An advanced voice/data telephone switching system

by J. M. Kasson

This paper describes a voice/data circuit-switching system known as the ROLM CBX II. The paper first discusses what the ROLM CBX II family does, describing the most important functions and relating them to popular voice and data applications. The second section describes how the CBX II works, delineating the architecture and giving some details of the system implementation. The final section offers an assessment of what the CBX II and similar products might become in the future.

A *private branch exchange*, or *PBX*, is a communication system that provides telephone service to one or more organizations. PBXs are usually located at users' sites and traditionally provide basic voice switching services for three classes of ports: (1) telephone users; (2) *attendants*, or operators, who perform the manual operations associated with the system; and (3) *trunk lines* (or simply *trunks*) that connect switching facilities to the public and private telephone networks. If trunk lines connect to other PBXs, they are called *tie trunks* or *tie lines*; if they connect to the public switched network, they are called *CO trunks* (the traditional name for the gateway to the public switched network is the *central office* or *central exchange*). The PBX routes incoming calls directly to telephones or to attendants, from which the calls may be extended to station (telephone) users. The PBX provides station-to-station calling without the use of the telephone network, and it allows station users to access the telephone network for outgoing calls. The PBX performs *circuit switching* on the voice signals: it gives the appearance to users that there are connections between pairs of telephones for the duration of the calls.

The first PBXs were called switchboards, and attendants made connections manually, using plug-and-socket patch-panels. Soon after the development of the manual switchboard, inventors developed automatic PBXs that used stepping switches driven by pulses from rotary dial telephones. This technology saw application from the 1890s until well into the 1960s. Crossbar switching became available in the 1930s and flourished into the 1970s. Reed relays and solid-state crosspoints were introduced in the 1960s, as well as time-division multiplexing (TDM).[1] In 1968, the FCC handed down the Carterfone Decision,[2] which began to open much of the telephone industry to competition. The combination of a competitive business environment and low-cost integrated circuits caused increased innovation in the 1970s, as demonstrated by several new switching technologies. Time-division switches employed two types of modulation: (1) *pulse amplitude modulation*, in which the input signal is periodically sampled, and the value of that sample is represented in the value of an analog pulse; and (2) *delta modulation*, which is a digital coding technique in which the input signal is periodically sampled, and the difference between the value of the current sample and the previous one is encoded and transmitted as a single bit. Space-division switches used solid-state crosspoints constructed with pnpn diodes, silicon-controlled rectifiers, bipolar transistors, and field-effect transistors. Introduced in 1975 were the first *pulse code modulation* (PCM) PBXs.[3,4] PCM, in which the digital signal represents the value of the signal at the instant of sampling, has gained wide acceptance and is now the predominant technology for systems larger than one hundred lines.

By necessity, the first automatic systems used distributed-control organizations as their stepping structures, thus integrating both control and switching technologies. Later systems designers specified control architectures based on centralized hardwired relay logic, followed by hardwired solid-state logic, and later by special-purpose computers. As computer technology advanced, general-purpose computer technology proved to be economical for PBX control. As in general-purpose computing, microprocessors are used increasingly for all but the most demanding applications (in this case, the largest PBXs).

Most businesses find that about 75 percent of their telephone expenses consists of charges for network usage and long-distance calls.

The ROLM CBX[5] was first shipped in April 1975 and was the first production pulse code modulation PBX. The original product handled from 100 to 800 telephone users. In 1979, a smaller system that allowed economical configurations as small as 40 lines and a 4000-line product were announced. In 1983, ROLM announced a successor product, the CBX II,[6] based on a new switching architecture that makes possible substantially more switching capacity and affords greater flexibility in allocating this capacity to users.

Features and functions of the CBX family

Presented here are the voice and data capabilities supported by the ROLM CBX, which is called simply the CBX in this paper.

Voice features. When designing the CBX, we looked into voice usage with a view toward introducing functions and features for cost savings. Most businesses find that about 75 percent of their telephone expenses consists of network usage charges, such as CO-trunk costs, and the charges associated with long-distance calls.The CBX provides several mechanisms to control these expenses.

Toll restriction. When using this cost-reduction facility, the system administrator assigns station users to different categories, each with its own set of rules for allowed calls.The CBX can restrict calls by central office prefixes as well as by area codes, to allow control of message-unit charges and long-distance costs.

Call-detail recording (CDR). CDR saves information about outgoing and incoming calls. For all or selected outgoing calls, the system records the calling number, the called number, the time of the call, its duration, an optional account code, and some additional information. From the collected records, local or remote analysis programs produce reports that allow businesses to bill telephone usage to departments or customers, detect abuses, and devise ways to change the network configuration to improve cost efficiency.

Least-cost routing. The CBX provides a routing facility that frees the station user from having to understand the configuration of special transmission facilities, by automatically placing each call over the least-cost service. If the lowest-cost trunks are busy, the system may allow the user access to more expensive facilities, or it may force the user to wait until one of the low-cost services becomes available. Efficient use of the network involves operating the system in such a way that the low-cost facilities are busy nearly all the time. To make this situation palatable to the user, the CBX offers a feature, called *call queuing,* that allows the user accessing a busy facility to wait in a queue with other callers. The system administers the queue on a first-in, first-out basis, dialing the number automatically when the caller reaches the head. If the lowest-cost facilities are not available after a predetermined waiting period, the system allows the call to go out over more expensive services.

System administrators can combine the toll restriction, call queuing, and least-cost routing facilities to allow instantaneous access for certain users to the direct-distance-dialing network, if the least-cost facilities are busy. Other calls would have to queue through a hierarchy of facilities.

Private networks. The CBX can operate with networks of other PBXs connected by leased tielines or earth-satellite facilities. It also works with private switched networks supplied by common carriers or regional operating companies. Many system administrators specify an integrated numbering plan for all sites on the network, allowing station users to dial remote stations in the same way as local extensions.

User features. The CBX offers many features to increase the speed and convenience of using the telephone. If an extension is busy, the caller may ask the system to call back when that number becomes free. People can answer ringing telephones from any location (by number or by group), easily arrange conference calls, and transfer calls to other stations. If a call is in progress when a new call comes in, the system informs the user of the new call and allows alternating conversations with the callers, even on single-line telephones. The user may put a station in do-not-disturb state, whereupon callers hear a special busy signal. The system forwards calls to alternate stations or to message systems if the called station is busy, if it is in the do-not-disturb mode, or if there is no answer. The user may also invoke unconditional forwarding. The CBX provides abbreviated dialing in three forms: (1) the user may save and repeat the last number dialed, (2) the user may select from a private library of numbers, or (3) the user may access a system-wide pool. For internal calls, the system displays the number of the calling station or the name of its principal user, while the phone is ringing.

Voice messaging. PhoneMail® is a computer-controlled voice messaging system that connects to the CBX. It sends and receives voice messages and answers telephones, by digitizing the messages and storing them on disk files.[7] Subscribers to the system retrieve these messages, send replies, forward the messages to their colleagues, or save them for later action from any tone-dial telephone or ROLMphone® telephone. The system guides users through their options with synthesized speech prompts. A communication link integrates the PhoneMail system with the CBX, thereby allowing such enhanced capabilities as flexible telephone answering, message notification, single-digit access, reply to internal callers, and distribution lists.

Specialized applications. In addition to widely used services, the CBX implements some specialized telephone services. One of these is the Automatic Call Distribution (ACD) system, which is useful to businesses in which several agents perform an undifferentiated service for callers, such as airline reservation centers, customer service operations, and catalog sales offices. The system distributes incoming calls to agents and equalizes their workloads. If all the agents are busy, the system plays a suitable recorded message to the caller, and provides music to encourage him to wait. When an agent is available, the caller who has waited the longest is connected. The CBX keeps statistics that allow supervisors and managers to evaluate the performance of both the system and the agents. Many businesses that are too small for stand-alone ACDs can afford an ACD system combined with a PBX.

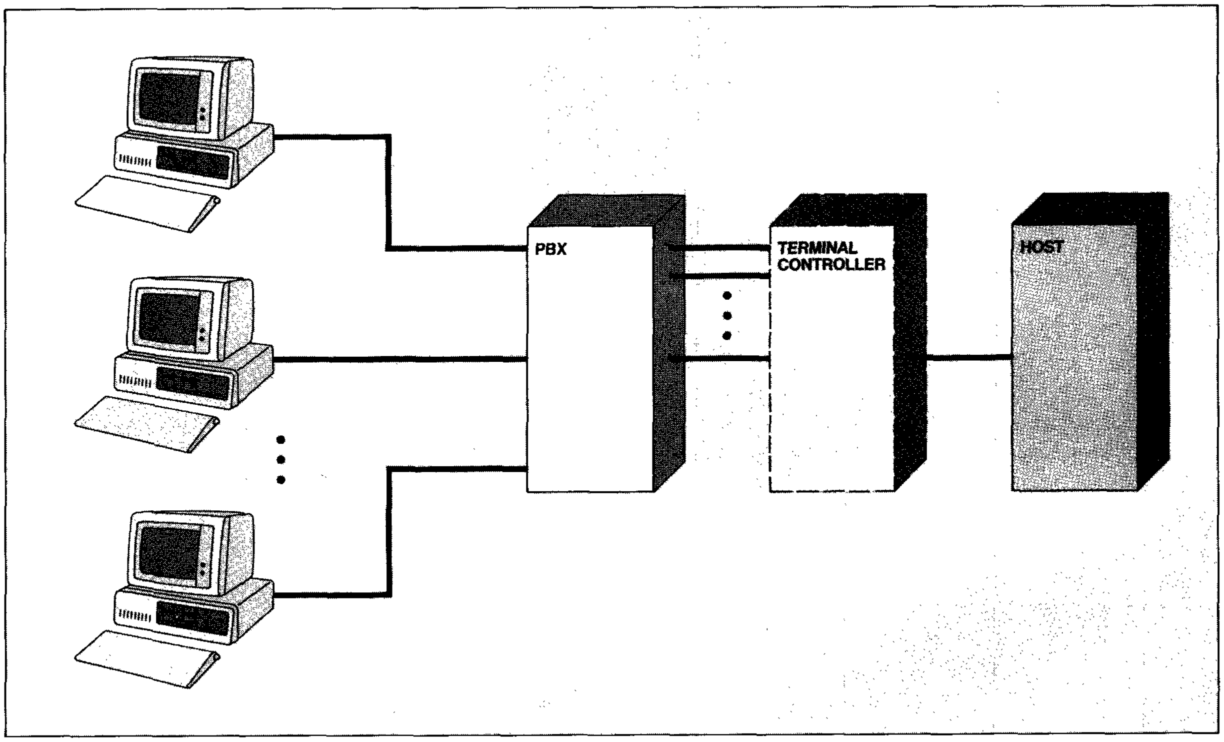
Most new PBXs today are digital circuit switches that switch analog information.

Another specialized system is Centralized Attendant Service (CAS), in which the attendants for a number of PBXs are concentrated at one location called the *CAS center* that connects to *CAS branch* PBXs by tie trunks. When a call comes into a branch, the system routes it over one of the tie trunks to the center. At the center, an attendant answers the call and extends it to an appropriate station in the branch, whereupon the system frees the tieline for the next call. A CAS system requires fewer attendants than would be necessary if each PBX were operated separately. Having all the attendants in one location simplifies administration.

Data features. Most new PBXs today are digital circuit switches that switch analog information by first encoding it into digital form. Many such systems can economically provide local digital circuit switching for data applications, because much of the equipment for data switching is needed for voice communications as well.

In today's office, two popular models for computation exist. In the mainframe interactive (MFI) model,

Figure 1 Local terminal attachment through a PBX



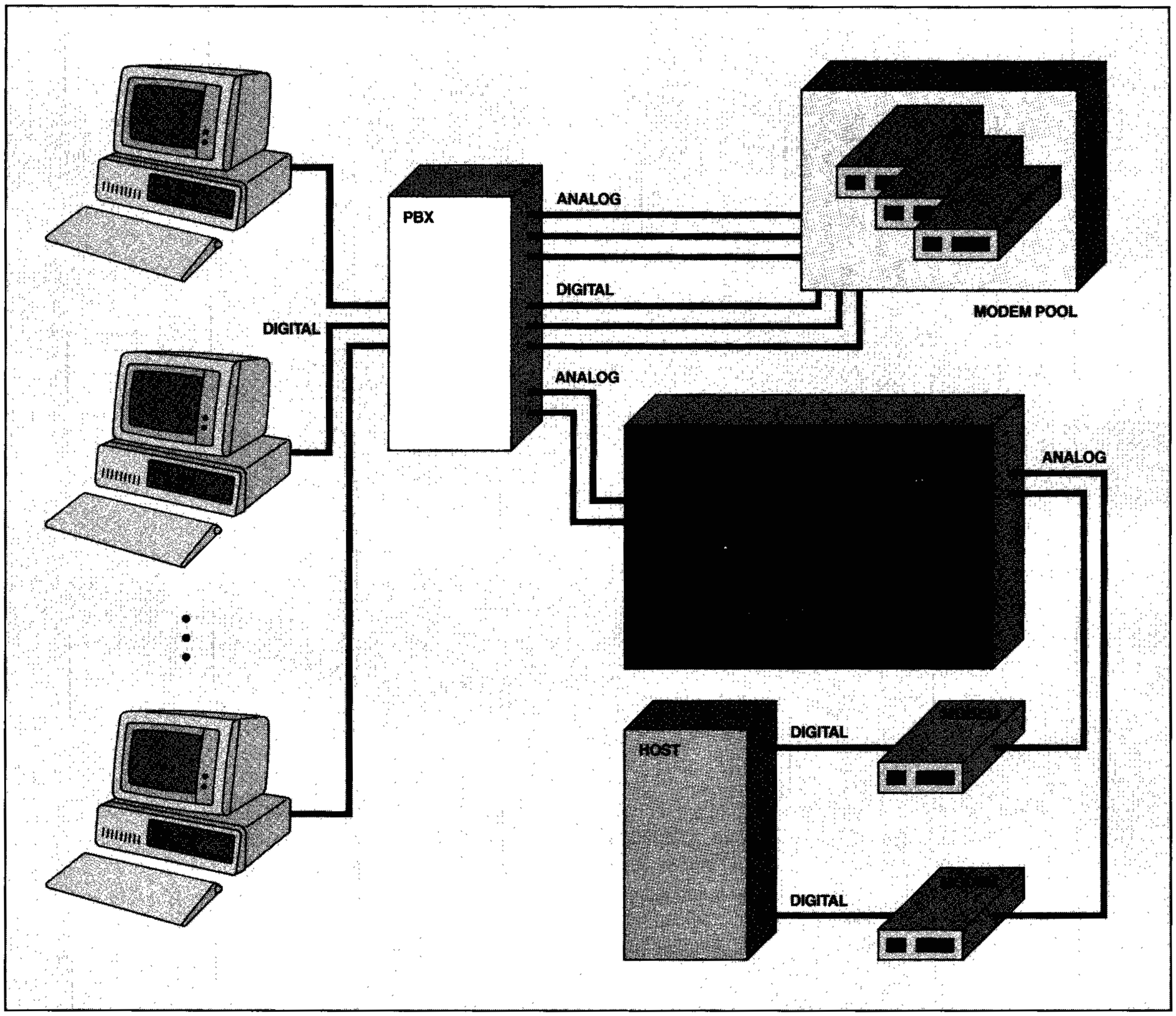
the user interacts with programs running on a time-shared machine, called the *host*, using a terminal of limited intelligence. In the Personal Computer (PC) model, the user programs run on a private, local machine. The MFI model has a long history and a large base of application programs. Indeed, because almost all host application programs were written for the MFI model, personal computers emulate MFI terminals for most local and virtually all remote communication.

If the user spends most of the time interacting with only one local host, the terminal can be cost-effectively wired directly to the host or to a terminal controller that is in turn connected to the host. The wiring from an MFI terminal to the host computer should be part of a uniform wiring scheme, so that changes in terminal locations do not require changes in wiring. The IBM Cabling System, which includes an adapter to allow most IBM 327X terminals to use unshielded telephone twisted-pair wiring, is a comprehensive implementation of such a scheme. If each user uses a terminal substantially less than full time, a switching element, such as the PBX shown in Figure 1, between the terminals and the computer (or terminal controller) allows more users than computer ports. This function, called *port contention*, can provide economic benefits in many circumstances.

A remote host computer allows several alternatives. First, each terminal can have a modem and can use the same PBX facilities as those that provide voice communications to access the remote host through the switched network. Also, a terminal controller can be installed near the terminal users, who communicate with the host through a modem that multiplexes the data from many terminals into one stream to save modem and communications costs. Figure 2 shows that the PBX can connect the user's terminal to a modem, and connect the modem to a trunk. This scheme is called *modem pooling*. Whether this is more economical than the second alternative depends upon data rates and usage patterns.

If the user requires access to application programs that reside on different hosts, there are two main alternatives. If the hosts support a common networking scheme, such as SNA, the user can communicate

Figure 2 Remote terminal access through a PBX

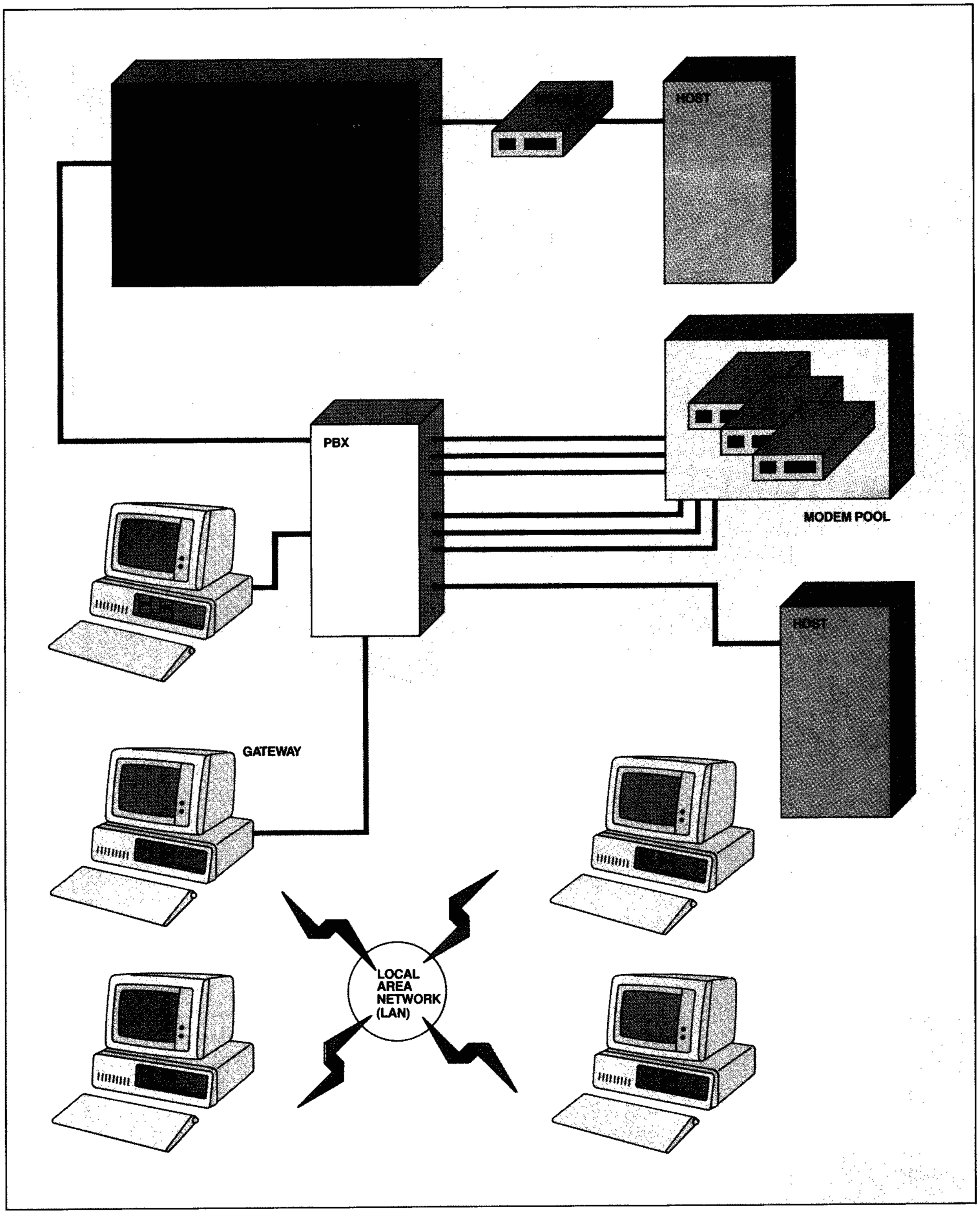


with one host, which then can provide connection to applications in other hosts. This approach can support connection to both local and remote application programs, but it may, at the same time, impose performance penalties and use host resources for the networking. Alternatively, a PBX may be introduced between the terminals and the hosts, possibly with protocol converters at either the host or terminal ends.

In the Personal Computer model, desktop computers often need to share information or peripheral equipment. Both PBXs and local-area networks (LANs) provide the means for such sharing. PBXs are usually limited to data transfer rates of 64K bits per second or less. These rates, although usually adequate for terminal-to-host communications, could impose performance limitations in some machine-to-machine applications. Local-area networks provide increased performance in these applications, because of their ability to deliver a large portion of their aggregate bandwidth to any user for short periods.

PBX connections are usually available at almost any location in a building, whereas smaller areas are usually wired for LANs. PBXs have access to wide-area communications facilities, which are useful to LAN users. PBXs and LANS can thus usefully be connected

Figure 3   PBX/LAN communications through gateway



with *gateways* or protocol converters that connect a circuit-switched network and a LAN. This provides interoperability for devices connected to each type of network, as shown in Figure 3. A LAN-connected user can go through a gateway, emulating an MFI terminal, to access remote or local hosts. A PBX-connected user can access LAN services through a gateway, albeit at lower performance than would be possible by direct connection to the LAN.

CBX data communications capabilities. The CBX provides circuit-switched, point-to-point communications between data devices at rates up to 64K bits

The user can have simultaneous voice and data connections to different destinations.

per second. Data devices may be synchronous or asynchronous, and may use simplex, half-duplex, or full-duplex communications. The circuits through the CBX are transparent to protocol and code set. Attached data devices may be terminals, personal computers (with communication interface and appropriate terminal emulation software), computer ports, modems, statistical multiplexors, protocol converters, LAN gateways, etc.

Desktop devices are connected to digital telephones, which provide standard 25-pin RS-232C connectors. Data Communications Modules (DCMs) in the phone provide signaling for the terminal devices. Computer room applications use rack-mountable versions of the DCM. Figure 4 shows a typical installation, indicating which signals flow on which wires.

The digital telephones multiplex voice and data on a high-speed link, but the two types of signals are switched independently by the CBX. Thus, the user can have simultaneous voice and data connections to different destinations. Every data device attached to the CBX has an extension number. In addition, the system groups extensions which provide an undifferentiated service (such as a set of computer ports attached to a host) and gives them a pilot number and an alphanumeric name. The CBX distributes calls to the extensions in such a data group on a round-robin basis. Users who call a data group that is entirely occupied enter a priority-ordered queue, with calls completing as facilities become available. The CBX keeps statistics on extension utilization, average and peak queue lengths, average time in queue, etc. Using such statistics, a computer center manager can eliminate underutilized equipment or identify oversubscribed services.[8]

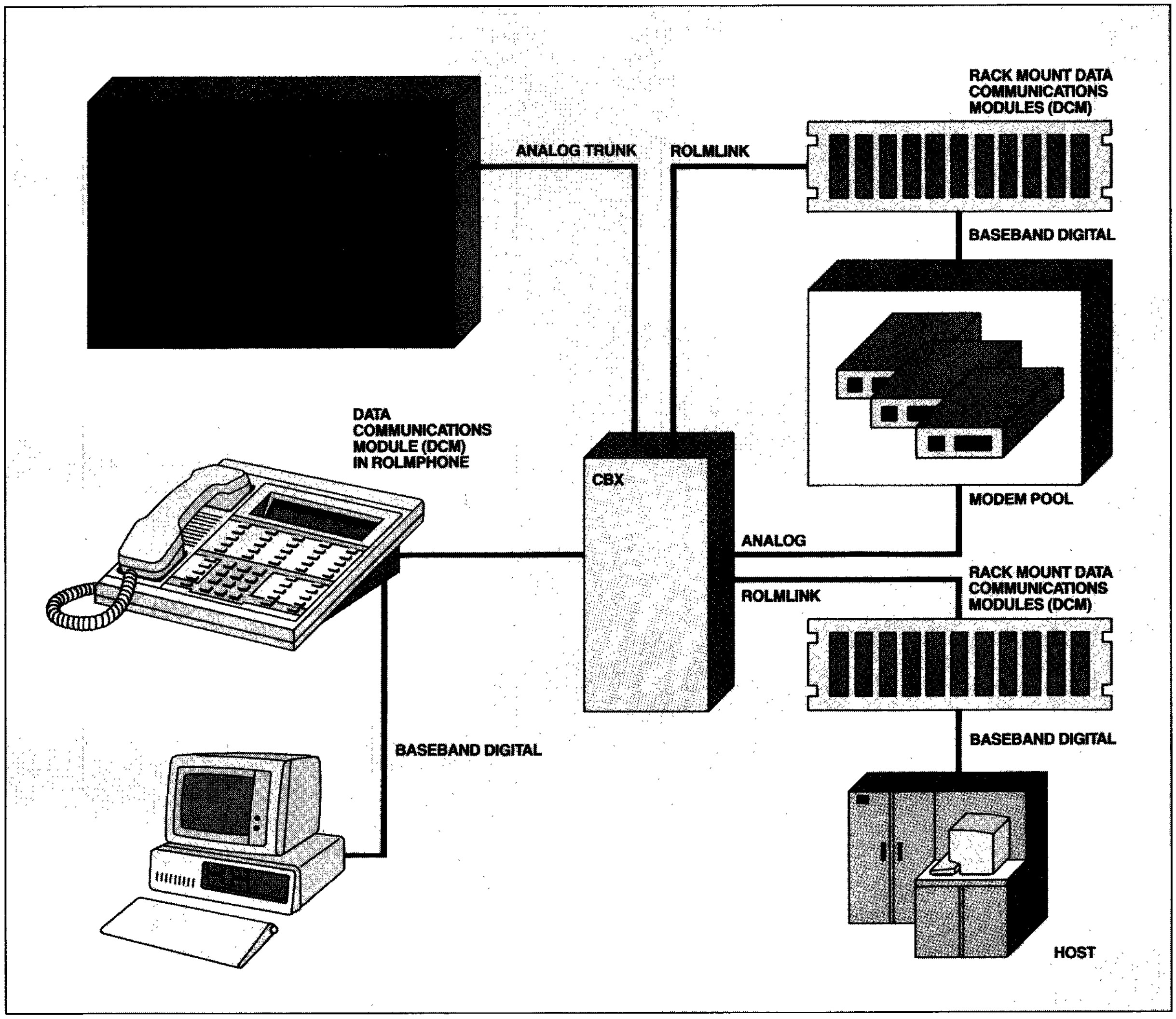
Because a data group may contain equipment with mixed communications characteristics, the CBX performs extensive interoperability checks. When a user calls such a group, the CBX searches the member extensions for a line that matches the characteristics of the calling terminal.

The user may request a data call from the keypad of the telephone or from the terminal itself, through an interactive dialog with the CBX processor. (Only asynchronous terminals using the ASCII code set may use the latter facility.) To establish a call from the phone, the user presses a "data" button followed by an extension number, a data group pilot number, or an external telephone number. In the last case, the CBX searches its pool of modems for one that matches the characteristics of the terminal. A digital circuit is established between the terminal and the modem and an analog connection is established between the modem and a selected trunk. The telephone number supplied by the user is sent out on the trunk. A timer is then started, and the CBX monitors the modem for receipt of carrier. If the carrier is detected before expiration of the timer, the call is considered completed.

When requesting a call from the terminal, the user enters a carriage return character. The CBX uses this character to compute the data rate of the terminal and issues a prompt. The user then enters the extension number, external number, or alphanumeric name of the desired party. When the user queues for a busy resource, the CBX periodically displays the user's current queue position on the terminal. The user may access other services, such as a directory of all of the callable data groups.

The system administrator may connect any two data devices from the CBX's administrative terminal. The advantage of establishing such connections through the CBX as opposed to hardwiring them is that the connections can easily be rerouted to bypass failed equipment, to meet time-of-day peak loads, or to

Figure 4 Digital communications over ROLMlink



reorganize computer room equipment. In this application, the CBX serves as the electronic equivalent of a patch panel.

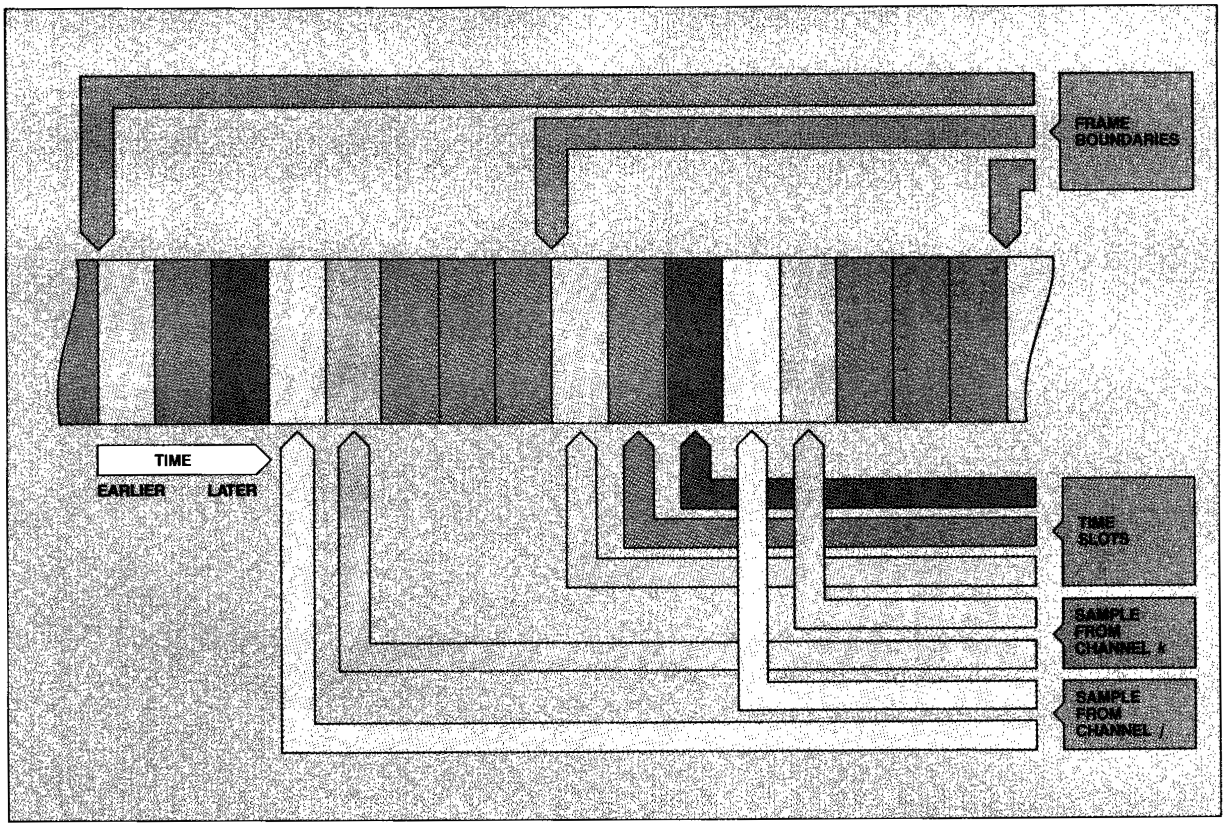
Organizations going from hardwired to switched connections may find themselves vulnerable to additional accidental or malicious breaches of computer security. The CBX can help prevent unauthorized access to computer information by assigning a security class to each terminal attached to the CBX. The CBX data base contains a list of the callable resources associated with each security class and completely hides resources from unauthorized users.

Callable resources can also receive password protection by which the CBX prompts the user for the password before completing the call. Passwords can also be associated with terminal devices. These security features complement the normal host computer security functions.

CBX II implementation

Most modern PBXs are best understood by considering the switching mechanisms separately from the control architecture. This section describes the switching structure first.

Figure 5 Time-division multiplexing



Switching. In *time-division multiplexing*, as conventionally applied to telephone systems, input analog signals are sampled at regular intervals at rates somewhat above twice the highest frequency of interest. The analog samples are converted to digital representations, which are multiplexed into a data stream along with other such digital signals. As shown in Figure 5, the format of the data stream consists of a succession of *frames* that contain a number of *time slots* in which signals can be transmitted. The duration of a frame is the sampling period (i.e., the inverse of the sampling frequency). One time slot in the frame is marked as a reference point or frame boundary, and the other time slots are identified by the delay from the reference time slot. Each active communications path is assigned a time slot in the data stream and transmits its sample in that time slot, once per frame.

Most PCM telephone switches, whether for central exchange or PBX application, cause samples to be time-multiplexed from the input lines or ports into a central switching element that can switch any input time slot to an arbitrary time slot on any output port. This approach has the following two disadvantages: (1) the central switch provides a fixed cost that increases the per-port cost for small systems, and (2) the bandwidth available to each port is fixed. The fixed cost of the central switch can be reduced by designing a family of central switches of varying complexity and capacity. Often the central switches employ multiple switching stages.[9] The fixed bandwidth that is available to each port is the information-carrying capacity of one frame. This capacity can be increased by increasing the number of time slots in one frame, but the link must be designed for the largest capacity required by any interface card, whether that capacity is needed or not.

The CBX II uses a bus-oriented basic switching method. Data from the source card are placed on a high-speed bus and transmitted to the destination

Figure 6 CBX II intranode switching structure

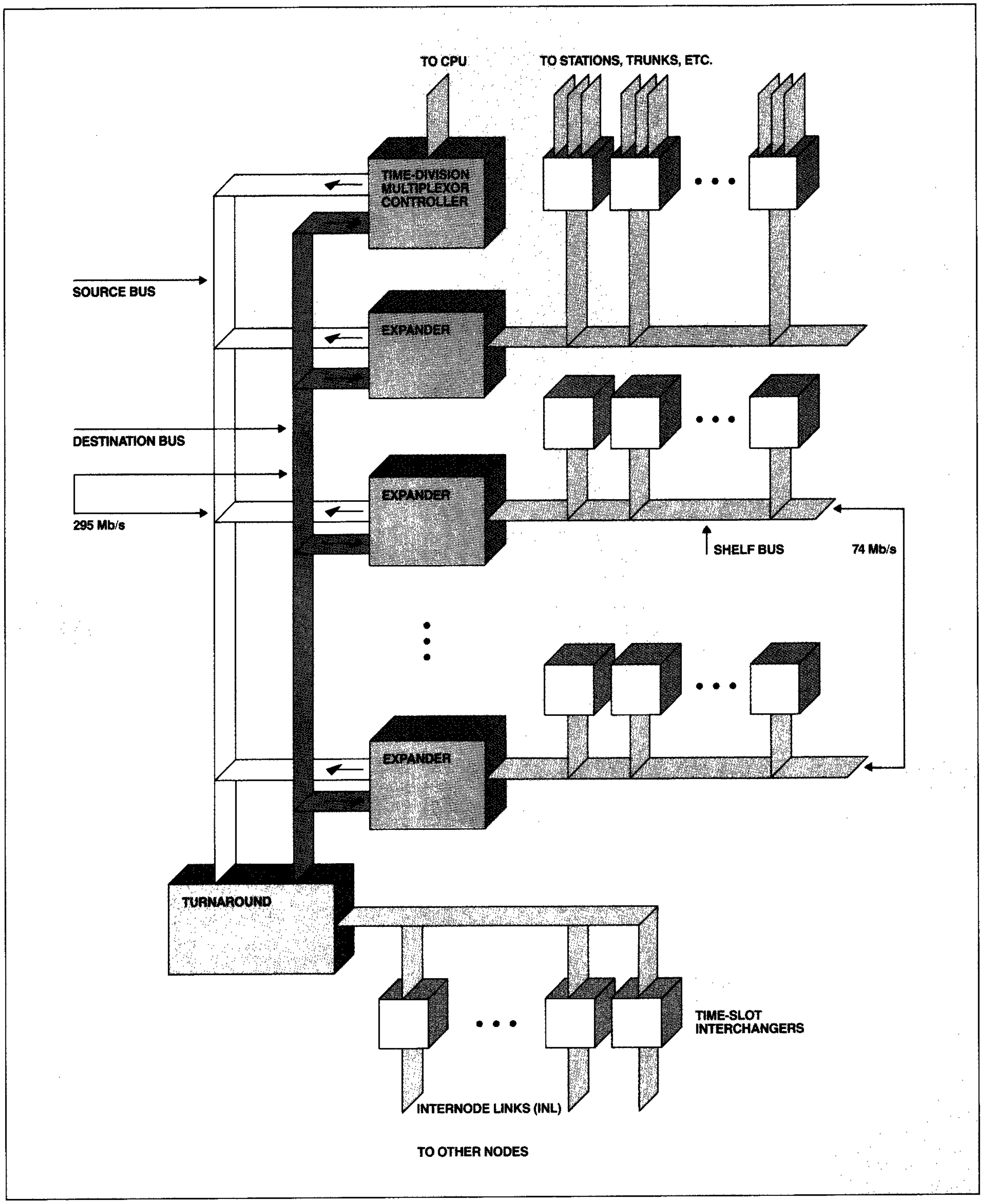
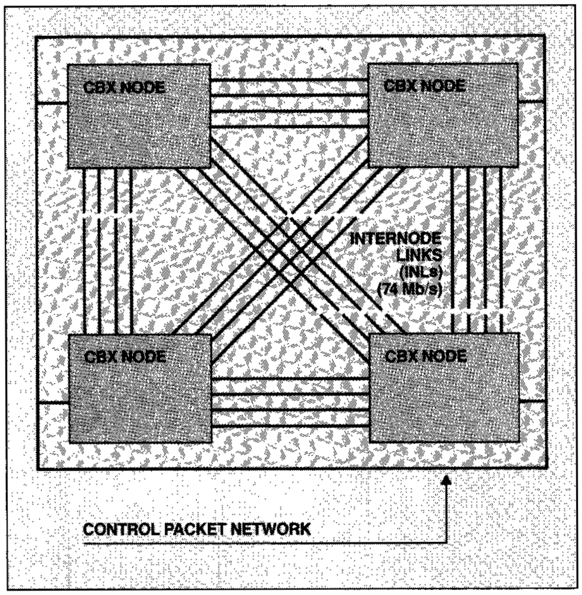


Figure 7 Multiple-node CBX II connection with internode links (INL)



card. Since each card has the entire bandwidth of the bus available to it, this approach provides great flexibility in assigning switching capacity to ports. In theory, it is possible to design a system that is capable of assigning its entire switching capacity to just two ports. For many years, the ROLM CBX family was unique in its use of bus-oriented switching, but recently other switches using this approach have been announced.[10]

In the CBX, six shelves make up a cabinet and one to three side-by-side cabinets form a *switching node.* Each shelf has its own 74M-bit-per-second backplane, and the shelf also contains an *expander* card that passes information between the shelf and system backplanes.

The CBX II provides two alternative system backplanes. In ROLMbus 74 systems, which have an aggregate switching capacity of 74M bits per second, the bus structure is "flat," with all the switched information flowing by every card in the system. In ROLMbus™ 295 systems, which have 295M bits per second switching capacity per switching node, a four-to-one concentration is performed at each shelf, where the 74M-bit-per-second shelf bus is connected to the 295M-bit-per-second intershelf bus. Figure 6 shows how these components fit together.

At the present time, it is not practical to build PBXs with bus structures of greater capacity than about 500M bits per second. Therefore, CBX systems requiring more than 295M-bit-per-second switching capacity are constructed by joining up to fifteen nodes through time-slot interchangers, which allow the assignment of time slots in each node independently of the assignments in the other nodes. For internode communication, the CBX II uses fiber optic *internode links* (INLs), each of which can carry up to 74M bits per second over distances up to 12000 meters, as shown in Figure 7.

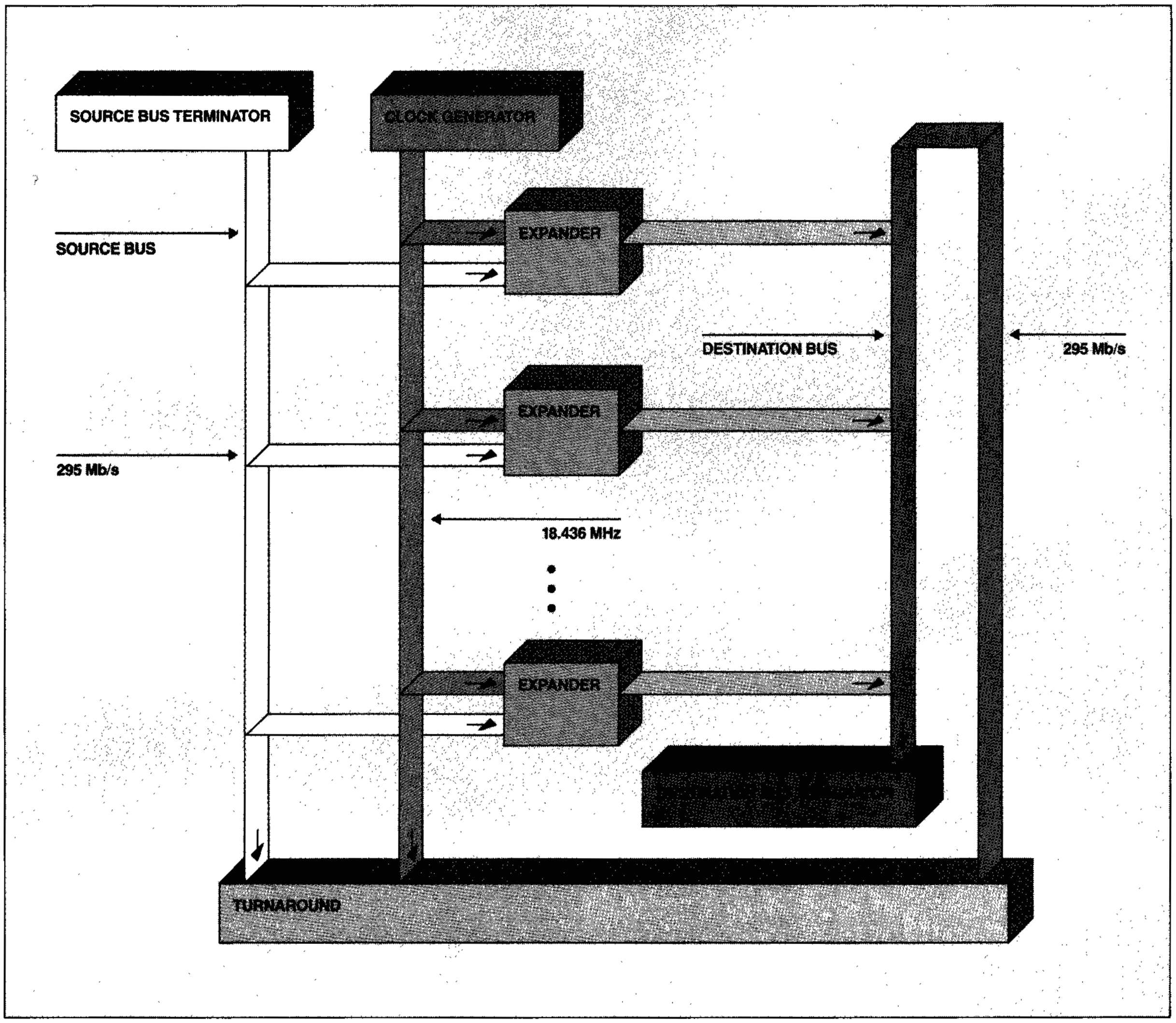
The interface shelf. Twisted-pair wiring brings all voice and data signals to the system. This wiring terminates in circuit cards that are arranged in interface shelves. The CBX II interface shelf contains interface cards and shared resources. Each interface card has two connectors: (1) the line interface connector connects to telephone and data equipment; and (2) the second connector goes to the data bus, carrying voice samples, data, and control information to the card. The expander card is used to exchange data and control information with other shelves. There are two different types of expander cards, one for ROLMbus 74 and another for ROLMbus 295.

The digital bus allows resources to be shared over many ports and to be accessed from anywhere in the system. The circuitry implementing these functions is packaged on cards that interface only to the backplane. One card generates call progress tones and dual-tone multifrequency tones using digital signal processing. Two multiplexed digital signal processing cards implement the conferencing circuitry. DTMF registers, which convert the analog tone signaling to a format the system controller can handle, are packaged four to a card. Dial-pulse registers and senders occur in groups of 16. For increased reliability, the system allows for the duplication of all shared resources.

ROLMbus 74 and 295. The capacity (74M bits per second) and clock speed (4.608 MHz) of ROLMbus 74 are identical to those of an interface shelf. The controller for ROLMbus 74 continuously supplies the expanders with addressing information that specifies which devices are to communicate in what time slots.

ROLMbus 74 is about 38 feet long, which is about one-quarter wavelength of the 4.608-MHz clock. The systems designers could not substantially increase the system capacity simply by increasing the clock speed. Instead they used a new transmission technique. ROLMbus 295 is a unidirectional, trav-

Figure 8 ROLMbus 295 detail



eling-wave bus that consists of *source* and *destination* buses.

In ROLMbus 295, the *turnaround* card acts as a repeater to isolate the source bus from the destination bus and generates clock signals that propagate along the destination bus, as illustrated in Figure 8. Each expander uses the clock for all timing. Therefore, although the expanders are all out of phase with one another, they maintain the correct phase relationship with the data, which travel along the bus in the same direction as the clock. The source shelf expander transmits a data word onto the source bus, which propagates it to the turnaround.The turnaround repeats the word back onto the destination bus, from which it is captured by the expander on the destination shelf.

In multinode systems, the turnaround allows efficient use of the backplane transmission capacity by placing information from other nodes on the destination bus. Suppose a call is in progress between a local node with phone X and a remote node with phone Y. When X transmits a voice sample, it flows out onto the local source bus and down to the turnaround, which captures that sample and sends

Table 1 System software components

System	Thousands of Lines of Code	System Memory
8000	200	1M byte
9000	600	4M byte

it directly to the remote node. The empty destination bus cycle is then filled in with a sample from *Y*.

Other switching modes. The CBX II has architectural provisions for three new data-handling modes, which allow partitioning of the ROLMbus 295 capacity into noninteracting segments of *shared access groups, broadcast groups, supermultiplexed channels,* and ordinary data or voice connections. By using supermultiplexing, blocks of transmission capacity can be allocated to provide full-duplex switching services at speeds up to 37M bits per second. Shared-access mode provides common broadcast capabilities to a number of cards and allows for access arbitration independently of normal call processing mechanisms. Shared-access mode uses special commands to instruct a group of device interfaces to share some part of the system transmission capacity. Broadcast mode allows one device to transmit to a number of listeners simultaneously. This mode uses special commands in order to instruct a number of cards to read data from a single source simultaneously.

Control subsystem. Each CBX II node uses a hierarchical control structure in which a single call-processing computer controls each node communicating with the interface cards over the same bus structure used for switching, as shown in Figure 9. Most of the interface cards carry out the signaling protocols with microprocessors to simplify the task of the main computer. In multinode systems, the system uses a peer-to-peer distributed processing arrangement in which the nodes communicate over a packet-switched network. Each node retains the intelligence it needs for local call processing, whereas the network of controllers share common processing tasks. To improve reliability, there are two independent packet networks, and each CPU connects to both. In geographically distributed systems, the internode link carries the packet network.

The CPU, TDM Network Controller, ROLMbus, and one expander for each shelf form the control subsystem. Systems with common control redundancy have duplicated the entire control structure of a node, including the CPU, ROLMbus, and expanders. Only one common control side is active at a time. The standby side runs diagnostics and keeps itself updated on connections and other aspects of the system status should it be required to take over.

The control processor can be either the 8000 (16-bit) processor or the 9000 (32-bit) processor. The 8000 and 9000 systems have different processors and system software. The 8000 processor is a 16-bit design, and its software is written entirely in assembly language. The 8000 systems operate in single-node configurations only, and they have restrictions on the number of lines supported.

The 9000 system, on the other hand, is designed to support multinode configurations. The 9000 processor is a 32-bit design. One 9000 processor design objective was to incorporate specialized instructions to assist in processing telephony applications. These applications include hardware support for the manipulation of nibble and bit items and linked-list operations. The 9000 software is written entirely in a ROLM-designed language that combines features of the C and Pascal languages.

Table 1 shows the size of the 8000 and 9000 systems, both in number of lines of code in each, in the system memory requirements, and in the total line sizes supported by each system.

The 9000 software is made up of the following component subsystems:

- *Telephony application and data switching.* The telephone application and data-switching subsystem is the largest software component in the 9000 processor system and implements all call-processing and data-switching features.
- *Front-end processor (FEP).* The FEP is the only software component that interfaces with the TDM network. It senses events, such as a station going off-hook, and sends messages to the telephony application software indicating this state change. It also receives requests from the application software and makes state changes in the TDM network. These changes include ringing phones, sending call progress tones, and making connections.
- *Configuration.* The configuration system lets the system administrator make changes in the configuration of the system. These changes include adding, moving, or deleting stations and trunks, or changing their characteristics. The configuration

Figure 9 CBX II control subsystem

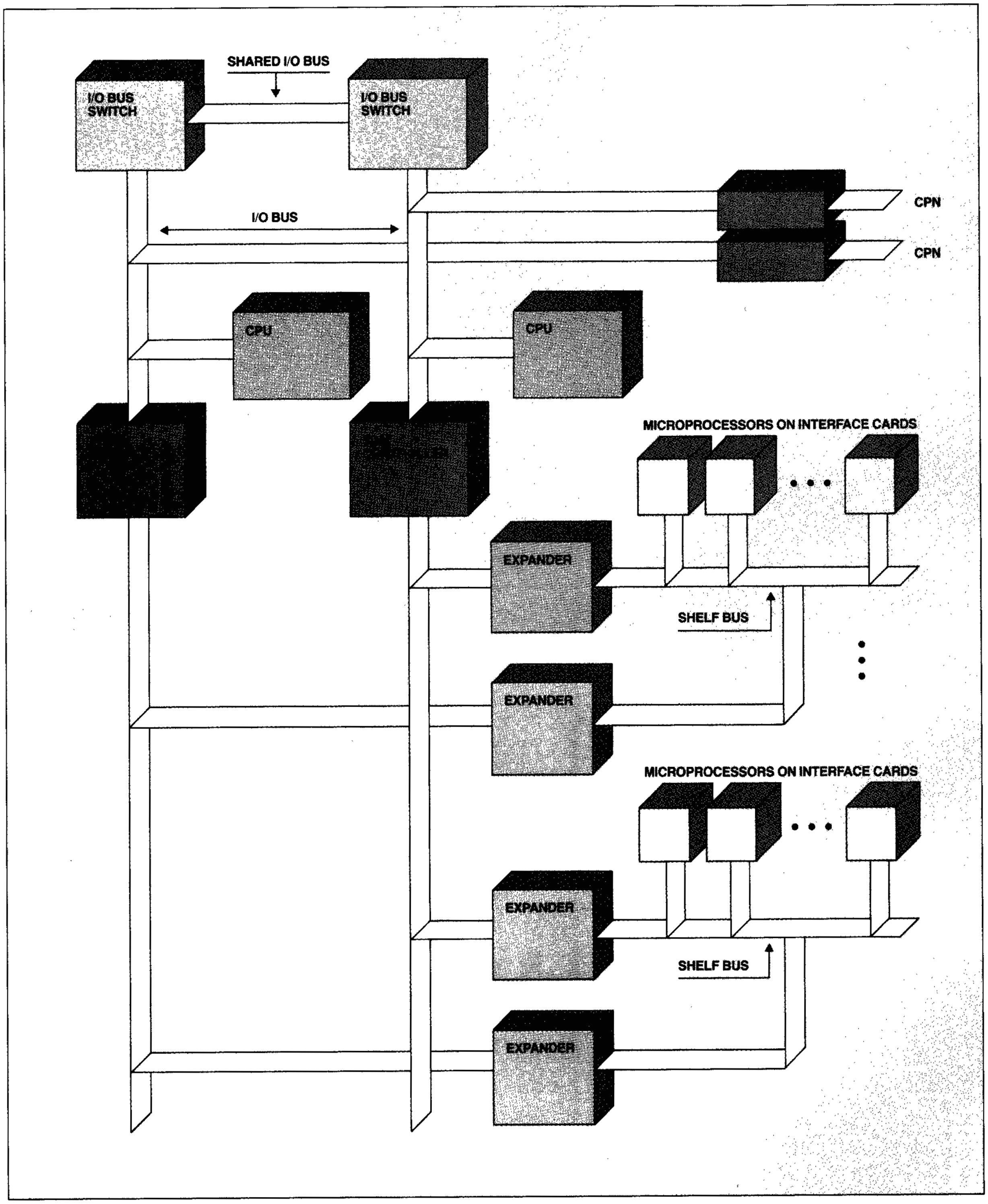


Figure 10 Software control of a telephone call: Phase 1

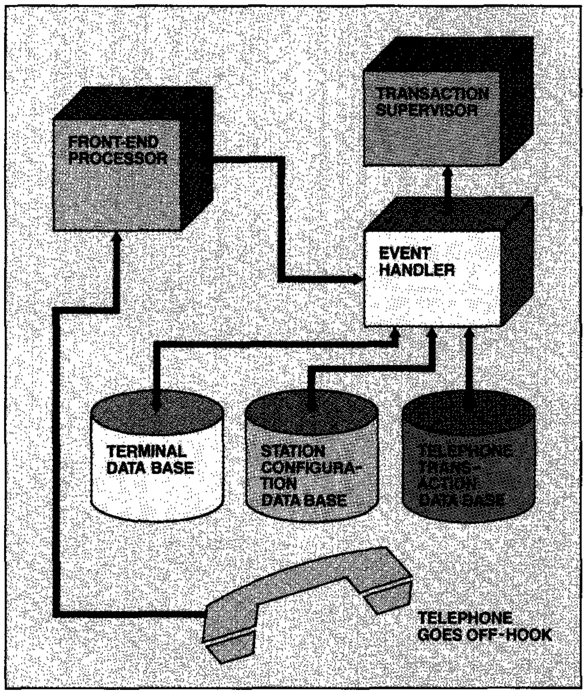
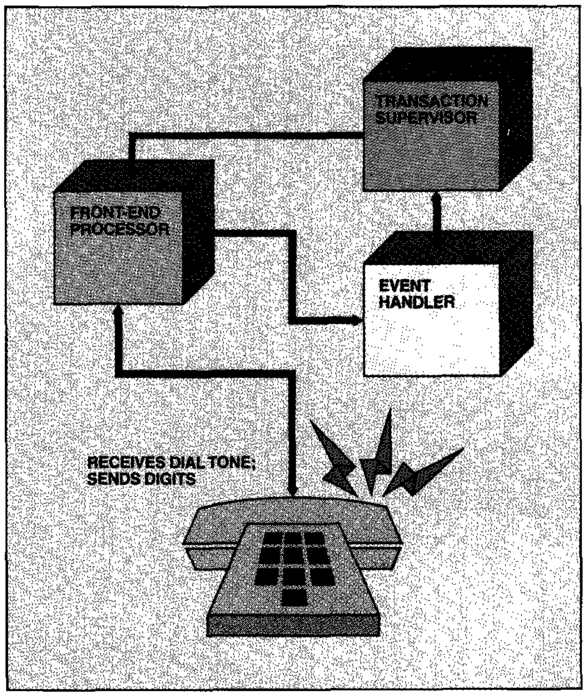


Figure 11 Software control of a telephone call: Phase 2



system maintains this information in data bases that the remainder of the system consults to complete calls.

- *Operating and file system.* The operating system provides a series of primitives that the remainder of the software uses to access system-level facilities. These facilities include the allocation of memory and the scheduling of tasks. The file system provides software access to storage on fixed- and flexible-disk devices.
- *System integrity.* A system integrity component performs the testing of both hardware and software subsystems.

Placing a call. The 9000 processor system is an event-driven system. That is, the 9000 processor detects an event happening in the TDM network and responds to it. The following example is for an internal call placed between two stations on a single node. Outgoing calls and calls between stations on different nodes follow the same general flow, but require additional processing steps.

Phase 1. The starting of the call. See Figure 10.

1. A call begins when a telephone goes off-hook. The front-end processor (FEP) detects this event at a particular physical location within the switch.
2. The front-end processor sends a message to the applications event handler. The event handler performs physical-to-logical address translation, and, hence, it is involved in all communications from the FEP to the applications software.
3. The event handler next looks up this station's address in several data bases. It determines the station's extension number from the station configuration data base and the characteristics of the device from the terminal data base. The event handler ensures that the extension is not involved in another transaction by consulting the telephone transaction data base.
4. The event handler then creates a transaction record in which important information about the call resides. It also creates a transaction supervisor task to handle subsequent steps in completing the call. The event handler passes control to the transaction supervisor via an operating system (OS) call.

Phase 2. Getting the dialing string. See Figure 11.

1. The transaction supervisor instructs the FEP to send the dial tone to the station and to collect dialed digits. It does not interact with the event

handler for this communication. The event handler passes a number (the station physical address) to the transaction supervisor to permit direct communication from the transaction supervisor to the FEP. The reverse communication path does not happen. After this (and all) calls on the FEP, the transaction supervisor suspends, waiting for a digit or another event that causes the event handler to reschedule it.
2. The FEP sends the dialed digits to the event handler. The event handler schedules the transaction supervisor to parse each digit as it comes in and to determine what kind of call this is. When it receives a parsable stream, the transaction supervisor engages the appropriate call-processing software to process the call.

Phase 3. Can the call be completed? See Figure 12.

1. The transaction supervisor requests the FEP to disconnect all common equipment, such as tone generators and digit collection registers.
2. The transaction supervisor determines from the class-of-service data base whether the phone is authorized to make the call requested. If so, it then determines the target extension of the called party from the extension data base, and determines whether the target extension is involved in another call, using the station transaction data base. If the transaction supervisor cannot complete the call, it requests the FEP to send appropriate tones to the station.

Phase 4. Completing the call. See Figure 13.

1. If the call can be completed, the transaction supervisor requests the FEP to ring the called phone and to give ringback to the calling phone.
2. If the called phone goes off-hook, the FEP sends a message to the event handler. The event handler reschedules the transaction supervisor and passes this information to it. The transaction supervisor requests the FEP to terminate the ring and ringback and to make a two-way connection. The transaction supervisor then exits the system through an OS call.

Transaction supervisors do not persist until the call terminates, because calls last for a relatively long and indeterminate time. The system creates a separate task to tear down the call when the call terminates.

Figure 12 Software control of a telephone call: Phase 3

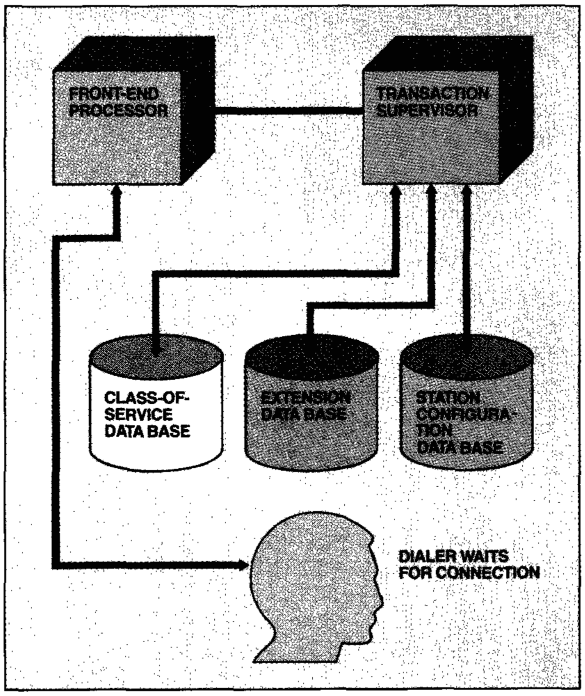
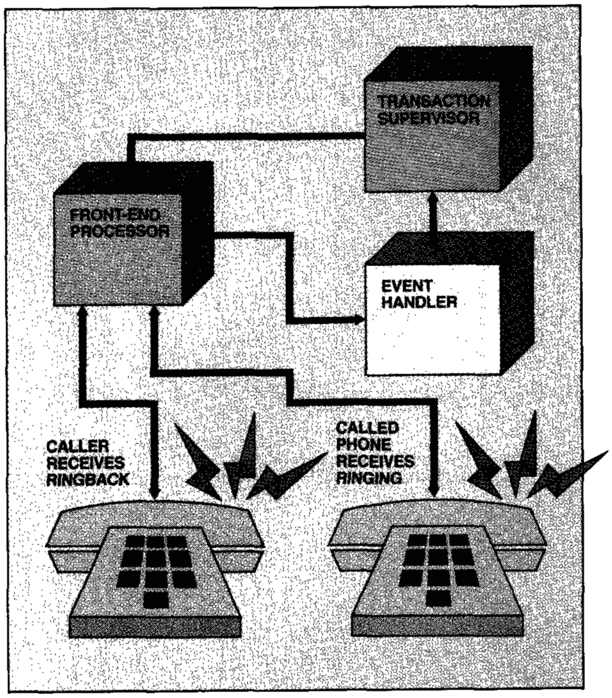


Figure 13 Software control of a telephone call: Phase 4



ROLMlink™

The CBX II supports many types of terminal links, the most common of which is a serial protocol called ROLMlink. ROLMlink provides a full-duplex communications channel of 256K bits per second from the switch to the terminal equipment, carried on unshielded telephone twisted pairs for distances up to 1300 meters. The link protocol is synchronous, operating bidirectionally over a single pair. Both ends transmit simultaneously, by separating the signals with hybrid line-balancing circuits. The link transmits both voice and data in 64K-bit-per-second subchannels.

Diagnostic programs run continuously as low-priority tasks.

The system uses this link to support a family of digital instruments, ranging from single-line telephones through telephones with both a speakerphone and a 60-character liquid crystal display (LCD) to integrated voice/data terminals and personal computers. Each instrument provides one-button feature access, configurable keys, and hands-free dialing. Each phone is configurable for quick access to popular features, such as conferencing, forwarding, transferring, and abbreviated dialing.

Maintenance

A built-in modem allows remote reconfiguration and access to diagnostic programs that normally run continuously as low-priority tasks. These programs are referred to collectively as *system integrity*. Access to the reconfiguration and system integrity system is controlled through a password protocol similar to those that are used on time-sharing systems.[11]

Once the security checks have been passed, the service person interacts with the CBX diagnostic programs. The computer may read and write patterns to the data bus directly. System integrity often takes advantage of this ability to create test connections to the modules under test, thereby allowing more accurate trouble-shooting than indirect methods.

When system integrity detects a fault, it enters the physical location of the offending circuit and a description of the failed test into an error table, and then activates an alarm indicator on the attendant's console. If a fault occurs during the night, the maintaining organization may detect it before the attendant does. Many such organizations remotely poll their CBX systems before normal working hours and dispatch a service technician when a fault is detected. In addition to the system integrity routines, the service technician uses status-monitoring and trace programs to localize subtle faults, including misconfigurations and user-training problems.[12]

Field reconfiguration of the CBX takes place using the same methods used to access system integrity. Because software determines virtually the entire CBX configuration, many changes do not involve changing hardware and are handled remotely. Technicians can remotely administer nearly all station and trunk attributes and many system tables, thus providing lower-cost implementation and more accurate record keeping.

Future directions

In the next five years, the characteristics of a telephone network will begin to change dramatically. Because the standards that are expected to play such an important role in the coming changes are now being settled, we can already predict the broad outlines of this change. Virtually all of the major European and most of the United States telephone operating companies and common carriers have expressed their support for a set of international standards, collectively termed Integrated Services Digital Network (ISDN). The ultimate objective of ISDN is to allow the implementation of public and private networks that provide services ranging from simple voice circuit switching, through data circuit and packet switching, to videotex, data base services, and computing services.[13] The standards, although not complete, are being developed in a bottom-up, hierarchical fashion, with the lower levels of the protocols being standardized first. There are several trial implementations already underway. Even though the widespread implementation of the ISDN standards is still several years away and actual implementations may not conform exactly to the present codifications, the trends are clear.

All ISDN communications will be digital, with both circuit- and packet-switching channels available. Transparent switching will allow voice, character-oriented data, and other coded information. Voice will be coded at 64K bits per second using PCM. There will be clear-channel digital circuits at 64K bits per second, and packet-switching channels at 16K bits per second and 64K bits per second. Two 64K-bit-per-second and one 16K-bit-per-second channel will be multiplexed onto one link that will be used to connect telephones, computers, and desktop devices to PBXs and public and private networks. Higher-bandwidth connections with 23 (30 in Europe) 64K-bit-per-second circuit-switched channels and one 64K-bit-per-second packet-switched channel will provide connection among PBXs, computers, and public and private networks.

Even without the advanced services promised under ISDN, inexpensive, high-speed global communications will make possible applications that are too costly or too slow today. The lower-speed interface will probably become a standard native terminal and personal computer interface. This will be the result of low cost in the terminal equipment, because of the high volume of large-scale integrated (LSI) circuits, low cost in the PBX and the public-switched network, and the commonality between the voice and the data protocols. The higher-speed interface will be used to connect computers and PBXs to the public-switched network. This higher-speed interface will act as a multiplexed interface between PBXs and computers and as a high-speed interface between data terminals and hosts.

Although the ISDN standards provide a reference point for the next decade of communications advances, much of the future is uncertain, because of simultaneous changes in business and technology. The following are some of those uncertainties:

- It will be many years before the results of the current communications industry deregulation become clear.
- The economic trade-offs among optical fiber, satellite, and switched-versus-leased transmission will continue to change in unpredictable ways.[14]
- Inexpensive desktop processing will cause new communications protocols to evolve in unexpected ways.
- More sophisticated modems, which make better use of the existing analog transmission facilities, will compete with new digital transmission offerings with uncertain near-term results.
- More sophisticated bandwidth compression techniques implemented in inexpensive large-scale integration will compete for marketplace acceptance with lower-cost transmission facilities. (Inasmuch as mankind appears to have an insatiable appetite for both communication and information, both forces will coexist for a long time.)

The following trends seem to be clear:

- High-speed digital wide-area communications facilities will continue to become more available at lower cost.
- Because of their low entry costs and abilities to handle high instantaneous bandwidths, multiple simultaneous sessions, and broadcast communications, local-area networks (LANs) will continue to proliferate, and the present gateways between LANs and the circuit-switched world will increase in capacity and features.
- Using the common-channel signaling capabilities provided by ISDN, private PBX networks will be designed to allow many more features to be supported across large geographical areas, as well as providing for advanced network management facilities.
- PBX management software will be more closely linked to computer system administration tools, thereby achieving an integrated view of management of all communications facilities.

It seems certain that the next five years, even more than the last five, will see new and exotic communications capabilities accompanied by confusion about their proper use.

Concluding remarks

The combination of computer-controlled time-division multiplexing with pulse-code modulation and bus architecture of the ROLM CBXs makes possible data communications features that are vital to business. Such data communications features include communications with host computers, desktop devices, and many types of local- and wide-area communications networks. This computer control structure has allowed the addition of features like automatic call distribution, centralized attendant service, and voice messaging.

The CBX II provides a switching system that can be constructed in large sizes without sacrificing traffic capacity. Using modular architecture, the CBX II provides many advanced features without serious cost penalties for smaller customers or those who do not need the switching flexibility.

Cited references

1. A. E. Joel, Jr., *A History of Engineering and Science in the Bell System*, Bell Telephone Laboratories, Inc., Murray Hill, NJ (1982), pp. 443–594.
2. G. Zorpette, "The telecommunications bazaar," *IEEE Spectrum* **22,** No. 11, 56–63 (November 1985).
3. J. M. Kasson, "Computer sophistication yields new versatility, economy in medium-size PBX," *Telephony* **188,** No. 12, 26 (1975).
4. G. L. Rainey and B. E. Voss, "SL-1: A Business Communications System with Digital Switching and Stored Program Control," *International Switching Symposium Record*, October 1976.
5. J. M. Kasson, "The ROLM computerized branch exchange: An advanced digital PBX," *Computer* **12,** No. 6, 24–31 (June 1979).
6. J. M. Kasson and H. W. Johnson, "The CBX II switching architecture," *IEEE Journal of Selected Areas in Communications* **SAC-3,** No. 4, 555–560 (July 1985).
7. J. Holding and S. Taylor, "Digital voice store and forward system answers the phones, takes messages," *Electronics* **56,** No. 8, 139–143 (April 23, 1983).
8. J. Edwards and D. Lieberman, "The ROLM CBX as an integrated voice/data communications switching system," *National Telecommunications Conference Record*, New Orleans, LA, November 1981; IEEE Catalog Number 81-CH-1679-0, Vol. 3 (pp. F3.1.1–F3.1.5).
9. J. M. Kasson, "A survey of digital PBX design," *IEEE Transactions on Communications* **COM-27,** No. 7, 1118–1124 (July 1979).
10. L. A. Baxter, P. R. Berkowitz, C. A. Buzzard, J. J. Herenkamp, and F. E. Wyatt, "System 75: Communications and control architecture," *AT&T Technical Journal* **64,** No. 1, 153–173 (January 1985).
11. J. M. Kasson, "System self test, administration and maintenance in the ROLM CBX," *National Telecommunications Conference Record*, New Orleans, LA, December 1–3, 1975; IEEE Catalog Number 75-CH-10115-7-CSCB, Vol. 1 (pp. 11-20–11-23).
12. J. M. Kasson and C. S. Plant, "ROLM CBX maintenance with Release IV software," *Proceedings of the National Electronics Conference*, October 1977, pp. 75–80.
13. N. Q. Duc and E. K. Chew, "ISDN protocol architecture," *IEEE Communications Magazine* **23,** No. 3, 15–22 (March 1985).
14. F. Guterl and G. Zorpette, "Fiber optics: Poised to displace satellites," *IEEE Spectrum* **22,** No. 8, 30–37 (August 1985).

James M. Kasson *ROLM Corporation, 4900 Old Ironsides Drive, Santa Clara, California 95050.* Mr. Kasson received his B.S.E.E. from Stanford University in 1964 and his M.S.E.E. from the University of Illinois in 1965. From 1963 to 1969, he performed research in speech bandwidth compression and speech recognition at Santa Rita Technology in Menlo Park, California. He then developed data acquisition and process control systems and subsystems at the Hewlett-Packard Company. In 1973, he went to work for the ROLM Corporation, where he led the development of the ROLM CBX. He has served as Vice President for Engineering and as General Manager of the Office Systems Division. He is currently Vice President for Research and Development, and he is also Director of the IBM Advanced Telecommunications Systems Laboratory. Mr. Kasson is a Fellow of the IEEE and an IBM Fellow.

Reprint Order No. G321-5282.