Multi-Microphone Conference Endpoint - DM816X

TARGET APPLICATIONS

IP Based Conference Phone System

IP Based Intercom

OVERVIEW

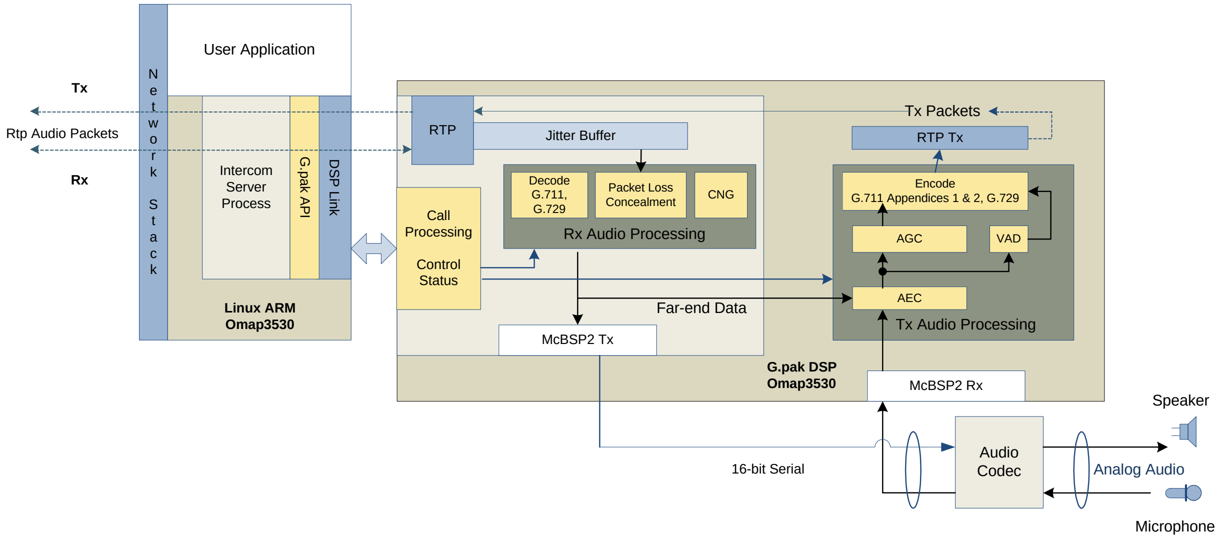
Adaptive Digital's IP Multi-Microphones Conference Endpoint product combines Adaptive Digital's IP Conference DSP software plus host API along with Texas Instruments' DM816X DSP to form a turnkey soft-chip for use in conference phone equipment.

SOFTWARE FEATURES

- Acoustic Ech Cancellation (AEC)
- Multi Microphone Input
- Digital Gain Control
- Noise Reduction
- Automatic Gain Control (AGC)
- G.711
- RTP + Jitter Buffer

Optional Software Features

- Video
- Wideband Support
- VoIP Codecs, such as G.723.1A, G.722, G729AB
- Tone Detection



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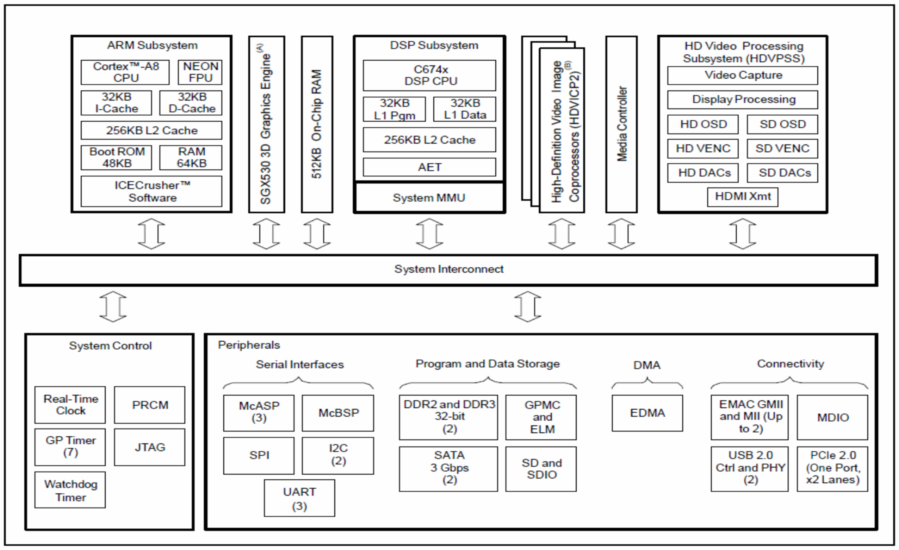
HARDWARE FEATURES

- Up to 1.35GHz* ARM Cortex™-A8 Core Coprocessor
- Up to 1.125-MHz* TMS320C674x™ Floating-Point DSP Core
- Fully Software-Compatible With C64x and ARM9™
- Endianess:
- ARM Instructions - Little Endian
- ARM Data – Configurable
- DSP Instruction/Data - Little Endian
- Enhanced Direct Memory Access (EDMA) Controller (4 of 128 available Independent Channels)
- C64x+ L1/L2 Memory Architecture
- 32K-Byte L1P and L1D RAM and Cache
- 256K-Byte L2 Unified Mapped RAM/Cache
- 1 of 3available Multichannel Buffered Serial Ports (McASP 2)
- 1 of 7 available 32-bit General Purpose Timers
- Two 10 Mbps, 100 Mbps, and 1000 Mbps • On-Chip ARM® ROM Bootloader (RBL) Ethernet MACs (EMAC)

ADDITIONAL HARDWARE FEATURES

- Dual 32-Bit DDR2 and DDR3 SDRAM Interfaces
- System Memory Management Unit(System MMU)
- Media Controller
- Up to Three Programmable High-Definition Video Image Coprocessing(HDVICP2) Engines
- SGX530 3D Graphics Engine
- HD Video Processing Subsystem (HDVPSS)
- Dual USB 2.0 Ports With Integrated PHYs
- General Purpose Memory Controller (GPMC)
- Test Interfaces
- Multichannel Buffered Serial Port (McBSP)
- Three Configurable UART, IrDA, and CIR
- One 40-MHz Serial Peripheral Interface (SPI)
- Up to 64 General-Purpose I/O (GPIO) Pins
- Power, Reset, and Clock Management



PRODUCT OVERVIEW

Adaptive Digital's IP Multi-Microphones Conference Endpoint product highlights our field proven AEC, voice quality enhancement algorithms in IP-based conference phone applications.

Adaptive Digital's Multi-Microphone Conference Endpoint product is a software subsystem that simplifies software design of an IP conference phone or IP intercom. It implements complete VoIP capability all the way from PCM to Packet and back. This includes a process running on the ARM under Linux as well as all the necessary voice processing running on the DSP core.

A user's application, co-resident on the ARM, can set up and tear down VoIP channels via an easy to use API. The DSP software takes care of everything else.

SPECIFICATIONS

Application	Product Number/Silicon	Channel Count	Description
Conference Speaker phone	DM8168/1.0GHz	2 Microphones 1 Speaker 1 RTP/Encoder/Decoder	AEC, Noise Reduction, G.711, RTP with Jitter Buffer, AGC, Digital Gain

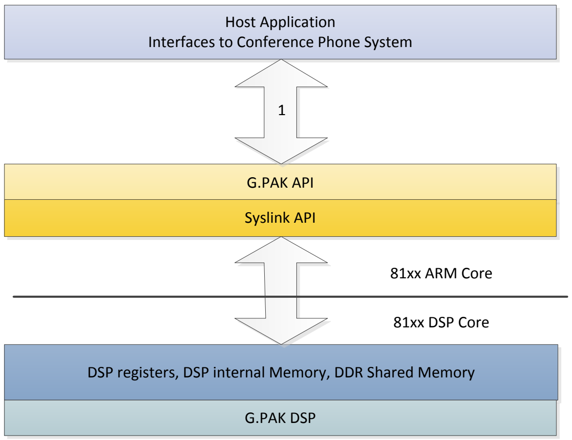
DETAILED DESCRIPTION

The Adaptive Digital's IP Multi-Microphones Conference Endpoint product runs on TI DM816X family of DaVinci processors. The ARM core is responsible for the call control, user interface, network interface, and protocol processing, while the DSP core is responsible for the audio processing. All control APIs functions on ARM core are non-blocking. Furthermore, video can be added by leveraging the DM816X's HD Video Processing Subsystem.

The host software runs as a Linux process on the ARM core of DM816X processor, calling the appropriate API functions, which in turn, control the DSP via TI's SYSLINK software. Subsequent status messages are returned to the ARM. In addition to supporting messages, the host process is responsible for booting and downloading the DSP portion of the DM816X device, controlling the DSP's operation, and the timely transfer of RTP packets between the network stack and the DSP software.

The DSP supports a voice over IP Interface application running a single voice channel with the following features:

- **Vocoders**: G.711 μ-Law, G.711 A-Law, and other optional VoIP codecs(G.729AB, AMR, etc) with packet loss concealment
- **Multiple Microphone Inputs**: Via the DM816X McASP TDM port.
- **Voice enhancement algorithms:** Acoustic Echo Cancellation (AEC Gen4), VAD/CNG, AGC, RTP with Jittter Buffer, Noise Reduction.
- **Digital Gains** on Microphones and Speaker
- **Conference**: Mixing the multiple microphones input



HOST API

Adaptive Digital Technologies IP Multi-Microphones Conference Endpoint *host software consists:*

- API C-code that enables the customer's host application to control the DSP software
- Host software C-code Example

REFERENCES

1. Adaptive Digital Technologies IP Multi-Microphones Conference Endpoint DM816X Users Guide

2. Texas Instruments TMS320DM816X DaVinci Video Processors (SPRS614D)

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.

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