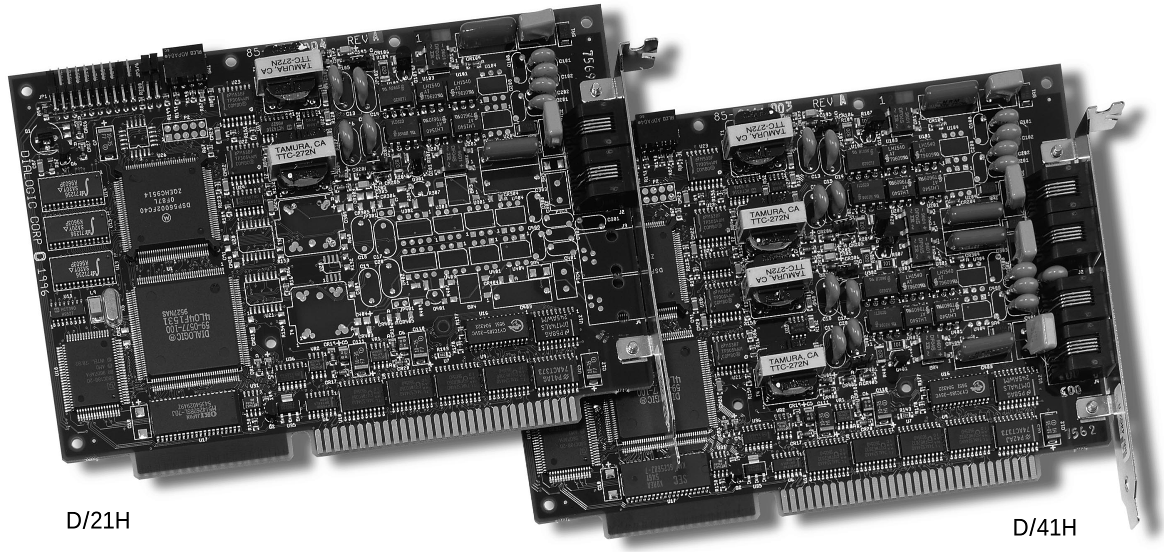

D/21H D/41H

D/21H D/41H

High-Performance 2- and 4-Port Voice Processing Boards

The four-line D/41H™ board and its two-line version, the D/21H™ board, are ideal for applications that need high-performance voice processing but don't require the large-scale system sophistication of SCbus™ or CT Bus™ based products. The D/21H and D/41H boards use the same Dialogic application programming interface (API) as their predecessors, making it easy to scale existing applications upward to take advantage of the power and features of these boards. The D/21H and D/41H boards also have improved voice quality and automatic gain control (AGC) over the legacy D/21D™ and D/41D™ boards. Even the weakest of telephone signals traveling over difficult telephone lines can be recorded and played back with complete clarity.

The D/21H and D/41H boards use the latest digital signal processor (DSP) voice processing technology, making them ideal for small- and medium-sized, server-based computer telephony (CT) systems — particularly under the Windows® operating systems. Windows support includes TAPI and WAVE APIs, which facilitate call control, recording, and playback of voice messages under the Microsoft Windows Open Services Architecture (WOSA). The D/21H and D/41H voice processing boards give Windows 95 and Windows NT® application developers a powerful platform for creating sophisticated interactive voice response (IVR) applications. The "H Series" boards also support use in MS-DOS®, OS/2®, and UNIX® operating system environments.

International Caller ID is supported on the D/21H and D/41H boards, allowing an application such as IVR to receive calling party information via a telephone trunk line. Caller ID is supported for North America (CLASS protocol), the United Kingdom (CLI protocol), and in Japan (CLIP protocol).

The Dialogic Global Dial Pulse Detection™ (DPD) algorithm is available for both boards, enabling application development for deployment in countries with limited touchtone telephone service. Global DPD™ is optimized for several countries and provides

FEATURES AND BENEFITS

- Freedom of choice: supports Windows® 95, Windows NT® (including TAPI/WAVE), MS-DOS®, OS/2®, and UNIX®
- Easily expands markets to satisfy international demands by providing a full suite of international telephone network approvals
- International Caller ID capable: supports North American Bellcore CLASS, UK CLI, and NTT CLIP
- Enables advanced call processing features for competitive differentiation by supporting software-based features such as Global Dial Pulse Detection™, TextTalk™ text-to-speech, SpeechWorks-Host™, continuous speech recognition, and PBXpert™ tone characterization utility
- Provides reliable DTMF detection during voice playback letting callers "type-ahead" through voice menus for quicker completion of call transactions
- Ensures reliability via call progress analysis which monitors outgoing call status quickly and accurately

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FEATURES AND BENEFITS (cont.)

- Offers flexible voice coding at dynamically selectable data rates, 24 to 64 Kb/s, selectable on a channel-by-channel basis for optimal tradeoff in disk storage and voice quality
- Offers superior voice quality through enhanced telephone circuitry and automatic gain control
- Enables developers to build cost-effective scalable systems from 2 to 64 ports
- Compatible with legacy telephone switches in the United Kingdom and Northern Europe that use Earth Break Recall
- Lets developers build flexible, cost-effective Internet telephony platforms for small-business applications

APPLICATIONS

- Voice messaging
- Automated attendant
- Interactive voice response
- Audiotex
- Inbound and outbound telemarketing
- Small call centers

superior dial pulse detection wherever it has been optimized.

Offered as additional software options, SpeechWorks-Host™ continuous speech recognition and TextTalk™ text-to-speech (TTS) software let you differentiate your offerings with state-of-the-art speech technologies for command and control of advanced IVR and unified messaging applications.

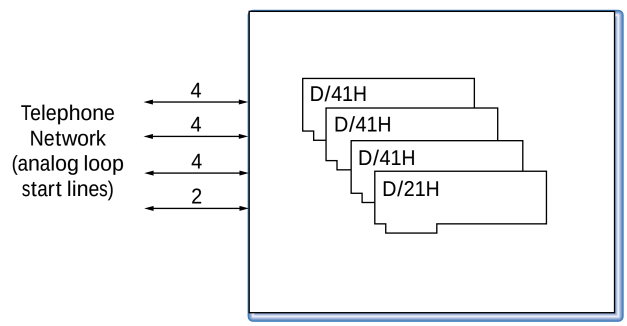
The D/21H board can also be utilized as a cost-effective platform to develop Internet telephony applications that are ideal for the small-business environment.

With all of these advanced features in a half-size ISA board footprint, the D/21H and D/41H boards are perfect for client or small server system development. Both boards offer enhanced DSP power and memory capacity that not only provides a base level of performance for today's requirements, but also provides the "head room" for future application expansion through software-based technologies.

CONFIGURATIONS

Use the D/21H and D/41H boards to build sophisticated messaging and IVR CT systems with optional technologies, such as automatic speech recognition (ASR), Global DPD, TTS, and PBXpert™. These boards share a common hardware and firmware architecture with other Dialogic voice boards for maximum flexibility and scalability. More ports and new features can be added while protecting your original investment in hardware and application code. With only minimum modifications, applications can be easily ported to higher line density platforms.

The D/21H and D/41H boards install in IBM® PC XT®/AT® (ISA bus) and compatible computers (80386, 80486 or Pentium™ based PC platforms). Both boards provide everything required for building integrated, non-SCbus voice solutions, scalable from 2 to 64 ports.



SOFTWARE SUPPORT

The D/21H and D/41H boards are supported by Dialogic System Software and SDK packages for Windows 95, Windows NT, MS-DOS, OS/2, and UNIX. These SDKs contain all the documentation, demo code, and tools necessary for developing complex multichannel applications. ■

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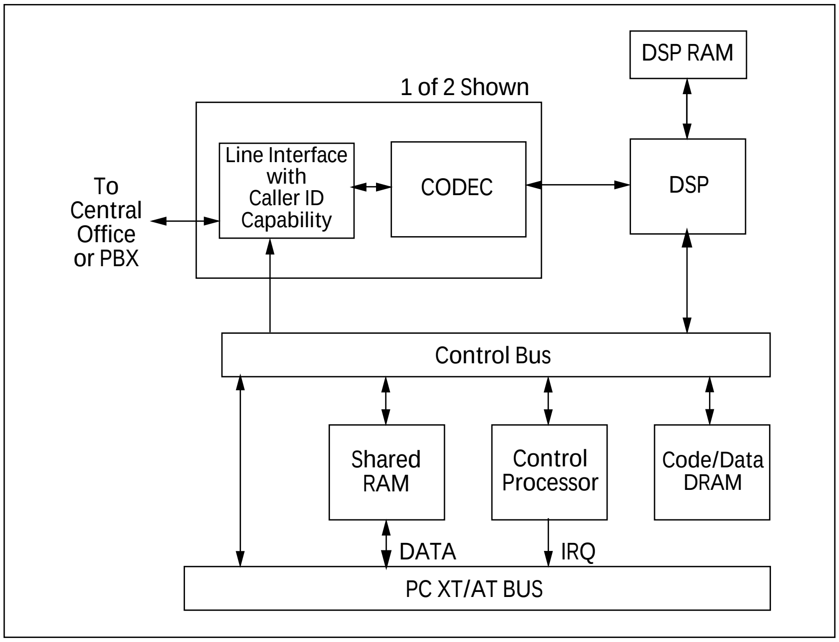
FUNCTIONAL DESCRIPTION



The D/21H and D/41H voice processing boards build on the patented Dialogic dual-processor architecture that combines the signal processing capabilities of a DSP with the decision-making and data movement functionality of a general-purpose control microprocessor by using faster processors and considerably more memory. This dual-processor approach offloads 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.

Each of the two (D/21H) or four (D/41H) analog loop start interfaces receives analog voice and telephony signaling information from the telephone network (see Block Diagram). Each telephone 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 also incorporates circuitry that protects against high-voltage spikes and adverse network conditions and lets applications go off-hook any time during ring cadence without damaging the board.

Part of the telephone interface for the D/21H and D/41H boards 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 (in some enhanced caller ID environments) the name of the person calling. The on-hook audio path can also detect touchtones while the line is on-hook. This capability lets the board operate behind PBXs that require on-hook touchtone detection for their signaling.

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Inbound telephony signaling (ring detection 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 bandpass filtered and conditioned by the line interface and then applied to a COder/DECoder (CODEC) circuit. The CODEC filters, samples, and digitizes the inbound analog audio signal and passes this digitized audio 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:

- uses AGC to compensate for variations in the level of the incoming audio signal. The D/21H and D/41H boards also include special circuitry to detect and amplify extremely weak line signals due to harsh telephone line conditions or back-to-back local loops often found in 800 service (toll-free) scenarios.
- 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-frequency tone
- uses 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

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 popular 11 kHz, 8-bit linear multimedia WAVE format is also supported on the D/21H and D/41H voice boards. 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 playing back 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 the 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 D/21H and D/41H boards 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 signaling 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 and thus increases 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 D/21H and D/41H boards installed in the PC share the same interrupt line. When the system is initialized, SpringWare firmware is downloaded from the host PC to the on-board code/data RAM and DSP RAM to control all board operations. This downloadable firmware gives the board all of its intelligence and enables easy feature enhancement and upgrades. ■

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Technical Specifications*

Number of ports	2 (D/21H) or 4 (D/41H)
Max. boards/system	16
Analog network interface	On-board loop start interface circuits
Microprocessor	Intel® 80C188
Digital signal processor	Motorola DSP56002
HOST INTERFACE:	
Bus compatibility	IBM PC XT/AT (ISA)
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, 3, 4, 5, 7, 10, 11, 12, jumper selectable. One IRQ is shared by all boards (D/21H or D/41H).
TELEPHONE INTERFACE:	
Trunk type	Loop start (or ground start for answer only)
Impedance	600 Ohms nominal
Ring detection	25 Vrms min., 15.3 to 68 Hz, 150 Vrms max.
Loop current range	20 to 120 mA, dc (polarity insensitive)
Crosstalk coupling	–70 dB at 3 kHz channel-to-channel
Frequency response	300 Hz to 3400 Hz ±3 dB (transmit and receive)
Connector	Two (D/21H) or four (D/41H) RJ-11
POWER REQUIREMENTS:	
+5 VDC	500 mA
+12 VDC	35 mA
–12 VDC	35 mA
Operating temperature	0°C to +50°C
Storage temperature	–20°C to +70°C
Humidity	8% to 80% noncondensing
Form factor	PC XT (ISA); 7.9 in. long, 0.75 in. wide, 3.85 in. high (excluding edge connector)
REGULATORY CERTIFICATIONS:	
United States	FCC part 68 ID#: EBZUSA-65588-VM-E REN: 1.0B UL: E143032
Canada	IC CS-03, 885 4452 A Load number: 5 ULC: E143032
Warranty	Lifetime

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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 –30 dBm results in full scale recording, configurable by parameter†.
Silence detection	–40 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 data rates. Adjustment range: ±50%, adjustable through application or programmable DTMF control.

WAVE AUDIO:

Supports 11 kHz linear PCM, 8-bit mono mode (available only when running Windows)

DTMF TONE DETECTION:

DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6
Dynamic range	Programmable, default set at –36 dBm to +0 dBm per tone
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 +0 dBm per tone

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SpringWare Technical Specifications* (cont.)

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-000506
Signaling mechanism	Complies with Bellcore LSSGR Sec 6, TR-NWT-000506
Dynamic range for detection	–25 dBm to –1 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.
Ringback detection	Default setting designed to detect 83 out of 87 unique ringback 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 tritone. Other SIT 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.

TONE DIALING:

DTMF digits	0 to 9, *, #, A, B, C, D; 16 digits per Bellcore LSSGR Sec 6, TR-NWT-000506
MF digits	0 to 9, KP, ST, ST1, ST2, ST3
Frequency variation	±0.5% of nominal frequency
Rate	10 digits/s max., configurable by parameter†
Level	–5 dBm per tone, nominal, configurable by parameter†

PULSE DIALING:

10 digits	0 to 9
Pulsing rate	10 pulses/s, nominal, configurable by parameter†
Break ratio	60% nominal, configurable by parameter†

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SpringWare Technical Specifications* (cont.)

ANALOG CALLER IDENTIFICATION:

Applicable standards	Bellcore TR-TSY-000030 Bellcore TR-TSY-000031 TAS T5 PSTN1 ACLIP: 1994 (Singapore) British Telecom SIN 242 (Issue 01) British Telecom SIN 227 (Issue 01) Japan NTT CLIP
Modem standard	Bell 202 or V.23, serial 1200 b/s (simplex FSK signaling)
Receive sensitivity	–48 dBm 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	600 Ohm
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

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.

‡ Average speech mandates +16 dB peaks above average and preserves –13 dB valleys below average.

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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