DSL Forum

Technical Report
TR-122

Base Requirements for Consumer-Oriented Analog Terminal Adapter Functionality

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Produced by DSL Home™ Technical Working Group

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Notice

Version History

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1	02/10/2005	Barbara Stark Jeff Multach	Initial version taken from 2004.430 as updated based on Conference call of 2/8/2005.
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Summary

This working text presents specifications for a SIP Based, Consumer-Oriented Analog Terminal Adapter (ATA) functionality to be used with DSL and with characteristics also identified in TR-069 and TR-104. When this functionality is incorporated into a stand-alone device (including a device with an embedded switch), specific requirements relating to the physical nature of that device are included in this document. When this functionality is incorporated into a device with router capabilities, additional requirements relating to the physical and non-VoIP operation of those devices are specified in other documents (e.g., WT-124, TR-068, and TR-098).

DSL Forum Working Text WT-122

Base Requirements for Consumer-Oriented Analog Terminal Adapter Functionality

1 Purpose

This working text presents specifications for a SIP Based, Consumer-Oriented Analog Terminal Adapter (ATA) functionality to be used with DSL and with characteristics also identified in TR-069 and TR-104. When this functionality is incorporated into a stand-alone device (including a device with an embedded switch), specific requirements relating to the physical nature of that device are included in this document. When this functionality is incorporated into a device with router capabilities, additional requirements relating to the physical and non-VoIP operation of those devices are specified in other documents (e.g., WT-124, TR-068, and TR-098).

2 Scope

The device, whose specification is presented in this document, supports packet telephony by terminating VoIP protocols and putting the additional phone lines out across an analog telephony interface or provides an alternative technology for distributing the lines within the premises.

These requirements specify a minimum set of specifications. It is expected that devices will include these in a superset of features.

2.1 Abbreviations

This technical Report defines the following abbreviations:

ac	alternating current
ADSL	Asynchronous Digital Subscriber Line
AKA	Authentication and Key Agreement
ANSI	American National Standards Institute
ASCII	American Standard Code for Information Interchange
ATA	Analog Terminal Adapter
BOOTP	Bootstrap Protocol
CAT3	Category 3
CAT5	Category 5
CPE	Customer Premises Equipment
CPN	Customer Premises Network
CSA	Canadian Standards Association
dBrnC	Decibels Above Reference Noise with C-Message Weighting
dc	direct current
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name Server

DoS	Denial of Service
DSCP	Differentiated Services Code Point
DSL	Digital Subscriber Line
DTMF	Dual Tone Multi-Frequency
EARP	Ethernet Address Resolution Protocol
EIA	Electronic Industries Alliance
FCC	Federal Communications Commission
FSD	Feature Specification Document
FXO	Foreign Exchange Office
FXS	Foreign Exchange Station
GMT	Greenwich Mean Time
GSM	Global System for Mobile Communications
GUI	Graphical User Interface
HTML	HypertextMarkup Language
HTTP	Hypertext Transfer Protocol
HTTPS	Secure Hypertext Transfer Protocol
Hz	Hertz
ICE	Interactive Connectivity Establishment
IEEE®	The Institute of Electrical and Electronics Engineers
IETF	The Internet Engineering Task Force
iLBC	Internet Low Bit Rate Codec
IP	Internet Protocol
ISO	International Organization for Standardization
ITU	International Telecommunication Union
IVR	Interactive Voice Response
Kbps	kilobits per second
LAN	Local Area Network
LSSGR	Local Switching System General Requirements
mA	milli-amp
MAC	Medium Access Control
MEGACO	Media Gateway Control
MGCP	Media Gateway Control Protocol
MIME	Multipurpose Internet Mail Extensions
MLTS	Multi-Line Telecommunications Systems
MoIP	Modem over IP
MOS	Mean Opinion Score
ms	milli-second
MTBF	Mean Time Between Failure

NANPA	North American Numbering Plan Adminstrator
NAT	Network Address Translation
NTP	Network Time Protocol
PC	Personal Computer
PCM	Pulse Code Modulation
PESQ	Perceptual Evaluation of Speech Quality
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
REN	Ringer Equivalence Number
RTCP	Real Time Control Protocol
RTP	Real-time Transport Protocol
SDP	Session Description Protocol (SDP)
SGCP	Simple Gateway Control Protocol
SIP	Session Initiation Protocol
SNTP	Simple Network Time Protocol
SRTP	Secure Real-time Transport Protocol
SRV	Service Type
SSL	Secure Sockets Layer
STUN	Simple Traversal of User Datagram Protocol
TCP	Transmission Control Protocol
TFTP	Trivial FTP
TIA	Telecommunications Industry Association
TLS	Transport Layer Security
TR	Technical Report
TSB	Telecommunications Systems Bulletin
TURN	Traversal Using Relay NAT
UDP	User Datagram Protocol
UL	Underwriters Laboratories
ULC	Underwriters Laboratories Canada
UPS	Uninterruptible Power Supply
URI	Uniform Resource Identifier
UTC	Coordinated Universal Time
UTF	Unicode Transformation Format
Vac	Volts ac
Vdc	Volts dc
VAD	Voice Activity Detection
VID	VLAN Identifier
VLAN	Virtual LAN

VoIP	Voice over IP
Vrms	Volts RMS
VSC	Vertical Service Code
W3C	World Wide Web Consortium
WAN	Wide Area Network
WEP	Wireless Encryption Protocol
WPA	Wi-Fi Protected Access
WT	Working Text
XR	Extended Reports

2.2 Conventions

In this document, several words are used to signify the relative importance of the specified requirements.

MUST This word, or the adjective "REQUIRED", means that the definition is an absolute requirement of the specification.

MUST NOT This phrase means that the definition is an absolute prohibition of the specification.

SHOULD This word, or the adjective "RECOMMENDED", means that there may exist valid reasons in particular circumstances to ignore this item, but the full implications must be understood and carefully weighted before choosing a different course.

MAY This word, or the adjective "OPTIONAL", means that this item is one of an allowed set of alternatives. An implementation that does not include this option MUST be prepared to inter-operate with another implementation that does include the option.

By Default These words indicate that this is a default setting or operation of the unit which **MUST** be configurable if provided.

Other ATA features not identified in this document may also be implemented in the ATA. An implementation that includes features not identified in this document MUST be prepared to inter-operate with implementations that do not include these features.

Requirements which are specific to North America start with [North America].

3 References

The following DSL Forum Technical Reports and other references contain provisions, which, through reference in this text, constitute provisions of this Technical Report. At the time of publication, the editions indicated were valid. All Technical Reports and other references are subject to revision; users of this Technical Report are therefore encouraged to investigate the possibility of applying the most recent edition of the Technical Report and other references listed below. A list of the currently valid DSL Forum Technical Reports is published at www.dslforum.org.

NOTE – The reference to a document within this Technical Report does not give it, as a stand-alone document, the status of a Technical Report.

NOTE – A number of IETF drafts are referenced in this document. Due to the fact that VoIP standards and technology are still being rapidly developed, this was considered necessary. If subsequent drafts or RFCs are published, they will obsolete the draft referenced in this document.

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4 Stand-alone ATA Requirements

The following requirements apply *only* to stand-alone ATAs, including those with one or more switched Ethernet ports. These requirements specifically *do not* apply to Residential Gateway devices with embedded ATA ports that are designed to provide routing and home networking functionality based on the requirements defined in DSL Forum WT-124.

4.1 Physical and Power

I - 1 The device **MUST** be compact and have a physical profile suitable for desktop.

I - 2 The device **SHOULD** be able to be wall mounted and stand on its side.

I - 3 The device **MAY** have the ability to be mounted horizontally or vertically.

I - 4 If wall mounted, the device **SHOULD** be oriented so that the cabling is routed toward the ground in order to reduce strain on the cabling.

I - 5 A detachable wall-mounting bracket **MAY** be added to the device.

I - 6 If the device can be wall mounted, specifications for screws and a template **SHOULD** be included with the device.

I - 7 [North America] The device **MUST** be UL 60950 listed. This is the most recent replacement for UL 1950.

I - 8 [North America] The device **MUST** display proof of CSA (Canadian Standards Association) or ULC (Underwriters Laboratories Canada) certification for CAN/CSA C22.2 No. 60950. This is the Canadian equivalent to, and is identical to, UL 60950.

I - 9 [North America] The device **MUST** meet all requirements when operating with the following line voltages:

Brownout	96 to 127 Vac @ 60 +/- 0.1 Hz
Reserve	105 to 129 Vac @ 60 +/- 3.0 Hz

I - 10 Outside of North America, voltage and powering must meet the requirements of the country where the device will be used.

I - 11 The power connector at the device **MUST** be securely connected to avoid accidental disconnect. This means that the connector **MUST** be either secured via a clip to the box or be held in place with significant force so that it does not readily pull out by minor pulling on the power cord.

I - 12 [North America] If the power supply is external to the ATA, it **MUST** be UL 1310 or UL 60950 listed and certified.

I - 13 If the power supply is external to the device, it **SHOULD** be labeled with the ATA device vendor's name and the model number of the ATA device.

I - 14 If the power supply is external to the device it **SHOULD** be either small enough, or appropriately positioned on the power cord, so as not to block other power outlets.

I - 15 If the power cable includes an external power supply, that power supply **MAY** have a power indicator light on it.

I - 16 The device **MUST** have a single-function recessed button in order to reset the device to the default factory settings.

I - 17 The reset button on the device **SHOULD** have a red circle around it.

I - 18 The reset button on the device **MAY** be labeled as "reset" so a help desk can more easily identify it to a user.

I - 19 Each port on the back of the device **MAY** have an icon displayed near it identifying the type of port.

I - 20 The ports on the device **MUST** be identified by color with the appropriate connection/interface color reflected above, below or around each port.

I - 21 The non-VoIP related ports **MUST** be colored as follows (see I - 136 for VoIP related ports):

- Ethernet Yellow
- Power Black

The preferred PANTONE® color for yellow is:

- Yellow 114C

I - 22 Each port on the back of the device **MUST** be labeled using icons and/or words, and any words must be spelled out completely (e.g., "Ethernet", "Power", etc.).

I - 23 The device **MUST** operate 24 hours a day, 7 days a week without the need to reboot.

I - 24 The MTBF (Mean Time Between Failure) of the device and operating system **SHOULD** be equal to or exceed 1 year (e.g., there **SHOULD** be no more than one unplanned reboot per year).

I - 25 The life expectancy of the device **SHOULD** be at least seven years.

I - 26 The device **MUST** complete power-up in 60 seconds or less (timing starts when the power is connected and stops when the On/Off power indicator light is "Solid Green").

I - 27 [North America] The device **MUST** comply with FCC Part 15 rules for Class B devices.

I - 28 [North America] The device **MUST** comply with Industry Canada ICES-003 Class B requirements.

I - 29 [North America] The device **MUST** comply with the requirements of Telcordia® GR-1089-CORE, Electromagnetic Compatibility and Electrical Safety - Generic Criteria for Network Telecommunications Equipment. Class A3 source voltages are not permitted.

I - 30 [North America] The device **MUST** support the following environmental conditions:

Environment	Temperature	Altitude	Relative Humidity	MWB
Operating (System Ambient)	0^{o} C to 40^{o} C	-60 to 2134 meters (-197 to 7000 feet)	8% to 95% non-condensing	23^{o} C
Shipping and Storage	-25^{o} C to 65^{o} C		low humidity for low temperatures,	29^{o} C

			90% at 45 ^{o} C, 30% at 65 ^{o} C	

I - 31 This device **MUST** preserve local configuration information during power-off and power interruption.

4.2 Visual Indicators

I - 32 In addition to the VoIP related light(s) identified in I - 141, the device **MUST** have the following indicator lights (which may be labeled with different words or symbols, according to local language and labeling practices):

`Power    Network`

I - 33 The Network indicator lamp **MUST** function as follows:

Solid Green = Unit has a working IP address.

Solid Red = Unit does not have a working IP address.

Off = Device power is off or there is no network connection.

I - 34 If multiple Ethernet ports are provided, then the device **MUST** have ports labeled to indicate which one is used for uplink and which one is used for PC/Hub.

I - 35 The port for the PC/HUB **SHOULD** be labeled "PC", or as according to the local language and labeling practices.

I - 36 If the device has an FXO port, then the port **MUST** be labeled "Wall", or as according to the local language and labeling practices.

I - 37 The device **MUST** have an "On/Off" power indicator light (identified in I - 32). The power indicator **MUST** be visible from the front of the unit and function as follows:

Solid Green = Power on

Off = Power off

Red = POST (Power On Self Test) failure (not bootable) or Device malfunction

I - 38 The Power indicator light **MUST** be labeled “Power”, or as according to the local language and labeling practices.

I - 39 All indicator lights on the unit **MUST** flash when the unit has software being downloaded to it.

I - 40 All physical network ports and bridged network connection types on the device (e.g., Ethernet, Wireless, HomePlug, HomePNA, FireWire, etc.) **MUST** each have a link integrity indicator lamp on the device (1 per port if a separate physical port is present or per connection type if a separate port is not present).

I - 41 Each port indicator light **MUST** be located next to the associated physical port. Port indicator lights for connection types with no physical ports (e.g., wireless) **MUST** be located on the front of the device.

I - 42 Summary indicator lights **MAY** be placed on the front of the unit for physical port indicator lights which are not visible from the front of the device.

I - 43 The physical port indicator lamps **MUST** function as follows:

Solid Green	=	Powered device connected to the associated port (includes devices with wake-on-LAN capability where a slight voltage is supplied to an Ethernet connection)
Flashing Green	=	LAN activity present (traffic in either direction)
Off	=	No activity present, device power is off, and no powered device is connected to the associated port.

I - 44 The network port indicator lights **MUST** be labeled as indicated in the parenthetic list of item I - 40, or as according to the local language and labeling practices.

I - 45 When flashing, the indicator lights **MUST** flash at 4 Hz with a duty cycle of 50% (except as specified otherwise in this document).

4.3 Networking Interfaces

I - 46 The device **MUST** have a 10/100 BASE-T Ethernet port (IEEE Std 802.3; RJ-45) for a WAN uplink which connects either directly or indirectly to the Internet (e.g., connects to a broadband router or associated routing device).

I - 47 The device **MAY** support an additional 10/100 BASE-T Ethernet port (RJ-45) for a downstream interface that connects to a hub or PC.

I - 48 The device **MUST** auto-negotiate for speed and duplex according to IEEE Std 802.3.

I - 49 If a downstream Ethernet port is provided (e.g., used to connect to a PC or local network), the device **MUST** function as a switch and not as a router. The device also **MUST NOT** support NAT or act as a DHCP server to LAN devices that may be connected to the downstream Ethernet port.

The above requirement is included simply to strengthen the definition of this section of requirements as being for a stand-alone ATA without routing functionality.

I - 50 If a downstream Ethernet port is provided, the device **SHOULD** be able to prioritize and schedule its own traffic upstream ahead of that received from any device on its downstream port.

I - 51 The device **SHOULD** support IEEE Std 802.11g client connectivity to the Customer Premises Network (CPN) with WPA, WPA2 and WEP 40/64 bit and WEP 128 bit encryption.

I - 52 The device **SHOULD NOT** function as a wireless access point.

4.4 Networking Functions

I - 53 The device **MUST** support the TCP, IP, UDP, routing and associated protocols identified here:

- IETF RFC 0768 User Datagram Protocol
- IETF RFC 0791 Internet Protocol
- IETF RFC 0792 Internet Control Message Protocol
- IETF RFC 0793 Transmission Control Protocol

- IETF RFC 0826 An Ethernet Address Resolution Protocol
- IETF RFC 0894 A Standard for the Transmission of IP Datagrams over Ethernet Networks
- IETF RFC 1042 A Standard for the Transmission of IP Datagrams over IEEE 802 Networks

I - 54 The device **MUST** support IP Version 4 (IETF RFC 791).

I - 55 The device **MUST** support a DHCP client. This includes support for the standards below:

- IETF RFC 2131 Dynamic Host Configuration Protocol
- IETF RFC 2132 DHCP Options and BOOTP Vendor Extensions

I - 56 The device **MUST** be able to obtain IP network information dynamically on its connection to the broadband interface using DHCP. This information includes an IP address, primary and secondary DNS addresses, and a default gateway address.

I - 57 By default, the device **MUST** identify itself as "atadevice<macaddress>" (where <macaddress> represents the MAC address of the device) in DHCP and other protocols in which the device name is specified.

4.5 Security

I - 58 The device **MUST** provide Denial of Service (DoS) protection for itself including protection from

a. Ping of Death [CERT® Advisory CA-1996-26]
b. SYN Flood [CERT® Advisory CA-1996-21]
c. LAND [CERT® Advisory CA-1997-28] and variant attacks
d. Port scans
e. Packets with same source and destination addresses (a variant of LAND attack)
f. Invalid fragmented IP packets
g. Packets with invalid TCP flag settings (NULL, FIN, Xmas, etc.)
h. Fragmented packet headers (TCP, UDP, and ICMP)
i. Invalid ICMP requests
j. Internally inconsistent IP, UDP, or TCP header lengths
k. Forwarded broadband packets with broadcast source address **MUST** be discarded.
l. Packets received from the WAN with a LAN source address (as per the IP subnet configured on any of the LAN interfaces in the device) **SHOULD** be discarded.
m. If an IP packet containing a TCP header is fragmented, the first fragment **SHOULD** be dropped.
n. The device **SHOULD** implement protection against TCP Sequence-number exploitation and reduce the probability of reset attack (which leads to unwanted connection-close) and the hijacking of a TCP connection by a third party's relying on guessing an allowed sequence number (based on overly large TCP window-size or on predictable TCP initial sequence number). Specifically, the device **SHOULD** implement the mechanisms identified in RFC 1948 and IETF draft draft-ietf-tcpm-tcpsecure-04.

I - 59 The device **SHOULD** drop or deny access requests from connections outside the LAN to the device itself except in direct response to outgoing traffic or as explicitly permitted through configuration of the device (e.g., for management).

I - 60 By default, the device **MUST NOT** allow "back door" entry to the unit (e.g., there must be no hidden telnet or web access using secret passwords).

I - 61 The device **SHOULD** support user level configuration password protection. This **SHOULD** be turned off by default.

I - 62 Unencrypted passwords **MUST NOT** be displayed nor broadcast in any way by the device (e.g., in a configuration file or on a web page).

This does not apply to the transmission of encrypted passwords used for network or session authentication.

I - 63 The device **MUST** be able to password protect configuration information and administrative functions.

I - 64 Access to the device configuration interface **MAY** be blocked by the service provider so as not to allow misconfiguration of the settings.

4.6 User Interface and Management

I - 65 Once a configuration server address is defined, the device **MUST** automatically connect to that configuration server after this definition has been made and upon each subsequent power-up. If a configuration server has not been defined and individual connection parameters have been defined (e.g., registration server address, proxy server address, username, password, …) then these **MUST** be used to establish a session after initial specification and upon power-up.

I - 66 A console port that allows end user access (e.g., placed on the outside of the device) **SHOULD NOT** be provided on the device.

I - 67 The device **SHOULD** be able to detect faults and, if in a lock-up condition or a condition that would lead to an improperly operating state, restart automatically.

I - 68 The device **SHOULD** be self-installable by an end user in under 10 minutes assuming the default configuration and mode of operation for the device. This is the time from when the box is opened to the time when the user is able to make a phone call (assuming no network complications and excluding installation of other related devices and customer ordering/registration).

I - 69 Configuration and installation of the device **SHOULD** minimize the number of restarts of the device when enabling changes. The device **SHOULD NOT** require a reboot to implement a configuration change.

I - 70 If software is loaded on LAN CPE for installation or configuration of the device, this software **MUST NOT** require the associated LAN CPE to restart.

I - 71 User installed software and drivers **MUST NOT** be required for proper and full use of the device.

I - 72 A PC **MUST NOT** be required for proper and full use of the device.

I - 73 The device **MUST** support the generic device TR-069 interface requirements as defined in TR-106.

I - 74 The device **SHOULD** support TR-111 Part 2 requirements for management of devices inside a home network.

I - 75 The device **MUST** be configurable and support management access using HTTP via embedded, easy-to-use web pages. Access may be limited or restricted by other functions of the device (e.g., as identified in I - 322, I - 324 and I - 325).

I - 76 The device **MAY** support software/firmware upgrades via TFTP as a client**.**

I - 77 The embedded GUI **MUST** be accessible from all Ethernet ports on the device.

I - 78 The device **MAY** support updating of its firmware via the embedded GUI by a user browsing and selecting an update file from a local PC and using HTTP to update the device using this file (see IETF RFCs 1867, 2388 and HTML 4.1 specifications for more details).

I - 79 The device **MUST** support loading firmware files using TR-069 with the SSL option.

I - 80 The device **MUST** support loading configuration files using TR-069 with the SSL option.

I - 81 GUI authorization, if enabled, **MUST** time out after 30 minutes.

I - 82 The device **MUST NOT** require browser support of Java, ActiveX nor VBSCRIPT in its web pages.

I - 83 The web pages **SHOULD** minimize internal page complexity (e.g., excessive use of frames, pop-ups, style sheets, JavaScript, etc…) that places demands on browser resources or causes interoperability problems with different browsers. In general, all pages **SHOULD** load within five seconds.

I - 84 The web pages and web server **MUST** interoperate with Opera, Mozilla®, Safari, Netscape® and Internet Explorer® web browsers, regardless of the computer operating system that these browsers are operating on. This does not apply to browsers for handheld devices.

I - 85 The web pages **MUST** work with Netscape® 6.0, Microsoft® Internet Explorer® 6 and later versions of these browsers.

I - 86 The device **MUST** use standard protocols when using TFTP and HTTP (e.g., TFTP – IETF RFCs 1350, 1785, 2347, 2348, 2349, HTTP - IETF RFC 2616, HTTPS - IETF RFCs 2246, 2818).

I - 87 The device **MUST** preserve its configuration across firmware updates.

I - 88 The vendor **SHOULD** have a web site where documentation is available.

I - 89 If the device supports updating of its firmware from the LAN side GUI, the vendor **SHOULD** have a web site where firmware updates are available.

I - 90 The documentation (from I - 88) **SHOULD** include manuals containing detailed installation procedures, corrective actions for troubleshooting, and subsequent release notes for all firmware versions.

I - 91 If firmware (from I - 89) is provided at a vendor's web site, the firmware **SHOULD** include all error-correcting updates for the device.

I - 92 All software revisions **SHOULD** be backward compatible with all previous versions. There **SHOULD** be no loss of existing functionality.

I - 93 Software revisions **MUST NOT** require service provider network changes to maintain proper operation of previous features.

I - 94 The vendor of the device **MUST** adhere to a vendor's self-defined standard numbering and revisioning scheme for all firmware releases and all documentation.

I - 95 If the device supports enabling telnet access, this option **MUST** be turned off by default.

I - 96 All firmware updates **MUST** be verified using security mechanisms. A checksum mechanism is a minimum requirement for achieving this.

I - 97 All firmware updates **SHOULD** be verified using a cryptographic "fingerprint" of at least 256 bits.

I - 98 In the event of a failure occurring during an update, the device **MUST** be able to back off to the prior version of the firmware installed on the device.

That is, the prior version of the device's firmware **MUST** continue to be useable in the event that a firmware update fails to complete.

This is not a requirement for a dual image, but that is one manner in which this requirement might be achieved.

I - 99 The device **MUST** provide detailed information for current connections and associated parameters including Ethernet connection speed, MAC address, IP address, IP gateway, DNS server and configuration server.

I - 100 The model number, serial number and MAC address **MUST** be visible via external markings on the device.

I - 101 The device **MUST** allow its assigned IP address and netmask to be specified through the XML and GUI interfaces.

I - 102 If the device is not configured, the device **MUST** have a quick start page allowing for specification of a configuration server.

I - 103 The model and firmware/software versions **MUST** be easily identifiable via the GUI interface.

4.7 Logs

I - 104 The device **MUST** maintain an internal log of all connection flows (e.g., DHCP, IP, etc). At a minimum, the log **MUST** record the last 250 events. This will include connection events initiated by the device or from the WAN side connections. The purpose of the log is to provide a troubleshooting aid in resolving line and connection problems.

I - 105 The device **MUST** timestamp each log entry.

I - 106 The factory default timestamp value for log entries **SHOULD** indicate the elapsed time since the unit was first powered on. The log entry timestamp **SHOULD** be formatted, consistent with ISO 8601:2004, as follows:

PYYYY-MM-DDThh:mm:ss

where:

P	=	the letter "P" used to indicate what follows is a time interval (period) data element
YYYY	=	number of years (digits)
MM	=	number of months (digits, 01 – 12; 1 month is the equivalent of 30 days for time interval purposes)
DD	=	number of days (digits, 01 – 30)
T	=	the letter "T", used to indicate the start of the time-units for hours, minutes and seconds
hh	=	number of hours (digits, 00 – 24)
mm	=	number of minutes (digits, 00 – 60)
ss	=	number of seconds (digits, 00 – 60)

Once the device has established connectivity to an Internet based time server, all subsequent log entry timestamps **SHOULD** be formatted for GMT or user specified time zone (24 hour military format), consistent with ISO 8601:2004, as follows:

YYYY-MM-DDThh:mm:ss±hh:mm or

YYYY-MM-DDThh:mm:ssZ

where:

YYYY	=	year (digits)
MM	=	month (digits, 01 – 12)
DD	=	day of month (digits, 01 – 31)
T	=	the letter "T", used to indicate the start of the time of day
Z	=	the letter "Z", used to indicate that the time is UTC (Coordinated Universal Time)
hh	=	hours (digits, 00 – 24)
mm	=	minutes (digits, 00 – 60)
ss	=	seconds (digits, 00 – 60)
±hh:mm	=	the difference between local time and UTC in hours and minutes (e.g., -05:00 would indicate Eastern Standard Time, 5 hours behind UTC)

I - 107 The device **MAY** be able to copy log files to a PC on the local LAN or network server in ASCII text format, using the GUI interface.

I - 108 The device log **SHOULD** reside on the device.

I - 109 The device log **SHOULD NOT** interfere with the normal performance of the device. That is, the prioritization of writing log entries to non-volatile storage **SHOULD NOT** be done at a priority or in a manner that would degrade the user experience nor the connection throughput.

I - 110 The device **MAY** be able to transmit log messages to an external device using the syslog protocol as identified in IETF RFC 3164.

4.8 Setting Date and Time

I - 111 The device **MUST** support an internal clock with a date and time mechanism.

I - 112 The device clock **MUST** be able to be set via an internal time client using NTP (IETF RFC 1305) or SNTP (IETF RFC 4330) from an Internet source.

I - 113 The device **MUST** support the use of time server identification by both domain name and IP address.

I - 114 If the device includes default time server values, they **SHOULD** be specified by domain name and not by IP address.

I - 115 The device **SHOULD** allow configuration of the primary and alternate time server values in addition to or in place of any default values.

I - 116 If the device includes default time server values or time server values are identified in documentation, these values **SHOULD** be selected using industry best practices.

I - 117 For example, IETF RFC 4330 identifies that the time server names used should be those of servers the manufacturer or seller operates as a customer convenience or those for which specific permission has been obtained from the operator of the time server.

I - 118 The time client **SHOULD** re-resolve any time server IP address obtained from a domain name on a periodic interval, but not less than the time-to-live field in the DNS response.

I - 119 The time client **SHOULD** support DNS responses with CNAMEs or multiple A records.

I - 120 The default frequency with which the device updates its time from a time server **MUST NOT** be less than 60 minutes.

I - 121 The default frequency with which the device updates its time from a time server **MUST NOT** be greater than 24 hours.

I - 122 The frequency with which the device updates its time from a time server **SHOULD** be configurable.

I - 123 The time server discovery and selection stage used by the time client **SHOULD** check each candidate time server in a round-robin fashion, with a response timeout between each request to each time server.

If no time server has responded during a round of checking, the response timeout **SHOULD** be exponentially incremented (e.g., doubled) and the time servers checked again.

The round-robin checking and exponential incrementing of the response timeout **SHOULD** continue until a time server is discovered or a search limit is reached.

I - 124 The device **SHOULD** support the [S]NTP access-refusal mechanism, so that a server returning a Stratum value of zero (0; sometimes termed a kiss-o'-death reply) in response to a client request causes the client to cease sending requests to that server.

If this occurs during the discovery and selection stage for a time server, then the discovery mechanism should continue on to the next time server in its list of those to check or increase the response timeout as identified above.

I - 125 If this occurs when the device is periodically updating its clock, then the discovery and selection stage for a time server **SHOULD** be re-initiated.

I - 126 The device **SHOULD** validate response packets for malformed time protocol packets (invalid flags – such as client query flag, bad packet size, etc.) and ignore invalid packets.

I - 127 The device **SHOULD** ignore time protocol response packets with a source IP address other than that of the time server that the ATA queried.

4.9 Packaging

I - 128 The device **MUST** be packaged with a quick start or installation guide.

I - 129 The Quick Start Guide **SHOULD** be made available in alternate formats including large print.

I - 130 All necessary end-user documentation **MUST** be included with the device.

I - 131 Additional detailed product documentation **SHOULD** be included with the device.

I - 132 The model and serial number **MUST** be visible via external markings on the product packaging (e.g., shipping or display box).

I - 133 All device firmware and associated system files **MUST** be pre-installed.

I - 134 All other supplied cable **MUST** be colored as identified in I - 349 and the same as the ports are colored in I - 21.

I - 135 A Category 5 (CAT5), or better, straight-through (patch) Ethernet cable with RJ-45 endpoints **MUST** be packaged with the product. The cable **MUST** be a minimum length of 2 meters (or 6 feet). The endpoints **MUST** meet the specifications for a miniature 8-position unkeyed plug in TIA-968-A.

5 General ATA Requirements

The following requirements apply to all devices with ATA functionality, including stand-alone ATA products, and Residential Gateway devices with embedded ATA function.

5.1 Physical and Power

I - 136 The FXS and FXO ports **MUST** be colored as follows:

- FXS Gray
- FXO Green

The preferred PANTONE colors for gray and green are:

- Gray Cool Gray 3U (matte)
- Green Hexachrome Green C

I - 137 The device **SHOULD** include sufficient non-volatile memory to accommodate future control and data plane protocol upgrades over a minimum of four years. The potential upgrades may include support for IPv6 and support for enhancements to the VoIP protocols (e.g., SIP, SDP and RTP).

I - 138 [North America] The FXS, and FXO (if available), ports of the device **MUST** comply with and be registered for compliance with TIA-968-A, Telecommunications – Telephone Terminal Equipment – Technical Requirements for Connection of Terminal Equipment to the Telephone Network, October 2002, and **MUST** be certified to meet FCC Part 68, or obtain the appropriate waiver.

I - 139 [North America] The device SHOULD conform to Clause 4.6.7 of Telcordia GR-1089 for improved surge protection beyond that required by TIA-968-A.

I - 140 The device **MAY** include UPS functionality (e.g., an internal battery) for support of continuous voice operation through a power interruption.

5.2 Visual Indicators

I - 141 The device **MUST** have, as a minimum, the following indicator light, in addition to any other indicator lights defined for the device type as defined above for stand-alone ATA devices, or in TR-068v2 and subsequent versions and other DSL Forum requirements documents for Residential Gateways devices (labeling may use different words or symbols, according to local language and labeling practices):

`Phone`

I - 142 If the device has multiple FXS (i.e., Phone) ports, the ports **MUST** be labeled numerically starting with the number "1" (i.e., "Phone 1", "Phone 2", etc.).

I - 143 Each FXS interface on the device **MUST** have its own indicator light.

I - 144 Each FXS indicator light **MUST** be visible from the front of the device.

I - 145 The FXS port indicator lamps **MUST** function as follows:

Solid Green	=	The associated FXS port has been registered with a SIP proxy server.
Flashing Green	=	Indicates when the associated telephone is off-hook.
Off	=	Line is not registered or device power is off.

5.3 Foreign Exchange Station (FXS) Port(s)

I - 146 The device **MUST** support at least one (1) RJ-11 jack with the signal present on the inner pins (pins 3 & 4) as a FXS interface (i.e., voice port).

I - 147 The RJ-11 jacks **MUST** meet the specification(s) for a miniature 6-position jack in TIA-968-A.

I - 148 The device **SHOULD** support two (2) FXS interfaces.

I - 149 If the device has two FXS interfaces and has been configured for only one line, the default behavior of the device **SHOULD** be that VoIP line 1 appears on the inner pairs (pins 3 & 4) of both jacks.

I - 150 If the device has two FXS interfaces and has been configured for two lines, the default behavior of the device **SHOULD** be that VoIP line 1 appears on the inner pair of the first jack (pins 3 & 4) and on the outer pair of the second jack (pins 2 & 5) and VoIP line 2 appears on the inner pair of the second jack and the outer pair of the first jack.

I - 151 If the device has multiple FXS interfaces, which line appears on which jack(s), and on which line pair(s) of these jacks, **SHOULD** be configurable.

I - 152 FXS interfaces **MUST** support loop-start signaling per T1.401-2000, "Network to Customer Installation Interfaces – Analog Voicegrade Switched Access Lines Using Loop-Start and Ground Start Signaling", and provide a RJ-11 interface per T1.TR.05-1999, "Network and Customer Installation Interface Connector Wiring Configuration Catalog", for each derived voice line.

I - 153 FXS interfaces **MUST** meet the requirements of ANSI/TIA-1063, Telecommunications - Telephony Aspects of MLTS and Packet-based Equipment, including VoIP - Analog Telephone Port Requirements for Packet-based User Premises Terminal Adapters, except where stricter requirements are provided below (in Section 5.3 and Section 5.6), or where alternate local, national, or regional requirements apply.

I - 154 The device **MUST** present an idle-line voltage of a nominal -48 volts dc (-42.5 to -56.5 Vdc) to derived voice ports to insure full compatibility with existing premises systems.

I - 155 The FXS interfaces **MUST** meet the following ringing requirements:

- The applied ringing voltage **MUST** meet the duration-limited source safety requirements of GR-1089-CORE.
- The frequency of the ringing voltage **MUST** be 20 ± 1 Hz.
- The ringing voltage **MUST** be least 55 Vrms superimposed on at least 15 Vdc across a ringing load of 5 REN at the end of any loop with a loop resistance 30 ohms.
- The dc component of the ringing voltage **MUST** be at least 15 Vdc and **MUST NOT** exceed 80 volts dc.
- During application of ringing and during ring trip, the bridged C-weighted noise referenced to 900 ohms **MUST** be less than or equal to 90 dBrnC and the analog voiceband lead-conducted emissions criteria of GR-1089-CORE **MUST** be met.
- All of the above criteria **MUST** be met with any applied ringing load from 0 REN to 5 REN.
- During the application of ringing, the ac ringing current **MUST** not exceed 220 mA at any phase angle into any ringing load.

I - 156 The crest factor (i.e., peak-to-rms voltage ratio) of the ac component of the ringing voltage **MUST** be greater than or equal to 1.35 and less than or equal to 1.45.

I - 157 [North America] The device **MUST** support a minimum Ringer Equivalence Number (REN) of 5 on each voice port to insure that multiple extensions to a single line can ring simultaneously.

I - 158 The ATA **SHOULD NOT** be damaged, **SHOULD NOT** have to be power-cycled, rebooted, etc, after the continuous application of a voltage source -- having the voltage and current characteristics falling anywhere within the 'Network Operating Region' of Figure 3 of ANSI Standard T1.401-2000 -- to the FXS port.

I - 159 It **MUST** be possible to configure the FXS ports to either provide a "fast busy" signal or "no voltage and no signal" in the case where the FXS port(s) are configured and enabled, but are unable to successfully register for SIP service.

I - 160 Line Testing Capabilities (Section 8.3) of ANSI/TIA-1063 **MUST** be supported.

5.4 Foreign Exchange Office (FXO) Port(s)

I - 161 The device **MAY** have a FXO interface to a POTS service.

I - 162 If the device has a FXO interface to a POTS service, it **MUST** recognize service on the inner pair if present.

I - 163 If the device has a FXO interface, then its REN **MUST** draw a maximum REN = 0.2 and a maximum dc current of 10 μA..

I - 164 If the device has a FXO interface, then the device **MUST NOT** use the local phone loop for power on this interface.

I - 165 If the device has a FXO interface to a POTS service, then all FXS interfaces **MUST** fail over to the FXO inner pair service.

I - 166 If the FXS interfaces switch over to the POTS service in a power failure, they **MUST** return to their derived operation when power is restored and the associated POTS line is on-hook.

I - 167 If the device has a FXO interface to a POTS service, then the device **SHOULD** support making a PSTN call through the FXO interface when the broadband network is unavailable or inaccessible. In this case, the device **SHOULD** present a special dial tone when a phone on an FXS interface is off hook.

I - 168 [North America] If the device has a FXO interface, the device **MUST** comply with Industry Canada's "Telecommunication Apparatus Compliance Specification" (IC document CS-03) and be registered with Industry Canada following the procedures highlighted in Industry Canada’s "Procedure for Declaration of Conformity and Registration of Terminal Equipment" document (IC document DC-01).

5.5 Codec Requirements

I - 169 The device **MUST** support ITU-T G.711 codec with 10, 20, and 30 ms frame sizes.

I - 170 The device **MUST** support ITU-T G.711 Appendix I and G.711 Appendix II.

I - 171 The device **MUST** support both μ-law and a-law PCM encoding for G.711.

I - 172 The device **MAY** support ITU-T G.722.

I - 173 The device **SHOULD** support ITU-T G.723.1 (5.3 and 6.3 encoding) with 30 ms frame sizes.

I - 174 If the device supports ITU-T G.723.1, then the device **MUST** support G.723.1 Annex A.

I - 175 The device **MAY** support ITU-T G.723.1 Annex B.

I - 176 The device **SHOULD** support ITU-T G.726 (16, 24, 32, and 40 kbps encoding) with 10, 20, and 30 ms frame sizes.

I - 177 The device **MAY** support ITU-T G.726 Annex A (16, 24, 32, and 40 kbps encoding).

I - 178 The device **MUST** support ITU-T G.729 Annex A with 10, 20 and 30 ms frame sizes.

I - 179 The device **SHOULD** support ITU-T G.729 Annex B.

I - 180 The device **MAY** support ITU-T G.729 Annex E.

I - 181 The device **MAY** support the iLBC™ codec as identified in IETF RFC 3951, Internet Low Bit Rate Codec (iLBC), and IETF RFC 3952, Real-time Transport Protocol (RTP) Payload Format for internet Low Bit Rate Codec (iLBC) Speech.

I - 182 The device **MAY** support the BroadVoice® BroadVoice16 (BV16) codec as identified in IETF RFC 4298, RTP Payload Format for BroadVoice Speech Codecs.

I - 183 The device **MAY** support the Speex codec as identified in draft-ietf-avt-rtp-speex-00, RTP Payload Format for the Speex Codec.

I - 184 The device **MAY** support GSM codecs.

I - 185 The device **MUST** support configuring a default codec to be used. The default codec is the first codec that is used during codec negotiation and selection (as identified in I - 188 and I - 191).

I - 186 [North America] The device **MUST** support G.711 μ-law as the initial default codec, as shipped from the factory, with a default frame size of 20 milliseconds.

I - 187 The device **SHOULD** support identifying which codecs to include or not to include in the signaled list of supported codecs (e.g., in the SIP SDP list of media formats).

I - 188 The device **MUST** support configuring the order of codecs to be negotiated.

I - 189 The device **MUST** support the codec assignment and selection (G.711, G.729 Annex A, etc) independently for each derived line.

I - 190 The device **SHOULD** negotiate an appropriate codec based on available upstream bandwidth or signal to the user that the call cannot be completed.

I - 191 When negotiating codec selection, the device **SHOULD** list codecs in order of priority as established using I - 188.

I - 192 When negotiating codec selection, the device **MUST** use the first matching codec listed by the far end if the requested codec is available in the device.

I - 193 The device **MUST** support dynamically switching (i.e., re-negotiated during the call) between available codecs.

For example, as might occur when changing from a codec which uses compression to a codec which does not use compression, to support facsimile, modem or text telephony pass-through.

I - 194 The device **MUST** support voice activity detection (VAD) on all connections using codecs supporting silence suppression.

I - 195 The device **SHOULD** negotiate the use of the comfort noise codec per IETF RFC 3389, "RTP Payload for Comfort Noise", and support VAD on all connections using codecs which do not inherently support silence suppression.

I - 196 The device **MUST** be able to enable and disable silence suppression on each line.

I - 197 The device **SHOULD** support configurable sensitivity of the VAD algorithm on a per port basis.

The applicability of configurable sensitivity and the manner in which this is done is codec and vendor dependent.

I - 198 The device **MUST** support ITU-T G.168-compliant echo cancellation on each voice port, including configurable tail-end delay up to at least 32 ms.

I - 199 The device **MUST** support comfort noise generation.

I - 200 The device **SHOULD** support packet loss concealment.

I - 201 The device **MUST** support a dynamic jitter buffer, which adjusts the size of the buffer based on detected delay.

I - 202 The device **SHOULD** support a jitter buffer up to 150 ms.

I - 203 The device **SHOULD** support ITU-T P.1010, "Fundamental voice transmission objectives for VoIP terminals and gateways".

5.6 Voice Services and Features

I - 204 The device **MUST** support DTMF tone detection on the FXS ports.

I - 205 The device **MUST** support local generation on the FXS interfaces of dial tone, DTMF tones, ringing and speaker signaling (e.g., busy).

I - 206 The device **MUST** generate local ringing and should ignore any early RTP media when a "180 Ringing" response is received. Any received media that is not early media (i.e., not received within the context of an early session, as specified in IETF RFC 3959, The Early Session Disposition Type for the Session Initiation Protocol) should be rendered as soon as it arrives in order to avoid speech clipping. The device **MUST** play the RTP stream for the established dialog and ignore any other RTP media streams when a "183 Session Progress" response is received.

I - 207 The device **SHOULD** obey the last 18x message received when multiple 18x responses are received.

If the last response is "180 Ringing" the client **SHOULD** generate local ringing. If the last response is "183 Session Progress" the client **SHOULD** play the RTP stream.

I - 208 All signaling on the FXS interfaces **MUST** be done on a per line basis.

I - 209 If the device supports multiple lines, all configuration and operation of connections **MUST** be done on a per line basis. This includes but is not limited to:

- Registration (user credentials, proxy server, configuration server, registration period, …)
- Feature configuration (SIP ports, RTP port range, codecs, timers, …),
- Line status (e.g., Do Not Disturb),
- Message Waiting Indictor, and
- Ring Tones.

I - 210 If the device supports multiple lines, they **SHOULD** be able to use common configuration settings for the lines (to the extent possible) or individual settings as identified in I - 209.

I - 211 The device **MUST** pass a voice quality test using PESQ MOS-LQO measurements and testing per ITU-T P.862, ITU-T P.862.1, and ITU-T P.862.3 in an isolated environment (such that the environment introduces negligible degradation of the input signal). The device **MUST** be able to achieve scores for 711a-law, 711u-law, and 729a that are at least 90% of the reference value listed in Table I.1 of ITU-T P.862.3, for a given language, filename, and codec. At least one language **MUST** be tested, and all filenames within that language **MUST** be tested.

I - 212 The device **MUST** support requirements appropriate to a consumer-grade ATA function specified in:

T1.401.02-2000, Network-to-Customer Installation Interfaces - Analog Voicegrade Switched Access Lines with Distinctive Ringing Features

T1.401.03-1998 (R2003), Network-to-Customer Installation Interfaces - Analog Voicegrade Switched Access Lines with Calling Number Delivery, Calling Name Delivery, or Visual Message-Waiting Indicator Features

T1.401-2000, Network to Customer Installation Interfaces - Analog Voicegrade Switched Access Lines Using Loop-Start and Ground Start Signaling

T1.401a-2001, Supplement to T1.401-2001 - Network - to - Customer Installation Interfaces - Analog Voicegrade Switched Access Lines Using Loop-Start and Ground-Start Signaling

T1.401b-2002, Supplement to T1.401-2001 - Network - to - Customer Installation Interfaces - Analog Voicegrade Switched Access Lines Using Loop-Start and Ground-Start Signaling (Supplement to T1.401-2000)

TIA/EIA 571-A, Telecommunications User Premises Equipment Environmental Considerations

I - 213 The device **MUST** support ANSI/TIA-912-A, Telecommunications - IP Telephony Equipment - Voice Gateway Transmission Requirements

I - 214 The device **MAY** support TSB-116, Telecommunications - IP Telephony Equipment - Voice Quality Recommendations for IP Telephony

I - 215 The device **MUST** support signaling of message waiting indication (over the FXS port(s)), on a per line basis.

I - 216 The device **MUST** support GR-1401-CORE, LSSGR: Visual Message Waiting Indicator Generic Requirements (FSD 01-02-2000) or as in ANSI/TIA-1063.

I - 217 The device **MUST** support stutter dial tone message waiting indication, as described in ANSI/TIA-1063.

I - 218 The device **MUST** support sending appropriate indicator-on/indicator-off messages (not stutter dial tone) to each FXS port requesting activation and deactivation of the message waiting visual indicators on telephony devices attached to the respective port.

I - 219 If the device is able to detect that a phone is connected to its FXS port(s) (i.e., the phone draws enough REN such that it can be detected), the device **MUST** report this when queried from a management system.

I - 220 The device **MUST** be able to report the on-hook/off-hook status of a phone attached to its FXS port when queried from a management system.

I - 221 The device **MUST** support ITU-T T.38 (e.g., it should support Facsimile Relay).

I - 222 The device **MAY** function as a MoIP gateway as identified in ITU-T V.150.0 and V.150.1 (e.g., it should support Modem Relay).

I - 223 When the device is transmitting or receiving modem data, it **MUST** be able to support modem speeds up to 33.6 kbps.

I - 224 The device **MUST** support TIA-1001, Transport of TIA-825-A Signals over IP Networks.

I - 225 The device **SHOULD** support ITU-T V.151.

I - 226 The device **SHOULD** support ITU-T V.152, Procedures for supporting voice-band data over IP networks .

I - 227 The device **MUST** support handling of data communications occurring over the voice stream (i.e., pass-through of facsimile, modem and text telephony).

In particular, the device **MUST**:

- Detect voice band data stimuli,
- Establish the appropriate mode of operation for voice band data after its detection (e.g., renegotiate to a non-compressed codec, etc…), and
- Recognize the stimuli and events which terminate sending of voice band data.

The ATA **MUST** follow ITU-T V.152 in identifying and supporting the three activities identified above. This is not a requirement to support ITU-T V.152 as that requires an associated ITU-T V.152-capable remote gateway as well as other encoding requirements.

As such, some functionality described in ITU-T V.152 may not be possible in this situation. Default values should be assumed in situations where negotiation would normally establish a value. For example, the default packetization period defined in IETF RFC 3550 **MUST** be used if not negotiable.

The preference, if negotiated and supported, is to utilize other methods of handling data in the voice stream (e.g., ITU-T T.38, TIA-1001, ITU-T V.150.1, ITU-T V.151, ITU-T V.152, IETF RFC 2833,…) over this mode of handling voice band data.

I - 228 The device **MUST** support all standard analog signaling for Distinctive Ringing. The device will provide specific distinctive ringing in response to proper messages from the Softswitch on call setup. Distinctive ringing cadences vary from country to country and region to region, throughout the world.

I - 229 The device **MUST** support all standard analog signaling for Call Waiting Tone.

I - 230 The device **MUST** meet the requirements as identified in Telecordia GR-571-CORE, LSSGR: Call Waiting (FSD 01-02-1201).

I - 231 The device **MUST** support all standard analog signaling for Calling Number and Calling Name Display.

I - 232 The device **MUST** support signaling as identified in the following:

ANSI/TIA-1063, Telecommunications - Telephony Aspects of MLTS and Packet-based Equipment, including VoIP - Analog Telephone Port Requirements for Packet-based User Premises Terminal Adapters, Section 12 (except where alternate local, national, or regional requirements apply)

Telcordia Technologies GR-416-Core, CLASS[SM] Feature: Call Waiting Deluxe, FSD 01-02-1215

I - 233 The device **SHOULD** be able to determine if there is sufficient bandwidth to establish a call.

I - 234 The device **SHOULD** be able to support changes in the characteristics of the broadband connection during a call (e.g., Internet performance changes, ADSL line synchronization speed changes, etc…).

I - 235 If a telephone number is dialed and there is too much traffic congestion in the home network to establish the call, the device **MAY** initiate an audible and/or visual indication that there is insufficient bandwidth. Such an audible signal should not be similar to audible signals currently used to indicate congestion in the PSTN.

I - 236 The device **MAY** be able to detect the dialing of a high-priority telephone number (where high-priority telephone numbers may be pre-defined) and if such a number is dialed, initiate steps to reduce local traffic congestion to the extent necessary to establish the call, if it is being blocked by local congestion.

I - 237 The device **MUST** be able to set the Ethernet user priority, identified in IEEE Std 802.1D and IEEE Std 802.1Q, of device generated traffic.

I - 238 The device **MUST** support setting the Differentiated Services Code Point (DSCP) in the IP header per IETF RFC 2474, of device generated traffic.

I - 239 The device **MUST** be able to set the Ethernet user priority in SIP packets to a configurable value.

I - 240 The device **MUST** be able to mark the Diffserv codepoint in SIP packets to a configurable value.

I - 241 The device **MUST** be able to set the Ethernet user priority in RTP packets to a configurable value.

I - 242 The device **MUST** be able to mark the Diffserv codepoint in RTP packets to a configurable value.

I - 243 The device **MUST** support setting the Ethernet VLAN Identifier (VID), defined in IEEE Std 802.1Q, of device generated traffic to a configurable value.

I - 244 The device **MUST** support enabling and disabling use of the IEEE Std 802.1Q tag header when the value of the VID is equal to the null VID (i.e., value is zero) and the user priority value is zero.

If the VID is not zero or the user priority value is not zero, then the use of the IEEE Std 802.1Q tag header can not be disabled.

5.7 Protocol Support - SIP and RTP Telephony Requirements

I - 245 The device **MUST** support IETF RFC 3261, SIP: Session Initiation Protocol, as modified by the requirements in this document.

I - 246 If the vendor supports VoIP signaling protocols other than SIP (e.g., H.323, SGCP, MEGACO, MGCP, etc.) for the device, then the device **SHOULD** support swapping out the signaling protocols without changing the hardware.

I - 247 The VoIP signaling protocol **SHOULD** be software definable and upgradeable, so service providers can select a network technology that best meet their needs.

I - 248 The device **MUST** be able to support at least two calls per line (i.e., appropriate handling of SIP messages and two or more concurrent RTP streams) as can occur with Call Waiting, Call on Hold and Three Way calling.

I - 249 The device **MAY** support device-based 3-way calling by mixing the audio streams of at least 2 separate calls.

I - 250 The device **MUST** support priority alerting, distinctive ringing and call waiting by making use of the Alert-Info header field as defined in IETF RFC 3261.

I - 251 The device **MUST** support UDP transport of SIP.

I - 252 The device **SHOULD** support IETF RFC 3310, HTTP Digest Authentication using Authentication and Key Agreement (AKA) when talking with the registration server.

I - 253 The device **MUST** support digest authentication per IETF RFC 3261.

I - 254 The device **SHOULD** support IETF RFC 3311, The Session Initiation Protocol (SIP) UPDATE Method.

I - 255 The device **MUST** support IETF RFC 3262, Reliability of Provisional Responses in SIP.

I - 256 The device **SHOULD** support IETF RFC 4028, Session Timers in the Session Initiation Protocol (SIP).

I - 257 The device **MUST** support Flow IV and Section 11 Implementation Recommendations from IETF RFC 3725, Best Current Practices for Third Party Control (3pcc) in the Session Initiation Protocol (SIP), in the sense that it may be the controlled device.

I - 258 The device **MUST** support RFC 3842, A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP).

I - 259 The device **MUST** support IETF RFC 3265, Session Initiation Protocol (SIP)-Specific Event Notification.

I - 260 The device **MUST** support IETF RFC 3515, The Session Initiation Protocol (SIP) Refer Method.

I - 261 The device **MUST** support IETF RFC 3892, The Session Initiation Protocol (SIP) Referred-By Mechanism.

I - 262 The device **MUST** support REFER and NOTIFY, as required to support transfer per IETF RFC 3515.

I - 263 The device **MUST** support escaped headers in the Refer-To: header.

I - 264 The device **MUST** support an unattended call transfer using the REFER method described in IETF RFC 3515.

I - 265 The device **SHOULD** comply with call flows which are described in the "Basic Transfer" section of draft-ietf-sipping-cc-transfer-05, Session Initiation Protocol Call Control - Transfer.

I - 266 The device **SHOULD** support attended call transfer.

I - 267 The device **SHOULD** comply with call flows which are described in the "Transfer with Consultation Hold" section of draft-ietf-sipping-cc-transfer-05.

I - 268 The device **MUST** provide a SIP response to the OPTIONS method of IETF RFC 3261. However, the device does not have to respond with a 200 OK to the options method. The device may respond with any SIP response code.

I - 269 The device **MUST** support IETF RFC 2976, The SIP INFO Method.

I - 270 When using (and only when using) a codec which does compression, the device **MUST** support DTMF named events as identified in IETF RFC 2833, RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals.

I - 271 The device **SHOULD** support coded audio packets as identified in IETF RFC 2833.

I - 272 When using IETF RFC 2833, the device **SHOULD** be able to send either named events, coded audio packets or both based on a configurable setting.

I - 273 Use of in-stream transmission of signals as is versus use of named events and coded audio packets identified in IETF RFC 2833 **MUST** be configurable.

I - 274 When the device is transmitting DTMF with IETF RFC 2833, fax or modem pass-through with a non-compressed codec (e.g., ITU-T G.711), it **SHOULD** be able to support the redundancy function defined in IETF RFC 2198, RTP Payload for Redundant Audio Data.

I - 275 Payload type negotiation **MUST** comply with IETF RFC 3264 and with the registered MIME types for RTP payload formats in IETF RFC 3555, MIME Type Registration of RTP Payload Formats.

I - 276 The dynamic payload type **MUST** remain constant throughout the session. For example, if an endpoint decides to renegotiate codecs or put the call on hold, the payload type for the re-invite **MUST** be the same as the initial payload type.

I - 277 The device **MAY** support Flow Identification as defined in IETF RFC 3388, Grouping of Media Lines in the Session Description Protocol (SDP).

I - 278 The device **MUST** support IETF RFC 2327, SDP: Session Description Protocol.

I - 279 The device **MUST** support IETF RFC 3264, An Offer/Answer Model with Session Description Protocol (SDP).

I - 280 By default, the device **MUST NOT** send RTP packets on a stream when a call is put on hold, when using call waiting, and with similar type of functionality. This **MUST** be a configurable option.

I - 281 The device **MUST** support IETF RFC 3550, RTP: A Transport Protocol for Real-Time Applications, for transport of speech packets.

I - 282 The device **MUST** support RTCP, sending RTCP traffic with the calculated data as specified in IETF RFC 3550.

I - 283 The device **SHOULD** use RTCP Extended Reports for logging and reporting on network support for voice quality per IETF RFC 3611, RTP Control Protocol Extended Reports (RTCP XR).

I - 284 The device **MUST** support IETF RFC 3551, RTP Profile for Audio and Video Conferences with Minimal Control.

I - 285 The device **MUST** support IETF RFC 3581, An Extension to SIP for Symmetric Response Routing.

I - 286 The device **MUST** support IETF RFC 3605, Real Time Control Protocol (RTCP) attribute in Session Description Protocol (SDP).

I - 287 The device **SHOULD** support IETF RFC 4145, TCP-Based Media Transport in the Session Description Protocol (SDP).

I - 288 The device **SHOULD** support IETF RFC 3489, STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs).

I - 289 If the device supports IETF RFC 3489, then the use of this feature by the device **MUST** be configurable.

I - 290 The device **SHOULD** support draft-rosenberg-midcom-turn-08, Traversal Using Relay NAT (TURN).

I - 291 If the device supports draft-rosenberg-midcom-turn-08, then the use of this feature by the device **MUST** be configurable.

I - 292 The device **SHOULD** support draft-ietf-mmusic-ice-06, Interactive Connectivity Establishment (ICE): A Methodology for Network Address Translator (NAT) Traversal for Offer/Answer Protocols.

I - 293 If the device supports draft-ietf-mmusic-ice-06, then the use of this feature by the device **MUST** be configurable.

I - 294 The device **SHOULD** support draft-ietf-sip-connect-reuse-05, Connection Reuse in the Session Initiation Protocol (SIP).

I - 295 The device **MAY** support IETF RFC 4240, Basic Network Media Services with SIP.

I - 296 The device **MUST** support a configurable digit map based on requirements in IETF RFC 3435, Media Gateway Control Protocol (MGCP), Version 1.0, Section 2.1.5. Although the digit map must be “based on” these requirements, the vendor is free to extend and modify the requirements to meet the needs and capabilities of the device. This digit map requirement applies to all signaling protocols, including SIP.

I - 297 The device **MUST** support a digit map of at least 256 characters.

I - 298 An extension to the digit map **MUST** be provided which includes the ability to strip and/or append prefix digits, and/or append suffix digits to a dialed string.

I - 299 By default, the device **MUST** recognize the use of the pound key (#) after one or more digits has been entered as termination of a dialed number.

I - 300 Be default, except as identified through the use of a digit map, the device **MUST** use five seconds as an indication that entry of a dial string has been completed.

I - 301 The device **MUST** support a First Digit Timer, Inter Digit Timer and Extra Digit Timer as identified in IETF RFC 2897, Proposal for an MGCP Advanced Audio Package.

I - 302 The device **MUST** support a digit map entry that matches a null dialed string (i.e., a timer only string).

I - 303 By default, the device **MUST** support dialing of numbers as identified in the North American Number Plan and ITU-T Recommendation E.164 (The International Public Telecommunication Numbering Plan).

I - 304 [North America] By default, the device **MUST NOT** prevent dialing of N11 codes, also known as service codes.

I - 305 [North America] By default, the device **MUST NOT** prevent dialing of Vertical Service Codes (VSCs; codes which begin with an asterisk) as identified by the administrator of the North American Numbering Plan Administrator (NANPA) or industry recognized # codes (e.g., #344).

For example, by default, the device **MUST NOT** prevent dialing of *73 (Call Forwarding Deactivation) because it recognizes *73738 (*RESET) as a sequence which resets the device to factory defaults.

I - 306 By default, device based functions accessed immediately via dialing **MUST** begin with **. This includes any access from an immediate off hook mode to enter any internal IVR but not for accessing functions once the IVR has been engaged.

I - 307 [North America] By default, if the device implements services using local VSCs, then these services **SHOULD** use the VSCs as identified by the NANPA.

I - 308 The device **MUST** allow the registration server and proxy server to be configured independently.

I - 309 The device **MAY** support local functions such as last called number redial, last call return, call hold/unhold, do not disturb, speed dial, unconditional call forwarding, call transfer, hotline, warmline, ring volume, ring mute, mute call, anonymous call reject, enabling and disabling of call waiting, etc. This means these are performed by the device and not as part of a network function.

I - 310 If local functions, such as those identified in I - 309, are supported, the device **MUST** allow for disabling these features as there may be a network implementation of the same features.

I - 311 If the device supports local enabling and disabling of call waiting, then this feature **MUST** be off by default.

I - 312 The device **MUST** support IETF RFC 4235, An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP).

I - 313 The device **SHOULD** support the call flows as shown in draft-ietf-sipping-service-examples-09, "Session Initiation Protocol Service Examples".

I - 314 The device **MUST** support IETF RFC 3911, The Session Initiation Protocol (SIP) "Join" Header.

I - 315 The device **MUST** support IETF RFC 3891, The Session Initiation Protocol (SIP) "Replaces" Header.

I - 316 The device **MUST** support IETF RFC 3326, The Reason Header Field for the Session Initiation Protocol (SIP).

I - 317 The device **MAY** support IETF RFC 3840, Indicating User Agent Capabilities in the Session Initiation Protocol (SIP).

I - 318 The device **MAY** support IETF RFC 3680, A Session Initiation Protocol (SIP) Event Package for Registrations.

I - 319 The device **MAY** support IETF RFC 3856, A Presence Event Package for the Session Initiation Protocol (SIP).

I - 320 The device **MUST** be able to parse and accept requests containing international characters encoded as UTF-8.

I - 321 The device **SHOULD** pass the current version of the SIP Forum Basic User Agent Test Suite (see http://www.sipforum.org/modules.php?name=Sections&op=viewarticle&artid=158 for details).

5.8 Security

I - 322 The device **SHOULD** support user level configuration password protection of VoIP configuration settings. This **SHOULD** be turned off by default.

I - 323 VoIP related passwords **MUST NOT** be displayed nor broadcast in any way by the device (e.g., a configuration file).

This does not apply to the transmission of encrypted passwords used for network or session authentication.

I - 324 The device **MUST** be able to password protect VoIP configuration information and administrative functions.

I - 325 Access to the device's VoIP configuration interface **MAY** be blocked by the service provider to prevent misconfiguration of the settings.

I - 326 The device **MUST** support the option to reject an incoming INVITE where the user-portion of the SIP request URI is blank or does not match a provisioned contact. This provides protection against war-dialer attacks, unwanted telemarketing and spam.

I - 327 The option to accept or reject an incoming INVITE per I - 326 **MUST** be configurable.

I - 328 When TLS is not used, the device **MUST** be able to reject an incoming INVITE when the message does not come from the proxy or proxies where the client is registered. This prevents callers from bypassing terminating call features on the proxy. For DNS SRV specified proxy addresses, the client must accept an INVITE from all of the resolved proxy IP addresses.

I - 329 The device **SHOULD** support SIP TLS per IETF RFC 2246, The TLS Protocol, Version 1.0.

I - 330 The device **SHOULD** support IETF RFC 3711, The Secure Real-time Transport Protocol (SRTP).

5.9 User Interface and Management

I - 331 Once VoIP connection parameters have been defined (e.g., registration server address, proxy server address, username, password, etc.) then these **MUST** be used to establish a session after initial specification and upon power-up.

I - 332 The device **MUST** provide a WAN-side configuration and reporting mechanism as defined in DSL Forum TR-069 and TR-104.

I - 333 By default, the device **SHOULD** be configurable via a phone attached to the FXS port. At a minimum, this **SHOULD** include configuration of the IP address of a configuration server.

I - 334 When configuring by using a phone attached to an FXS port, the device **MAY** be configurable using voice prompts.

I - 335 The device **SHOULD** provide, using voice response, status settings upon entry of a dial string from a phone attached to an FXS port. At a minimum, this **SHOULD** include the IP address of the device, the subnet mask and the gateway IP address.

I - 336 The dial strings to access configuration and status setting from a phone attached to an FXS port **MUST** be configurable.

I - 337 When a phone attached to an FXS port goes off-hook, the device **MAY** automatically provide system status to that phone if the network or device is in a state that would prevent network access (e.g., tone or voice announcement indicating IP address could not be obtained or registration failed).

I - 338 If the device uses TFTP, then it **MUST** do so securely following draft-rescorla-dtls-05.txt, Datagram Transport Layer Security.

I - 339 The device **MUST** have diagnostics tools that allow the user to identify the precise nature of any connection or performance problem. It **MUST** be able to indicate if the problem is at the IP layer or VoIP service layer and the specific problem (e.g., unable to obtain an IP address using DHCP, failing to find registration server, etc.). These tools **MUST** be accessible from the GUI.

I - 340 The device **MUST** provide detailed information for current connections and associated parameters including SIP proxy server, configured line id, displayed line id and current codec in use.

I - 341 The device **MUST** support restarting the connection to the SIP proxy server (all layers) via the GUI and through the use of a dial access code.

I - 342 The device **MUST** follow all standards required to perform an orderly tear down of the associated connections involved at the associated network levels (e.g., issue a SIP BYE, DHCP Release) and then restart the connections.

I - 343 The device **MUST** support remote testing, remote diagnostics, performance monitoring, surveillance information access and other information access.

I - 344 The device **MUST** have a software mechanism by which the user can reset it to default factory settings through the GUI interface and through a phone attached to the POTS port.

I - 345 The device **SHOULD** support a configurable language setting.

I - 346 If the language is set, the associated language **SHOULD** be used by any internal GUI or IVR.

I - 347 The configurable language setting **SHOULD** be able to be set and queried using the mechanism identified in I - 332.

5.10 Logs

I - 348 The device **MUST** maintain an internal log of VoIP service connection flows (e.g., SIP & RTP sessions). At a minimum, the log **MUST** record the last 250 events. This will include connection and registration events initiated by the device or from the WAN side connections. The purpose of the log is to provide a trouble shooting aid in resolving line and connection problems.

5.11 Packaging

I - 349 Supplied phone cables for use with FXS or FXO ports **MUST** be colored Gray using the color identified in I - 136.

I - 350 If the device has a FXO connection, a phone cable with two pairs and RJ-11 endpoints **MUST** be packaged with the product to connect the device to an associated wall jack. The cable **MUST** have a minimum length of 6 feet. The endpoints **MUST** meet the specifications for a miniature 6-position plug in TIA-968-A.

I - 351 If the device has a FXO port, the phone cable **SHOULD** be Category 3 (CAT3) or better and have a length of 4.3 meters (or 14 feet).