

28-30 November, 2000

Sophia-Antapolis, France

To: S2
Cc:
Source: S3
Contact: Krister Boman (krister.boman@emw.ericsson.se)
Title: **Clarification on the role of the P-CSCF**
Ref. tdocs: S3z000037

TSG S3 is studying how to provide access security for IP-based services and asks for advice on the role for the CSCFs.

TSG S3 is now looking on different alternatives to provide integrity protection for the SIP-messages. Specifically there are two alternatives for terminating the integrity protection: 1) S-CSCF and 2) P-CSCF.

In the second alternative it is assumed for access independence that the P-CSCF will be the entry point in the HN for a roaming subscriber for other accesses than 3GPP.

Therefore TSG SA3 considers the following alternatives for future work, on which guidance/advice is kindly asked from TSG SA2:

- What modifications will the P-CSCF perform on the SIP messages
- If the P-CSCF should be the entry point from other accesses than 3GPP

TSG SA3 would be grateful if TSG SA2 could provide an answer for TSG SA3 #17, scheduled February 27th-March 1st, 2001.

Att: S3z000027

Ad-hoc meeting S3#15bis

Munich, 8-9 November, 2000

3GPP CN1

455ge/09/TR/00/021

Source: BT**Subject:** SOME NOTES ON 3GPP TSG CN1 SIP #1 MEETING 17TH – 19TH OCTOBER 2000, SOPHIA ANTIPOLIS, FRANCE**Purpose:** Discussion**Agenda item:** 4.2

1 Ungraceful session termination in the IM domain

During a SIP session in a mobile network, the Mobile may go out of radio coverage or the Mobile may be switched off. The Call Control session in the S-CSCF needs to be terminated gracefully (and any billing needs to be stopped).

Solutions to solve this issue have been invited. Any S3 views?

2 Proposal to maintain CSCF in Call Path

In the case of a Mobile originated request, the next hop is based on the service control (S-CSCF) selected during the registration rather the called party (Request URI). There is no defined method in SIP registration to actually do this.

SIP protocol needs to be enhanced to follow the TSG S2 architecture requirement that P-CSCF shall always be in the loop. This requires the P-CSCF need to remember the address of the next hop (the S-CSCF). The S-CSCF has to pass this information to the P-CSCF at SIP registration time (200 OK response). However, STANDARD SIP can not do this. Two solutions have been identified, both requiring changes to SIP.

- a) To use Record-Route in the REGISTER method. Currently the SIP specification does not allow the use of Record-Route mechanism for the REGISTER message. If the Record-Route header is used in the REGISTER message, the S-CSCF can insert itself into the path. The P-CSCF can therefore read and remember this information (a stateful P-CSCF created by definition!).
- b) NEW 3GPP Path Header. This solution allows the addition of a new header, called *Path*, as part of a SIP extension. *Path* header can be used to specify the next hop for the P-CSCF. The P-CSCF can remember the S-CSCF's address, which is returned to it in the 200 OK message using the *Path* header. The P-CSCF can therefore read and remember this information (again a stateful P-CSCF created by definition!)

No decision made at this meeting. Any preference should be indicated before the next meeting (November 2000). Any S3 input?

3 Termination of SIP REGISTER or Transparent Relay through P-CSCF

Ad-hoc meeting S3#15bis

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3GPP CN1

455ge/09/TR/00/021

TSG S2 introduces P-CSCF as a mandatory node through which the SIP REGISTER message shall go through before ending up at the S-CSCF. Does NOT work with standard SIP. SIP has to be modified. Two ways of doing this.

a) The P-CSCF relays almost transparently the REGISTER message from the UE to the next node, with the CONTACT field completely changed. This requires that CONTACT field shall not be encrypted. ***Drawback: Integrity checking will no longer be possible between the UE and the S-CSCF. This security issue is a matter for TSG S3 to consider if relevant.***

b) The REGISTER message from the UE is terminated in the P-CSCF. A NEW REGISTER message is created in the P-CSCF and sent onto the next node. This approach has the drawback on feature transparency e.g. non-standardised information sent by the UE will be lost, unless the P-CSCF “copies” this in the new REGISTER message.

A decision on this will be made at the next N1 meeting in November. Any S3 input?

4 Routing of SIP requests in call flows

The SIP RESPONSE message back the destination network have to follow the SAME path back as for the initial REQUEST message. To enable this, the SIP protocol has the Route Record Header in SIP messages. The Record-Route headers list the entities between originating entity UE(A) and destination network S-CSCF. In standard SIP, The Destination terminal (UE) would store this information. For mobile networks, this is not sensible and the mobile can be out of radio coverage when a message needs to be sent back to the originating network.

- The meeting agreed that the UE should not store the call path for subsequent call modification or clearing but this will need to be the task of either P-CSCF or S-CSCF.
- Open issue. Should the S-CSCF store the information or the P-CSCF

The next N1/S2 joint meeting will make a decision on this issue. Any S3 views?