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NICC Document

NGA Telephony SIP User Profile

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Contents

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Foreword

This NICC Document (ND) has been produced by NICC AP WG

1 Scope

The present document The present document specifies the interface between an NGA-T Service Provider and NGA-T Access Provider based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) to enable an IP based telephony service.

This document provides a profile of SIP and SDP according to IETF RFCs where possible supplemented by endorsement of 3GPP and ETSI TISPAN procedures where necessary. This profile provides a signalling capability suitable for a NGA-T Service as defined in ND1645 [20].

Specific Annexes describe how particular UE use the SIP profile described in the main body. Therefore, in order to fully understand the signalling capabilities available the appropriate Annex should be read as well as the main body.

2 References

For the particular version of a document applicable to this release see ND1610 [1].

2.1 Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

[1] ND1610 Next Generation Networks, Release Definition
[2] RFC 3261 Session Initiation Protocol
[3] RFC 3262 Reliability of Provisional Responses in the Session Initiation Protocol
[4] RFC 3265 Session Initiation Protocol (SIP)-Specific Event Notification
[5] RFC 3311 The Session Initiation Protocol (SIP) UPDATE Method
[6] RFC 2976 The SIP INFO Method
[7] RFC 2782 A DNS RR for specifying the location of services (DNS SRV)
[8] RFC 4566 SDP: Session Description Protocol
[9] RFC 3264 An Offer/Answer Model with the Session Description Protocol
[10] RFC 4733 RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals
[11] RFC 4028 Session Timers in the Session Initiation Protocol
[12] RFC 5009 Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media
[13] RFC 3842 A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
[14] ETSI TS 124 229 Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3
[15] ETSI TS 183 043 Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS - based PSTN/ISDN Emulation; Stage 3 specification
[16] draft-ietf-sipping-config-framework-17 A Framework for Session Initiation Protocol User Agent Profile Delivery.

[17] RFC 3551 RTP Profile for Audio and Video Conferences with Minimal Control
[18] ND1646 NGA-Telephony Management Interface
[19] 3GPP TS 24.616 3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; Malicious Communication Identification (MCID) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (Release 8)
[20] ND1645 NGA Telephony; Architecture & Requirements
[21] RFC 3550 RTP: A Transport Protocol for Real-Time Applications
[22] 3GPP TS 24.607 3rd Generation Partnership Project;Technical Specification Group Core Network and Terminals; Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[23] ND1016 Requirements on Communication Providers in Relation to Customer Line Identification Display Services and Other Related Services
[24] 3GPP TS 24.615 3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[25] ND1418 NGA Telephony SIP User Profile, VQM Information

3 Definitions and abbreviations

3.1 Definitions

Next Generation Access Telephony (NGA-T) Service Provider : As defined in ND1645 [20].

Next Generation Access Telephony (NGA-T) Access Provider : As defined in ND1645 [20].

Analogue Telephony Adapter (ATA): As defined in ND1645 [20].

Outbound Proxy: An outbound proxy is a device to which a UE directs any SIP request. The outbound proxy may process the request or forward it to another entity.

3.2 Abbreviations

3GPP	3rd Generation Partnership Project
ABNF	Augmented Backus Naur Form
ATA	Analogue Terminal Adaptor
CGPN	Calling Party Number
CW	Call Waiting
DNS	Domain Name Server
DTMF	Dual-Tone Multi-Frequency
ETSI	European Telecommunications Standards Institute
FQDN	Fully Qualified Domain Name
FSK	Frequency Shift Keying
ID	Identity
IETF	Internet Engineering Task Force
MCID	Malicious Call Identity
MIME	Multipurpose Internet Mail Extensions
MWI	Message Waiting Indication

NGA	Next Generation Access
NGA-T	NGA-Telephony
NICC	NICC Standards Limited
P-E-M	P-Early-Media
Req-URI	Request-Uniform Resource Identifier
RFC	Request For Comment
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SRV	Service (record)
TEL	Telephone
TISPAN	Telecommunications and Internet converged Services and Protocols for Advanced Networking
TLS	Transport Layer Security
UDP	User Datagram Protocol
UA	User Agent
UE	User Entity
UNI	User Network Interface
URI	Uniform Resource Identifier
XML	Extensible Markup Language

4 General

4.1 URI support

4.1.1 UE originating calls

The SIP UNI shall support the sending of SIP-URI format in the req-URI, From and To headers within SIP requests and responses.

The userpart of the SIP URI in the req-URI and To header shall contain either a dialled digit string or the event that has occurred. The format of these shall be as follows:

- SIP URI's containing dialled digits in the userpart
  sip:<dialled digits>@<domain>; user=phone.

- SIP URI's that do not contain dialled digits (e.g. a hotline call)
  sip:<event>@<domain>.

The domain name portion shall comply with ND1645 [20].

4.1.2 UE terminating calls

The SIP UNI shall support the receipt of both SIP and TEL URI formats in the req-URI, From and To headers contained within SIP requests and responses.

4.2 SIP Transport support

The transport layer used for transmission of SIP transactions shall be UDP.

4.3 Outbound Proxy Address Discovery

The UE shall be capable of obtaining the Outbound Proxy address and port. To obtain this information, the UE shall use one of the following mechanisms:

- Static Outbound Proxy configuration:
  The UE shall support provisioning of a Primary and one or more alternative Outbound Proxy IP addresses and ports

- DNS:
  The UE utilizes DNS SRV queries as described in RFC 2782 [7] to determine the IP address(es) and port(s) of the Outbound Proxy(s) associated with the Service Provider's domain name. This option assumes that the UE has been pre-configured with the domain name of the NGA-T Service Provider network.

4.4 Failover and recovery

The UE shall first attempt to register towards the Primary Outbound Proxy. If registration towards the Primary Outbound Proxy fails, the UE shall attempt to register towards a Secondary Outbound Proxy, if available. Where registration fails towards an Outbound Proxy, that proxy is quarantined for a period of time (Proxy Wait Period). The value of the Proxy Wait Period is described in this clause.

Registration shall not be attempted towards an Outbound Proxy as long as its Wait Period is active.

The UE shall cancel the Proxy Wait Period for all its Proxies upon detection of any non-signalling event which causes the UE to restart its registration procedures.

When unregistered the UE should inform the user of the condition.

4.4.1 UE behaviour upon non-Register request failures/error responses

To support a high grade of service to the end user, the UE shall immediately change to an unregistered state and commence initial registration for any of the following events:

- The response to any non-register request is 403 (forbidden):
  In this case the UE shall perform initial registration procedures as described in clause 8.1, towards the Primary Outbound Proxy.

- Where a SIP request receives no response and the associated timer B or F has expired (indicative of an outbound proxy server failure): In this case the UE shall not attempt initial registration towards the same Outbound Proxy for 5 minutes, but shall perform initial registration procedures as described in clause 8.1, towards another Outbound Proxy, if available.

In these cases the UE should inform the user with an appropriate indication.

4.4.2 UE behaviour upon Registration failures / error responses

4.4.2.1 SIP Timer F expiry handling

When registration towards an Outbound Proxy fails due to SIP Timer F expiry, the UE shall not automatically re-attempt registration towards the same Outbound Proxy for a Proxy Wait Period of 5 minutes and attempt initial registration as described in clause 8.1, towards another Outbound Proxy, if available.

4.4.2.2 Registration Error handling

4.4.2.2.1 Less than five consecutive Registration Errors

If registration fails due to an error code other than 403 (Forbidden) or 423 (Interval too brief) then the UE shall attempt registration as described in clause 8.1 towards the same proxy.

If a Retry-After header field is present in the registration response, the UE shall:

- Not automatically re-attempt the registration request towards the same proxy until after the period indicated by the Retry-After header field contents; and

- If the Retry-After header field value is greater than an implementation specific value, then the UE shall attempt initial registration as described in clause 8.1 towards another proxy, if available.

  Note: The implementation specific value affects the grade of service and should be chosen accordingly.

The handling of 403 (Forbidden) error code is as described in clause 4.4.2.3.

The handling of 423 (Interval too brief) is described in clause 8.1.

4.4.2.2.2 Five consecutive Registration Error responses

If the fifth consecutive registration attempt towards the same proxy fails due to error code responses then:

- further attempts to the same proxy should be prevented for a Proxy Wait Period of at least:
  - the period of time indicated in the Retry-After header field, if the Retry-After header field was present; or
  - 5 minutes in case of initial registration failure, if the Retry-After header field was not present; or
  - 30 minutes in case of re-registration failure, if the Retry-After header field was not present.

- Initial registration as described in clause 8.1 shall be attempted towards another proxy, if available.

4.4.2.3 Authentication Failure handling

4.4.2.3.1 Single authentication failure

If registration fails due to a 403 (Forbidden) error code, then the UE shall attempt initial registration as described in clause 8.1 towards the same proxy.

4.4.2.3.2 Two consecutive authentication failures

Upon receipt of two consecutive 403 (Forbidden) error codes from the same proxy the UE shall:

- Not automatically re-attempt registration towards the same proxy for an implementation specific time; and

- Attempt initial registration as described in clause 8.1 towards another proxy, if available.

4.4.3 Recovery to Primary Outbound Proxy

When the UE has to perform re-registration and the UE is currently registered towards the Secondary Outbound Proxy then the UE shall attempt initial registration towards the Primary Outbound Proxy unless a call has been established or is in the process of being set up.

If registered towards the Secondary Outbound Proxy and a call has been established or is in the process of being set up then re-registration shall be attempted towards the Secondary Outbound Proxy.

4.4.4 UE behaviour upon detection of a physical fault

Upon detection of a physical fault (e.g. fibre disconnected), the UE shall move to an unregistered state. Once the UE has detected that the fault has been rectified the UE shall attempt initial registration, as described in Clause 8.1, towards the Primary Outbound Proxy.

4.4.5 UE behaviour upon modification of Registration affecting parameters

The UE shall consider itself deregistered and attempt initial registration towards the Primary Outbound Proxy if any of the following parameters are changed:

- Public-user-Id
- Authentication
- Domain Name
- UE address
- Default Gateway address
- Proxy Server address currently used in an active registration

5 SIP Method support

Table 5.1 below lists the methods that shall be supported across the NGA-T UNI.

Table 5.1: SIP Method support

Method	**RFC**
ACK	3261 [2]
BYE	3261 [2]
CANCEL	3261 [2]
INVITE	3261 [2]
OPTIONS	3261 [2]
REGISTER	3261 [2]
PRACK	3262 [3]
NOTIFY	3265 [4]
UPDATE	3311 [5]
INFO	2976 [6]

6 SIP Header support

The support of SIP Headers shall comply with Tables 2 and 3 of RFC 3261 [2] with additions applied by the RFCs referenced in the present document.

7 SDP

The SIP UNI shall utilise the Session Description Protocol (SDP) as described in RFC 4566 [8]. Media capabilities shall be negotiated according to RFC 3264 [9].

Support of Silence Suppression and Comfort Noise is not required over this interface.

When operating with a G.711 codec, the support of DTMF Relay is optional. When DTMF Relay is enabled, the SDP offer shall indicate its support of DTMF Relay per RFC 4733 [10]. When DTMF Relay is disabled or is not negotiated with the far end, the DTMF digits shall be transmitted in-band in the G.711 RTP payload.

Support of T.38 Fax Relay is not required over this interface.

7.1 Packet time handling

The selection of the packetisation interval on the SIP UNI shall comply with RFC 3551 [17]. Specific UE behaviour is covered in the following sub-clauses.

7.1.1 Originating UE

7.1.1.1 SDP Offer sent by originating UE

Where the preferred Codec is G.711 A-law, the ptime attribute in the SDP Offer shall be set to 10 ms.

7.1.1.2 SDP Answer received at originating UE

Upon completion of SDP negotiation the UE should set its transmit RTP packetisation interval as shown in Table 7.1 below:

Table 7.1

Received ptime in SDP Answer	**UE transmit RTP packetisation interval**
Supported for negotiated Codec	Set to received ptime value
ptime not received	Set to default codec packetisation interval in RFC 3551 [17]
Not supported for the negotiated Codec	Set to default codec packetisation interval in RFC 3551 [17]

Where the SDP Answer received is G.711 A-law with ptime= 10 ms, the UE shall set the transmit RTP packetisation interval to 10 ms.

7.1.2 Terminating UE

7.1.2.1 SDP answer sent by terminating UE

Upon reception of an SDP Offer the UE transmit RTP packetisation interval and the ptime in the SDP Answer should be set as indicated in Table 7.2 below:

Table 7.2

Received ptime in SDP Offer	**ptime in SDP Answer**	**UE transmit RTP packetisation interval**
Supported for negotiated Codec	Set to Received ptime value	Set to Received ptime value
ptime not received	Set to default codec packetisation interval in RFC 3551 [17]	Set to default codec packetisation interval in RFC 3551 [17]
Not supported for the negotiated Codec	Set to default codec packetisation interval in RFC 3551 [17]	Set to default codec packetisation interval in RFC 3551 [17]

Where the negotiated codec is G.711 A-law and the SDP Offer ptime = 10 ms, the UE shall set the transmit RTP packetisation interval to 10 ms, and upon the completion of SDP negotiation, transmit RTP packets at 10 ms.

8 Procedures

The present clause provides details of the procedures with respect to the UE.

8.1 Registration, Re-registration, De-registration and authentication

TS 124.229 Clause	Title	Comment
5.1.1	Registration and authentication	UK: No change
5.1.1.1	General	UK: Modify the sub-clause as follows: Paragraph 1: Delete the text “(see table A.4/1 and dependencies on that major capability)”. Delete Note 1. Paragraph 2: Delete the paragraph which begins “In case a UE…” . Paragraph 3: Delete the last sentence which begins “When binding any…”. Paragraph 4: Delete the paragraph which begins “If SIP digest without TLS…”. Paragraph 5: Delete the paragraph which begins “In case a device performing…”.
5.1.1.1A	Parameters contained in the ISIM	UK: Void
5.1.1.1B	Parameters provisioned to a UE without ISIM or USIM	UK: Modify the sub-clause as follows: Delete “without ISIM or USIM” from the title. Copy the text from 5.1.1B.1 paragraph 2 which begins “The following IMS…” to the text “REGISTER request to.”.
5.1.1.1B.1	Parameters provisioned in the IMC	UK: Void
5.1.1.1B.2	Parameters when UE does not contain ISIM, USIM or IMC	UK: Void

TS 124.229 Clause	**Title**	**Comment**
5.1.1.2	Initial registration	UK: No change
5.1.1.2.1	General	UK: Modify the sub-clause as follows: Paragraph 1: Delete the second and third sentences beginning with the text “The UE can register…” to the text “…request has timed out.”. Paragraph 2: Delete the paragraph which begins “When registering any…”. Paragraph 3: Delete the paragraph which begins “ When binding any one…”. Paragraph 4: Delete the paragraph which begins “ If the UE detects…”. Paragraph 5: Replace the three instances of “P-CSCF” with “Outbound Proxy”. Delete Note 1 which begins “The UE will only…”. Paragraph 6: Delete the text “sub-clause 5.1.1.1A or” from the first sentence. Delete the second sentence beginning with the text “A public user…” to the text “….by the end user.”. Paragraph 7: Delete sentences 2 to 4 of item c) beginning with the text “If the UE supports…” to the text “….and RFC 3840 [62].”. Delete sentence 2 of item d) beginning with the text “For TCP, the…” to the text “….the request was sent.”. Replace the text in item e) beginning with the text “a registration expiration…” to the text “….of the registration.” with “an Expires header with the value set according to the provisioned value (default 3600 seconds).”. Delete “(S-CSCF)” from Note 3. Delete the following text from item g) beginning with the text “, and 1) if GRUU is…” to the text “….option tag “outbound””.

TS 124.229 Clause	Title	Comment
		Replace the text in item h) with the following "a Call-Id header with a freshly generated value." Paragraph 8: Delete the colon from the first sentence which begins "On receiving the 200 (OK)…". Replace ";" at the end of item a) with ".". Delete items b) to f) beginning with the text "store as the default…" to the text "….as defined in RFC 5626 [92].". Remove the bullet marks. Paragraph 9: Insert the text "ignore the content of the contact header field (in order to prevent security attacks) and" after "…the UE shall". Delete item a) which begins "release all IP CAN…". Replace "sub-clause 9.2.1" with "ND1033 Clause 4.3." in item b). Replace "the three instances of "P-CSCF" with "Outbound Proxy" in item b). Replace "P-CSCF" with "Outbound Proxy" in item c). Paragraph 12: Replace "sub-clause 9.2.1" with "ND1033 Clause 4.3."' in items a) and b). Replace all instances of P-CSCF" with "Outbound Proxy" in item a). Replace the first instance of "P-CSCFs" with "Outbound Proxies" and replace the second instance of "P-CSCF" with "Outbound Proxy" in item b). Paragraph 13: Delete the paragraph which begins with "After a maximum of 5…" and ends with "…a consequence of a failed reregistration". Paragraph 14: Delete the paragraph which begins with "These limits do not…".

TS 124.229 Clause	Title	Comment
5.1.1.2.2	Initial registration using IMS AKA	UK: Void
5.1.1.2.3	Initial registration using SIP digest without TLS	UK: No change
5.1.1.2.4	Initial registration using SIP digest with TLS	UK: Void
5.1.1.2.5	Initial registration using NASS-IMS bundled authentication	UK: Void
5.1.1.2.6	Initial registration using GPRS-IMS bundled authentication	UK: Void
5.1.1.3	Subscription to the registration-state event package	UK: Void
5.1.1.3A	Subscription to the debug event package	UK: Void
5.1.1.4	User-initiated reregistration and registration of an additional public user identity	UK: Modify the sub-clause as follows: Delete “and registration of an additional public user identity” from the title.
5.1.1.4.1	General	UK: Modify the sub-clause as follows: Paragraph 1: Delete the paragraph which begins “The UE can perform the reregistration of a previously registered public user identity bound …”. Paragraph 2: Delete the paragraph which begins “The UE can perform the reregistration of a previously registered public user identity over any …”. Paragraph 3: Delete the paragraph which begins “The UE can perform the reregistration of a previously registered public user identity via an initial…”.

TS 124.229 Clause	Title	Comment
		Paragraph 4: Delete the paragraph which begins "The UE can perform registration of additional public user identities at any time after…". Paragraph 5: Delete the text beginning with ", or when the UE intends…" to the text "….media feature tag". Paragraph 6: Delete the paragraph beginning with the text "When sending a…". Paragraph 7: Delete "sub-clause 5.1.1.A or" from the paragraph which begins with the text "The UE shall extract…". Paragraph 8: Delete the text in item c) beginning with ". and containing the instance…" to the text "….sub-clause 7.9.3 and RFC 3840 [62].". Delete the text in item d) beginning with ". For the TCP…" to the text "….the request was sent". Replace the text in item e) beginning with the text "a registration expiration interval…" to the text "….of the registration" with "an Expires header with the value set according to the provisioned value (default 3600 seconds)". Delete "(S-CSCF)" from Note 1. Delete the text in item g) beginning with ", and if GRUU is…" to the text "….option tag "GRUU"; and". Insert "." at the end of item g). Delete item h) beginning with the text "if available to the…" to the text "….sub-clause 7.2A.4)." Paragraph 9: Delete the colon after the text "…, the UE shall". Delete items b) and c) beginning with the text "store the list…" to the text "….that was registered." Remove the bullet marks. Paragraph 10: Replace the reference "5.1.1.5.1" with "5.1.1.5.4".

TS 124.229 Clause	**Title**	**Comment**
		Paragraph 13: Insert the text "ignore the content of the contact header field (in order to prevent security attacks) and" after "…the UE shall". Delete item a) beginning with the text "release all IP CAN…" to the text "….in sub-clause 9.2.2;" Replace "sub-clause 9.2.1" with "ND1033 clause 4.3."' in paragraph 13 item b). Replace all instances of "P-CSCF" with Outbound Proxy" in item b). Replace "a P-CSCF" with "an Outbound Proxy" in item c). Paragraph 14: Delete item 2) beginning with the text "after releasing all…" to the text "….in sub-clause 9.2.2." Replace "sub-clause 9.2.1" with "ND1033 clause 4.3" in paragraph 14 item a). Replace "P-CSCFs" with "Outbound Proxies" and "P-CSCF" with "Outbound Proxy" in item a). Replace "sub-clause 9.2.1" with "ND1033 clause 4.3" in paragraph 14 item b). Replace all instances of "P-CSCF" with "Outbound Proxy" in item b). Replace ", and" with "." in item c). Delete item d) beginning with the text "the UE shall perform…" to the text "….RFC 5626 [92] shall apply.".
5.1.1.4.2	IMS AKA as a security mechanism	UK: Void
5.1.1.4.3	SIP digest without TLS as a security mechanism	UK: No change
5.1.1.4.4	SIP digest with TLS as a security mechanism	UK: Void

TS 124.229 Clause	Title	Comment
5.1.1.4.5	NASS-ISM bundled authentication as a security mechanism	UK: Void
5.1.1.4.6	GPRS-ISM bundled authentication as a security mechanism	UK: Void
5.1.1.5	Authentication	UK: No change
5.1.1.5.1	IMS AKA - general	UK: Void
5.1.1.5.2	Void	UK: No change
5.1.1.5.3	IMS AKA abnormal cases	UK: Void
5.1.1.5.4	SIP digest without TLS – general	UK: Modify the sub-clause as follows: Paragraph 2: Replace “S-CSCF” with “Registrar”.
5.1.1.5.5	SIP digest without TLS – abnormal procedures	UK: Modify the sub-clause as follows: Paragraph1: Replace the text which begins “On receiving a 403…” with “Refer to ND1033 clause 4.4.2.3.”.
5.1.1.5.6	SIP digest with TLS – general	UK: Void
5.1.1.5.7	SIP digest with TLS –abnormal procedures	UK: Void
5.1.1.5.8	NASS-IMS bundled authentication – general	UK: Void
5.1.1.5.9	NASS-IMS bundled authentication – abnormal procedures	UK: Void
5.1.1.5.10	GPRS-IMS bundled authentication – general	UK: Void
5.1.1.5.11	GPRS-IMS bundled authentication – abnormal procedures	UK: Void

TS 124.229 Clause	**Title**	**Comment**
5.1.1.5.12	Abnormal procedures for all security mechanisms	UK: Modify the sub-clause as follows: Paragraph 1: Replace the text which begins “A UE shall only respond…” with “Refer ro ND1033 clause 4.4.2.3.”.
5.1.1.5A	Network-initiated re-authentication	UK: Void
5.1.1.5B	Change of IPv6 address due to privacy	UK: Void
5.1.1.6	User-initiated deregistration	UK: No change
5.1.1.6.1	General	UK: Modify the sub-clause as follows: Paragraph 1: Delete the second sentence beginning with the text “The UE shall protect…” to the text “….,if one is available.”. Paragraph 2: Delete ‘sub-clause 5.1.1.1A or” from the paragraph which begins “The UE shall extract…”. Paragraph 3: Delete the second sentence beginning with the text “However: -if the dialog…” to the text “….not release this dialog.”. Paragraph 4: Delete the text in item c) beginning with the text “, and containing the Instance…” to the text “….described in RFC 5626 [92]”. Insert “and” after “requirements of the user;” in item e). Replace “;” with “.” at the end of item f). Delete item g) beginning with the text “if available to the UE…” to the text “….see sub-clause 7.2A.4.”. Paragraph 5: Delete the paragraph beginning with the text “For a public user…”. Paragraph 6: Delete the paragraph beginning with the text “The UE can deregister…”. Paragraph 7: Replace the reference “5.1.1.5.1” with “5.1.1.5.4”.

TS 124.229 Clause	**Title**	**Comment**
		Paragraph 9: Delete the paragraph beginning with the text "If there are no more public…". Paragraph 10: Delete the paragraph beginning with the text "If all public user…".
5.1.1.6.2	IMS AKA as a security mechanism	UK: Void
5.1.1.6.3	SIP digest without TLS as a security mechanism	UK: No change
5.1.1.6.4	SIP digest with TLS as a security mechanism	UK: Void
5.1.1.6.5	NASS-ISM bundled authentication as a security mechanism	UK: Void
5.1.1.6.6	GPRS-ISM bundled authentication as a security mechanism	UK: Void
5.1.1.7	Network-initiated deregistration	UK: Void

8.2 Generic procedures for all Methods except Register

hhhh

TS 124.229 Clause	Title	Comment
5.1.2A	Generic procedures applicable to all methods excluding the REGISTER method	UK: No change
5.1.2A.1	UE-originating case	UK: No change
5.1.2A.1.1	General	UK: modify the sub-clause as follows: Paragraph 2: Delete the paragraph which begins “When the UE sends…”. Paragraph 3: Delete the paragraph which begins “If SIP digest without TLS…”. Paragraph 5: Delete the paragraph which begins “Where a security association…” . Paragraph 6: Delete the paragraph which begins “In accordance with RFC 3325…”. Paragraph 7: Delete the paragraph which begins “When sending any initial request…”. Paragraph 8: Delete the paragraph which begins “When privacy is required…”. Paragraph 9: Delete the paragraph which begins “The UE shall determine…”. Paragraph 10: Delete the paragraph which begins “The UE shall not include…”. Paragraph 11: Replace the text beginning with “as follows…” until “…that applies for the dialogue” with “with a SIP URI containing the contact address of the UE.”. Paragraph 13: Delete the paragraph which begins “If the UE support…”. Paragraph 14: Delete the paragraph which begins “If this is a request…”.

TS 124.229 Clause	Title	Comment
		Paragraph 15: Delete the paragraph which begins “If an IMS application…”. Paragraph 16: Delete the paragraph which begins “After the dialog is…”. Paragraph 17: Delete the paragraph which begins “The UE can indicate...”. Paragraph 18: Delete the paragraph which begins “If resource priority…”. Paragraph 19: Delete the paragraph which begins “If available to the…”. Paragraph 20: Delete the text from starting with “a) the P-CSCF URI…” until the end of clause 5.1.2A.1.1 ending with “…sub-clause 5.1.6.10 apply”. Insert the following text: “a) the Outbound Proxy which contains the FQDN or the IP address which has been learnt through the Outbound Proxy discovery procedures; and b) the Outbound Proxy port which has been used during the registration procedure. The UE shall support receipt of forked responses in accordance with RFC 3261 [2]. NOTE: The UE can accept or ignore any of the forked responses, for example, if the UE is capable of supporting a limited number of simultaneous transactions or early dialogs.”.
5.1.2A.1.2	Structure of Request-URI	UK: Refer to clause 4.1 of ND1033
5.1.2A.1.3	UE without dial string processing capabilities	UK: No change
5.1.2A.1.4	UE with dial string processing capabilities	UK: No change
5.1.2A.1.5	Setting the ‘phone-context’ tel URI	UK: Void

TS 124.229 Clause	Title	Comment
	parameter	
5.1.2A.1.6	Abnormal cases	UK: Modify the sub-clause as follows: Replace the text with the following: "The UE shall initiate restoration procedures by performing an initial registration sub-clause 5.1.1, as modified by ND1033 clause 8.1 when: a) the UE receives a 403 (forbidden) response; or b) the UEs SIP request receives no response and associated SIP timer B expires.".
5.1.2A.2	UE-terminating case	UK: Modify the sub-clause as follows: Paragraphs 2 to 16: Delete the text which begins "Where a security association…" to the text "…log the request or response.". Insert the following text as a new paragraph after Paragraph 1: "The UE shall include, in the address of the Contact header field, the unprotected port value used in the initial registration.".

9 SIP Session Timers

9.1 General

The SIP UNI shall support the SIP Session Timer procedures defined in RFC 4028 [11]. The NGA-T Service Provider network is responsible for determining on a per call basis whether to invoke session timers.

A UA which has the refresher role should use SIP UPDATE requests rather than re-INVITE requests to perform session refreshes, provided that the peer has included the UPDATE method in its ALLOW header. The UPDATE request should not contain an SDP offer, but a re-INVITE shall contain one, even if the details of the session have not changed.

9.1.1 Actions at the UE

If the UE supports session timer, it must include SIP “Supported: timer” and “Session-Expires” headers in all SIP INVITE and UPDATE requests, and 200 OK responses to those requests. The value of the Session-Expires header SHOULD be at least 1800 seconds. The “Min-SE” header, with a value greater than or equal to 90 seconds must be included in all SIP INVITE and UPDATE requests, and in 422 (session interval too small) responses.

When sending an INVITE request that includes “Supported: timer”, the Session-Expires header should not include a “refresher” parameter.

9.1.2 Actions at the NGA-T Service Provider Network

In dialogues for which SIP Session Timer procedures are not used, the NGA-T Service Provider may fall back to a simple method of sending periodic OPTIONS requests to verify that the dialogue is still active.

10 Services

If the SIP ua-profile provisioning mechanism is supported then, in order to ensure that the UE has the most up-to-date ua-profile information, the NGA-T Service Provider shall send a NOTIFY with a ua-profile XML body as described in Annex A of TS 183 043 [15] each time the NGA-T Service Provider completes an initial registration request from the UE.
Where the SIP ua-profile provisioning mechanism is not supported then settings are pre-provisioned as per prior agreement between the access and service provider.

10.1 Malicious Call

10.1.1 Provisioning

When the Malicious Call service is provisioned, if the SIP ua-profile provisioning mechanism is supported then the NGA-T Service Provider shall send a NOTIFY request containing a ua-profile XML body as described in Annex A of TS 183 043 [15] with the mcid-service element set to mcid-service-provisioned.

When the Malicious Call service is withdrawn , if the SIP ua-profile provisioning mechanism is supported then the NGA-T Service Provider shall send a NOTIFY request containing a ua-profile XML body as described in Annex A of TS 183 043 [15] with the mcid-service element set to mcid-service-withdrawn.

10.1.2 Procedures

10.1.2.1 Actions at the UE

The UE behaviour is described in the appropriate annex.

10.1.2.2 Actions at the NGA-T Service Provider network

When the Malicious Call service is provisioned and the NGA-T Service Provider receives a re-INVITE request with either no XML body or an XML mcid body with MCID XML Request schema as per 3GPP 24.616 [19] containing a McidRequestIndicator set to 1, the NGA-T Service Providers call server shall:

- Record appropriate information including the date and time and calling and called party identities.
- Take appropriate actions to inform the Service Provider or relevant authorities.

The NGA-T Service Provider network must delay the sending of the BYE to the MCID enabled ATA for 5 seconds thus allowing the called user to activate the service. The NGA-T Service Provider network should release the ingress call leg as normal.

10.2 Hotline

10.2.1 Provisioning

When the Hotline service is provisioned, if the SIP ua-profile provisioning mechanism is supported then the NGA-T Service Provider shall:

- Store the address to be used for hotline call setup.
- Send a NOTIFY request containing a ua-profile XML body as described in Annex A of TS 183 043 [15] with the no-dialling-behaviour element set to immediateCallSetup.

When the Hotline service is withdrawn, if the SIP ua-profile provisioning mechanism is supported then the NGA-T Service Provider shall send a NOTIFY request containing a ua-profile XML body as described in Annex A of TS 183 043 [15] with the no-dialling-behaviour element set to rejectCall.

10.2.2 Procedures

10.2.2.1 Actions at the UE

When the Hotline service is provisioned and the user initiates call setup, the UE shall immediately send an INVITE with the request-URI set to “sip:hotline@<domain>”as specified in clause 4.1.1.

10.2.2.2 Actions at the NGA-T Service Provider network

When the NGA-T Service Providers network receives an INVITE request with the request-URI set to “sip:hotline@<domain>” it shall retrieve the stored hotline address and use basic call setup procedures to establish a call with this destination.

10.3 Call Forward Unconditional

10.3.1 Provisioning

Call Forward Unconditional is performed in the NGA-T Service Provider’s network and so no provisioning actions take place across the SIP UNI.

10.3.2 Procedures

10.3.2.1 Actions at the UE

Upon receipt of a SIP NOTIFY containing an XML body as per Annex A of TS 183 043 [15] from the NGA-T Service Provider indicating that Call Forward Unconditional is active the UE should inform the user.

10.3.2.2 Actions at the NGA-T Service Provider network

When an INVITE is received addressed to a subscriber that has Call Forward Unconditional provisioned, the NGA-T shall invoke its procedure to divert the call to the forwarded to party. When a subscriber has activated or de-activated the Call Forward Unconditional service , the NGA-T Service Provider’s network shall update the UE’s Dial Tone Management by means of sending a NOTIFY request with a ua-profile XML body as described in Annex A of TS 183 043 [15].

10.4 Three Way Call

Note: Network-based conferencing is for further study.

10.4.1 Provisioning

When the Three Way Call service is provisioned, if the SIP ua-profile provisioning mechanism is supported then the NGA-T Service Provider shall send a NOTIFY request containing a ua-profile XML body as described in Annex A of TS 183 043 [15] with the following values:

Element	Value
hold-service	hold-service-provisioned
toggle-service	toggle-service-provisioned
three-pty-service	three-pty-service- provisioned

When the Three Way Call service is withdrawn, if the SIP ua-profile provisioning mechanism is supported then the NGA-T Service Provider shall send a NOTIFY request containing a ua-profile XML body as described in Annex A of TS 183 043 [15] with the following values:

Element	Value
hold-service	hold-service-withdrawn (Note 1)
toggle-service	toggle-service-withdrawn
three-pty-service	three-pty-service-withdrawn
Note 1: If the Call Waiting service is provisioned then the hold-service shall not be withdrawn	

10.4.2 Procedures

10.4.2.1 Actions at the UE

The UE behaviour is described in the appropriate annex.

10.4.2.2 Actions at the NGA-T Service Provider network

The hold/resume procedures are implemented using the procedures in RFC 3264 [9].

10.5 Call Waiting Service

10.5.1 Provisioning

When the Call Waiting service is provisioned, if the SIP ua-profile provisioning mechanism is supported then the NGA-T Service Provider shall send a NOTIFY request containing a ua-profile XML body as described in Annex A of TS 183 043 [15] with the following values:

Element	Value
hold-service	hold-service-provisioned
cw-service	cw-service-provisioned

When the Call Waiting service is withdrawn, if the SIP ua-profile provisioning mechanism is supported then the NGA-T Service Provider shall send a NOTIFY request containing a ua-profile XML body as described in Annex A of TS 183 043 [15] with the following values:

Element	Value
hold-service	hold-service-withdrawn (Note 1)
cw-service	cw-service-withdrawn
Note 1: If the Three Way Call service is provisioned then the hold-service shall not be withdrawn	

10.5.2 Procedures

10.5.2.1 Actions at the UE

The UE behaviour is described in the appropriate annex.

10.5.2.2 Actions at the NGA-T Service Provider network

When the Call Waiting service is provisioned and the NGA-T Service Provider receives a call setup request for a subscriber that is involved in a single established call, the NGA-T Service Providers call server shall:

- Start a network CW timer. This timer should be in the range as specified in TS 24.615 [24] and less than the UE-CW timer.
- Use basic call procedures to setup a second SIP dialogue with the subscribers UE.
- Cancel the network CW timer on receipt of a final response on the dialogue for the call waiting call.

On expiry of the Network CW Timer, the NGA-T Service Provider shall send a CANCEL for the dialogue for the call waiting call and follow the procedures as specified in TS 24.615 [24].

The hold/resume procedures are implemented using the procedures in RFC 3264 [9].

10.6 Caller Display Service (CLIP/CLIR)

The Caller Display service must comply with ND1016 [23] and should be implemented according to the minimum functionality described in 3GPP 24.607 [22] needed to meet this.
Note: the display-name field of a header may contain the same information as the URI.

Annex <A> (normative): ATA specific functionality

A.1 General

A.1.1 Event Notifications

The ATA shall support Implicit Subscription for the following Event packages:

- ‘ua-profile” Event (draft-ietf-sipping-config-framework-17 [16]) and support of the Profile document defined in Annex A of ETSI TS 183 043 [15].
- Message Summary Event defined in RFC 3842 [13].

Upon successful registration, the ATA shall be prepared to receive NOTIFY events from the above event packages.

A.1.1.1 UA Profile Notification

The ATA shall support receipt of a NOTIFY request with a ua-profile XML body as described in Annex A of TS 183 043 [15]. The ATA shall process the received XML elements as described in sections A.1.1.1.1 and A.1.1.1.2 and receipt of any other ua-profile XML element should be ignored.

UA-profile information shall be discarded when the ATA performs initial registration as per clause 8.1.

A.1.1.1.1 Dial-Tone-Management XML Element

On receipt of a NOTIFY request containing a ua-profile XML body with a “dial-tone-management” element the ATA shall update its storage of the last received dial-tone-pattern value.

Table A.1.1: dial-tone-pattern value enumerations

XML dial-tone-pattern value
standard-dial-tone
special-condition-tone
message-waiting-tone (note 1)

Note 1: RFC 3842 [13] shall be used for message waiting notification, therefore the XML element dial-tone-pattern is not expected to be included the message-waiting-tone value.

The default dial-tone pattern is “standard-dial-tone”.

A.1.1.1.2 Service Provisioning XML Elements

The following ua-profile XML elements are used for service provisioning in the ATA:

- mcid-service
- no-dialling-behaviour
- hold-service
- toggle-service
- three-party-service
- cw-service

On receipt of a NOTIFY request containing a ua-profile XML as described in Annex A of TS 183 043 [15] body with a one or more of these elements the ATA shall update its provisioned state for each received element.

Table A.1.2 below shows the mapping of XML elements to NICC services.

Table A.1.2: Determination of service state

NICC Service	**XML Elements required (see Note 1)**	**Service Provisioned** (xml element values)	**Service Withdrawn** (xml element values)
Call Waiting	“hold-service” “cw-service”	“hold-service-provisioned” and “cw-service-provisioned”	“hold-service-withdrawn” (Note 2) or “cw-service-withdrawn”
3 Way Call	“hold-service” “toggle-service” “three-pty-service”	“hold-service-provisioned” and “toggle-service-provisioned” and “three-pty-service-service-provisioned”	“hold-service-withdrawn” (Note 2) or “toggle-service-withdrawn” or “three-pty-service-withdrawn”
Malicious Caller Identify (see Note 3)	"mcid-service”	"mcid-service-provisioned"	"mcid-service-withdrawn"
Hotline	"no-dialling-behaviour"	"immediateCallSetup”	"rejectCall"
Note 1: It is recommended that when a ua-profile is sent that it includes all elements in this column. Note 2. If another service which requires hold is active then “hold-service” shall not be withdrawn. Note 3: When the Malicious Caller Identity service is provisioned, it shall take precedence over the Call Waiting and 3 Way Call Services.			

A.1.1.1.3 ATA behaviour when a ua-profile is not supported or has otherwise not been received

Upon successful initial registration, if a ua-profile had not been received, then the ATA shall provide:

- The XML based dial-tone state set to normal

- default service settings as agreed with the NGA-T Access Provider.

A.1.1.2 Message Waiting Notification

The ATA shall support Implicit Subscription to the Message Summary Event defined in RFC 3842 [13].

On receipt of a NOTIFY request reporting the “message-summary” event with a “Message-Waiting” field set to ‘yes’, the ATA shall:

- Update its MWI state to active

On receipt of a NOTIFY request reporting the “message-summary” event with a “Message-Waiting” field set to ‘no’, the ATA shall:

- Update its MWI state to inactive

The default MWI state is inactive.

A.1.1.3 Dial Tone Application

When a subscriber goes off hook the ATA shall:

- Retrieve the ua-profile XML based dial-tone-pattern value state stored as per section A.1.1.1.1
- Retrieve the MWI state stored as per section A.1.1.2
- Apply the appropriate dial tone indication shown in Table A.1.3 below

Table A.1.3: Determination of Dial Tone to be Applied

MWI State	Ua-profile XML Stored dial-tone-pattern value	Dial Tone
Inactive	standard-dial-tone	Standard Dial Tone
Inactive	special-condition-tone	Special Dial Tone
Inactive	message-waiting-tone	Special Dial Tone
Active	Don’t care	Special Dial Tone

A.1.2 SIP Provisional Responses

The ATA shall support Reliable Provisional Responses as per RFC 3262 [3].

A.1.2.1 Terminating ATA procedures

Upon receipt of an INVITE containing:

1. a Supported or Require header with value ‘100rel’
2. PRACK in the allow header, and
3. an SDP offer

the ATA shall send a 180 (ringing) response reliably (see RFC 3626 [3]) without SDP.

A.1.2.2 Originating ATA procedures

When generating an INVITE request the originating ATA shall include:

1. ‘100rel’ in the Supported header,
2. PRACK in the Allow header, and
3. a P-Early-Media header set to ‘supported’.

Upon receipt of a 18x provisional response containing a Require header with tag ‘100rel’ then the ATA shall respond with a PRACK as described in RFC 3262 [3].

A.1.3 SIP Forked Responses Handling

The ATA shall support forked responses as defined in RFC 3261 [2].

SDP offer/answer and P-Early-Media rules are applied to each early dialogue independently.

A.1.4 SIP digit Sending

A.1.4.1 General

The Dialled Digits sending method shall be configurable via management as one of the following:

- En-bloc sending;
- Overlap sending In-Dialog method
- Overlap sending Multiple INVITE method

A.1.4.2 En-bloc Sending

A digit map is configured on a per port basis using the structure defined in ND1646 [18].
The en-bloc timers are described in Table A.1.4 below.

Table A.1.4: En-bloc Signalling Timers for interworking

Symbol	Time-out value	Cause for initiation	Normal termination	At expiry
T-FirstDigit	1 s to 99 s (default=20 s)	Upon start of dial tone injection	On receipt of the first dialled digit	Release Call (Note 1)
T-InterDigit	1 s to 20 s (default=20 s)	Upon receipt of an additional dialled digit, provided that for the given sequence of digits there is no current match but one or more matches are possible upon receipt of additional digits.	On receipt of a subsequent dialled digit.	Release Call (Note 4)
T-ShortDigit	4 s to 6 s (default=4 s) (Note 3)	If a matched pattern is followed by the symbol for ShortDigit timer in the digit map.	Upon receipt of a subsequent digit.	1) Send an Invite message with all the digits received thus far. 2) Disable Digit Receiver. (Note 2)

Symbol	Time-out value	Cause for initiation	Normal termination	At expiry
Note 1: The announcement 'General' (see A.1.7.2) shall be injected. Note 2: If the Call Server replies to the Invite with 4xx (including 404, 484), 5xx or 6xx the ATA shall inject a Tone/Announcement. Note 3: T-ShortDigit timer must be less than the T-InterDigit timer. Note 4: The announcement 'Unrecognised number' (see A.1.7.2) shall be injected.				

Upon sending an INVITE message the ATA shall not send any further digits in subsequent INVITEs. Where the received dialled digits can never result in a match then the ATA shall inject the announcement "Unrecognised number'.

When the Hash digit received is part of a Service Activation/De-Activation/Check Code, the ATA shall not remove the Hash digit. In all other cases the Hash digit indicates end of dialling, the ATA shall remove the Hash and disable the DTMF receiver.

When sending the Hash digit character in the INVITE message it shall be escaped as per RFC 3261 [2].

A.1.4.3 SIP Overlap Sending

The ATA shall be provisioned with a Dial Plan. The digit map structure format is specified in ND1646 [18]. As soon as there is a match with a digit map pattern, the ATA shall send an INVITE with the digits gathered thus far.

The Overlap signalling procedures for the In-Dialog method shall comply with TS 183 043 Annex F.3. [15].

The Overlap signalling procedures for the Multiple INVITE method shall comply with TS 183 043 Annex F.2. [15].

The Timers for interworking shall comply with TS183 043 Annex F.4. [15].

A.1.5 Distinctive Ringing / CW Alerting

An Alert-Info header may be included in an INVITE message sent from the NGA-T Service Provider to the ATA to control one of the following:

- the Ringing Current Cadence of the alerting signal sent on the analogue line; or

- the Frequency & Cadence of the Call Waiting Tone bursts sent on the analogue line that are used to alert the customer that a new call has arrived.

If the Alert-Info header is not included in the INVITE message then the default value of the Cadence Code shall be '01'. (See Table A.1.5 below).

The absoluteURI portion of the Alert-Info header shall be constructed with

- scheme set to "http"
- a hier-part with net-path set to www.uktel.org.uk/SIPAlertInfoExtns and a query portion containing a value conforming to the following ABNF syntax for CadInd:

```
CadInd = %x52.43 code                      ; RCxx
       / %x43.57.54 code "-" interval      ; CWTxx-yyyy
code = 2HEXDIG
interval = NZDIGIT 3DIGIT
NZDIGIT = %x31-39                          ; non-zero digit
```

The interpretation of the cadence is shown in Table A.1.5 below.

Table A.1.5 - Alert Cadence Indicator values

<table>
<tr>
<td>Alerting Cadence Indicator</td>
<td>RCxx or CWTxx-yyyy
Where xx is the Cadence Code (ASCII-Hexadecimal coded) and where yyyy is the interval (ms) between successive Call Waiting Tone pulses (allowable range 1000 – 9999, default 5000).
The coding of the Cadence Code is shown below:
<table>
<tr><td>Cadence Code (xx)</td><td>Ringing Current Cadence</td></tr>
<tr><td>00</td><td>Not Used – treat as value 01</td></tr>
<tr><td>01 (default)</td><td>0.4sec On, 0.2sec Off, 0.4sec On, 2.0sec Off</td></tr>
<tr><td>02</td><td>0.4sec On, 0.8 sec Off</td></tr>
<tr><td>03</td><td>0.25sec On, 0.25sec Off, 0.25sec On, 0.25sec Off, 0.25sec On, 1.75sec Off</td></tr>
<tr><td>04</td><td>2.0sec On, 4.0sec Off</td></tr>
<tr><td>05</td><td>Continuous ringing</td></tr>
<tr><td>06</td><td>1.0sec On, 2.0sec Off</td></tr>
<tr><td>07</td><td>No ringing current.</td></tr>
<tr><td>08 to FF</td><td>Not Used – treat as value 01</td></tr>
<tr><td>Cadence Code (xx)</td><td>Call Waiting Tone - Frequency & Duration of Pulses (interval between successive pulses determined by yyyy field)</td></tr>
<tr><td>00</td><td>Not used – treat as value 01</td></tr>
<tr><td>01 (default)</td><td>400Hz for 0.1 sec (this timing is controlled by the ATA)</td></tr>
<tr><td>02</td><td>400Hz for 0.03sec On, 0.01sec Off, 0.03sec On (these timings are controlled by the ATA)</td></tr>
<tr><td>03 to FF</td><td>Not used – treat as value 01</td></tr>
</table>
</td>
</tr>
</table>

Examples:

a) Alert-info: <http://www.uktel.org.uk/SIPAlertInfoExtns?RC01>

This Alert-Info header indicates the default Ringing Cadence 01

b) Alert-info: <http://www.uktel.org.uk/SIPAlertInfoExtns?CWT01-5000>

This Alert-info header indicates the default Call Waiting Tone cadence 01, with CW tone bursts spaced at intervals of 5 seconds.

A.1.6 Early Media and Ringback Tone

The originating ATA shall not provide a forward media path until receipt of 200 OK (invite) unless explicitly instructed by the receipt of a P-Early-Media header.

A.1.6.1 Early Media

The ATA shall support the P-Early-Media header as described in IETF RFC 5009 [12].

For originating (outgoing) calls the INVITE request shall include a P-Early-Media header field including the parameter “supported”.

On receipt of a 18X response when the early media authorisation status, according to RFC 5009 [12], is either “Backward” or “Backward and Forward”, the ATA shall:

a) stop generation of local ringback tone (if previously started)
b) play-out RTP received early media towards local subscriber.

The receipt of a non-180 18x with SDP and without a P-Early-Media header, shall cause the ATA to open the path in the backward direction.

Receipt of a non-180 18x (no SDP) and without P-Early-Media shall be ignored.

A.1.6.2 Local Ringback Tone

On receipt of a 180 (ringing) (with or without SDP) the ATA shall play a Local Ring (Ringback) Tone towards the analogue line unless the early media authorisation status, according to RFC 5009 [12], is either “Backward” or “Backward and Forward”.

A.1.7 Tones and announcements

A.1.7.1 Call Progress Tones

The ATA shall support generation of the following call progress tones:

Table A.1.6: Call Progress Tones

Call Progress Tone	Comments
Dial Tone (Proceed Indication)	
Special Dial Tone (Special Proceed Indication)	Stutter Dial Tone
Ring Tone (Ring-back tone) (Awaiting Answer Indication)	
Busy Tone (Number Engaged Tone)	
Congestion Tone (Path Engaged Tone)	
Number Unobtainable tone (NU)	
Howler	
Call Waiting Indication	(default cadence)
Special Call Waiting Indication (special cadence)	Cadence is indicated in Alert-Info header
Special Information Tone (SIT)	

A.1.7.2 Local Announcements

The ATA shall have capability to store the announcements listed below. Some of the listed announcements shall be preceded by Special Information Tone (SIT) as indicated in the table below. The ATA shall support play-out of the following announcements towards the analogue subscriber line:

Table A.1.7: List of Announcements

Signal Name	SIT	Words
Unrecognised number	Y	The number you have dialled has not been recognised. Please check and try again
Fault	Y	Sorry there is a fault. Please try again.
No reply	Y	Sorry, there is no reply.
All lines busy	Y	Sorry, lines are busy. Please try later.
Call cannot be connected	Y	Sorry, your call cannot be connected at present. Please try again.
General	N	Please hang up and try again
Isolation	N	Sorry, you may not make calls from this line at the moment. Please contact your Telephone Service Provider
Anonymous Call Reject	N	The person you are calling is not accepting anonymous calls. Please re–dial without withholding your number.
Other user cleared	N	The other person has hung up. You will be reconnected to your call.
Invalid switching order received	N	You have dialled the code incorrectly. You will re-connected to your call
Supplementary	N	You will be re-connected to your call

Signal Name	SIT	Words
service re-connect		
Supplementary service call hold failure	Y	Sorry there is a fault. Please hang up.

A.1.7.3 SIP Final response to Tone / Announcement mappings

Upon reception of a SIP final response to an initial INVITE request the ATA shall inject the following tone / announcement towards the analogue subscriber line:

Table A.1.8: Final failure code to tone/announcement mapping

SIP Response Code	Audible Tone or Announcement
401 Unauthorized	None
402 Payment Required	Call cannot be connected announcement
403 Forbidden	Call cannot be connected announcement.
404 Not Found	En-bloc Sending mode: Unrecognised number announcement Overlap Sending mode: None
405 Method Not Allowed	Call cannot be connected announcement
406 Not Acceptable	Call cannot be connected announcement
407 Proxy Authentication Required	None
408 Request Time-out	No reply announcement.
423 Interval Too Brief	Call cannot be connected announcement
433 Anonymity Disallowed	Anonymous Call Reject announcement.
484 Address Incomplete	En-bloc Sending mode: Unrecognised number announcement Overlap sending mode: None (see A.1.4.3)
485 Ambiguous	Unrecognised number announcement
486 Busy Here	Number Engaged Tone
500 Server Internal Error	Path Engaged Tone
503 Service Unavailable	Path Engaged Tone
504 Server Time-out	Fault announcement
600 Busy Everywhere	Number Engaged Tone
606 Not Acceptable	Call cannot be connected announcement
Any Other Response Code	Number Unobtainable tone (NU tone)
Response Code not expected by the protocol	Number Unobtainable tone (NU tone)

A.1.7.4 Mapping of other events to Tones / Announcements

The ATA shall map the following events to tone/announcement towards the analogue line as follows:

Table A.1.9: Other events to tone/announcement mapping

Event	Tone / Announcement
Reception of 180 Ringing without	Ring-back tone

Event	Tone / Announcement
early media authorization	
Reception of INVITE during existing conversation (See A.2.1)	Call Waiting Tone
BYE	Number Unobtainable tone (NU tone)
CANCEL	Number Unobtainable tone (NU tone)
Session Timer Expiry	Number Unobtainable tone (NU tone)
ATA is unregistered and subscriber goes Off-Hook	Isolation announcement
SIP Timer B expiry	Call cannot be connected announcement
ETSI TS 183 043 [15] overlap timers Ta2 or Ta3 expiry	Unrecognised Number announcement

A.1.8 Flash Hook Management

ETSI TS 183 043 Clause	Title	Comment
B.4.2	Flash Hook Management	UK: No change
B.4.2.1	Void	UK: No change
B.4.2.2	Flash Hook Management for analogue access	UK: No change
B.4.2.2.1	General rules	UK: Modify the text as follows: Paragraph 1: Insert “are” after “Flash-hook events”. Replace “AGCF/VGW” with “ATA”. Delete the text “are reported to the feature manager using a **Feature Request** internal primitive”. Paragraph 2 Replace the text “two different methods exist:” with “the Loose Coupling method is used.”. Delete bullets 1 and 2. Paragraph 3: Delete the paragraph which begins “Among others,…”. Paragraph 4: Delete the paragraph which begins “The execution of the…”. Paragraph 5: Delete the text which begins “Both methods are…” to “characterized as follows:”. In the Loose Coupling bullet replace the two instances of “AGCF/VGW” with “ATA”.

ETSI TS 183 043 Clause	Title	Comment
		In the first sentence of the Loose Coupling bullet replace “register recall” with “flash-hook”. In the second sentence of the Loose Coupling bullet replace “register recall” with “receipt of a flash-hook”. In the third sentence replace “AS” with “NGA-T Service Provider” . Delete the bullet heading “Loose Coupling”. Delete the whole of the second bullet which begins with “Tight Coupling:.”. Remove the bullet markings. Paragraph 6: Delete the paragraph which begins “The decision to…”.
B.4.2.2.2	Loose Coupling Procedures	UK: Modify the text as follows: Paragraph 3: Replace the text “SIP UA” with “ATA”. Paragraph 4: Delete the paragraph which begins “On receipt of a **feature request**…”. Paragraph 5: Replace the text starting with “If the digit collection…” until the text “…Otherwise, the events” with “On receipt of a flash-hook, the event”. Paragraph 7: Replace the text “SIP UA” with “ATA”. Insert the following text below the paragraph: “If MCID is provisioned the ATA shall proceed to option 2, otherwise it shall proceed to option 1.” Paragraph 8: Format the paragraph as list item “1.” and indent all bullets within this paragraph. Delete the text from the paragraph starting with “from the MGC component…” until “…profile delivery procedure, ”.

ETSI TS 183 043 Clause	Title	Comment
		Replace the text “Feature Manager” with “ATA”. Delete the text “Request the SIP UA to ” from the first bulleted item. Replace the text “AS” in Note 1 with “NGA-T Service Provider”. Delete the text “Request the MGC component to ” from the second bulleted item. Insert the following text below the paragraph: “2. On receipt of a flash-hook notification, the ATA shall send a re-INVITE request towards the NGA-T Service Provider network with: ▪ The Request URI set to the B party’s identity. ▪ An XML-MIME body containing an MCID XML request schema (as described in 3GPP 24.616 [19]) with McidRequestIndicator set to 1.” Paragraph 10: Replace the text “SIP UA” with “ATA”. Paragraph 11: Replace the text starting with “from the MGC Component,…” until “…in order to:” with “the actions of the ATA depend on which service is currently in operation.” Insert the following text as new paragraphs above the first bullet: “If the ATA is in the Call Waiting state the ATA shall act as described in A.2.1. If the ATA is in the 3-Way Call state the ATA shall:” Paragraph 12: Replace both occurrences of the text “AGCF/VGW” with “ATA”. Paragraph 13: Replace the text “a Network operator” with “an NGA-T Access Provider”. Replace the text “feature logic” with “ATA”.

ETSI TS 183 043 Clause	Title	Comment
		Replace the text "AS" in Note 2 with "NGA-T Service Provider." Paragraph 14: Delete the paragraph which begins "Processing of the feature…". Paragraph 15: Delete the paragraph which begins "With loose coupling…". Paragraph 16: Replace the text "Feature Manager" with "ATA". Delete the three occurrences of "/waiting" from the bullets. Delete the 4th bulleted item which contains the text "The served user wishes invoke the MCID service.". Paragraph 17: Replace the text "FM" with "ATA". Replace the text starting with "requests the MGC component…" until "…in order to play" with "plays". Paragraph 18: Delete the text "by the MGC component". Replace the text "AGCF/VGW" with "ATA". Paragraph 19: Delete the text "/waiting" from the first sentence. Replace the text starting with "unless an explicit…" until "…the Feature Manager" with "the ATA". Delete the first bulleted item starting with "Request the SIP UA". Delete the first sub-bulleted item starting with "a 200 OK response". Insert the text "Send" at the start of the second sub-bulleted item. Delete the text from the second sub-bulleted item starting with "if the dialogue…" until "…re-

ETSI TS 183 043 Clause	Title	Comment
		INVITE request is". Delete the text starting from "If a AGCF…" until "…If a VGW is used then:". Delete the text "/waiting" from the first sub-bulleted item of the 3rd bulleted item. Paragraph 20: Delete the text "/waiting" from the first sentence. Replace the text "Feature Manager" with "ATA". Delete the text "Request the SIP UA to" from the first bulleted item. Delete the text "a 603 response or" from the first bulleted item. Delete "waiting/" from the first bulleted item. Delete the text "Request the MGC component to interact with the media gateway in order to" from the second bulleted item. Paragraph 21: Delete the text starting with ", unless an explicit…" and ending with "…delivery procedure,". Replace the text "Feature Manager" with "ATA". Delete the text "Request the MGC component to interact with the media gateway in order to" from the first bulleted item. Delete the text "Request the SIP UA to" from the second bulleted item. Delete the text "/waiting" from the second bulleted item. Delete the text "/waiting" from the first sub-bulleted item. Delete the text "If an AGCF is in use" from the second sub-bulleted item.

ETSI TS 183 043 Clause	**Title**	**Comment**
		Delete the text from the second sub-bulleted item starting with “the contents of the local…” until “…address and port are set according to”. Paragraph 22: Delete the paragraph which begins “If the feature code received…”.
B.4.2.2.3	Tight Coupling Procedures	UK: Void

A.1.9 Void

A.2 Services

A.2.1 Call Waiting

A.2.1.1 General

The CW service shall be provisioned as described in sub-clause 10.5.

The UK Simplified CW service uses flash-hook activation (without the need to collect a feature code) in order to:

- Hold Active Call and Answer Waiting Call

- Toggle between Active and Held Calls

The Hold/Resume procedures used during CW service shall comply with RFC 3264 [9].

The CW service is not allowed when a CW enabled user either:

1. Is off-hook but not in an answered call state, or
2. Already has two call legs.

If either of the above is true then upon receipt of an INVITE the ATA shall send a 486 (Busy Here).

The following sections describe the actions of User B's ATA in the relevant Call Waiting states.

A.2.1.2 Actions whilst in a Call Waiting allowed state

A.2.1.2.1 Call Waiting presentation procedures

Upon receipt of an INVITE message to the busy subscriber, while there is no other waiting or held call the ATA shall:

1) Start an ATA-CW Timer.

   NOTE: The ATA-CW timer is used as a protection timer for the exception case where the CS has not acted upon it's own CW timer expiry. The ATA-CW timer shall be greater than the CS CW timer.

2) Provide a CW indication to User B by means of an in-band Call Waiting Tone (see section A.1.7.1).

3) Send FSK Caller-ID indication, if User B has the Caller ID service (see section A.2.2).

4) Send a 180 (Ringing) response to the INVITE according to the provisional response procedures described in sub-clause A.1.2.

A.2.1.3 Actions whilst in a Call Waiting alerting state

A.2.1.3.1 User B actions

The following actions can be performed by User B:

c) User B flashes to accept the waiting call and place User A in the held state.

In this case the ATA shall:

- Stop the ATA-CW timer
- Remove the CW indication
- Send a re-INVITE to User A placing the bearer on hold
- Send a 200 OK (invite) to User C Upon receipt of the 200 OK (invite) from User A.

d) User B hangs up to accept the waiting call and release the dialogue with User A.

In this case the ATA shall:

- Stop the ATA-CW timer
- Remove the CW indication
- Send a BYE to User A
- Apply Ringing towards User B
- The call now continues as a basic call including the normal ringing timer.

e) User B ignores the CW indication

In this case then either of the following will occur:

- The ATA will receive a CANCEL from the NGA-T Service Provider network; or
- the ATA-CW timer will expire.

If a CANCEL is received the ATA shall follow procedures described in section A.2.1.3.2.

If the ATA-CW timer expires the ATA shall remove the CW indication and send a 486 (busy here) to User C.

A.2.1.3.2 User C clears

Upon reception of a CANCEL request or BYE request from User C during a call waiting condition, User B’s ATA shall:

- Stop the ATA-CW timer
- Remove the CW indication
- Apply standard SIP procedure as described in RFC 3261 [2] related to CANCEL / BYE request.

A.2.1.3.3 User A clears

Upon reception of a BYE request from User A during a call waiting condition, User B's ATA shall wait for one of the following actions to be performed by User B:

a) User B flashes to accept the waiting call.
In this case the ATA shall:

- Stop the ATA-CW timer
- Remove the CW indication
- Send a 200 OK (invite) to User C.

b) User B hangs up to accept the waiting call.

In this case the ATA shall:

- Stop the ATA-CW timer
- Remove the CW indication
- Apply Ringing towards User B
- The call now continues as a basic call including the normal ringing timer.

c) User B ignores the CW indication

In this case then either of the following will occur:

- The ATA will receive a CANCEL from the NGA-T Service Provider network; or
- The ATA-CW timer will expire.

If a CANCEL is received the ATA shall:

- Remove the CW indication, and
- Play Number Unobtainable tone to User B

If the ATA-CW timer expires the ATA shall:

- Remove the CW indication
- Play Number Unobtainable tone to User B
- Send a 486 (busy here) to User C.

A.2.1.3.4 Handling of non-2xx responses to a Re-INVITE

Handling shall comply with RFC 3261.

Upon reception of a second 491 Request Pending the ATA shall stop sending the re-INVITE.

Upon receipt of a 408/481 or no response to the re-INVITE, the ATA shall:

- Play announcement ‘Supplementary service call hold failure’.
- Ignore Hook Flash during the announcement

If User B hangs up during the announcement while the waiting call is still present, the ATA shall:

- Present the waiting call by applying Ringing towards User B
- Continue as a basic call including the normal ringing timer.

If the announcement completes the ATA shall:

- Terminate any remaining dialogues.
- Continue with normal clearing procedures.

A.2.1.4 Actions whilst in a held call state.

A.2.1.4.1 User B actions

The following actions can be performed by User B:

a) User B flashes to toggle between the held and the active call.

In this case the ATA shall:

- Send a re-INVITE to active user requesting to place the bearer in the held state
- Set the bearer in the held state upon receipt of a 200 OK (invite)
- Send a re-INVITE to the held user requesting to place the bearer in the active state
- Set the bearer in the active state upon receipt of a 200 OK (invite)

b) User B hangs up

In this case the ATA shall:

- Send a BYE to the far end user of the current call leg
- Continue as described in A.2.1.5.

A.2.1.4.2 Held User A or C clears

If the held User clears then User B's ATA shall complete clearing of the dialogue as per basic call.

If User B attempts to toggle to the held call the ATA shall:

- Send a re-I**NVI**TE to the active user requesting to place the bearer in the held state
- Set the bearer in the held state upon receipt of a 200 OK (invite)
- Play announcement 'Other User Cleared' to User B

If, during the announcement phase, User B hangs up the ATA shall continue as described in A.2.1.5.

If either the announcement completes or User B flashes then the ATA shall:

- Send a re-INVITE to the held user requesting to place the bearer in the active state
- set the bearer in the active state upon receipt of a 200 OK (invite).

A.2.1.4.3 Active User A or C clears

If the active User (A or C) clears user B's ATA shall:

- Release the dialogue as per basic call
- Play announcement 'Other User Cleared' to User B.

If, during the announcement phase, User B hangs up the ATA shall continue as described in A.2.1.5.

If either the announcement completes or User B flashes then the ATA shall:

- Send a re-INVITE to held user requesting to place the bearer in the active state
- Set the bearer in the active state upon reception of a 200 OK (invite).

A.2.1.4.4 Handling of non-2xx responses or no response to a Re-INVITE

Upon receipt of a non-2xx response to any re-INVITE, the ATA shall:

- Send a BYE to any existing dialogue.

  Note: For any response other that 481/408 or no response this action is triggered by the ATA service layer and not the SIP protocol.

- Play announcement 'Supplementary service call hold failure'.

If the announcement completes the ATA shall continue with normal clearing procedures.

A.2.1.5 Re-Ring procedure, single Held Call exists

If User B hangs up and a held call exists, then the ATA shall:

- Apply Ringing current to User B
- Start Re-Ring Timer awaiting off-hook indication from User B
  Note: This timer shall be of an implementation specific value
- Await an off-hook indication from User B.

Upon receipt of an Off-Hook indication the ATA stall:

- Stop Re-Ring Timer
- Send a re-INVITE to the held user requesting to place the bearer in the active state upon receipt of an off hook condition from User B
- Set the bearer in the active state upon receipt of a 200 OK (invite)
- Continue as per basic call

If the Re-Ring Timer expires the ATA shall:

- Remove Ringing current from User B
- release the dialogue with the held User.

A.2.2 Caller Display

A.2.2.1 General

Information received in the URI and “display name” of the FROM header of an INVITE request is used by the ATA to derive the appropriate FSK parameters for display by terminal equipment.

The URI portion of the FROM header received from the Call Server may be one of the following:

- An international E.164 calling party number in a TEL URI or userportion of a SIP URI where the number comprises the country code and national part; or
- <sip:anonymous@anonymous.invalid> (used as an indication that Caller ID display has been restricted); or
- <sip:unavailable@unknown.invalid> (used as an indication that Caller ID is unavailable, e.g. due to network interworking); or
- <sip:notavailable@notavailable.invalid>; (used to indicate that the terminating subscriber has no Caller ID subscription).

The “display name” of the FROM header may be used by the NGA-T Service Provider to provide additional text to the called party. Possible values may include Operator, Payphone and

International but does not preclude the use of other values. The “display name” must only contain characters in token form and not quoted string as quoted string allows a character set which is greater than the IA5 code used in FSK delivery.

A.2.2.2 FSK parameter derivation

When an ATA receives an INVITE it shall generate FSK signalling prior to the first alerting indication unless the “userinfo” portion of the FROM header is set to “notavailable”.

The FSK parameters “Date and Time” and “Call Type” shall always be included.

The “Call Type” parameter shall be set to ‘Voice Call’.

The “Date and Time” parameter shall be set from the ATA clock which shall take into account the Daylight Saving Time rules.

FSK parameters “Calling Line Identity”, “Reason for Absence of Calling Line Identity” and “Calling Party Name” shall be populated as detailed in Table A.2.1 below:

Table A.2.1: FSK Call set-up message – parameter derivation

INVITE – From header		**FSK Call Set-UP Message – Parameters**		
“userinfo” portion	**“display name” present**	**Calling Line Identity**	**Reason for absence of Calling Line Identity**	**Calling Party Name**
E.164 number	No	Set according to contents of “From” header. (Note 1)	Not sent	Not sent
	Yes	Set according to contents of “From” header. (Note 1)	Not sent	Set to value contained in the “display name”.
anonymous	Don’t care	Not sent	Set to “50H” (Private)	Not sent
unavailable	No	Not sent	Set to “4FH” (Unavailable)	Not sent
	Yes	Not sent	Set to “4FH” (Unavailable)	Set to value contained in the “display name”.
Note 1: If the country code is “44” remove the “+44” from the beginning of the userinfo portion of the URI and replace with “0” before mapping to the parameter indicated above. Otherwise remove the “+” from the beginning of the userinfo portion of the URI and replace with “00” before mapping to the parameter indicated above. In either case the maximum number length in the Calling Line Identity parameter shall be 20 and strings longer than this shall be truncated.				

A.2.3 Message Waiting

Upon receipt of a NOTIFY containing a message-summary event package as per RFC 3842 [13] the ATA shall apply the appropriate dial tone pattern as described in clause A.1.1.

A.2.4 3-Way call

A.2.4.1 General

The 3-Way Call service is provisioned via ua-profile NOTIFY as specified in section A.1.1 or pre-configuration.

A.2.4.2 Enquiry Call - service invocation

In the following section, it is assumed that User A has the 3-Way Call service provisioned and is in an existing conversation with User B. User A is at the service invocation side.

A.2.4.2.1 Actions at the ATA at the service invocation side

Following the establishment of a confirmed dialogue between User A and User B and at any time during the active phase, the subscriber may attempt to establish a second call instance towards User C by pressing the flash-hook (recall) button.
Upon reception of the “flash-hook”, the ATA shall carry out the procedures defined in section A.1.8 Flash Hook management for “Call Configuration: Stable 2-party”.
Note: These procedures set User B on Hold, and apply dial tone towards User A.

If User A flashes during dial tone the ATA shall re-connect User A to User B.

If User A hangs up during dial tone the ATA shall re-connect User A to User B as described in A.2.4.5.

If either insufficient digits or no digits are received then announcements ‘Unrecognised Number followed by ‘Supplementary Service Re-connect’ will be played.

If User A flashes during the announcements the ATA shall re-connect User A to User B.

If User A hangs up either during dial tone or the announcements the ATA shall re-connect User A to User B as described in A.2.4.5.

If the duration of the announcement ‘Supplementary Service Re-connect’ expires the ATA shall proceed as follows:

- if the dialogue with User B still exists then re-connect User A to User B; or
- if the dialogue with User B has been released, apply normal clearing sequence towards the analogue line of User A.

On receipt of dialled digits, the ATA shall open a new dialogue towards User C by sending an INVITE request.

Reception of 2xx response to INVITE request towards user C:

If a 2xx message is received at ATA for the INVITE message sent out, a second confirmed dialogue shall be established and the procedures of section A.1.8 Flash Hook Management for “Call Configuration: Stable 2-party call with additional held/waiting party” will then be applicable.

For each successful second call establishment, the ATA of User A shall assign its call legs as follows:

- “First call” is the call to User B;
- “Second call” is the call to User C.

Reception of 4xx, 5xx or 6xx response to INVITE request towards user C:

If a 4xx, 5xx, or 6xx message is received at ATA as a final response to the attempted establishment of the second call instance, then the ATA shall play-out a Tone /Announcement per section A.1.7.
If the duration of Tone/Announcement expires the ATA shall proceed as follows:

- if the dialogue with User B still exists, play announcement 'Supplementary Service Re-connect' and then re-connect User A to User B; or

- if the dialogue with User B has been released, apply normal clearing sequence towards the analogue line of User A.

During play-out of Tone / Announcement the subscriber may return to the held call leg by means of:

- Flash-hook procedure; or

- By going On-Hook. On-hook will activate the Re-Ring procedure as described in A.2.4.5.

Subscriber Initiated Cancel of the Second Call Instance Establishment

The subscriber may cancel the call setup at any stage before a confirmed dialog has been established for the second call instance by either pressing the "Flash-Hook" (Recall)or by going On-Hook.

If User A flashes the ATA shall:

1. Stop any tone (e.g. Dial Tone) being applied to the subscriber line.

2. Stop collecting any digits and discard any digits already collected.

3. Send a CANCEL message if an INVITE has already been sent

If a dialogue with User B still exists, the ATA shall re-connect User A with User B. If the dialogue with User B has been released, the ATA shall apply normal clearing sequence towards the analogue line of User A.
If User A goes On-Hook the ATA shall proceed as follows:

- If the dialogue still exists, re-connect User A to User B as described in A.2.4.5.

- If the dialogue with User B has been released, apply normal clearing sequence towards the analogue line of User A.

A.2.4.3 User actions upon established Enquiry Call

In this scenario the service invoking "User A" has an active call with "User C" and a held call with "User B".
Upon reception of the "flash-hook", the ATA shall carry out the Flash Hook Management procedures of section A.1.8 for "Call Configuration: Stable 2-party call with additional held/waiting party".

Note: According to these procedures user C will be placed on Hold, user A will get dial tone and is expected to dial a feature code.

The ATA analyses the feature code according to a local mapping table. For each feature code, this table indicates the user expected feature:

- User A wishes to have separate calls and wishes to toggle between the two calls:
    - The ATA shall connect the service invoking user to the previously held User (User which was on Hold before last flash-hook invocation).

- User A wishes to end the call with User B ("first call") only:
    - The ATA shall send a BYE to User B.
    - The ATA shall re-connect the service invoking User A to User C.

- User A wishes to end the call with User C ("second call") only:
    - The ATA shall send a BYE to User C.
    - The ATA shall re-connect the service invoking User A to User B.

- User A wishes to establish a 3-party conference with both of the other parties:
    - See section A.2.4.4

Feature Code: No match or no feature code is dialled

If the feature code does not match any known feature or no feature code is dialled, the ATA shall play announcement 'Invalid Switching Order Received'.

If the announcement completes then the ATA has two held calls. The ATA shall proceed as follows:

- Send a re-INVITE to the held user of the last active party requesting to place the bearer in the active state.
  Note: the last active party is the active party before flash hook invocation.

- Set the bearer in the active state upon receipt of a 200 OK (invite).

- Continue as described in A.2.4.3.

If User A flashes during the announcement the ATA shall apply dial tone and await a feature code to be dialled.

If User A hangs up during announcement then the ATA has two held calls. The ATA shall proceed as follows:

- Apply Ringing current to User A

- Send a re-INVITE to the held user of the last active party, requesting to place the bearer in the active state upon receipt of an off hook condition from User A.

  Note: the last active party is the active party before flash hook invocation.

- Set the bearer in the active state upon receipt of a 200 OK (invite)

- Continue as described in A.2.4.3.

User A hangs up while having an active and a held call

If User A hangs up the ATA shall apply the Re-Ring procedure per section A.2.4.5.

A.2.4.3.1 Held User B or C clears

If the held Party clears then User A’s ATA shall complete clearing of the dialogue as per basic call.

Upon reception of a “flash-hook”, the ATA shall carry out the procedures defined in section A.1.8 Flash Hook management for “Call Configuration: Stable 2-party with additional held/waiting party”.
Note: These procedures set User B on Hold, and apply dial tone towards User A.

If User A dials the appropriate feature code to toggle to/or conference the other party (who has cleared down) the ATA shall play announcement ‘Other User Cleared’ to User A.

If the announcement completes or User A flashes during the announcement the ATA shall:

- Send a re-INVITE to held user requesting to place the bearer in the active state
- Set the bearer in the active state upon reception of a 200 OK (invite)
- Continue as per basic call

If User A hangs up during the announcement while a single held call exists the ATA shall apply the Re-Ring procedure per section A.2.4.5.

A.2.4.3.2 Active User B or C clears

If the active User (B or C) clears then User A’s ATA shall:

- Release the dialogue as per basic call
- Play announcement ‘Other User Cleared’ to user A.

If the announcement completes or User A flashes during the announcement the ATA shall:

- Send a re-INVITE to held user requesting to place the bearer in the active state
- Set the bearer in the active state upon reception of a 200 OK (invite)
- Continue as per basic call

If User A hangs up during announcement while a single held call exists the ATA shall apply the Re-Ring procedure per section A.2.4.5.

A.2.4.4 3-Way Conference Call

A.2.4.4.1 3-Way Conference Call – Internal Conference Bridge

Upon establishment of an Enquiry Call, the service invoking ATA may receive a feature code indicating that User A wishes to establish a 3-way conference.

The ATA shall implement a conference bridge following the signalling procedures described in Flash Hook management section A.1.8 for "Call Configuration: Stable 2-party call with additional held/waiting party". According to these procedures the ATA shall activate both call legs and set them into a 3-way conference context (i.e. apply a conference call algorithm and mix the media streams accordingly).

A.2.4.4.1.1 User actions upon established 3-way conference

User A goes on-Hook:
If User A goes on-hook during the 3-way conference call then the ATA shall send a BYE to both Users B and C.

User A activates Hook-Flash:
Upon reception of a flash-hook during an established 3-way conference call, the ATA shall:

- Send a re-INVITE to both User B and C requesting to place the bearers in the held state
- Set the bearers in the held state upon receipt of a 200 OK (invite)
- Apply a dial tone once both bearers are in a held state (the dial tone pattern shall applied according to section A.1.1.3)
- Await a feature code

User A activates flash-hook during dial tone:
If User A flashes during dial tone the ATA shall play announcement 'Invalid Switching Order Received'. Once the announcement completes the ATA shall place User A back in a 3-way conference with Users B and C.

User A goes On-Hook during dial tone:
If User A hangs up during dial tone or announcement 'Invalid Switching Order Received' the ATA shall:

- Send a BYE on both dialogues
- Continue with normal clearing procedures

Feature Code received:
The ATA analyses the received feature code according to a local mapping table. For each feature code, this table indicates the user expected feature:

- User A wishes to end the call to User B (end "first call") only:
    - The ATA shall send a BYE towards User B
    - The ATA shall re-connect the service invoking User A to User C.

- User A wishes to end the call to User C (end "second call") only:
  - The ATA shall send a BYE towards User C.
  - The ATA shall re-connect the service invoking User A to User B.

- User A wishes to return to a 3-way conference
  - Send re-INVITEs to the two held users to request activate both call legs
  - Set both of the other parties into a 3-way conference context (i.e. apply a conference call algorithm and mix the media streams accordingly) upon receipt of both 200 OKs.
  - Continue in clause A.2.4.4.

If the feature code does not match any known feature for this call state or no feature code is dialled, the ATA shall play out announcement 'Invalid Switching Order Received'.

If the announcement completes the ATA shall:

- Send re-INVITEs to the two held Users requesting to place the dialgies in the active state.

- Set both of the other parties into a 3-party conference context (i.e. apply a conference call algorithm and mix the media streams accordingly) upon receipt of both 200 OKs.

- Continue in clause A.2.4.4.

If User A flashes during the announcement the ATA shall apply dial tone and await a feature code to be dialled.

If User A hangs up during the announcement the ATA shall:

- Send a BYE on both dialogues

- Continue with normal clearing procedures

A.2.4.4.1.1.1 Held User B or C clears

Users B and C will have been placed in a held state when User A 'flashes' whilst in a 3-way conference.

If one of the held Users clear then User A's ATA shall complete clearing of the dialogue as per basic call. The following User A actions can then occur:

User A flashes whilst a Feature Code is awaited:
If User A flashes during dial tone the ATA shall play announcement 'Supplementary Service Re-connect' to User A.

When the announcement completes the ATA shall:

- Re-connect User A with the remaining active user.

- Continue as per basic call.

If User A flashes during the announcement the ATA shall take no action.

User A goes On-Hook:
If User A goes On-Hook during dial tone or the announcement the ATA shall:

- Apply the re-ring procedure to the remaining active dialogue as described in A.2.4.5

Feature Code dialled: End call toward User that cleared or establish 3 way conference with User that cleared
If User A dials the feature code to either end the call to a party that cleared or to activate a 3-way conference, the ATA shall play announcement 'Other User Cleared' to user A.
If the announcement completes or User A flashes during the announcement, the ATA shall:

- Send a re-INVITE requesting to place the remaining held bearer in the active state
- Set the bearer in the active state upon reception of a 200 OK (invite)
- Continue as per basic call

Feature Code dialled: End call toward remaining User (while other User has already cleared)
If User A dials the appropriate feature code to end the call to the remaining held party the ATA shall apply normal clearing sequence towards the analogue line of user A as per basic call.

Feature Code : no match or no feature code dialled
If the feature code does not match any known feature for this call state or no feature code is dialled, the ATA shall play out announcement 'Invalid Switching Order Received'. If the announcement completes or User A flashes during the announcement, the ATA shall:

- Send a re-INVITE requesting to place the remaining held bearer in the active state
- Set the bearer in the active state upon reception of a 200 OK (invite)
- Continue as per basic call

If User A hangs up during dail tone or the announcement while a single held call exists, then the ATA shall apply the Re-Ring procedure per section A.2.4.5.

A.2.4.4.1.2 User B or C actions upon established 3-way conference

If either User B or C clear during 3-way conference User A's ATA shall:

- Release the dialogue as per basic call
- Continue as per basic call with the remaining active dialogue

A.2.4.4.2 3-Way Conference Call – External Conference Server

For further study.

A.2.4.5 Re-Ring procedure, single Held Call exists

If User A hangs up and a single held call exists, then the ATA shall:

- Apply Ringing current to User A
- Start Re-Ring Timer awaiting off-hook indication from User A
  Note: This timer shall be of an implementation specific value
- Await an off-hook indication from User A.

Upon receipt of an Off-Hook indication the ATA stall:

- Stop Re-Ring Timer
- Send a re-INVITE to the held user requesting to place the bearer in the active state upon receipt of an off hook condition from User A
- Set the bearer in the active state upon receipt of a 200 OK (invite)
- Continue as per basic call

If the Re-Ring Timer expires the ATA shall:

- Remove Ringing current from User A
- release the dialogue with the held user.

A.2.4.6 Handling of non-2xx responses or no response to a Re-INVITE message

Scenario with a single established dialogue

The dialogue handling shall comply with RFC 3261.

Upon reception of a second 491 Request Pending the ATA shall stop sending the re-INVITE.

Scenario with more than one established dialogue

Upon reception of a non-2xx response where there is more than one established dialogue, the ATA shall:

- Send a BYE to any existing dialogue

  Note: For any response other that 481/408 or no response this action is triggered by the ATA service layer and not the SIP protocol.

- Play announcement 'Supplementary service call hold failure'.

If the announcement completes the ATA shall continue with normal clearing procedures.

A.2.5 Call Return

P-Early-Media as described in A.1.6.1 must be supported in order for this service to work.
There are a number of ways this service can be realised and the ATA shall support all of them.

A.2.5.1 Call to last caller set up using 302 (moved temporarily)

This option is covered by SIP procedures as per RFC 3261 [2].
When constructing the INVITE subsequent to the receipt of a 302 (moved temporarily) as per RFC 3261 [2], the ATA shall ensure that a new Call-Id value used.
Note: The creation of a new Call-Id avoids the call being incorrectly terminated due to loop detection at the NGA-T Service Provider as the transport is UDP.

A.2.5.2 Call to last caller set up using UPDATE

This option is covered by SIP procedures as per RFC 3261 [2] and RFC 3311 [5].

A.2.5.3 Call to last caller set up using forked responses

This option is covered by SIP procedures as per RFC 3261 [2].

A.2.6 Hotline

A.2.6.1 Procedures

Upon detection of an off-hook indication and the hotline service is provisioned the ATA shall continue as described in 10.2.2 of the present document.

A.2.7 Malicious Call Identity

A.2.7.1 General

The MCID service is provisioned via ua-profile delivery as specified in section A.1.1 or manually by the Access Provider's management system.

A.2.7.2 Terminating ATA procedures

If the called user 'flashes' during an active dialogue the ATA shall send a re-INVITE and an XML-MIME body with XML mcid body with MCID XML Request schema as per 3GPP 24. 616 [19] containing a McidRequestIndicator set to 1.

If the called user goes on-hook either before or after flashing the ATA shall send a BYE.

If the ATA receives a BYE from the NGA-T Service Provider network then normal call clearing applies.

A.2.8 Ring Back When Free

P-Early-Media as described in A.1.6.1 must be supported in order for this service to work.

History

Document history		
V.1.1.1		Publication
V1.2.1	Nov 2012	Publication
V1.3.1	Sept 2014	Published on the public web site