ภาคผนวก

ภาคผนวก ก

แนวคำถามในการสัมภาษณ์ ผู้ที่เกี่ยวข้องกับระบบ

แนวคำถามในการสัมภาษณ์ ความพึงพอใจของผู้ใช้บริการ

1. ในแต่ละครั้งที่แจ้งเหตุขัดข้องเข้ามามีความประสบความสำเร็จเพียงใดหากไม่ประสบความสำเร็จเกิดขึ้นเนื่องจากสาเหตุใด
2. เวลาในการตอบรับหลังจากที่แจ้งเหตุขัดข้องเข้ามารวดเร็วเพียงใด
3. คำแนะนำขั้นตอนต่าง ๆ ที่ระบบแจ้งเข้าใจเพียงใด
4. ถ้อยคำ , น้ำเสียง ที่ระบบตอบรับอัตโนมัติโต้ตอบเหมาะสมเพียงใด
5. ระยะเวลาที่ท่านใช้ในการแจ้งเหตุขัดข้องแต่ละครั้งนั้นเหมาะสมเพียงใด
6. ความมั่นใจที่มีต่อระบบรับแจ้งเหตุขัดข้อง 17 อัตโนมัติว่าหมายเลขโทรศัพท์ที่แจ้งเข้าไปจะได้รับการแก้ไข
7. ความคาดหวังว่าหลังจากที่แจ้งเหตุขัดข้องเข้าระบบ กองงานตรวจแก้จะมาแก้ไขหมายเลขโทรศัพท์ให้สามารถใช้งานได้ตามปกติรวดเร็วเพียงใด
8. เมื่อหมายเลขโทรศัพท์ได้รับการแก้ไขจนสามารถใช้งานได้เป็นปกติ ได้รับแจ้งจากองค์การโทรศัพท์หรือไม่อย่างไร
9. ทัศนคติที่มีต่อการองค์การโทรศัพท์แห่งประเทศไทยที่นำเอาระบบตอบรับอัตโนมัติมาใช้ในการรับแจ้งเหตุขัดข้อง
10. เมื่อหมายเลขโทรศัพท์ขัดข้องครั้งต่อไปจะเลือกแจ้งเหตุขัดข้องโดยวิธีใด

แนวคำถามในการสัมภาษณ์ ความพึงพอใจของเจ้าหน้าที่ 17

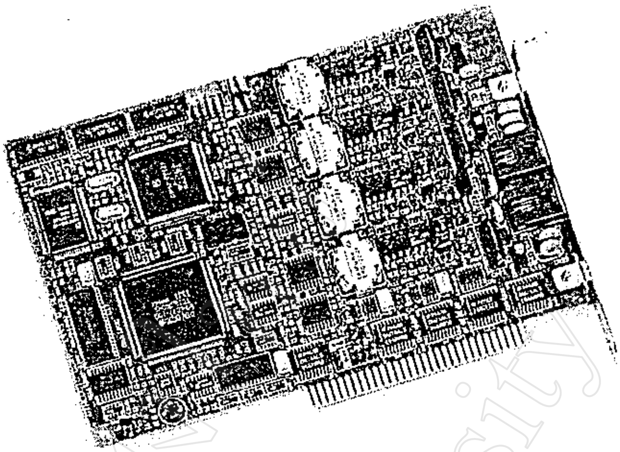
1. ทัศนคติที่มีต่อการองค์การโทรศัพท์แห่งประเทศไทยที่นำเอาระบบตอบรับอัตโนมัติมาใช้ในการรับแจ้งเหตุขัดข้อง
2. ระบบอัตโนมัติช่วยให้การบริการผู้ใช้บริการที่แจ้งเหตุขัดข้องเข้ามาและกองงานตรวจแก้ที่ต้องการข้อมูลคู่สายได้เร็วขึ้นได้มากน้อยเพียงใด
3. ระบบอัตโนมัติช่วยให้ทำงานได้มากขึ้นเพียงใด
4. หากไม่สามารถแก้ไขปัญหาที่เกิดขึ้นกับระบบรับแจ้งเหตุขัดข้อง 17 ได้จะแจ้งใคร เพื่อขอความช่วยเหลือ
5. ผู้ที่ท่านร้องขอความช่วยเหลือจะดำเนินการแก้ปัญหาให้รวดเร็วเพียงใด
6. ต้องการให้มีการปรับปรุงระบบตอบรับอัตโนมัติ 17 ด้านใดบ้าง

แนวคำถามในการสัมภาษณ์ ความพึงพอของกองงานตรวจแก้

1. ใช้บริการใดบ้างจากระบบรับแจ้งเหตุขัดข้องอัตโนมัติ
2. ความถูกต้องของการให้ข้อมูลคู่สายที่ระบบแจ้งทราบมีความถูกต้องหรือไม่เพียงใด
3. ระยะเวลาใช้บริการข้อมูลจากระบบตอบรับอัตโนมัติเหมาะสมหรือไม่อย่างไร
4. ระบบอัตโนมัติช่วยให้แก้ไขหมายเลขโทรศัพท์ที่ขัดข้องได้เร็วขึ้นได้หรือไม่เพียงใด เพียงใด
5. ระบบอัตโนมัติช่วยให้ท่านทำงานได้มากขึ้นหรือไม่เพียงใด
6. เมื่อต้องการสอบถามข้อมูลคู่สายหรือแจ้งการคืนดีหมายเลขโทรศัพท์ครั้งต่อไป จะเลือกใช้วิธีการใด
7. ความคิดเห็นและทัศนคติ ต่อการที่องค์การโทรศัพท์ นำเอาระบบตอบรับอัตโนมัติมาใช้ในการรับแจ้งเหตุขัดข้อง

ภาคผนวก ข

รายละเอียดของแผงวงจรตอบรับอัตโนมัติ



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

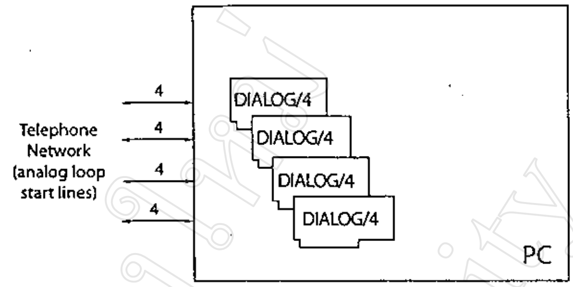
Dialogic Asia/Pacific
(Tokyo)
81-3-5430-3252
fax: 81-3-5430-3373
japan.sales@dialogic.com

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. ■

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

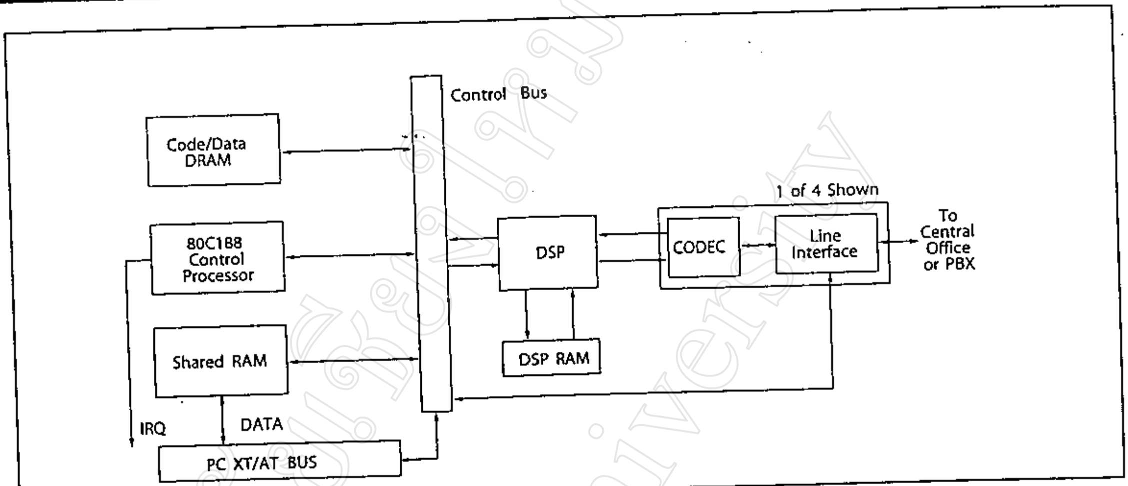
Dialogic North American Sales
1-800-755-4444
or 1-973-993-3030
fax orders: 1-973-631-9631
sales@dialogic.com

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

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.

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

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 UL: 143032
Canada	DOC: 885-4452A ULC: 143032
Warranty	Lifetime

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

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 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

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

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

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

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.

00-2419-004

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

2. Using the Voice Reference Library

This chapter provides a description of the voice library and the programming requirements. The following topics are included:

- Voice Library and its function categories (*Section 2.1*)
- Programming requirements for the Voice Library (*Section 2.2*).

2.1. Voice Library

The Voice Library functions provide an interface to the Voice Device Driver. The functions can be divided into the following major categories:

Device Management	• open and close devices
Configuration	• alter configuration of devices
I/O	• transfer data to and from devices
Convenience	• simplify play and record
Call Status Transition Event	• set and monitor events on devices
Route	• for SCbus boards, connect the receive (listen) channel of an SCbus board to an SCbus time slot; the transmit of each channel device is connected to a unique and unchangeable SCbus time slot at system initiation and download.
Global Tone Detection	• enable user-defined tone detection
Global Tone Generation	• enable user-defined tone generation
R2MF Convenience	• detect and generate R2MF tones
Speed and Volume	• enable play-speed and play-volume control
Convenience	• convenience functions for adjusting speed and volume control
Structure Clearance	• clear data structures
Extended Attribute	• retrieve device information

This section lists the functions that belong to each category and describes the characteristics of each category.

In the *Function Reference* (*3. Voice Function Reference*) each function is described in detail, and the function header includes the category to which the function belongs.

2.1.1. Device Management Functions

dx_close( )	• close a board or channel
dx_open( )	• open a board or channel

The Device Management functions open and close devices (boards and channels). For SCbus configurations using a D/240SC-T1 or D/300SC-E1 board, each board comprises a digital interface device with independent channels/time slots (dtiBxTx) and a voice device with independent channels (dxxxBxCx); where B is followed by the unique board number, C is followed by the number of the voice device channel (1 to 4) and T is followed by the number of the digital interface device time slot (digital channel)(1 to 24 for T-1; 1 to 30 for E-1).

Before you can use any of the other library functions on a device, that device must be opened. When the device is opened using **dx_open( )** the function returns a unique Dialogic device handle. The handle is the only way the device can be identified once it has been opened. The **dx_close( )** function closes a device via its handle.

Device Management functions do not cause a device to be busy. In addition, the Device Management functions will work on a device whether the device is busy or idle.

NOTES:

1. Issuing a **dt_open( )**, **dx_open( )**, **dt_close( )** or **dx_close( )** while the device is being used by another process will not affect the current operation of the device.
2. The device handle which is returned is Dialogic defined. The device handle is not a standard Windows NT file descriptor. Any attempts to use operating system commands such as **read( )**, **write( )**, or **ioctl( )** will produce unexpected results.
3. In an application that starts a process, the device handle is not inheritable by the child process. Devices must be opened in the child process.

2.1.2. Configuration Functions

dx_clrdigbuf( )	• clear the firmware digit buffer
dx_getparm( )	• get a board/channel device parameter
dx_setdigtyp( )	• set digit collection type
dx_sethook( )	• set hookswitch state
dx_setparm( )	• set device parameters
dx_wtring( )	• wait for number of rings

Configuration functions allow you to alter, examine, and control the physical configuration of an open device. The configuration functions operate on a device only if the device is idle. All configuration functions cause a device to be busy and return the device to an idle state when the configuration is complete. See *Section 2.2.3. Busy and Idle States* for information about busy and idle states.

NOTE: The **dx_sethook( )** function can also be classified as an I/O function and can be run asynchronously or synchronously.

2.1.3. I/O Functions

dx_dial( ) (enable/disable call analysis)	• dial an ASCIIZ string of digits
dx_getdig( )	• get digits from channel digit buffer
dx_play( )	• play voice data from one or more sources
dx_playiottdata( )	• play voice data from multiple sources
dx_rec( )	• record voice data to one or more destinations
dx_reciottdata( )	• record voice data to multiple destinations
dx_setdigbuf( )	• set digit buffering mode
dx_stopch( )	• stop current I/O
dx_wink( )	• wink a channel

NOTES: 1. **dx_playtone( )**, which is grouped with the Global Tone eneration functions, is also an I/O function and all I/O characteristics apply.

2. **dx_sethook( )**, which is grouped with the Configuration functions, is also an I/O function and all I/O characteristics apply.
3. **dx_wink( )**, cannot be called for a digital T-1 configuration that includes a D/240SC-T1 board. Transparent signaling for SCbus digital interface devices is not supported in System Release 4.1SC.

The purpose of an I/O function is to transfer data to and from an open idle channel. All I/O functions cause a channel to be busy while data transfer is taking place and return the channel to an idle state when data transfer is complete. The **dx_stopch( )** function stops any other I/O function, except **dx_dial( )** (see **dx_dial( )** and **dx_stopch( )** in the *Chapter 3. Voice Function Reference* for information).

I/O functions can be run synchronously or asynchronously. When running synchronously, they return after completing successfully or after an error. When running asynchronously they will return immediately to indicate successful initiation (or an error), and continue processing until a termination condition is satisfied. See the *Standard Runtime Library Programmer's Guide for Windows NT*, for a full discussion on asynchronous and synchronous operation.

A set of termination conditions can be specified for I/O functions (except **dx_stopch( )** and **dx_wink( )**). These conditions dictate what events will cause an I/O function to terminate. The termination conditions are specified just before the I/O function call is made. Obtain termination reasons for I/O functions by calling the Extended Attribute function **ATDX_TERMMSK( )**. See *Section* 2.2.4. I/O Terminations for information on I/O terminations.

NOTE: The **dx_stopch( )** function will not stop all I/O functions. Do not use this function to stop **dx_wink( )** or **dx_dial( )**(without Call Analysis enabled). See *Chapter 3. Voice Function Reference* for more information on these functions.

2.1.4. Convenience Functions

dx_playf( )	• play voice data from a single file
dx_playvox( )	• play a VOX file
dx_playwav( )	• play a WAVE file
dx_recf( )	• record voice data to a single file
dx_recvox( )	• record voice data to a single VOX file
dx_recwav( )	• record voice data to a single WAVE file

These functions simplify synchronous play and record.

dx_playf( ) performs a playback from a single file by specifying the filename. The same operation can be done by using **dx_play( )** and supplying a *DX_IOTT* structure with only one entry for that file. Using **dx_playf( )** is more convenient for a single file playback, because you do not have to set up a DX_IOTT structure for the one file, and the application does not need to open the file. **dx_playvox( )**, **dx_playwav( )**, **dx_recvox( )**, **dx_recwav( )**, and **dx_recf( )** provide the same single-file convenience for the **dx_playiottdata( )**, **dx_reciottdata( )**, and **dx_rec( )** function.

Source code is included for **dx_playf( )** and **dx_recf( )** in the function descriptions in *Chapter 3. Voice Function Reference*.

NOTE: **dx_playf( ), dx_playvox( ), dx_playwav( ), dx_recf( ), dx_recvox( )** and **dx_recwav( )** run synchronously only.

2.1.5. Call Status Transition Event Functions

dx_getevt( )	• get call status transition event
dx_setevtmsk( )	• set call status transition event notification

Call Status Transition (CST) Event functions set and monitor Call Status Transition events that can occur on a device. Call Status Transition events indicate changes in the status of the call. For example, if rings were detected, if the line went onhook or offhook, or if a tone was detected. The full list of Call Status

Transition events is contained in *Section* 4.1.3. DX_CST - *call status transition structure* which describes the Call Status Transition structure (DX_CST).

dx_setevtmsk( ) enables detection of CST event(s).

dx_getevt( ) retrieves events in a synchronous environment. To retrieve CST events in an asynchronous environment, use the Standard Runtime Library's Event Management functions.

2.1.6. SCbus Routing Functions

See the *SCbus Routing Function Reference for Windows NT* for function descriptions and the nomenclature used to identify devices, channels and time slots in an SCbus configuration. The SCbus routing functions can only be used in SCbus configurations.

2.1.7. Global Tone Detection Functions

dx_addtone( )	• add a user-defined tone
dx_blddt( )	• build a dual frequency tone description
dx_blddtcad( )	• build a dual freqency tone cadence description
dx_bldst( )	• build a single frequency tone description
dx_bldstcad( )	• build a single frequency tone cadence description
dx_deltones( )	• delete user-defined tones
dx_enbtone( )	• enable detection of user-defined tones
dx_distone( )	• disable detection of user-defined tones
dx_setgtdamp( )	• sets amplitudes used by Global Tone Detection (GTD)

Use the Global Tone Detection (GTD) functions to define and enable detection of single and dual frequency tones that fall outside those automatically provided with the Voice Driver. This includes tones outside the standard DTMF range of 0-9, a-d, * and #.

The GTD **dx_blddt( )**, **dx_blddtcad( )**, **dx_bldst( )**, and **dx_bldstcad( )** functions define tones which can then be added to the channel using **dx_addtone( )**. This enables detection of the tone on that channel.

See the *Voice Features Guide for Windows NT* for a full description of Global Tone Detection.

2.1.8. Global Tone Generation Functions

dx_bldtngen( )	• build a user-defined tone generation template
dx_playtone( )	• play a user-defined tone

Use Global Tone Generation functions to define and play single and dual tones other than those automatically provided with the Voice driver.

dx_bldtngen( ) defines a tone template structure, *TN_GEN*. **dx_playtone( )** can then be used to generate the tone.

See the *Voice Features Guide for Windows NT* for a full description of Global Tone Generation, and see 4. Voice Data Structures and Device Parameters for a description of the TN_GEN structure.

2.1.9. R2MF Convenience Functions

r2_creatfsig( )	• create R2MF forward signal tone
r2_playbsig( )	• play R2MF backward signal tone

These are convenience functions which enable detection of R2MF forward signals on a channel, and play R2MF backward signals in response. For more information about Voice Support for R2MF, see the *Voice Features Guide for Windows NT*.

2.1.10. Speed and Volume Functions

dx_adjsv( )	• adjust speed or volume
dx_clrsvcond( )	• clear speed or volume digit adjustment conditions

dx_setsvcond( )	• set speed or volume digit adjustment conditions
dx_getcursv( )	• get current speed and volume settings
dx_getsvmt( )	• get Speed/Volume Modification Table
dx_setsvmt( )	• set Speed/Volume Modification Table

NOTE: Speed and Volume Control are available on D/41ESC, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1, and D/320SC boards.

Use these functions to adjust the speed and volume of the play. A 21-entry Speed Modification Table and Volume Modification Table is associated with each channel. This table can be used for increasing or decreasing the speed or volume. This table has default values which can be changed using the **dx_setsvmt( )** function.

dx_adjsv( ) and **dx_setsvcond( )** both use the Modification Table to adjust speed or volume; **dx_adjsv( )** adjusts speed or volume immediately, and **dx_setsvcond( )** sets conditions (such as a digit) for speed or volume adjustment. **dx_clrsvcond( )** to clear the speed or volume conditions.

dx_getcursv( ) retrieves the current speed or volume settings. **dx_getsvmt( )** retrieves the settings of the current Speed or Volume Adjustment Table.

See the *Voice Features Guide for Windows NT* for more information about voice software support for speed and volume.

2.1.11. Speed and Volume Convenience Functions

dx_addspddig( )	• add speed adjustment digit
dx_addvoldig( )	• add volume adjustment digit

dx_addspdig( ) and **dx_addvoldig( )** are convenience functions that specify a digit and an adjustment to occur on that digit, without having to set any data structures. These functions use the default settings of the Speed/Volume Modification Tables.

2.1.12. PerfectCall Call Analysis Functions

dx_chgdur( )	• change PerfectCall Call Analysis signal duration
dx_chgfreq( )	• change PerfectCall Call Analysis signal frequency
dx_chgrepcnt( )	• change PerfectCall Call Analysis signal repetition count
dx_initcallp( )	• initialize PerfectCall Call Analysis on a channel
dx_chg( )	• functions can be used to change the definition of default PerfectCall Call Analysis tones.
dx_initcallp( )	• enables PerfectCall Call Analysis.

2.1.13. Structure Clearance Functions

dx_clrcap( )	• clear DX_CAP structure
dx_clrtpt( )	• clear DV_TPT structure

These functions do not affect a device. The **dx_clrcap( )** and **dx_clrtpt( )** functions provide a convenient method for clearing the DX_CAP and DV_TPT Voice Library data structures. These structures are discussed in *Chapter 4. Voice Data Structures and Device Parameters*.

2.1.14. Extended Attribute Functions

ATDX_ANSRSIZ( )	• Returns duration of answer detected during
Call Analysis	
ATDX_BDNAMEP( )	• Returns pointer to the device name string
ATDX_BDTYPE( )	• Returns board type
ATDX_BUFDIGS( )	• Returns number of digits in firmware since last dx_getdig( ) for a given channel
ATDX_CHNAMES( )	• Returns pointer to an array of channel name strings
ATDX_CHNUM( )	• Returns channel number on board associated with the channel device handle

ATDX_CONNTYPE( )	• Returns connection type for a call
ATDX_CPERROR( )	• Returns call analysis error
ATDX_CPTERM( )	• Returns last call analysis termination
ATDX_CRTNID( )	• Returns the identifier of the tone that causedthe most recent Call Analysis termination
ATDX_DEVTYPE( )	• Returns device type
ATDX_DTNFAIL( )	• Returns the dial tone character that indicates which dial tone Call Analysis failed to detect
ATDX_FRQDUR( )	• Returns duration of first frequency
ATDX_FRQDUR2( )	• Returns duration of 2nd SIT tone frequency
ATDX_FRQDUR3( )	• Returns duration of 3rd SIT tone frequency detected
ATDX_FRQHZ( )	• Returns frequency of first detected tone
ATDX_FRQHZ2( )	• Returns frequency of second detected SIT tone
ATDX_FRQHZ3( )	• Returns frequency of third detected SIT tone
ATDX_FRQOUT( )	• Returns % of frequency out of bounds detected during Call Analysis
ATDX_FWVER( )	• Returns firmware version
ATDX_HOOKST( )	• Returns current hook status
ATDX_LINEST( )	• Returns current line status
ATDX_LONGLOW( )	• Returns duration of longer silence detected during Call Analysis
ATDX_PHYADDR( )	• Returns physical address of board
ATDX_SHORTLOW( )	• Returns duration of shorter silence detected during Call Analysis
ATDX_SIZEHI( )	• Returns duration of non-silence detected during Call Analysis
ATDX_STATE( )	• Returns current state of the device
ATDX_TERMMSK( )	• Returns termination bitmap

ATDX_TONEID( )	• Returns the tone id
ATDX_TRCOUNT( )	• Returns last record or play transfer count

Voice Library Extended Attribute functions return information specific to the Voice device indicated in the function call. Many are related to specific Voice features:

Basic Call Analysis uses:

ATDX_ANSRSIZ( )
ATDX_CPERROR( )
ATDX_CPTERM( )
ATDX_FRQ( )
ATDX_LONGLOW( )
ATDX_SHORTLOW( )
ATDX_SIZEHI( )

PerfectCall Call Analysis uses:

ATDX_ANSRSIZ( )
ATDX_CPERROR( )
ATDX_CPTERM( )
ATDX_FRQ( )
ATDX_CRTNID( )
ATDX_DTNFAIL( )

The Call Status Transition event detection uses:

ATDX_HOOKST( )

Global Tone Detection uses:

ATDX_TONEID( )

ประวัติผู้เขียน

ชื่อ - สกุล	นายพนม มาลีพัตร
วัน เดือน ปี เกิด	1 มีนาคม 2514
ประวัติการศึกษา	สำเร็จการศึกษาระดับประกาศนียบัตรวิชาชีพชั้นสูง (ปวส.) แผนกเทคนิคคอมพิวเตอร์สถาบันเทคโนโลยีราชมงคล วิทยาเขตภาคพายัพ ปีการศึกษา 2534 สำเร็จการศึกษาระดับปริญญาวิศวกรรมศาสตรบัณฑิต สาขาอิเล็กทรอนิกส์- สื่อสาร คณะวิศวกรรมศาสตร์ สถาบันเทคโนโลยีราชมงคล กรุงเทพฯ ปีการศึกษา 2536
ประสบการณ์ทำงาน	พ.ศ. 2536 - 2539 อาจารย์ 1 ระดับ 4 ภาควิชาไฟฟ้า แผนกอิเล็กทรอนิกส์ วิทยาเขตภาคพายัพ สถาบันเทคโนโลยีราชมงคล พ.ศ. 2539 - 2540 วิศวกรประจำศูนย์บริการ บริษัท ชินวัตร คอมพิวเตอร์แอนด์คอมมิวนิเคชั่น จำกัด (มหาชน) พ.ศ. 2540 - ปัจจุบัน วิศวกร 5 งานเทคโนโลยีสารสนเทศ ส่วนอำนวยการฝ่ายโทรศัพท์ภาคเหนือ องค์การโทรศัพท์แห่งประเทศไทย