VOCAL

Analog Telephone Adapter Reference Design Kit

The ATA Reference Design Kits enable licensees to develop a variety of standards-based next generation product configurations with extensive features and world-wide configurability. VoCAL's highly optimized On-One™ DSP technology is used to reduce system cost by controlling all ATA operations and performing advanced signal processing on a single state-of-the-art DSP. These designs far surpass the competition in terms of cost advantage and time-to-market potential, and VoCAL's extensive customization support allows quick and easy implementation of value added features.

Configurations

Product configurations (please see diagrams on next page):

1 Analog Port (FXS), 1 Ethernet
2 Analog Ports (FXS), 1 Ethernet
2 Analog Ports (FXS), 2 Ethernet

The designs include PSTN lifeline support via a failover relay. A full FXO is supported with the addition of a codec and DAA hardware.

Features

Included are all the standard features expected of an ATA, plus innovative customer demanded premium features. With the patent-pending VoCAL VoATA control software, an end user can manage up to four VoIP service providers with a single ATA. User access may be automatic on a per-number basis, or can be as simple as using a standard long distance prefix such as "1010825…".

Compatibility and Interoperability

All VOCAL Software benefits from over 15 years of service history. Compatibility has been assured by many current licensees and extensive testing with popular industrial products such as Cisco, Quintum and Asterisk. Configurations for service provides such as Vonage, Free World Dialup (FWD) and many others are available. Individual user configuration can be completely managed from the web interface built into the VoATA software, or restricted as desired by the licensee.

Provisioning

The VoCAL VoATA software is designed to support many common provisioning requirements. All tone and ring cadences and frequencies are configurable to meet world-wide expectations. Individual advanced features may be configured based on the customer's service plan, and administrators may prepare configuration change files and firmware updates for automatic distribution to deployed ATA's.

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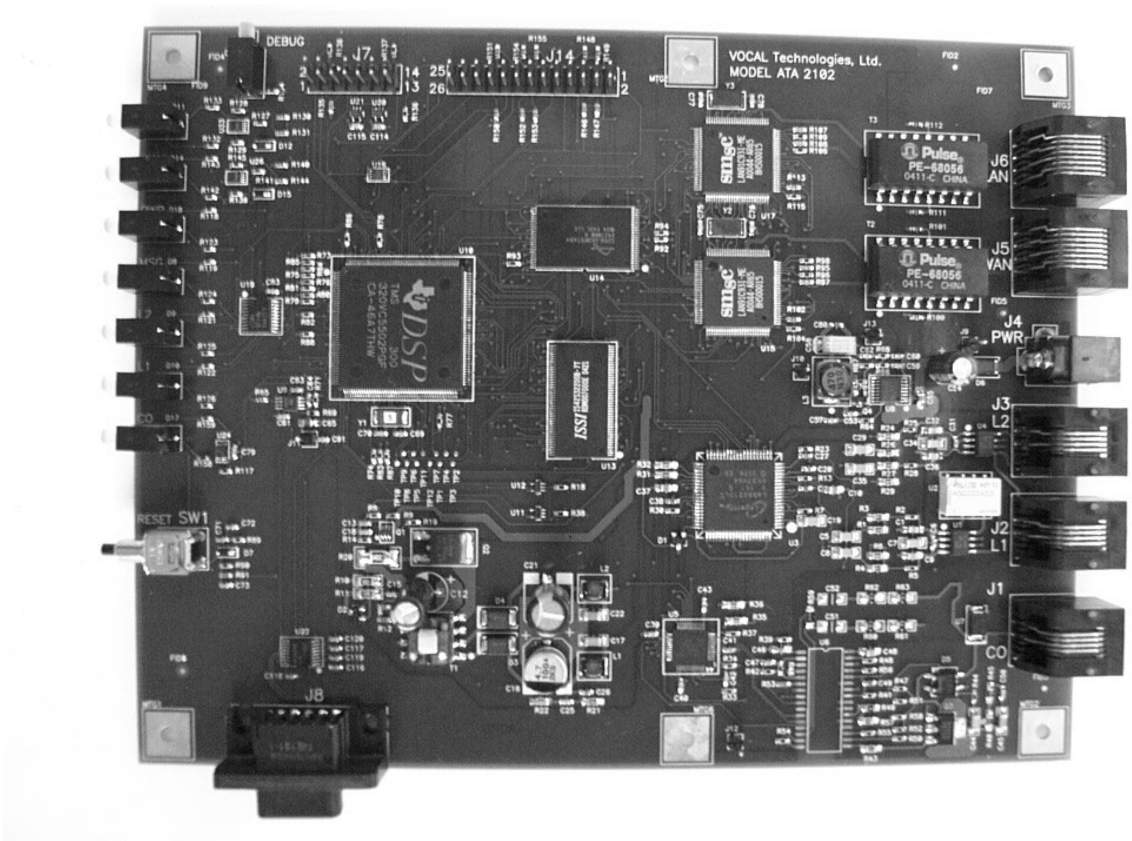

2x2+1 Development board

Unlike traditional architectures, the 2x2+1 ATA hardware and software designs fully utilize leading DSP resources and advanced patent-pending algorithms to eliminate the need for an additional RISC processor, reducing the cost of typical bill of material by as much as $8 per design. The solution is ideal for VoIP equipment suppliers desiring to add a low-cost, compact ATA solution to their portfolios.

The 2x2+1 ATA solution supports up to two telephone lines, two Ethernet ports and a PSTN life-line port for automated voice service switching in case of a power outage or network disconnection and is stackable to larger port configurations. End users can connect their homes and businesses to VoIP services using conventional wired and cordless phones without incurring additional monthly service fees or adding extra hardware. The 2x2+1 ATA solution initiates calls using the rapidly emerging IP telephony standard Session Initiation Protocol (SIPv2) and supports multiple advanced codecs to optimize performance. For VoIP service providers looking for a design that is

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easily configurable with their networks, the solution provides secure and sophisticated Web-based provisioning and firmware update technology.

The 2x2+1 ATA solution is reconfigurable in the field and designed to give OEMs and service providers the ability to bring their products to market quickly and deploy them with immediate interoperability in most broadband VoIP service provider networks. OEM customers can also take advantage of VOCAL's high-end customization capabilities, including reference design assistance and integration of value-added features.

Reference Design Kit

The ATA Reference Design Kits from VoCAL offer the following capabilities:

Technical Specifications

Voice-over-IP (VoIP) protocols
SIPv2 - Session Initiation Protocol (RFC 3261, 3262, 3263, 3264)
SDP - Session Description Protocol (RFC 2327)
RTP - Real-Time Protocol (RFC 1889, 1890)
RTCP - Real-Time Control Protocol (RFC 1889)
RFC 2833 X-NSE Tone Events for SIP/RTP
RFC 2833 AVT Tone Events for SIP/RTP

Fax Support
G.711 Fax Pass-Through
T.38 - Real-Time Fax Over IP
T.38 using UDP
T.38 using RTP

Network Protocols
IPv4 - Internet Protocol Version 4 (RFC 791)
TCP - Transmission Control Protocol (RFC 793)
UDP - User Datagram Protocol (RFC 768)
ICMP - Internet Control Message Protocol (RFC 792)
RARP - Reverse Address Resolution Protocol (RFC 903)
ARP - Address Resolution Protocol (RFC 826)
DNS- Domain Name Server
DHCP Client - Dynamic Host Control Protocol (RFC 2131)
NTP - Network Time Protocol (RFC 1305)
SNTP - Simple Network Time Protocol (RFC 2030)
STUN - Simple Traversal of UDP over NATs (RFC 3789)
HTTP - HyperText Transfer Protocol
TFTP - Trivial File Transfer Protocol (RFC 1350)

NAT/Firewall Support (2x2 product configuration)
Router
NAT Firewall
Gateway and DMZ Port Forwarding
DHCP Server
PPPoE - Point to Point Protocol over Ethernet (RFC 2516)

Voice Codecs
G.711 - Pulse Code Modulation
G.723.1 - 6.4 and 5.3 kbps ACELP/MP-MLQ
G.726 - 16, 24, 32 and 40 kbps ADPCM
G.728 - 16 kbps LD-CELP
G.729A - 8 kbps CS-ACELP
G.729B - Silence Detection/Comfort Noise Generation
iLBC - Internet Low Bitrate Codec

Telephony
Q.24 DTMF Generation with Zero Crossing Cutoff
Q.24 DTMF Detection exceeding Bellcore Specifications
Configurable Tone Generation for 4 Sets of Frequencies and 4 Sets of On/Off Cadence
Caller ID Type I (On-hook) Generation
Caller ID Type I (Off-hook) Generation
Caller ID Type I Detection
Caller ID Type II Detection

Line-echo cancellation
G.168 Line Echo Cancellation
Echo Canceller for Each Port
16 ms Echo Length
Nonlinear Echo Suppression (ERL greater than 28 dB for f = 300 to 3400 Hz)
Double-Talk Detection

Quality of Service
Layer 2 Class-of-Service (CoS) Tagging (802.1P)
Layer 2 (802.1Q VLAN)
Layer 3 Type-of-Service (ToS) Tagging (RFC 791/1349)
Layer 3 DIFFServ (RFC 2475)

Hardware Features

Data Network
Ethernet - 10baseT RJ-45
Ethernet - 10baseT/100baseTx RJ-45 (Optional)
Ethernet WAN Port RJ-45
Ethernet LAN Port RJ-45
Configurable MAC Address (IEEE 802.3)

Analog Telephone Ports
FXS Analog RJ-11 Ports (#)
Up to 5 REN
Configurable Terminating Impedance - 8 Settings
-48V Nominal Battery
85V Ringing
Sinusoidal or Trapezoidal Ringing

PSTN Port
FXO Analog RJ-11 Ports (#)
Lifeline Port (Processor Controlled Relay)
Dial Plan Accessible
Relay Deactivated on Power Fail

Indicators

Tri-Color LEDs (Red, Orange, Green)
POWER LED (Power, Registration, Use)
LAN LED (Activity and Link Fail)
MESSAGE LED (Message Waiting, Update Failure)
LINE LEDS (Line Status)

Reset Button
System Reset
Reset Configuration to Factory Defaults when Held

Power
12V DC Input
0.8 Amp (2 FXS with 4 REN)

Feature List

Phone User Services
Place/Cancel Outgoing Calls
Answer Incoming Calls
Full Duplex Audio
Flexible Dial Plan Support
Fax Tone Detection (Pass Through)
ITU T.38 Fax Support
PSTN Line Support (Dial and Backup)
PSTN Emergency 911/999

Call-in Profiles
IPBX (Internet Private Branch Exchange)
IHT (Internet Home Termination)
SIHT (Simple Internet Home Termination)
SIOT (Small Internet Office Termination)

Voice features
Voice Activity Detection (VAD)
Silence Suppression (DTX)
Comfort Noise Generation (CNG)
Packet Loss Concealment (PLC)
Dynamic Jitter Buffer (Adaptive)
Audio Codec Preferences
Dynamic Payload Negotiation
Codec Name Assignment
Adjustable Audio Frames per Packet

Telephony
CLASS Features
Call Waiting Enable/Disable
Caller ID Display Enable/Disable
Call Waiting Caller ID Enable/Disable
Blocked Call List for a Specified Number
Distinctive Ring for a Specified Number
Block/Unblock Caller ID
Block/Unblock Caller ID for One Call
Accept Priority Call of a Specified Number
Busy Number Redial
Call Return (Call the Last Caller)
Deactivate/Activate Call Waiting for Current Call
Call Forwarding
- Forward Priority Call of a Specified Number
- Forward on Busy
- Forward on No Answer
- Forward all Calls

Speed Dial (8 + 20 Numbers)
Block Anonymous Calls
Do Not Disturb
Call Transfer
3-way Conference Calling with Local Mixing
Redial
Call Hold
Call Waiting/Flash
Flash Hook Timer
Delay Disconnect
Hot Line and Warm Line Calling
Call Blocking with Toll Restriction

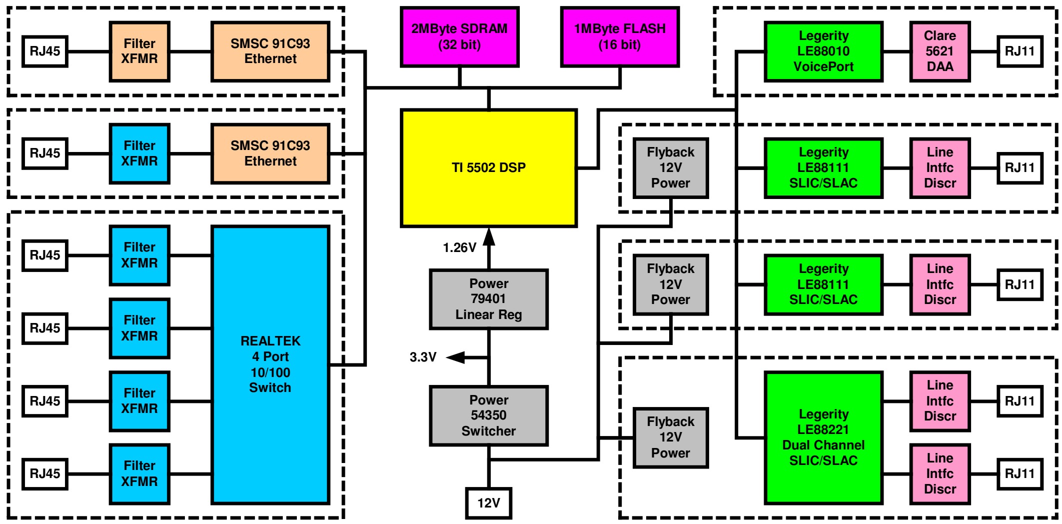
Caller ID Generation (Name & Number) - Bellcore, DTMF, ETSI
Call Waiting Caller ID with Name/ Number
Distinctive Ringing
Distinctive Call Waiting
MWI - Message Waiting Indicator Tone and Visual
VMWI Via FSK
Polarity Control

Call Progress Tones
Programmable Tone Generation Patterns
Four Tones, Four On/Off Time Pairs
- Initial Dial Tone
- Secondary Dial Tone
- Stuttered Dial Tone
- Message Wait Dial Tone
- Call Forward Dial Tone
- Pre-Ringback Dial Tone
- Ring Back Tone
- Call Waiting Tone
- Call Holding Tone
- Call Disconnect Tone
- Call Conference Tone
- Busy Tone
- Reorder Tone (Network/Fast Busy)
- Off Hook Warning Tone (Howler Tone)
- SIT Tones 1 to 4
- Prompt Tone
- Confirm Tone
- Input Error Tone
- Number Error Tone

Ringing Patterns
Programmable Ring Patterns
Four On/Off Time Pairs
- Default Ring
- Hold Rering
- Call Back Ring
- Call Back Ring Splash
- Call Forward Ring Splash
- Message Waiting Ring Splash
- 8 Distinctive Ring Patterns

Distinctive Call Waiting
Programmable Tone Generation
Four Tones, Four On/Off Time Pairs
8 Distinctive Call Waiting Patterns

LED Display Patterns
Tri-Color LEDs (Red, Orange, Green)
Programmable LED Patterns
Four On/Off Time Pairs
Configurable Display Priority
- Use and Waiting
- Ringing or Waiting
- PSTN in Use
- Use and Hold
- Line in Use
- Call Holding
- Do Not Disturb
- Call Forward
- Message Waiting



VOCAL ATA Design Kit Components

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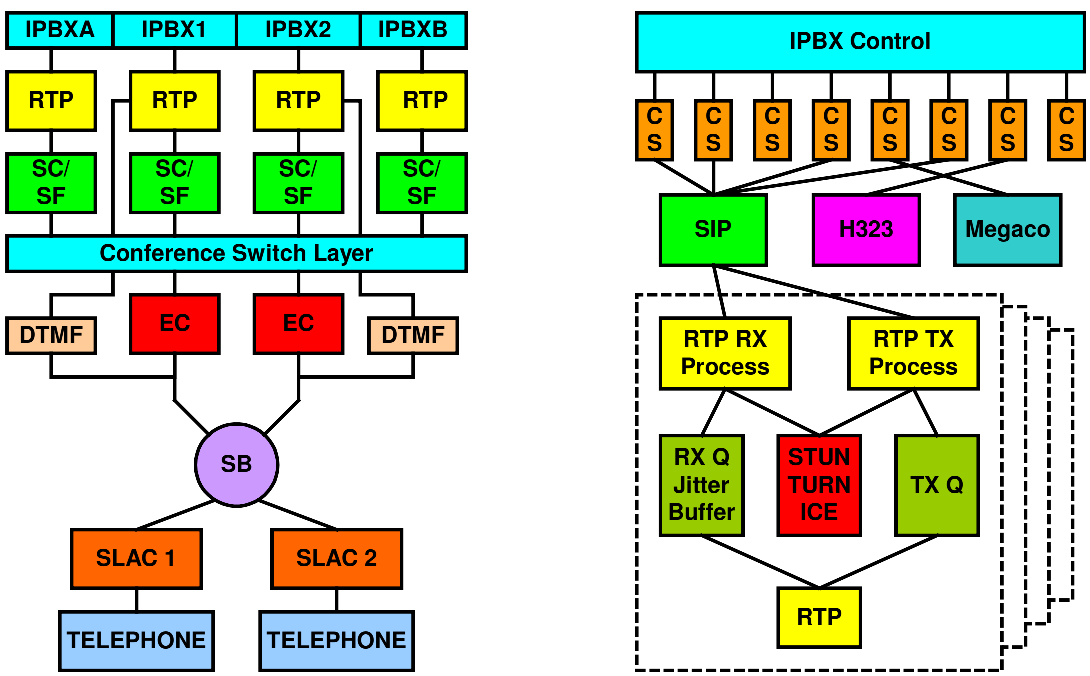

VOCAL IPBX Innards

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Voice-over-IP (VoIP) protocols
Power-on Auto Registration
Re-registration with SIP Proxy Server
SIP over UDP
SIP Authentication (HHP Digest with MD5)

Quality of Service
Port Priority for VoIP Packets from Application
High and Low Priority Transmit Queues for Interface

NAT/Firewall Support
Built-in Router
Automated NAT Traversal Without Manual Manipulation of Firewall/NAT
NAT Traversal for Private Networks with STUN (RFC3489)
NAT Firewall
Gateway and DMZ Port Forwarding
LAN Pass Through
Voice Priority
PPPoE – Point-to-Point Protocol over Ethernet (RFC2516)

Security
Provisioning/Configuration/Authentications
Password Protected Web based Administration
RC4 Encryption for TFTP Configuration Profiles
Authentication (DIGEST using MD5)

Remote Configuration/Maintenance
Web Configuration via Built-in Web Server
Configuration Update via TFTP or HTTP
Firmware Upgrade via TFTP or HTTP
SYSLOG Update/Upgrade Processing Notifications

Documentation
Administration Guide
Installation Guide
Configuration Guide