaling requirements. The technique employed should be capable of adapting to the future without major modification, complete replacement, separate additions of hardware, or substantial changes in software.

3.4. Additional Study Areas

This report describes a prototype design for a digital time division switch suitable for use at the nodes of an access area network. These switch nodes are automatically operated by processors with stored programs. The processors in turn communicate with each other and are remotely controled via signaling systems. Traditional techniques for this interprocessor signaling are reviewed in the second half of this report. The more advanced signaling systems are only briefly introduced.

A continuation of this study is currently in progress. Subsequent reports will further define, and attempt to resolve, the complex signaling issues which arise in an integrated voice/ data network that operates in a military switching environment. Several advanced signaling concepts, including common channel interswitch signaling, are being evaluated in detail. When this evaluation is completed, the advantages and disadvantages of all available signaling systems can be compared. Part of the AADSS tasking is to select promising signaling systems in terms of operational effectiveness. Given the most promising signaling systems and network service environments, it may be possible to determine the traffic management capabilities of available processors. Phrased in a different way, this approach should enable AADSS to establish the processors speed and memory requirements for a given system, access network and traffic statistics.

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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.)*

This report covers two key tasks of the Access Area Digital Switch (AADSS) program being conducted by NTIA/ITS for the U.S. Army Communications Systems Agency.

First, a brief introduction to digital electronic private automatic branch exchanges (PABX or EPABX) with stored program control is given, followed by some examples of system design. These examples offer a background, against which AADSS switching and signaling concepts, issues and alternatives can be reviewed. Furthermore, these systems provide integrated interfaces and digital switching to local access areas of the Defense Communications System (DCS). System functions and service features are discussed and initial cost projections given for installation sizes of interest.

Second, a digital and analog signaling techniques of all existing types are reviewed. The main concepts in establishing and maintaining circuit connections and other message transactions are outlined in present day and near future technology. Interface issues during the foreseeable DCS transition from analog to digital integrated systems, as well as other signaling problems in the access area, are summarized.

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