



# Fusion VoIP Phone Reference Design

The Fusion VoIP Phone reference design is a complete software/silicon solution that offers a full-featured platform that supports current and emerging IP phone standards, and has expansion capabilities for product differentiation.

The Unicoi IP Phone Reference Design reduces the Bill of Materials (BOM) costs as well as the time and complexity typically associated with developing an IP Phone. Designed around the ADI Blackfin BF536, the reference design software delivers the critical processing (RTOS, call manager, voice algorithms, etc.), communication (TCP/IP, SIP, RTP, etc.) and peripheral functions (Ethernet, Flash and Interface controllers, etc.) required to build a basic or advanced IP Phone.

## Software Peripheral Set Includes

- Fully standards compliant SIP/RTP module
- Security options for Secure SIP and Secure RTP
- Acoustic echo cancellation for full duplex speaker phone capability
- Support for up to 6-way conferencing
- Support for a wide range of low bit rate and wideband codecs
- Full featured Voice Engine
- Firmware management with remote update module
- An extensible Call Manager that is easily customer configured

## Hardware Development Board Includes

- LCD controller, keypad interface, UART serial interface, 3 port Ethernet Switch, High quality wideband codec, SPI Interface, and several general purpose I/O ports
- Standard connectivity options are available for integrating a coprocessor or other peripherals
- During IP Phone operation, the ADI Blackfin BF536 has ample remaining horsepower for the customer to implement value-added features.

## The Unicoi Advantage

The Fusion VoIP Phone Reference Design provides an exceptionally efficient framework for real-time voice processing and has been pre-integrated and rigorously tested with many customer deployments. Fusion VoIP provides many benefits for all levels of users, from networking equipment manufacturers to designers who are providing next-generation equipment to service providers marketing these services to business and home users.



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## Voice Codecs

- G.711U/A
- G.722
- G.723.1
- G.726 (40, 32, 24 & 16 Kbits/sec)
- G.729 A/B
- DVI4 Narrow & WideBand

## Starter Kit Features

For immediate demonstration and development, the Starter Kit includes:

- Software Source Code with Development
- License (voice algorithms are delivered as object code)
- One Hardware Development Board with complete schematics and Gerber files
- Fusion RTOS and fully-integrated network protocol stacks (TCP/IP, SIP, RTP, etc.) enabling rapid implementation of customer's unique features and services
- RFC 2327 (SDP)
- RFC 2617 (Basic and Digest Authentication)
- RFC 2833 (RTP Payload for DTMF Digits)
- RFC 2976 (Info Method)
- RFC 3261 (SIP version 2)
- RFC 3262 (Reliability of Provisional Response in SIP)
- RFC 3263 (Locating SIP servers)
- RFC 3264 (Offer answer model with SDP)
- RFC 3265 (Specific Event Notification)
- RFC 3420 (Internet Media Type message/sipfrag)
- RFC 3515 (Refer method)
- RFC 3891 (SIP "Replaces" Header)



- RFC 3892 (SIP Referred-By Mechanism)
- RFC 2778 (A Model for Presence and Instant Messaging)
- RFC 3428 (Extensions for instant messaging)
- RFC 3856 (A presence Event Package for SIP)
- RFC 3857 (A Watcher Information Event Template Package for SIP)
- RFC 3858 (XML Based Format for Watcher information)
- RFC 3863 (A Presence information data format PIDF)
- RFC 1889, 1890 (RTP/RTCP)
- RFC 3550 (RTP)
- RFC 3551 (RTP A/V Profile)
- RFC 3959 (sender & receiver tables for session members)
- Acoustic Echo Cancellation
- Silence Detection
- Dynamic jitter buffer (adaptive)
- Packet Loss Compensation Module
- AGC TX & RX
- Volume Control for TX & RX
- Sampling rate Conversions (8K, 16K, 32K & 48K)
- Call Progress Tone Generator
- DTMF detection and generation
- 6-way call conferencing
- Dial tone
- Busy signal
- Ring back
- Distinctive ringtones
- Call transfer
- Transfer attended and unattended
- Call hold
- Caller ID
- Call waiting
- Message waiting indicator
- Supports narrow and wide band calls
- DTMF: In-Band and Out-of-Band-SIP INFO
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG) QoS
- RFC 1349 (IPToS function priority quest for upstream/downstream traffic based on ToS field)
- Voice priority always guaranteed Dialing Engine
- Support for various digit-length dialing as well as international and long distance

- Phonebook support for speed dialing
- Provides support for shortcut dialing to other pre-configured VoIP users
- Point-to-Point Dialing Management
- DHCP client

### Network Management

- SNMP Support Available
- TFTP firmware upgrade and configuration backup/restore
- Web-based configuration
- Command-line Interpreter (CLI)
- Telnet remote management
- Built-in diagnostic tool

### Hardware Specs

- LAN: 2 port 10/100 ethernet switch with auto crossover capability; IP Phone pulls off packets as necessary
- Reset button
- LEDs Indicators for network activity
- Power: Requires 12V DC 1A switching power supply or Power Over Ethernet (PoE)
- Software Upgrade
- Firmware is upgradeable either remotely or locally
- TFTP support
- HTTP support

