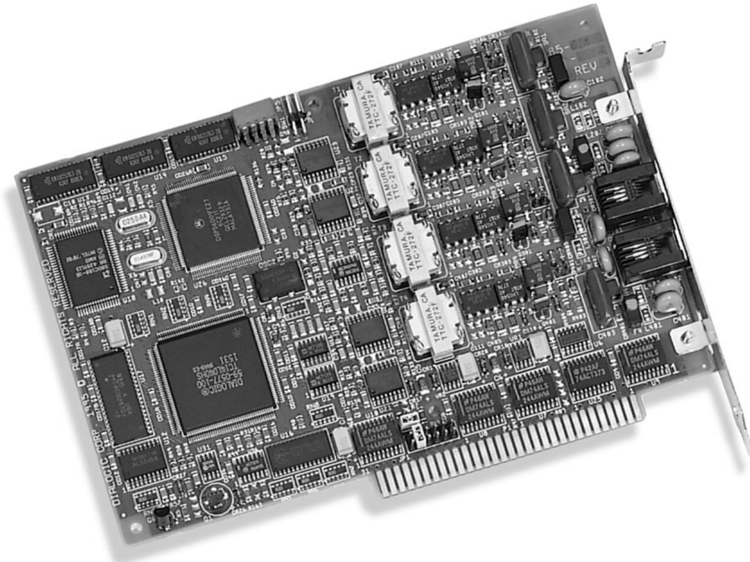


# DIALOG/4

## Half-Size, Four-Port Voice Processing Board

The DIALOG/4 board, with its half-size footprint, is an ideal solution for computer telephony installations that cannot take full-size voice boards. It provides four telephone line interface circuits that are approved 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. Multiple DIALOG/4 boards can be installed in a single PC chassis enabling system expansion up to 64 ports.

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.

Downloaded firmware algorithms, SpringWare™, executed by the on-board DSP, provide voice coding at 24 and 32 Kb/s ADPCM, and 48 and 64 Kb/s 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. Enhanced telephone circuit design and automatic gain control maintains recorded voice quality even at extremely low signal levels.

The DIALOG/4 voice board

- connects directly to the telephone line
- automatically answers calls
- detects touchtones
- plays voice messages to a caller
- digitizes, compresses, and records voice signals
- places outbound calls and automatically reports the results

all in real time on four independent channels

### FEATURES AND BENEFITS

- Four independent voice processing ports in a single, half-size PC ISA slot supporting low- to medium-density voice systems
- Dialogic downloadable signal and call processing firmware, SpringWare™, facilitates feature enhancement and provides field-proven performance based on over two million installed ports
- C language application program interfaces (APIs) for MS-DOS®, Windows® 95, Windows NT®, OS/2®, and UNIX® shorten your development cycle so you can get your applications to market faster
- Application generators available from third-party providers
- Configure multiple DIALOG/4™ boards in a single PC for easy and cost effective system expansion, and to build scalable systems from 4 to 64 ports
- Voice coding at dynamically selectable data rates, 24 Kb/s to 64 Kb/s, selectable on a channel-by-channel basis for optimal tradeoff in disk storage and voice quality



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

**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 & Caribbean**
(Buenos Aires)
54-1-328-1531 or -9943
fax: 54-1-328-5425
dlac.sales@dialogic.com

**Dialogic Europe**
(Brussels)
32-2-712-4311
fax: 32-2-712-4300
dte.sales@dialogic.com

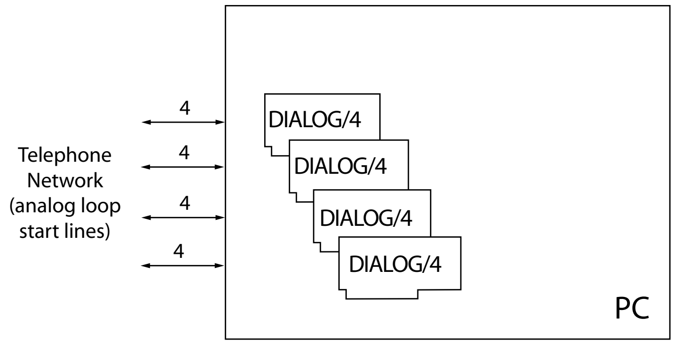
**Dialogic Asia/Pacific**
(Tokyo)
81-3-5430-3252
fax: 81-3-5430-3373
japan.sales@dialogic.com

FEATURES AND BENEFITS, (cont.)

- Enhanced telephone circuitry and automatic gain control maintains recording quality over a wide dynamic range
- PerfectDigit™ DTMF (touchtone) provides reliable detection during voice playback — allows callers to "type-ahead" through menus
- Patented outbound call progress analyzes outgoing call status quickly and accurately
- Supports PBXpert™ and PBXpert/32™, free utilities that simplify switch integration
- Lifetime warranty

APPLICATIONS

- Voice mail/voice messaging
- Interactive voice response
- Audiotex
- Inbound and outbound telemarketing
- Operator services
- Dictation
- Auto dialers
- Telecomputing servers
- Notification systems
- On-line data entry/query




## CONFIGURATIONS

The DIALOG/4 board shares a common hardware and firmware architecture with other Dialogic voice boards for maximum flexibility and scalability. You can easily add new features and/or expand the size of the system while protecting your original investment in hardware and application code. Applications can be ported to lower or higher line-density platforms with minimal modifications.

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

## SOFTWARE SUPPORT

The DIALOG/4 is supported by Dialogic System Software and SDK for MS-DOS®, Windows NT®, Windows 95®, OS/2®, and UNIX®. These packages contain a set of tools for developing complex multichannel applications. ■



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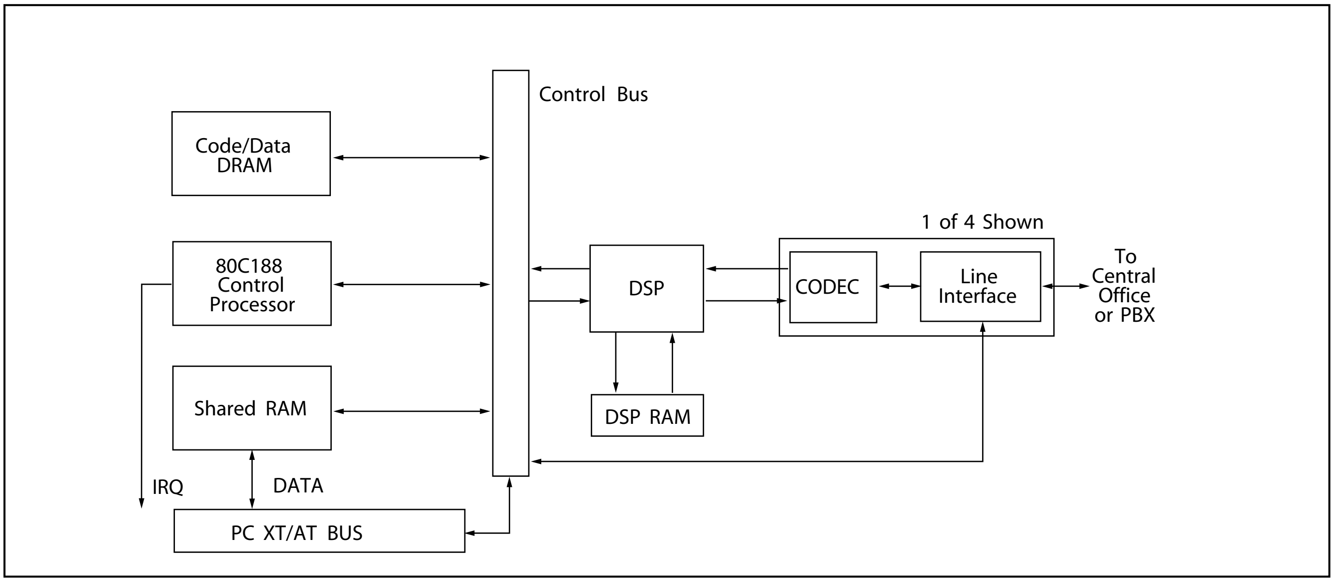
## Functional Description




The DIALOG/4 board uses a unique dual-processor architecture that combines the signal processing capabilities of a DSP with the decision-making and data movement functionality of a general-purpose 80C188 control microprocessor. This dual processor approach offloads many low-level decision-making tasks from the host computer enabling 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.

Each of four loop start telephone line interfaces on the DIALOG/4 board receives analog voice and telephony signaling information from the telephone network (see block diagram). Each line interface uses reliable, solid-state hook switches (no mechanical contacts) and 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 incorporates circuitry that protects against high-voltage spikes and adverse network conditions allowing applications to go off-hook any time during ring cadence without damaging the board.

Inbound telephony signaling (ring and loop current detection) are conditioned by the line interface and routed via a control bus to the control processor. The control processor responds to these signals, informs the application of telephony signaling status, and instructs the line interface to transmit outbound signaling (on-hook/off-hook) to the telephone network.

The audio voice signal from the network is sent through a bandpass filter, conditioned by the line interface, and then applied to a CODEC (COder/DECoder) circuit. The CODEC filters, samples, and digitizes the inbound analog audio signal and passes the digitized signal to a Motorola DSP.

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

- automatic gain control to compensate for variations in the level of the incoming audio signal
- applies an ADPCM (Adaptive Differential Pulse Code Modulation) or PCM (Pulse Code Modulation) 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-frequency tone
- silence detection 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 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, an answering machine, a facsimile machine, or a modem.



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japan.sales@dialogic.com

## Functional Description, (cont.)

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 can be selected on a channel-by-channel basis and can be changed each time a record or play function is initiated. Outbound processing is the reverse of inbound processing. The DSP processed speech is transmitted by 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 and passes it to the DSP which converts the file into digitized voice. The DSP sends digitized voice to the CODEC to be converted into analog voice and then to the line interface for transmission to the telephone network.

The on-board microprocessor controls all operations of the DIALOG/4 board 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 shared RAM that acts as an input/output buffer increasing the efficiency of disk file transfers. This RAM interfaces to the host PC via the XT/AT bus. All operations are interrupt-driven to meet the demands of real-time systems. All DIALOG/4 boards installed in the PC share the same interrupt line. When the system is initialized, SpringWare firmware to control all board operations is downloaded from the host PC to the on-board code/data RAM and DSP RAM. This downloadable firmware gives the board all of its intelligence and enables easy feature enhancement and upgrades. ■

## ■ Technical Specifications*

| | |
|---|---|
| Number of ports | 4 |
| Max. boards/system | 16 |
| Analog network interface | On-board loop start interface circuits |
| Microprocessor | 80C188 |
| Digital signal processor | Motorola DSP56001 |
| **HOST INTERFACE:** | |
| Bus compatibility | IBM PC XT/AT (ISA) |
| Bus speed | 4 to 12 MHz, 70 nsec back-to-back bus cycle |
| Shared memory | 8 KB page, switch selectable on 8 KB boundaries |
| Base addresses | D000h (default), A000h or C000h |
| Interrupt level | IRQ 2 to IRQ 7 jumper selectable; one IRQ is shared by all DIALOG/4 boards |
| **TELEPHONE INTERFACE‡:** | |
| Trunk type | Loop start (or ground start for answer only) |
| Impedance | 600 ohms nominal |
| Ring detection | 40 Vrms min; 15.3 to 68 Hz, 130 Vrms max. |
| Loop current range | 20 to 120 mA, dc (polarity insensitive) |
| Receive signal/noise ratio | 70 dB, referenced to −15 dBm |
| Crosstalk coupling | −70 dB at 1 kHz channel to channel |
| Frequency response | 300 Hz to 3400 Hz ±3 dB (transmit and receive) |
| Connector | Two RJ-14 type |
| **POWER REQUIREMENTS:** | |
| +5 VDC | .75 A |
| +12 VDC | 40 mA |
| −12 VDC | 40 mA |
| Operating temperature | 0°C to +50°C |
| Storage temperature | −20°C to +70°C |
| Humidity | 8% to 80% noncondensing |
| Form factor | PC (ISA) half size: 7 in. long, 0.652 in. wide, 4.5 in. high (excluding edge connector) |
| **REGULATORY CERTIFICATIONS:** | |
| United States | FCC part 68 ID#: EBUSA-65588-VM-E<br>UL: 143032 |
| Canada | DOC: 885-4452A<br>ULC: 143032 |
| Warranty | Lifetime |



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sales@dialogic.com

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dte.sales@dialogic.com

**Dialogic Asia/Pacific**
(Tokyo)
81-3-5430-3252
fax: 81-3-5430-3373
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## SpringWare Technical Specifications*

**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 and legal limit cap |
| 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 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 | Default set to –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<br>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 per tone into 600 ohm load impedance |



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## ■ SpringWare Technical Specifications* (cont.)

**DTMF TONE DETECTION (cont.):**

| | |
|---|---|
| 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 125 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 |

**MF SIGNALING:**

| | |
|---|---|
| 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-506 |
| Signaling mechanism | Complies with Bellcore LSSGR Sec 6, TR-NWT-506 |
| Dynamic range for detection | –25 dBm to –3 dBm per tone |
| Acceptable twist | 6 dB |
| Acceptable freq. variation | Less than ±1 Hz |

**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 uses both frequency and cadence detection; application can select frequency only for faster detection in specific environments |



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## SpringWare Technical Specifications* (cont.)

**CALL PROGRESS ANALYSIS, (cont.):**

| | |
|---|---|
| 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; uses both frequency and cadence detection |
| Positive Voice Detection™ accuracy | >98% based on tests on a database of real-world calls |
| Positive Voice Detection speed | Detects voice in as little as 1/10th of a second |
| Positive Answering Machine Detection™ accuracy | 80 to 90% based on application and environment |
| Fax/modem detection | Preprogrammed |
| Intercept detection | Detects entire sequence of the North American tri-tone; other SIT sequences can be programmed |
| Dial tone detection before dialing | Application enable/disable; supports up to three different user definable dialtones; programmable dialtone drop out debouncing |

**TONE DIALING:**

| | |
|---|---|
| DTMF digits | 0 to 9, *, #, A, B, C, D; 16 digits per Bellcore LSSGR Sec 6, TR-NWT-506 |
| MF digits | 0 to 9, KP, ST, ST1, ST2, ST3 |
| Frequency variation | Less than ±1 Hz |
| Rate | 10 digits/s max., configurable by parameter‡ |
| Level | –4.0 dBm per tone, nominal, configurable by parameter‡ |

**PULSE DIALING:**

| | |
|---|---|
| 10 digits | 0 to 9 |
| Pulse rate | 10 pulses/s, nominal, configurable by parameter‡ |
| Break ratio | 60% nominal, configurable by parameter‡ |

**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.
** Average speech mandates +16 dB peaks above average and preserves –13 dB valleys below average.
‡ Analog levels: 0 dBm0 corresponds to a level of +3 dBm at tip-ring analog point. Values vary depending on country requirements; contact your Dialogic Sales Engineer.

**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.



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