AudioCodes VoIP Processor Solution Guide

The Clear Sound of Quality

Featuring Rich VoIP Processor Solutions
from the world's leading enabler of the new voice infrastructure

AudioCodes

HD VoIP Sounds Better

Solution Guide

Application	Solution	VoIP Processor Product Number	Number of VoIP Channels	Number of Data Ports	Product Description	Remarks	Page
IP Phone	AC494 IP Phone VoIP Toolkit	AC494 AC495 AC495L	2	2	Hardware & Software Reference Design, Linux support, IP Phone SIP demo application	IP Phone solution available	4 - 5
IP Phone	AC494E (Orchid) IP Phone Toolkit	AC494E AC495E	2	2	Hardware & Software Reference Design Gigabit Eth support, Linux support, IP Phone SIP demo application	IP Phone solution available	10 - 11
Analog Telephone Adapter (ATA) and CPE VoIP Gateway	ATA/CPE Gateway VoIP Toolkit	AC494 AC495 AC496	Up to 8	2	Hardware & Software Reference Design, Linux support, ATA SIP demo application, FXS/FXO support		6 - 7
Analog Telephone Adapter (ATA) and CPE VoIP Gateway	Tulip ATA	AC494 AC495 AC496	Up to 8	2	Production ready module, production files available, Complete ATA SIP application, FXS/FX0 support	Optional customization of the software	7
Analog Telephone Adapter (ATA) and CPE VoIP Gateway	CPE VoIP Toolkit	AC483	Up to 4	-	Hardware & Software Reference Design, Linux support, ATA SIP demo application, FXS/FXO support		14
Analog Telephone Adapter (ATA) and CPE VoIP Gateway	AC48x VoIP DSP	AC48x	Up to 4	-	API Drivers OS independent		13
SMB VoIP Gateway and IP-PBX	SMB VoIP Toolkit - Gateway	AC494E + AC498	4 - 16	2	Hardware & Software Reference Design, Linux support, 16 port Gateway SIP demo application, FXS support	Scalable solution	8 - 9
SMB VoIP Gateway and IP-PBX	Risk Free Asterisk based IP-PBX Toolkit	AC494E + AC498	4 - 16	2	IP-PBX application		9
SMB VoIP Gateway and IP-PBX	Gladiolus SDK	AC501x	8 - 24	-	Hardware & Software Reference Design, API Drivers OS independent		9
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SMB VoIP Gateway and IP-PBX	Voice Over Packet Processor	AC501x	8 - 24	-	Hardware Reference Design, API Drivers OS independent		16
Enterprise VoIP Gateway	Voice Over Packet SoC	AC5042	Up to 32		Hardware Reference Design, API Drivers OS independent, Linux support, SIP application		12
Enterprise VoIP Gateway	Voice Over Packet Processor	AC491/ AC491L	32, 64, 96		Hardware Reference Design, API Drivers OS independent		17
Enterprise VoIP Gateway	Voice Over Packet Processor	AC503x	*168, 204, 234 *Dynamically allocated VoIP channels	-	Hardware Reference Design, API Drivers OS independent		18

AC494 Family System On a Chip (SoC)

- Single chip complete IP Phone solution
- Superior voice quality at a competitive price
- Field-proven VoIP processor and software
- VoIPerfectHD™ (High Definition VoIP) support
- Combined hardware & software reference design
- Ready to use IP Phone open platforms

The AC494 IP Phone Toolkit is a combined hardware and software IP Phone reference design based on the AC494 System on Chip (SoC) family. AudioCodes' AC494 SoC (System on Chip) family provides IP Phone developers and manufacturers with a single chip field-proven complete IP Phone solution, integrated with superior voice quality at a competitive price.

AC494 SoC (System on Chip)

In a single chip, the AC494 combines all IP Phone components; including a MIPS 4KEC CPU, AC49x DSP and a 3-port Ethernet Switch (MACs and PHYs) together with a rich set of peripherals such as internal CODECS and serial ports. The AC494 is based on the VoIPerfectHD™ architecture (High Definition VoIP) support, AudioCodes' underlying, best-of-breed, core media gateway technology for its product line. Utilizing its highly integrated structure and requiring only a minimal number of peripheral devices, time to market is rapid with little risk.

Several silicon derivatives of this family allow cost optimization per application, which includes the following devices: AC494, AC495, AC496 and AC495L.

Device	CPU Clock	DSP Clock	USB	Voice CODECs	Eth. Ports
AC494	165 Mhz	125 Mhz	V	2	2
AC495	125 Mhz	100 Mhz	-	2	2
AC495L	87.5 Mhz	87.5 Mhz	-	2	2
AC496	165 Mhz	125 Mhz	-	2	2

Channel Density
- 2 VoIP line support

Hardware
- CPU – MIPS32 4KEC, 165 Mhz
- DSP – AC49x, 125 Mhz
- 2 Integrated 10/100 Base-T MAC/PHY ports
- Integrated 3-port layer-2 Ethernet Switch
- 8x8 Keypad interface
- LCD Interface – Alphanumeric, Graphic, Rasterized
- Integrated SSP (SPI/I2C) Controller
- USB 1.1 Host/Device
- External Memory I/F – Two SDRAM banks, up to 128 MB, Flash up to 32 MB
- Telephony Interface - Integrated dual channel 16 bit CODEC, sampling rate 8/16 KHz
- Five inputs to ADC, four outputs from DAC
- PCM Interface - 2.048 MHz A/μ-Law serial port
- Power Supply - +3.3V (+1.5V core via integrated voltage regulator)
- Power Consumption - 1.88 W (max)
- Operational Case Temperature Range - 0ºC - 70ºC (commercial)
- Package - 324 pin BGA, 23x23 mm, 1.00 mm ball spacing

Telephony Signaling
- DTMF – TIA464B
- Programmable Call-Progress Tones

VoIP Signaling Protocols
- SIP - RFC 3261, 3262, 3263, 3264, 2327

Packetization
- RTP/RTCP Packetization – RFC 3550, 3551, 2198
- Caller ID - Telcordia/ETSI/NTT Type I/II (on hook / off hook)
- DTMF Relay - RFC 2833, RFC 4733

Security
- SRTP (Secured RTP) per RFC 3711, 128 bit AES, HMAC SHA1

Media Processing
- Voice Coders - G.711, G.723.1, G.729A/B, G.726, GSM FR
- Wideband Coders – G.722, G.722.2 (WB-AMR), Opus
- Echo Cancelation – G.168-2004 compliant, 64 msec tail length
- Acoustic Echo Canceler
- Silence Suppression – VAD, CNG
- Adaptive Jitter Buffer 300 msec

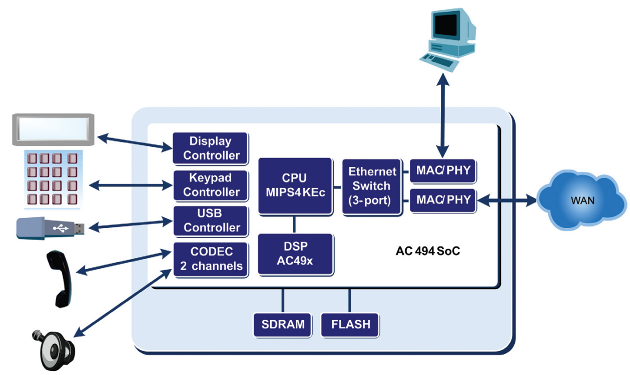
Telephony Features
- 3-Way conferencing (on each channel)
- Call Forward
- Call Hold
- Call Transfer
- Call Waiting

Configuration/Management
- Embedded Web
- TFTP
- FTP
- Telnet
- Filesystem Support

Operating System
- Linux Kernel
- Board Support Package

IP Phone Toolkit Hardware Design

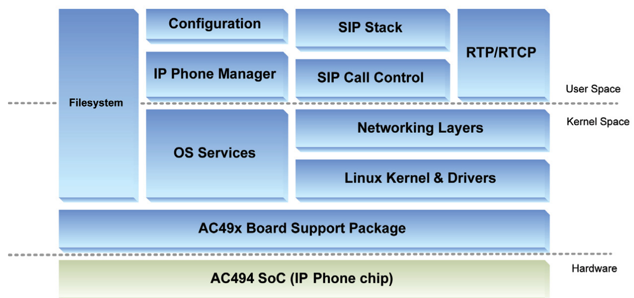
The AC494 IP Phone Toolkit hardware reference design includes all the peripherals for a complete high-end IP Phone. The AC494 SOC includes the internal interfaces for LCD Display, a keypad, USB, speaker and handset. Additionally, it has a 3-port Eswitch and internal MACs/PHYs.



IP Phone Toolkit Hardware Design

IP Phone Software Development Kit (sdk)

The AC494 IP Phone Toolkit includes a complete Linux Software Development Kit (SDK) supporting voice and network protocols. The software is available in open source code, comprised of separate modules allowing for maximum customization.



IP Phone Software Architecture

Ready to use IP Phones for developers

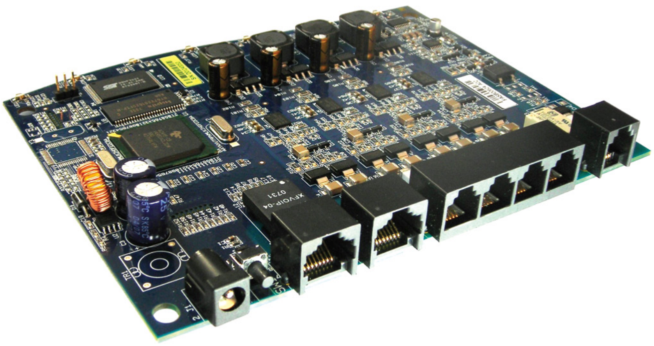
The AC494 IP Phone toolkit is also available on ready-to-use IP Phones. This allows OEMs and ODMs to develop their software on an actual IP Phone platform.

AC494 IP Phone

Toolkit Contents

- Reference & Development IP Phone Platform
- Linux Operating System
- Free Development Environment (Linux Tool-chain)
- Ready-to-use IP Phone Application (open source code)

ATA/CPE Gateway VoIP Toolkit



- Complete ATA/CPE Gateway VoIP solution
- Up to 4 FXS line interfaces with assembly options
- 3-Way conferencing per line support
- VoIPerfectHD™ (High Definition VoIP) support, enabling superior voice quality at a competitive price
- Field-proven VoIP processor and software
- Perfect platform for stand-alone ATA developers
- Production ready hardware and software reference design

The ATA Toolkit is a combined hardware and software ATA reference design based on the AC494 System on Chip (SoC). The AC494 SoC (System on Chip) provides ATA developers and manufacturers with single chip field-proven complete ATA solution, integrated with superior voice quality at a competitive price.

AC494 SoC (System on Chip) Family

The AC494 family combines in a single chip the main ATA components; it includes a MIPS 4KEC CPU, AC49x DSP and a 3-port Ethernet Switch (MACs and PHYs) together with a rich set of peripherals. The AC494 is based on the VoIPerfectHD™ architecture (High Definition VoIP) support, AudioCodes' underlying, best-of-breed, core media gateway technology for all of its products.

Channel Density
- Up to 4 FXS line interfaces

Hardware
- CPU – MIPS32 4KEC, 165 Mhz
- DSP – AC49x, 125 Mhz / 100 Mhz
- 2 Integrated 10/100 Base-T MAC/PHY ports
- Integrated 3-port layer-2 Ethernet Switch
- Integrated SSP (SPI/I2C) Controller

Telephony Signaling
- DTMF – Detection and Generation, TIA464B
- Caller ID – Telcordia/ETSI/NTT Type I/II (on hook/off hook)
- Programmable Call-Progress Tones

Packetization
- RTP/RTCP Packetization – RFC 3550, 3551, 2198
- DTMF Relay - RFC 2833, RFC 4733

Security
- SRTP (Secured RTP) per RFC 3711, 128 bit AES, HMAC SHA1

Telephony Support
- FXS and FXO supported

Media Processing
- Voice Coders - G.711, G.723.1, G.729A/B, G.726, GSM FR
- Wideband Coders – G.722, G.722.2 (WB-AMR)
- Fax – T.38 Relay, Bypass , T.38 over RTP
- Echo Cancelation – G.168-2004 compliant, 64 msec tail length
- Silence Suppression – VAD, CNG
- Adaptive Jitter Buffer 300 msec

Telephony Features
- 3-Way conferencing (on each line)
- Call Forward
- Call Hold
- Call Transfer
- Call Waiting

Data Protocols
- IPv4, TCP, UDP, ICMP, ARP
- PPPoE
- Layer 2 switching
- DHCP Client/Server – RFC 2132
- WAN to LAN Layer-3 routing
- DHCP Client/Server – RFC 2132
- NAT – RFC3022, Application Layer Gateway (ALG)
- IEEE 802.1p/q QoS (VLAN tagging)

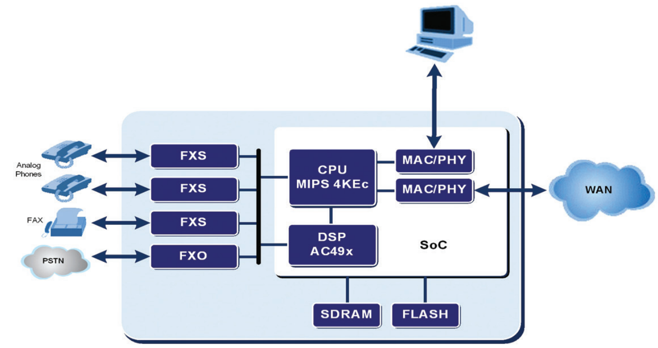
VoIP Signaling Protocols
- SIP - RFC 3261, 3262, 3263, 3264, 2327

Configuration/Management
- Embedded Web
- TFTP
- FTP
- Telnet
- Filesystem support

Operating System
- Linux Kernel
- Board Support Package

Development Platform and Reference Design

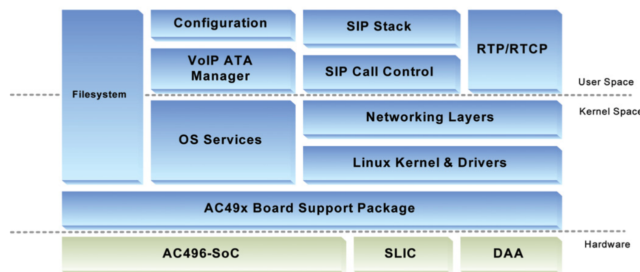
The ATA VoIP Toolkit is a combined hardware and software ATA reference design. The ATA VoIP Toolkit is based on the AC494 System on Chip (SoC) family. The SoC combines in a single chip, all the ATA components; it inlcudes a MIPS 4KEC CPU, AC49x DSP and 2-ports MACs/PHYs together with a rich set of peripherals. The hardware reference design includes also the analog FXS module.



The ATA Toolkit Hardware Design

ATA Software Development Kit (SDK)

The ATA VoIP Toolkit couples the hardware with a complete Linux software suite supporting voice and network protocols, call features and management applications. The software is available in open source code, combined of separate modules allowing customers to easily differentiate themselves.



ATA Software Architecture

Toolkit Contents

- Reference Design & Development Platform based on AC494-SoC
- Ready-to-use scalable ATA modules
- Linux Operating System
- Free Development Environment (Linux Tool-chain)
- Ready to use VoIP ATA Application (open source code)

Tulip ATA

The Tulip ATA based on AC494-SoC family is a complete, ready to-use reference design of an ATA and Gateway with data routing capabilities. Utilizing AudioCodes field-proven VoIPerfectHD™ DSP software and the integrated AC494 System on Chip (SoC) family, the Tulip ATA offersOEMs and ODMs an excellent and cost effective solution for the rapidly growing residential and Small Office/Home Office (SOHO) VoIP market.

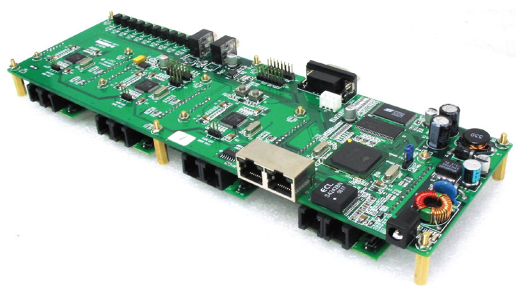
The Tulip ATA have a costeffective Bill of Materials (BOM) to support aggressive market cost demands. An additional data port provides the option to convert a PC to the ATA without supplementary hardware. Special attention was given to designing the hardware and PCB to shorten customer investment in the homologation process.

Tulip ATA

Toolkit Contents

- Production ready Analog Telephone Adapter (ATA) reference design
- Available as a module or through production license
- Data Routing Capabilities
- Minimizing risk and allowing quick time to market for ODMs and OEMs rolling out VoIP-based services

SoC Based SMB VoIP Toolkit



- Complete SMB VoIP Gateway Solution
- Up to 16 FXS lines at a competitive price
- Field-proven VoIP processors and software
- VoIPerfectHD™ (High Definition VoIP) support
- Ideal solution for SOHO & SMB VoIP applications that require superior voice quality, scalability and reliability with cost-effective deployment

The SMB VoIP Toolkit is a combined hardware and software reference design for VoIP gateways with up to 16 channels, providing a complete, cost-effective solution for various applications. Utilizing AudioCodes field-proven VoIPerfectHD™ architecture (High Definition VoIP) support, the AC494E System on Chip (SoC) and the AC498 DSP, allows this reference design to offer OEMs and ODMs a cost-effective and quick time to market solution for the rapidly growing SOHO and SMB VoIP markets.

AC494E SoC (System on Chip) and AC498

In a single chip, the AC494E combines the main components of a 4 VoIP channel gateway. It includes a MIPS 4KEC CPU, AC49x DSP and 2-port MACs/PHYs and a rich set of peripherals.

The AC498 DSP is an add-on DSP to the AC494E based on the AC49x DSP core. Each device provides extra 4 VoIP channels to the AC494E. Up to three AC498 devices can be concurrent with the AC494E providing a configurable VoIP density between 4 to 16 channels.

The AC494E and the AC498 are based on VoIPerfectHD™ architecture (High Definition VoIP) support, AudioCodes' underlying, best-of-breed, core media gateway technology for its product portfolio. Utilizing its highly integrated structure, requiring only aminimal number of peripheral devices, time-to-market The AC494E and the AC498 are based on VoIPerfectHD™ architecture (High Definition VoIP) support, AudioCodes' underlying, best-of-breed, core media gateway technology for its product portfolio. Utilizing its highly integrated structure, requiring only aminimal number of peripheral devices, time-to-market

Device	CPU Clock	DSP Clock	USB	Max VoIP Channels	Eth. Ports
AC494E	300 Mhz	150 Mhz	-	4	2
AC498	-	150 Mhz	-	4	-

Channel Density
- Up to 16 FXS lines

Hardware
- CPU – MIPS32 4KEC, 300 Mhz
- DSP – AC49x, 150 Mhz
- 2 Integrated 10/100/1000 Base-T MAC/PHY ports
- Integrated 3-port layer-2 Ethernet Switch
- Integrated SSP (SPI/I2C) Controller

Telephony Signaling
- DTMF – Detection and Generation, TIA464B
- Caller ID – Telcordia/ETSI/NTT Type I/II (on hook/off hook)
- Programmable Call-Progress Tones

Packetization
- RTP/RTCP Packetization – RFC 3550, 3551, 2198
- DTMF Relay - RFC 2833, RFC 4733

Security
- SRTP (Secured RTP) per RFC 3711, 128 bit AES, HMAC SHA1

Telephony Support
- FXS

Media Processing
- Voice Coders - G.711, G.723.1, G.729A/B, G.726, GSM FR
- Wideband Coders – G.722, G.722.2 (WB-AMR)
- Fax – T.38 Relay, Bypass , T.38 over RTP
- Echo Cancelation – G.168-2004 compliant, 64 msec tail length
- Silence Suppression – VAD, CNG
- Adaptive Jitter Buffer 300 msec

Telephony Features
- N-Way conferencing
- Call Forward
- Call Hold
- Call Transfer
- Call Waiting

Data Protocols
- IPv4, TCP, UDP, ICMP, ARP
- PPPoE
- IEEE 802.1p/q QoS (VLAN tagging)
- Layer 2 switching
- DHCP Client/Server – RFC 2132

VoIP Signaling Protocols
- SIP - RFC 3261, 3262, 3263, 3264, 2327

Configuration/Management
- Embedded Web
- TFTP
- FTP
- Telnet
- Filesystem support

Operating System
- Linux Kernel
- Board Support Package

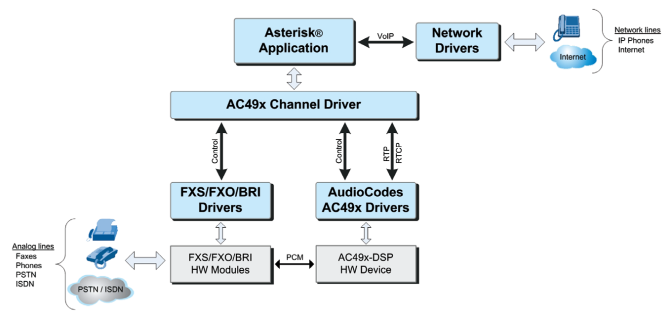
Toolkit Contents

- Reference Design & Development Platform based on AC494E-SoC and AC498-DSP (up to 16 VoIP channels)
- FXS modules
- Linux Operating System
- Free Development Environment (Linux Tool-chain)
- Ready-to-use VoIP Gateway SIP Application (open source code)

RISC Free IP-PBX Toolkit

The RISC Free IP-PBX Toolkit enables developers to use the advantages of the embedded Asterisk IP-PBX application while offloading the CPU from the voice processing heavy tasks, and using AudioCodes superior voice quality and enhanced VoIP features.

The RISC Free IP-PBX Toolkit is a combined hardware and software reference design for IP-PBX with up to 16 analog channels support and up to 32 IP Phones support, providing a complete, cost-effective solution for various applications. Utilizing AudioCodes field-proven VoIPerfectHD™ architecture (High Definition voice support) and the embedded Asterisk IP-PBX application, the AC494E System on Chip (SoC) and the AC498 DSP, allows this reference design to offer OEMs and ODMs a cost effective and quick time to market solution for the rapidly growing SOHO and SMB VoIP markets.

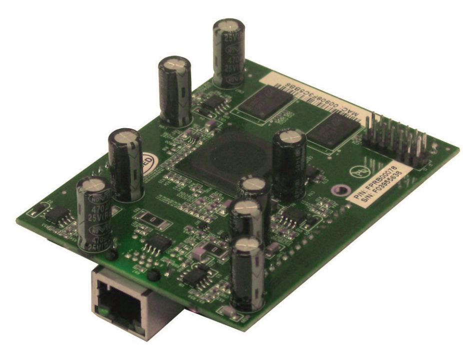


Asterisk AC49x Channel Driver Architecture

Gladiolus Software Development Kit (SDK)

The Gladiolus SDK is a Linux-based, Software Development Kit (SDK) for the AudioCodes' AC501x DSP family. The Gladiolus SDK provides the infrastructure and medium density (up to 28 LBR channels with AC494E SOC) sample applications. The Gladiolus SDK can be ported to any host platform. The Gladiolus SDK instructs users how to integrate the AC501x with any host platform and then develop their own VoIP application.

The Gladiolus can be provided with AC494E SMB board and a SLIC card with two FXS interfaces, as a complete development platform. So, customers can write their application with it, and start their development. A reference demo application (Linux based) is provided with the SDK.



The Gladiolus Board

AC494E Family (Orchid) SoC

- High Performance VoIP System on Chip Family
- Suitable for IP-Phones, Gateways and IP-PBX solutions
- Gigabit Ethernet throughput
- Security accelerator
- Static packet filtering
- Enhanced integrated CPU and DSP
- VoIPerfectHD™ (High Definition Voice) Support
- Combined hardware and software reference designs and toolkits

The Orchid family is the next-generation AudioCodes System on Chip (SoC) devices that integrates Gigabit Ethernet, security capabilities and superior voice quality features, and it combines the AC494E, AC495E and the AC496D devices.

The Orchid devices are dual-core based on MIPS 24KEc RISC CPU at 300MHz and AC49x-DSP at 150MHz allowing ODMs and OEMs to develop IP Phones, Gateways and IP-PBX with enhanced data and voice performance. These devices are based on VoIPerfectHD™ architecture (High Definition voice support), AudioCodes' underlying, best-of-breed, core media gateway technology for all of its products.

Gigabit Ethernet Support

The Orchid devices integrates a gigabit Ethernet (10/100/1000 Mbps) switch and two Fast Ethernet (10/100 Mbps) transceivers, allowing for the optional development of traditional Fast Ethernet designs without the additional cost of adding external transceivers.

The two Ethernet MACs support a high-speed and flexible switching mechanism allowing connection to PCs and/or a local area network (LAN) via three Gigabit Ethernet ports. The Orchid devices also integrated with two 10/100 Ethernet PHYs, enabling developers to design a cost effective product when gigabit technology is not required.

Device	CPU Clock	DSP Clock	USB	Voice CODECs	Eth. Ports
AC494E	300 Mhz	150 Mhz	V	2	2
AC495E	125 Mhz	125 Mhz	-	2	2
AC496D*	300 Mhz	300 Mhz	V	-	2

* With ADPCM I/F, without 2 channel CODEC

Channel Density
- Up to 4 compressed VoIP channels

Hardware
- CPU – MIPS32 4KEC, 300 Mhz/125Mhz
- DSP – AC49x, 150 Mhz/125Mhz
- 2 Integrated 10/100 Base-T MAC/PHY ports
- 2 MII or RGMII Interfaces
- Integrated 3-port layer-2 Ethernet Switch
- 8x8 Keypad interface
- LCD Interface – Alphanumeric, Graphic, Rasterized
- Integrated SSP (SPI/I2C) Controller
- USB 1.1 Host/Device
- External Memory I/F – 16-bit DDR SDRAM and ASync EMIF
- Telephony Interface - Integrated dual channel 16 bit CODEC, sampling rate 8/16 KHz
- Five inputs to ADC, four outputs from DAC
- PCM Interface - 2.048 MHz A/µ-Law serial port
- Power Supply - +3.3V (+1.5V core via integrated voltage regulator)
- Power Consumption - 1.9 W
- Operational Case Temperature Range - 0ºC - 70ºC (commercial)
- Package - 376 pin BGA, 23.2x23.2 mm, 1.00 mm ball spacing

Telephony Signaling
- DTMF – TIA464B
- Caller ID - Telcordia/ETSI/NTT Type I/II (on hook / off hook)
- Programmable Call-Progress Tones

VoIP Signaling Protocols
- SIP - RFC 3261, 3262, 3263, 3264, 2327

Packetization
- RTP/RTCP Packetization – RFC 3550, 3551, 2198
- Caller ID - Telcordia/ETSI/NTT Type I/II (on hook / off hook)
- DTMF Relay - RFC 2833, RFC 4733

Security
- SRTP (Secured RTP) per RFC 3711, 128 bit AES, HMAC SHA1

Media Processing
- Voice Coders - G.711, G.723.1, G.729A/B, G.726, GSM FR, AMR, iLBC
- Wideband Coders – G.722, G.722.2 (WB-AMR)
- Echo Cancelation – G.168-2004 compliant, 64 msec tail length
- Acoustic Echo Canceler
- Silence Suppression – VAD, CNG
- Adaptive Jitter Buffer 300 msec

Telephony Features
- 3-Way conferencing (on each channel)
- Call Forward
- Call Hold
- Call Transfer
- Call Waiting

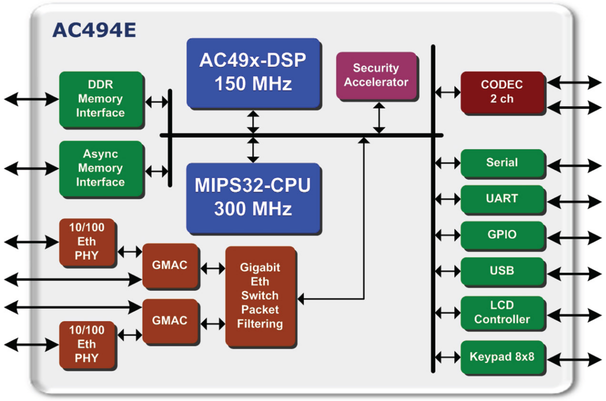
Configuration/Management
- Embedded Web
- TFTP
- FTP
- Telnet
- Filesystem Support

Operating System
- Linux Kernel
- Board Support Package

AC494E/AC495E

The AC494E/AC495E cocombines is a single chip all the IP-Phone components; it includes a MIPS 24KEc CPU, AC49x DSP and 3-port gigabit Ethernet Switch together with a set of peripherals such as internal CODECs and serial ports,on-chip peripherals include a 24-bit color LCD controller, 8 x 8 keypad interface, USB controller and two serial UARTs.

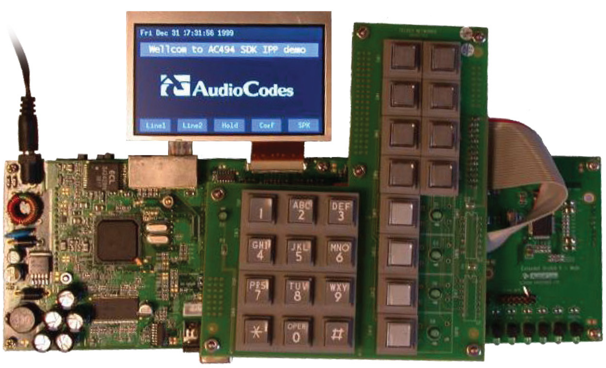
The AC494E/AC495E is integrated with PCM (TDM) interfaces allowing connecting it with analog devices such as FXS, FXO and BRI. The AC49x-DSP integrated inside the AC494E/AC495E can support up to 4 VoIP channels and together with the VLYNQ high speed serial interface extra AC498- DSP devices can be added with support up to 16 VoIP channels.



AC494E SoC Block Diagram

AC494E/AC495E IP Phone Reference Design

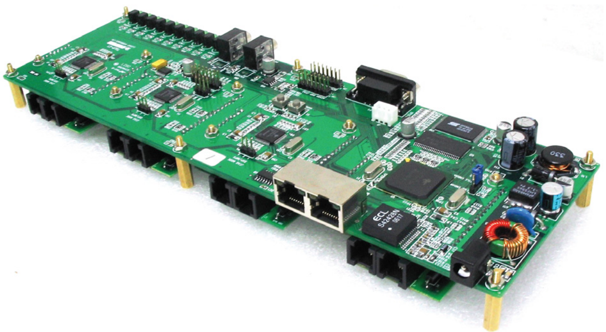
- 10/100/1000 Eth interfaces
- Security Accelerator Engine
- High resolution 24 bit LCD
- HD VoIP support
- Bluetooth and USB accessories support



AC494E IP Phone Toolkit

AC494E/AC495E GW/IPPBX Reference Design

- 10/100/1000 Mbytes Ethernet Interfaces
- Security Accelerator Engine
- Integrated 300 MHz CPU for IP-PBX applications
- USB support*
- Up to 16 FXS/FXO configurable ports support
- HD VoIP support
- Embedded Asterisk IP-PBX application is available



* AC494E only

Enterprise VoIP Gateway

AC 5042 (Sunflower) System on Chip

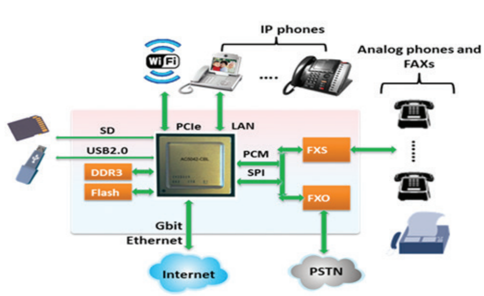
- Powerful multi purpose dual core SoC
- Ideal solution for IPPBX & VoIP Gateway applications scalability and reliability with cost-effective deployment
- VoIPerfectHD™ support
- Dynamically allocated Up to 32 VoIP channels

The AC5042 AudioCodes System on Chip (SoC) device integrates Gigabit Ethernet and GMAC Switch, Video coprocessor, PCI Express I/F, security capabilities and superior voice quality features. The AC5042 device is dual-core based on ARM Cortex A8™ at 1GHz and AC50x-DSP core at 750MHz allowing ODMs and OEMs to developGateways and IP-PBX with enhanced data and voice performance. The Sunflower is based on VoIPerfectHD™ architecture (High Definition voice support), AudioCodes' underlying, best-of-breed, core media gateway technology for all of its products.

AC5042 Software Development Kit (SDK) and Reference Design

The AC5042 SDK couples the hardware with a complete Linux software suite supporting voice and network protocols, call features and management applications. This software is available in open source code allowing customers to easily differentiate themselves. The SDK implements the SIP control protocol and Asterisk based IPPBX application.

The reference design can be used as an infrastructure for IP-PBX and Gateways products. The scalability and flexibility of this reference design and its open source code software allows customers to rapidly customize and actively deploy a differentiated VoIP gateway product to any market.



Sunflower IPPBX Sample Application

Hardware
- CPU – ARM Cortex A8™ at 1GHz
- DSP – AC50x- core at 750MHz
- Enhanced HD Coprocessor
- 2 x Gigabit EMAC, GMII
- PCI Express I/F
- USB 2.0 OTG I/F
- 6xUART, 3xI2C,4xSPI,3x MMC/SD/SDIO

Telephony Signaling
- DTMF – Detection and Generation, TIA464B
- Caller ID – Telcordia/ETSI/NTT Type I/II (on hook / off hook)
- Programmable Call-Progress Tones

Packetization
- RTP/RTCP Packetization – RFC 3550, 3551, 2198
- DTMF Relay - RFC 2833, RFC 4733

Security
- SRTP (Secured RTP) per RFC 3711, 128 bit AES, HMAC SHA1

Telephony Support
- FXS supported

Media Processing
- Up to 32 VoIP channels
- Voice Coders - G.711, G.723.1, G.729A/B
- Wideband Coders – G.722
- Fax – T.38 Relay, Bypass, T.38 over RTP
- Echo Cancellation – G.168 compliant, up to 128 msec tail length
- Silence Suppression – VAD, CNG
- Adaptive Jitter Buffer 300 msec

Telephony Features
- 3-way conferencing (on each channel)
- Call Forward
- Call Hold
- Call Transfer
- Call Waiting

Data Protocols
- IP, TCP, UDP, ICMP, ARP
- PPPoE
- Layer-2 switching
- WAN to LAN Layer-3 routing
- DHCP Client/Server – RFC 2132
- NAT – RFC3022, Application Layer Gateway (ALG)
- IEEE 802.1p/q QoS (VLAN tagging)

VoIP Signaling Protocols
- SIP - RFC 3261, 3262, 3263, 3264, 2327

Configuration/Management
- Embedded Web
- TFTP
- FTP
- Telnet
- Filesystem support

Operating System
- Linux Kernel

** AC5142 - Video Enabling - Roadmap*

ATA and CPE VoIP Gateway

AC48x VoIP DSP

- Feature rich DSP for low density gateways with toll quality voice compression
- Independent multi-channel operation
- Pin compatible devices

The AC48x is an ideal low cost voice processing engine for a variety of Voice over IP, Voice over DSL and other voice over packet client applications. The AC48x VoP processor is from one up to four ports voice over packet processor that combines toll quality low bit rate voice compression, T.38 compliant fax relay and other voice band processing functions. Field-proven, feature-rich software enables the rapid development and fast time-to-market of the complete solution.

The AC48x is supported by the VoicePacketizerTM software stack, which enables the processor to create a VoIP-compliant media stream as part of a client entity. VoicePacketizerTM is an ANSI-C operating system, independent software stack that supports the RTP/RTCP protocol. The software stack also provides a simple API for initialization and configuration of the AC48x and for run-time call control.

Channel Density
- AC483 4 channels Single LBR

Data Functions
- Voice/Fax/Data - Automatic Detection and Switching
- Fax support – T.38 compliant G3 Fax Relay, 2.4-33.6 kbps or PCM bypass
- Modem support – Automatic switch to PCM for up to V.92 rates

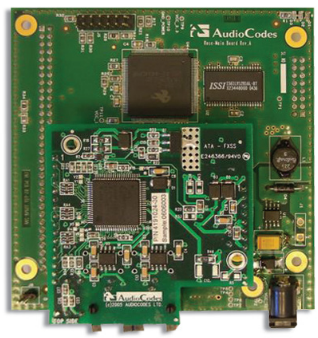
Signaling
- Inband Signaling
  - DTMF TIA 464B
  - MF R1, R2
- Detection and Generation - User Defined Call Progress Tones
- Out of band signaling – CAS ABCD (From Standard Framers)
- Caller ID Detection and Generation
  - Telecordia (Bellcore) On Hook / Off Hook (Type 1 & 2)
  - ETSI Onhook and Offhook Service (Type 1 & 2)
  - NTT Number Display (Type 1), Name Display

Voice Functions
- Voice Coders
  - G.711 PCM (A / u-law) at 64 kbps
  - G.726 ADPCM at 16-40 kbps
  - G.727 E-ADPCM at 16-40 kbps
  - G.729AB CS-ACELP at 8kbps
  - G.723.1 MP-MLQ at 6.3 kbps
  - G.723.1 ACELP at 5.3 kbps
  - GSM 6.10 Full Rate at 13.2 kbps
  - G.722
- Echo Cancellation - G.168-2004 compliant, up to 64msec tail length
- 3-Way Conference - Conferencing of 3 participants from PSTN or IP
- Quality Enhancement - Voice Activity Detection (VAD)
  - Comfort Noise Generation (CNG)
  - Packet Loss Concealment (PLC)
  - Adaptive Jitter Buffer (up to 300 msec)
- IPmedia™ Features
  - Energy and Answer detectors
  - Packet to packet Transcoding

Packetization
- RTP/RTCP – RFC 3550, 3551, 2198
- DTMF Relay - RFC 2833, RFC 4733

ATA and CPE VoIP Gateway

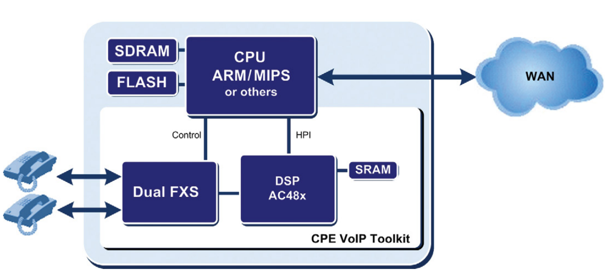
CPE VoIP Toolkit



- Combined hardware & software reference design for up to 4 compressed VoIP channels
- Ideal add-on VoIP solution for broadband access products such as FTTx, Cable, xDSL
- Superior voice quality at a competitive price
- Minimizing risks and allowing quick time to market for ODMs and OEMs rolling out VoIP-based services

The AC48x CPE VoIP Toolkit is a combined hardware and software VoIP reference design for up to 4 VoIP channels. This reference design can be used as an add-on VoIP module for CPE broadband products. Utilizing AudioCodes field-proven VoIPerfect™ software, the AC48x Voice over Packet Processor (VoPP) and the VoicePacketizer™ Voice over Streaming Protocol Stack, enables this reference design to offer manufacturers of broadband access products a cost-effective, high quality and market ready solution for the rapidly growing residential VoIP market. The AC48x CPE VoIP Toolkit hardware is based on the AC48x Voice over Packet Processor (VoPP) family offering a cost-effective Bill of Materials (BOM) that fully addresses the needs of aggressive market cost demands The AC48x CPE VoIP Toolkit couples the hardware with complete Linux software available in source code.

Device	DSP Clock	Voice Ports
AC483	100 Mhz	up to4



AC48x CPE VoIP Toolkit Hardware Design

Channel Density
- Up to 4 compressed VoIP channels

Media Processing
- Voice Coders - G.711, G.723.1, G.729A/B, G.726, G.722
- Fax – T.38 Relay, Bypass
- Echo Cancelation – G.168-2004 compliant, 32 msec tail length
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)

VoIP Signaling Protocols
- SIP - RFC 3261, 3262, 3263, 3264, 2327

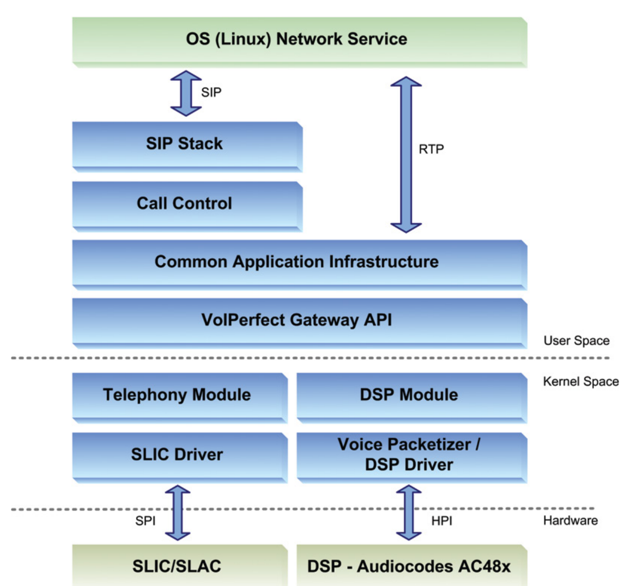
Packetization
- RTP/RTCP Packetization – RFC 3550, 3551
- DTMF Relay - RFC 2833, RFC 4733

Telephony Signaling
- DTMF – Detection and Generation, TIA464B
- Caller ID – Telcordia/ETSI/NTT Type I/II (on hook/off hook)
- Programmable Call-Progress Tones

Telephony Support
- FXS and FXO supported

Telephony Features
- 3-Way conferencing (on both channels)
- Call Hold
- Call Transfer
- Call Waiting

Operating System
- Embedded Linux

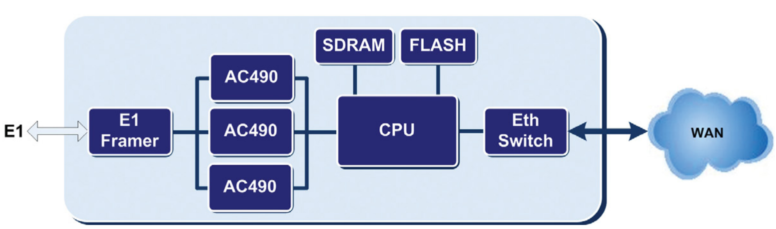


AC48x CPE VoIP Toolkit Software Architecture

AC490 – VoIP DSP

- Feature rich DSP for medium density gateways with superior voice quality
- Variety of vocoders for various VoIP applications
- No external memory required

The AC490xx family of DSPs is an ideal solution for Small Medium Business (SMB) VoIP gateways and IP-PBXs that have an existing CPU or network processor. Field-proven, feature rich software and a full reference design, enable the rapid development and quick time to market of the complete solution. The AC490xx is based on VoIPerfect™ architecture, AudioCodes' underlying, best-of-breed core media gateway technology.



E1 to VoIP sample application using the AC490

Channel Density
- AC49008 – Up to 8 compressed channels
- AC49012 – Up to 12 compressed channels

Data Functions
- Voice/Fax/Data - Automatic Detection and Switching
- Fax support - T.38 compliant G3 Fax Relay, 2.4-14.4 kbps or PCM bypass
- Modem support – Automatic switch to PCM for up to V.92 rates

Signaling
- Inband Signaling - DTMF TIA 464B
  - MF R1,R2
- Detection and Generation - User Defined Call Progress Tones
- Out of band signaling – CAS ABCD (From Standard Framers)
- Caller ID Detection and Generation
  - Telecordia (Bellcore) On Hook/Off Hook (Type 1 & 2)
  - ETSI On Hook and Off Hook Service (Type 1 & 2)
  - NTT (Type 1 and 2)

Security
- SRTP (Secured RTP) per RFC 3711, 128 bit AES, HMAC SHA1

Host Services
- HDLC Framing
  - For CCS signaling (ISDN, V5.2)
  - MTP2 per Q.703

Software Functionalities
- System Features
  - Selection of vocoders on the fly for each channel
  - Dynamic packet size programming

Voice Functions
- Voice Coders
  - G.711 PCM (A/u-law) at 64 kbps
  - G.726 ADPCM at 16-40 kbps
  - G.727 E-ADPCM at 16-40 kbps
  - G.729AB CS-ACELP at 8kbps
  - G.723.1 MP-MLQ at 6.3 kbps
  - G.723.1 ACELP at 5.3 kbps
  - GSM 6.10 Full Rate at 13.2 kbps
  - GSM EFR
  - AMR at 4.75-12.2 kbps
  - iLBC at 13.33 & 15.2 kbps
- Echo Cancelation - G.168-2004 compliant, 128 msec tail length
- 3-Way Conference - Conferencing of 3 participants from PSTN or IP
- Quality Enhancement
  - Voice Activity Detection (VAD)
  - Comfort Noise Generation (CNG)
  - Packet Loss Concealment (PLC)
  - Adaptive Jitter Buffer (up to 320 msec)
- IPmedia™ Features
  - Automatic Gain Control
  - Packet-to-packet Transcoding

Packetization
- RTP/RTCP – RFC 3550, 3551, 2198
- DTMF Relay - RFC 2833, RFC 4733

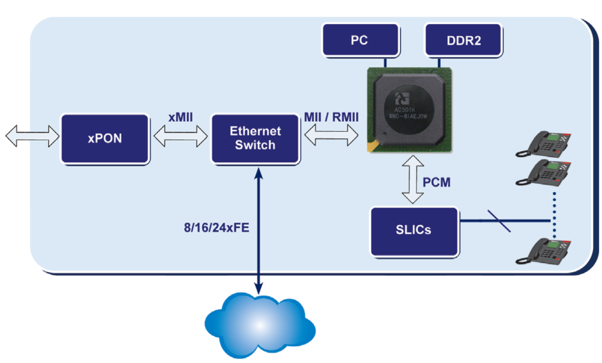
SMB VoIP Gateway and IP-PBX

AC501x - VoIP DSP

- Feature rich DSP with superior voice quality
- Variety of vocoders for various VoIP applications
- Offers excellent VoIP add-ons for designs with Stand-alone CPU
- Off loading the CPU with Ethernet interface
- Pin Compatible Devices

The AC501x family of DSPs is an ideal solution for GPON MDUs, Small Medium Business (SMB) VoIP gateways and IP-PBXs that have a standalone CPU or network processor. Field-proven, feature rich software, and advanced HW architecture that contains Ethernet interface for off loading the CPU, enable the rapid development and fast time to market of the complete solution. The AC501x is based on VoIPerfect™ architecture, AudioCodes' underlying best-of-breed core media gateway technology.

Device	Compressed Channel	HPI	RMII/MII	Ext. Memory
AC5011	8	V	V	DDR2
AC5012	16	V	V	DDR2
AC5015 IND	16	V	V	DDR2
AC5014 IND	22	V	V	DDR2
AC5014	24	V	V	DDR2



PON MDU sample application using AC501x

Channel Density
- Up to 8/16/24 compressed channels

Data Functions
- Voice/Fax/Data - Automatic Detection and Switching
- Fax support – T.38 compliant G3 Fax Relay, 2.4-33.6 kbps or PCM bypass
- Modem support - Automatic switch to PCM for up to V.92 rates

Signaling
- Inband Signaling
  - DTMF TIA 464B
  - MF R1,R2
- Detection and Generation - User Defined Call Progress Tones
- Out of band signaling – CAS ABCD (From Standard Framers)
- Caller ID Detection and Generation
  - Telecordia (Bellcore) On Hook/Off Hook (Type 1 & 2)
  - ETSI On Hook and Off Hook Service (Type 1 & 2)
  - NTT (Type 1 and 2)

Security
- SRTP (Secured RTP) per RFC 3711, 128 bit AES, HMAC SHA1

Voice Functions
- Major Voice Coders Supported
  - G.711 PCM (A / u-law) at 64 kbps
  - G.726 ADPCM at 32 kbps
  - G.729AB CS-ACELP at 8kbps
  - G.723.1 MP-MLQ at 6.3 kbps
  - G.723.1 ACELP at 5.3 kbps
  - G.722
  - G.722.2 WB-AMR
  - AMR
  - EVRC
  - GSM FR 6.10
  - Silk
  - Opus
- Echo Cancellation - G.168-2004 compliant, 128 msec tail length
- N Way Conference - Conferencing of N participants from PSNT or IP
- Quality Enhancement
  - Voice Activity Detection (VAD)
  - Comfort Noise Generation (CNG)
  - Packet Loss Concealment (PLC)
  - Adaptive Jitter Buffer (up to 300 msec)
- IPmedia™ Features*
  - Automatic Gain Control
  - Answer detector
  - Packet to packet Transcoding

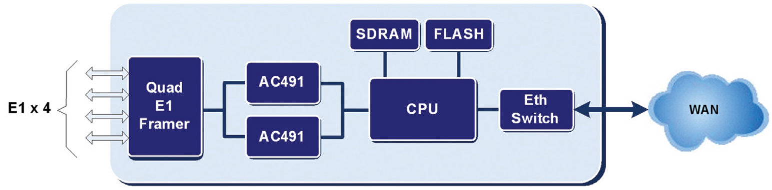
Packetization
- RTP/RTCP – RFC 3550, 3551, 2198
- DTMF Relay - RFC 4733, 2833

Enterprise VoIP Gateway

AC491 / AC491L

- Feature rich DSP for enterprise gateways with superior voice quality
- Variety of vocoders for various VoIP applications
- No external memory required

The AC491/AC491L family of DSPs is an ideal solution for enterprise VoIP gateways - digital and analog. Featuring high channel density (up to 192 channels of non-compressed voice, or 96 channels of compressed voice), low power consumption (less than 6mW/ch for non-compressed voice), and low footprint (16x16 mm) with no external memory, the AC491/AC491L provides an ideal building block for enterprise gateways and IP-PBXs. The AC491/AC491L is based on VoIPerfect™ architecture, AudioCodes' underlying, best-of-breed core media gateway technology.



E1 x 4 to VoIP sample application using the AC491

Channel Density
- AC491L32 – Up to 32 compressed channels
- AC491064 – Up to 64 compressed channels
- AC491096 – Up to 96 compressed channels

Data Functions
- Voice/Fax/Data - Automatic Detection and Switching
- Fax support – T.38 compliant G3 Fax Relay, 2.4-14.4 kbps or PCM bypass
- Modem support – Automatic switch to PCM for up to V.92 rates

Signaling
- Inband Signaling
  - DTMF TIA 464B
  - MF R1,R2
- Detection and Generation - User Defined Call Progress Tones
- Out of band signaling – CAS ABCD (from Standard Framers)
- Caller ID Detection and Generation
  - Telecordia (Bellcore) On
  - On Hook/Off Hook (Type 1 & 2)
  - ETSI On Hook and Off Hook Service (Type 1 & 2)
  - NTT (Type 1 and 2)

Security
- SRTP (Secured RTP) per RFC 3711, 128 bit AES, HMAC SHA1

Host Services
- HDLC Framing
  - For CCS signaling (ISDN, V5.2)
  - MTP2 per Q.703

Software Functionalities
- System Features
  - Selection of vocoders on the fly for each channel
  - Dynamic packet size programming

Voice Functions
- Voice Coders
  - G.711 PCM (A / u-law) at 64 kbps
  - G.726 ADPCM at 16-40 kbps
  - G.727 E-ADPCM at 16-40 kbps
  - G.729AB CS-ACELP at 8kbps
  - G.723.1 MP-MLQ at 6.3 kbps
  - G.723.1 ACELP at 5.3 kbps
  - GSM 6.10 Full Rate at 13.2 kbps
  - G.729E at 11.8 kbps
  - AMR at 4.75-12.2 kbps
  - iLBC at 13.33 & 15.2 kbps
  - EVRC Up to 8.55 kbps
  - G.722
  - G.722.2 WB-AMR
- Echo Cancelation - G.168-2004 compliant, 128 msec tail length
- 3-Way Conference - Conferencing of 3 participants from PSTN or IP
- Quality Enhancement
  - Voice Activity Detection (VAD)
  - Comfort Noise Generation (CNG)
  - Packet Loss Concealment (PLC)
  - Adaptive Jitter Buffer (up to 300 msec)
- IPmedia™ Features
  - Automatic Gain Control
  - Energy and Answer detectors
  - Packet-to-packet Transcoding

Packetization
- RTP/RTCP - RFC 3550, 3551, 2198
- DTMF Relay - RFC 2833, RFC 4733

Enterprise VoIP Gateway

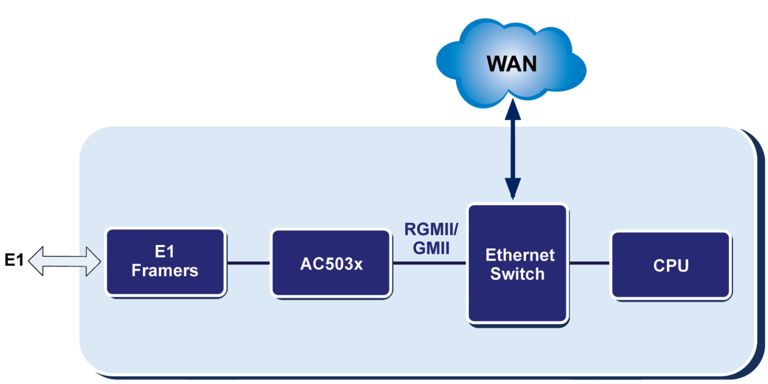
AC503x - VoIP DSP

- Feature rich DSP for enterprise gateways with superior voice quality
- Variety of vocoders for various VoIP applications
- Offer excellent VoIP add-ons for designs with Stand-alone CPU
- Off loading the CPU with Ethernet interface
- Pin compatible devices

The AC503x family of DSPs is an ideal solution for Enterprise VoIP gateways and IP-PBXs that have an stand alone CPU or network processor. Field-proven, feature rich software, and advanced HW architecture that contains Ethernet interface for off loading the CPU, enable the rapid development and fast time to market of the complete solution. The AC503x is based on VoIPerfect™ architecture, AudioCodes' underlying, best-of-breed core media gateway technology.

Device	Voice Channels**
*AC5033	160
AC5039 IND	204
AC5039	234

* Please contact AudioCodes representative for specific software availability and channel density support
** Dynmically allocated VoIP channels



Data Functions
- Voice/Fax/Data - Automatic Detection and Switching
- Fax support – T.38 compliant G3 Fax Relay, 2.4-33.6 kbps or PCM bypass
- Modem support - Automatic switch to PCM for up to V.92 rates

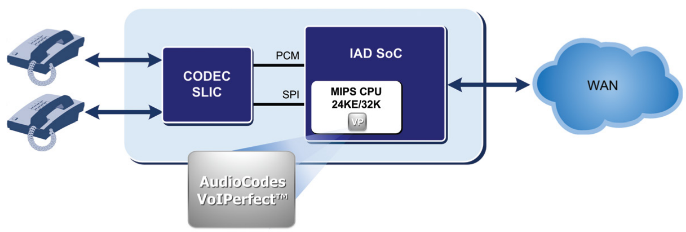
Signaling
- Inband Signaling
  - DTMF TIA 464B
  - MF R1,R2
- Detection and Generation - User Defined Call Progress Tones
- Out of band signaling – CAS ABCD (From Standard Framers)
- Caller ID Detection and Generation
  - Telecordia (Bellcore) On Hook/Off Hook (Type 1 & 2)
  - ETSI On Hook and Off Hook Service (Type 1 & 2)
  - NTT (Type 1 and 2)

Security
- SRTP (Secured RTP) per RFC 3711, 128 bit AES, HMAC SHA1

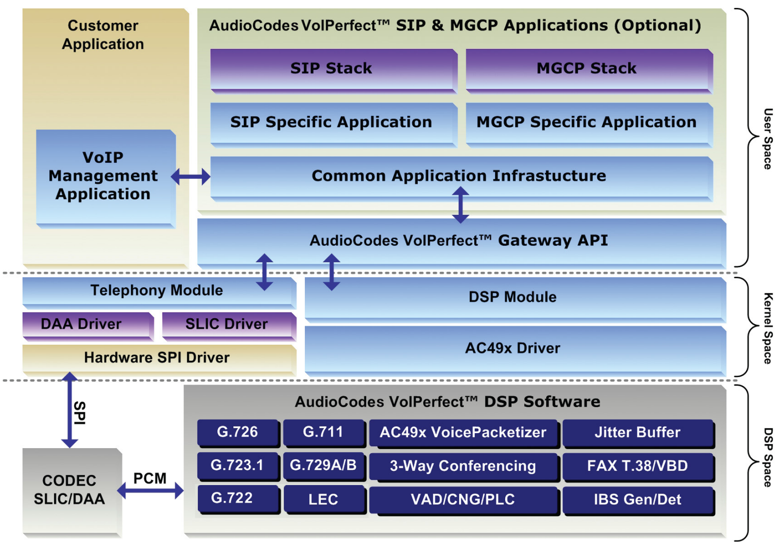
Voice Functions
- Major Voice Coders Supported
  - G.711 PCM (A / u-law) at 64 kbps
  - G.726 ADPCM at 16-40 kbps
  - G.729AB CS-ACELP at 8kbps
  - G.723.1 MP-MLQ at 6.3 kbps
  - G.723.1 ACELP at 5.3 kbps
  - GSM 6.10 Full Rate at 13.2 kbps
  - AMR at 4.75-12.2 kbps
  - iLBC at 13.33 & 15.2 kbps
  - EVRC Up to 8.55 kbps
  - G.722
  - G.722.2 WB-AMR
  - Silk
  - Opus
- Echo Cancellation - G.168-2004 compliant, 128 msec tail length
- N Way Conference - Conferencing of N participants from PSNT or IP
- Quality Enhancement
  - Voice Activity Detection (VAD)
  - Comfort Noise Generation (CNG)
  - Packet Loss Concealment (PLC)
  - Adaptive Jitter Buffer (up to 300 msec)
- IPmedia™ Features
  - Automatic Gain Control
  - Answer detector
  - Packet to packet Transcoding

Packetization
- RTP/RTCP – RFC 3550, 3551, 2198
- DTMF Relay - RFC 2833, 4733

VoIPerfect™ Soft DSP Solutions



- AudioCodes field-proven VoIPerfect DSP Software optimized for various host processors
- Featuring AudioCodes' certified and interoperable VoIPerfect™ SIP application
- Ideal VoIP solution for broadband CPEs such as EPON/GPON ONU, xDSL Residential Gateways and Cable Set-top box
- Complete VoIP application minimizes efforts and risks to ODMs and OEMs rolling out VoIP products



VoIPerfect™ Software Architecture

Channel Density
- Up to 4 compressed channels

Media Processing
- Major Voice Coders Supported: G.711, G.729A/B, G.723.1, G.726
- Wideband Coder: G.722
- 3-Way Conference: Up to 4 independent 3-Way conferences
- Fax Support: T.38 compliant G3 Fax Relay, Bypass
- Echo Cancelation: G.168-2004 compliant, 32 msec tail length
- Quality Enhancement:
  - VAD - Voice Activity Detection
  - CNG - Comfort Noise Generation
  - PLC - Packet Loss Concealment
  - Adaptive Jitter Buffer, up to 300 msec with re-ordering
  - Configurable Digital Gain

Telephony Signaling
- Programmable Call-Progress Tones Detection and Generation
- DTMF Detection and Generation
- Caller ID Detection and Generation (On Hook & Off Hook):
  - Telcordia (Type 1 and 2)
  - ETSI (ETS 300 659-1 and 2)
  - NTT (Type 1 and 2)

Packetization
- RTP/RTCP Packetization: RFC 3550, 3551
- RTP Redundancy: RFC 2198
- DTMF Relay: RFC 2833, RFC 4733
- RTCP XR: RFC 3611 (optional)

VoIP Signaling Protocols
- SIP: RFC 3261, 3262, 3264, 2327, 2976

Telephony Support
- FXS analog interface

Telephony Features
- 3-Way conference
- Call Forward
- Call Hold
- Call Transfer
- Call Waiting

Operating System
- Embedded Linux, Linux SMTC

Notes:

AudioCodes HD VoIP

HDVoIP refers to the use of wideband technology, providing deeper clarity and a better audio experience in VoIP Communications. The traditional Public Switch Telephony Network (PSTN) is limited to 300-3400 Hz for narrowband voice. Voice signals are sampled at a rate of 8 kHz, causing limitations in communication quality and comprehension. In HD VoIP, wideband telephony refers to transmitting voice signals with bandwidths ranging between 50-7000 Hz and a sampling rate of 16 kHz. This effectively doubles the narrowband voice signal bandwidth and offers the caller "true voice" conversation. Compared to narrowband telephony, wideband technology establishes a sense of presence, resulting in a natural and comfortable conversation.

About AudioCodes

AudioCodes Ltd. (NasdaqGS: AUDC) designs, develops and sells advanced Voice over IP (VoIP) and converged VoIP and Data networking products and applications to Service Providers and Enterprises. AudioCodes is a VoIP technology market leader focused on converged VoIP & data communications and its products are deployed globally in Broadband, Mobile, Enterprise networks and Cable. The company provides a range of innovative, cost-effective products including Media Gateways, Multi-Service Business Routers, Session Border Controllers (SBC), Residential Gateways, IP Phones, Media Servers and Value Added Applications. AudioCodes' underlying technology, VoIPerfect HDTM, relies on AudioCodes' leadership in DSP, voice coding and voice processing technologies. AudioCodes High Definition (HD) VoIP technologies and products provide enhanced intelligibility and a better end user communication experience in Voice communications.

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