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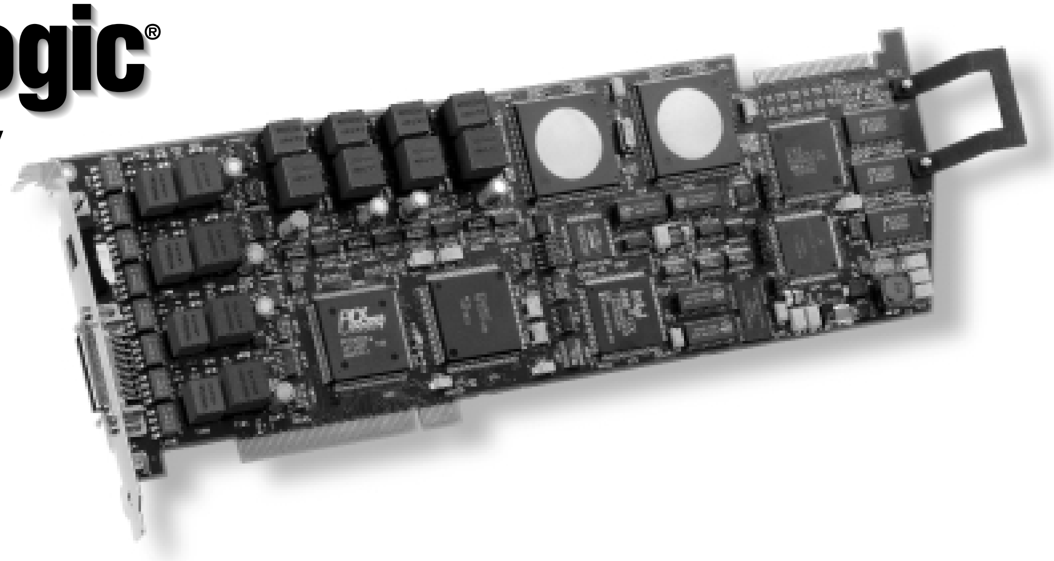
an Intel company



SCSA Hardware Model | CT Bus

D/82JCT-U

Revolutionary PBX Integration Platform

The D/82JCT-U™ board offers advanced digital connectivity to today's most popular private branch exchanges (PBXs) for unified and Internet-ready call, voice, and fax processing applications. Featuring programmable soft-ports capable of supporting voice, fax, call handling, and host-based speech technologies, the D/82JCT-U board reduces cost of ownership for systems requiring multimedia functionality. The D/82JCT-U board's universal PBX interface offers a downloadable selection of the switches that make up close to 90 percent of the telephone equipment market. Support for advanced PBX features such as Called and Calling Party Identification, message waiting notification, busy lamp fields, disconnect supervision, and many others is useful for developing unique unified messaging, interactive voice response (IVR), and call management solutions. By choosing the D/82JCT-U board for enterprise applications, developers eliminate the complexities associated with analog or T-1 integration as well as costly investments in proprietary computer-telephone integration (CTI) links.

Features Programmable Resources for Multimedia Functionality

Beyond support for eight ports of leading voice-processing features from Dialogic, the D/82JCT-U board lets developers select from standard coders such as GSM and G.726 for transmitting voice messages through the private or public Internet. In addition to voice media, the D/82JCT-U board also features SoftFax: two ports of mappable 14,400 b/s (V.17) transmit and 9600 b/s (V.29) receive fax instances. In the near future, Dialogic will introduce host-based speech platforms to complement the D/82JCT-U board, letting developers offer host based automatic speech recognition (ASR) and text-to-speech (TTS) as part of a solution without the need to add expensive hardware. No other product on the market offers integration to multiple PBXs and support for multimedia including voice, fax, and speech recognition on a single hardware platform.

Features and Benefits

- Eight digital interfaces connecting to Definity, Meridian, Mitel, NEC, Norstar, and Siemens Rolm and Hi-Com provide tight, direct switch integration to the most popular PBXs on the market
- Digital interfaces to the PBX eliminate the need for expensive digital/analog conversion cards or serial ports
- Access to switch information like Called and Calling Party ID, Date/Time, Call Forward Reasons — Advanced Digital Connectivity provides more information than analog without the need for separate links and third-party call control to get data from the PBX
- Phone-emulated Supervised/Unsupervised transfers provide faster, more reliable transfers than analog
- Disconnect supervision offers faster disconnects than analog
- Supports GSM and G.726, the coders of choice for Internet-ready unified messaging applications while enabling messaging integration with the most popular email servers and applications.
- Two channels of SoftFax per card, sharable across CT Bus enables scalable fax capability at a low cost by eliminating the need for additional fax hardware

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Features and Benefits (cont.)

- H.100 connectivity allows systems to scale from 8 through 64 ports, meeting the density requirements of the most widely deployed enterprise computer telephony (CT) solutions.

Applications

- Unified Messaging (UM)
- Interactive Voice Response (IVR)
- Automatic Call Distribution (ACD)
- Telecom/Data equipment convergence solutions

Offers Advanced Digital Connectivity to the Most Popular PBXs on the Market

Developers will be able to choose from any of the following switches and phone types:

PBX Switch Manufacturers	Phone Emulations Supported	Available Switches
Lucent	7434 (4-wire)	Definity, System 75/85
Lucent	8434 (2-wire)	Definity (G3 ver. 4) only
Mitel	Superset 420	SX-50 only
Mitel	Superset 430	SX-200ML, SX-2000
NEC	Dterm III	KSU: Electra Professional II PBX: NEAX 2000 IVS, 2400 ICS
Nortel Networks	M7324	Norstar
Nortel Networks	M2616	Meridian 1
Siemens	Optiset E	HiCom 150E, HiCom 300 E
Siemens	ROLMphone 400 (RP400)	CBX 9000 Series

Compared to other methods, PBX integration offers more features and greater reliability. Call transfers are speedier and developers do not run the risk of dropped transfers as they do with analog "hook flash" methods. The switch offers the application a wealth of information including called and calling party ID, call forward reasons, and date and time stamps. Such features are valuable for many customer premises applications. In addition, the D/82JCT-U board offers a single programming interface eliminating the need for complex, switch-specific programming interface management as well as voice, fax, and speech programming interfaces.

Hardware and Software Environments

The D/82JCT-U board is offered in a PCI long-card form factor for easy implementation in the most widely available chassis. Systems can readily scale from eight ports to 64 ports and beyond, using the industry-standard H.100 bus. The D/82JCT-U board is supported under Windows NT. Dialogic plans to support the product under CT Media™ server software, with a developer’s choice of the S.100 media application programming interface (API) or Microsoft TAPI 3.0 API. Call your Dialogic technical sales representative at 1-800-755-4444 for further details.

CONFIGURATIONS

The D/82JCT-U board is based on the latest Dialogic PBX integration platform with an industry-standard CT Bus interface enabling access to all CT Bus-supported complementary technologies. A unique, dual-processor architecture comprised of a digital signal processor (DSP) and a general purpose microprocessor handles all telephony signaling and performs DTMF (touchtone) and audio/voice signal processing tasks. When used with the supported PBXs, the D/82JCT-U board provides a flexible platform for developing integrated computer telephony (CT) applications. Dialogic developers can port current applications on the Dialogic D/42 Series of PBX integration boards to D/82JCT-U boards with minimal software modifications, and create more efficient applications that are portable for use with multiple PBXs.

Use the D/82JCT-U board to build sophisticated, multimedia communications systems that incorporate capabilities such as voice processing, fax, text-to-speech (TTS), and ASR. The D/82JCT-U board shares a common hardware and firmware architecture with other Dialogic SCbus™ and CT Bus boards for maximum flexibility and scalability. You can add features and grow the system while protecting your investment in hardware and application code — with only minimum modifications, applications can be easily ported to different PBX integration solutions.

The D/82JCT-U board installs in any PCI-based Pentium™ processor PC or server (PCI bus or mixed PCI/ISA) and compatible computers. Each board occupies a single expansion

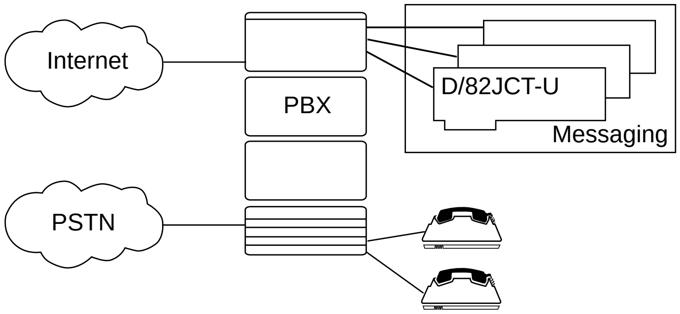

Figure 1: D/82JCT-U configuration diagram

slot and up to eight boards can be configured in a single system. The number of boards and channels supported varies depending on the application, the operating system, the amount of disk I/O required, the number of CT Bus loads per board, and the host computer’s CPU(s) and power supply.

The D/82JCT-U board can operate within a mixed chassis containing Dialogic PCI and ISA products. The board's forward-looking design incorporates the new H.100 connector to simplify the connection to next-generation CT Bus products.

The D/82JCT-U board can access called number identification for calls transferred from within the PBX, access trunk identification for calls originating outside the PBX, and control message waiting indicators for message notification.

Downloaded firmware algorithms executed by the on-board DSP provide variable voice coding at 24 and 32 Kb/s adaptive differential pulse code modulation (ADPCM), and 48 and 64 Kb/s µ-law or A-law pulse code modulation (PCM). Sampling rates and coding methods are selectable on a channel-by-channel basis. Applications may dynamically switch sampling rate and coding method to optimize data storage or voice quality as the need arises. These firmware algorithms also provide reliable DTMF detection, DTMF cut-through, and talk off/play off suppression over a wide variety of telephone line conditions.

Dialogic voice products offer a rich set of advanced features including innovative DSP technology and signal processing algorithms for building the core of any CT system. With industry-standard PCI bus expansion boards and a variety of channel densities to choose from, you can integrate Dialogic voice products easily into your PBX system at a price and performance level unmatched in the CT industry.

SOFTWARE SUPPORT

The Dialogic System Releases for Windows NT and Windows 2000 provide support for the D/82JCT-U board and contain a set of tools for developing sophisticated, multimedia communications applications.

The D/82JCT-U board also supports BoardWatch™, the SNMP-compatible software for remote CT board management. BoardWatch software simplifies the management of CT devices and lowers the total cost of operation. Centralized management capabilities provide a single point of configuration and inventory for all network devices. Fault management for high availability systems includes diagnostics, detection, and recovery capabilities.

The D/82JCT-U also supports the D/42 R4 API (Unified API™), which provides developers with a single set of basic functions that can be used by any supported switch and that can be sent directly to the switch without additional hardware support. Functioning as an extension of the Dialogic standard voice API, the Unified API allows developers to take advantage of the following advanced PBX features:

- Called/calling number ID — Usually two sets of digits representing either a trunk line or extension. Tells an application where a call originated and to what extension it was directed. (Not the same as Caller ID)
- Retrieve LCD/LED prompts and indicators — Allows an application to determine what kind of prompts or indicators have been set
- Read displays — Allows an application to "read" display information such as hook state, messages, features, and other ASCII text

■ Functional Description

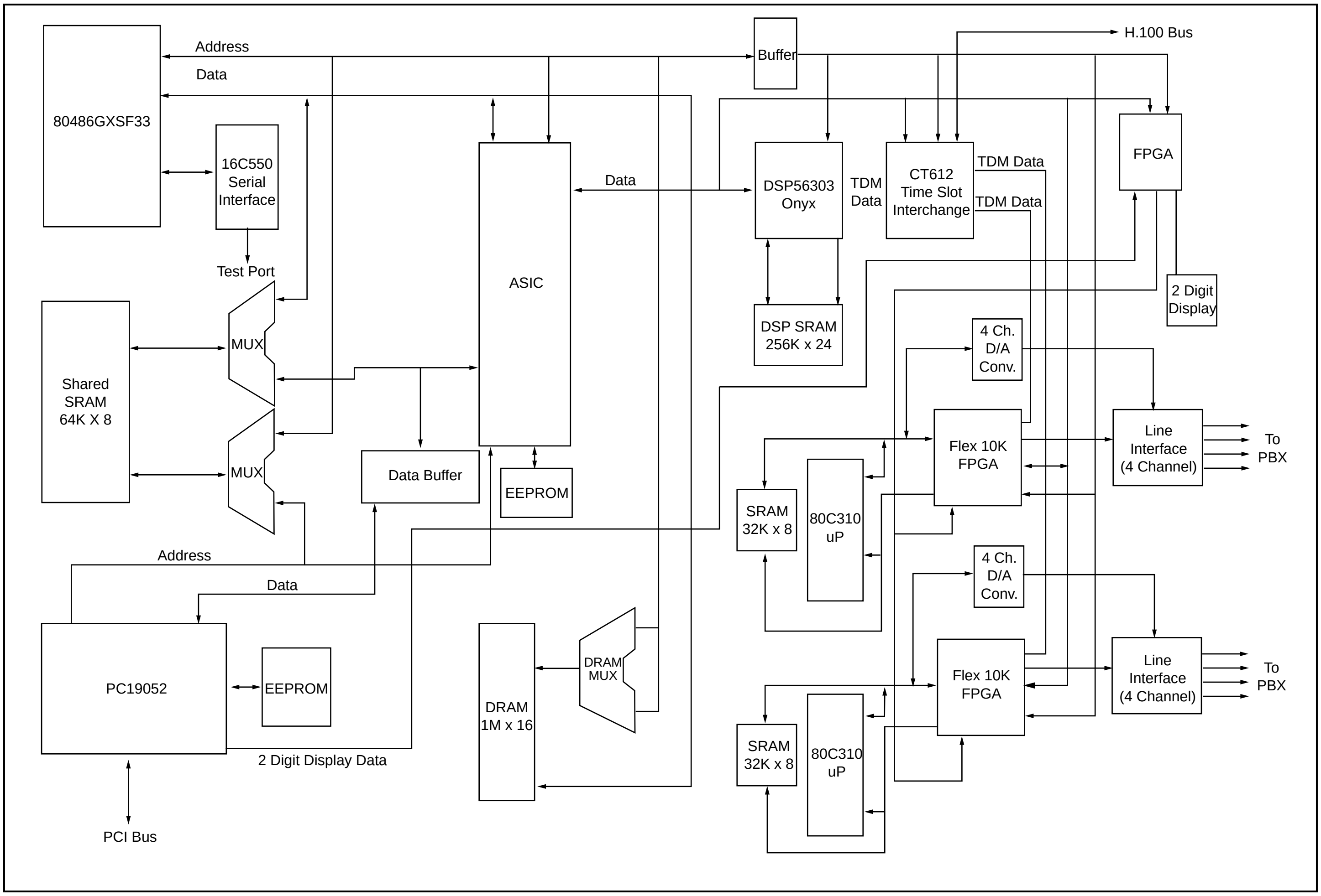

Figure 2: D/82JCT-U block diagram

The D/82JCT-U board is an 8-port voice processing/PBX integration board with Group 3 fax support. Each port is a digital interface that connects directly to the station interface of various PBXs when the appropriate downloadable software is selected. The D/82JCT-U board is designed around the proven Dialogic dual-processor architecture and consists of two primary sections: the PBX interface, and the voice and fax processing engine.

In addition to having all the standard features of a Dialogic voice processing board, the D/82JCT-U board can access enhanced PBX station set features such as:

- call transfer
- turn phone message waiting indicators on or off
- receive called party identification
- receive positive disconnect supervision
- access PBX features using dial strings

Each line interface on the D/82JCT-U board receives PCM digital voice and control data from the PBX station interface port (see block diagram). The digital voice signals are compressed and processed by a DSP.

Control data from the PBX passes through the digital duplexer to a command processor where it is converted from PBX format to D/41D format. This serial bit stream is then converted into a parallel bit stream that is sent via the local bus to the on-board control processor which either acts on the information or passes the event to the application.

Incoming data for each channel is divided into separate voice and control data signaling portions. The voice portion contains the digitized voice data for every channel, while the control data-signaling portion contains the telephone and special feature signaling information for every channel.

The D/82JCT-U has the Dialogic standard, dual-processor architecture that combines the signal processing capabilities of a DSP with the decision-making and data movement functionality of a general purpose 80486 control microprocessor. This dual processor approach off-loads many low-level decision-making tasks from the host computer and thus enables easier development of more powerful applications. This architecture handles real time events, manages data flow to the host PC for faster system response time, reduces host PC processing demands, processes DTMF and telephony signaling, and frees the DSP to perform signal processing on the incoming call.

The PBX interface is the electrical and functional link between the D/82JCT-U board and PBX station interface. The purpose of this interface is to transmit and receive voice, fax, and signaling information to and from the PBX.

Functional Description

In the receive mode, each of the eight PBX interfaces converts network PBX data to PCM data and applies the incoming bit stream to the CT612 chip. The CT612 ASIC distributes the data onto pre-assigned time slots on the local TDM bus. Voice (and fax) transmissions are routed to the DSP voice processing engine. In transmit mode, the DSP voice processing engine transmits voice in assigned time slots to the CT612 device. Voice is passed on the local TDM bus to the line interface, and is then transmitted to the PBX.

The CT612 ASIC also serves to exchange voice and fax between a time slot on the local TDM bus and a time slot on the CT Bus. The D/82JCT-U board can transmit or receive voice or fax information to and from an external device over the CT Bus.

The D/82JCT-U board has an on-board microprocessor (80486) and a high-speed DSP (Motorola 56303 Onyx) to provide voice and call processing. When the system is initialized, SpringWare™ firmware is downloaded from the host PC to the board and all board operations. SpringWare gives the board all of its intelligence and enables easy feature enhancements and upgrades.

SpringWare offers several features including PerfectPitch™ Speed Control, PerfectLevel™ Volume Control, Global Tone Detection™, and Positive Voice Detection™. PerfectPitch allows callers to change the speed of messages played back without any pitch distortion. PerfectLevel allows callers to adjust the volume of messages before or during playback. Global Tone Detection allows applications to detect special intercept tones, fax tones, modem tones, non-standard PBX tones, or user-defined tones such as tones in international networks.

The D/82JCT-U board processes voice with a Motorola 56303 Onyx DSP voice processing engine. The DSP resource receives voice and fax via the CT612 chip. The Motorola 56303 Onyx DSP processes the digitized voice data based on SpringWare firmware loaded in code/data RAM. The Motorola 56303 Onyx DSP performs the following signal analysis and operations

- automatic gain control to compensate for variations in the level of the incoming audio signal
- applies an ADPCM or PCM algorithm to compress the digitized voice and save disk storage space
- detects the presence of tones — DTMF, MF, or an application-defined single or dual tone
- silence detection to determine whether the line is quiet and the caller is not responding

For outbound voice, the DSP performs the following operations

- expands stored, compressed audio data for playback
- adjusts the volume and rate of speed of playback upon application or user request
- generates tones — DTMF, MF, or any application-defined general-purpose tone

The dual-processor combination also performs outbound dialing and call progress monitoring

- transmits an off-hook signal to the telephone network
- dials out (makes an outbound call)
- monitors and reports results: line busy or congested; operator intercept; ring, no answer; or if the call is answered, whether answered by a person (PVD™), an answering machine (PAMD™), a facsimile machine, or a modem.

When recording speech, the DSP can use different digitizing rates from 24 to 64 Kb/s as selected by the application for the best speech quality and most efficient storage. The digitizing rate is selected on a channel-by-channel basis and can be changed each time a record or play function is initiated. Outbound signal processing is the reverse of inbound processing. The DSP- processed speech is transmitted via the control microprocessor to the host PC for disk storage. When replaying a stored file, the microprocessor receives the voice information from the host PC via a shared buffer memory and passes it to the DSP. The DSP then sends the digitized voice to the digital duplexer in PBX format for transmission to the PBX.

Signaling data (on-/off-hook, ringing, Caller ID, and special feature signaling data) is converted from KSU message format by the command processor, passed to the on-board control processor, and transmitted to the application via a dual-port shared RAM and the host PCI bus.

An onboard control processor accomplishes the flow of information between the voice engine and the host PC. The control processor controls all operations of the D/82JCT-U board and interprets and executes commands from the host PC. The processor handles real-time events, manages data flow to the host PC to provide faster system response time, reduces PC host processing demands, processes DTMF and PBX signaling before passing them to the application, and frees the DSP to perform signal processing.

Communications between the control processor and the host PC is via the shared RAM that acts as an input/output buffer, increasing the efficiency of disk file transfers. The RAM interfaces with the host PC via the PCI bus. All operations are interrupt-driven to meet the demands of real-time systems. When the system is initialized, SpringWare firmware is downloaded from the host PC to the onboard code/data RAM and DSP RAM to control all board operations.

The traffic controller ASIC (TCA) is the 80486 control processor interface that handles all peripheral devices (CT612, DSP, PBX interface) and host PC functions across the PCI bus. The D/82JCT-U board's hardware is plug-and-play enabled. The board ID/slot number configuration is handled exclusively by software, using the Dialogic Configuration Manager (DCM). The assigned board ID/slot number is presented on a hexadecimal display located on the PCB bracket.

Technical Specifications*

Number of ports	8
Total ports/system	64
Max. boards/system	8
Microprocessor	Intel® 80486GXSF running @ 28.5 MHz with 2MB DRAM
Digital signal processor	Motorola DSP56303 (Onyx) @ 100 MHz, 24-bit
DSP SRAM	256K SRAM

HOST INTERFACE:

Bus compatibility	PCI
Bus speed	33 MHz
Shared memory	64 KB SRAM
Interrupt level	One IRQ is shared by all D/82JCT-U boards

TELEPHONE INTERFACE:

Support	Lucent 7434 (4-wire), Lucent 8434 (2-wire), Siemens Optiset E, Siemens ROLMphone 400, MITEL Superset 420, MITEL Superset 430, Nortel M7324, Nortel M2616
Connectors	36-position mini D cable plug

POWER REQUIREMENTS:

+5 VDC	2 A at 5 volts per board
Operating temperature	0ºC to +50ºC
Storage temperature	-20ºC to +70ºC
Humidity	8% to 80% non-condensing

FORM FACTOR: 5V PCI long form factor card. 12.283 in. long and 4.200 in. high

SAFETY & EMI CERTIFICATIONS:

United States	FCC part 68
Canada	CSO3

■ SpringWare Technical Specifications*

AUDIO SIGNAL:

Usable receive range	–42 to +2.5 dBm0
Silence detection	–38 dBm0 nominal, software adjustable**
Transmit level	–12.5 dBm0 nominal (weighted average)**
Frequency response	24 Kb/s: 300 Hz to 2600 Hz ±3 dB
	32 Kb/s: 300 Hz to 3400 Hz ±3 dB
	48 Kb/s: 300 Hz to 2600 Hz ±3 dB
	64 Kb/s: 300 Hz to 3400 Hz ±3 dB

AUDIO DIGITIZING:

Method	G.711 A-law and µ-law PCM; GSM 610; G.726
Sampling rates	6 kHz, 8 kHz for PCM
Data rates	G.711 A-law and µ-law PCM; 48 Kb/s

TONE DIALING:

DTMF digits	0 to 9, *, #, A, B, C, D
MF digits	0 to 9, KP, ST, ST1, ST2, ST3
Level	Network compatible
Rate	10 digits/s maximum, software adjustable

DTMF TONE DETECTION:

DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6
Dynamic range	-39 dBm0 to +0 dBm0 per tone**
Minimum tone duration	32 ms, software adjustable
Acceptable twist	10 dB
Signal/noise ratio	10 dB (referenced to lowest amplitude tone)
Talk off	Detects 0 digits while monitoring MITEL speech tape #CM7291. Detects less than 10 digits while monitoring Bellcore TR-TSY-000763 standard speech tapes (LSSGR requirements specify detecting no more than 470 total digits).

MF TONE DETECTION:

MF digits	0 to 9, KP, ST, ST1, ST2, ST3

SPEED CONTROL:

Pitch controlled	Available for 24 and 32 Kb/s data rates
Adjustment range	50%

VOLUME CONTROL:

Adjustment range	40 dB, with programmer-definable increments

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* All specifications are subject to change without notice.

** Analog levels: 0 dBm0 corresponds to a level of +3dBm at tip-ring analog point.

HARDWARE SYSTEM REQUIREMENTS

Minimum 90 MHz Pentium™5- or the equivalent Celeron®-based platform with an available PCI bus slot for an 8-port system. The host system must provide a CPU of Pentium or Celeron class at 266 MHz speed or higher for a 64-port system, including eight available PCI slots. The D/82JCT-U board occupies a single expansion slot, and up to eight boards can be configured in a system, with each board sharing the same interrupt level. The maximum number of ports supported is 64, dependent on the application, the amount of disk I/O required, and the host computer's CPU. The computer must run the Windows NT or Windows 2000 operating system.

OPTIONAL COMPONENTS

- Optional multidrop SCbus or CT Bus cables
- Optional CT Bus/SCbus Adapter
- Optional Lab Cable Kit