

ANSI SCbus

H.100 CT Bus PCI

H.110 CT Bus cPCI

D/41ESC

Global SCSA 4-Port Voice Processing Board

The D/41ESC™ voice processing board brings DSP-based call technology to the global marketplace. This SCSA four-channel, loop start voice board complements the series of global telecomputing products provided by Dialogic. The unique interface circuitry of the D/41ESC is approvable for connection to analog networks in over 30 countries. (See your Dialogic sales engineer for a list of the latest approvals.)

The D/41ESC provides four telephone line interface circuits for direct connection to analog loop start lines. A unique dual-processor architecture, comprising a DSP (digital signal processor) and a general purpose microprocessor, handles all telephony signaling and performs DTMF (touchtone) and audio/voice signal processing tasks. This architecture allows the board to run SpringWare™, the advanced set of call processing firmware features by Dialogic, including selectable rate, high-quality voice coding with speed control, outstanding DTMF detection with cut-through, and advanced outbound call progress analysis.

Multiple D/41ESCs can be installed in a single PC chassis enabling system expansion up to 64 ports. For adding resources such as facsimile, speech recognition and text-to-speech, the D/41ESC provides an SCbus™ option or PCM Expansion Bus™ (PEB™) option as well as an Analog Expansion Bus™ (AEB™). With the D/41ESC you can create applications that allow hands-free speed dialing from cellular car phones, hands-free voice mail, and automatic dialing of spoken numbers or names. Complicated numeric menu systems can be reduced to a small set of user friendly spoken commands.

Downloaded firmware algorithms, SpringWare™, executed by the onboard DSP, provide variable voice coding at 24 and 32 Kb/s ADPCM, and 48 and 64 Kb/s μ-law or A-law 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. SpringWare also provides reliable DTMF detection, DTMF cut-through, and talk off/play off suppression over a wide variety of telephone line conditions.

FEATURES AND BENEFITS

- Four independent voice processing ports in a single PC ISA slot supports low- to medium-density voice systems
- Approved for use in numerous countries throughout North and South America, Europe, and Asia/Pacific
- SCSA™ SCbus™ connectivity enables applications requiring switching and allows access to additional resources such as fax, text-to-speech, and automatic speech recognition
- Supports Windows NT® and Windows® 95 including TAPI/WAVE®
- A-law or μ-law voice coding at dynamically selectable data rates, 24 Kb/s to 64 Kb/s, selectable on a channel-by-channel basis for optimal tradeoff between disk storage and voice quality
- Dialogic SpringWare™, downloadable signal and call processing firmware, provides easy feature enhancement and field-proven performance based on over four million installed ports

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FEATURES AND BENEFITS

- International Caller ID capability via on-hook audio path. Supports Bellcore CLASS™, UK CLI, and other international protocols.
- PerfectDigit™ DTMF (touchtone) provides reliable detection during voice playback — allows callers to “type-ahead” through menus
- Patented outbound call progress analysis monitors outgoing call status quickly and accurately
- Configure multiple boards in a single PC for easy and cost-effective system expansion and to build scalable systems from 4 to 64 ports
- C language application program interfaces (APIs) for MS-DOS®, UNIX®, OS/2®, Windows NT, and Windows 95 shorten your development cycle so you can get your applications to market faster
- Support for Global Dial Pulse Detection (DPD™) pulse-to-tone conversion software
- Supports software-based speech technologies, including TextTalk™ TTS and SpeechWorks™ host-based speech recognition
- Supports PBXpert™, a free utility that simplifies switch integration

Dialogic voice products offer a rich set of advanced features, including state-of-the-art DSP technology and signal processing algorithms, for building the core of any computer telephony system. With industry-standard ISA bus expansion boards and a variety of channel densities to choose from, you can integrate Dialogic voice products easily into exactly the type of system you require at a price and performance level unmatched in the computer telephony industry.

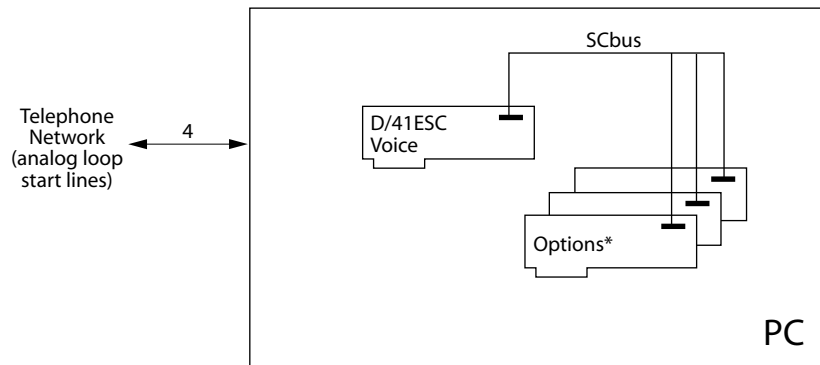
The D/41ESC board

- connects directly to analog loop start telephone lines
- offers application-controlled call answering
- detects touchtones
- plays voice messages to a caller
- digitizes, compresses, and records voice signals
- places outbound calls and automatically monitors their progress
- all in real time on four independent channels.

Configurations

Use the D/41ESC board to build sophisticated, computer telephony systems to which capabilities such as speech recognition, facsimile, and text-to-speech can be added. The D/41ESC shares a common hardware and firmware architecture with other Dialogic SCbus, PEB, and AEB-based boards for maximum flexibility and scalability. Features can be added and systems can grow while protecting investment in hardware and application code. With only minimum modifications, applications can be easily ported to lower or higher line-density platforms.

The D/41ESC installs in IBM® PC AT® (ISA bus) and compatible computers (80386, 80486, or Pentium™-based PC platforms). The D/41ESC provides everything required for building integrated voice solutions scalable from 4 ports to 64 ports. The maximum number



- *GammaLink CP4/SC Facsimile
- *Antares™ DSP Platform
- *Other third-party SCbus resources

of lines that can be supported is dependent on the application, the amount of disk I/O required, and the host computer CPU and power supply.

Applications developed to run on the Proline/2V™, DIALOG/4™, D/41D™, or D/41H™ family will run on a similar D/41ESC configuration. Developers can choose from a wide selection of Dialogic products to build scalable, reliable, and economical computer telephony.

SOFTWARE SUPPORT

The D/41ESC is supported by Dialogic System Software and Software Development Kits for many popular operating systems including MS-DOS, OS/2, UNIX, Windows NT, and Windows 95. These packages contain a set of tools for developing complex multichannel applications.

Based on SpringWare firmware loaded in DSP SRAM, the DSP performs the following signal analysis and operations on this incoming data:

- applies automatic gain control to compensate for variations in the level of the incoming audio signal
- applies an Adaptive Differential Pulse Code Modulation (ADPCM) or Pulse Code Modulation (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
- detects silence to determine whether the line is quiet and the caller is not responding

For outbound data, 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 the following outbound 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
 - if answered, whether answered by a person, an answering machine, a fax machine, or a modem

The D/41ESC also supports Global Dial Pulse Detection (DPD) Software that recognizes dial pulse digits even in the most difficult telephony environments.

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. The DSP-processed speech is transmitted by the control processor to the host PC for disk storage. When replaying a stored file, the processor retrieves the voice information from the host PC and passes it to the DSP, which converts the file into digitized voice. The DSP sends digitized voice and appropriate signaling responses to the CODEC to be converted into analog format for transmission to the telephone network.

Signaling data (on-/off-hook, ringing, Caller, etc.) is passed to the onboard control processor and transmitted to the application via a dual-port shared RAM and the host PC ISA bus.

When using the D/41ESC board with SCbus or PEB, digital voice and signaling information from a network board or other resource enter the board via the SCbus interface. These signals are managed by a SC2000 chip that acts as the traffic coordinator and matrix switch to buffer the high-speed digital data from the bus until the data for each channel can be transmitted to the DSP.

The SC2000 chip transmits several lower speed data streams over a single high-speed channel, either the SCbus or the PEB. The bus configuration is set when the firmware is downloaded at system initialization. This chip incorporates matrix-switching capabilities. Under control of the onboard control processor, the SC2000 chip can connect any call being processed to any of the four analog lines or to any SCbus or PEB time slot (1024 for the SCbus or 24 for the PEB in T-1 mode, or 32 in E-1 mode). This enables the application to switch calls to or from other resources, such as facsimile or speech recognition, as they are needed, or to reroute calls.

The SC2000 chip can bundle time slots to carry high bandwidth data and can broadcast to multiple resources over the SCbus.

The onboard microprocessor controls all operations of the D/41ESC via a local bus and interprets and executes commands from the host PC. This microprocessor 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 telephony signals before passing them to the application, and frees the DSP to perform signal processing. Communications between this microprocessor and the host PC is via the dual port shared RAM that acts as an input/output

buffer and thus increases the efficiency of disk file transfers. This RAM interfaces to the host PC via the AT® (ISA) bus. All operations are interrupt driven to meet the demands of real-time systems. When the system is initialized, SpringWare firmware to control all board operations is downloaded from the host PC to the onboard code/data RAM and DSP SRAM. This downloadable firmware gives the board all of its intelligence and enables easy feature enhancement and upgrades.

The Board Locator Technology circuit operates in conjunction with a rotary switch to determine and set nonconflicting PC memory and IRQ interrupt-level parameters. This feature eliminates the need to set confusing jumpers or DIP switches.

■ Technical Specifications*

Number of ports	4
Max. boards/system	16. Number may be limited by application and system performance.
Analog network interface	Onboard loop start interface circuits
Resource sharing bus	AEB; SCbus, or PEB
Control microprocessor	Intel 80C186 @ 16 MHz
Digital signal processor	Motorola DSP56002 @ 49 MHz, with 32 K word private, 0 wait state SRAM

HOST INTERFACE:

Bus compatibility	IEEE P996 ISA compatible (IBM PC XT/AT)
Bus speed	12.5 MHz maximum
Bus mode	Automatically configures to 8- or 16-bit transfer mode
Shared memory	8 KB page
Base addresses	8000h to E800h, on 32 K boundaries. All D/41ESC boards share the same base address. Shared memory is page mapped in/out dynamically as needed.
Interrupt level	IRQ 2/9, 3, 4, 5, 6, 7, 10, 11, 12 software selectable. One IRQ is shared by all D/41ESC boards.
I/O ports	None

TELEPHONE INTERFACE†:

Trunk type	Loop start
Loop current range	20 to 120 mA
Impedance	600 Ohms nominal
Ring detection	15 Vrms min., 13 to 68 Hz (configurable by parameter)
Echo return loss	Configurable by software parameter
Cross talk coupling	Less than -70 dB at 1 KHz channel to channel
Receive signal/noise ratio	70 dB referenced to -15 dBm
Freq. response	200 Hz to 3400 Hz ±3 dB (transmit and receive)
Connector	Four RJ-11 type

POWER REQUIREMENTS:

+5 VDC	820 mA max.
+12 VDC	113 mA max.
-12 VDC	86 mA max.
Operating temperature	0°C to +50°C
Storage temperature	-20°C to +70°C
Humidity	8% to 80% noncondensing
Form factor	PC AT, 13.34 in. long, 0.79 in. wide, 4.8 in. high

SAFETY AND EMI CERTIFICATIONS:

United States	FCC Part 15 class A; FCC Part 68 EBZUSA-75385-VM-T UL: E-143032 UL1950, 3rd edition
Canada	DOC: 885-5542A
Europe	For specific country approval designation, see the Dialogic Global Approvals list or contact a sales engineer. Use the D/41ESC-Euro card in CTR21 member countries in Europe.
Warranty	3 years standard

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AUDIO SIGNAL:

Receive range	-50 to -13 dBm (nominal), for average speech signals± configurable by parameter†
Automatic gain control	Application can enable/disable. Above -18 dBm results in full scale recording, configurable by parameter†
Silence detection	-38 dBm nominal, software adjustable†
Transmit level (weighted average)	-9 dBm nominal, configurable by parameter†
Transmit volume control	40 dB adjustment range, with application definable increments
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:

24 Kb/s	ADPCM @ 6 kHz sampling
32 Kb/s	ADPCM @ 8 kHz sampling
48 Kb/s	μ-law PCM @ 6 kHz sampling
64 Kb/s	μ-law PCM @ 8 kHz sampling
Digitization selection	Selectable by application on function call by call basis
Playback speed control	Pitch controlled; available for 24 and 32 Kb/s ADPCM data rates; adjustment range: ±50%; adjustable through application or programmable DTMF control

DTMF TONE DETECTION:

DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6
Dynamic range	-45 dBm to +3 dBm per tone, configurable by parameter†
Minimum tone duration	40 ms, Can be increased with software configuration
Interdigit timing	Detects like digits with a 40 ms interdigit delay. Detects different digits with a 0 ms interdigit delay.
Twist and frequency variation	Meets Bellcore LSSGR Sec 6 and EIA 464 requirements
Acceptable twist	10 dB
Signal/noise ratio	10 dB (referenced to lowest amplitude tone)
Noise tolerance	Meets Bellcore LSSGR Sec 6 and EIA 464 requirements for Gaussian, impulse, and power line noise tolerance.
Cut through	Detects down to -36 dBm per tone into 600 Ohm load impedance
Talk off	Detects less than 20 digits while monitoring Bellcore TR-TSY-000763 standard speech tapes (LSSGR requirements specify detecting no more than 470 total digits). Detects 0 digits while monitoring MITEL speech tape #CM 7291.

GLOBAL TONE DETECTION™:

Tone type	Programmable for single or dual
Max. number of tones	Application dependent
Frequency range	Programmable within 300 to 3500 Hz
Max. frequency deviation	Programmable in 5 Hz increments
Frequency resolution	Less than 5 Hz — Note: certain limitations exist for dual tones closer than 60 Hz apart
Timing	Programmable cadence qualifier, in 10 ms increments
Dynamic range	Programmable, default set at -36 dBm to +3 dBm per tone

GLOBAL TONE GENERATION™:

Tone type	Generate single or dual tones
Frequency range	Programmable within 200 to 4000 Hz
Frequency resolution	1 Hz
Duration	10 msec increments
Amplitude	-43 dBm to -3 dBm per tone, programmable