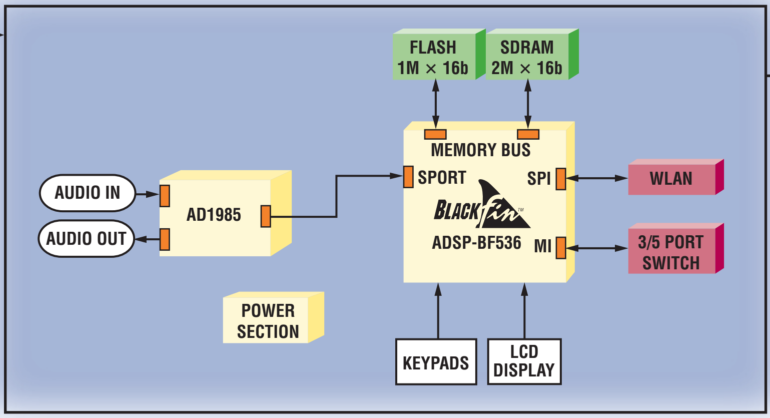
Blackfin BRAVO Voice-over-IP (VoIP)—Communications Chipset/Reference Design



Blackfin BRAVO VoIP Communications Reference Design

Complete Voice-over-IP Solution Reduces Time-to-Market by Up to 80%

Do you need to get to market quickly with a VoIP solution that is highly integrated, cost-effective, standards compliant, and fully tested? Would you rather spend your time differentiating your product instead of trying to get software modules from various suppliers to work together?

Analog Devices' Blackfin® BRAVO™ Voice-over-IP reference design is a complete system solution for building feature-rich, high performance, low cost VoIP desktop phones and telephone adaptors. The design includes the complete suite of software for VoIP applications, all controlled by a comprehensive set of application program interfaces (APIs). The easy to use APIs provide the functionality for customization and control of core system functions, letting you focus on adding high value through product differentiation.

High Quality Audio with Full Duplex Speakerphone and Call Control Features

No compromise on performance here. The Blackfin BRAVO VoIP solution with the AD1985 audio codec, provides exceptional audio quality, advanced call control features, and more. For I/O, the chipset supports 64-keys on the input (IR optional), and on the output supports LCDs and Dot Matrix displays. At the heart of the chipset is the Analog Devices Blackfin Processor with integrated Ethernet MAC. The processor includes an enhanced instruction set to support multimedia audio and video functionality, with an architecture that is optimized to perform equally well on both control and numeric algorithms.

For audio, the design supports ITU standard compliant G.711, G.722*, G.722.1*, G.723.1, and G.729AB audio codecs, with support for multiline SLICs, G.168 ITU standard compliant network echo cancellation, and acoustic echo cancellation (AEC) for enhanced audio clarity. Optionally, RF transceivers can be included in the design to provide cordless audio capability. The design supports ITU Standard H.323 and SIP compliant software stacks and includes autodetection.

Reference Design Kit

The reference design kit includes the VoIP development platform, full system software, documentation, and technical support.

Features

- H.323 and SIP protocol compliant with autodetection, RTP, RTCP, and static NAT traversal.
- Supports G.711, G.723.1, G.729AB, G.722*, and G.722.1* audio codecs.
- Integrated acoustic echo cancellation (AEC) with noise cancellation for high quality full duplex speakerphone functionality.
- Adaptive jitter buffer, packet loss concealment (PLC), voice activity detection (VAD), and comfort noise generation (CNG) for smooth high quality audio under various network conditions.
- Ethernet and WLAN interfaces with QoS.
- Optional WLAN/IEEE 802.11x connectivity supports WEP, WPA, and QoS.
- Quality of service (QoS) features include Difserve, 802.1p, and TOS.
- Advanced call control features include call forward, call transfer, call hold, call waiting, and 3-way conferencing.
- Other features include 7-segment LCD and dot matrix support, LED indicators for power-on and conversation, and 64-key keypad support.
- SIP features include instant messaging, presence, HTTP digest access authentication, message wating indicator, and many more.
- Gatekeeper registration
- Flexible software to enable customized GUIs
- Complete, highly integrated, and cost-effective system design.
- Development board, sample applications, and all necessary support included.

Applications

- Desktop VoIP phone
- VoIP telephone adaptor
- VoIP-enabled router

ANALOG DEVICES

www.analog.com/processors

Hardware Specifications

External Memory

- 2 MB Flash
- 4 MB (SDRAM)

Video Display

- 7-Segment LCD and Dot Matrix

Audio I/O

- Microphone—1 channel for dynamic or condenser microphone
- Speaker—1 W @ 8 Ω
- 4 kHz to 48 kHz sampling rates
- Programmable gain
- Optional RF transceiver for cordless phone function

Functional Specifications

Protocol Stacks

- ITU standards H.323, SIP
- TCP/IP, UDP, DHCP, ARP
- Static NAT, STUN*, UPnP*, SMS*
- RTP and RTCP

Audio

- G.711, G.723.1, G.729AB, G.722*, and G.722.1*
- Acoustic echo cancellation (AEC) with noise cancellation
- Voice activity detection (VAD)
- Comfort noise generation (CNG)
- Audio mute
- Multiline SLIC interface*

SIP Features

- SIP according to RFC 3261
- SIP instant messaging, RFC 3428
- SIP presence, RFCs 3856, 3863, 3903
- Offer answer model with SDP, RFC 3264
- HTTP digest access authentication, RFC 2617
- Method to locate SIP servers, RFC 3263
- Configurable signaling and RTP ports
- Out-of-band DTMF tones, RFC 2833
- DNS lookup
- Call hold using reINVITE method
- Call forwarding
- Call waiting
- Call transfer
- 3-way conference
- Message waiting indication, RFC 3842

SIP Features (continued)

- RFC3262 reliability of provisional responses in session initiation protocol (SIP)
- URLs for telephone calls, RFC 2806
- Hypertext transfer protocol (HTTP) digest authentication using authentication and key agreement (AKA), RFC 3310*
- REGISTER SUBSCRIBE/NOTIFY, RFC 3680*

Communications Ports

- Ethernet and UART
- IEEE 802.11x*

Reference Design Includes

- System Architecture
- User Manuals
- Schematics, BOM, and Gerber files
- User interface source code
- Core functions in library/object code
- Diagnostics and test plan
- Training kit

Voice-over-IP (VoIP) Reference Design

- Complete system solution for building VoIP phones. Reduces time to market up to 80%.
- VoIP reference design engine can be used as a standalone product or integrated into a larger system.
- Software includes the full suite of software/firmware required for the application, including audio codecs and H.323/SIP/network stacks.
- A comprehensive set of application program interfaces (APIs) allow easy customization and control of core functions.
- Documentation includes software and hardware manuals that cover system architecture, operation, APIs, schematics, BOM, Gerber files, diagnostics, and test plan.
- Development platform: The VoIP reference design kit (RDK) serves as the development platform and includes VoIP development boards and full documentation.
- Remote software update support—allows updating of the software on the board remotely via a standard PC.

Ordering Information

Contact Analog Devices, Inc. for ordering information and for information concerning available development platforms and software tools. Designers of products using this reference design will be required to sign a license agreement with ADI. For more information on the BRAVO VoIP solution, email systems.solutions@analog.com.

*Future support

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