

DATA SHEET

CONFERENCING— C6416

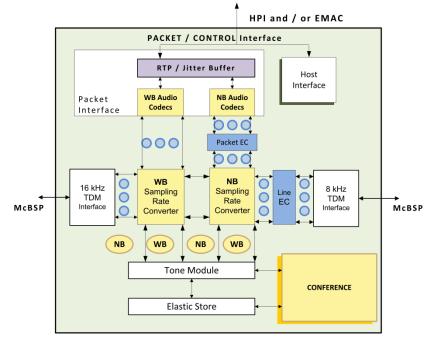
WB / NB Conferencing

TARGET APPLICATIONS

Conferencing

OVERVIEW

The Conferencing chip C6416x (Conf-C6416x) provides the DSP software components necessary in a voice-overpacket system and provides an application interface to allow easy integration into a user's application.



Software Block diagram

SOFTWARE FEATURES

- G.168 EC Certified by AT&T Voice Quality Lab
- Available in wideband, narrowband or mixedband versions
- Supports RTP type packet payloads
- ITU G.168-2002 Compliant
- Caller ID
- Excellent voice quality maintained even in large conferences

Voice Playout

Voice Record

Supports TDM or packet interface for input and output data

OPTIONAL SOFTWARE FEATURES

VoIP Codecs, such as G.723.1A, G.722, G729AB



DATA SHEET

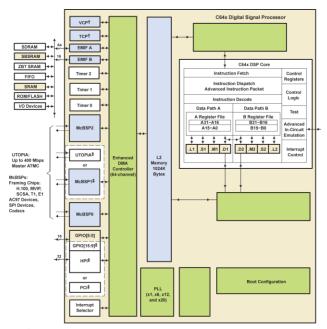
In addition to the conferencing feature, the conferencing chip provides tone detection and tone generation capabilities. Tone detection is typically used in a conference server to allow the user to enter a conference ID and password, or to perform other control functions. Tone generation is used to generate alerting signals.

HARDWARE FEATURES -

- 600-/720-/850-MHz, 1-GHz Clock Rate
- 4800, 5760, 6800, 8000 MIPS
- L1/L2 Memory Architecture 128K-Bit (16K-Byte)
 L1P Program Cache
- 128K-Bit (16K-Byte) L1D Data Cache
- 8M-Bit (1024K-Byte) L2 Unified MappedRAM/Cache
- Two External Memory Interfaces (EMIFs) 1280M-Byte Total Addressable ExternalMemory Space
- Host-Port Interface (HPI) User-Configurable Bus Width (32-/16-Bit)
- 32-Bit/33-MHz, 3.3-V PCI Master/SlaveInterface Conforms to PCI Specification2.2
- Three Multichannel Buffered Serial Ports Direct Interface to T1/E1, MVIP, SCSAFramers – Up to 256 Channels Each

ADDITIONAL HARDWARE FEATURES

- VelociTI.2 Extensions to VelociTI Advanced
 Very-Long-Instruction-Word(VLIW) TMS320C64x
 DSP Core
- Sixteen General-Purpose I/O (GPIO) Pins
- Flexible PLL Clock Generator



TVCP and TCP decoder coprocessors are applicable to the C6416T device only.

For the C6415T and C6416T devices, the UTOPIA peripheral is muxed with McBSP1, and the PCI peripheral is muxed with the HPI peripheral and the GPIO(15:9] port. For more details on the multiplexed pirs of these peripherals, see the Device Configurations section of this data short.

Hardware Block diagram

SPECIFICATIONS

Application	Product Number/Silicon	Channel Count	Description
Conferencing	TMS320C6416	512	Packet EC, G.711, VAD/CNG, AGC, Tone Gen, Tone Detect, DTMF High Performance Fixed-Point C64x+ core 600-/720-/850-MHz, 1-GHz Clock Rate 32-/16-Bit Host-Port Interface (HPI) 3 McBSPs

DETAILED DESCRIPTION

The VoIP Packet Conferencing Chip-C6416x provides the DSP components necessary in a voice-over-packet system and provides an application interface to allow easy integration into a user's application.

A system built with CC-C6416x consists of a host control processor connected to one or more DSPs loaded with CC-C6416x. The host processor typically connects to a communication network and the DSPs typically connect to one or more TDM serial streams.

- The conference chip supports RTP type packet payloads. The DSP and host will transfer RTP payload packets once every 10 mS.
- The system provides the following functionality:
- Tone detection: DTMF, MFR1, MFR2 Forward, MFR2 Backward, and TAS, FAX CNG/CED
- Detection of tones with non-standard tone on and off times. Tone detection events are reported to the host when the DSP first detects the presence of a tone, and again when the DSP determines that a detected tone has ended. The exception to this rule is FAX CNG/CED detection; for this tone type only the tone-on event is reported to the host.
- Complete tdm-input-side tone suppression
- Tone generation under host control in either the TDM out or packet out direction. Tone generation is only allowed in one direction at a time.
- Automatic gain control (AGC) or fixed gain block. Fixed gain blocks can replace an AGC block wherever it occurs
 in the system. In addition, fixed gain blocks are available for conference output data. The fixed gains can be
 changed dynamically during the course of a call.
- Voice activity detection (VAD). VAD status is reported to the host once per 10 mS.
- Comfort noise generation (CNG)
- V.23 and Bell202 fsk modem to support type 1, 2, and ADSI caller ID. The FSK input and output signals, as well as associated Caller Id tone signals are transported via the TDM interface. A caller Id channel can operate as transmit or receive, but not both simultaneously. The host can change a channel's cid state (tx or rx) in real-time via an API command.
- Conferencing with noice cancellation. The chip supports up to 170 conferences. Each conference can have up to 12 members plus 1 composite output channel. Conference participants can be a combination of TDM and packet channels

Each CC-C6416x DSP supports a fixed number of conferences. The number of conferences available in a DSP is determined at build time. Each CC-C6416x channel can be used as a conference member. Figure below shows a block

Device Overview 3

diagram of a CC-C6416x conference. Refer to the Conferencing Algorithm Software User's Guide for a detailed description of the conferencing algorithm.

Members of a conference can be Conference PCM and Conference Packet type channels. The input data from all member channels is summed and normalized. Each member channel's output data is the normalized sum minus the channel's own input data.

A Conference Composite type channel can be assigned to a conference to output the normalized sum of all input channels as PCM data and/or as packet data.

Deliverables

The deliverable items are platform dependent. In general, there is a single DSP-downloadable binary image along with host API software in C source code format. Also included in the deliverables is product documentation, which includes a users guide and usually includes release notes. Sample/test code may be included as well.

Adaptive Digital is a member of the Texas Instruments Developer Network, and ARM Connected Community.

CONTACT INFORMATION

Web: www.adaptivedigital.com
Email: information@adaptivedigital.com

Tel: 610.825.0182 ~ Toll Free: 1.800.340.2066

Fax: 610.825.7616

Address: 525 Plymouth Road, Suite 316

Plymouth Meeting, PA 19462





IMPORTANT NOTICE: Data subject to change, for the most up to date information visit our website. Customers are advised to obtain the most current and complete information about Adaptive Digital products and services before placing orders.

All trademarks are property of their respective owners.



Device Overview 4