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Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Feasibility study on new methods for Overlap Sending

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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN).

Introduction

The SIP protocol has been designed for terminal types that send the address information en-bloc rather than a digit at a time. Whilst this is not an issue for terminals such as PCs and mobile phones, this is an issue for stimulus mode terminals such as telephones.

If the gateway controller that is controlling the access gateway for telephones, cannot determine the number length from the initial dialled digits, the procedures for an O-IWU described in ITU-T Recommendation Q.1912.5 [i.4] and its ETSI endorsement are either to apply a timer to collect digits or to send an INVITE message with the digits collected so far to avoid call setup delays due to this timer. If the INVITE contains incomplete digits, a SIP proxy in the session establishment path can return a SIP 404 "not found" or SIP 484, 'Number Incomplete Message' error response. For instance, the I-CSCF [i.4] acting as entry point to the terminating IMS will return a 404 response. An I-IWU as described in ITU-T Recommendation Q.1912.5 [i.4] will return a SIP 484 error response.

On receipt of the next digit the Gateway Controller may send a new INVITE with all the digits collected. If this is not enough then it will be rejected again along with sending the 484 or 404 message. This can be repeated digit by digit, with the session attempts progressing deeper and deeper into the network with the associated waste of signalling bandwidth and processing.

The present document proposes a number of mechanisms that will help minimize these wasteful overheads without impacting on the original mechanism and SIP. It also describes alternative mechanisms, one using SIP in-dialog messages, to transport the additional digits, once an early dialog has been established with the remote SIP entity.

In addition, impacts to overlap routeing in SIP networks are also investigated.

1 Scope

The present document purpose is to investigate new methods for providing overlap sending originating from PSTN Networks and devices. The expectation is that such methods would save on processing that the present method for overlap sending deploys. It is also the aim of this investigation to be fully backward compatible, have no impact upon the SIP signalling protocols, and have a minimal impact upon the existing SIP nodes.

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific.

- For a specific reference, subsequent revisions do not apply.
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  - for informative references.

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2.1 Normative references

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

Not applicable.

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area**.** For non-specific references, the latest version of the referenced document (including any amendments) applies.

[i.1] ETSI ES 283 003: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 [3GPP TS 24.229 (Release 7), modified]".

[i.2] IETF RFC 3261: "SIP: Session Initiation Protocol".

[i.3] IETF RFC 3578: "Mapping of Integrated Services Digital Network (ISDN) User Part (ISUP) Overlap Signalling to the Session Initiation Protocol (SIP)".

[i.4] ITU-T Recommendation Q.1912.5: "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".

[i.5] ETSI TR 184 005: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Types of numbers used in an NGN environment".

[i.6] ETSI TS 129 228: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia (IM) Subsystem Cx and Dx Interfaces; Signalling flows and message contents (3GPP TS 29.228)".

[i.7] ETSI TS 129 229: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Cx and Dx interfaces based on Diameter protocol; Protocol details (3GPP TS 29.229)".

[i.8] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163)".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

deterministic routing: routeing method ensuring that subsequent INVITE requests for the same call are forwarded to the same next hop

Incoming Interworking Unit (I-IWU): As defined in ITU-T Recommendation Q.1912.5 [i.4].

Outgoing Interworking Unit (O-IWU): As defined in ITU-T Recommendation Q.1912.5 [i.4].

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACK	ACKnowledge
AGCF	Access Gateway Control Function
AGW	Access GateWay
AS	Application Server
B2BUA	Back-to-Back User Agent
BGCF	Border Gateway Control Function
BICC	Bearer Independent Call Control
CSCF	Call Server Control Function
DDI	Direct Dialling In
DNS	Directory Name Server
DTMF	Dual Tone Multi Frequency
ENUM	Electronic Number
IAM	Initial Address Message
IBCF	Interconnect Border Control Function
I-CSCF	Interrogating - CSCF
ID	IDentity
IETF	Internet Engineering Task Force
I-IWU	Incoming Interworking Unit
IMS	IP Multimedia Subsystem
INFO	Information message
ISUP	ISDN User Part
ITU-T	International Telecommunications Union - Telephony
MGC	Media Gateway Controller
MGCF	Media Gateway Control Function
MIME	Multipurpose Internet Mail Extensions
NICC	Network Interoperability Consultative Committee
O-IWU	Outgoing Interworking Unit
OS-IWF	Overlap Signalling Interworking Function
PBX	Private
PC	Personal Computer

P-CSCF	Proxy CSCF
PES	PSTN Emulation Subsystem
PSTN	Public Service Telephony Network
PUID	Public User Identity
S-CSCF	Serving CSCF
SDP	Service Description Protocol
SIP	Session Initiation Protocol
SLF	Server Local Function
S-CSCF	Service Call Server Control Function
ST	Signal Termination
TrGW	Trunking GateWay
UAC	User Agent Client
UE	User Equipment
URI	Uniform Resource Identifier
VGW	Voice Gateway
XML	eXtensible Markup Language

4 Requirements and Issues

4.1 Requirements

1) The use of the new overlap signalling mechanism(s) minimize the additional signalling and processing load.

2) Any new overlap signalling mechanism is to be fully backward compatible with the overlap Release 1 SIP mechanism as described in RFC 3578 [i.3].

3) The entities collecting digits within the IMS network should be able to distinguish between unknown and incomplete numbers.

4) Routeing Database requirements:

   - There needs to be a mechanism in the database to handle an incomplete string, and the means of signalling a unique response for a valid incomplete string.

   - Support of a minimal length for the N(S)N part of Tel-URIs and "Types of Numbers" as drafted in TR 184 005 [i.5], should only apply to numbers starting with country codes for countries that support overlap dialling.

5) For the mechanism(s) based on RFC 3578 [i.3] and Q.1912.5 [i.4] every node in a network supporting overlap signalling ensures that subsequent INVITE requests for the same call are forwarded to the same next hop as the previous INVITE.

6) Networks supporting overlap and interfacing to networks that do not support overlap signalling will have to interoperate with theses networks. Networks that do not support SIP overlap signalling will then be unaffected when interworking with networks using SIP overlap dialling.

7) Networks supporting only specific overlap scenarios only need to implement the overlap extensions needed for these scenarios.

8) The solution for overlap sending will not cause any unnecessary impacts on systems not directly concerned with the use of overlap or en-bloc sending.

9) In the PES network the service level provided to the user should not be dependent on using overlap or en-bloc sending.

4.2 Issues

The point when overlap sending required is assumed by the present document however this decision is made is outside the scope of the present document and needs to be documented elsewhere.

4.2.1 Routing related issues:

1) When the S-CSCF or routing functions query ENUM, it needs to identify a valid incomplete string:

- **Issue 1a:** Separate entries are required in the database for each valid incomplete string.

Conclusion:

- An incomplete digit string may still be a valid entry for the routeing database, assuming that a routeing decision based on this entry can be made:

  - **Issue 1b:** The means of signalling a unique response for a valid incomplete string is required.

Conclusion:

- If an incomplete string is provisioned as a valid database entry, then the routing decision can be made. In this case the session setup is progressed to the next hop and no SIP response is generated by the routing hop.

2) **Deterministic routing** is recommended based on text in ITU-T Recommendation Q.1912.5 [i.4] and RFC 3578 [i.3]. RFC 3578 [i.3], paragraph 3.1 "One vs. Several Transactions":

- **Issue 2a:** All intermediate routing entities such as Application Servers, I-MGCF, Terminating UE and terminating S-CSCF all need to route deterministically and therefore route the INVITE requests to the same next hop.

NOTE: This is primarily an issue when routing to an MGCF, because multiple MGCFs may support routing to the same final destination in the SIP to ISUP direction, there may be a need to associate caller id etc with the IP addresses, etc. and different MGCF may be reached for reach new .SIP invite setup.

Conclusion:

- Only the entities that further propagate received overlap address information need to support the call-id correlation. Terminating entities apply the normal terminating procedures:

  - **Issue 2b:** Application Servers, I-MGCF, Terminating UE and terminating S-CSCF need to identify that subsequent INVITE requests with the same call ID are related.

The BGCF needs to be able to identify a valid incomplete string for numbers not in ENUM (after ENUM query failure before the BGCF is reached).

4.2.2 Interrelation with number portability

- **Issue 3:** At the point at which number porting information is retrieved, to enable portability in the DDI enterprise cases, in the normal case is that you cannot port until the full number is received.

NOTE: It is not decided yet whether Number Portability is supported in TISPAN R2.

Conclusion:

- It could be necessary first to collect all digits transmitted in the overlap mode before a valid portability response can be received.

On the other side the database of a number portability server could also include incomplete numbers like it is in Germany where blockbuilding is done. Also PBX users cannot be forced to store their PBX dialling schema in an official database. So for routing purposes the answer given for issue 1a is valid.

4.2.3 Interrelation with the support of wildcard PUID

- **Issue 4:** The network needs to deliver a valid incomplete strings owned by IP PBX networks, that enable a IP PBX UE to setup to another part of the PBX network UE across the public NGN. E.g. S-CSCF and Application Servers will need to deliver valid incomplete strings to PBXs.

NOTE: It is not decided yet whether wildcard PUID is supported in TISPAN R2. However, deferring to TISPAN R3 is not a feasible option, as the support of wildcard PUID is needed to support PBX in TISPAN R2.

Conclusion:

- The routing based on wildcard PUIDs is the same as on an incomplete numberstring.

4.2.4 No IMS defined solutions

Historically the defined procedures for the overlap sending and receiving are focusing on the interworking elements only. The IMS core was not taken in to account. This leads to the following problem (see figure 4.2.4-1) if no additional procedures are defined for the support of the overlap by IMS. The subsequent INVITE may get routed to the different MGC due to the load sharing policies, or any other routing decisions made by the core IMS.

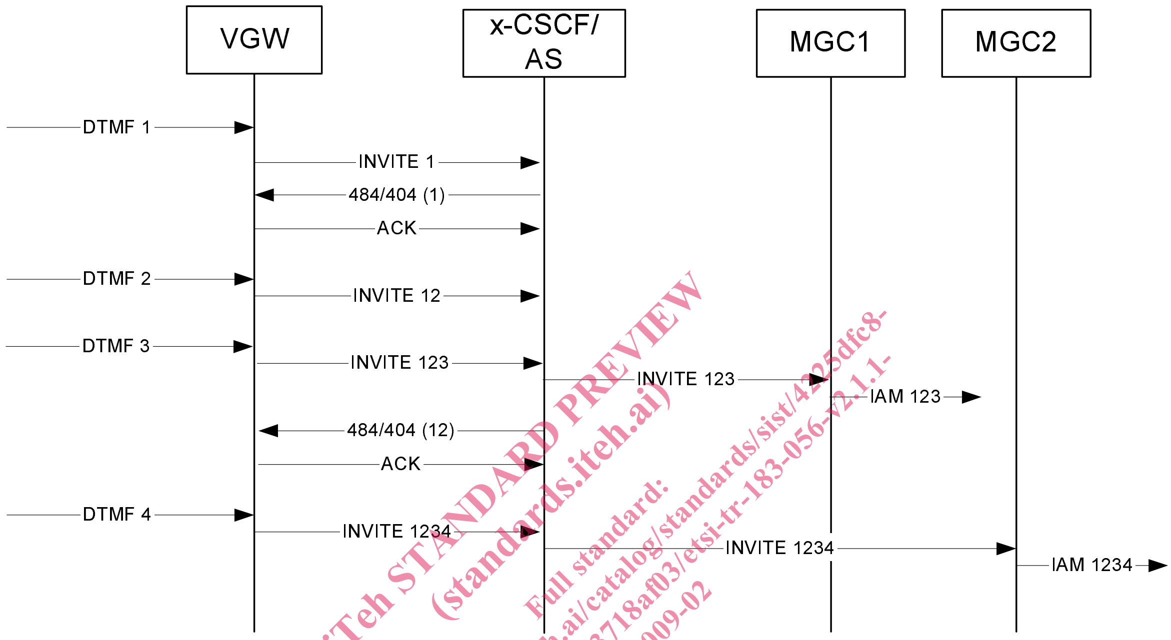

Figure 4.2.4-1

4.2.5 Overlap Scenarios supported

The impacts on IMS and IMS-PES network entities and functions depend on the supported overlap scenarios. This clause defines three cases of overlap support. For example, an IMS network may only support overlap for transit calls.

In all the scenarios reflected in the following clauses, the same basic principle should apply: the first network able to resolve the overlap signalling conversion, either totally or partially, is responsible for such conversion.

4.2.5.1 Overlap transit

In this scenario, the terminating subscriber is not known to the IMS network, and the call is not routed via the S-CSCF. The transit function(s) within the network route the call towards the next network.

The transit network can choose to perform full en-bloc conversion, or perform partial conversion and forward the call towards the next network once enough digits have been received in order to find the next network.

NOTE: If the next network does not support overlap, full en-bloc conversion is performed by the (transit) IMS network.

4.2.5.2 Terminating overlap

In this scenario, the terminating subscriber is registered to the terminating IMS/IMS PES network. The call enters the terminating network from a PSTN network (via a MGCF), or from another IMS/IMS PES network (via IBCF). The terminating subscriber uses the DDI service, but the additional digits for DDI are not relevant to identify the subscriber for the purpose of the service logic in the IMS network or to route the call towards the subscriber through the IMS based network.

The terminating IMS/IMS-PES network will perform full en-bloc conversion when the network knows the numbering length of the terminating subscriber.

For IMS PES, in the particular case of subscribers with a DDI service, there may be scenarios where the IMS PES network does not know the full length of the number. In such a case, the IMS PES will perform partial en-bloc conversion required to perform the service logic in accordance to the subscription, and forward the call once enough digits have been received in order to identify the terminating subscriber (AGCF).

NOTE 1: Currently there are no requirements to support overlap for IP-PBXs, so the current assumption is that DDI services are provided by IMS PES via an AGCF. If there is a requirement to support overlap for IP-PBXs, it is handled separate from cases handled via an AGCF.

NOTE 2: The AGCF/VGW is an IMS PES entity.

4.2.5.3 Originating overlap (IMS PES)

The originating IMS-PES network (the overlap comes via an AGCF) can choose to perform full en-bloc conversion, or perform partial conversion and forward the call towards the next network (which could be the same physical IMS network although in the context of overlap should be seen as a different logical network, and the transit or terminating overlap case applies) once enough digits have been received in order to identify the next network.

NOTE: If the next network does not support overlap the originating IMS-PES network performs full en-bloc conversion.

4.2.6 Different error responses for incomplete and unknown number

SIP proxies, which require a minimum number of digits to forward the call, such as a node performing a routeing decision or the I-CSCF at the B-side that looks up the S-CSCF assigned to the request URI in the SLF, return an error response when receiving an INVITE with insufficient digits. According to current procedures in TS 129 229 [i.7], the SLF does not keep apart unknown and uncomplete numbers and the I-CSCF will therefore reply with the generic 404 response. The procedures could be modified by identifying uncomplete numbers. The Cx interface would need to allow the SLF to return different results for uncomplete and unknown numbers so the I-CSCF can generate a 484 response as appropriate, in order for the MGCF to continue overlap timing or to reject the call immediately.

According to ITU-T Recommendation Q.1912.5 [i.4] the receipt of a 404 response would trigger an ISUP REL message to the PSTN/ISDN side of the gateway.

According to TS 129 163 [i.8] the receipt of a 404 response could, when overlap is used, be treated in the manner as the receipt of a 484 response i.e. a timer Ti/w3 is started and the gateway will await receipt of further digits.

Separate 404/484 processing would apply to all nodes collecting digits for overlap sending.

The advantage of different processing is that sessions can be rejected quicker e.g. a 404 response would trigger an immediate release.

5 Overview of technical solutions

5.1 General

The following clauses describe different mechanism alternatives, and IMS entity impacts, related to SIP overlap signalling:

- Mechanisms to indicate minimum number of digits needed by the network to forward the call, in order to reduce signalling load by not sending digits until the minimum number of digits are available.
- Mechanisms to transport additional digits in the IMS network.
- Mechanism to interwork between different overlap transport mechanisms and networks that do not support overlap.

5.2 Mechanisms for the Reduction of Signalling Load

5.2.1 Provisioning of Number Length Information within Extension of the Error-Info Field

5.2.1.1 Procedures

Upon receipt of an INVITE request with incomplete digits, some SIP proxy or B2BUA identifies that the number is uncomplete, looks up information about a minimum number length as derived from the digits received in the INVITE, and then rejects the INVITE request using a SIP 484 error response that encodes this information about a minimum number length.

The SIP proxy could be:

1) The caller's S-CSCF that will need to take a routing decision based upon request URI. While the routeing procedure is not fully standardized in TS 129 229 [i.7], this specification lists as a typical example that the S-CSCF applies ENUM to query an external database to transform a Tel URI into a SIP URI including a host portion for that purpose and then uses the host portion of the SIP URI for routeing.

2) An AS attached to the caller's S-CSCF, which could also use ENUM for the same purpose as described under bullet 1.

3) An IBCF at the interconnection between the caller's and the callee's network.

4) The callee's I-CSCF, which needs to interact with the SLF to identify the S-SCSF of the called user and requires the complete number for that purpose.

The min length information would allow the original gateway controller to optionally withhold sending further SIP INVITE until the minimum number length is accumulated, thus removing an ineffective SIP INVITE that may have been sent before that number length were reached.

This process may be repeated if the session establishment progressed to another SIP Proxy that recognized that even more digits were required.

Figures 5.2-1 and 5.2-2 show example callflows. These figures also assume some digit collection functionality as described in clause 5.2.3 for the first digits at the VGW.