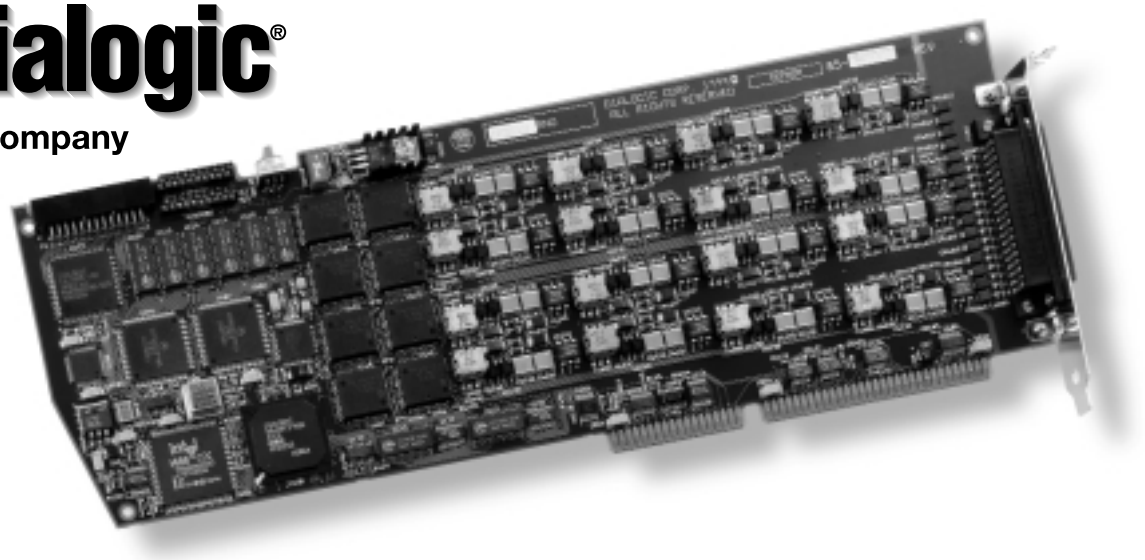
Dialogic®
an Intel company



SCSA Hardware Model | AEB Hardware Model | PEB Hardware Model

D/160SC-LS-HIZ

16-Port Voice Processing and Analog Hi-Impedance Interface Board

Dialogic voice products offer a rich set of advanced features, including state-of-the-art digital signal processor (DSP) technology and signal processing algorithms, for building the core of any computer telephony (CT) 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 CT industry.

The D/160SC-LS™-HIZ board provides 16 channels of call processing interfaces in a single PC slot. A unique dual-processor architecture comprised of DSPs and a general-purpose microprocessor handles all telephony signaling and performs all DTMF (touchtone) and audio/voice signal processing tasks. The D/160SC-LS-HIZ board uses the Signal Computing System Architecture™ (SCSA). SCSA provides an open architecture that enables developers to use products from multiple vendors to build a unified CT solution. SCSA provides features such as distributed switching, logical addressing, and location-independent resource management.

Downloaded firmware algorithms such as Dialogic SpringWare™ provide variable voice coding at 24 and 32 Kb/s Adaptive Differential Pulse Code Modulation (ADPCM), and 48 and 64 Kb/s Pulse Code Modulation (PCM) µ-law or A-law. Sampling rates and coding methods are selectable on a channel-by-channel basis. Applications can dynamically switch sampling rate and coding method to optimize data storage or voice quality as the need arises. SpringWare firmware also provides reliable DTMF detection and talk off/play off suppression over SCbus.

Features and Benefits

- 16 independent Hi-Z telephone interfaces, combined with 16 channels of voice processing in one ISA slot, provide effective solutions for building high-density applications
- High impedance, on-hook record capability enables high-density call logging and transaction record applications
- Highest density analog interface voice processing platform in the industry enables system integrators and developers to lower costs by incorporating more ports per chassis, by using less expensive desktop-style machines, and by easing configuration/installation effort
- Create more cost-effective switching solutions via access to the SCSA™ SCbus™ with its 1024 time slot capability; SCxbus™ interbox communications provides the capability to build higher density systems and large, multinode systems
- Downloadable signal and call processing firmware by Dialogic, called SpringWare™, provides easy feature enhancement and field-proven performance based on over four million installed ports
- Playback via SCbus

Dialogic Corporation
1515 Route Ten
Parsippany, NJ 07054
1-973-993-3000
fax: 1-973-993-3093
http://www.dialogic.com

Intel Dialogic North American Sales
1-800-755-4444
or 1-973-993-3030
fax orders: 1-973-631-9631
sales@dialogic.com

Dialogic Latin America and Caribbean
(Buenos Aires)
54-11-4328-1531
fax: 54-11-4328-5425
dlac.sales@dialogic.com

Dialogic Europe
(Brussels)
32-2-712-4311
fax: 32-2-712-4300
info.europe@dialogic.com

Dialogic Japan
(Tokyo)
81-3-5430-3252
fax: 81-3-5430-3373
japan.sales@dialogic.com

Dialogic China
(Beijing)
86-10-6590-0055
fax: 86-10-6590-7989
china.sales@dialogic.com

Dialogic Asia
(Singapore)
65-339-9833
fax: 65-339-9211
asia.sales@dialogic.com

Features and Benefits (cont.)

- Two independent Motorola 56002 digital signal processors (DSPs), clocked at 65 MHz; each with private, high-speed SRAM, permit execution of high performance SpringWare signal processing algorithms
- Intel® 486 GX microprocessor off-loads call processing tasks from host PC, giving more power to the application
- Board Locator Technology™ eliminates confusing DIP switch or jumper settings and simplifies installation
- C language application program interfaces (APIs) for MS-DOS®, UNIX®, Solaris®, and Windows NT® shorten your development cycle so you can get your applications to market faster
- Caller ID capability for "screen pop" applications (supports Bellcore CLASS protocols)
- Configure multiple boards in a single PC (ISA bus) for easy and cost-effective system expansion on the best computing platform that best fits your needs

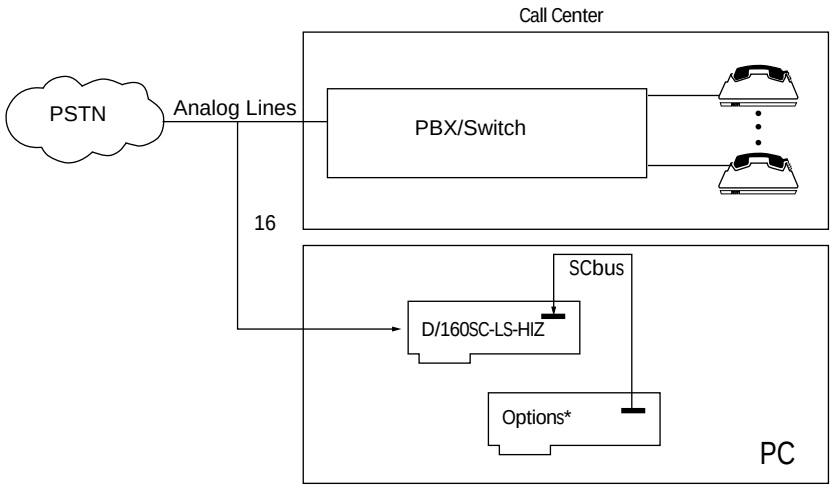
The D/160SC-LS-HIZ voice board

- connects to 16 analog telephone channels
- detects touchtone
- digitizes, compresses, and records voice signals

Configurations

Use the D/160SC-LS-HIZ board to develop sophisticated, multifunction CT systems incorporating capabilities such as voice processing, speech recognition, and text-to-speech (TTS). The D/160SC-LS-HIZ board shares a common hardware and firmware architecture with other Dialogic SCbus based boards for maximum flexibility and scalability. You can add features or grow the system while protecting your investment in hardware and application code. Applications can be easily ported to lower or higher line-density platforms, with only minimum modifications.

The D/160SC-LS-HIZ board installs in IBM® PC AT® (ISA bus) and compatible computers (80386, 80486, and Pentium® processor-based PC platforms). The D/160SC-LS-HIZ board occupies a single expansion slot and up to 16 boards can be configured in a system with each board sharing the same interrupt level. The maximum number 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.



*MSI/960SC, D240/JCT-T1

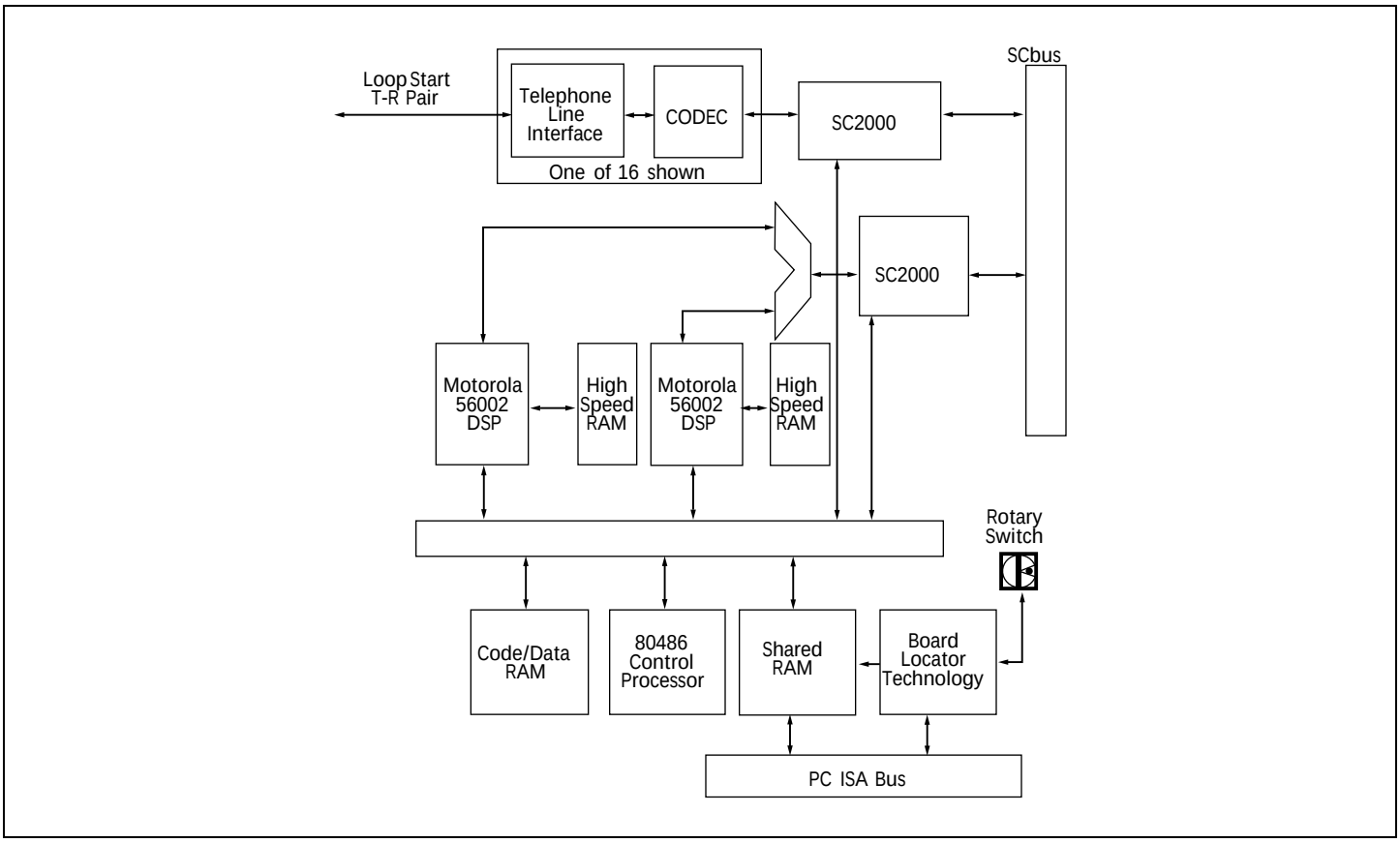
Software Support

The D/160SC-LS-HIZ board is supported by Dialogic System Software and Software Development Kits (SDKs) for Windows NT, UNIX, Solaris, and MS-DOS. These packages contain a set of tools for developing complex multichannel applications.

Applications

- Call logging
- Voice messaging
- Telemarketing/call center
- Transaction record

■ Functional Description



The D/160SC-LS-HIZ board connects 16 analog (loop start) telephone lines to 16 on-board call processing resources or to other resources via the SCbus. This board provides:

- interference suppression
- ring signaling control
- tone detection
- digitization and of voice files (via SCbus)

The signals from the 16 loop start telephone lines connected to the D/160SC-LS-HIZ board first pass through a high impedance telephone line interface that provides transient protection and electromagnetic interference (EMI) suppression (see block diagram). These telephone line interfaces use FCC-part 68 class B ring detection circuitry. This FCC-approved ring detector is less susceptible to spurious rings created by random voltage fluctuations on the network. Each interface also incorporates circuitry that protects against high-voltage spikes and adverse network conditions.

The telephone line interface applies the inbound signal including the ring to analog/digital inputs of a signal converter called a COder/DECoder (CODEC) for sampling and digitization. These digitized signals are sent to an SC2000™ chip where they are routed via the SCbus either to an onboard DSP or to an external resource on any of the 1024 SCbus time slots. This enables the application to reroute calls to any added resource, such as speech recognition, facsimile, or TTS.

The D/160SC-LS-HIZ board's high impedance telephone interface includes an on-hook audio path that detects Caller ID information. Depending on the level of service offered by the local telephone provider, Caller ID can include the date, time, caller's telephone number, and the name of the person calling (in some enhanced Caller ID environments). The on-hook audio path can also detect touchtones while the line is on-hook. This capability lets you use the D/160SC-LS-HIZ board behind PBXs that require on-hook touchtone detection for their signaling.

When onboard call processing resources are used, the network signals are extracted and passed to the onboard control processor, which can change channel status and communicate channel events to the application via a shared RAM and the host PC ISA bus.

The DSP processes the digitized voice data based on SpringWare firmware loaded in code/data RAM. Each DSP performs the following signal analysis and operations.

On the incoming data:

- applies 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
- detects silence to determine whether the line is quiet and the caller is not responding

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. The D/160SC-LS-HIZ board can record incoming signals with the telephony

interface in the high-impedance on-hook state. When replaying a stored file via the SCbus only, 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 the digitized voice responses to the CODEC, which is controlled by a pair of SC2000 chips. The CODEC converts the digitized voice into analog voice and transmits the voice response to the SCbus only.

When the system is initialized, SpringWare firmware is downloaded from the host PC to the board. It controls all board operations. SpringWare gives the board all of its intelligence and enables easy feature enhancements and upgrades.

The onboard control processor manages all operations of the D/160SC-LS-HIZ board via a local bus and interprets and executes commands from the host PC. This 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 telephony signaling before passing them to the application, and frees the DSP to perform signal processing. Communication between the processor and the host PC is via the 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 ISA bus. All operations are interrupt-driven to meet the demands of real-time systems.

The Board Locator Technology circuit operates in conjunction with a rotary switch that eliminates the need to set confusing jumpers or DIP switches.

■ Technical Specifications*

Number of ports	16
Max. boards/system	6 (MS-DOS); 16 (UNIX, Windows NT). Number may be limited by application and system performance.
Analog network interface	Onboard hi-z interface
Resource sharing bus	SCbus or PEB
Control microprocessor	Intel® 80486 GX @ 32.768 MHz, 0 wait state
Digital signal processors	Two Motorola DSP56002 @ 49 - 66 MHz, each with 32 K word private, 0 wait state SRAM

HOST INTERFACE:

Bus compatibility	IEEE P996 ISA compatible (IBM PC AT)
Bus speed	8 MHz typical
Bus mode	Automatically configures to 8- or 16-bit transfer mode
Shared memory	32 Kbytes page
Base addresses	8000h to E800h, on 32 K boundaries. All D/SC 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, 14, 15, software selectable. One IRQ line must be shared by all D/SC boards.
I/O ports	None

TELEPHONE INTERFACE:**

Trunk type	Hi-Z analog interface
Impedance	>1 Meg Ohm
Loop current range	N/A
Ring detection	40 to 130 Vrms, 15.3 to 68.0 Hz
Echo return loss	N/A
SNR	–40 dB
Cross talk coupling	–70 dB
Speech digitization	64 Kb/s, µ-law PCM (companding to ADPCM performed in SpringWare)
Freq. response	300 to 3400 Hz ±3 dB
Connector	DB-37

POWER REQUIREMENTS:

+5 VDC	1.5 A max.
–12 VDC	250 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, 13.3 in. long, 4.5 in. high (excluding edge connector)

SAFETY & EMI CERTIFICATIONS:

United States	UL: 1459, with optional adapter
Canada	CSA: 225 (by UL)
Estimated MTBF	163,000 hours per Bellcore Method I
Warranty	3 years standard

■ SpringWare Technical Specifications*

AUDIO SIGNAL:

Receive range	–40 to +2.5 dBm0 nominal, configurable by parameter**
Automatic gain control	Application can enable/disable. Above –18 dBm0 results in full scale recording, configurable by parameter.**
Silence detection	–38 dBm nominal, software adjustable**
Transmit level (weighted average)	N/A
Transmit volume control	N/A
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	OKI® ADPCM @ 6 kHz sampling
32 Kb/s	OKI® 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	N/A

DTMF TONE DETECTION™:

DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6
Dynamic range	–36 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
Acceptable twist and frequency variation	Meets Bellcore LSSGR Sec 6 and EIA 464 requirements
Noise tolerance	Meets Bellcore LSSGR Sec 6 and EIA 464 requirements for Gaussian, impulse, and power line noise tolerance
Cut-through	Local echo cancellation permits 100% detection with a >4.5 dB return loss line (only via SCbus)
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.

MF SIGNALING: **R1**

MF digits	0 to 9, KP, ST, ST1, ST2, ST3 per Bellcore LSSGR Sec 6, TR-NWT-000506 and CCITT Q.321
Transmit level	Complies with Bellcore LSSGR Sec 6, TR-NWT-000506
Signaling mechanism	Complies with Bellcore LSSGR Sec 6, TR-NWT-000506
Dynamic range for detection	–25 dBm0 to +3 dBm0 per tone
Acceptable twist	6 dB
Acceptable freq. variation	Less than ±1 Hz

■ SpringWare Technical Specifications* (cont.)

CALL PROGRESS ANALYSIS:

Busy tone detection	Default setting designed to detect 74 out of 76 unique busy/congestion tones used in 97 countries as specified by CCITT Rec, E., Suppl, #2. Default utilizes both frequency and cadence detection. Application can select frequency only for faster detection in specific environments.
Ring back detection	Default setting designed to detect 83 out of 87 unique ring back tones used in 96 countries as specified by CCITT Rec, E., Suppl, #2. Utilizes both frequency and cadence detection.
Positive Voice Detection™ accuracy	>99% based on tests on a database of real world calls in North America. Performance in other markets may vary.
Positive Voice Detection speed	Detects voice in as little as 1/10th of a second
Positive Answering Machine Detection™ accuracy	>85% based on tests on a database of real world calls in North America. Performance in other markets may vary.
Fax/modem detection	Preprogrammed
Intercept detection	Detects entire sequence of the North American tri-tone. Other intercept tone sequences can be programmed.
Dial tone detection before dialing	Application enable/disable; supports up to three different user definable dial tones; programmable dial tone drop out debouncing.

ANALOG CALLER IDENTIFICATION:

Applicable standards	Bellcore TR-TSY-000030 Bellcore TR-TSY-000031 TAS T5 PSTN1 ACLIP: 1994 (Singapore)
Modem standard	Bell 202 or V.23, serial 1200 b/s (simplex FSK signaling)
Receive sensitivity	–48 dBm (–50 dBv) to –1 dBm
Noise tolerance	Minimum 18 dB SNR over 0 to –48 dBm dynamic range for error-free performance
Data formats	Single Data Message (SDM) and Multiple Data Message (MDM) formats via API calls and commands
Line impedance	AC coupled 600 Ohm (@ 1.8 kHz) termination during Caller ID on-hook detection interval
Message formats	ASCII or binary SDM, MDM message content

ANALOG DISPLAY SERVICES INTERFACE (ADSI):

FSK generation per Bellcore TR-NWT-000030
CAS tone generation and DTMF detection per Bellcore TR-NWT-001273

*All specifications are subject to change without notice.
**Configurable to meet country-specific PTT requirements. Actual specification may vary from country to country for approved products.

Hardware System Requirements

- 80386, 80486, or Pentium IBM PC AT (ISA) bus or compatible computer. Operating system hardware requirements vary according to the number of channels being used.

Additional Components

- Multidrop SCbus cable
- Required: Station Adapter, 37-pin to 50-pin cable
- Optional: UL 1459 compliance adapter

Dialogic Offices

Dialogic World Headquarters

Dialogic Corporation
1515 Route Ten
Parsippany, NJ 07054-4596 USA
1-973-993-3000 (auto attendant)
1-973-993-3030 (live operator — 8:30 am to 5:30 pm EST)
fax: 1-973-993-309*3*

North America

1-800-755-4444
1-973-993-3030 (outside US and Canada)
fax: 1-973-631-9631
sales@dialogic.com
Dialogic Online Information Retrieval System (fax-on-demand)
1-800-755-5599 or 1-973-993-1063 (outside US and Canada)
Sales offices: Waltham, MA
Atlanta, GA; Schaumburg, IL
Austin, TX; Kirkland, WA
Santa Clara, CA; Lowell, MA

Canada

Dialogic Canada Ltd.
1033 Oak Meadow Road
Oakville, Ontario L6M 1J6
Canada
1-800-755-4444 • fax: 1-905-827-1857

Dialogic Latin America and Caribbean

Roque Saenz Peña 730
Tercer Piso Oficina 34
1035 Buenos Aires, Argentina
54-11-4328-1531 • fax: 54-11-4328-5425
dlac.sales@dialogic.com

Brazil

Av. Brigadero Faria Lima, 1306 Andar 5
01451-001 São Paulo, Brazil
55-11-3067-4787 • fax: 55-11-3067-4554

Dialogic Europe

Europe, Middle East, and Africa

Dialogic Telecom Europe N.V.-S.A.
Airway Park
Lozenberg 23 (3rd Floor)
B-1932 Sint-Stevens-Woluwe (Brussels)
Belgium
32-2-712-4311 • fax: 32-2-712-4300
info.europe@dialogic.com
DTE Online Information Retrieval System
32-2-712-4322
DTE BBS: 32-2-725-7846

France

Dialogic Telecom France
"Le Debussy"
77-81 Bld de la République
92250 La Garenne Colombes
France
33-1-41-30-88-88 • fax: 33-1-41-30-88-80

Germany, Switzerland, and Austria

Dialogic Telecom Deutschland GmbH
Industriestrasse 1
D-82110 Germering (Munich)
Germany
49-89-894-362-0 • fax: 49-89-894-362-77

Israel

Dialogic Israel
99 Medinat Hayehudim Str. 3rd floor
POB 12242
Herzelia Pituach, Israel 46766
972-9-9500-968 • fax: 972-9-9500-984

Italy

Dialogic Telecom Italia S.r.l.
Via Fermi, 20
I-20090 Assago (Milan)
Italy
39-2-488-2582 • fax: 39-2-488-1176

United Kingdom, Ireland, and Scandinavian Countries

Dialogic Telecom U.K. Ltd.
Dairy Walk
Hartley Wintney, Hampshire RG27 8XX
United Kingdom
ukhotmail@dialogic.com
44-1252-844-000 • fax: 44-1252-844-525

Spain

Dialogic Telecom España SL
Julián Camarillo, 29 D1-2
28037 Madrid
34-91-440-2840 • fax: 34-91-304-7807

Dialogic Asia/Pacific

Japan

Dialogic Systems K.K.
Carrot Tower 19F
4-1-1 Taishido
Setagaya-ku, Tokyo 154
Japan
phone: 81-3-5430-3252
fax: 81-3-5430-3373
japan.sales@dialogic.com
japan.tech@dialogic.com
Sales Office: Seoul

China

Dialogic Beijing Representative Office
Landmark Building, Suite 1308
8 North Dong-Sanhuan Road
Chaoyang District, Beijing 100004
People's Republic of China
phone: 86-10-6590-0055
fax: 86-10-6590-7989
china.sales@dialogic.com
china.tech@dialogic.com
Sales Office: Hong Kong

Singapore

Dialogic S.E.A. Pte. Ltd.
9 Temasek Boulevard
#37-02 Suntec Tower Two
Singapore 038989
phone: 65-339-9833
fax: 65-339-9211
asia.sales@dialogic.com
asia.tech@dialogic.com
Sales Offices: N. Sydney, Bangalore

Dialogic Web site

http://www.dialogic.com

Dialogic®
an Intel company

Printed on recycled paper.
00-6977-001 09/00 Printed in U.S.A

THE DIALOGIC EDGE®