OVERVIEW

The DSP SDK provides an environment for customer development of software for the Voiceboard PMC41 series of PMC DSP mezzanine cards. The PMC41 mezzanine integrates up to 12 TI TMS320C5441 533MIPS DSP’s, an 8245 Power PC processor, 128MB of SDRAM and 16MB of flash. RS232 and JTAG debug ports are included.

The SDK supports software developed under TI Code Composer Studio™. The SDK includes example source code for supporting the DSP McBSP serial ports and HPI host port interfaces. I/O protocols are provided for interfacing the DSP’s with the 8245 PPC.

The 8245 code supports flash file memory, DSP load utility, TCP/IP and UDP/IP communication over dual 10/100bT Ethernet, web server, and TFTP file transfers. Source code for a host resident API is provided that communicates with the PMC41 mezzanine via ping-pong memory buffers for data plane transfers and queued buffers for control plane messaging. Host API drivers are available for the Linux, Solaris, VxWorks and Windows operating systems.

A wide variety of common telecommunications software libraries are available for integration with customer developed code, including tone generators/detectors, conferencing and most CCITT vocoders.

Voiceboard professional engineering services is available to support customer engineering projects ranging from technical support to turnkey development of applications and web server software.

DSP SDK FEATURES

- Field proven off the shelf software libraries may be integrated with customer provided code to speed delivery schedules, reduce development cost and risk
- Drivers for DSP McBSP serial ports and HPI port included
- Embedded 8245 TCP/IP and UDP/IP stack includes a full complement of features, including encryption
- Host API and portable host driver for most popular OS software supports multi-threaded communications between host CPU and the PMC41 mezzanine card
- Universal messaging channel supports direct communication from the host application to the customer DSP API.
- DSP and host API example application code for common applications included.

TYPICAL APPLICATIONS

- Cellular base station voting algorithm
- Custom vocoders not included in our VoIP library for cellular, satellite and radio communications
- Simulation of radar threat signatures for aircraft crew training
- Simulation of airport marker beacons
- Custom modem protocols
- Voice and data encryption

AVAILABLE DSP RESOURCE LIBRARIES

- Telephony Functions:
  - DTMF and MF detection and generation.
  - Universal call progress tone detection and generation.
  - G.168 echo cancellation.
  - Caller ID detect.
  - Modem carrier detect.
- Conferencing and mixing:
  - Dynamic Conferencing: Used in a large PSTN conference environment.
  - Static Conferencing: Used primarily in military, radio communications, conference and intercom environments.
  - Mixing: Permits a user to monitor multiple audio sources with individually adjustable audio levels.
- Vocoders: G.711 µ-Law/a-Law PCM; G.726 40/32/24/16 Kbps ADPCM; G.729 A/B 8 KHz toll quality CS-ACELP; G.723.1 IMTC Compliant - 5.3/6.3 Kbps; Contact Voiceboard regarding availability of EVRC, MELPe, FR/EFR and AMR vocoders.
The PMC41-DSP mezzanine is logically divided into the following areas:

- **Motherboard control interface**
  The PMC41-DSP communicates with the host processor and with peer resources, using the motherboard control interface. The control interface consists of the motherboard interface, and the DSP I/O channels, tied together by the local control bus.

- **Data Interface**
  PMC41-DSP boards utilize the motherboard local TDM bus for transfer of data to and from the network. The data interface consists of the DSP serial and parallel I/O ports and the TDM switch, tied together by the local TDM bus.

- **TDM Switch**
  Flexible timeslot interchange routing is accomplished via an industry standard H.110 (cPCI) or SCSA (VME) bus switch, providing 4,096 (cPCI) or 2,048 (VME) timeslots, usable as either inputs or outputs.

  Under the MediaPro hardware and software implementation, the TDM bus is capable of time slot bundling (multiple channels forming a wide bandwidth channel) and sub-rate switching (multiple channels occupying a single timeslot) with a low latency.

- **Digital Signal Processors**
  Digital Signal Processor resources are shared over the TDM bus and are not dedicated to the network interface board they are physically mounted on.

  Resources are dynamically assigned and released from the available resource pool by the system host. Resource pooling reduces the number of resources required on a system wide basis and provides more available processing power for implementing MIPS intensive tasks than a dedicated DSP architecture.
PMC41 DSP Block Diagram

8245 DSP Controller
1. Host Post Interface (HPI) Module
2. Software Load Manager
3. Resource Allocation
4. PCI Interface Module
5. TCP-UDP/IP Driver Module
6. Protocol Processing for VoIP
7. G3 FAX Management Layer
8. Modem Error Correction
9. Modem AT Command Set

860/8260 Processor
1. Telephony Signaling Module
2. System Services
3. Local Resource Allocation
4. Alarms and Watchdog
5. TSI Control

Service | Max Port Capacity, each PMC-41
---|---
G.711 VoIP | 480
G.723, G.726, G.729 VoIP | 240
Universal Voice Messaging/IVR | 480
G.168 Echo Canceller | 480
Multi-party Conferencing | 480
SELECTED DSP SDK ALGORITHM SPECIFICATIONS

- **Call Progress Tone Detection**: Detects North America precise tones for dial, busy, audible ring, intercept, reorder, message wait, and recall dial.
- **MF Tone Detection**: Detects the presence of all 15 R1 Multifrequency signals, all 15 R2 Forward and 15 R2 Backward Multifrequency signals, Line Signaling tones for both R1 and R2 signaling and detects leading and trailing edge.
- **Tone Generation**: Generates dial, ringback, busy, and reorder tones, all 16 DTMF digits and all MF tones. High and low frequencies (±1Hz), power levels (±.5db steps), and make and break time (in 1 msec units) set through configurable parameters.
- **AGC (VOX)**: Classifies voice activity as “early” or “sustained” according to user specified parameters, adapts to different background ambient noise or line noise and adapts to echo when used in conjunction with the PMC41-DSP network echo canceller. AGC adjusts voice signal to a specified target power level within a specified adaptation time.
- **Network Echo C canceller**: ITU (CCITT) G.168-2000 compliant, operates in fast adapt, slow adapt, and fixed modes.
- **G.711 PCM Vocoder**: Selectable input format; linear, a-law, or mu-law.
- **G.723.1 MP-MLQ/ACELP Vocoder**: Compresses codec or linear data to either 5.3 or 6.3 kbps code, verified with the G.723.1 test vector to guarantee interoperability with other G.723.1 coders.
- **G.726 ADPCM Vocoder**: Compresses voice data to 40 kbps, 32 kbps, 24 kbps, or 16 kbps. 32 kbps ADPCM meets the “toll quality” requirements of speech applications.
- **G.729 CS-ACELP Vocoder**: Compresses codec or linear data to 8 kbps code, verified with the G.729 test vector to guarantee inter-operability with other G.729 coders.
- **Dynamic Conferencing**: Allows for up to 16 active speakers per DSP. Number of listeners and participants is only limited by the number available inputs and outputs in the system. Conferences can be conducted with or without AGC, gain normalization and echo cancellation techniques.
- **DTMF Directional Removal**: Detects the presence of all 16 DTMF digits or 12 DTMF digits produced by different telephones; eliminates DTMF signal within 12 msec of tone onset. Detects leading and trailing edge and eliminates tone bounces; Programmable twist, dynamic range, and tone duration: 24, 32 and 40 msec

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**Vocoder Performance**

![Vocoder Performance Chart](chart.jpg)

Represented By

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