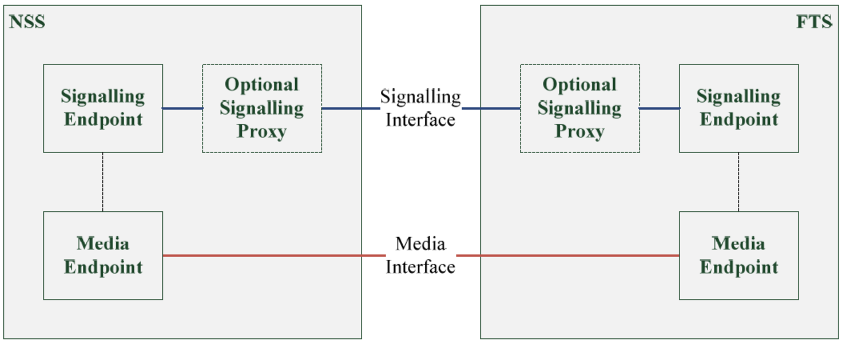
MEMENTO GSM-R – ETSI TS 103 389
SIP Profile for NSS / FTS interface

Reference GSM-R System Architecture at the NSS /FTS interface:



Provided services : <u>Teleservices:</u> Telephony, Emergency calls, Voice Group call services (VGCS), Voice Broadcast Service (VBS) - <u>Supplementary Service:</u> Calling Line Identification Presentation (CLIP) , Connected Line Identification Presentation (CoLP), - Call waiting (CW) [O] - Call hold (HOLD) - Multi Party Service (MPTY)/Conference [O], Enhanced Multi-Level Precedence and Preemption (eMLPP), Explicit Call Transfer (ECT) [O], User-to-User Signalling 1 (UUS1)

SIP profile:

- IPv4 (recommended Qos based on DiffServ (DSCP) [O], no NAT), only UDP
- RFC 3261 (no fork - UA should acts as B2B UA), only “early” SDP offer)
- Methods: INVITE, ACK, CANCEL, BYE, OPTIONS (request: [O], response: [M]) , PRACK (RFC 3262), UPDATE (RFC 3311; request: [O], response: [M]), INFO (RFC 6086)
- Specific headers:
  - “P-Asserted-Identity” (RFC 3325 & 5876) - shall be used for indicating a Call Transfer
  - “Privacy” (RFC 3323 - only with "none" value and when P-Asserted-Identity is present),
  - “User-to-User” (RFC 7433 & ETSI TS 103 389) shall be used to transport User-to-User information element: only present in INVITE, end-to-end responses to INVITE and BYE
    E.g.: User-to-User: 0005067370050005F1; encoding=hex; content=gsmr-uui.
  - “Resource-Priority” (RFC 4412 with only Q.735 namesspace; default value: “q735.4”)
  - “Reason” (RFC3326)
  - “P-Early-Media” (RFC 5009) [O]
  - “Alert-Info: <urn:alert:service:call-waiting>” in 180 Ringing response when “call waiting”.
- Only SIP URI with user = Eirene-user / e164user & user-param = "user=("gsmr" / "phone")"
- Option tags: 100rel (RFC 3262), privacy (RFC 3323), resource-priority (RFC 4412), timer (RFC4028)
- Feature Parameter “isFocus” [O] (RFC 3840) – used when a conference service is available.
- No SIP authentication
- If Proxy present => always “record-route”
- Early Media (183 with SDP): shall be supported by the caller subsystem.

SDP profile:

- RFC 4566, RFC 3264

Media Profile:

- IPv4 (recommended Qos based on DiffServ (DSCP) [O], no NAT), UDP, RTP
- Media inactivity detection using a RTP timer; if timer expires ⇨ SIP dialog is released (BYE)
- Media: audio; codecs: PCMU (0), PCMA (8) with 20 ms packetization, telephone-event (101) (RFC 4733 - feature “Telephony Tones and Multiple Events into One Packet” not supported)

[O]: optional - [M]: Mandatory (implicit when not mentioned)

Services to SIP Interface Mapping

Basic Call/Emergency Call

INVITE, CANCEL, BYE and responses generated in accordance with RFC 3261 (SIP), RFC 3264 (SDP) with:

- Only "early" SDP offer shall be allowed (INVITE with SDP offer).
- “Resource-Priority” header (RFC 4412) to indicate the operational and resource priority of the call => Require: resource-priority in INVITE request.
- “Reason” header in final INVITE response (E.g. upon Precedence Call Blocking) or in BYE request (E.g.: upon Pre-emption) with "SIP" or "Q.850" cause parameter value.
- "User-to-User" header in INVITE, end-to-end responses to INVITE and BYE requests - shall be used to transport the User-to-User information (User-to-User Signalling 1).
- Provisional Response Acknowledgement (PRACK - RFC 3262) ⇨ “Require: 100rel” in an INVITE request.
- No SIP authentication/challenging.
- Only two SIP UAs shall be involved in a SIP session's signalling flow => no Fork at the interface.

Connected party identity updates during call establishment or established call, a final INVITE response or a new Re-INVITE/UPDATE is sent with:

- “P-Asserted-Identity header”,
- “Privacy: none” (RFC 3323).

Media Session Renegotiation and Call Hold during established call:

- if the NSS or FTS SIP Endpoint wishes to put a media session on hold in an established dialog, a new SDP offer shall be sent with a re-INVITE request that contains:
  - the "inactive" SDP attribute if the remote side should generate a hold tone; or
  - the "sendonly" SDP attribute if an "On Hold Tone" or "Music On Hold" will be provided to the remote party.
- If the media session shall be resumed, a new SDP offer containing the "sendrecv" SDP attribute shall be sent in a new re-INVITE request.

Early media:

- INVITE may contain “P-Early-Media: supported” (RFC 5009).
- Upon receipt of a 18x with SDP answer or with a “P-Early-Media header”, the originating subsystem shall:
  - suppress local tone generation and shall instead present the media packets received to the user,
  - sends a PRACK request according with RFC 3262.

Note: In the absence of early media, the originating subsystem, FTS or NSS, shall generate and provide in-band progress indication tones to the user.

Session timer (a keep alive mechanism) shall be performed as described in RFC 4028:

- ⇨ Initial INVITE request with “supported: timer”, “Session-Expires: 600; refresher=uac” and “Min-SE: 600”,
- ⇨ Periodically re-INVITE or UPDATE to refresh session.

Group Call and Broadcast Call Control

For controlling VGCS/VBS calls:

- DTMF as specified in IETF RFC 4733; or
- INFO method with header “Info-Package” set to “etsi.groupcall.control” and dedicated body

Conferencing [O] (RFC 4579):

- Conference server/focus shall use “isfocus” feature tag (RFC 3840) in the Contact header as long as the conference established condition is true.
- “Replaces” header (RFC 3891) shall be used to replace an existing dialog at the NSS signalling endpoint. This shall be done, when turning a point-to-point call into a conference call

Call Forwarding

- If call has been forwarding, “History-Info” header (RFC 4244) with a cause-param URI parameter (RFC 4458) shall be used in initial INVITE or in “181 Call Is Being Forwarded” response

Call Waiting

- If activated ⇨ “Alert-Info” header (RFC 3261 [3]) set to “urn:alert:service:call-waiting" in the “180 Ringing”.