Vodafone
Telephony Interface Specification

Version: 1.08
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Terminal devices are only interface-compliant if they can be used by suitable technical measures in such a way that ensure that they are not altered in any way by software or hardware manipulation that they no longer meet the requirements of this interface description.

In particular, the installation of modified firmware must be effectively prevented.

With the release of a new version of this interface description, previous versions will lose their validity.

For the purpose of technical testing, Vodafone GmbH, Vodafone Deutschland GmbH and Vodafone West GmbH reserve the right to make deviating implementations at any time in geographically limited regions.

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Contents

Conventions

Throughout this document, the words that are used to define the significance of particular requirements are capitalized. These words are:

"MUST"	This word means that the item is an absolute requirement of this specification.
"MUST NOT"	This phrase means that the item is an absolute prohibition of this specification.
"SHOULD"	This word means that there may exist valid reasons in particular circumstances to ignore this item, but the full implications should be understood and the case carefully weighed before choosing a different course.
"SHOULD NOT"	This phrase means that there may exist valid reasons in particular circumstances when the listed behavior is acceptable or even useful, but the full implications should be understood and the case carefully weighed before implementing any behavior described with this label.
"MAY"	This word means that this item is truly optional. One vendor may choose to include the item because a particular marketplace requires it or because it enhances the product, for example; another vendor may omit the same item.

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1 Scope

This document contains the requirements for a telephony device or application to be used in the cable access network of Vodafone GmbH, Vodafone Deutschland GmbH and Vodafone West GmbH.

2 References

In the case of a conflict between specific requirements in this document with requirements in any of the directly or indirectly referenced documents, the specific requirements of this document are applicable.

2.1 Normative References

[G.168]	ITU-T, G.168 : Digital network echo cancellers
[G.711]	ITU-T, G.711 : Pulse code modulation (PCM) of voice frequencies
[G.722]	ITU-T, G.722 audio codec
[G.729]	ITU-T, G.729 audio codec
[G.726]	ITU-T, G.726 audio codec
[RFC1034]	Domain Names – Concepts and Facilities
[RFC1035]	Domain Names – Implementation and Specification
[RFC2131]	Dynamic Host Configuration Protocol
[RFC2617]	HTTP Authentication: Basic and Digest Access Authentication
[RFC2782]	A DNS RR for specifying the location of services (DNS SRV)
[RFC2915]	The Naming Authority Pointer (NAPTR) DNS Resource Record
[RFC3261]	SIP: Session Initiation Protocol
[RFC3262]	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
[RFC3263]	Session Initiation Protocol (SIP): Locating SIP Servers
[RFC3264]	An Offer/Answer Model with the Session Description Protocol (SDP)
[RFC3311]	The Session Initiation Protocol (SIP) UPDATE Method
[RFC3323]	A Privacy Mechanism for the Session Initiation Protocol (SIP)
[RFC3326]	The Reason Header Field for the Session Initiation Protocol (SIP)
[RFC3550]	RTP: A Transport Protocol for Real-Time Applications
[RFC3551]	RTP Profile for Audio and Video Conferences with Minimal Control
[RFC3611]	RTP Control Protocol Extended Reports (RTCP XR)
[RFC3960]	Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
[RFC4244]	An Extension to the Session Initiation Protocol (SIP)for Request History Information
[RFC4566]	SDP: Session Description Protocol
[RFC4733]	RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals
[RFC6337]	Session Initiation Protocol (SIP) Usage of the Offer/Answer Model
[UAK-S]	“Specification of the NGN-Interconnection Interface” version 4.0.3 or higher www.AKNN.de: UAK S

3 Informative References

[RFC3312]	Integration of Resource Management and Session Initiation Protocol (SIP)
[RFC3325]	Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
[RFC3329]	Security Mechanism Agreement for the Session Initiation Protocol (SIP)
[RFC4032]	Update to the Session Initiation Protocol (SIP) Preconditions Framework
[RFC5379]	Guidelines for Using the Privacy Mechanism for SIP
[T.38]	ITU-T, T.38 : Procedures for real-time Group 3 facsimile communication over IP networks

3.1 Reference Acquisition

- Arbeitskreis für technische und betriebliche Fragen der Nummerierung und der Netzzusammenschaltung AKNN, http://www.aknn.de
- Internet Engineering Task Force (IETF) RFCs, http://www.ietf.org/
- ITU Recommendations: http://www.itu.int

4 Definitions and Abbreviations

4.1 Definitions

This specification uses the following terms:

Public User Identity	A logical identity for purposes of communication with a User.
Server	A network element that receives requests in order to service them and sends back responses to those requests. Examples of servers are proxies, User Agent servers, redirect servers, and registrars as defined by [RFC3261].
User	A person who, in the context of this document, uses the telephony service.
User Agent (UA)	A software entity contained in a device that acts on behalf of the user to send requests to and receive responses from the network for a particular application. In the context of this document, a UA refers to a SIP User Agent as defined by [RFC3261].
User Equipment (UE)	The User device or application that is compliant to this specification, used by the User that wants to get telephony service.

4.2 Abbreviations

This specification uses the following abbreviations:

DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name System
DSx	Dynamic Service flow Add/Change/Delete
DTMF	Dual-tone multi-frequency
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
NA(P)T	Network Address (and Port) Translation
NAPTR	Name Authority Pointer
QoS	Quality of Service
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
TLS	Transport Layer Security
TTL	Time To Live
UA	User Agent

UDP	User Datagram Protocol
UE	User Equipment
URI	Uniform Resource Identifier
VSC	Vertical Service Code

5 IP Connectivity

The SIP UE MUST either use the same IP interface that is provisioned for data service, or either a separate IP interface.

The SIP UE must support IPv4 and IPv6. SIP must prefer IPv6 over IPv4. SIP UE must use only one IP protocol.

The SIP UE MUST NOT be used behind a NAT.

The SIP UE MUST obtain an IP address using standard DHCP [RFC2131].

The SIP UE MUST obtain an IP address using PPPoE for xDSL Access [RFC2516].

The SIP UE MUST NOT announce itself as a PacketCable device by containing string like “pktc1.0”, “pktc1.5” or “pktc2.0” in DHCP option 60.

The SIP UE MUST at least request the following options (Parameter Request List) to the DHCP server: 1 = Subnet Mask, 3 = Router, 6 = Domain Name Server.

The SIP UE MUST NOT be configured with a static IP address.

The SIP UE MUST conform to the requirements in DNS standards: [RFC1034], [RFC1035], [RFC2782], [RFC2915].

The SIP UE MUST follow standard industry best practice behavior with regards to usage of TTL.

The SIP UE should prefer IPv6.

6 SIP Profile

6.1 Configuration of SIP client

Vodafone Deutschland GmbH will provide a set of SIP credentials to the end user, one set for each assigned phone number.

The following entries MUST be configurable in the SIP UE (separately for each assigned phone number):

a. Phone Number (below referred to as “PHONE_NUMBER”)

b. SIP Domain (below referred to as “SIP_DOMAIN”)

c. Outbound SIP Proxy

d. (Authentication) Username

e. (Authentication) Password

The SIP UE MUST support a Username in the format of this regular expression:
“[a-zA-Z0-9_]{9,20}”

The SIP UE MUST support a password in the format of this regular expression:
“[a-zA-Z0-9!$/()=?*+#-_.:,]{20,32}”

6.2 Standard SIP Support

The SIP UE MUST be compliant to the base SIP specification [RFC3261], implementing the “User Agent (UA) role” in particular.

The SIP UE MUST support UDP as transport protocol.

The SIP UE should not exceed UDP packet size of 1400 bytes.

The SIP UE should support IP fragmentation.

The SIP UE MUST use the default SIP port 5060 to contact its outbound proxy.

The Public User Identity MUST take the form of a SIP URI as specified in [RFC3261].

The Public User Identity MUST be built as follows (based on the phone-number and SIP domain:
sip:PHONE_NUMBER@SIP_DOMAIN

Any other URI format (e.g., tel URI) MUST NOT be used by the SIP UE.

The SIP UE MUST never put "anonymous" in any outgoing message.

The SIP UE MUST be compliant to [RFC3262] (Provisional Acknowledgement – PRACK/support: 100rel).

The SIP UE MUST be compliant to [RFC3311] (aka. SIP-UPDATE).

The SIP UE MUST be compliant to [RFC3960] (Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)).

The SIP UE must be compliant to [RFC4244] for Request History Information (histinfo).

The SIP UE MUST be compliant to [RFC6337] (Session Initiation Protocol (SIP) Usage of the Offer/Answer Model).

The SIP UE MUST NOT subscribe to any event packages (via SUBSCRIBE/NOTIFY).

The SIP UE MUST NOT use the Privacy mechanism for SIP specified in [RFC3323].

The SIP UE MUST NOT use the Private Extensions to SIP specified in [RFC3325].

The SIP UE MAY be compliant to [RFC3326] (Reason Header Field).

6.3 SIP Registration and Redundancy

The SIP UE MUST be compliant to [RFC3263] (Locating SIP Servers).

The SIP UE MUST support DNS NAPTR [RFC2915], DNS SRV [RFC2782], DNS A [RFC1035] and DNS AAAA [RFC3596] record queries for locating the SIP server as defined in [RFC3263]. These are to be used to resolve the provided Outbound Proxy into an IP address.

If more than one SIP servers are resolved, the SIP UE MUST always try to register to the first priority SIP server coming from DNS SRV response. The second priority SIP server is only used in case of an outage of the first priority SIP server.

The SIP UE shall support up to four SIP servers.

The SIP UE MUST NOT register against two SIP servers in parallel.

The SIP UE MUST always register all of its phone numbers with the same SIP server to avoid a situation where registrations are sent to different SIP server addresses.

In case of a SIP UE failover and registration to the second priority SIP server, the SIP UE MUST try with the next re-registration attempt to register all phone numbers back to the first priority SIP server again.

If the SIP UE re-registration attempts towards first priority SIP server fail, the SIP UE MUST stay registered on the second priority SIP server but the fallback to the first priority SIP server MUST be retried again with the next re-registration attempt.

If SIP UE fallback registration attempt to the first priority SIP server is successful for one phone number all other phone numbers MUST fallback too.

Unless either the user or the application within the SIP UE has determined that a continued registration is not required, the SIP UE MUST re-register an already registered Public User Identity either 600 seconds before the expiration time if the previous registration was for greater than 1200 seconds, or when half of the time has expired if the previous registration was for 1200 seconds or less.

If the IP address of the SIP UE changes (e.g., upon DHCP Renew), the SIP UE MUST start a new SIP registration.

6.4 SIP Security

The SIP Digest method as specified in [RFC3261] MUST be supported.

Any other mechanisms or protocols to protect the SIP signaling MUST NOT be used.

On receiving a “401 (Unauthorized)” response to the REGISTER request, and where the "algorithm" Authorization header field parameter is "MD5", the SIP UE MUST extract the digest-challenge parameters as indicated in [RFC2617] from the WWW-Authenticate header field, calculate digest-response parameters as indicated in [RFC2617], send another REGISTER request containing an Authorization header field containing challenge response as indicated in [RFC2617]. The SIP UE MUST set the Call-ID of the REGISTER request which carries the authentication challenge response to the same value as the Call-ID of the 401 (Unauthorized) response which carried the challenge. The SIP UE MUST NOT include [RFC3329] header fields with this REGISTER.

Similarly, upon receiving a “407 (Proxy Authentication Required)” response to an initial request, the originating SIP UE MUST extract the digest-challenge parameters as indicated in [RFC2617] from the Proxy-Authenticate header field, calculate the response as described in [RFC2617], and send a new request containing a Proxy-Authorization header in which the header fields are populated as defined in [RFC2617] using the calculated response.

The SIP UE MUST NOT authenticate the SIP server.

The SIP UE SHOULD only accept SIP requests from the SIP server IP addresses that are resolved via DNS SRV / A records (SIP server whitelist).

If an incoming SIP request is received by the SIP UE from a SIP server which is not on the SIP server whitelist, the SIP UE SHOULD respond with "403 Forbidden".

6.5 SIP Sessions – Originating

A SIP INVITE message MUST only be sent out by SIP UE during an active registration and only to the SIP server on which the SIP UE is currently registered.

The SIP UE as Caller (side A) MUST apply local ring back tone towards the connected telephone set when it receives “180 (Ringing)” without SDP.

The SIP UE as Caller (side A) MUST NOT apply local ring back tone towards the connected telephone set when it receives “180 (Ringing)” with SDP.

The SIP UE as Caller (side A) MUST NOT apply local ring back tone towards the connected telephone set when it receives “183 (session progress)” with SDP.

An outgoing INVITE MUST use the following format for the “Request-URI” field:
sip:CALLED_PHONE_NUMBER@SIP_DOMAIN

An outgoing INVITE MUST use the following format for the “To:” field:
<sip:CALLED_PHONE_NUMBER@SIP_DOMAIN>

6.6 SIP Sessions – Terminating

When it receives an INVITE the SIP UE as Callee (side B) MUST respond with “180 (Ringing)” without SDP.

When it receives an INVITE the SIP UE as Callee (side B) MUST NOT send out any in-band ring back tone via RTP.

When it receives an INVITE the SIP UE as Callee (side B) MUST apply local ringing towards the connected telephone.

6.7 SDP Profile

The SIP UE MUST be compliant to [RFC3264] (Offer/Answer Model with SDP). This includes the correct handling of dialogues using more than one m-line as described for i.e. Real Time Text.

The SIP UE MUST be compliant to [RFC4566] (Session Description Protocol).

An INVITE request generated by a SIP UE MUST contain an SDP offer with at least one media description.

This SDP offer MUST reflect the calling user's terminal capabilities and user preferences for the session.

The SIP UE MUST NOT use the precondition mechanism specified in [RFC3312] and [RFC4032].

The SIP UE MUST NOT request/enable authentication/encryption for the media streams.

Upon sending an SDP answer to an SDP offer (which included one or more media lines which was offered with several codecs) the terminating SIP UE MUST select exactly one codec per media line and indicate only the selected codec for the related media stream.

In addition, the SIP UE MAY indicate support of the in-band DTMF codec.

The SIP UE MUST support configuration of dynamic RTP payload type number which is used for DTMF RTP Events as defined by [RFC4733].

If the SIP UE is configured to use DTMF RTP Events as defined by [RFC4733], then it MUST add the following in its outgoing SDP:

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

The SIP UE MUST include the "a=ptime" attribute for all "audio" media lines as described in [RFC4566], with value 20.

If a SIP UE receives an "audio" media line with "a=ptime" specified, the SIP UE MUST transmit at the specified packetization rate.

If the SIP UE supports RTCP Extended Reports per [RFC3611], then this MUST be indicated in its SDP offer per [RFC3611] with encoding “a=rtcp-xr:voip-metrics”.

The SIP UE MUST NOT try to set up any video sessions.

Other SDP parameters as “b=AS”, “b=TIAS”, “a=maxprate” SHOULD NOT be present.

For the forming of SDP the SIP UE MUST follow the examples as given in Annex B.

7 Call features / Supplementary Services

The SIP UE MUST NOT interpret Vertical Service Codes (VSC) locally.

Overview of VSCs:

Service	**VSC**
Calling Line Identification Restriction	*31#<target phone number>
Calling Line Identification Delivery per Call	#31#<target phone number>
Call Forwarding Busy – Activation	*67*<target phone number>#
Call Forwarding Busy – Deactivation	#67#
Call Forwarding No Answer – Activation	*61*<target phone number>#
Call Forwarding No Answer – Deactivation	#61#
Call Forwarding Unconditional – Activation	*21*<target phone number>#
Call Forwarding Unconditional - Deactivation	#21#
Call Waiting – Activation	*43#
Call Waiting – Deactivation	#43#
Call Forwarding on Subscriber Not Registered (CFNL) Activation	*62*<forward-to number>#
Call Forwarding on Subscriber Not Registered (CFNL) Deactivation	#62#
Anonymous Call Rejection (ACR) activation	*336#
Anonymous Call Rejection (ACR) deactivation	#336#

VSCs beginning with an asterisk (*) or hash (#) MUST be sent transparently and unchanged by the SIP UE to the SIP server.

If the SIP UE is to be used for Call Hold / 3PTY Conference, then the SIP UE MUST implement this per [UAK-S] chapter 2.1.5.5, CASE A.

8 Quality of Service

Quality of Service for voice calls (signaling and media) set up per this specification will be taken care of by the network.

The SIP UE MUST NOT in any way try to request quality of service. E.g, the SIP UE must not initiate any DSx signaling by itself.

9 CODEC-MEDIA

9.1 Codecs

The SIP UE MUST support [G.711] a-Law voice codec (PCMA).

The SIP UE MAY support [G.722] voice codec.

The SIP UE MAY support [G.726] voice codec.

The SIP UE MAY support [G.729] voice codec.

Silence suppression, comfort noise and echo cancellation MUST be implemented according to [G.168].

9.2 RTP and RTCP

The SIP UE MUST send and receive RTP and RTCP packets as defined in [RFC3550] and [RFC3551] for transport of audio flows.

The SIP UE MAY support RTCP Extended Reports per [RFC3611].

The SIP UE MUST use a packetization time of 20ms for RTP packets.

The SIP UE MUST NOT apply any authentication/encryption on RTP and RTCP.

9.3 DTMF Relay

The SIP UE MUST be compliant to [RFC4733] (RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals).

The SIP UE MUST send DTMF RTP Event packets strictly according to 20ms packetization time.

The SIP UE MUST stop the audio RTP stream temporarily for duration of DTMF RTP Event transmission.

10 Support of Real-Time Text

10.1 Real-Time Text References

[uaks-29] IETF RFC 4103 (Juni 2005): "RTP Payload for Text Conversation"

[uaks-30] IETF RFC 5194 (Juni 2008): "Framework for Real-Time Text over IP Using the Session Initiation Protocol (SIP)"

[uaks-31] IETF RFC 9071 (Juli 2021): "RTP-Mixer Formatting of Multiparty Real-Time Text"

[uaks-32] IETF RFC 2198 (September 1997): "RTP Payload for Redundant Audio Data"

[uaks-33] IETF RFC 7656 (November 2015): "A Taxonomy of Semantics and Mechanisms for Real-Time Transport Protocol (RTP) Sources"

[uaks-34] EN 301 549 V3.2.1 (2021-03): "Accessibility requirements for ICT products and services"

[uaks-35] EN 103 919 V0.0.8 (2024-05): "Emergency Communications (EMTEL); Accessibility and interoperability of emergency communications and for the answering of emergency communications by the public safety answering point (PSAPs) (including to the single European Emergency number 112)"

[uaks-36] ITU-T T.140 (02/98): "Protocol for multimedia application text conversation"

[uaks-37] ITU-T T.140.1 (02/00): "Protocol for multimedia application text conversation", Addendum I

[uaks-38] BFSG (22. Juli 2021): "Barrierefreiheitsstärkungsgesetz; Gesetz zur Umsetzung der Richtlinie (EU) 2019/882 des Europäischen Parlaments und des Rates über die Barrierefreiheitsanforderungen für Produkte und Dienstleistungen (BFSG)"

[uaks-39] BFSGV (22. Juni 2022): "Verordnung zum Barrierefreiheitsstärkungsgesetz; Verordnung über die Barrierefreiheitsanforderungen für Produkte und Dienstleistungen nach dem Barrierefreiheitsstärkungsgesetz – BFSGV"

[uaks-40] EU 2019/882 (17. April 2019): "Richtlinie über die Barrierefreiheitsanforderungen für Produkte und Dienstleistungen"

10.2 General

Based on the “Barrierefreiheitsstärkungsgesetz” telephony user equipment must support RTT.

Therefore, the user-network-interface (UNI) and the devices used shall support RTT according to the standards listed below:

- RFC 2198 [uaks-32],
- RFC 3264 [146],
- RFC 3840 [56],
- RFC 4103 [uaks-29],
- RFC 5194 [uaks-30],
- RFC 4566 [147],
- ITU-T T.140 [uaks-36],
- ITU-T T.140.1 [uaks-37] and
- EN 301 549 [uaks-34]

The UE and UNI equipment (i.e. CPEs) shall support beyond the support of SIP sessions with audio media:

- SDP negotiation within SIP signalling incl. media section for m=text and further a/b lines as defined for RTT
- through-connect RTP/RTCP payload types:
    - text media (t140/1000)
    - redundant media (red/1000)

      NOTE: It is mandatory to support RTP redundancy for text media with redundant RTP packets (coded red/1000). Nevertheless, request for redundancy in SDP Offers (with appropriate "a=fmtp" line) is optional and redundancy might be rejected in SDP Answers (by withdrawal of the relevant <fmt> value in m-line and the associated "a=fmtp <fmt>…" line).

      Appropriate bandwidth according to TS 26.114 [11] shall be provided.
- "text" feature tag in Contact and Accept-Contact header fields within INVITE message and response for INVITE message shall be forwarded transparently if present.

"text" media shall be supported only in conjunction with "audio" media. Hereby "text" media might be

- established together with initial establishment of the "audio" media session,
- added during an ongoing "audio" media session (i.e. add m-line "m=text…" in SDP) or
- removed during an ongoing "audio" & "text" session (i.e. set port=0 in "m=text" line in SDP).

10.3 Rejecting text media

Even though the network supports RTT, network and/or user equipment on either side of the NNI might not support RTT. Even in this case the SDP negotiation on the UNI shall support appropriate, generic, standardised capabilities to ensure support of "audio only" communication between the end points.

If a SIP client/proxy/B2BUA which does not support RTT receives an SDP offer with "m=text" it shall indicate non-support by setting the port in the media line to the value 0 (see RFC 3264 [146]) in the SDP answer. Setting the port in the "m=text" line to 0 might also be applied within the SDP offer (typically in a Re-INVITE) in order to remove a text media stream in an existing session.

Even in case RTT is disabled with port=0, the syntax for the media-field does not allow to skip all <fmt> parameter (at least one is mandatory).

For the UNI it is not required that an SDP answer or a (subsequent) SDP offer with "m=text" and port equal zero provides any a/b lines in the m=text section (i.e. also no "a=rtpmap" line).

NOTE: Nevertheless, it has been observed that some clients (which do not support RTT) expect within an SDP Offer with "m=text" section appropriate "a=rtpmap" line(s) which explain the <fmt> parameter(s) from the m-line, even in case port within m-line is equal to zero. Otherwise, such devices provide an SDP answer which results in call failure.

SDP example with:

m=text **20000** RTP/AVP 110 112

b=AS:3

b=RS:0

b=RR:0

a=rtpmap:110 t140/1000

a=rtpmap:112 red/1000

a=fmtp: 112 110/110/110

a=fmtp: 110 cps=30

a=sendrecv

Rejecting "text" media shall result in SDP:

m=text **0** RTP/AVP 110 112

optional: further a/b lines

Specific services (e.g. Call Hold, Audio Conference, Network Announcements) might result in temporary or permanent disabling of the "text" media stream. Hereby SDP offer/answer negotiations result in RTT disabled ("m=text 0 …") respective RTT resumed ("m=text <port>….") as described above.

10.4 Supplementary Services

10.4.1 Call Hold

In case of Call Hold, the same rules and procedures used for audio only sessions shall apply for text media, typically by sending a=sendonly, a=inactive or port=0 in the new SDP offer. The other party will put a=recvonly, a=inactive or port=0 in the SDP answer. With or after call resume the text media might also be resumed.

10.5 Exclusions

The following functionalities need not to be supported on the UNI. They may however be implemented in networks if this is opaque to the NNI signalling.

10.5.1 Teletypewriter (TTY)

Support for TTY and interworking between RTT and TTY and v.v. is out of scope for the NNI. This does not preclude TTY-RTT interworking in other networks if this is opaque to the NNI signalling.

10.5.2 Total Conversation

Total Conversation Service refers to a multimedia real time conversation service that provides bidirectional symmetric real time transfer of motion video, real-time text, and voice between users.

Support for video is out of scope.

10.5.3 Early Text Media

A text media stream for an early dialogue and in advance to a final call setup (200 OK [INVITE]) may be signalled on UNI ~~NNI~~ as per RFC 3264 [146], e.g. in a SIP 18x + SDP response. This is e.g. required to allow the calling party's network to trigger the setup of the dedicated bearer required for text media. This does not define whether any "early text" stream is rendered to the calling party.

NOTE: For the time being network announcements are not expected to play out any "early text" to a calling party.

10.5.4 RTT Conference

RTT conference refers to a text-mixer capability in parallel to an audio conference, i.e. forwarding text media from any participant towards all other participants. RFC 9071 [uaks-31] defines the technical approach.

RTT conferences are not supported on the NNI. This does not preclude RTT conferencing in other networks if this is opaque to the NNI signalling.

10.5.5 Text to Speech

"Text to Speech" or "Speech to Text" conversion within networks or devices is not required by legal obligations. This does not preclude such conversions if this is opaque to the NNI signalling.

Annex A
Example SIP messages

This Annex provides some example supported SIP messages:

A.1 REGISTER

REGISTER sip:SIP_DOMAIN SIP/2.0
Via: SIP/2.0/UDP UE_ADDRESS:UE_PORT;rport;branch=aaaabbbbcccc
From: <sip:PHONE_NUMBER@SIP_DOMAIN>;tag=123456789
To: <sip:PHONE_NUMBER@SIP_DOMAIN>
Call-ID: xxxxyyyyzzzz
CSeq: 123 REGISTER
Contact: <sip:PHONE_NUMBER@UE_ADDRESS>
Authorization: Digest username="AUTH_USER", realm="SIP_REALM", nonce="NONCE_VALUE", uri="sip:SIP_DOMAIN", response="AUTH_RESPONSE", algorithm=MD5
Max-Forwards: 70
Expires: 3600
Allow: ACK,BYE,CANCEL,INVITE,OPTIONS,PRACK,UPDATE
Content-Length: 0

A.2 INVITE

INVITE sip:CALLED_PHONE_NUMBER@SIP_DOMAIN SIP/2.0
Via: SIP/2.0/UDP UE_ADDRESS:UE_PORT;rport;branch=aaaabbbbcccc
From: <sip:PHONE_NUMBER@SIP_DOMAIN>;tag=123456789
To: <sip:CALLED_PHONE_NUMBER@SIP_DOMAIN>
Call-ID: xxxxyyyyzzzz
CSeq: 123 INVITE
Contact: <sip:PHONE_NUMBER@UE_ADDRESS>
Proxy-Authorization: Digest username="AUTH_USER", realm="SIP_REALM", nonce="NONCE_VALUE", uri="sip:CALLED_PHONE_NUMBER@SIP_DOMAIN", response="AUTH_RESPONSE", algorithm=MD5
Max-Forwards: 70
Expires: 120
Supported: 100rel
Allow: ACK,BYE,CANCEL,INVITE,OPTIONS,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 321

v=0
o=<username> <sess-id> <sess-version> IN IP4 <unicast-address>
s=-
c=IN IP4 <connection-address>
t=0 0
m=audio <port> RTP/AVP 9 8 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=rtcp-xr:voip-metrics

Annex B
Example SDP encodings

This Annex shows allowed options for SDP contents:

a. SDP Option 1

SIP UE Support	Answer
G.722 (ptime 20ms)	No
PCMA (ptime 20ms)	Yes
DTMF RTP Events (RFC4733)	No
RTCP-XR Reports (RFC3611)	No

v=0
o=<username> <sess-id> <sess-version> IN IP4 <unicast-address>
s=-
c=IN IP4 <connection-address>
t=0 0
m=audio <port> RTP/AVP 8
a=ptime:20

b. SDP Option 2

SIP UE Support	Answer
G.722 (ptime 20ms)	Yes
PCMA (ptime 20ms)	Yes
DTMF RTP Events (RFC4733)	No
RTCP-XR Reports (RFC3611)	No

v=0
o=<username> <sess-id> <sess-version> IN IP4 <unicast-address>
s=-
c=IN IP4 <connection-address>
t=0 0
m=audio <port> RTP/AVP 9 8
a=ptime:20

c. SDP Option 3

SIP UE Support	Answer
G.722 (ptime 20ms)	Yes
PCMA (ptime 20ms)	Yes
DTMF RTP Events (RFC4733)	Yes
RTCP-XR Reports (RFC3611)	No

v=0
o=<username> <sess-id> <sess-version> IN IP4 <unicast-address>
s=-
c=IN IP4 <connection-address>
t=0 0
m=audio <port> RTP/AVP 9 8 101

a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20

d. SDP Option 4

SIP UE Support	Answer
G.722 (ptime 20ms)	Yes
PCMA (ptime 20ms)	Yes
DTMF RTP Events (RFC4733)	Yes
RTCP-XR Reports (RFC3611)	Yes

v=0
o=<username> <sess-id> <sess-version> IN IP4 <unicast-address>
s=-
c=IN IP4 <connection-address>
t=0 0
m=audio <port> RTP/AVP 9 8 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=rtcp-xr:voip-metrics

e. SDP Option 5

SIP UE Support	Answer
G.722 (ptime 20ms)	No
PCMA (ptime 20ms)	Yes
DTMF RTP Events (RFC4733)	Yes
RTCP-XR Reports (RFC3611)	No

v=0
o=<username> <sess-id> <sess-version> IN IP4 <unicast-address>
s=-
c=IN IP4 <connection-address>
t=0 0
m=audio <port> RTP/AVP 8 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20

f. SDP Option 6

SIP UE Support	Answer
G.722 (ptime 20ms)	No
PCMA (ptime 20ms)	Yes
DTMF RTP Events (RFC4733)	Yes
RTCP-XR Reports (RFC3611)	Yes

v=0
o=<username> <sess-id> <sess-version> IN IP4 <unicast-address>
s=-
c=IN IP4 <connection-address>

t=0 0
m=audio <port> RTP/AVP 8 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=rtcp-xr:voip-metrics

g. SDP Option 7

SIP UE Support	Answer
G.722 (ptime 20ms)	No
PCMA (ptime 20ms)	Yes
DTMF RTP Events (RFC4733)	No
RTCP-XR Reports (RFC3611)	Yes

v=0
o=<username> <sess-id> <sess-version> IN IP4 <unicast-address>
s=-
c=IN IP4 <connection-address>
t=0 0
m=audio <port> RTP/AVP 8
a=ptime:20
a=rtcp-xr:voip-metrics

History

Document history		
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