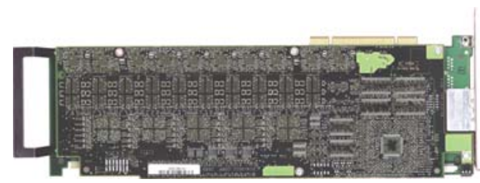
Intel in
Communications

Intel® NetStructure™ DM/V480A-2T1, DM/V600A-2E1, DM/V960A-4T1, and DM/V1200A-4E1 Combined Media Boards



The Intel® NetStructure™ DM/V480A-2T1, DM/V600A-2E1, DM/V960A-4T1, and DM/V1200A-4E1 combined media boards are ideal for developers seeking to rapidly build and globally deploy high-density media server solutions for the enterprise and public networks. They provide a robust media feature set, including voice processing and speech recognition, and on select media loads, fax and/or conferencing capabilities, combined with an extensive suite of network protocols in a single PC slot.

Features	Benefits
Built on the industry-standard telephony bus — ECTF H.100/H.110 CT Bus	Lets applications expand (up to 1200 ports per system) through access to other communications boards, such as IP telephony, ATM, HDSI, and SS7
Supports TrueSpeech* voice coder (a default coder with Microsoft* Windows* supported by Windows Media Player)	Lets developers play Internet content and develop unified messaging systems without creating and supporting custom clients
Ability to select between 16 ms, 32 ms, and 64 ms echo cancellation tail on select media loads	The longer tail lengths are useful for environments and applications when optimum audio quality and clarity is a necessity
Supports rich conferencing media load on select boards	Lets you deploy network-grade conferencing systems with comparable features, audio quality, and density as typical proprietary solutions, but at significantly reduced costs

The platforms, available in both H.100 (PCI) and H.110 (CompactPCI*) compliant universal form factors, are ideal for service providers and large enterprise applications. This flexibility lets developers build single applications to be deployed on either industry-standard form factor.

Support for continuous speech processing (CSP) technology — the digital signal processing (DSP)-based solution optimized for speech recognition — enables 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 the audio quality.

Other improvements 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 into 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 comparable features, audio quality, and density as 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 multifunction (DTMF; touchtone) clamping to limit audible enter and exit tones.

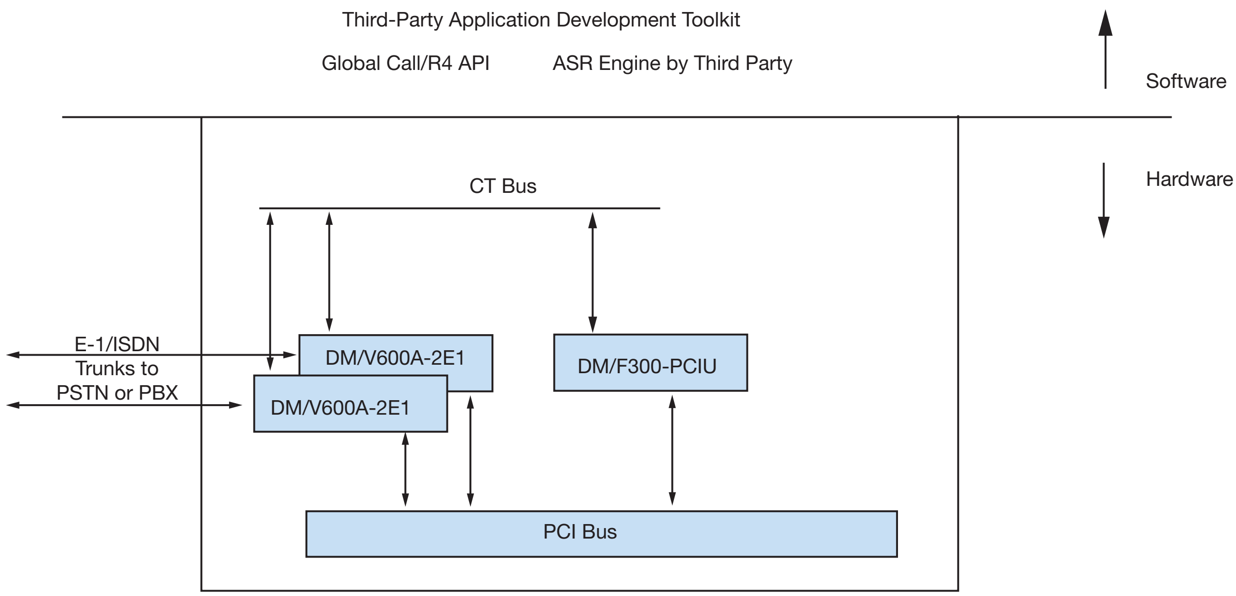
In addition, another new conferencing feature is bridging, also known as cascade conferencing. Bridging lets you bridge together conferences from different DSPs and boards, consuming just one extra time slot per bridge. This maximizes the flexibility of your conferencing solution by letting you create 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.

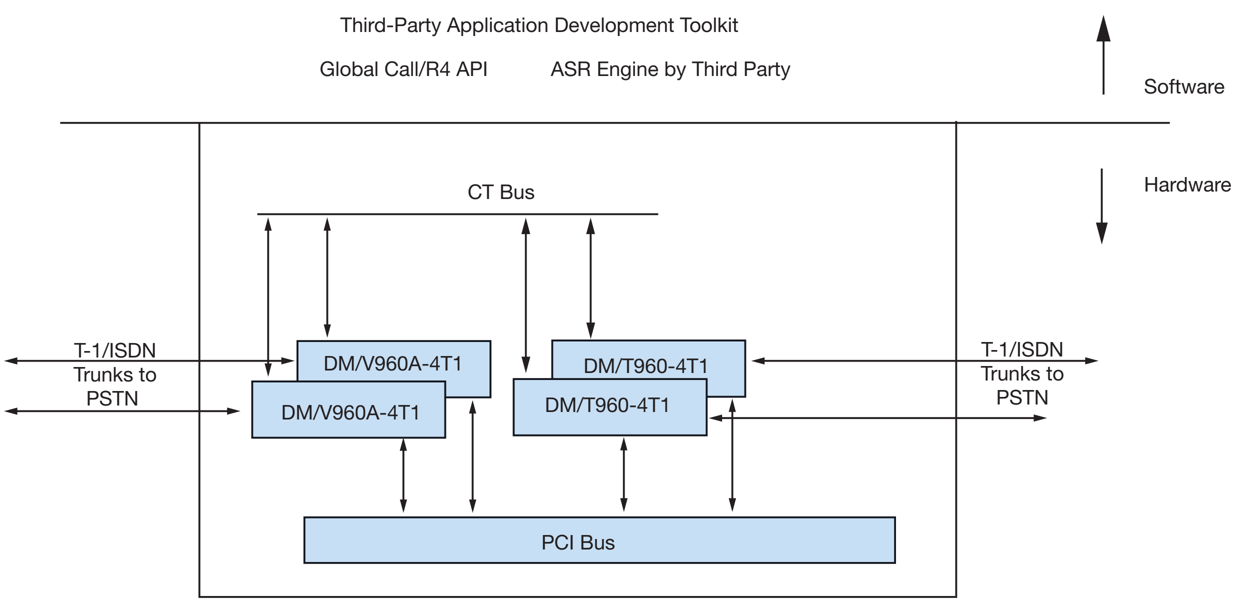
These combined media boards are based on the DM3 architecture, which provides a development environment that accelerates application development while providing a path for future growth. With support for the R4 application programming interface (API), it's easy for these boards to interoperate with other CT Bus and SCbus boards from Intel. Applications can be ported easily to lower or higher density platforms, or new features can be added with only minimum modifications — thus protecting your investment in hardware and application code.

Applications

- Messaging and enhanced services
- Wireless and fixed-line short message service (SMS)
- Color ringback
- Voice portal
- Contact center and e-Business
- PC-PBX
- Audio conferencing server
- Web conferencing
- Switching and call completion
- Prepaid/debit card
- Gateway switch



Terminate Configuration: Unified Messaging Application



Hairpin Configuration: Call Completion or Prepaid/Debit Card Applications

Configurations

Use multiple Intel NetStructure DM/V480A-2T1, DM/V600A-2E1, DM/V960A-4T1, and DM/V1200A-4E1 combined media boards to develop sophisticated, mixed-media communication applications that include voice processing, speech recognition, fax, and conferencing.

Note, for added functionality, Intel NetStructure DMV600BTEP and DMV1200BTEP combined media boards offer universal media loads which support mixed media resources, including voice, fax, and conferencing all on the same media load. This essentially combines three boards into one, reducing the total cost of ownership by increasing flexibility, reducing inventory, and simplifying the purchasing process and test effort.

These boards occupy a single computer backplane slot and multiple boards can be installed in a single computer. The maximum number of supported lines 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 like 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, these products handle the processing of digital audio and telephony signaling. Additional system resources can access calls via the CT Bus. This configuration is ideal for voice messaging, unified messaging, voice portal, and interactive voice response (IVR) applications.

In a hairpin configuration, the boards are connected via the CT Bus and can continuously pass all T-1/E-1 time slots through to each other. This configuration can switch call traffic between separate T-1 or E-1 lines, or can be placed inline between a T-1/E-1 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 during "flow through" mode, these boards can still monitor the lines, listening for DTMF or voice commands. This configuration is ideal for call center, PC-PBX, voice portal, prepaid calling card, international callback, and telecom resale applications.

Software Support

The Intel NetStructure DM/V480A-2T1, DM/V600A-2E1, DM/V960A-4T1, and DM/V1200A-4E1 combined media boards support the SNMP agent software for remote CT board management and other software development kits (SDKs) for Windows NT*, Windows 2000*, Windows 2003*, and Linux* operating systems.[1]

Global Call

These boards support 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 ideal for high-density, network-enabled solutions for voice, data, and video, 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 Intel NetStructure DM/V480A-2T1, DM/V600A-2E1, DM/V960A-4T1, and DM/V1200A-4E1 combined media boards are based on the DM3 architecture. The architecture consists of a set of core specifications and firmware modules that are implemented on boards with various processors, including

- RISC processor for centralized control
- DSP(s) for media processing
- TDM bus interface (H.100/H.110)
- Two or four digital telephony network interfaces
- PCI and CompactPCI bus interfaces

These boards support up to 96 (T-1) or 120 (E-1) channels of voice processing via a bank of DSPs and up to four T-1 or E-1 digital trunk interface (DTI) circuits. The DTI circuits contain signaling services (ISDN, channel associated signaling [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

- loop mode — transmit clocking is slaved to the external network
- independent mode — transmit clocking is derived from an onboard oscillator
- expansion or system mode — transmit clocking is slaved to the TDM; receive clocking is always slaved to the trunk interface

The control processor is a general-purpose Intel i960® RISC microprocessor, responsible for the initialization, configuration, and control of the various elements that make up these specific boards. It controls the TDM bus

[1] Supported operating systems vary, depending on the specific system release.

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 a daughterboard using Motorola* DSPs. The DSPs process the digitized voice data using downloaded resource firmware. Each DSP can perform the following signal analysis and operations:

For 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 whether the line is quiet and the caller is not responding

For 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, a fax machine, or modem

While recording speech, the DSP can use different digitizing rates from 8.5 Kb/s to 176 Kb/s, 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, cache prompts now let you store 4 MB 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 i960 control processor. A bank of global memory is also provided to facilitate communications 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 Intel NetStructure DM/V480A-2T1, DM/V600A-2E1, DM/V960A-4T1, and DM/V1200A-4E1 combined media boards consists of a baseboard with a RISC processor and two or four DS-1 digital network interfaces. (Different assemblies are used for T-1 and E-1.) An array of DSPs resides on a low-profile daughterboard. Telephony signaling protocols and voice 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 T-1 versions of these boards support all T-1 robbed-bit signaling protocols and are fully compatible with all interface devices that use, or can be set to use, 1.544 MHz clocking and μ-law PCM. The E-1 versions of these boards support all CEPT CAS protocols and are fully compatible with interface devices that use, or can be set to use, 2.048 MHz clocking and A-law PCM (ITU-T Recommendation G.703/704/711). The boards also support the clear channel feature, thus providing up to 124 bearer channels.

The boards also support ISDN PRI access for both T-1 and E-1. The T-1 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). The E-1 protocol implementations comply with the E-1 ISDN PRI protocols. The E-1 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.

Note, for added functionality, Intel NetStructure DMV600BTEP and DMV1200BTEP combined media boards offer software selectable T-1/E-1 trunks which help reduced the total cost of ownership by increasing flexibility, reducing inventory, and simplifying the purchasing process and test effort.

The key ISDN PRI features include

- Non-Facility Associated Signaling (NFAS) lets a single D-channel control up to 20 PRI trunks, providing significant savings in ISDN service subscription costs available on NI-2, 4ESS, Lucent 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) let applications work with network-specific supplementary services
- Direct Dialing In (DDI), also known as Dialed Number Identification Service (DNIS), lets an application route incoming calls by automatically identifying the number the caller dialed
- Call-by-call service selection lets an application select the most efficient bearer channel service, such as an 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
- The ability to dynamically set protocol timers through a configuration file
- A maskable Layer 2 Control lets the application toggle between bringing Layer 2 up and down as desired

Intel maintains an extensive number of product approvals in international markets. See the list of globally approved products at http://resource.intel.com/globalapproval/globalapproval.asp.

Voice Processing

Voice processing features, downloaded to the onboard DSPs at power up, let these combined media boards play and record voice messages to and from callers through the digital network interface. Messages can be stored using G.711 μ-law or A-law PCM, at a rate of 64 Kb/s, as is used by the public switched telephone network (PSTN). To reduce storage requirements and help developers implement unified messaging applications that meet VPIM standards, voice coding algorithms can compress recordings as low as 8.5 Kb/s using low-bit rate coders such as TrueSpeech, GSM, and G.726. 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 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.

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 conferencing solution on the combined media boards 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) and tone clamping (TC) integrated on the board. The advanced algorithm distinguishes between noise and speech dynamically and prevents noise build up. 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) lets you bridge together conferences (up to 60 participants in each conference) from different DSPs and boards, consuming just one extra time slot per bridge. This maximizes the flexibility of your conferencing solution by letting you create high-density conferences where any party has the capability to speak and be heard by other participants.

The conferencing resource supports 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 during configuration time or change dynamically during runtime. For a small number of parties, pure summation might be preferred so all conferees are heard; and as a 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 Intel NetStructure DM/V480A-2T1, DM/V600A-2E1, DM/V960A-4T1, and DM/V1200A-4E1 combined media boards 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 capability to detect and generate user-defined tones for solving special application situations, such as integration with 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. Patented DSP-based algorithms are used to accurately discriminate human speech from recorded human voice and from network noise.

High Availability CompactPCI*

Intel NetStructure DM/V480A-2T1, DM/V600A-2E1, DM/V960A-4T1, and DM/V1200A-4E1 combined media boards provide a whole range of high-availability features.

Hot Swap (PICMG* Specifications)

The hot swap capability includes like-for-like board replacement while the system is operational.

System Management

Three types of system management are provided.

- Configuration management — includes features like plug-and-play configuration, individual board validation, automatic addressing, and automatic board configuration to decrease the likelihood of procedural errors caused by inexperienced personnel
- Performance management — detailed monitoring at the port, DSP, or board level lets administrators balance system capacity and plan for future growth
- SNMP — SNMP-enabled computer telephony (CT) components lower the cost of ownership. You can integrate SNMP into an existing infrastructure, or deploy a standard, off-the-shelf SNMP management platform. Remote monitoring and configuration are possible at the board, network, or port level.

Clock Fallback

A fallback clock is provided on a separate board to provide redundancy in case of clock failure. In the event that the master clock fails, the fallback clock takes over to prevent any loss of data. An alarm message is generated in the system log, without interrupting service.

Rugged and Durable Design

CompactPCI uses the Eurocard 6U format and is especially suitable for large-scale PSTN systems where availability and reliability are critical.

CT Bus Compatibility

The Intel implementation of the ECTF H.110 standards-based CT Bus on CompactPCI provides 4096 time slots for exchanging voice, network interface, speech recognition, or other media resources.

Technical Specifications

	Digital interfaces	Two or four T-1/E-1
	Max. boards/system	Application, call traffic, and CPU dependent
	Control processor	Intel® i960CF
	Control processor memory	8 MB
	Baseboard global memory	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 mode	Target and DMA master mode operation
Support	3.3 V or 5 V signaling environment (universal connectivity)

Platforms

	PCI	CompactPCI
Form factor	PCI long card, single-slot width	6U Eurocard form factor, single-slot width
Digital signal processors	Motorola* 56311 10 DSPs @ 150 MHz each	Motorola 56307 15 DSPs @ 100 MHz each
DSP memory	512 K word SRAM local to each DSP	256 K word DRAM local to each DSP 128 K word SRAM local to each DSP
Bus compatibility	Rev 2.2 of PCI Bus Specification	Rev 2.1 of PCI Bus Specification
Bus mode	Target and DMA master mode operation	Target and DMA master mode operation
Computer telephony bus	ECTF H.100 compliant CT Bus, offering • Onboard switching access to 4096 bidirectional 64 Kb/s DS0 time slots • SCbus interoperability through Intel provided adapter • 68-pin ribbon cable connector	ECTF H.110 compliant CT Bus, offering onboard switching access to 4096 bidirectional 64 Kb/s DS0 time slots
Network connectors	Two or four RJ-48C on front bracket	Provided through rear I/O transition modules (not included with board) • BNC for 75 Ohm lines • RJ-48C for 100 Ohm and 120 Ohm lines

Technical Specifications (cont.)

Telephone Interface

DSX-1 T-1

Clock rate	1.544 Mb/s ±32 ppm
Level	3.0 V (nominal)
Pulse width	323.85 ns (nominal)
Line impedance	100 Ohm ±10%
Other electrical characteristics	Complies with AT&T* TR62411 and ANSI T1.403-1989
Framing	SF (D3/D4) ESF for ISDN
Line coding	AMI AMI with B7 stuffing B8ZS
Clock and data recovery	Complies with AT&T TR62411 and Bellcore* TA-TSY-000170
Jitter tolerance	Complies with AT&T TR62411 and ANSI T1.403-1989
Connectors	RJ-48C
Telephony bus connector	H.100 (PCI) and H.110 (CompactPCI) style connectors
Loopback	Supports switch-selectable local analog loopback and software selectable local digital loopback
Zero code suppression	Bell* ZCS (Jam bit 7) GTE* ZCS (Jam bit 8) Digital Data Service* ZCS No zero code suppression

Telephone Interface

CEPT E-1

Network clock rate	2.048 Mb/s ±50 ppm
Internal clock rate	2.048 Mb/s ±32 ppm
Level	2.37 V (nominal) for 75 Ohm lines 3.0 V (nominal) for 120 Ohm lines
Pulse width	244 ns (nominal)
Line impedance	75 Ohm, unbalanced 120 Ohm, balanced
Other electrical characteristics	Complies with CCITT Rec. G.703
Framing	CCITT G.704-1988 with CRC4
Line coding	HDB3
Clock and data recovery	Complies with CCITT Rec. G.823-1988
Jitter tolerance	Complies with CCITT Rec. G.823, G.737, G.739, G.742-1988
Connectors	BNC for 75 Ohm lines RJ-48C for 120 Ohm lines
Telephony bus connector	H.100 (PCI) and H.110 (CompactPCI) style connectors
Loopback	Supports switch-selectable local analog loopback and software selectable local digital loopback

Power Requirements

Configuration	+5 VDC	+12 VDC	–12 VDC	+3.3 VDC
PCI				
DM/V960A-4T1-PCI	22.5 W	N/A	N/A	N/A
DM/V1200A-4E1-PCI	22.5 W	N/A	N/A	N/A
DM/V480A-2T1-PCI	22.5 W	N/A	N/A	N/A
DM/V600A-2E1-PCI	22.5 W	N/A	N/A	N/A
CompactPCI				
DM/V960A-4T1-cPCI	19.25 W	1.1 W	N/A	9.8 W
DM/V1200A-4E1-cPCI	19.25 W	1.1 W	N/A	9.8 W
DM/V480A-2T1-cPCI	19.25 W	1.1 W	N/A	9.8 W
DM/V600A-2E1-cPCI	19.25 W	1.1 W	N/A	9.8 W

Technical Specifications (cont.)

Cooling Requirements

Operating temperature	0°C to +50°C
Cooling condition for maximum operating temperatures for the PCI board are	50°C — 2.3 CFM per board 40°C — 1.5 CFM per board 30°C — 1.1 CFM per board
Cooling condition for maximum operating temperatures for the CompactPCI board are	50°C — 3.1 CFM per board 40°C — 2.1 CFM per board 30°C — 1.6 CFM per board
Storage temperature	–20°C to +70°C
Humidity	8% to 80% noncondensing

Approvals

Safety and EMC Certifications	
Canada	ICES-003 Class A ULc 60950 File E96804
Europe	EN60950 EN55022 EN55024
Japan	VCCI Class A
US	FCC Part 15 Class A UL 60950 File E96804
International	IEC60950 CISPR 22 CISPR 24
Telecom Approvals	For country-specific approval information, see the Global Product Approvals list at http://resource.intel.com/globalapproval/globalapproval.asp or contact your Authorized Distributor.
United States	EBZUSA-31207-XD-T
Canada	IC:885-7969A
European Union	DoC 01/10/2003

Reliability/Warranty

Estimated MTBF	PCI: 96,000 per Bellcore Method I CompactPCI: 64,000 per Bellcore Method I
Warranty	Intel® Telecom Products Warranty Information at http://www.intel.com/network/csp/products/3144web.htm

Resource Technical Specifications

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 Kb/s	300 Hz to 2600 Hz ±3 dB
	32 Kb/s	300 Hz to 3400 Hz ±3 dB
	64 Kb/s	300 Hz to 3400 Hz ±3 dB
Audio Digitizing		
	8.5 Kb/s	TrueSpeech
	13 Kb/s	GSM (TIPHON, MSGSM)
	16 Kb/s, 24 Kb/s, 32 Kb/s, and 40 Kb/s	G.726
	24 Kb/s	OKI* ADPCM @ 6 kHz sampling
	32 Kb/s	OKI ADPCM @ 8 kHz sampling
	32 Kb/s	IMA ADPCM @ 8 kHz sampling
	48 Kb/s	G.711 PCM (μ-law for T-1 and A-law for E-1) @ 6 kHz sampling rate
	64 Kb/s	G.711 PCM (μ-law for T-1 and A-law for E-1) @ 8 kHz sampling rate
	64 Kb/s	Linear 8 kHz 8-bit WAV
	128 Kb/s	Linear 8 kHz 16-bit WAV
	88 Kb/s	Linear 11 kHz 8-bit WAV
	176 Kb/s	Linear 11 kHz 16-bit WAV
	Digitization selection	Selectable by application on function call-by-call basis Pitch controlled
	Playback speed control	Available on the following 8 kHz coders: OKI ADPCM, G.711 PCM, Linear 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	(T-1) –36 dBm to +3 dBm per tone, configurable by parameter** (E-1) –39 dBm to 0 dBm per tone, configurable by parameter**
	Minimum tone duration	32 ms; can be increased with software configuration
	Interdigit timing	Detects like digits with a >45 ms interdigit delay Detects different digits with a 0 ms interdigit delay
	Acceptable twist and frequency variation	(T-1) Meets Bellcore LSSGR Sec 6 and EIA 464 requirements (E-1) Meets ITU-T Q.23 recommendations**
	Noise tolerance	Meets Bellcore LSSGR Sec 6 and EIA 464 requirements for Gaussian, impulse, and power line noise tolerance
	Cut-through	(T-1) Local echo cancellation permits 100% detection with a >4.5 dB return loss line. (E-1) Digital trunks use separate transmit and receive paths to network. Performance dependent on far-end handset's match to local analog loop.
	Talk off	Detects less than 10 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.

Resource Technical Specifications (cont.)

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	(T-1) Default set at –36 dBm to +3 dBm per tone, programmable (E-1) Default set at –39 dBm to +0 dBm per tone, programmable
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	(T-1) –43 dBm0 to –3 dBm0 per tone nominal, programmable (E-1) –40 dBm0 to +0 dBm0 per tone nominal, programmable
MF Signaling (T-1)	**R1**	
	MF digits	0 to 9, KP, ST, ST1, ST2, ST3 per Bellcore LSSGR Sec 6, TR-NWT-000506 and CCITT Q.321
	Transmit level	Complies with Bellcore LSSGR Sec 6, TR-NWT-000506
	Signaling mechanism	Complies with Bellcore LSSGR Sec 6, TR-NWT-000506
	Dynamic range for detection	–25 dBm to +3 dBm per tone
	Acceptable twist	6 dB
	Acceptable freq. variation	Less than ±1 Hz
MF Signaling (E-1)	**R2**	
	MF digits	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
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 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	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	Application enable/disable Supports up to three different user-definable dial tones Programmable dial tone drop out debouncing (when not part of regulatory approval)

Resource Technical Specifications (cont.)

Tone Dialing		
	DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6, TR-NWT-000506, ITU-T Q.23
	Frequency variation	Less than ±1 Hz
	Rate	10 digits/s, configurable by parameter**
	Level	(T-1) –4.0 dBm per tone, nominal, configurable by parameter** (E-1) –7.0 dBm per tone, nominal, country-specific**
Conferencing		
	Max. parties per conference	Up to 60
	Bridging/cascade conferencing	Lets you bridge together conferences from different DSPs and boards, consuming just one extra time slot 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 of which party is talking so the application can process that information and act accordingly
	Number of active talkers	Dynamically selectable
	Modes	Duplex, monitor, coach, pupil
Facsimile		
	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	Single page Multipage
	Compression	MH (ITU T.4, 1D) MR (ITU T.4 2D) MMR (ITU T.6) Onboard, on-the-fly
	ECM	Supported
	ASCII to TIFF	On-board,on-the-fly
	Page headers	Generated on board, on-the-fly
	Width	A4
	Polling	Normal and turnaround
	Resolution	Standard (100 dpi x 200 dpi) Fine (200 dpi x 200 dpi) Superfine (200 dpi x 400 dpi)
	JPEG/JBIG	Color fax and gray scale fax pass-through feature
Protocols		
	T-1 CAS	E&M (wink start, immediate start), loop start, ground start; feature group A, B, and D
	T-1 ISDN	NI-2, 4ESS, 5ESS*, DMS100, DMS250, INS1500, Q.Sig
	E-1 CAS	Many country-specific MFC-R2 variants. For more details, refer to the latest Global Call Protocols Package release notes.
	E-1 ISDN	NET5, DPNSS, DASS2, Q.Sig

Additional Components (with Item Market Names)

- Multidrop CT Bus cables (CBLCTB68C3DROP, CBLCTB68C4DROP, CBLCTB68C8DROP, CBLCTB68C12DROP, CBLCTB68C16DROP)
- CT Bus/SCbus adapter (CTBUSTOSCBUSADP)
- SCbus terminator kits (1SCBUS1TERMKIT, 2SCBUS1TERMKIT, 3SCBUS1TERMKIT)
- Rear I/O module for CompactPCI boards
  — **Unkeyed** (works in all chassis): CPCIREARRJ48, CPCIREARE1120, REARIOV19E175
  — **Keyed** (works only in keyed/guided chassis): CPCIREARRJ48KYD, CPCIREARE1120KY, REARIOV19E175KY
- 120 Ohm to 75 Ohm converter (supplied by a third party)

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Positive Answering Machine Detection/Positive Voice Detection
These performance results were measured using specific computer systems and/or components within specific lab environments and under specific system configurations. Any difference in system hardware, software design, or configuration may affect actual performance. The results are furnished for informational use only and should not be construed as a commitment by Intel. Intel assumes no responsibility or liability for any errors or inaccuracies.

Outbound Dialing/Telemarketing
Outbound dialing systems may be subject to certain laws or regulations. Intel makes no representation that Intel products will satisfy the requirements of any such laws or regulations (including, without limitation, any regulations dealing with telemarketing).

** Configurable to meet country-specific PTT requirements. Actual specification may vary from country to country for approved products.

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