

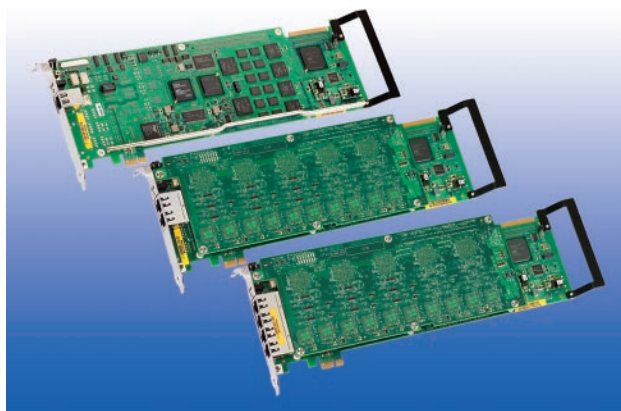
## Products Discussed in This Datasheet

- Dialogic® DM/V300BTEPEQ Media Board
- Dialogic® DM/V600BTEPEQ Media Board
- Dialogic® DM/V1200BTEPEQ Media Board

## With One, Two, or Four ISDN PRI Trunks

The Dialogic® DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ Media Boards are among the industry's most powerful media platforms for developers seeking to rapidly build and globally deploy some of the highest density media server solutions for the enterprise and public networks.

They provide a truly universal port solution with a robust media feature set — including voice processing, speech recognition, fax, and conferencing capabilities — combined with an extensive suite of network protocols in a single PCI Express slot.



Features	Benefits
<b>Software selectable trunks to configure the board to be either T1 or E1</b>	Reduces the total cost of ownership by increasing flexibility, decreasing inventory, and simplifying the purchasing process and test effort
<b>Universal media loads offer mixed media resources including voice, fax, and conferencing</b>	By combining three boards into one, reduces development, inventory, and solution costs by eliminating the need for dedicated media boards
<b>Provides the ability to mix select protocols on each span</b>	Maximizes slot efficiency and reduces total cost of ownership in environments where there are multiple protocols (for example, call centers)
<b>Built on the industry-standard telephony bus — ECTF H.100 CT Bus</b>	Lets applications expand (up to 1200 ports per system) through access to other communications boards, such as IP telephony, ATM, HDSI, and SS7
<b>Supports TrueSpeech voice coder (a default coder with Windows® supported by Windows® Media Player)</b>	Lets developers play Internet content and develop unified messaging systems without creating and supporting custom clients
<b>Ability to select between 16 ms, 32 ms, and 64 ms echo cancellation tail on select media loads</b>	Longer tail lengths are useful for environments and applications when optimum audio quality and clarity is a necessity
<b>PCI Express interface</b>	Supports new full-length PCI Express form factor compatible with x1 slots

## Applications

- Messaging and enhanced services
- Wireless and fixed-line Short Message Service (SMS)
- Color ring back tone
- Voice portal
- Contact center and e-Business
- PC-PBX
- Audio conferencing server
- Web conferencing
- Switching and call completion
- Prepaid/debit card
- Gateway switch

The boards, which are H.100-compliant, are suitable for service providers and large enterprise applications.

As part of the next generation of media span products from Dialogic, these boards include a host of robust features including:

- Software selectable T1/E1 Digital Network Interfaces (DNIs)
- A-law and  $\mu$ -law conversion with Dialogic® System Release 6.0 PCI for Windows® and Dialogic® System Release 6.1 for Linux
- A universal media load offering simultaneous
  - Fax
  - Conferencing
  - Voice processing
- Improved media densities
- The ability to mix select protocols
- Hardware selectable non-loopback mode
- Different conferencing media loads from which to choose

Support for Continuous Speech Processing (CSP) technology — Dialogic's Digital Signal Processor (DSP)-based solution optimized for speech recognition — enables a friendlier user interface and seamless integration of speech recognition software from the leading speech technology vendors. CSP reduces system latency, increases recognition accuracy, and improves overall system response time for high-density speech solutions. Also available on select media loads is Enhanced Echo Cancellation (EEC), which offers the ability to select longer echo cancellation tail lengths of 32 ms and 64 ms (beyond the normal 16 ms) to further improve and refine audio quality.

Other features include silence-compressed streaming to the host, which improves performance by removing the silence when data is sent to the host CPU. In addition, streaming to the CT Bus lets echo cancelled data be streamed onto the TDM bus.

The onboard conferencing solution offers an advanced feature set, presenting both a satisfying conferencing experience for the end user and one that can be used to deploy network-grade conferencing systems with features, audio quality, and density comparable to typical proprietary solutions, but at significantly reduced costs. Its optimized algorithm prevents noise build-up and echo in the conference. It also equalizes participant voice volumes, offering optional Dual Tone Multi-Frequency (DTMF and touchtone) clamping to limit audible enter and exit tones. The DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ are available with two different types of conferencing — rich and standard — to optimize a conferencing solution based on system configuration and requirements. (See the Conferencing section for details.)

Another conferencing feature is bridging, also known as cascade conferencing. Bridging lets you bridge together conferences from different DSPs and boards, consuming just two conferencing resources per bridge. This maximizes the flexibility of your conferencing solution by letting you create high-density conferences where any party can speak, and be heard, by the other conference participants.

Powerful DSPs provide a rich set of media processing features, including various rates of voice compression, recording and playback, conferencing with echo cancellation and active talker algorithm, telephony tone signaling, reliable DTMF detection using local echo cancellation, and automated outbound call progress analysis with positive voice detection and positive answering machine detection.

The DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ are based on DM3 architecture, which provides an environment that accelerates application development while also providing a path for growth. With support for the Dialogic® R4 API, it is easy for these boards to interoperate with other CT Bus boards from Dialogic. Applications can be ported easily to lower or higher density platforms, or new features can be added with only minimum modifications — thus protecting investment in hardware and application code.

## Configurations

Use the DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ to develop sophisticated, mixed-media communication applications that include voice processing, speech recognition, fax, and conferencing. These boards occupy a single computer backplane slot and multiple boards can be installed in a single chassis. The maximum number of supported lines and features depends on the application type, call module, and host computer CPU. For media-intensive applications, 600 ports in a chassis are reasonable. For other applications, such as call completion, where media processing is less intensive, systems of 1200+ ports per chassis are possible.

These boards can operate in either terminate or hairpin configurations. In a terminate configuration (see Figure 1), the boards handle the processing of digital audio and telephony signaling. Additional system resources can access calls via the CT Bus. This configuration is suitable for voice messaging, unified messaging, voice portal, and Interactive Voice Response (IVR) applications.

In a hairpin configuration (see Figure 2), the boards are connected via the CT Bus and can continuously pass all T1/E1 time slots through to each other. This configuration can switch call traffic between separate T1 or E1 lines, or can be placed inline between a T1/E1 public network trunk and a digital switch. Calls on individual channels can either terminate at a call processing resource on a board, or “flow through” transparently from one board to the other. Even in “flow through” mode, these boards can still monitor the lines, listening for DTMF or voice commands. This configuration is well suited for call center, PC-PBX, voice portal, prepaid calling card, international callback, and telecom resale applications.

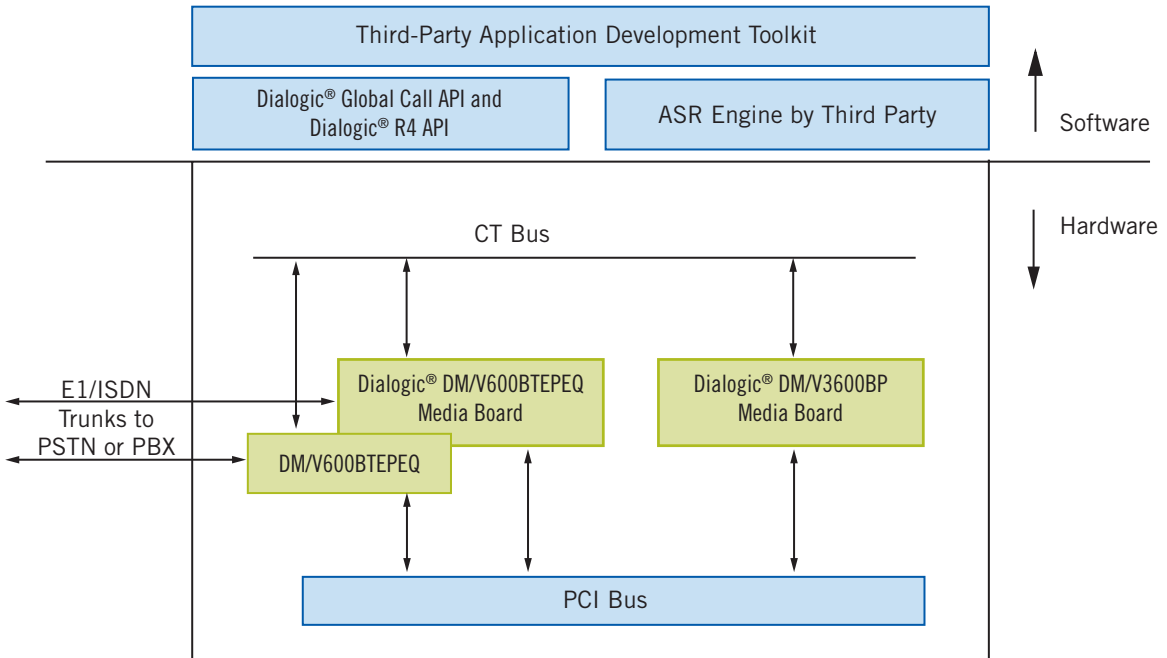


Figure 1. Terminate Configuration: Unified Messaging Application

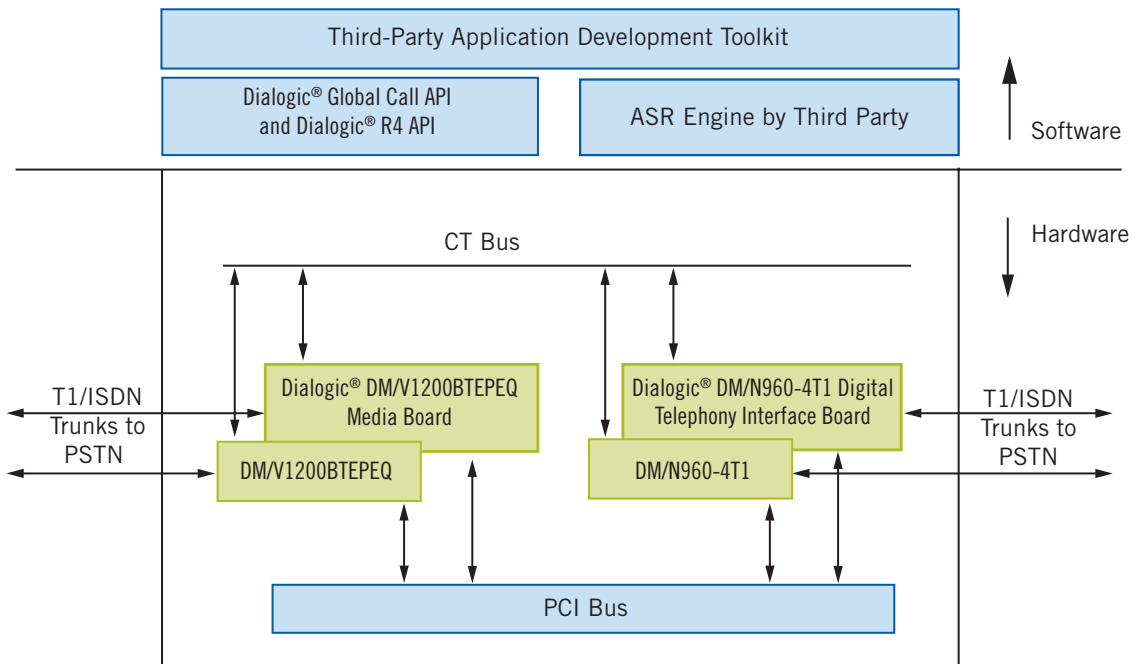


Figure 2. Hairpin Configuration: Call Completion or Prepaid/Debit Card Applications

## Software Support

The DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ are supported by system software and Software Development Kits (SDKs) for Windows and for Linux. They also support the Simple Network Management Protocol (SNMP) agent software for remote CT board management. See the application note at <http://www.dialogic.com/systemreleases> for details about the operating systems currently supported. The SDKs contain all the documentation, demonstration code, and tools necessary for developing complex multichannel applications.

## Dialogic® Global Call Software

The DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ support Dialogic Global Call Software, a unified call control programming interface and protocol engine that makes it easier to provide worldwide application portability and can shorten development time by using the same API for almost any network protocol.

Global Call Software provides a common signaling interface for network-enabled applications, regardless of the signaling protocol needed to connect to the local telephone network. Global Call is the recommended API for unified call control for Springware and DM3 architectures. The signaling interface provided by Global Call facilitates the exchange of call control messages between the telephone network and virtually any network-enabled application. Global Call lets developers create an application that can work with signaling systems worldwide, regardless of the network to which they are connected.

Global Call is well suited for high-density, network-enabled solutions for voice and data where the supported hardware and signaling technology can vary widely. Rather than requiring the application to handle the low-level details, Global Call Software offers a consistent, high-level interface to the user, handling each country's unique protocol requirements in a way that is transparent to the application.

## Functional Description

The DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ are based on the DM3 mediastream architecture. The architecture consists of a set of core specifications and firmware modules that are implemented on boards with various processors, including:

- Intel XScale processor for centralized control
- DSP(s) for mediastream processing
- TDM bus interface (H.100)
- One, two, or four software selectable digital telephony network interfaces
- PCI Express interface

These boards support up to 120 channels of voice processing via a bank of DSPs and up to four T1/E1 digital trunk interface (DTI) circuits. The DTI circuits contain signaling services (ISDN, CAS, and CCS), plus any alarm handling and line maintenance services required by the installed networks. Each DTI includes software-switchable clock circuits that can be set to one of three modes:

- **Loop** — Transmit clocking is slaved to the external network
- **Independent** — Transmit clocking is derived from an onboard oscillator
- **Expansion or system** — Transmit clocking is slaved to the TDM; receive clocking is slaved to the trunk interface

The control processor is the Intel 80200 based on Intel XScale microarchitecture. It is responsible for the initialization, configuration, and control of the various elements that make up these boards. It controls the TDM bus interface, as well as the signaling protocols for the DTIs installed on the platform.

The boards support various DSP configurations for voice processing and call progress analysis capabilities. These features are provided by baseboard and daughterboard configurations using Motorola 56321 DSPs. The DSPs process the digitized voice data using downloaded resource firmware. Each DSP can perform the following signal analysis and operations.

### Incoming Data

- Automatic Gain Control (AGC), which compensates for variations in the level of the incoming audio signal
- Adaptive Differential Pulse Code Modulation (ADPCM), Pulse Code Modulation (PCM), LinearWAV, Global System for Mobile Communications (GSM), G.726, and TrueSpeech algorithms that compress digitized voice and save disk storage space
- Tone detection of DTMF, MF, or application-defined single or dual tones
- Silence detection to determine if the line is quiet and the caller is not responding

### Outbound Data

- Expands stored, compressed audio data for playback
- Adjusts the volume and pitch of playback upon application or user request
- Generates tones — DTMF, MF, or any application-defined general-purpose tone
- Performs outbound dialing
- Monitors call progress functions, including:
  - Line busy
  - Operator intercept
  - Ring
  - No answer
  - Answered; the DSP detects whether the answering party is a person, answering machine, fax machine, or modem

While recording speech, the DSP can use different digitizing rates from 8.5 kbps to 176 kbps, selectable 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.

DSP-processed speech is transmitted by the control processor to the host for disk storage. When playing back a stored file, the processor retrieves voice information from the host CPU and passes it to the DSP, which converts the file into digitized voice. The DSP sends the digitized voice responses to the caller via the network interface or TDM bus. In addition, the cache prompt feature provides 4 to 8 MB of onboard cache for the storing and playback of voice files directly on the board, eliminating the need to send voice files to and from the host/server.

Shared RAM on these boards enables communication between the host system and the XScale control processor. A bank of global memory is also provided to facilitate communication between the control processor and the various DSPs. In addition to providing a data pathway between processors, the global memory can also serve as a repository for data that is to be shared among processors, or which may not be storable within local memory associated with the processor.

## Downloadable Firmware

The hardware for the DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ consists of a baseboard with a control processor and one, two, or four DS-1 DNIs. An array of DSPs resides on the baseboard and a low-profile daughterboard on the DM/V600TEPEQ and DM/V1200TEPEQ. The DM/V300TEPEQ has no daughterboard. Telephony signaling protocols and media processing features are downloaded as firmware to the board on power up and reside on the various onboard processors. This downloadable firmware approach enables easy feature upgrade and expansion. Individual firmware components, such as a network interface protocol or a voice recording function, are referred to as resources.

## Network Interface

The DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ have software-selectable trunks that can enable the board to be configured as T1 or E1 to increase flexibility, simplify the purchasing process and test effort, and help reduce inventory and the total cost of ownership. With Dialogic® System Release 6.0 PCI for Windows® and Dialogic® System Release 6.1 for Linux, these boards support A-law/ $\mu$ -law conversion and the ability to mix individual spans as T1 or E1. They also support ISDN PRI access for both T1 and E1.

- **Configured as T1** — Support for all T1 robbed-bit signaling protocols and fully compatible with all interface devices that use, or can be set to use, 1.544 MHz clocking and  $\mu$ -law PCM. In addition, supports the clear channel feature, thus providing up to 96 bearer channels when used in this mode. The T1 protocol implementations comply with the North American standard ISDN PRI and the INS-1500 standard used in Japan. In North America and Japan, the ISDN Primary Rate includes 23 voice/data channels (B channels) and one signaling channel (D channel).
- **Configured as E1** — Support for all CEPT Channel Associated Signaling (CAS) protocols and fully compatible with interface devices that use, or can be set to use, 2.048 MHz clocking and A-law PCM. In addition, supports the clear channel feature, thus providing up to 124 bearer channels when used in this mode. The E1 protocol implementations comply with the E1 ISDN PRI protocols. The E1 ISDN Primary Rate includes 30 voice/data channels (B channels) and two additional channels: one signaling channel (D channel) and one framing channel to handle synchronization.

Key ISDN PRI features include:

- **Non-Facility Associated Signaling (NFAS)** — Lets a single D-channel control up to 10 PRI trunks, providing significant savings in ISDN service subscription costs available on NI-2, 4ESS, 5ESS, DMS100, and DMS250
- **D-channel backup (on NI-2 only)** — Lets another D-channel takeover should the main D-channel fail
- **Facility, notify, and optional Information Elements (IEs)** — Lets applications work with network-specific supplementary services
- **Direct Dialing In (DDI)** — Lets an application route incoming calls by automatically identifying the number the caller dialed. Also known as Dialed Number Identification Service (DNIS).
- **Call-by-call service selection** — Lets an application select the most efficient bearer channel service, such as a toll-free line or a WATS line, on a call-by-call basis
- **User-to-user information** — Lets an application send proprietary messages to remote systems during call establishment
- **LAP-D Layer 2 access** — Lets developers build a customized Layer 3 protocol
- **Protocol timer setting** — Lets protocol timers be set dynamically through a configuration file
- **Mask-able Layer 2 Control** — Lets the application toggle between bringing Layer 2 up and down as desired



Dialogic maintains an extensive number of product approvals in international markets. See the list of globally approved products at <http://www.dialogic.com/declarations>.

## Voice Processing

Voice processing features, downloaded to the onboard DSPs at power up, enable these combined media boards to play and record voice messages to and from callers through the digital network interface. Messages can be stored using G.711  $\mu$ -law or A-law PCM, at a rate of 64 kbps, as is used by the Public Switched Telephone Network (PSTN). To reduce storage requirements and help developers implement unified messaging applications that meet Voice Profile for Internet Messaging (VPIM) standards, voice coding algorithms can compress recordings as low as 8.5 kbps using low-bit rate coders such as GSM, G.726, and TrueSpeech. 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 needed.

AGC is provided to automatically adjust the signal level of incoming calls for recording at normal levels, compensating for adverse line conditions, distance, and other factors. Playback volume can also be dynamically adjusted over a 40 dB range using DTMF input or directly from the application.

DTMF detection is provided to control record and play functions using DTMF input. Local echo cancellation techniques are used to improve DTMF cut-through and talk-off/play-off suppression over a wide variety of telephone line conditions.

The voice player and recorder resources are linked with the DTMF detection resources using Run-Time Control (RTC) messages. This lets play or record functions be initiated or terminated quickly using DTMF input from the caller. The RTC function off-loads the host application from involvement in every interaction, thereby enabling voice processing applications to scale to hundreds of ports per system.

Continuous Speech Processing (CSP) enables software-based Automatic Speech Recognition (ASR) and the ability to speak over speech prompts. It processes the incoming voice signal using DSP-based Echo Canceller (EC) and Voice Activity Detector (VAD) integrated on the board. The incoming voice signal is then streamed to the host system and the ASR engines only when voice energy is detected. Features such as the pre-speech buffer and the onboard VAD let the system attain higher accuracy and efficiency. Also now available on select media loads is Enhanced Echo Cancellation (EEC), which offers the ability to select longer echo cancellation tail lengths of 32 ms and 64 ms beyond the standard 16 ms to further improve and refine audio quality.

The transaction record feature lets voice activity on two channels be summed and stored in a single file, or in a combination of files, devices, and/or memory. When it is necessary to archive a verbal transaction or record a live conversation, the silence-compressed record feature, when enabled, eliminates silence from recorded data, thus saving disk storage space. Speed and volume control are also provided to let the application or user adjust the speed and volume during playback. Silence compressed streaming to the host improves performance by removing the silence when data is sent to the host CPU. Streaming to the CT Bus lets echo cancelled data be streamed into the TDM bus.

## Conferencing

The DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ are available with two different types of conferencing (rich and standard) to maximize conferencing solutions based on system configuration and requirements (see Table 1).

Conferencing Type	Echo Cancellation	Tone Detection and Generation	Tone Clamping
Rich Conferencing	✓	✓	✓
Standard Conferencing	—	✓	✓

Table 1. Conferencing Media Loads



The conferencing solution is implemented using onboard DSPs. The conferencing resource sums incoming voice signals on the board. Higher quality conferencing is attained using sophisticated summing algorithms and Echo Cancellation (EC). The advanced algorithm dynamically distinguishes between noise and speech and prevents noise buildup. The incoming voice signal is then streamed out to the CT Bus where it can be transmitted to any network interface — either PSTN or IP telephony.

This enables very large size conferences (1000+) such as analyst calls or broadcast calls where most of the callers are in listen-only mode. In addition, bridging (also known as cascade conferencing) enables conferences to be joined together via a bridge (up to 64 participants in each conference) from different DSPs and boards, consuming just two conferencing resources per bridge. This maximizes the flexibility of a conferencing solution by allowing the creation of high-density conferences where any party has the ability to speak and be heard by other participants.

Both types of conferencing media loads support the active talker feature that identifies which conferees are actively talking at any given time and suppresses the background noise from all the silent parties. It also lets applications dynamically choose between the summing mode — active talker vs. pure summation — based on the conference size. Applications can set this parameter at configuration or change it dynamically at runtime. For a small number of parties, pure summation might be preferred so all conferees are heard; and as conference size increases, the active talker feature might be enabled so conferees can hear the most active participants.

## Tone Signaling

In addition to the DTMF signaling commonly used for voice processing, the DM/V300BTEPEQ, DM/V600BTEPEQ, and DM/V1200BTEPEQ also contain a robust set of features used for network tone signaling and control. The Global Tone Detection (GTD) and Global Tone Generation (GTG) features provide the ability to detect and generate user-defined tones for solving special application situations, such as integration with Private Branch Exchange (PBX) or dealing with unique tones.

Perfect Call call progress analysis accurately monitors outbound calls, detects when calls are answered, and distinguishes:

- Line ringing with no answer
- Line busy
- Problem completing call (such as operator intercept)
- Call answered by a human or answering machine
- Call answered by a fax machine or modem

Perfect Call is intelligently tolerant of the wide variation in call progress signaling tones found in central offices and PBXs around the globe and offers accurate performance right out of the box. DSP-based algorithms are used to accurately discriminate human speech from recorded human voice and from network noise.

## Technical Specifications

Digital interfaces	One, two, or four T1/E1
Max. boards/system	Application, call traffic, and CPU dependent
Control processor	Intel 80200
Control processor memory	48 MB
Baseboard global memory	16 MB 32-bit wide DRAM accessible to all signal processors and control processor
Cache prompts	4 MB to 8 MB

### Host Interface

Host interface memory	512 KB
Bus compatibility	PCI Express Card Electromechanical Specification Revision 1.0a
Bus mode	Target and DMA master mode operation

### Platform

Form factor	Full-length PCI Express (x1 compatible)
Digital Signal Processor (DSP)	Motorola 56321 DM/V300BTEPEQ: 4 DSPs @ 220 MHz each DM/V600BTEPEQ: 10 DSPs @ 220 MHz each DM/V1200BTEPEQ: 10 DSPs @ 220 MHz each
DSP memory	512 k word SRAM local to each DSP
Computer telephony bus	ECTF H.100-compliant CT Bus with: <ul style="list-style-type: none"> <li>• Onboard switching access to 4096 bi-directional 64 kbps DS0 time slots</li> <li>• 68-pin ribbon cable connector</li> </ul>

### Telephone Interface

	<b>DSX-1 T1</b>
Clock rate	1.544 Mbps $\pm$ 32 ppm
Level	3.0 V (nominal)
Pulse width	323.85 ns (nominal)
Line impedance	100 Ohm $\pm$ 10%
Other electrical characteristics	Complies with AT&T TR62411 and ANSI T1.403-1989
Framing	<ul style="list-style-type: none"> <li>• SF (D3/D4)</li> <li>• ESF for ISDN</li> </ul>
Line coding	<ul style="list-style-type: none"> <li>• AMI</li> <li>• AMI with B7 stuffing</li> <li>• B8ZS</li> </ul>
Clock and data recovery	Complies with AT&T TR62411 and Telcordia TA-TSY-000170
Jitter tolerance	Complies with AT&T TR62411 and ANSI T1.403-1989
Connectors	One, two, or four RJ-48C on front bracket

**Technical Specifications (continued)**

Telephony bus connector	H.100 (PCI)
Loopback	Supports switch-selectable local analog loopback and software-selectable local digital loopback
Zero code suppression	<ul style="list-style-type: none"> <li>• Bell ZCS (Jam bit 7)</li> <li>• GTE ZCS (Jam bit 8)</li> <li>• Digital Data Service ZCS</li> <li>• No zero code suppression</li> </ul>

**Telephone Interface**

Network clock rate	2.048 Mbps $\pm$ 50 ppm
Internal clock rate	2.048 Mbps $\pm$ 32 ppm
Level	<ul style="list-style-type: none"> <li>• 2.37 V (nominal) for 75 Ohm lines</li> <li>• 3.0 V (nominal) for 120 Ohm lines</li> </ul>
Pulse width	244 ns (nominal)
Line impedance	<ul style="list-style-type: none"> <li>• 75 Ohm, unbalanced</li> <li>• 120 Ohm, balanced</li> </ul>
Other electrical characteristics	Complies with ITU-T Rec. G.703
Framing	ITU-T G.704-1988 with CRC4
Line coding	HDB3
Clock and data recovery	Complies with ITU-T Rec. G.823-1988
Jitter tolerance	Complies with ITU-T Rec. G.823, G.737, G.739, G.742-1988
Connectors	One, two, or four RJ-48C on front bracket
Telephony bus connector	H.100 (PCI)
Loopback	Supports switch-selectable local analog loopback and software-selectable local digital loopback

**Power Requirements**

Configuration	+12 VDC	+3.3 VDC
DM/V300BTEPEQ	0.6	0.58
DM/V600BTEPEQ	1.25	0.82
DMV1200BTEPEQ	1.25	0.82

**Operating Requirements**

Operating temperature	+32°F (0°C) to +122°F (+50°C)
Storage temperature	-4°F (-20°C) to +158°F (+70°C)
Humidity	8% to 80% noncondensing

**Approvals and Compliance**

Hazardous substances	RoHS compliance information at <a href="http://www.dialogic.com/rohs">http://www.dialogic.com/rohs</a>
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## Technical Specifications *(continued)*

### Safety and EMC

United States	<ul style="list-style-type: none"> <li>• UL 60950-1 File E96804</li> <li>• FCC Part 15 Class A</li> </ul>
Canada	<ul style="list-style-type: none"> <li>• ULc CSA 60950-1 File E96804</li> <li>• ICES-003 Class A</li> </ul>
Europe	<ul style="list-style-type: none"> <li>• EN60950</li> <li>• EN55022</li> <li>• EN55024</li> </ul>
International	<ul style="list-style-type: none"> <li>• IEC60950-1</li> <li>• CISPR 22</li> <li>• CISPR 24</li> </ul>

### Telecom Approvals

United States	US:EBZUSA-31207-XD-T
Canada	IC:885A 7969 A
Europe	DoC TBR4

For country-specific approval information, see the global product approvals database at <http://www.dialogic.com/declarations>

### Reliability/Warranty

Estimated MTBF	<p>DM/V300BTEPEQ — 270,780 hours per Telcordia Method I</p> <p>DM/V600BTEPEQ — 239,839 hours per Telcordia Method I</p> <p>DM/V1200BTEPEQ — 235,000 hours per Telcordia Method I</p>
Warranty	Warranty information at <a href="http://www.dialogic.com/warranties">http://www.dialogic.com/warranties</a>

### Audio Signal

Usable receive range	–40 dBm0 to 0 dBm0 nominal, configurable by parameter**
Automatic gain control	Application can enable/disable output level, configurable by parameter**
Silence detection	–40 dBm nominal, software adjustable**
Transmit level (weighted average)	–12.5 dBm nominal, configurable by parameter**
Transmit volume control	40 dB adjustment range, with application-definable increments and legal limit cap

### Frequency Response

24 kbps	300 Hz to 2600 Hz $\pm 3$ dB
32 kbps	300 Hz to 3400 Hz $\pm 3$ dB
64 kbps	300 Hz to 3400 Hz $\pm 3$ dB

**Technical Specifications (continued)****Audio Digitizing**

8.5 kbps	TrueSpeech
13 kbps	GSM (TIPHON, MSGSM)
16 kbps, 24 kbps, 32 kbps, and 40 kbps	G.726
24 kbps	OKI ADPCM @ 6 kHz sampling
32 kbps	OKI ADPCM @ 8 kHz sampling
32 kbps	IMA ADPCM @ 8 kHz sampling
48 kbps	G.711 PCM ( $\mu$ -law for T1 and A-law for E1) @ 6 kHz sampling rate
64 kbps	G.711 PCM ( $\mu$ -law for T1 and A-law for E1) @ 8 kHz sampling rate
64 kbps	Linear 8 kHz 8-bit WAV
128 kbps	Linear 8 kHz 16-bit WAV
88 kbps	Linear 11 kHz 8-bit WAV
176 kbps	Linear 11 kHz 16-bit WAV
A-law/ $\mu$ -law conversion	Standard (with Dialogic® System Release 6.1 for Linux)
Digitization selection	Selectable by application on function call-by-call basis
Playback speed control	<ul style="list-style-type: none"> <li>• Pitch controlled</li> <li>• Available on the following 8 kHz coders: OKI ADPCM, G.711 PCM, Linear</li> <li>• Adjustment range: <math>\pm 50\%</math></li> <li>• Adjustable through application or programmable DTMF control</li> </ul>

**DTMF Tone Detection**

DTMF digits	0 to 9, *, #, A, B, C, D per Telcordia LSSGR Sec. 6
Dynamic range	<ul style="list-style-type: none"> <li>• (T1) <math>-36</math> dBm to <math>+3</math> dBm per tone, configurable by parameter**</li> <li>• (E1) <math>-39</math> dBm to <math>0</math> dBm per tone, configurable by parameter**</li> </ul>
Minimum tone duration	32 ms, can be increased with software configuration
Interdigit timing	<ul style="list-style-type: none"> <li>• Detects like digits with a <math>&gt;45</math> ms interdigit delay</li> <li>• Detects different digits with a <math>0</math> ms interdigit delay</li> </ul>
Acceptable twist and frequency variation	<ul style="list-style-type: none"> <li>• (T1) Meets Telcordia LSSGR Sec 6 and EIA 464 requirements</li> <li>• (E1) Meets ITU-T Q.23 recommendations**</li> </ul>
Noise tolerance	Meets Telcordia LSSGR Sec 6 and EIA 464 requirements for Gaussian, impulse, and power line noise tolerance
Cut-through	<ul style="list-style-type: none"> <li>• (T1) Local echo cancellation permits 100% detection with a <math>&gt;4.5</math> dB return loss line</li> <li>• (E1) Digital trunks use separate transmit and receive paths to network. Performance dependent on far-end handset's match to local analog loop.</li> </ul>
Talk-off	<ul style="list-style-type: none"> <li>• Detects less than 10 digits while monitoring Telcordia TR-TSY-000763 standard speech tapes. (LSSGR requirements specify detecting no more than 470 total digits.)</li> <li>• Detects 0 digits while monitoring MITEL speech tape #CM 7291.</li> </ul>

**Technical Specifications (continued)****Global Tone Detection**

Tone type	Programmable for single or dual
Max. number of tones	Application-dependent
Frequency range	Programmable within 300 Hz to 3500 Hz
Max. frequency deviation	Programmable in 5 Hz increments
Frequency resolution	±5 Hz. Separation of dual frequency tones is limited to 62.5 Hz at a signal-to-noise ratio of 20 dB.
Timing	Programmable cadence qualifier, in 10 ms increments
Dynamic range	<ul style="list-style-type: none"> <li>• (T1) Default set at –36 dBm to +3 dBm per tone, programmable</li> <li>• (E1) Default set at –39 dBm to +0 dBm per tone, programmable</li> </ul>

**Global Tone Generation**

Tone type	Generate single or dual tones
Frequency range	Programmable within 200 Hz to 4000 Hz
Frequency resolution	1 Hz
Duration	10 ms increments
Amplitude	<ul style="list-style-type: none"> <li>• (T1) –43 dBm0 to –3 dBm0 per tone nominal, programmable</li> <li>• (E1) –40 dBm0 to +0 dBm0 per tone nominal, programmable</li> </ul>

**MF Signaling (T1)**

MF digits	<b>R1</b> 0 to 9, KP, ST, ST1, ST2, ST3 per Telcordia LSSGR Sec 6, TR-NWT-000506 and ITU-T Q.321
Transmit level	Complies with Telcordia LSSGR Sec 6, TR-NWT-000506
Signaling mechanism	Complies with Telcordia LSSGR Sec 6, TR-NWT-000506
Dynamic range for detection	–25 dBm to +3 dBm per tone
Acceptable twist	6 dB
Acceptable freq. variation	Less than ±1 Hz

**MF Signaling (E1)**

MF digits	<b>R2</b> All 15 forward and backward signal tones per ITU-T Q.441
Transmit level	–8 dBm0 per tone, nominal, per ITU-T Q.454; programmable
Signaling mechanism	Supports the R2 compelled signaling cycle and non-compelled pulse requirements per ITU-T Q.457 and Q.442
Dynamic range for detection	–35 dBm to –5 dBm per tone
Acceptable twist	7 dB
Acceptable freq. variation	Less than ±1 Hz

**Technical Specifications (continued)****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 ITU-T Rec. E., Suppl. #2. Default uses 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 ITU-T Rec. E., Suppl. #2. Uses both frequency and cadence detection.
Positive voice detection	Standard
Positive voice detection speed	Detects voice in as little as 1/10th of a second
Positive answering machine detection	Standard
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	<ul style="list-style-type: none"> <li>• Application enable/disable</li> <li>• Supports up to three different user-definable dial tones</li> <li>• Programmable dial tone drop out debouncing (when not part of regulatory approval)</li> </ul>

**Tone Dialing**

DTMF digits	0 to 9, *, #, A, B, C, D per Telcordia LSSGR Sec 6, TR-NWT-000506, ITU-T Q.23
Frequency variation	Less than $\pm 1$ Hz
Rate	10 dps, configurable by parameter**
Level	<ul style="list-style-type: none"> <li>• (T1) –4.0 dBm per tone, nominal, configurable by parameter**</li> <li>• (E1) –7.0 dBm per tone, nominal, country-specific**</li> </ul>

**Conferencing**

Max. parties per conference	Up to 64 (without bridging) on select media loads
Bridging/cascade conferencing	Can bridge together conferences from different DSPs and boards, consuming just two conferencing resources per bridge
Echo cancellation	16 ms
Tone clamping	Enable/disable at board level
Summing modes	Automatically configures to active talker or pure summation based on number of parties in a conference. Application can specify minimum number of parties before active talker mode is used.
Automatic gain control	Normalizes the parties' power levels to a unified target. Key features include speech/noise discrimination, tolerance to impulsive noise, faster convergence, and increased steady-state stability.
Tone detection/generation	Generates tariff tones and warning tones. Detects DTMF from each party and can clamp those tones so that other members of the conference do not hear them.
Active talker notification	Can notify the application as to which party is talking so the application can process that information and act accordingly
Number of active talkers	Dynamically selectable



**Technical Specifications** *(continued)*

Modes	<ul style="list-style-type: none"> <li>• Duplex</li> <li>• Monitor</li> <li>• Coach</li> <li>• Pupil</li> </ul>
<b>Facsimile</b>	
Fax compatibility	T.30, T.4, T.6, V.17, V29, V27ter, V.21
Speed	14.4 kbps with automatic fallback send and receive concurrently on all channels
TIFF	<ul style="list-style-type: none"> <li>• Single page</li> <li>• Multipage</li> </ul>
Compression	<ul style="list-style-type: none"> <li>• MH (ITU T.4, 1D)</li> <li>• MR (ITU T.4 2D)</li> <li>• MMR (ITU T.6)</li> <li>• Onboard, on-the-fly</li> </ul>
ECM	Supported
ASCII to TIFF	Onboard, on-the-fly
Page headers	Generated onboard, on-the-fly
Width	A4
Polling	Normal and turnaround
Resolution	<ul style="list-style-type: none"> <li>• Standard (100 dpi × 200 dpi)</li> <li>• Fine (200 dpi × 200 dpi)</li> <li>• Superfine (200 dpi × 400 dpi)</li> </ul>
JPEG/JBIG	Color fax and gray scale fax pass-through feature
<b>Protocols</b>	
T1 CAS	E&M (wink start, immediate start), loop start, ground start; feature group A, B, and D
T1 ISDN	NI-2, 4ESS, 5ESS, DMS100, DMS250, INS1500, Q.Sig
E1 CAS release notes	Many country-specific MFC-R2 variants. For more details, refer to the latest Global Call Protocols Package
E1 ISDN	NET5, DPNSS, DASS2, QSIG

**Additional Components**

- Multidrop CT Bus cables (CBLCTB3DROPQ, CBLCTB4DROPQ, CBLCTB8DROPQ, CBLCTB12DROPQ, CBLCTB16DROPQ)
- 120 Ohm to 75 Ohm converter (supplied by a third party)

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