New Zealand Telecommunications Forum

SIP ATA Standard for LFC Wholesale Service (Loose Coupling)

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Contents

1. Scope and Purpose

1.1 Introduction

This document is an output of the TCF VoIP Working party and aims to help define the wholesale standard for VoIP access services using the UFB Voice Access Service.

This version of the document focuses specifically on SIP Loose Coupling implementations.

This document should be read in conjunction with the “ATA Voice Service” Service description available from each LFC.

1.2 Document Purpose

The purpose of this document is to:

- Define an architecture and specify the requirements to enable a Voice Service Provider (VSP) to offer a telephony service to customers served by an analogue telephone adaptor (ATA) using the UFB network;

- To specify the interface between the VoIP telephony Service Provider and VoIP Telephony Access Provider based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) to enable an IP based telephony service and further to describe the base set of configurations required in order to carry the base services across it.

- To facilitate the Voice Service Provider to easily develop consistent services across Local Fibre Companies allowing identical end user experiences.

1.3 Scope

1.3.1 In scope

This document is relevant to LFCs only and is used as an input to VSP services.

1.3.2 Out of Scope

The following items are viewed as out of scope within this document

- VSP services using Telephony
- Definition of TCF ATA Voice Service
- Definition of Handover Connection Services
- Service Provisioning Methods and Parameters (addressed in Business Interaction Framework Working group)
- Case Studies

Note : The Service Provisioning methods are defined within the OSS/BSS documentation which can be found at http://www.tcf.org.nz/ossbssstandards

1.3.3 Normative Documents

The following documents must be read in conjunction with this code:

- LFC's Service Description for ATA Voice
- LFC's Service Description for Baseband
- TCF Ethernet Access Service Description (http://www.tcf.org.nz/content/433bebfa-5dc8-4bf3-a408-5c43cf886c38.cmr) Under review by the TCF 2014 Q3

The LFC Service descriptions are part of the LFC/Chorus's agreement with Crown Fibre Holdings (CFH) and take precedence over this document. These documents are published on each LFC's and Chorus's websites.

The UFB Baseband service is an A-EVPL service for use when there is no data Bitstream service (e.g. BS2, BS3, BS3a) implemented to the ONT.

The UFB ATA Voice Service is the SIP User Agent that supplements either the data based Bitstream service or the UFB Baseband service.

1.3.4 Conformance to the Standard

LFC's Conformance to this standard and differences with the standard will be highlighted in the individual LFC compliance documentation.

Each LFC publishes documentation relating to this standard on the Web as follows:

- Chorus
- Enable Networks
- Northpower Fibre
- Ultrafast Fibre: http://www.ultrafastfibre.co.nz/rsps/publications-resources-and-tools/user-guide

1.4 Standards Glossary

This document will make use of the following standards:

Australia/New Zealand standards

- AS/NZ 60950-1 *"Information technology equipment - Safety - General requirements"*

1.4.1 ETI/NICC

- NICC ND 183 043 V1.1.1 (2006-05) *"Telecommunications and Internet Converged Services and Protocols for Advanced Networks (TISPAN); IMS-based PSTN/ISDN Emulation Stage 3 specification"*

1.4.2 International Electrotechnical Commission (IEC)

- IEC/EN 60950-1 *"Information technology equipment – Safety – Part 1: General requirements"*
- IEC/EN 61000 series *"Electromagnetic compatibility (EMC) – Part 6-1: Generic standards – Immunity for residential, commercial and light-industrial environments"*

1.4.3 ITU

- E.180) (03/1998) *"Various Tones used in National Networks"*
- G.122 *"Influence of national systems on stability and talker echo in international connections"*
- G.161 (06/12) *"Interaction aspects of signal processing network equipment"*
- G.168 (02/12) *"Digital network echo cancellers"*
- G.711 *"Pulse code modulation (PCM) of voice frequencies"*
- G.729 *"Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)"*
- Q.23 (11/88) "*Technical features of push-button telephone sets"*
- Q.24 (11/8) "*Multifrequency push-button signal reception"*
- T.38 (09/10) *"Procedures for real-time Group 3 facsimile communication over IP networks"*
- V.150 (01/03) *"Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs"*
- V.152 "*Procedures for supporting voice-band data over IP networks"*

1.4.4 Broadband Forum

- TR 122 Issue 1 Amendment 1 (Nov 2006) *"Base Requirements for Consumer-Oriented Analog Terminal Adapter Functionality"*

1.4.5 Internet Engineering Task Force (IETF)

- RFC 2833 "RTP Payload for DTMF digits, Telephony tone and Telephony signals"
- RFC 3398 "Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping"
- RFC 3550 *"*A transport Protocol for Real-time Applications"
- RFC 4033 "DNS Security Introduction and Requirements"
- RFC 4497 "Interworking between the Session Initiation Protocol (SIP) and QSIG"
- RFC 6840 "Clarifications and Implementation Notes for DNS Security (DNSSEC)"

1.4.6 Telecordia

- GR-30 "LSSGR Voiceband Data Transmission Interface, Section 6.6"

- GR-1401 “Visual Message Waiting Indicator Generic Requirements (FSD 01-02-2000)

1.4.7 Telecom NZ

- TNA 102 “1996 Telecom Public Switched Telephone Network (PSTN) Analogue Line Interface” - plus amendments to sections 10 (Analogue On-Hook Data Transmission); and a new section 12 (Analogue Calling Line Identification Presentation)
- TNA 151 July 1996 “Telecom Telephone Network Transmission Plan”
- PTC 107 “PABX External Port Interface Requirements”
- PTC 200 “Requirements for Connection of Customer Equipment to Analogue Lines may 2006 (minor amendments September 2006)
- PTC 331 “Telecom Network Interconnection using ITU-T no.7 Signalling Part C ISUP Specification Recommendation Q.763 - Message and Parameter Formats and Codes and Recommendation Q.764 - Signalling Procedures “ (http://www.telepermit.co.nz/PTC331%202012%20Part%20C.pdf)

1.4.8 New Zealand Telecomunications Forum (TCF)

- Code for Emergency Voice Calling Services.

1.4.9 Telecommunications Industry Association

- TIA-968-A “Telecommunications. Telephone Terminal Equipment. Technical Requirements for Connection of Terminal Equipment to the Telephone Network

1.4.10 NAD - Number Administration Deed

- Telecommunications Numbering Plan - Number Allocation Rules; Link is: http://www.nad.org.nz/assets/Deed-and-Rules/NAD-Rules-v6.1.pdf
- Number Register; Link is: http://nad.org.nz/number-register/

1.5 Acknowledgement

Thanks is given to Chorus NZ for providing the template for this specification.

1.6 Requirements Identification

Where requirements are identified, they are classified using the following:

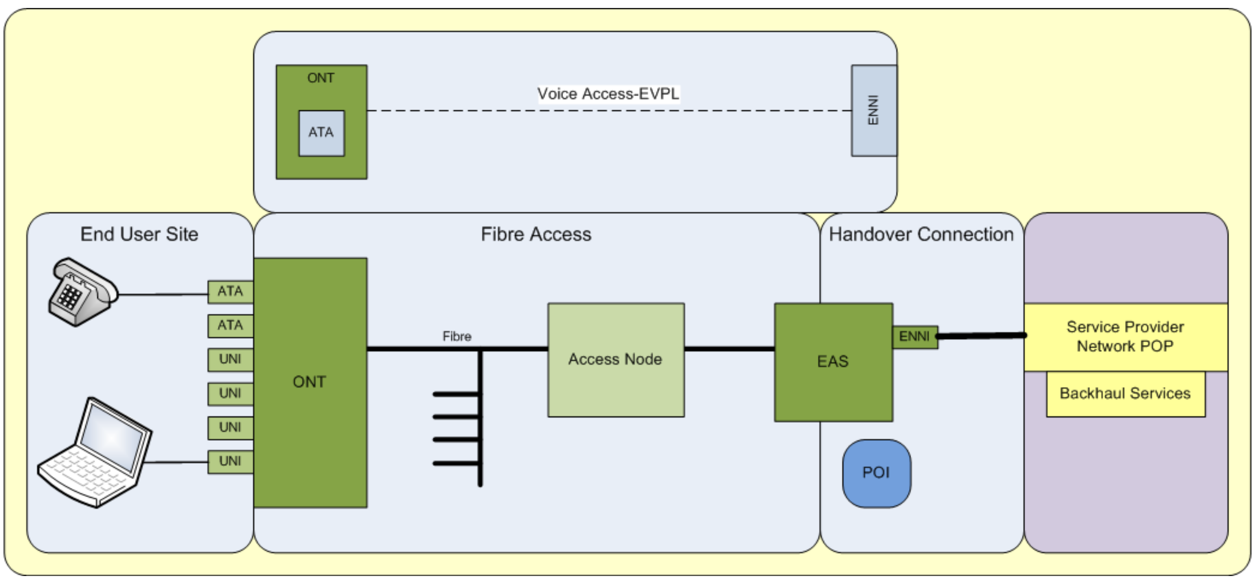
- **Must** have: requirements that are fundamental to the service. The service cannot be implemented without them.

- **May** have: requirements that can be omitted if the LFC requires for commercial or technical reasons. This may include requirements that, while valuable, do not need to be implemented immediately.
- **Must Not** have: Requirements that are prohibited from being included.

2. Specification of Service Connection

ATA Voice Service and Baseband is an intermediate input service which the Voice Service Provider can combine with its own network or other products to provide an analogue voice service to their customers. The analogue voice service at the end user premises is converted to a SIP-controlled VoIP service and delivered to the Service Provider at a UFB POI, as shown below:

Figure 1: ATA Voice and Baseband solution



2.1 Network Topology

ATA: The analogue Terminal Adapter (ATA) provides the ability for End Users to use analogue telephones and telephony services on a GPON-based Bitstream access. Specifically, it converts an analogue PSTN-compatible 2-wire voice band call into a Bitstream at the ONT, and converts a Bitstream received from the Service Provider into an analogue PSTN-compatible 2-wire voice band call at the ONT.

The Access - EVPL service: The Access Ethernet Virtual Private Line service provides the underlying connectivity for carrying the VoIP packets between the ATA port on the ONT and the 'External Network to Network Interface' (E-NNI) port where the Service Provider interconnects with the service.

SIP User Agent: The SIP User Agent located in the ONT interacts with the Service Provider soft switch (Applications Server (AS)) to manage the ATA Port‘s voice functions. The SIP User Agent is configured using parameters provided by the VSP and interacts with the AS using Session Initiation Protocol (SIP).

2.2 Physical Design

The physical design of the ONT deployed will vary per LFC. A minimum of one RJ11 analogue FXS port must be provided.

The ONT will connect to the GPON (Gigabit Passive Optical Network) service with a single optical uplink connection.

The voice traffic will be carried over the PON (Passive Optical Network) on the ONT optical uplink.

There will be an External Network to Network Interface (E-NNI) at the POI. This E-NNI can be carrying traffic for other Voice or Bitstream services.

The POI supports a coverage area of the LFC network, and hence a Service provider will require multiple E-NNIs if they offer service in different coverage areas.

2.3 Voice Network Conditions

Voice traffic will be carried as a unique VLAN (Access EVPL) from the ONT through to the Handover port (E-NNI), as per TCF Ethernet Access Service Description.

2.4 Logical Service Design

The ATA voice and Baseband service is a 1:1 VLAN service construct between the Service Provider handover and the end-user, i.e. all traffic must traverse the E-NNI.

The key service definitions are as follows:

- ATA Voice and baseband services are defined as having a single high-priority Class-of-Service.
- A minimum of one SIP User Agent (UA) will be provided per ONT
- At the E-NNI, there will be a single SVLAN:CVLAN per SIP User Agent (ONT).
- ATA voice and Baseband service has a maximum MTU of 2000 (inclusive of CVLAN tag).
- Encryption is not required.
- PCP marking for Voice to be in the high Priority Queue as appropriate (PCP 5).
- The Operator Virtual Circuit between the ATA and the Handover connection must comply with the TCF Ethernet Access Standard.

The Service Provider is responsible for providing the IP addressing and any DHCP and/or DNS services that may be required.

The Remote ID and TR-156 Circuit ID can be inserted into the option 82 and can be used as an unique service identification, i.e. validate the CPE trying to connect to the network. It is recommended that the Remote ID is used as this ID is more operationally stable.

RTP media flows and RTCP signalling between two or more ATA voice end points must be routed via a Service Provider router. Layer 2 traffic that is bridged between two ATA voice end points will be dropped.

2.4.1 DNS Security

To protect against malicious DNS attacks, DNSSEC (DNS Security certificates) are available to authenticate the information provided by the DNS.

Use of a DNS by the ATA Voice service is configurable and defined by the VSP.

The ATA Voice Service would implement DNSSEC.

While out of scope of this document, it is highly recommended that VSPs who provide a SIP service use DNSSEC as soon as possible, in accordance with international and New Zealand best practice.

2.4.2 UA Client Identification

During provisioning the VSP must supply for each instance of ATA Voice Service, as a minimum:

- Username
- Password
- DHCP connection information or IP address details
- Domain

Note: information may be supplied by the VSP using direct ATA provisioning methods (for example FTP/TFTP file transfer or TR-069)

2.4.3 URI definition

The ATA Voice Service must support the use of a Uniform Resource Identifier as defined in RFC 3966 and are as follows:

- tel:+1-201-555-0123: This URI points to a phone number in the United States. The hyphens are included to make the number more human readable; they separate country, area code and subscriber number.

- tel:7042;phone-context=example.com: The URI describes a local phone number valid within the context "example.com".

- tel:863-1234;phone-context=+1-914-555: The URI describes a local phone number that is valid within a particular phone prefix.

2.5 Emergency Services

The SIP ATA must support the VSP's commitment to the TCF Emergency Voice Calling Services Code. The TCF's home page for the Code is:
http://www.tcf.org.nz/es

The Code itself (formally known as the Code for Emergency Voice Calling Services is:
http://www.tcf.org.nz/library/a4a3ad46-2e90-42f3-8733-a66f7bd1065c.cmr

The Code is technology/network neutral and it covers the general technical and operational requirements for providing a high quality, robust, and ubiquitous access to 111 services within the NZ PSTN. Among other things, the Code specifically covers:

- Availability and Quality of Emergency Calls
- Emergency Calling
- Handling of Emergency Calls
- Caller Information
- Customer Information Standards
- Customer Complaints

Parties to the Code are obligated to comply with the Code if they wish to describe their network as providing a "Code Standard Voice Service". Compliance certification to the Code is required annually.

The following information is provided to assist VSP's understanding of how the ATA Voice Service handles emergency calls:

- The ATA or the LFC network does not detect an emergency call and so does not handle these calls any differently to other calls.
- The VSP is responsible for defining within the digit map all the short codes required to call the emergency service (e.g. 111, 911 etc.) that are handled by the VSP's network.
- All calls are sent to the VSP's SIP proxy as defined within the ATA Voice Service configuration. All subsequent routing is the responsibility of the VSP.
- Where the VSP wishes to impose call limiting on a service, the LFC is unable to limit the calls made from the ATA Voice Service when the service is operational. Any call limiting, including any call destinations which are limited, is the responsibility of the VSP.

- When the ATA Voice service is terminated with the LFC, no calls will be able to made or received via the ATA Voice Service. This includes the implication that no emergency calls will be able to be made from the ATA.
- All ATA Voice Service traffic is carried at the same Class of Service. The LFC undertakes to keep the frame loss, frame delay and frame delay variation within the published SLA.
- Any traffic congestion occurring at the Handover Connection, being the demarcation between the LFC and the VSP, is the responsibility of the VSP.
- The LFC will make available facilities (e.g. LAG) for multiple/redundant Handover Connections between the VSP and the LFC. The VSP must purchase the level of resiliency required to support the grade of voice service provided by the LFC.
- The ATA Voice Service is not responsible for maintaining any location information required to be provided to the Emergency Services.
- The ATA Voice Service does not prevent the End user from terminating the call by going on-hook. That is, there is no special treatment of how a BYE method is handled within the ATA for an emergency call.

2.6 Loose and Tight Coupling

The difference between loose and tight coupling is around how features are handled that require a switch hook flash enabler.

In **Tight Coupling**, the SIP server has full control over features - The ATA does not implement any service-independent logic on detection of a specific call event.

In **Loose Coupling**, the ATA implements independent feature logic for dealing with the call event.

For the purposes of this document, loose and tight coupling are used as they are defined in ETSI TS 183 043 V2.3.1:

Loose coupling: on-hook and flash-hook are analysed in the Access Gateway Control Function/VoIP Gateway; much like a simulation endpoint would operate.

Tight coupling: on-hook and flash-hook are interpreted by the Application Server.

This document is based on SIP loose coupling between the ATA Voice Service and VSP AS.

3. Safety Standards Compliance

This section describes the ATA Voice Service compliance with safety standards.

3.1 Equipment safety standard compliance

The ATA complies with the following safety standards:

- IEC/EN 60950-1 *“Information technology equipment – Safety – Part 1: General requirements”*
- AS/NZ 60950-1 *“Information technology equipment - Safety - General requirements”*

3.2 EMC compliance

The ATA complies with the following Electro-Magnetic Compatibility and related standards:

- IEC/EN 61000 series *“Electromagnetic compatibility (EMC) – Part 6-1: Generic standards – Immunity for residential, commercial and light-industrial environments”*

4. ATA and VSP Supplied Services

The total range of services available to customers (generally and collectively known as Supplementary Services), can be categorised into ATA Voice Service Supplied Services, and VSP Provided Services, depending on the method of the implementation of each service (particularly; whether Tight or Loose SIP coupling is used). However, at this stage, the following tables of services list and describe the expected total range of services that service can be offered; and later, each service will be designated as either ATA Voice Service Supplied; or VSP Supplied.

(a) CUSTOMER SERVICES:

Customer Service	Description	Comment
Standard Calling	Basic POTS service	
Conferencing; - National - International - Three-way - Multi-party	Audio-conferencing between three or more parties.	
Call Diversion; - Immediate - Non Answer - Busy	Manages and activates the diversion of incoming calls to another number.	
Call Transfer	The transfer of an existing call (in the speech state) to another line, or to an Operator.	Mainly a CTX/PBX feature. Not a current PSTN feature. Maybe so in the future.
Call Deflection	Similar to Call Transfer, but the call is transferred to the other line during the ringing state.	Mainly a CTX/PBX feature. Not a current PSTN feature. Maybe so in the future.
Hot Line; - Immediate - Delayed	Automatic call setup on off-hook, to a pre-set number	
Call Restriction/Barring	Managing and blocking of chargeable calls; or of call types	

Caller Display; - Calling Number - Calling Name	Displays the callers number/name (aka CLIP)	
Number Withhold; - Per call - Automatic - Override	Manages the caller's number/name display/withhold choice (aka CLIR)	
Voice Messaging; - Message Waiting - audible - Message Waiting - Visual	Diverts incoming calls to your voice mailbox	
Distinctive Ringing	Different ringing cadences to a line; based on the specific number dialled	
Multiple Number; - Dual Number - Multiple Number	Multiple numbers for the same line; eg; if the main number has Call Minder/Call Diversion/Do not Disturb activated; the call will proceed if the other number is dialled.	
Call Waiting	Alerts you to an incoming call, if you are already on a call.	
Call Track - per caller billing	Tracks calls from your phone - to enable billing to the actual caller	
PBX	PBX calling and services	Small scale and limited capabilities
Code Access	Alternate Service Provider (for Toll services) accessed by a prefix code (Regulatory requirement)	
Non-Code Access	Alternate Service Provider (for Toll services) accessed by a pre-defined (non-dialled) prefix code (Regulatory requirement)	

Number Portability; - Local - Mobile - Freephone	Number Portability between Service providers (Regulatory requirement)	
Payphone	Payphone services	
Call Hold	Enables an existing call in the speech state, to be held (with no speech), while another call is set up; or the CPE is transferred to another outlet in the premises, and speech is then re-established.	Mainly a CTX/PBX feature. Not a current PSTN feature. Maybe so in the future.

Table 2: Customer Services

(b) SUPPLEMENTARY FEATURE CODES AND NUMBERING

Supplementary features generally fall into two categories - viz:

- Accessible services that are permanently available to all customers in a network or across all networks, without requiring the customer to subscribe to them, eg 111 (EMG); 018 (Directory Service); 196 (Number Withhold) etc. These are also generally known as Special Services.

- Services that a customer needs to subscribe to, to get access to them, eg Call Diversion; Call Waiting; WakeUp etc. Usually, the numbers used to activate/use these services are unique to each Service Provider, and are only used within that Service Provider's network. For example, Spark tends to use codes such as 16X(Y) and 18X(Y) for such services. In general, it is these sorts of services that fall into the category of Supplementary Features.

A common feature of the codes used for these services, is that the codes are typically 3 to 5 digits in length - hence the generic term Short Codes is sometimes used.

Many of the codes currently used for these services come about for historical reasons - the services were implemented long ago and, in particular, implemented prior to the formation of the NAD and the NAD's Numbering Rules. Some of the current usages of these pre-existing codes do not comply with the Rules, but are allowed to continue to be used on the basis they were allocated and used in good faith, under the conditions operating at that time.

The NAD’s Numbering Allocation Rules provide for two specific groups of short codes, in the context of the overall Service Category of Special Services Codes. The two groups are:

- 01XY: generally used for services that may invoke call charges
- 1XYZ: generally used for accessing and controlling Supplementary Services

The NAD Number Allocation Rules has conditions for the use of these Special Service Codes - viz:

- codes can only be allocated at the 1XYZ+ level; and may not be used at less than the 1XYZ level - eg; 1867 is OK; 186 is not.
- codes can be used at longer number lengths - eg; the above 1867 could be used as 1867X, which could enable 10 different services - each with different value of X.

Special Service Codes are a limited resource, and should be used carefully - the shorter the Code, the greater the consumption of the number space. Large, important services may justify a 4 digit code; whereas a minor service should use a code of 5 digits or greater.

(c) NETWORK SERVICES:

Network Service	Description	Comment
EMG (111) Calling	ENG (111) (and other EMG codes such as 911; etc) calling and services in the PSTN. Covered by the TCF Emergency Calling Code	Semi regulatory requirement. Manual Hold of EMG calls is not a regulatory requirement, and it is not implemented in the PSTN.
Network Interconnection	Calling to/from/between alternate networks	Regulatory requirement
Lawful Intercept	Monitoring and collection of call data, on request from an approved body (Regulatory requirement)	Regulatory requirement.
ENUM; - User ENUM - Carrier ENUM	Call control and call routing management	
Geographic Based	Call routeing based on the geographic	

Network Service	Description	Comment
Routeing	location of the caller. Esp. for 111 (EMG) and Freephone calling. Also for general routeing based on the geographic location of the caller/called.	
Billing	Specifically, for inter-party peering billing and billing reconciliation	
Malicious Call Tracing	The detection; tracing; holding; and alerting (by Hookflash) of malicious/nuisance calls	Alerting by Hookflash is not a requirement; but may be implemented.

Table 3: Network Services

(d) ENABLERS:

Enabler	Description	Comment
Switchhook Flash	Customer activation for specific “in-call” services	See section 4.7
CLR FWD	LIne signalling for PBX call control	See section 4.12
ANS REV	Line signalling for Pay-Phones and PBX calling and billing	See section 4.13
Codec Negotiation	Resolution of CODEC choices	See section 4.2
RNG TO		See section 4.1.1
B Party Hold	For 111 (EMG) calls and Malicious Call Tracing	See section 4(c) for EMG (111) Calling; and Malicious Calling.
Short Codes		See section 4(b)
DTMF		See section 4.5
Geo Location	For geographic-based call routeing	See section 4(c)
Identity Delivery	CLIP/CLIR	See section 4.11.3

Enabler	Description	Comment
Ringing Cadences		See section 4.9.1
Database Lookup	External database for routeing; Number Portability; ENUM; etc	
High & Dry	Line lock-out and alerting under permanent seizure conditions.	High and Dry is not a requirement; but may be implemented.
DigiMap		See section 4.19.1

Table 4: Enablers

4.1 Supervisory Tones

The supervisory tones described in this section are based on those currently used in the NZ PSTN network[1].

For clarification, this section covers tones provided by the ATA Voice Service and the network, but not tones provided by devices connecting to it.

"Supervisory Tones" includes all tones delivered to customers as information during the various events of call set-up, congestion and call termination. Certain tones described are dependent upon the customer subscribing to a service (e.g. Call waiting).

The ATA Voice Service may provide a supervisory timeout for playing the supervisory tone.

4.1.1 Detail of Supervisory Tones

The ATA Voice Service must provide the following supervisory tones[2]. These tones are published in E.180. The timeout values are supported and are typical across the industry, but are not mandatory.

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[1] TNA 102 § 7 Supervisory Signals
[2] TNA 102 § 7.1 Supervisory Tones

Tone Name	Abbrev	Frequency	Output Level (dBm)	Cadence	Provided by
Busy Tone	BT	400 Hz	-15	500 ms on 500 ms off Sequence repeated until timeout (40+ secs)	ATA
Dial Tone	DT	400 Hz	-15	Continuous until timeout (15+ secs)	ATA
Dial Tone (with Message Waiting Indication) (also known as Stutter Dial Tone or Message Waiting Tone)	MWT	400 Hz	-15	100 ms on, 100 ms off, repeated for 2.5 secs, then continuous until timeout (15+ secs)	ATA
Call Waiting Tone	CWT	400 Hz	-15	200 ms on, 3 sec. off, 200 ms on, 3 sec. off, 200 ms on, 3 sec. off, 200 ms on, not repeated	ATA
Disconnect Tone (also known as Congestion Tone, or Reorder Tone)	DSCT	400 Hz or 900 Hz	-15	250 ms on, 250 ms off, repeated until timeout (40+ secs)	ATA
Number Unobtainable Tone	NUT	400 Hz	-15	75 ms on, 100 ms off, 75 ms on, 100 ms off, 75 ms on, 100 ms off, 75 ms on, 400 ms off, repeated until timeout (40+ secs)	ATA

Tone Name	Abbrev	Frequency	Output Level (dBm)	Cadence	Provided by
Ringing Tone (also known as Ringing Back Tone)	RBT	400 Hz + 450 Hz	-18	400 ms on 200 ms off 400 ms on 2.0 sec off repeated until timeout (4+mins) (same as Ringing timeout)	ATA
Switching Complete Tone	SCT	400Hz + 450 Hz	-18	200 ms on, 400 ms off, 2 sec. on, 400 ms off, repeated until timeout (15+ secs)	AS
Recall Dial Tone (also known as Transfer Dial Tone or 3 Short tones)	RDT	400 Hz	-15	100ms on, 100 ms off, 100ms on, 100 ms off, 100ms on, 100 ms off, Sequence followed by continuous DT until timeout (15+ secs)	ATA
Call Holding Tone	CHT	400Hz	-15	500ms on 500s off; then 400Hz+450Hz 500ms on 2.5sec off repeated	ATA

Table 5: Supervisory Tones

Notes:

(a) The tolerances on frequencies, levels and cadences of supervisory tones are as follows:

- Frequencies maintained within ± 5% of nominal values.
- The output level of tones as specified at the zero transmission reference point ±[5%]

- Cadences maintained within ± 10 % of nominal values.

(b) There are no formal standards covering the timeout durations for tones. Alignment with the above timings is recommended.

(c) If a UA is not Registered (or, a customer's service has been relinquished), no tone should be played.

(d) If a customer's service has been suspended (eg for credit control purposes, or to limit disruptive calling) the tone played is up to the VSP.

(e) The use of a Howler tone is not required, but it can be implemented.

4.1.2 SIP Response to tone mapping

The ATA Voice Service provides a fixed mechanism for mapping SIP responses received from the AS to the relevant supervisory tone.

For the NZ PSTN, the current mapping between "network events" and the resultant tones is as described in PTC 331 - Telephone Network Interconnection using ITU-T No. 7 Signalling PART C: ISUP SPECIFICATION; table 1; page C-36. In this specification, the "network events" are the various ISUP Cause Values, and hence relate only indirectly to the SIP Responses.

The following table is a summary of the basic relationships between some generic call events and tones:

- Customer Idle: RBT
- Customer Busy: BT
- Unallocated Customer Number: NUT
- No Routing for the Code: NUT
- Call Event Timeout: DSCT
- Call Resource Blocked/Unavailable: DSCT
- Invalid or Errored Call Data Encountered: DSCT
- Special Service is Currently Active: SCT
- Special Service Invocation Accepted: SCT
- Other - generally the default is: DSCT

For SIP, the direct mapping between SIP Response Messages and the consequent tones appears to be undocumented. Two RFC's do, however, provide some insight. Both RFC's describe the linkages between SIP Responses and ISUP Cause Values - and hence indirectly they do enable linkages between SIP Responses and Tones to be derived. The two RFC's are:

- RFC3398; ISUP to SIP Mapping; and

- RCF4497; Interworking Between SIP and QSIG (ISUP)

On this basis, the following is recommended:

SIP Response Message Received by ATA	Message Description	Tone Played
180	Ringing	RBT
183	Session in progress	Early Media from AS.
400	Bad Request	DSCT
402	Payment Required	DSCT or Early Media from AS.
403	Forbidden	DSCT
404	Not Found (user not found)	NUT
405	Method not Allowed	NUT
406	Not Acceptable	NUT
410	Gone	NUT
423	Interval Too Brief	NUT
433	Anonymity Disallowed	DSCT
480	Temporarily Unavailable	DSCT
484	Address Incomplete	DSCT
485	Ambiguous	NUT
486	Busy Here	BT
500	Server Internal Error	DSCT
503	Server Timeout	DSCT
600	Busy Everywhere	DSCT
603	Decline	DSCT
604	Does not exist anywhere	NUT

SIP Response Message Received by ATA	Message Description	Tone Played
606	Not Acceptable	DSCT
	Any Other Response Code	DSCT

Table 6: SIP Response to Tones mapping

Note: where a specific SIP Response appears to have no related tone, DSCT has been assumed to be applicable.

4.2 Codec

The ATA Voice Service must support the following codecs:

- ITU G.711A-Law (Current NZ PSTN standard)
- ITU G.711μ-Law
- ITU G.729 Annex A[3]

The ATA Voice Service may support the following additional codecs:

- ITU G.729 Annex B[4]
- Any other codec supported by the LFC equipment Vendor.

The LFC must publish in their compliance statement to this code, which codecs they support.

The ATA Voice Service must support, for each applicable codec:

- Voice Activity Detection (VAD).
- Comfort Noise Generation (CNG).
- Packet Loss Concealment (PLC).

Support for faxes is described in section 4.3.

When the ATA Voice Service and the remote UA cannot establish a common codec, the call setup must fail.

[3] TR 122 § I - 178
[4] TR 122 § I - 179

CODEC	Bit Rate(kbps)	Sampling Period (ms)	Packet Size (Bytes)
G.711a/u	64	10	134
G.711a/u	64	20	214
G.729 Annex A	8	10	10

Table 7: Codec parameters

4.2.1 Codec Packetisation

The ATA Voice Service must support a codec packetisation rate of 10 ms and 20 ms. It may support additional packetisation rates.

4.2.2 Codec Prioritisation

Codec prioritisation is where each UA specifies in decreasing order of desirability, the codecs that it supports. The desirability of a particular codec may depend on whether it is a voice only service and can utilise a low bit rate codec or it supports faxes and therefore requires a more accurate codec.

The ATA Voice Service must provide a facility to order the codecs[5] offered during the call setup dialogue in decreasing order of priority.

Where an ATA Voice Service provides more than one POTS port, each POTS port must be able to be prioritised independently of the other POTS port[6].

4.2.3 Jitter Buffer

A jitter buffer is provided on the receiver side to compensate for variable frame delay (jitter) and out of order packets. Further guidance can be found at ITU G.1021 “Buffer models for development of client performance metrics”.

The ATA must provide a dynamic jitter buffer for voice calls with a minimum size of 20 ms.

The ATA Voice Service may support a jitter buffer up to 150 ms[7].

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[5] TR 122 § I - 191
[6] TR 122 § I - 189
[7] TR 122 § I - 202

4.2.4 Codec Transport

The codec data must be transported using the Real-time Transport Protocol (RTP) as described in RFC 3550.

4.2.5 Voice Band Data

ATA Voice Service supports G.711 transparent pass through mode of operation that will be used for Voice Band Data (VBD) communication and which has the following characteristics:

- Echo cancellers and non linear processors are switched off as per ITU-T G.168 and G.161
- Uses fixed jitter buffer (not adaptive) and auto selected on detection of VBD. The fixed jitter buffer size is 100 ms.

Note:

- Voice Activity Detections (VAD) and Comfort Noise Generation (CNG) are turned off.
- Packet Loss Concealment (PLC) algorithms are turned on for voice calls and off for VBD calls. Refer to ITU-T Recommendation G.113.

4.2.6 Transcoding

The ATA Voice Service is unable to perform transcoding as it used as an end point. The appropriate codec should be selected during the call setup or if the desired codec is not available, the VSP must undertake any transcoding required.

4.3 FAX Support

The ATA Voice Service supports faxes calls when a fax device is connected via the ONT.

Support for fax calls is limited to data speeds up to and including 14.4 kbit/s. Operating at higher speeds may be possible, but is not supported.

- The service supports end-to-end fax services using the VBD triggers identified below
- The service supports Voice band data as specified in ITU recommendation G.711 using A-law transparent mode. T.38 may be supported with fall back to G.711 A-law.

4.4 Modem Standards

The ATA Voice Service analogue voice interface supports modem calls when a modem device is connected via the ONT.

Support for modem calls is limited to data speeds up to and including 14.4 kbit/s[8]. Operating at higher speeds may be possible, but is not supported.

The ATA Voice Service may support end-to-end modem services as per ITU V.152 non-assured VBD mode[9].

The ATA Voice Service may support V.150 (modem-over-IP networks)[10]

4.4.1 Modem Data

The ATA Voice Service must support the handling of modem data[11] for the pass through of facsimile, modem and text telephony.

.

The Service must support the following stimuli for invoking Voice Baseband Data (VBD) mode of operation (note that the VBD mode of operation disables the Non Linear Processor in the echo canceller (amongst other things)). However, it does not disable the linear processing part of the echo canceller. This is further detailed in ITU V.152 § 9.

VBD Stimuli	Protocols / Signals
980Hz	V.21L Mark, Edt
1100Hz	T.30 CNG, CED?
1270Hz	Bell103
1650,1850 Hz	V.21 flag
2100Hz	ANS
2100Hz with phase reversal	/ANS
2100Hz with amplitude modulation	ANSam
2100Hz with amplitude modulation and phase reversal	/ANSam
2225Hz single tone	Belltone
2250Hz	V.22 USB1

[8] TR - 122 § I - 223
[9] TR - 122 § I - 226
[10] TR - 122 § I - 222
[11] TR - 122 § I - 227

Table 9: VDB Stimuli

The stimuli that completely disable the echo canceller are listed below.

The ATA Voice Service must support the following stimuli for disabling echo cancellers/suppressors.

Echo Canceller Disabling Stimuli	Protocols / Signals
2100Hz with phase reversal	/ANS
2100Hz with amplitude modulation and phase reversal	/ANSam

Table 10: Echo Canceller disablement stimuli

4.5 Dual Tone Multi Frequency (DTMF)

All DTMF signalling will be exchanged between the ATA POTS port and the End User CPE.

RFC 2833 may be used if agreed as part of the VSP onboarding process with the LFC.

DTMF Tones as used in the interaction between customers (CPE) and ATA Voice Service comply with ITU-T Recommendation Q.23 and Q.24, and are applied as follows:

DTMF Tones				
Low Group (Hz)	High Group (Hz)			
	1209	1336	1477	1633
697	1	2	3	A
770	4	5	6	B
852	7	8	9	C
941	*	0	#	D

Table 11: DTMF tones

Note 1: The tone duration needed from CPE is at least 60 ms for each digit, with at least 60 ms between digits.

Note 2: The frequency tolerances and permissible intermodulation products needed from CPE are defined as follows:

- Each transmitted frequency to be within ± 1.8% of the nominal frequency;
- Total distortion (resulting from harmonics or intermodulation) to be at least 20 dB below the fundamental frequencies.

Note 3: the A, B, C, D signals are not in general use

Transmit levels are not defined in ITU-T Recommendations Q.23 and Q.24, as the level conditions depend upon national transmission plans.

- The ATA tone receiver is able to accept tones within a range between 0 dBm and -21 dBm, as received at the analogue Interface.
- The acceptable “twist” (i.e. level difference between the two tones) is within a range of a minimum of 0 dB and a maximum of 5 dB
- Where customer CPE generates DTMF or single frequency tones after call setup, for applications that are within the voice band, these signals are transmitted to the distant end so that they can be recognised by the appropriate application.

4.5.1 RFC 2833 support

While a call is in progress, either RFC 2833 DTMF relay or in-band methods may be used to communicate the DTMF tones between the ATA Voice Service and the AS.

The ATA Voice Service must support RFC 2833, and the ATA Voice Service must fall back to G.711 if it unable to negotiate or support named telephone events.

Separate RTP payload formats are desirable since low-rate voice codecs cannot be guaranteed to reproduce tone signals accurately enough for successful recognition.

4.6 POTs Port Characteristics

Foreign Exchange Station (FXS) ports must be supported. FXS ports must be coloured gray[12].

Foreign Exchange Office (FXO) ports may be supported but are outside the scope of this document. FXO ports must be coloured green[13].

The FXS ports must be RJ11[14] [15].

[12] TR 122 § I - 136
[13] TR 122 § I - 136
[14] TR 122 § I - 138
[15] TIA - 968-A § 6.2.2

The following Line Interface characteristics are based on the existing NZ PSTN Standard Analogue Line Interface requirements - viz: TNA102; section 4; DC Line Conditions ; and also PTC220; section 5; FXS Requirements

However, some existing line interface capabilities are not available, for example:

- Decadic dialling;
- Party lines

PBXs are not explicitly supported, however, they may be supported if they use functionality already specified in this document.

In this section, the term "Interface" refers to the FXS interface provided by the ATA Voice Service to the CPE.

4.6.1 DC Line Voltages

The Interface operates on a nominal DC supply voltage of -50 Volts (generally in a range between -44 and -56 Volts).

The line feed is normally applied as negative battery (relative to earth) on one wire and earth on the other. CPE devices are required to be polarity insensitive so that it is not necessary to specify a particular polarity for line feed.

The normal DC polarity of the line Interface is defined by its polarity during the idle condition, as follows:

- negative on idle wire connected to the negative lead (-50 V battery).
- positive on idle wire connected to the positive lead (earth).

4.6.2 DC Line Current

FSX ports provide a DC line feed. This is a nominal 50V source feed via a constant impedance (usually 400 Ohm).

The maximum current under a line fault condition must not exceed 125 mA[16] being the maximum current that CPE and premises wiring must support for a lengthy period without damage.

Following timeouts, the line may be placed into a "high and wet" state indefinitely, until the CPE is returned to on-hook. During the high and wet state the normal DC voltage feed continues to be applied to the line.

[16] TNA 102 § 2.6.2

4.6.3 Loop Resistance

The DC feed must be capable of supplying not less than 20mA into a load of 450 Ohms.

Many CPE devices have features such as 'last number redial' or 'memory dial' which depend on a small on-hook line current for the maintenance of memory information. The DC line feed from the Interface provides a DC power source capable of supplying a continuous on-hook current of at least 150 µA into a load of 33 kOhm in order to maintain such memories.

4.6.4 Line Event Management

The Interface does the detection of DC line conditions that signify line events required in providing basic call services, such as:

- Off-hook detection.
- On-hook detection.
- Switch-hook flash.

4.6.4.1 Off-hook Detection

The Interface must detect off-hook when the DC feed current is greater than 15mA for a period not less than 10 ms for a load of 1000 Ohm[17].

Any line event of duration less than 35 ms (the de-bounce time) will be considered not to have occurred.

4.6.4.2 On-hook Detection

The interface must detect on-hook when the DC feed current is less than 5mA for a period not less than 1000 ms for a load of 10 kOhm[18].

4.6.5 Switchhook Flash

See also section (4.7).

The ATA Interface must recognize a switch-hook flash (also known as register recall) when a temporary open circuit (ie; on-hook; followed by off-hook) with a duration of between 300 ms and 800 ms[19], is detected during a call. The upper timer and lower timer values can be set from 0 to 1400 ms during the on-boarding process.

[17] PTC 200 § 6.10 (5) (a)
[18] PTC 200 § 6.10 (5) (b)
[19] PTC 200 § 6.8

4.7 Switch-hook flash Handling

Two methods of handling switch-hook flash signalling and are referred to as loose and tight coupling.

SIP signalling standards that are relevant to switchhook flash are:

- RFC2833; RTP Payload for DTMF Digits; Telephony Tones; and Telephony Signals:
- The following sections are relevant:
- section 3.5; Payload Format: Defines the "Event" parameter that can contain the Flash event
- section 3.10; DTMF Events: digit 16 in the field represents the Flash Event.

There appear to be two options for the ATA to signal to the VSP that a Hookflash Event has been detected - viz:

- A SIP INVITE message; containing the Flash Event
- A SIP INFO message; containing the Flash Event

Note that in the ATA Spec, in Appendix 1; section 1.4 (Call Waiting) shows the use of the INVITE message to contain the Flash Event

All the above options may be supported but are not mandatory.

4.7.1 Switch-Hook Flash Handling (Loose Coupling)

Loose coupling refers to the ATA Voice Service handling the feature.

For loose coupled implementations, the hook flash is handled by the ATA Voice Service and is not reported back to the AS. The ATA Voice Service may subsequently send SIP messages back to the AS such as to place a call on hold.

The ATA Voice services may implement the following User Interface (UI):

- To answer the waiting call Hook flash + 2
- To toggle between calls Hook flash + 2
- To drop active call Hook flash + 1
- To make 3-way call Hook flash + 3

4.7.2 Switch-Hook Flash Handling (Tight Coupling)

Tight coupling refers to how the ATA Voice Service signals to the VSP the initiation of a Switch-Hook Flash for the VSP's Application Gateway to handle the call feature.

The ATA Voice Service must send a SIP INVITE to the AS[20].

4.8 Line and Information Signalling at the FXS Interface

4.8.1 Information Signalling

DTMF signalling and the data transmission associated with CLIP/MWI are the only forms of information signalling supported. The standards for DTMF signalling are covered in section 4.5

Dial-pulse dialling is not supported.

4.8.2 Line Signalling

Customer line signalling is provided at the Interface using DC line conditions as shown in the following examples.

4.8.3 Basic originating calls originating from a CPE device, delivered via the ATA to the network

Note: See also section Appendix 1 for examples of typical SIP call flows.

Note: The Direction, State, and Polarity are as at the CPE-ATA Voice Service interface point.

[20] ETSI TS 183 043 V1.1.1 § B.4.2.2.3

Line Signal	Direction	Originating Line State	Originating Line Polarity
Idle	None	Open circuit	Normal polarity
Seize	CPE to ATA (DT heard by Calling Party)	Loop	Normal polarity
Information Signalling (DTMF)	CPE to ATA	Loop	Normal polarity
Answer Answer (with ANS REV)	None ATA to CPE	Loop Loop	Normal polarity Reversed polarity
If Switch-hook Flash	CPE to ATA	Momentary open circuit (ref sections 4.6.5 and 4.7)	Polarity maintained
Release			
Either:			
Calling Party Clear (= Idle)	CPE to ATA	Open circuit	Normal polarity
or:			
Called Party Clear, then	None (DSCT heard by Calling Party)	Loop	Normal polarity
Calling Party Clear (= Idle)	CPE to ATA	Open circuit	Normal polarity

Table 12: Originating calls - line signalling

4.8.4 Basic Terminating calls received from the network via the ATA, delivered to the CPE device

Line Signal	Direction	Terminating Line State	Terminating Line Polarity
Idle	None	Open circuit	Normal polarity
Ringing sequence	ATA to CPE	Open circuit (CPE ringing)	Normal polarity
Answer (standard)	CPE to ATA	Loop	Normal polarity
If Switch-hook Flash	CPE to ATA	Momentary open circuit (ref section 4.6.5)	Normal polarity
Release			
Either:			
Called Party Clear (= Idle)	CPE to ATA	Open circuit	Normal polarity
or:			
Calling Party Clear (= Idle); then	None (DSCT heard by Called Party	Loop	Normal polarity
Called Party Clear,	CPE to ATA	Open Circuit	Normal polarity

Table 13: Terminating calls - line signalling

4.8.5 Metering Pulses

Metering Pulses are not required.

4.9 Ringing

The Ringing attributes and standards covered in this section are based on those currently used in the NZ PSTN network[21].

4.9.1 Ringing Cadences

The ATA Voice Service must apply ringing to the customer's Interface to alert the user as a consequence of an SIP INVITE message from the VSP to the URI for the port and that port is idle.

The current NZ PSTN uses four Ringing (or Distinctive Alert (DA)) patterns/cadences[22] for indicating the specific function required by the caller (for example telephone or facsimile). Customer lines may have distinctive ringing cadences applied, usually activated by dialling different telephone directory numbers. Note that the caller hears the same standard Ringing Tone cadence for all of the ringing cadence options.

When the ATA Voice Service rings the line, a SIP 180 ALERTING response must be returned, followed by a 200 OK when the called party goes off hook.

DA1 must be the default cadence if another cadence is not specified in the SIP INVITE.

During ringing and when CLIP is enabled, the Calling Line Information (CLI) in the SIP INVITE header must be sent to line using FSK signalling. This must be sent to line during the first long silent period in the first ringing cadence.

Note that the TNA102 standard does not provide for any form of "initial burst of ringing" prior the sending of the first cadence.

The first and subsequent cadences sent to line should be "full" cadences - noting that the final cadence may be truncated by a ring trip, or by the caller releasing the call.

The cadence must be repeated until a ring trip, call abandonment or timeout occurs.

The following table gives details of the standard cadences:

Cadence Name	Abbrev	Use	Cadence
Distinctive Alert Pattern 1	DA1	Standard pattern for customers	400 ms on 200 ms off 400 ms on

[21] TNA 102 § 6 Ringing Characteristics
[22] TNA 102 § 6.4 Ringing Cadences

			2000 ms off
Distinctive Alert Pattern 2	DA2	Alert pattern used for customers with a second directory number	400 ms on 2600 ms off
Distinctive Alert Pattern 3	DA3	Alert pattern used for customers with a third directory number	400 ms on 200 ms off 400 ms on 200 ms off 400 ms on 1400 ms off
Distinctive Alert Pattern 4	DA4	Alert pattern used for customers with a fourth directory number. Note: used in the New Zealand PSTN for Faxability service	400 ms on 800 ms off 400 ms on 1400 ms off

Table 14: Ringing cadences

Note: The selection of the DA4 cadence does not implicitly disable the "Call Waiting" service and the VSP must deselect the "Call Waiting" service separately[23].

Note: The TNA102 standard does not provide for any form of "initial burst of ringing" prior the sending of the first cadence, however other standards allow for an initial ringing pulse prior to sending the first cadence. In this case the FSK signalling is sent between the initial ringing pulse and the first ringing cadence.

The ringing cadences must be within ± 10%[24]

4.9.2 FSX Ringing Characteristics

The local ringing source will satisfy the following requirements:

- The ringing supply will comply with the electrical safety requirements of AS/NZS 60950.

[23] TNA 102 § 6.6 (2)
[24] TNA 102 § 6.5

- The ringing voltage and duration will comply with the TNV requirements of AS/NZS 60950, Section 2.3.1(b) annex M.

Ringing will be connected to the port as a loop connection, i.e. one terminal of the ringer supply connected to one wire of the port with the other wire serving as a ring return path to the other side of the ringer supply.

The ringing voltage may be 42 Volts RMS with a frequency of 25 Hz.

The open circuit ringing voltage applied to line at the exchange is nominally 90 Vrms dropping to 75 Vrms under the maximum rated load of the ringing generator.

Under normal operating conditions, the ringing voltage across the ringing detector (in the CPE), can be expected to be within the range of 30 to 80 Vrms.

The crest factor of the ringing waveform will be between 1.35 and 1.45. (The crest factor is defined as the ratio of the peak to RMS voltage, and this equates to a value of 1.414 for a pure sine wave.)

To ensure reasonable ringing performance, the total ringing load connected to a line should not exceed a RAL[25] of 5 (roughly equivalent to a load of 5 normal CPE devices).

A default ringing timeout supervision period of at least 4 minutes applies at the ONT for terminating calls only. It must be software configurable. When a timeout occurs, a SIP CANCEL must be sent to the ATA Voice Service.

4.9.3 Ring Trip

When ringing is being sent to CPE, the ring trip d.c. would normally have to be maintained for 40 ms or more[26].

The interface will remove ringing from the line within a maximum of 100ms from detection of an off-hook condition[27]. This timing will apply during both the active and silent periods of ringing.

4.10 Analogue Transmission Characteristics

This section outlines the analogue transmission characteristics for the ATA Voice Port. The digital access EVPL service is described in the TCF Ethernet Access Standards document.

The current NZ practices for Transmission Characteristics are based on:

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[25] TNA 102 § 6.3
[26] PTC 200 § 6.10 (5) (a) Note
[27] PTC 220 § 5.6.1

- TNA151; Telecom Telephone Network Transmission Plan (1996); updated by:
- Telecom NZ document ASG0008 (2003)[28] ; Telephony Voice Quality

The following is based on these documents.

4.10.1 Loss Plan

The loudness rating objective nominal values and permitted ranges are illustrated below in a series of half circuit diagrams. Loudness ratings are summed to give the overall send loudness rating (SLR) or receive loudness rating (RLR) of the half circuit. The overall loudness rating (OLR) of a connection is calculated by adding the SLR of one half circuit to the RLR of the half circuit to which it is connected. Objective nominal value of OLR = 10 dB.

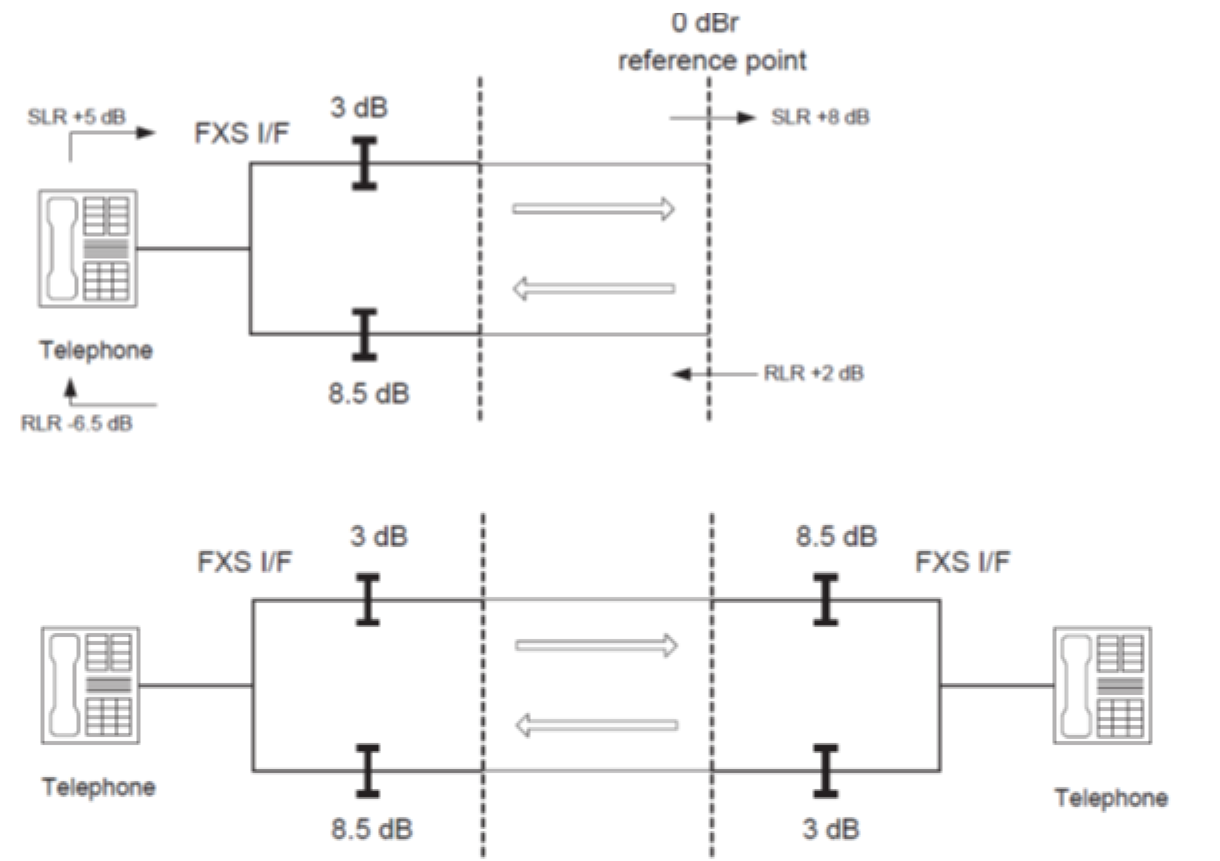

Figure 5: ATA Voice and Baseband objective loss plan

Note: In the above figure, FXS I/F means the ATA Voice Service Port Interface

[28] This document is not publicly available

4.10.2 Objective impedances

An impedance plan is required to reduce impedance mismatches in connections. Impedance mismatch can lead to impairment of voice quality due to echo and/or less than optimal telephone sidetone.

To minimise mismatch it is necessary to specify an objective standard impedance for the input impedance of telephones, input impedance of 2-wire analogue ports, input impedance of digital switches, and balance networks associated with 2-wire/4-wire hybrids.

There are two impedances associated with the ATA Voice Service port. Firstly, there is the input impedance of the port, and secondly the balance impedance.

4.10.3 Port Input Impedance BT3 Port Input Impedance

The nominal input impedance used by the current NZ PSTN is a 370 ohm resistor in series with a parallel combination of a 620 ohm resistor and a 310 nanofarad capacitor (known as BT3)[29].

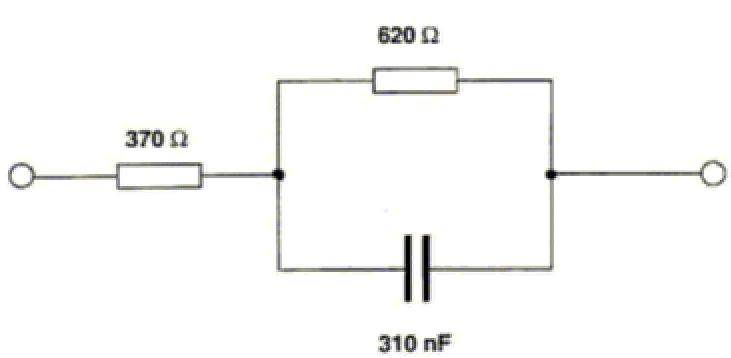

Figure 6: BT3 impedance

Each ATA Voice Service port must provide a BT3 impedance for both port input impedance and balance impedance.

The port input impedance is measured as a Return Loss against BT3, at the following frequencies: 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 3150, 4000 Hz:

- The Return Loss is not less than 12dB at any of the above frequencies.
- The Echo Return Loss (ELR) is not less than 14 dB.

The ELR is calculated according to the method given in ITU-T Recommendation G.122.

[29] TNA 102 § 8.2 Network Impedance

4.10.4 Echo Cancellers

The ATA Voice Service must provide G.168 echo cancellers in their networks to eliminate any hybrid imbalance and handset conduction.

The ATA Voice Service port must support a near-end 32[30] ms G.168 echo-canceller capability.

4.10.5 Noise

Noise is any unwanted electrical energy which can be measured sophometrically at the output of a device or line. For the purposes of this specification, 'noise' also includes unwanted electrical energy at frequencies outside the normal speech band of 300 Hz to 3400 Hz[31].

With the ATA Voice Service port terminated in 600 Ohms (off-hook with speech path open but quiet) the noise must be less than -65dBm[32].

With ATA Voice Service port terminated in 10 kOhms (on-hook) the noise must be less than -65dBm[33].

4.10.6 Frequency range

The voice frequency range from 300 - 3400 Hz must be supported[34].

4.11 Analogue Data Transmission

4.11.1 Overview

Analogue data transmission carries service specific data to analogue data capable CPE. The applicable NZ standard for general analogue data transmission is TNA 102; section 11.

This facility does not include the use of standard techniques such as DTMF for data transfer.

Services using this capability may include:

- Analogue Calling Line Identity Presentation (CLIP), together with Calling Name service;
- Visual Message Waiting Indication (VMWI);
- Calling Line Identity Presentation (CLIP) on Call Waiting service;

Analogue data can be transmitted when the Line interface is in an "On-hook" condition for Analogue CLIP and VMWI, and in an "Off-hook" condition for CLIP on Call Waiting.

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[30] TR 122 § I - 198
[31] PTC 200 § Definition of Noise
[32] PTC 220 § 5.3.4.4(1)
[33] PTC220 5.3.4.4(2)
[34] TNA 102 § 8.1

4.11.2 Generic data transmission standard

The ATA Voice Service may implement analogue data transmission in compliance with Telcordia Specification GR-30-CORE (formerly Bellcore TR-NWT-000030) and Telcordia Specification GR-1401-CORE (MWI below)

When analogue data transmission functionality is supported, both the SDMF and MDMF implementations must be supported.

On-hook data transmission with ringing (or distinctive ringing) is implemented in conjunction with the ringing cadences and must occur after the first ringing cadence and before the second ringing cadence[35].

4.11.3 Analogue Calling Line Identity Presentation (CLIP)

When the Interface Line is idle, analogue CLIP is sent in accordance with Telcordia Specification GR-31-CORE and TNA 102 section 11. This includes:

- Up to 10 digits coded in IA5 with no parity,

or

- (ii) IA5 character "P" if an anonymous indication is to be delivered in lieu of the calling line directory number as reason for absence of directory number,

or

- (iii) IA5 character "O" if an out-of-area/unavailable indication is to be delivered in lieu of the calling line directory number as reason for absence of directory number.

The number format provided from the ATA Voice Service must use the user part of the URI in the “FROM” field for normal calls.

The ATA Voice Service must not add a “0” prefix for the national access code. Insertion of a leading “0” for a dial-back is the responsibility of the Caller Display equipment and guidelines are provided in PTC200 section 5.5.2 including when the leading “0” should not be inserted.

Where the calling party has requested to remain anonymous, the “From” header will be received in the format <sip:anonymous@anonymous.invalid>[36]. The ATA Voice Service must send a “P” as the caller ID.

The sending of the “O” character is not required to be supported.

The date and time must be derived from the ATA Voice Service time source and is in local time[37].

[35] TNA 102 § 10.1
[36] PTC 228 § 4.11.5 (a) (iii)
[37] PTC 200 and TNA 102 are silent on the time zone to be used and local time (NZDT/NZST) is assumed.

Where Calling Line Identity Restriction (CLIR) is required, the withholding of the CLI is a VSP responsibility and the ATA Voice Service must be regarded as an untrusted end point.

4.11.4 Message Waiting Indication (MWI)

MWI is sent in accordance with Telcordia Specification GR-1401-CORE.

The Multiple Data Message Format (MDMF) must be supported. MWI information is included by the ATA in a MDMF message sent to the CPE. The CPE is responsible for displaying a visual message waiting indicator.

The ATA must support the playing of the supervisory stutter dial tone (MWT) as per section 4.1.1.

Implied subscription to the message waiting indication service may be the default, that is, the ATA Voice Service may not send a ‘SUBSCRIBE” message to the AS following registration.

The NOTIFY method must be used for message waiting indication service. The body text in the NOTIFY message must be:

- Turn on waiting indication Messages-Waiting: yes
- Turn off waiting indication Messages-Waiting: no

For example, see the sample NOTIFY message below, the BODY standard is defined in ETSI 183 043, on how the tone may be formatted:

```
NOTIFY sip:77770271@171.1.20.108:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP
171.23.111.11:5060;branch=z9hG4bKccebcuoypnebn9edf7pu9ntfd;Role=1;Dpt=75d2_36
Call-ID: jlz4s8kjm2nllncscnsi238jnk33c034@171.1.20.108
From: <sip:77770271@ont.huawei.com>;tag=sdssd7i7-CC-51
To: <sip:77770271@ont.huawei.com>;tag=nkpk1n34
CSeq: 5 NOTIFY
Contact: <sip:172.23.111.25:5060>
Max-Forwards: 68
Event: ua-profile
Subscription-State: active;expires=3188
Content-Length: 314
Content-Type: application/simservs+xml

<?xml version="1.0"?>
<simservs>
  <dial-tone-management>
    <dial-tone-pattern>special-condition-tone</dial-tone-pattern>
```

</dial-tone-management>
</simservs>

4.11.5 Calling Line Identity Presentation (CLIP) on Call Waiting

When the Interface Line is off-hook and Call Waiting is invoked, analogue data is sent in accordance with the Telcordia Specifications GR-30-CORE and GR-575-CORE.

For this service the first of the Call Waiting tones is immediately followed by a CPE Alerting Signal (CAS). If the CPE can support the service it will respond to the CAS with an Acknowledgement (ACK) signal, which comprises either a DTMF —A or —D digit (reference Section D.6.1). The FXS will then in turn proceed to send analogue CLIP signals in accordance with Telcordia Specification GR-31-CORE and as outlined above.

CLIP Type 2 (off hook) can be disabled independently of providing CLIP Type 1 (on hook).

4.12 Clear Forward

On receipt of a BYE message the ATA Voice Service may apply (VSP defined) a clear forward signal to the line to signal to the CPE that the other party has terminated the call.

This is principally used on CO trunks to signal to PABX systems and to voice message systems that message recording should stop[38].

The “clear forward” is a 800 - 1,100 ms break on the idle positive lead (i.e. the lead that is positive when in the idle condition). In the future this may be changed to the idle negative lead.

The ATA Voice Service may support Clear Forward.

4.13 Polarity Reversal Answer (Answer Supervision)

A reversal of polarity[39] is to indicate to payphones[40] and private call loggers that the called party has answered the call[41]. This may be used to indicate that call charging may commence and is only active for the calling party.

The ATA Voice Service may reverse the polarity on the line on receipt of 200 OK of Invite.

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[38] PTC 107 § 6.5
[39] TNA 102 § 4.4
[40] PTC 213: 2004 § 8.4 (3)
[41] TNA 102 § 4.5

Answer supervision is NOT part of the standard NZ PSTN service. It was historically available only on business lines which service a PBX system or payphone.

4.14 SIP Standards Support

The following SIP-related sections should be read in conjunction with the typical SIP Call Flow diagrams shown in Appendix 1 (if any).

The ATA Voice Service must conform to the specifications listed in the following table.

Standard	Description
RFC 791	Internet Protocol (IPv4) Note: IPv6 is not currently supported.
RFC 2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
RFC 2976	The SIP INFO method
RFC 3261	SIP: Session Initiation Protocol The transport method supported for SIP signalling is UDP (RFC 768). The use of TCP (RFC 793) is not supported.
RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
RFC 3264	An Offer/Answer Model with Session Description Protocol (SDP)
RFC 3311	The Session Initiation Protocol (SIP) UPDATE Method
RFC 3323	A Privacy Mechanism for the Session Initiation Protocol (SIP)
RFC 3550	RTP: A Transport Protocol for Real-Time Applications
RCF 3398	Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping
RFC 3842	A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
RFC 4033	DNS Security Introduction and Requirement
RCF 4497	Interworking between the Session Initiation Protocol (SIP) and QSIG”

RFC 4566	SDP: Session Description Protocol
RFC 5246	The Transport Layer Security (TLS) Protocol Version 1.2

Table 16: SIP Standards Support

4.15 SIP Methods Support

The minimum set of SIP methods that require support include those shown in the following table.

Method	Use	RFC	Section
REGISTER	The SIP REGISTER method is used by the ONT to establish and maintain registration with the VSP's softswitch.	3261	
INVITE	The SIP INVITE method is used to invite another party to participate in a call session. The INVITE method can also be used within an existing dialog to change SDP characteristics once a call session has been established (in which case the INVITE is commonly called a REINVITE).	3261	
ACK	The SIP ACK method confirms that a client has received a final response (2xx, 3xx, 4xx, 5xx or 6xx response) to an INVITE request. The VSP must be able to send and receive SIP ACK requests. If the INVITE (or REINVITE) request sent to the VSP did not contain a SDP offer, then the SDP offer must be included in the 200 OK (INVITE), and the SDP answer must be included in the ACK. The VSP must be able to receive SDP answers within the ACK requests.	3261	
BYE	The SIP BYE method terminates a call. The VSP must be able to send and receive a SIP BYE request. The VSP must only send a BYE if an INVITE dialog is confirmed (i.e. a 200 OK INVITE and ACK have been successfully exchanged between the VSP and the ONT). If the dialog has not reached the confirmed state, a SIP CANCEL must be used instead.	3261	

CANCEL	The SIP CANCEL method terminates a pending INVITE before a 200 OK (INVITE) has been received. The VSP must be able to send and receive a CANCEL request. The VSP must only send (or receive) a CANCEL if an INVITE dialog is not confirmed (that is, a 200 OK INVITE and ACK have not been successfully exchanged between the VSP and the ONT). If the dialog is confirmed, a SIP BYE must be used instead.	3261	
NOTIFY	The SIP NOTIFY method is only required if a VSP intends to support the Message Waiting Indication supplementary service.	3265	
PRACK	The PRACK (Provisional Response Acknowledgement) method provides provisional responses to certain SIP messages.	3262	
OPTIONS	The SIP method OPTIONS allows a UA to query another UA or a proxy server as to its capabilities. This allows a client to discover information about the supported methods, content types, extensions, codecs, etc. without "ringing" the other party.	3261	
UPDATE	UPDATE allows a client to update parameters of a session (such as the set of media streams and their codecs) but has no impact on the state of a dialog.	3311	
INFO	The INFO method is used for communicating mid-session signalling information along the signalling path for the call.	2976	

Table 17: SIP Methods Support

4.16 SIP Signaling General

This section is planned to be completed as part of a later release of this document, post 2015.

4.17 SIP Transport support

The transport layer used for transmission of SIP transactions must be UDP and must support use of IP version 4, however the service design must not unreasonably preclude the use of IPv6 in the future.

The ATA Voice Service must not apply Port or IP address translation.

The ATA Voice Service may support SIPS.

4.18 SIP Header support

The support of SIP Headers complies with Tables 2 and 3 of RFC 3261 with additions applied by the RFCs referenced in this document.

4.19 Digit Handling

The New Zealand Number Register (which reflects the NZ dial plan) is maintained by the Number Administration Deed (http://www.nad.org.nz) and the Number Register can be accessed at http://www.nad.org.nz/number-register/.

4.19.1 Digit Map

A Digit Map will minimize post-dialling delay by avoiding, as much as possible, the need to incur a time-out to determine number length.

The ATA Voice Service must, as a minimum, support a configurable digit map as described in IETF RFC 3435, Media Gateway Control Protocol (MGCP), Version 1.0, Section 2.1.5[42].

The length of the Digit Map must not be less than 256 characters[43].

The Service Provider is responsible for specifying and providing updates of their preferred Digit Map including any specific feature activation codes. The initial Digit Map must be provided by the Service provider during on-boarding.

4.19.2 Digit Map example

The following digit map is provided as an example of a digit map.

(111|911|999|00xxxxx.T|001xx{9}|0061[02-9]x{8}|01xxx.T|01[08]|0110|015[013-9]|01681800x{7}|017[0239]|019[67]|0198xx|02xxxxxxxx.T|020[1278]x{6}|0210[03-7]x{5}|021[12]x{6}|021[3-9]x{5}|022x{7}|026[02-9]x{6}|027x{7}|028[037]x{6}|029x{7}|0[3469]x{7}|07xxxxxxxx.T|070[03]x{6}|07[2-9]x{6}|05xxxxx.T|0508x{6}|08xxxxx.T|0800x{6}|0830xx|08321x|087[459]x|1xxx.T|12[0-8]|129x|17[459]x|19[67]|19[34589]x|[2-9]x{6}|[*#]xx.T)

[42] TR 122 § I-296
[43] TR 122 § I-297

111	Emergency services fixed length 3 digits
911	Emergency services fixed length 3 digits
999	Emergency services (routed to a PSTN announcement) fixed length 3 digits
00xxxxx.T	International variable length, timeout after 6 or more digits received (ie "catch-all" for other 00 codes not shown below)
001xx{9}	North America fixed length 13 digits
0061[02-9]x{8}	Australia fixed length codes 13 digits
01xxx.T	Special service codes variable length, timeout after 4 or more digits received (ie "catch-all" for other 01 codes not shown below)
01[08]	NA/DA operator codes fixed length 3 digits
0110	Auto-collect code fixed length 4 digits
015[013-9]	TelstraClear service codes fixed length 4 digits
01681800x{7}	USA Freephone Access fixed length 15 digits
017[0239]	Yabba and International operator codes fixed length 4 digits
019[67]	CLIP/CLIR override codes fixed length 4 digits (then a pause in dialling, receipt of second dial tone from the Service Provider's soft-switch, then dialling the called number)
0198xx	Call Plus card fixed length 6 digits
02xxxxxxxx.T	Mobile codes variable length, timeout after 9 or more digits received ()
020[1278]x{6}	Orcon mobile etc fixed length 10 digits
0210[03-7]x{5}	Vodafone fixed length 10 digits
021[12]x{6}	Vodafone fixed length 10 digits
021[3-9]x{5}	Vodafone fixed length 9 digits

022x{7}	2degrees mobile fixed length 10 digits
26[02-9]x{6}	Telepaging fixed length 10 digits
027x{7}	Telecom NZ and WXC mobile fixed length 10 digits
028[037]x{6}	Compass mobile etc fixed length 10 digits
029x{7}	TelstraClear mobile fixed length 10 digits
0[3469]x{7}	National codes fixed length 9 digits
07xxxxxxxx.T	National 07 codes variable length, timeout after 9 or more digits received (ie "catch-all" for other 07 codes not shown below)
070[03]x{6}	WXC PCS, etc fixed length 10 digits
07[2-9]x{6}	National codes fixed length 9 digits
05xxxxx.T	Interconnect codes variable length, timeout after 6 or more digits received (ie "catch-all" for other 05 codes not shown below)
0508x{6}	TelstraClear freephone fixed length 10 digits
08xxxxx.T	Misc 08 codes variable length, timeout after 6 or more digits received (ie "catch-all" for other 08 codes not shown below)
0800x{6}	Freephone fixed length 10 digits
0830xx	Audio-conf fixed length 6 digits
08321x	VSP fixed length 6 digits
087[459]x	EFTPOS/Packet dial-up fixed length 5 digits
1xxx.T	Special service codes variable length, timeout after 3 or more digits received (ie "catch-all" for other 1 codes not shown below)
12[0-8]	Service codes fixed length 3 digits

129x	Service codes fixed length 4 digits
17[459]x	EFTPOS/Packet dial-up fixed length 4 digits
19[34589]x	Service codes fixed length 4 digits
19[67]	CLIP/CLIR override codes fixed length 3 digits (then a pause in dialling, receipt of second dial tone from the Service Provider’s soft-switch, then dialling the called number)
[2-9]x{6}	Local codes fixed length 7 digits

Note:

- **0**-**9**, ***** and **#** represent the respective dialled digits
- The symbol "**x**" is used as a wildcard, designating any event corresponding to digits in the range **0**-**9** (but not ***** or **#**)
- A set of “alternative” digit symbols can be enclosed in brackets **[ ]**,
  - represents one occurrence of any of the enclosed digit symbols
  - allows for ranging using hyphen symbol **-**
- **{ n }** signifies n occurrences
- **.** signifies zero or more occurrences
- **x.T** defines an end of dialling timer, which is set to 5 seconds. The inter-digit timer is also set to 5 seconds.
- **E** intended to be used as a suffix to denote an Emergency number but it should not be used as it results in the following:
  - alarm generation
  - use of unsupported SIP priority headers in the INVITE message
  - switch-hook flash is ignored
  - terminating calls receive SIP response 486 (eg call waiting is not invoked)

  no change to normal calling party release (ie manual hold is **not** invoked)

Table 18: Digit Map

4.19.3 Digit Reception, Timing, and Digit Timeout Conditions

Timeouts are a standard way of managing digit reception from the customer, in order to:

- mimimise the overall holding time of a call that will arise because of slow dialling sequences by a customer, and
- to determine the number length for variable length codes - typically, when calling overseas numbers

The ATA Voice Service must handle a minimum of 20 digits[44].

The following table outlines the current inter-digit reception time-out settings for the NZ PSTN (ref TNA102; section 5.4):

Supervision Requirement	Definition	Time Period
Pre-Dialling Timeout (aka DT timeout)	This is the maximum time from when DT is played, to the receipt of the first digit. Upon timeout DT is removed and DSCT is played.	15-16 secs Default: 16 sec[45]
Inter-Digits Timeout	Maximum time between each of digits 1 to 8 Maximum time between each digit after 8 digits (desirable)	10-11 secs Default: 11 sec 5-6 secs Default: 6 secs

Table 19: Digit Reception Timing Supervision

Notes:

- The above timings reflect Spark's current implementation in the PSTN. Exact compliance with these timings is not required; but it is recommended that general alignment with these timings will ensure consistency from a user perspective

The # key must be supported to indicate the end of the entry of the digits[46]; but the actual use of the # by a customer for this purpose, is optional.

[44] TNA 102 § 2.7.1(1)
[45] TNA 102 § 5.4 (2)(a)
[46] TR 122 § I - 299

5. Glossary

Advanced Gateway	Advanced Gateway – from 3.1
ATA	Analogue Terminal Adapter. In this context ATA refers to a telephony adapter on the ONT located at the end user's premises
BSS	Business Support System
CIR	Committed Information Rate
CVLAN	Customer Virtual Local Area Network
CWMP	CPE WAN Management Protocol.
DHCP	Dynamic Host Configuration Protocol (RFC 2131)
DNS	Domain Name System
DSCP	Diff Serve Code Point. A field in an IP header used for setting a packet's class of service or prioritisation
DTMF	Dual Tone Multi frequency
E-NNI	External Network-Network Interface
FSK	Frequency-Shift Keying, a modulation method for sending data over an analogue circuit.
GPON	Gigabit Passive Optical Network
HTTP	Hyper Text Transfer Protocol. Used for communication over a computer network.
IETF	Internet Engineering Task Force. A leading standards body for the Internet. IETF standards are published as RFC documents. www.ietf.org
I-NNI	Internal Network – Network Interface
IPv4	Internet Protocol version 4, RFC 791, IETF, September 1981
ITU-T	The ITU Telecommunication Standardization Sector (ITU-T) is one of the three sectors (divisions or units) of the International Telecommunication Union (ITU); it coordinates standards for telecommunications.
LAG	Link Aggregation Group. Logical termination that aggregates one or more physical link into one logical link. Used to aggregate port capacity and provide redundancy. Also allows port rearrangements without impacting L2 terminations on the LAG.

Loose Coupling	Call control intelligence is embedded in the Access Gateway Control Function rather than in a separate SIP Application Server.
OLT	Optical Line Termination. A GPON Access Node that provides for the delivery of LFC services.
ONT	Optical Network Termination. A single subscriber device that terminates any endpoint of an optical distribution network.
OVC	Operator Virtual Circuit.
PCP	Priority Code Point. A three bit field in the 802.1q header that identifies what class a particular frame is associated with.
PIR	Peak Information Rate. This is the "burstable" rate.
POI	Point of Interconnect
POTS	Plain Old Telephone System – analogue phone service as provided by the Public Switched Telephone Network (PSTN)
QOS	Quality of Service. A means whereby things (such as data) are treated as priorities.
RSP	Retail Service Provider
RTP	Real Time Protocol (RFC 3550)
RCP	Remote Procedure Call methods, encoded in SOAP. That allows a device to access the services of a remote application.
SBC	Session Border Controller
SDP	Session Description Protocol (RFC 4566)
SIP	Session Initiation Protocol (RFC 3261)
SSP	Self Service Portal.
SVID	Service VLAN ID. This is the VLAN identifier contained in the outer 802.1ad tag delivered on the E-NNI.
SVLAN	Service Virtual Local Area Network (refer SVID)
TISPAN	Telecoms and Internet converged Services and Protocols for Advances network
TSO	Telecom Service Obligations
TTY	Teleprinter/Teletype/Typewriter
VLAN	Virtual LAN. A Virtual LAN has the same attributes as a physical local area network

	(LAN), but it allows for end stations to be grouped together even if they are not located on the same network switch.
Voice-AVPL	Voice Access Virtual Private Line
VoIP	Voice over Internet Protocol.
VSP	Voice Service Provider, a retail service provider that provides PSTN voice service
UA	User Agent. The end user SIP signalling function in the ONT supporting ATA Voice and Baseband.
UDP	User Datagram Protocol
UFB	Ultrafast Fast Broadband. New Zealand's Fibre roll out project.
XML	Extensible Markup Language

6. Appendix 1. Typical SIP Call Flows

This section is planned to be completed as part of a later release of this document, post 2015.

7. TNA 102 Conformance Matrix

The following matrix reviews this document's compliance with TNA 102.

Section	Compliance	Details
1.1 Telecom PSTN	N/A	Introductory information on the Telecom PSTN
1.2 CPE	N/A	• The ATA Voice service does not require Telecom Telepermits. • The ATA Voice service assumes CPE comply with PTC 200
1.3 legal Requirements	Comply	The Telecommunications Act acknowledges the right for a Network Service Provider to require compliance testing. That is out of scope of this document.
1.4 Compliance with International Standards	N/A	Provided for information by Telecom
2.1 Mode of presentation	N/A	Provided for information by Telecom
2.2 Network Demarcation	Non-compliance	The Network demarcation is specified in the UFB Service descriptions and differs from this document. The UFB demarcation is the UNI-V port of the ATA Voice Service.
2.3 Service Delivery Points	N/A	Provided for information by Telecom
2.4 Responsibilities	Comply	Subject to the compliance with 2.2 above, the responsibilities are largely the same
2.5 Network Interface Characteristics	N/A	Provided for information by Telecom
2.6 Protection from line Interference	N/A	Provided for information by Telecom. The ATA Voice Service works within these requirements. Refer § 3 & 4.10.12
2.7 Network Numbering	Comply	Provided for information by Telecom. The requirement for storing 20 digits is included in §

		4.19.3
2.8 Supplementary services and Centrex	N/A	Provided for information by Telecom No additional functionality is provided to support Centrex functionality as this would be provided by the VSP.
3 Definitions	N/A	Provided for information by Telecom
4.1 General	N/A	Provided for information by Telecom
4.2 Exchange line Feed Equipment	Comply	Detailed in § 4.8 TNA 102 § 4.2 (1): The additional voltage boost is not supported as long lines will not be a feature of premises wiring.
4.3 Derived Circuits	N/A	Derived circuits are not expected to be used in conjunction with the ATA Voice Service
4.4 Line Polarity	Comply	Detailed in § 4.8.3
4.5 Answer Supervision	Comply	Detailed in § 4.13
4.6 Voltage Transients	N/A	Provided for information by Telecom
4.7 Telecom Test	N/A	Provided for information by Telecom
4.8 Requirements for Terminal Equipment	N/A	Provided for information by Telecom. TNA102 section 4.8 notes that these requirements are covered by the PTC 200 specification series.
5.1 Signalling types	N/A	Provided for information by Telecom Decadic dialling is not supported by the ATA Voice Service
5.2 DTMF	Comply	Detailed in § 4.5
5.3 Decadic Signalling	Non compliance	Decadic signalling is not supported by the ATA Voice Service.
5.4 Timeout	Comply	General alignment of timings is expected. Detailed in § 4.19.3
5.5 Recall	Comply	(aka Switch Hook Flash) Detailed in § 4.7
6.1 Ringing Frequency	Comply	Detailed in § 4.10.6

6.2 Ringing Voltage	Comply	Detailed in § 4.9.2
6.3 Ringing Current	Comply	Detailed in § 4.9.2
6.4 Ringing Cadences	Comply	Detailed in § 4.9.1
6.5 Tolerance on Cadences	Comply	Detailed in § 4.9.1
6.6 Allocation of Cadences	N/A	The allocation of the cadences is undertaken by the VSP.
6.7 PABX Ringing	N/A	Provided for information by Telecom
6.8 Multiparty ringing	N/A	Provided for information by Telecom. The ATA Voice Service does not support multi-party ringing.
7.1 Supervisory Tones	Comply (subsection 1) Non Compliance (subsection 2)	Detailed in § 4.1.1.
7.2 Tolerances on Frequencies and Cadences	Comply	Detailed in § 4.1.1.
7.3 Received levels of Supervisory Tones	Comply	Detailed in § 4.1.1.
8.1 Frequency Range	Comply	Detailed in § 4.10.6
8.2 Network Impedance	Comply	Detailed in § 4.10.2
8.3 Impedance balance about Earth	N/A	Provided for information by Telecom
8.4 Limits for Transmitted Speech and Data	N/A	No limits on the maximum power limits are included in this document. (8.4 refers to the PTC200 series of standards)
8.5 Network and Local Circuit Losses	Comply	Detailed in § 4.10.1

8.6 Performance of facsimile and Data Modems	N/A	Provided for information by Telecom The same caveat applies to the ATA Voice Service
8.7 Received Speech Levels	N/A	Provided for information by Telecom
8.8 Echo	Comply	Detailed in § 4.10.4
9.1 Customer Interface Arrangements - General	N/A	Provided for information by Telecom
9.2 Telecom Standard	Comply	The ATA Voice Service works with these requirements.
9.3 Physical Network Connection Methods	Comply	The ATA Voice Service works with these requirements.
10 Analogue on-hook Data Transmission	Comply	Detailed in § 4.11
11 Analogue Calling Line Identification Presentation	Comply	Detailed in § 4.11.3