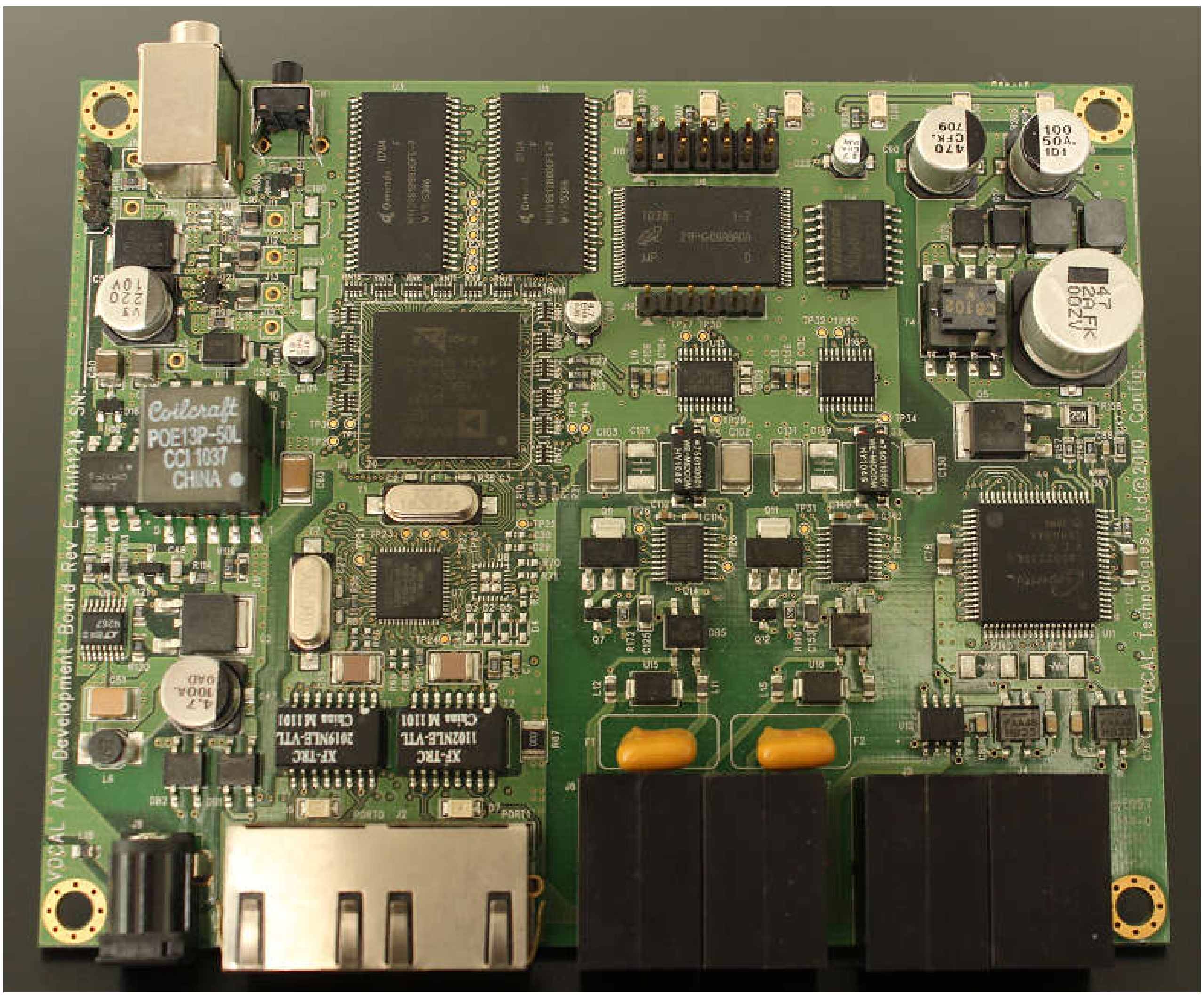
VOCAL

Analog Modem/Telephone Adapter

The AMA/ATA enable licensees to deploy a variety of standards-based next generation product configurations with extensive features and world-wide configurability. VOCAL AMAs include support of all common PSTN data and fax modems including V.90/V.34, Point of Sale (POS) and Alarm Panels, with T.38 support for Group 3 and Group3+ (V.34) fax machines. As an ATA, all common voice coders (ITU, GSM and IETF) are supported as well as specialty military voice coders (MELPe, TSVCIS, LPC10) in certain configurations. VoCAL's highly optimized On-One™ DSP technology is used to reduce system cost by controlling all AMA/ATA operations and performing advanced signal processing on a single state-of-the-art DSP. VOCAL's devices 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.



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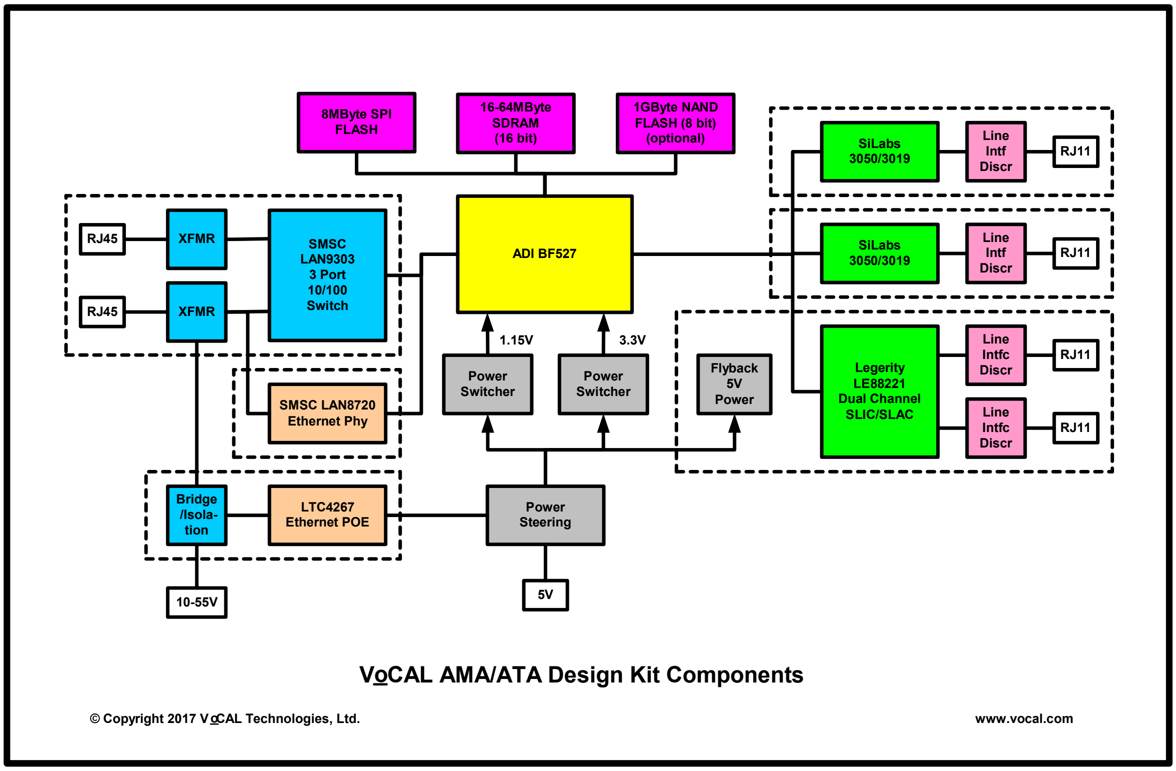
Product configurations (please see diagram below)

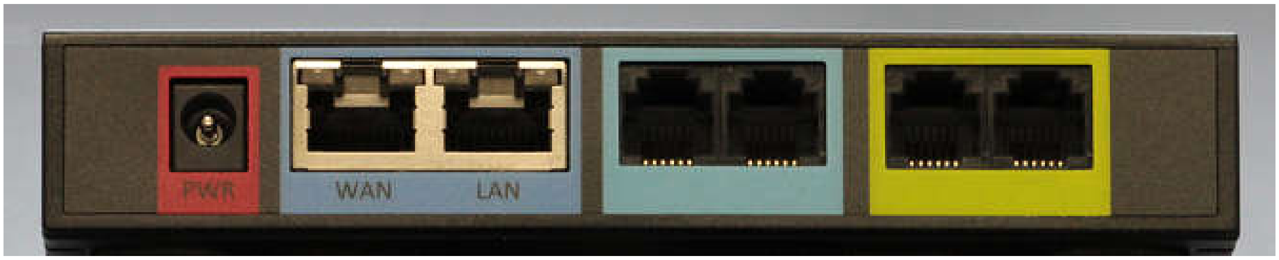
2 Telephone FXS Ports, 2 Telephone FXO Ports, 2 Ethernet (standard configuration)
2 Telephone FXS Ports, 2 Ethernet
2 Telephone FXS Ports, 1 Ethernet (standard configuration)
4 Telephone FXO Ports, 2 Ethernet
4 Telephone FXO Ports, 2 Ethernet (standard configuration)
1 Telephone FXS Port, 1 Ethernet
Optional POE
5 volt wall adapter standard
10-55 volt isolated power option



Standard configurations identify those directly available from VOCAL for unique applications as COTS hardware/software. These configurations are available with customized software for meeting the unique requirements of specific customers.

The VOCAL AMA/ATA Blackfin-based solutions support one to four telephone lines (FXS or FXO), and one or two Ethernet ports. A PSTN life-line port for automated voice service switching in case of a power outage or network disconnection is available for some 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 VOCAL ATA solution initiates calls using the 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 easily configurable with their networks, the solution provides secure and sophisticated Web-based provisioning and firmware update technology.

The VOCAL AMA/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. Customers can also take advantage of VOCAL's high-end customization capabilities, including integration of value-added features.



Connectors (left to right):

5.5/2.1mm barrel connector for 5VDC or optional 10-50V isolated power+
RJ45 Ethernet with optional POE
RJ45 Ethernet (optional)
RJ11 Telephone FXO Port 4
RJ11 Telephone FXO Port 3
RJ11 Telephone FXS Port 2
RJ11 Telephone FXS Port 1

+ - Optional isolated power supply also supports locking barrel connector. Requires resistor straps and eliminates Ethernet POE.

Telephone FXS/FXO Ports (RJ11 – 6 position, 2 contact):

Pin 3 – Tip
Pin 4 – Ring

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 "91716…" or "81716…".

Compatibility and Interoperability

All VoCAL Software benefits from over 20 years of service history. Compatibility has been assured by many current licensees and extensive testing with popular industrial products such as Cisco, Avaya, 3CX, Freeswitch and Asterisk. Configurations for service provides such as Vonage, Level3, Century Link, Verizon, AT&T, Cox, Twilio, Nextiva, Ring Central 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.

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Provisioning

The VoCAL VoATA software is designed to support many common provisioning requirements including TR-069 and SNMP. 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.

Modem Over IP (MoIP) or RAS over IP (RASoIP)

There are several ways to support data modems over an IP VoIP network. The most standardized way (V.150.1) is also the most complex and is used predominately by government communications networks for secure telephone communications. This is a modem relay approach whereby two end-point modems are supplemented with two more modems (one in the AMA and the other in a VoIP/PSTN gateway). VOCAL supports this method but actually recommends the methods that follow.

Many applications use a pure dial-up connection for a device to be in contact with a server. Either end-point may initiate the communications depending on the application. Server applications commonly use RFC 2217 for access of remote COM ports on terminal servers or deprecated Cisco modem server banks. VOCAL's AMA pushes the modem to the remote location and utilizes VOCAL's SAMS (SIP Analog Modem Server) for managing connections between remote devices and the applications servers. This works well for many PSTN replacement scenarios including for remote utility meter access and management of remotely controlled equipment (MV90, PrimeRead and Autosol).

Another class of applications requires access of either a public or private internet. These formerly relied upon now deprecated Cisco (and other vendor) RAS modem server banks. VOCAL AMA devices can be used in several modes to service these applications. The modem in the AMA ensures a high quality signal for support of the fastest data connections. The packetized application data may be transported back to a central site for access to a local PPP/RADIUS server (using a SAMS). Alternatively, the AMA may use its internal PPP stack and either deliver packets directly to a locally connected IP network or tunneled to remote IP network. Communications for all modes can be secured using TLS or DTLS as per configuration.

Alarm Over IP

Alarm panels use multiple communications methods including DTMF, tonal and low speed modems (actually some use V.32/V.34 modems while most simply use FSK Bell 102 for V.21). VOCAL's AMA may be optionally configured to support alarm panel relay operations with transparent protocol passthrough. Supported protocols include SIA DC-02/Pulse Format, DC-03/SIA FSK Format, DC-04/SIA 2000, and DC-05/Contact ID.

Point-of-Sale Over IP (POSoIP)

The AMA supports for Point-of-Sale (POS) terminals which use typically V.22 modems with a quick connect process (used by Veriphone) to send ASYNC or HDLC framed data. VOCAL supports all common non-standard methods for POS applications including byte timing in both the AMA and SAMS configurations. Again VOCAL POS support is an option for its AMA devices.

FAX Over IP (FoIP)

T.38 Real-time Fax is supported for reliable end-to-end transport and confirmation of image pages. T.38 allows the use of ordinary group 3 facsimile as well as high speed V.34 machines on modern Internet Protocol (IP) networks, permitting the system to relay fax calls through a VoIP service. Without the T.38 protocol, successful calls can be placed, but these can be less reliable than facsimile transmissions through an ordinary telephone line, because modem protocols were not designed to handle the issues a packet network introduces. The VoIP provider's T.38 FoIP gateway converts between the T.30 protocol used for analog fax, and the T.38 data stream used on the IP side. All T.38 versions are supported.

Telocator Alphanumeric Protocol (TAP) Gateway

TAP is utilized for transmitted paging messages over traditional analog telephone networks for critical notification and monitoring. TAP messages must be sent from TAP endpoints to paging terminals and gateways which are typically owned and controlled by cellular network operators. As both analog telephone lines and legacy cellular services are being phased out, most carriers are terminating their TAP services. The AMA supports TAP v1.8 gateway functionality to terminate paging calls internally and forward them to IP based messaging services, E-mail, or through customizable interfaces. This is ideal for services such as Critical/On-Call Notifications (Health Care, Service industries), Network Monitoring Systems, Environmental Sensors, and Smart Building Management Systems.

Specifications and Features

The AMA/ATA product family from VoCAL offers the following capabilities:

Technical Specifications

Voice-over-IP (VoIP) protocols
SIPv2 - Session Initiation Protocol (RFC 3261, 3262, 3263, 3264)
SDP - Session Description Protocol (RFC 4566)
RTP - Real-Time Protocol (RFC 3550, 3551)
RTCP - Real-Time Control Protocol (RFC 3550)
RFC 4733 X-NSE Tone Events for SIP/RTP
RFC 4733 AVT Tone Events for SIP/RTP
STUN - Simple Traversal of UDP over NATs (RFC 3789)

Fax Support
G.711 Fax Pass-Through
T.38 Real-Time Fax Over IP
T.38 using UDP
T.38 using RTP
Group 3 (V.17, V.29, V.27ter)
Group 3+ (V.34)

Modem Support
V.151 Modem Signal Pass-Through (G.711 with redundancy)
V.150.1 Modem Over IP
RAS Over IP (for access of remote COM ports)
V.92/V.90 - Server and Client PCM Modems (optional)
V.34/V.32/V.32bis - High Speed Full Band Modem
V.21/V.23/V.22/V.22bis - Low Speed Modems
POS Support Optional
TAP v1.8 Gateway Optional

Alarm Panel Support Optional
SIA DC-02: Pulse Format
SIA DC-03: SIA FSK Format
110/300 baud with automatic detection
SIA DC-04: SIA 2000
SIA DC-05: Contact ID

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)
HTTP - HyperText Transfer Protocol
TFTP - Trivial File Transfer Protocol (RFC 1350)
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
GSM, GSM HR, GSM FR and GSM AMR
iLBC - Internet Low Bitrate Codec
Speex/Opus – Nonproprietary VDR Codec
MELPe - 2400/1200/600 bps Codec
TSVCIS - Wideband VDR MELPe extension

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
16 to 64 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/100baseT
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 (#)
Configurable Terminating Impedance - 16 Settings
Dial Plan Accessible
Optional Relay Deactivated on Power Fail
Lifeline Port (Processor Controlled Relay)

Indicators
Optional 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
5V DC Input
1.1 Amp (2 FXS with 4 REN total)

Feature List

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)
Secure SIP (SIPS)
Secure RTP (SRTP)
TLS 1.1 or later

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

Telephony User Features

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 (DOCOMO available on special order)
- 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 Pattern