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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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Contents

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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.
- Non-specific reference may be made only to a complete document or a part thereof and only in the following cases:
  - if it is accepted that it will be possible to use all future changes of the referenced document for the purposes of the referring document;
  - for informative references.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

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.

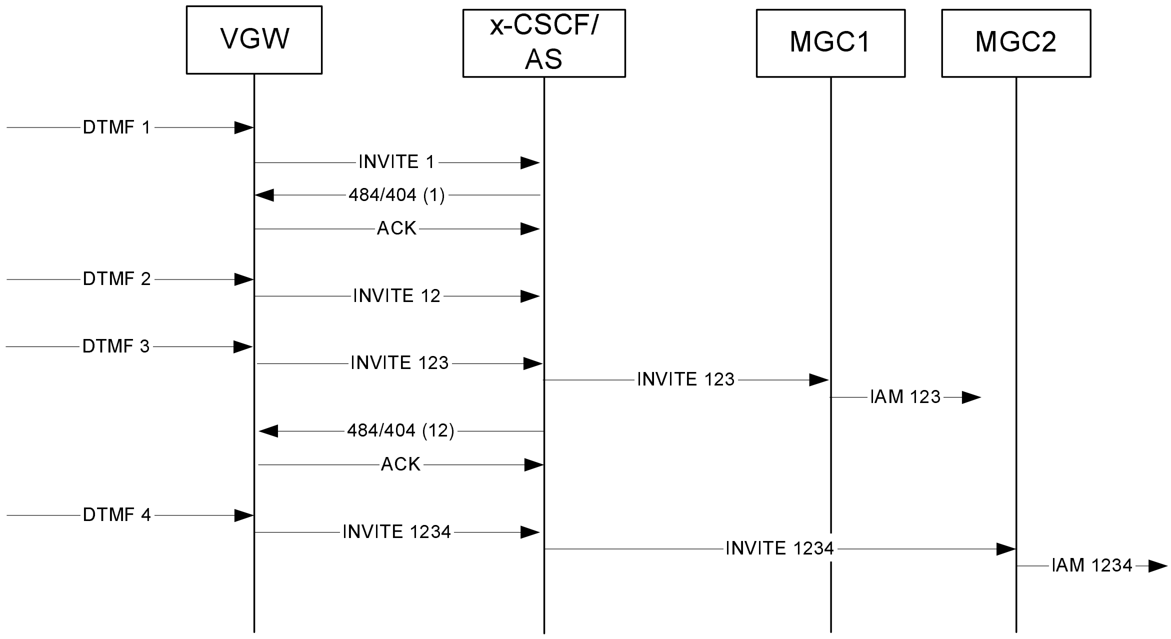

Figure 4.2.4-1

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.

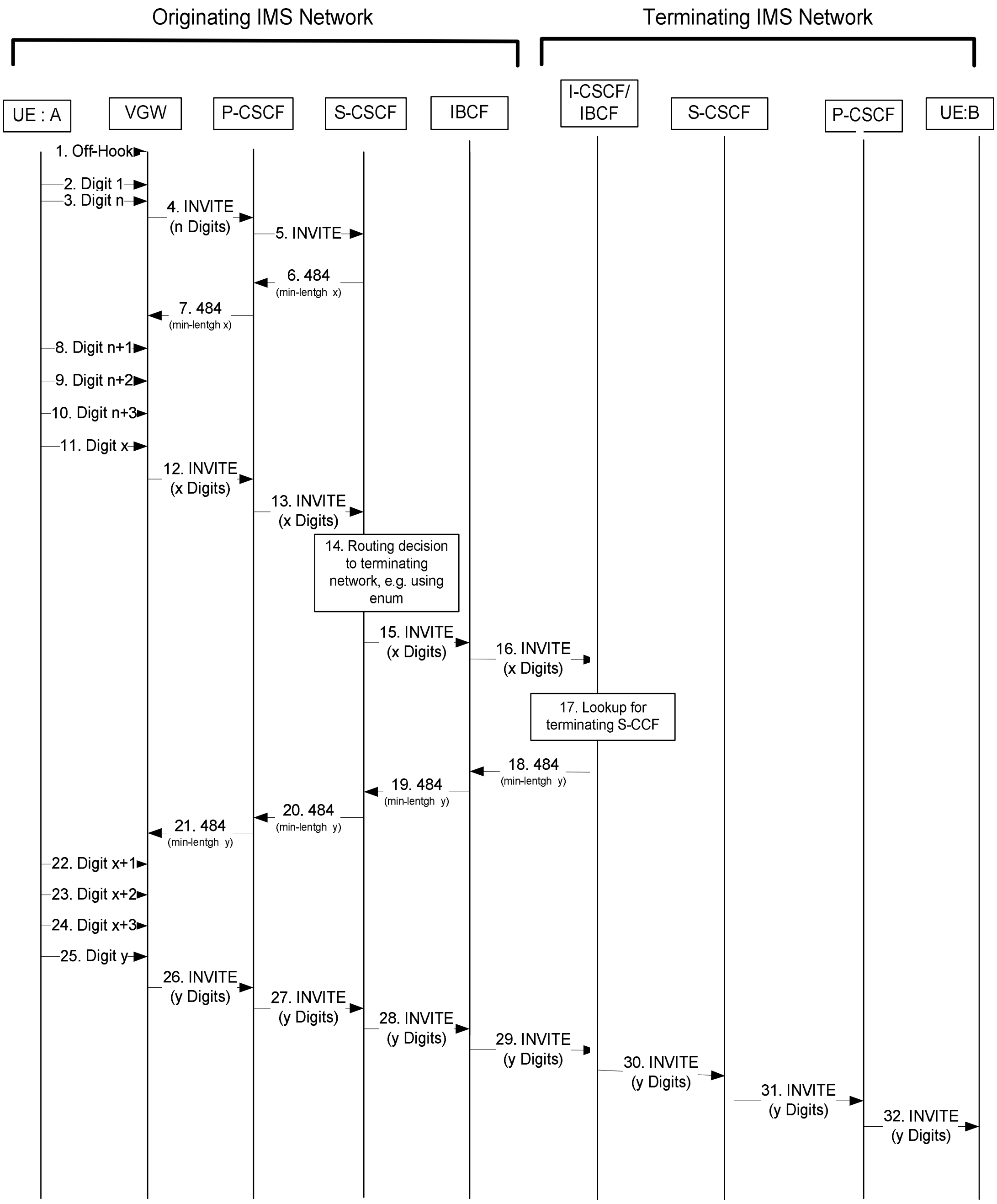

Figure 5.2.1.1-1: Example of overlap communication including min number length information provisioned by S-CSCF A and I-CSCF B

1-3: Analogue Phone is dialling the first digits.

4-5: A initial INVITE is sent towards the S-CSCF. The S-CSCF looks up the routeing database and obtains information about the min Length of Digits required to route the call.

6-7: A 484 including the minimum number of digits required is sent back.

8-11: The VGW is collecting the further needed digits to reach the minimum of needed numbers.

12-14: The following INVITE includes the Request URI including all digits collected so far. The S-CSCF has now the information to route the call.

15-16: The INVITE is forwarded to the terminating network.

17: The I-CSCF selects the S-CSCF assigned to the terminating user. If the numbers is incomplete, the I-CSCF sends a 484 to request further digits, encoding the required number of digits within.

18-21: A 484 with the information about the minimum number of digits required is sent back towards the originating VGW.

22-25: The VGW is collecting the further digits needed.

26-32: The INVITE containing all numbers requested is sent towards the destination. The terminating I-CSCF and S-CSCF can route upon the delivered numbers towards UE-B.

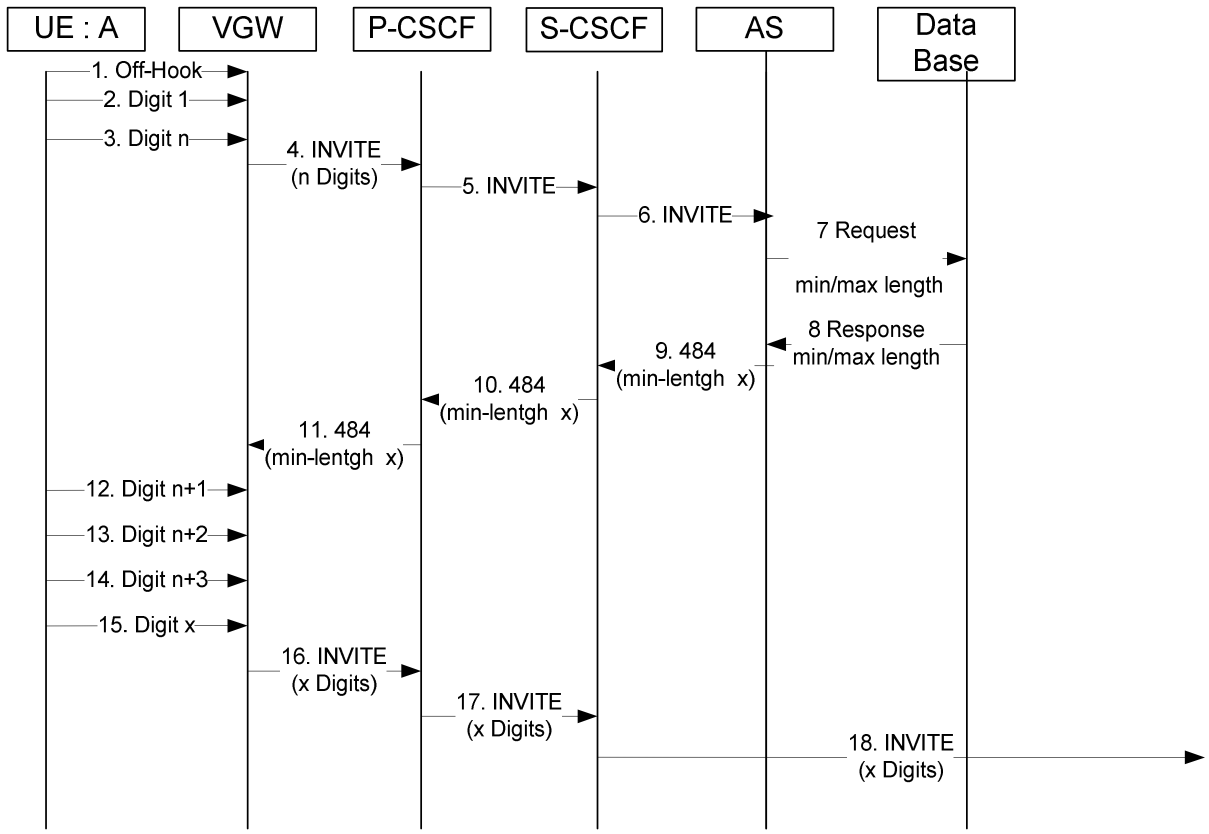

Figure 5.2.1.1-2: Example of overlap communication including min number length information provisioned by overlap AS

5.2.1.2 Encoding

One encoding proposal is to use the existing Error-Info SIP header field as defined in RFC 3261 [i.2], which can be optionally used by any SIP proxy that rejects a SIP INVITE with the response '484'. When sending a 484 response, a SIP proxy can optionally include this Error-Info SIP header field according to the profile endorsed by ETSI TISPAN in ES 283 003 [i.1].

The error-info field is intended to carry a URI which points to where further information can be found on the minimum numbers required at this point in the network. This URI can point to an announcement server or a HTTP URI to a web page.

The URI could encode the min length information as parameters to the URI. The format of the URI parameters needs to be standardized and be suitable for an automatic access by the AGCF or VGW.

As an example in the UK it is proposed to profile this field to point to an NICC web page with a parameter that indicates the minimum number length to be displayed. A PSTN call server, with an AGCF serving overlap sending lines would recognize this URI and directly read the parameter field without referencing to any announcement servicer or web page (see figure 5.2-2). However it is also proposed that a web pages exist that would produce an appropriate display in a web browser should an appropriate terminal (e.g. PC soft phone) with access to the public internet, made a call to the PSTN with insufficient digits.

A reject to an INVITE could be optionally constructed as:

```
SIP/2.0 484 Address Incomplete
Via: …...
Via:…….
To: ……
From:…………
Call-ID:………
CSeq:………
Content-Length: …….. 
Error-Info: http://www.etsi.tispan.org/SIPErrInfoExtns?MinNumLen=nn
```

Where nn is the total minimum number length as recognized by the rejecting SIP node.

E.g. where an INVITE with 6 digits was sent and it was recognized that another 5 were required to get a complete address then MinNumLen=11.

This proposal does not change any of the SIP protocol and is backwards compatible with implementations that do not recognize this profile. Therefore it does not need approval from the IETF and such a mechanism also has advantage in working with international calls.

5.2.2 Provisioning of Number Length Information within min/max Digit MIME body

5.2.2.1 Procedures

See clause 5.2.1.1.

5.2.2.2 Encoding

An encoding proposal is to standardize a MIME type containing the min length. When sending a 484 response, a SIP proxy can optionally include this MIME type as body of the SIP message.

Preferably, the MIME type contains XML contents. This XML body can be used to provide a minimum number length indication as recognized by the rejecting SIP proxy.

A reject to an INVITE could be optionally constructed as:-

```
SIP/2.0 484 Address Incomplete
Via: …...
Via:…….
To: ……
From:…………
Call-ID:………
CSeq:………
Content-Length: …….. 

Media type name: application
Media subtype name: OVERLAP

Min-length …xx
```

Where xx is the total minimum number length as recognized by the rejecting SIP/IMS node.

E.g. where an INVITE with 6 digits was sent and it was recognized that another 5 were required to get a complete address then Min-Length=11.

This proposal does not change any of the SIP protocol and is backwards compatible with implementations that do not recognize this profile. Therefore it does not need approval from the IETF and such a mechanism also has advantage in working with international calls.

5.2.3 Digit Collection at AGCF or VGW

Although an AGCF or VGW will not be able to determine the end of dialling without applying unacceptable long timers in all cases, the AGCF/VGW may apply similar procedures as already implemented in some current local ISUP exchanges at the caller's side to reduce the overlap signalling load:

- collect a fixed minimum number of digits before sending the first invite; or
- implement an incomplete numbering plan that contains information about the minimum number of digits for different destinations (e.g. keeping apart local, national and international calls, or discriminating different cities in the country);
- use short digit collecting timer (e.g. 1 s) to collect digits typed in as a sequence without any unusual interrupts. On expiry of this timer a new INVITE will be sent and digit collection will continue.

The applied fixed minimum number of digits or incomplete numbering plan will need to be harmonized with the routing database of the network (e.g. an ENUM database). For a call towards another IMS network, it needs to be guaranteed that the number of digits is sufficient for a routing decision, in order to prevent that a call is routed to some default destination such as the PSTN.

5.2.4 Provisioning of Number Length Information within the SIP Reason header

5.2.4.1 Procedures

See clause 5.2.1.1.

5.2.4.2 Encoding

An encoding proposal is to extend the Reason header by defining a way to transport the min digit value. This can be done by either defining a new "protocol" type for the header, or to define new reason parameters for an existing type.

5.3 Routeing Methods for Transmitting SIP networks

5.3.1 Overlap signalling using SIP in-dialog messages

This clause describes an alternative mechanism, which uses SIP in-dialog messages (e.g. SIP INFO) to transport additional digits once an early dialog has been established with the remote SIP peer. Until the early dialog has been established, multiple INVITEs may be sent and rejected with an indication that not enough numbers have been received.

Until this SIP peer is reached, subsequent INVITEs are also used to transport additional digits, but deterministic routeing of the subsequent INVITEs is not required. An in-dialog based mechanism has the following advantages over the mechanism based on RFC 3578 [i.3]:

1) Once the early dialog with the remote peer has been established the in-dialog messages will be routed to the remote peer using normal SIP mid-dialog routing mechanisms. No new IMS core routing procedures will be needed, and no database/DNS lookups are needed for the INFO messages.

2) Since the INVITE establishing the early dialog, and the in-dialog messages within that dialog, will have the same Call-Id value and From tag value it does not matter if some B2BUA entity in the network modifies those values. Since the messages belong to the same dialog, the B2BUA modification would be identical for each message. Using a mechanism with INVITEs, each INVITE is sent for a separate dialog, and a B2BUA may modify that Call-Id and/or From tag value differently in each message.

3) Databases (e.g. ENUM) do not need to support each incomplete string. The databases may support a single part of the string, which is enough to route the INVITE request to the correct terminating peer. No database lookups are needed for the additional digits sent in the in-dialog messages.

A disadvantage with the mechanism using SIP in-dialog messages is that it is not compatible with RFC 3578 [i.3] and ITU-T Recommendation Q.1912.5 [i.4] based solutions. It needs to be studied whether it can be assumed to exist networks where the RFC based solution is used, and whether the overlap signalling functionality needs to interwork with those networks.

The additional digits may be sent using mid-dialog SIP requests, or on the media plane.

Mid-dialog SIP requests:

- Using this mechanism, the additional digits are transported using mid-dialog requests. The SIP method that normally has been seen as the most obvious choice is INFO. However, the in-dialog method as such does not require INFO to be used, in case there are more appropriate SIP methods.

When using mid-dialog requests, the additional digits cannot be encoded into the Request-URI (as for the multiple-INVITE method), since the Request-URI is generated using the SIP routing procedures. Some options are:

1) Use a new SIP header.

2) Use a SIP message body:

   a) Dedicated message body for overlap signalling digits.

   b) Use message body for DTMF transport.

   c) Use message body used to transport PSTN information (e.g. the XML body for transporting ISDN/ISUP information defined as part of the IMS work in ETSI TISPAN WG3).

Some of the alternatives would require IETF standardization work, while other alternatives do not require that.

Media plane transport:

- Using this mechanism, a media plane channel will be established for transporting the additional digits. The advantage is that the impacts on intermediate SIP entities will be reduced even further, but it means that any entity which are interested in terminating and/or processing have access to the media plane, and it is assumed that media plane traffic will not be terminated in the network during the early phase of the dialog. This assumes the support of early media.

If the additional digits are transported on the media plane, some options are:

1) Send additional digits as out-of-band DTMF digits.

2) Send additional digits using MSRP.

3) Send additional digits using dedicated media plane protocol.

Some of the alternatives would require IETF standardization work, while other alternatives do not require that.

5.3.2 Multiple INVITE method

Routeing logic as outlined in clauses 5.3.2.1 and 5.3.2.2, can coexist and can both be allowed as deployment options.

5.3.2.1 Call ID Extended Routeing logic

Another solution would be to extend the routing logic of the core IMS to always route the subsequent requests that have the same Call-Id before any dialog is established to the same destination (see figure 5.3.2-1).

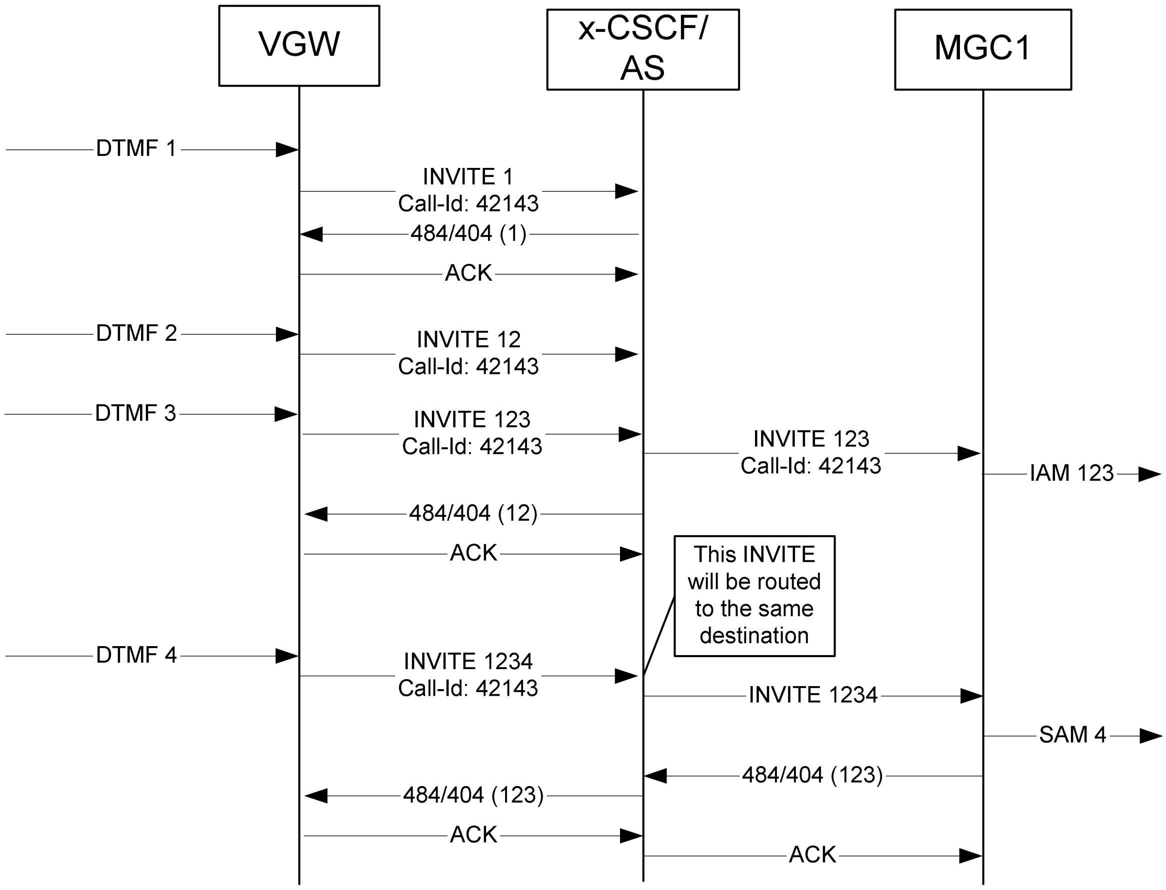

Figure 5.3.2.1-1

The disadvantage of this solution is that it requires an extension of the current core IMS routing logic. This extension thought does not break any IMS functionality and does not bring any major performance penalty as the preservation of the routing decision is only needed until the dialog is established. After that the resources can be freed.

The advantage of this solution is that it does not change the procedures defined by the ITU-T Recommendation Q.1912.5 [i.4] and RFC 3578 [i.3]. That means that the overlap interoperability with non IMS network that supports the overlap procedures according to RFC 3578 [i.3] and ITU-T Recommendation Q.1912.5 [i.4] can be guaranteed.

The SIP entities terminating SIP overlap procedures would not require establishing an early dialog, as for the mechanism in clause 5.4, in order to receive subsequent address information.

5.3.2.2 Use of deterministic routeing configuration.

Paths between UE and S-CSCF are established when the UE registers. INVITE request follow these deterministic paths from UE-A/VGW A/AGCF A to S-CSCF A and again from S-CSCF B to UE B, AGCF B or VGW B. This is accomplished using the SIP path header. Thus, intermediate nodes in these parts of the path do not need to apply any special procedures to accomplish deterministic routeing. This does not apply to the O-MGCF as it does not register with an S-CSCF.

Routeing algorithms at other network entities, such as the S-CSCF A or a BGF, are not fully standardized. However, these network entities may apply routeing databases that yield a deterministic result for a given number prefix. If such databases are used, a deterministic routeing of INVITE request relating to the same call in the encoding described in clause 5.2.2 will be accomplished without an inspection of the call ID. Features that would yield a non-deterministic routeing, such as algorithms for a dynamic load sharing, need to be disabled. This will apply to all sessions including non overlap cases.

As the same signalling format can be used for proposals in clauses 5.2.2 and 5.2.3, nodes behaving according to those proposals can coexist and can both be allowed as deployment options of the operator.

5.4 Interworking

5.4.1 Interworking between in-dialog and multiple-INVITE methods

5.4.1.1 General

In order for to support overlap signalling interworking between network supporting the in-dialog method and networks supporting the multiple-INVITE method an overlap signalling interworking function (OS-IWF) is defined. The OS-IWF supports both overlap signalling method.

The OS-IWF is a SIP B2BUA. It may for example be implemented in an IBCF, or as a separate entity.

5.4.1.2 Call from network supporting the in-dialog method

When the OS-IWF receives an initial INVITE request, it will forward it towards the other network, and it will establish an early dialog towards the originating party. When additional digits are received the OS-IWF will forward them in additional initial INVITE requests, according to procedures defined for the multiple-INVITE method.

When the OS-IWF receives responses, indicating that more digits are needed, it will not forward the response towards the originating party, unless additional digits are received within a specific digit collection time.

When the OS-IWF receives a response (provisional or final), not indicating that more digits are needed, the OS-IWF will forward the response towards the originating party.

The following two scenarios can be distinguished:

- Scenario 1: INVITE with SDP from "in-dialog" network to "multi-INVITE" network.

  This scenario shows the interworking with the originating side using the in-dialog method and the terminating side as using the multi-INVITE method. In this case, the initial INVITE request includes SDP.

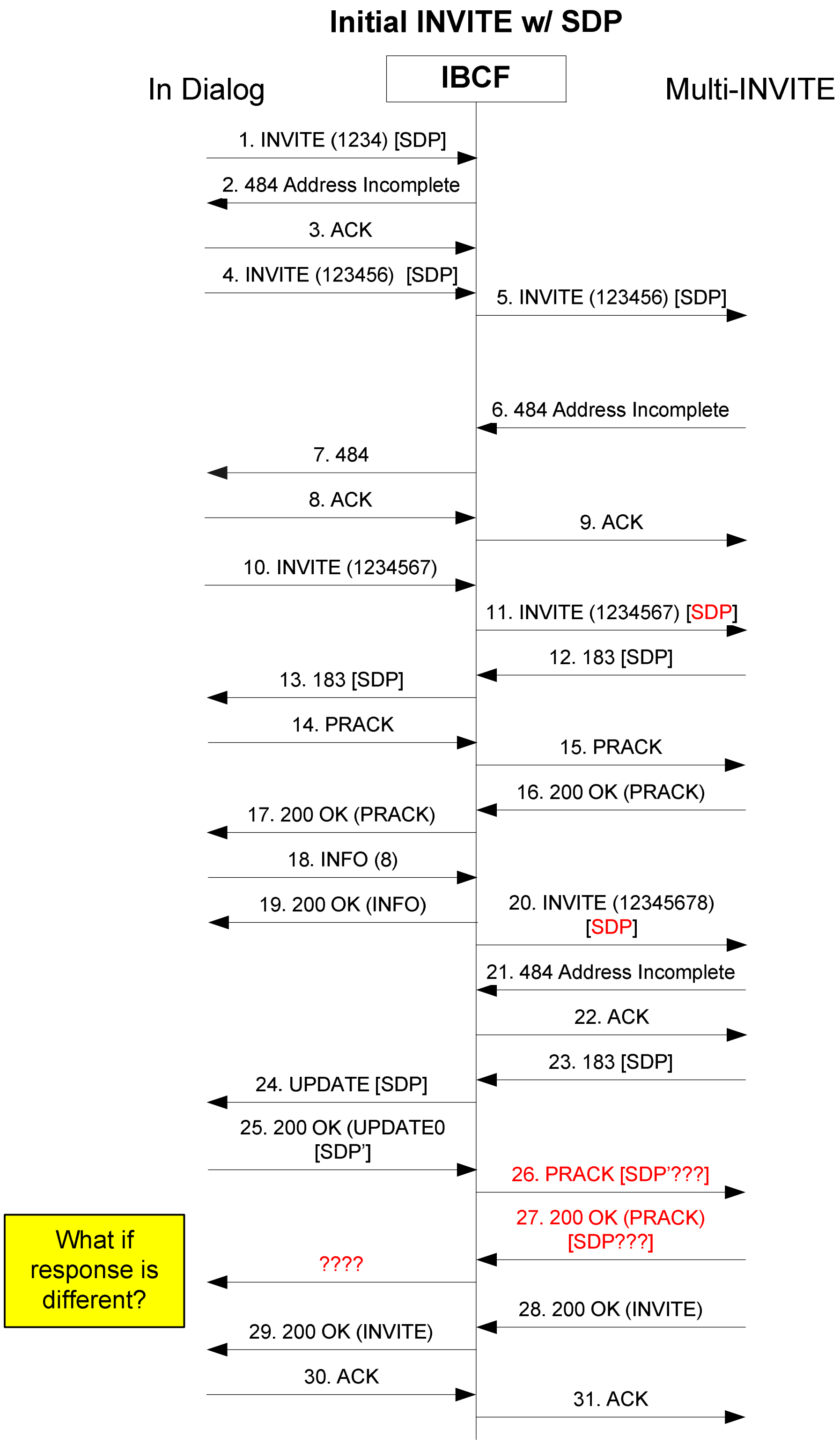

Figure 5.4.1.2-1

Each time a new INVITE request is sent into the terminating network, it will provide the latest SDP received from the originating side. Should the originating side provide new SDP (e.g., via UPDATE once initial offer/answer exchange is completed – not shown), this SDP provided for subsequent INVITE requests is sent. This requires the IBCF to remember the latest received SDP, even when no TrGW is used for the call.

When the terminating side returns a SDP answer, this will be propagated back into the originating network - completing the SDP offer/answer exchange in both networks. However, when subsequent INVITE requests are sent to the terminating network because of additional digits, the terminating network will repeat the SDP answer (perhaps with an updated SDP). To ensure that the most current SDP is provided to the originating side, this SDP is propagated through. However, since the SDP offer/answer exchange has already completed in the originating network, this SDP is sent as a new SDP offer, thus requiring a subsequent SDP answer. At this point, the SDP offer/answer exchange is imbalanced between the networks and the IBCF will then mediate between the two. Since it is always possible for a SDP change with each subsequent exchange, this cycle may never terminate. The interworking point has to be prepared to handle such a condition.

NOTE 1: Another alternative may be to send the SDP answer in a provisional response, using a separate To tag value.

NOTE 2: It may be possible to solve this issue by placing additional SDP requirements on the endpoints performing in-dialog overlap.

- Scenario 2: INVITE without SDP from "in-dialog" network to "multi-INVITE" network.

  This scenario shows the interworking with the originating side using the in-dialog method and the terminating side as using the multi-INVITE method. In this case, the initial INVITE request does not include SDP.

NOTE 3: IMS does not allow UEs to send INVITE without SDP, but there are scenarios where an AS is allowed to send INVITE without SDP.

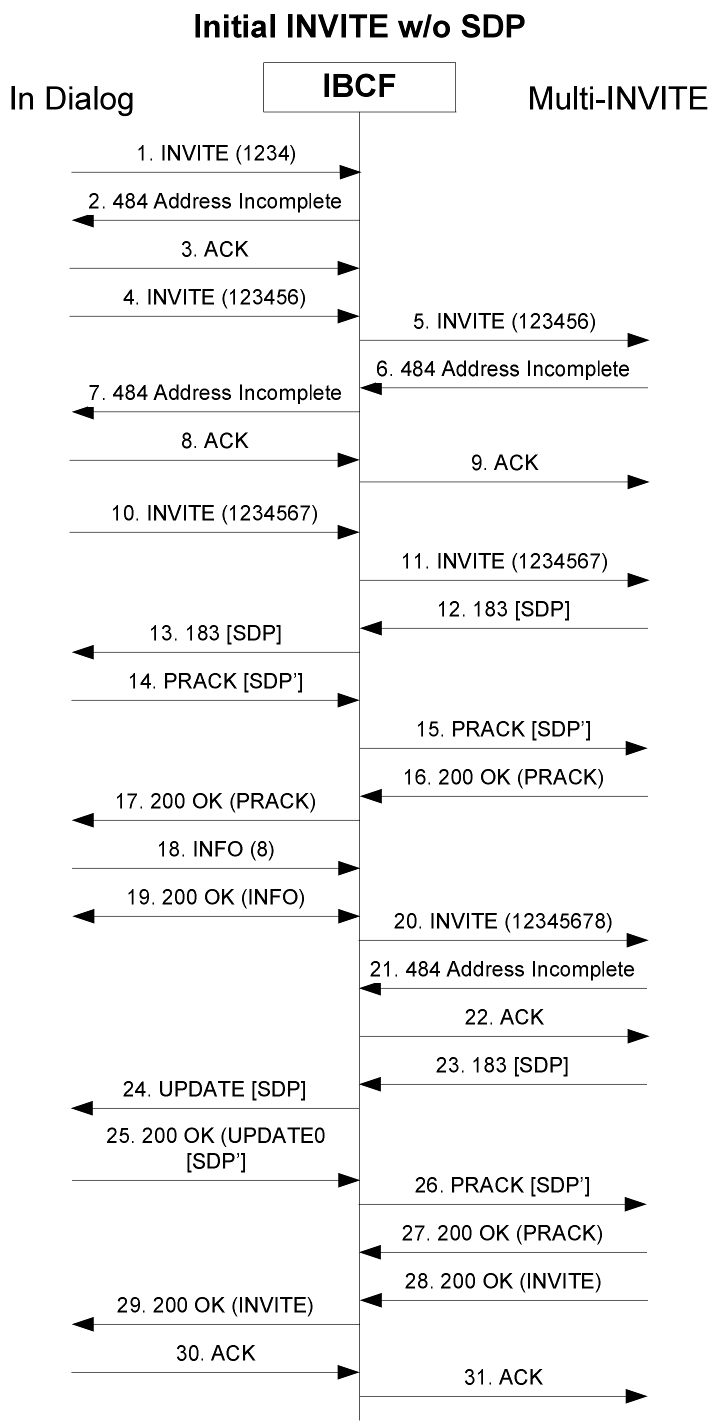

Figure 5.4.1.2-2

5.4.1.3 Call from network supporting the multiple-INVITE method

When the OS-IWF receives an initial INVITE request, it will forward it towards the other network. When additional digits are received from the originating user, and an early dialog has been established towards the other network, the additional digits will be forwarded according to the procedures defined for the in-dialog method.

When the OS-IWF receives additional responses (provisional or final), the OS-IWF will forward the response towards the originating party.

The following two scenarios can be distinguished:

- Scenario 1: INVITE with SDP from "multi-INVITE" network to "in-dialog" network.

  This scenario shows the interworking with the originating side using the multi-INVITE method and the terminating side as using the in-dialog method. In this case, the initial INVITE request includes SDP.

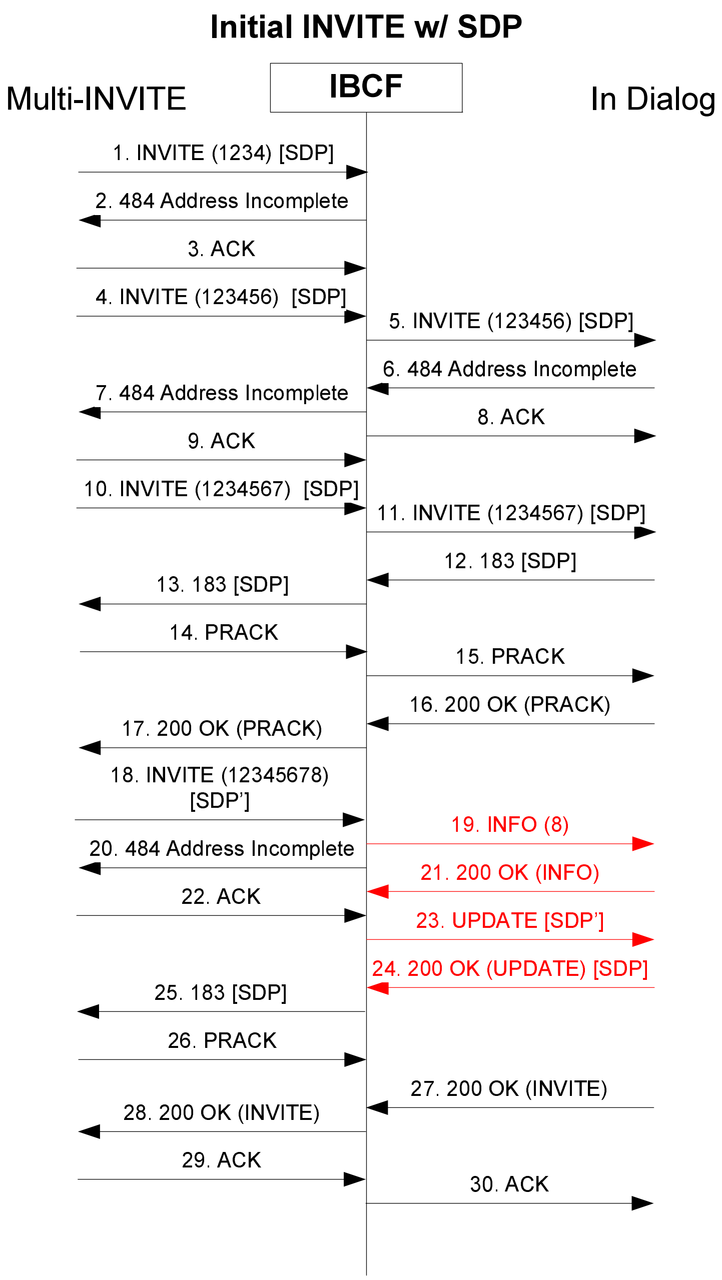

Figure 5.4.1.3-1

In this scenario, once the dialog is established on the terminating side, it will be necessary to interwork additional INVITE requests with new digits into both an INFO request to forward the new digits and an UPDATE request to propagate the SDP from INVITE request, if the SDP has changed.

NOTE 1: That it is possible that the SDP in additional INVITE requests be different from previously received SDPs.

- Scenario 2: INVITE without SDP from "multi-INVITE" network to "in-dialog" network.

  This scenario shows the interworking with the originating side using the multi-INVITE method and the terminating side as using the in-dialog method. In this case, the initial INVITE request does not include SDP.

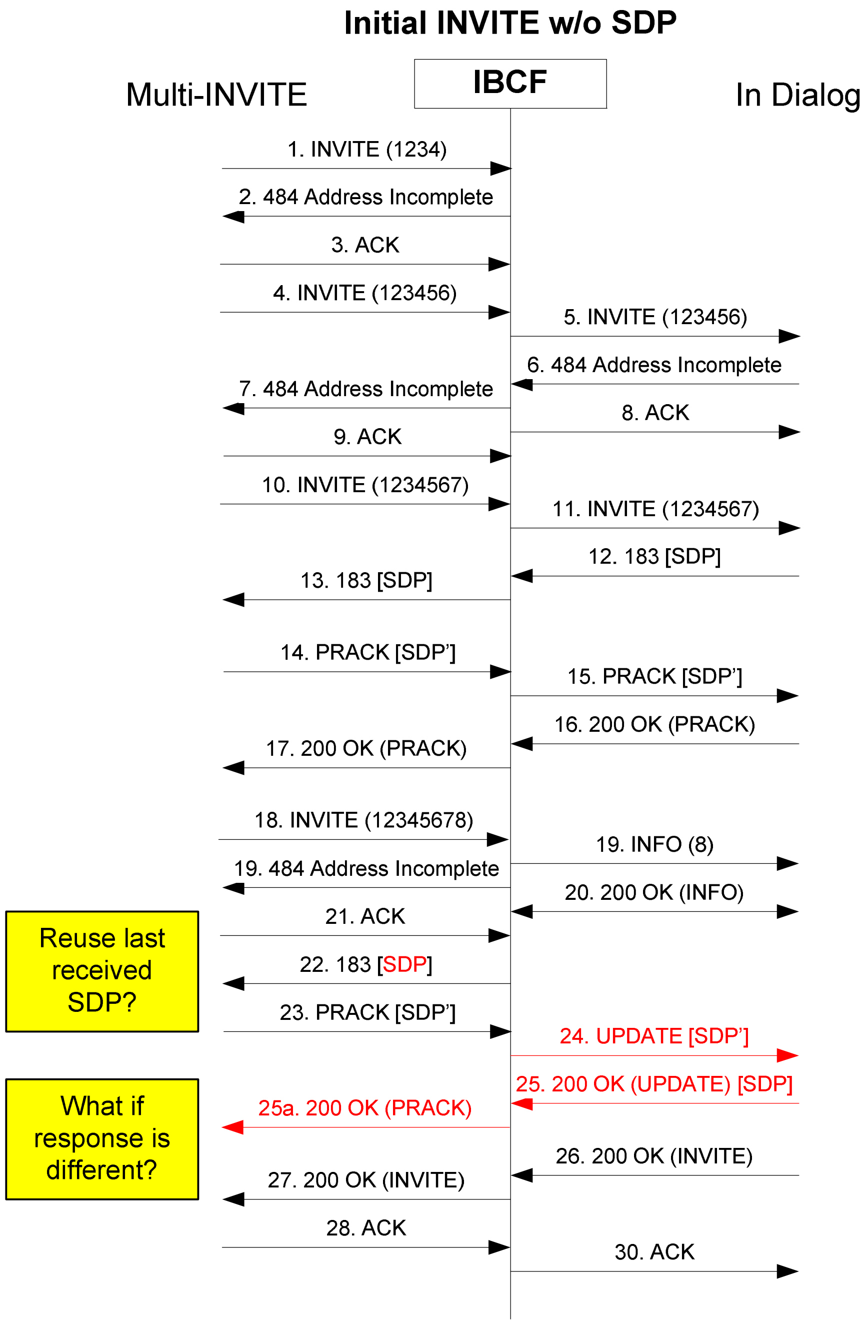

Figure 5.4.1.3-2

In this scenario, each time a new INVITE request is received with additional digits, it is necessary to repeat previously sent SDP offers. The SDP returned to the originating side is then considered to be the latest SDP received from the terminating side. When this SDP is returned due to additional INVITE requests received, the IBCF SDP offer/answer exchange will become imbalanced between the two networks resulting in the same issues as mentioned for scenario 1.

If the terminating side does not support the UPDATE method, then it is not even possible to forward the SDP. If the originating side SDP has changed, then the terminating side will not have valid SDP. This will result in the call being continued without a valid media path until a re-INVITE can be sent.

NOTE 2: This issue is solved if endpoints performing in-dialog overlap support the UPDATE method.

5.4.2 Interworking towards networks not supporting overlap signalling

5.4.2.1 Forwarding overlap signalling and handling error responses from the network not supporting overlap in appropriate manner in originating network as specified in TS 129 163

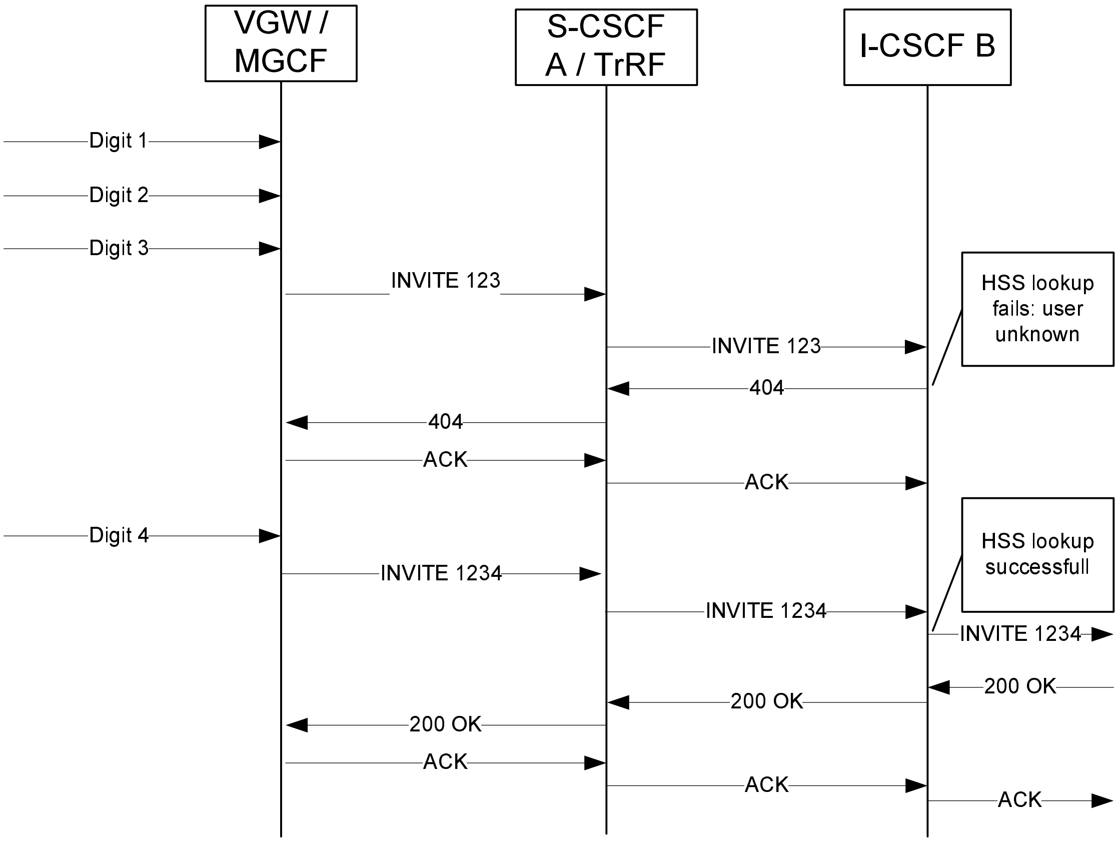

Figure 5.4.2.1-1: Forwarding overlap signalling and handling error responses from the network not supporting overlap in appropriate manner in originating network

A SIP proxy in a network not supporting overlap will not understand an uncomplete number in a request URI and reply with a 404 (not found) SIP error response according to SIP procedures.

The node as acting as originating UAC, e.g. the O-MGCF or the VGW, waits to receive extra digits upon this error response, and then retries the call setup with a more complete digit, treating the 404 response in the same manner as the 484 response. 3GPP has already specified this behaviour for the O-MGCF in TS 129 163 [i.8].

To reduce the signalling load, the node acting as originating UAC can use mechanisms as described in clause 5.2.

6.4.2.2 Digit Collection in originating network and forwarding en-bloc signalling.

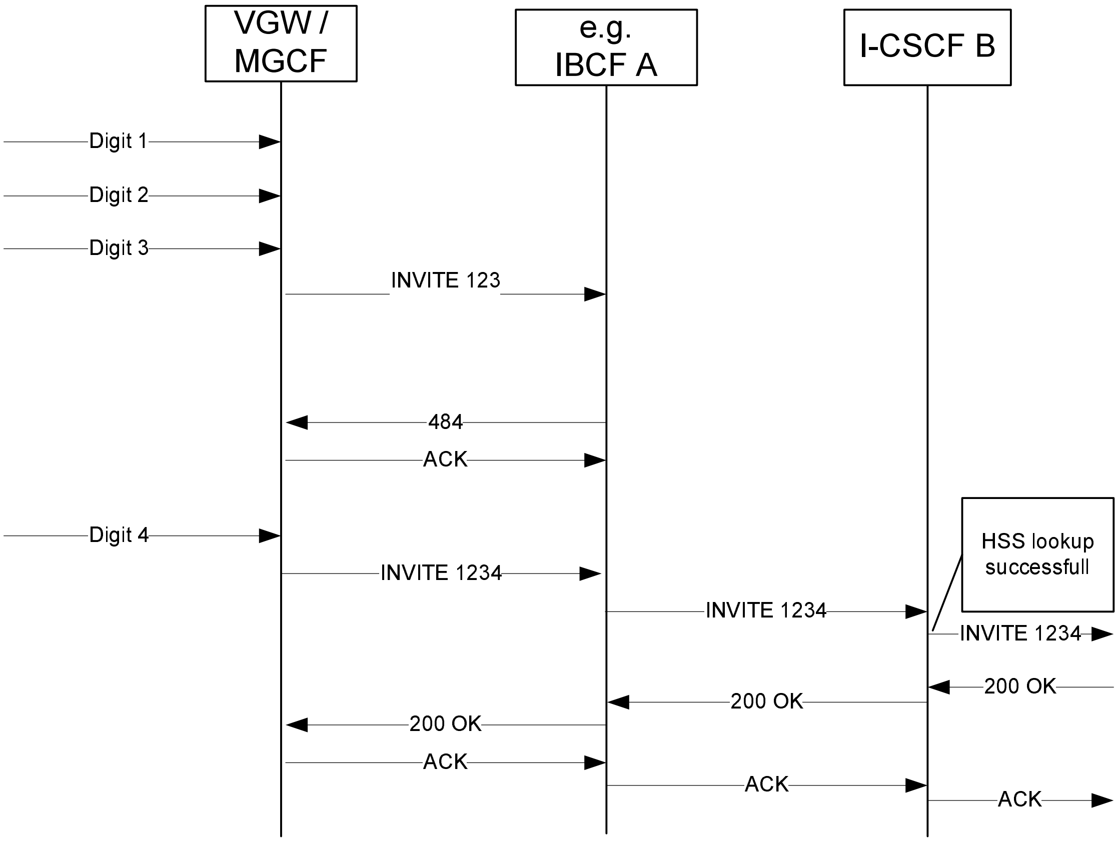

Figure 5.4.2.2-1: Digit Collection in originating network and forwarding en-bloc signalling

Some node in the originating network, e.g. the IBCF at the interconnection towards the terminating network, performs digit collection and forwards INVITE request only when they contain a complete number.

This node either has access to the numbering plan of the terminating network, e.g. as configured information within the node or some attached database, or it uses a long inter-digit timer (e.g. 12 s) to collect further incoming digits before forwarding the INVITE request.

5.5 Impacts on other interfaces/nodes in the IMS network

5.5.1 I-CSCF Impacts

When the I-CSCF receives each initial INVITE request, the I-CSCF starts the user location query procedure to the SLF.

1) If the query is successful, then it continues normal call processing and forwards the INVITE request to the identified S-CSCF per existing procedures.

2) If the query is unsuccessful, then the call will be handled per existing unsuccessful query procedures. Based on a network option, if the I-CSCF is not supporting transit functionality, then an unsuccessful SIP response will be returned according to existing procedures. This response may be a 404 (Not found) or 604 (Does not exist anywhere) in the case the user is not a user of the home network. If the I-CSCF is accommodating transit functionality then the INVITE may then be passed to a subsequent transit routing function.

3) If the query returns an indication of limited success, then the I-CSCF returns a 484 (Address Incomplete) response.

4) If a subsequent INVITE request is received the I-CSCF will repeat the user location query procedures to the SLF according to existing procedures and repeat the above steps.

Only step 3, which is only required for proposed solutions "Extension of the Error-Info Field" (see clause 5.2) and "Definition of an Overlap min Digit MIME body" (see clause 5.3), is an extension compared to existing procedures.

5.5.2 SLF

Upon receiving the user location query, the SLF attempts to determine if the provided Public Identity is known:

1) If the Public Identity known based on either the full subscriber DN being provided or based on a wildcard Public Identity, existing procedures in TS 129 228 [i.6], clause 6.1.4.1 for a known Public Identify are applied.

2) If the Public identity is not known, the existing procedures in TS 129 228 [i.6], clause 6.1.4.1 for unsuccessful match are applied (i.e., the Experimental-Result-Code is set to DIAMETER_ERROR_USER_UNKNOWN).

3) For proposed solutions "Extension of the Error-Info Field" (see clause 5.2) and "Definition of an Overlap min/max Digit MIME body" (see clause 5.3), if the SLF is able to identify a potential match of Public Identity then the SLF returns user location response with a Result-Code of DIAMETER-LIMITED-SUCCESS. A potential match occurs when the Public Identify digit string matches the leading digits of at least one identity, including potentially a wildcard Public Identity.

Only step 3, which is only required for proposed solutions "Extension of the Error-Info Field" (see clause 5.2) and "Definition of an Overlap min/max Digit MIME body" (see clause 5.3), is an extension compared to existing procedures.

5.5.3 S-CSCF

5.5.3.1 Originating

The originating S-CSCF performs a routeing of the call based upon the request URI. The mechanism to perform the routeing in the S-CSCF are not fully specified, but TS 129 229 [i.7] 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. A typical behaviour of an S-CSCF if an ENUM lookup fails is to route the call towards a BGCF. Thus, the routing decision of the S-CSCF for a shorter number of digits may be to route the call to the PSTN and for a subsequent INVITE with more digits may be to route to another IMS.

If overlap signalling is used and INVITES with incomplete numbers are received, the S-CSCF requires a sufficient number of digits to avoid an erroneous routeing to the PSTN.

This can be accomplished by aligning the routing database with the incomplete numbering plan at AGCF or VGW proposed in solution in clause 5.2.3. Alternatively, the routing database has the capability to identify incomplete numbers. As yet another alternative, the S-CSCF reject calls with unknown numbers with a 404 response, using entries in the routing database to identify calls towards the PSTN. The S-CSCF could also forward such calls to the BGCF, if the BGCF is configured to reject calls to unknown destinations with a 404 response.

If ENUM is used, the ENUM database in a typical deployment contains sufficient information about the first digits, as required to identify the destination IP domain. Therefore, ENUM is able to handle incomplete numbers in such deployments.

For proposed solutions "Extension of the Error-Info Field" (see clause 5.2) and "Definition of an Overlap min/max Digit MIME body" (see clause 5.3), the originating S-CSCF or an attached AS should feedback information about the minimum number of digits. This information should typically be requested from the routing database. For typical deployment that use ENUM as routing database, the ENUM protocol would need to be extended by IETF to provide information about the min/max digit length.

Furthermore, the ENUM protocol does not indicate if query fails because a number is incomplete or unknown. Again extensions to the ENUM protocol in IETF would be required if an indication of incomplete numbers is required.

If Overlap signalling encoded according to RFC 3578 [i.3] and ITU-T Recommendation Q.1912.5 [i.4] is used, the S-CSCF will also need to route the subsequent INVITE's related to a single call in deterministic fashion to the same destination, as discussed under solution proposals in clauses 5.3.1 or 5.3.2.

6 Comparison of solutions

6.1 Comparison of Signalling load reductions mechanisms

6.1.1 URL method (see clause 5.2)

6.1.1.1 Pros

- Backward compatible with existing overlap sending mechanism.
- Backward compatible with the originating terminals outside of the IMS network, as they can deal with the URL information in the sense that call failures do not occur.

6.1.1.2 Cons

- Can only be used with all the entities in IMS networks that have been upgraded to support the new overlap signalling mechanism.
- There is no reduction in the signalling load at the gateway once the peer SIP entity has been reached and is now interworking to the terminating PSTN for instance.
- Signalling load reduction is not as high in networks where the difference between minimum digit length that can be derived from first digits (e.g. city code), and average digit lengths applicable for those first digits, is high.
- Signalling load reduction is not as high due to possible race conditions (applies to the multiple INVITE methods) between 484 responses providing min length information and subsequent INVITES with additional digits.

6.1.2 MIME method (see clause 5.3)

6.1.2.1 Pros

- Backward compatible with existing overlap sending mechanism.

6.1.2.2 Cons

- Can only be used within the IMS network that supports the new overlap signalling mechanism.
- There is no reduction in the signalling load at the gateway once the peer SIP entity has been reached and is now interworking to the terminating PSTN for instance.
- May lead to call failures if call is propagated to SIP UAs not supporting multipart MIME.
- Signalling load reduction is not as high in networks where the difference between minimum digit length that can be derived from first digits (e.g. city code), and average digit lengths applicable for those first digits, is high.
- Signalling load reduction is not as high due to possible race conditions (applies to the multiple INVITE methods) between 484 responses providing min length information and subsequent INVITES with additional digits.

6.1.3 Digit Collection at AGCF or VGW (see clause 5.2.3)

6.1.3.1 Pros

- Fully backward compatible with existing overlap sending mechanism.
- Can achieve signalling load reductions for calls towards a PSTN.
- Can achieve signalling load reductions in the for calls towards an IMS not supporting overlap.
- Does not require database information about number lengths in the AGCF/AGW.

6.1.3.2 Cons

- Short digit collecting timer introduces short call set-up delay (e.g. 1 s).

6.1.4 Reason method (see clause 5.2.4)

6.1.4.1 Pros

- Does not require usage of timers.
- Backward compatible with existing overlap sending mechanism.

6.1.4.2 Cons

- Method only useful with entities that are able to interpret the information:
  - Additional digits may be sent even before the min number of digits have been received.
- Signalling load reduction is not as high in networks where the difference between minimum digit length that can be derived from first digits (e.g. city code), and average digit lengths applicable for those first digits, is high.
- Signalling load reduction is not as high due to possible race conditions (applies to the multiple INVITE methods) between 484 responses providing min length information and subsequent INVITES with additional digits

6.1.5 Conclusions

Void.

6.2 Comparison of Routeing mechanisms for Transmitting SIP networks

6.2.0 Comparison Criteria

For the purpose of comparison, the following criteria have been established:

- additional signalling and processing load;
- compatibility with the overlap Release 1 SIP mechanism as described in RFC 3578 [i.3];
- impact on existing nodes;
- impacts on systems not directly concerned with the use of overlap or en-bloc sending;
- impact on service level provided to the user;

- impact on transit networks.

6.2.1 In Dialog method (see clause 5.5)

Table 6.2.1-1: Evaluation of In Dialog method

Criteria	Pro	Con
Additional signalling and processing load	- No database/DNS lookups are needed for the in dialogue messages. - Option to transport the min digits according to clause 5.2.	- Will only reduce processing load, but not reduce substantially the number of messages.
Compatibility with RFC 3578 [i.3]/ITU-T Recommendation Q.1912.5 [i.4]		- Not backward compatible, without enabling interworking (see clause 5.4.1), with the existing method of deployed overlap sending as specified in RFC 3578 [i.3] and ITU-T Recommendation Q.1912.5 [i.4].
Impact on Existing Nodes	- No deterministic routing procedures will be needed. - A possible B2BUA in the IMS will not be required to apply special overlap related procedures when modifying the Call-Id and/or From tag. (see second bullet in clause 5.5).	- Mechanism for the encoding of digits for in dialogue method is yet to be defined. - Contradicts to 3GPP (TS 129 163 [i.8]: see clause 7.2.3.2.1a (in R7 and R8) would probably need revision to benefit from the proposed in-dialog approach).
Impact on other nodes and functions	- No impact on Nodes and functions not involved in Session signalling identified.	-
Impacts on service level	- Call success probability not impacted as only 1 Invite subject to Node overload control.	- May slow down call setup times until the early dialogue is established, as current overlap procedures allow that several INVITE transactions are open in parallel to forward overlap signalling quickly, but the proposal would require that only one transaction is open in order to decide if subsequent digits are to be signalled via INFO or INVITE.
Impacts on Transit networks	- Can be used across transit networks without any specific support for overlap signalling in those networks.	

6.2.2 Multiple INVITEs method (see clause 5.5 and related to clauses 5.2 and 5.3)

Table 6.2.2-1: Evaluation of Multi Invite method

Criteria	Pro	Con
Additional signalling and processing load	- option to transport min/max digits according to clause 5.3.	- Will not reduce substantially the number of messages. - Some higher performance degradation to be expected.
Compatibility with RFC 3578 [i.3]/ITU-T Recommendation Q.1912.5 [i.4]	- Backward compatible with the existing method of deployed overlap sending. - Aligned with ITU-T Recommendation Q.1912.5 [i.4]. - Aligned with RFC 3578 [i.3]. - Aligned with ETSI TISPAN. - Aligned with 3GPP (TS 129 163 [i.8]: see clause 7.2.3.2.1a).	
Impact on Existing Nodes	-	- Some extension to TS 129 163 [i.8]: see clause 7.2.3.2.1a (in R7 and R8) may be needed. - Extensions to current routeing functionality, as described in TS 129 229 [i.7], required to compare Call IDs (Call ID Extended Routeing logic only) of different INVTE requests. - Extensions to current routeing functionality, as described in TS 129 229 [i.7], required to route based upon configured prefixes (Use of deterministic routeing configuration only). - Causes limitation of basic IP routing capabilities (e.g. DNS , ENUM) , and hence be limiting to inherent distribution capabilities (supporting load sharing , redundancy schemes).
Impact on other nodes and functions	-	- Will impact statistics as every invite message may be counted as a call attempt. - May cause disturbances to charging systems if CDRs are generated for unsuccessful calls.
Impacts on service level	-	- At high network load call success probability may be lower as all INVITES are subject to Node overload control. - May slow down call setup times as in-dialog messages may be processed with preference over Initial requests without priority.
Impacts on Transit networks	-	- Cannot be used across transit networks unless those networks also support the multiple INVITE/deterministic routeing method.

6.3 Comparison of mechanisms for interworking towards networks not supporting overlap signalling

6.3.1 Forwarding overlap signalling and handling error responses from the network not supporting overlap in appropriate manner in originating network

6.3.1.1 Pros

- No dedicated interworking function required.

6.3.1.2 Cons

- Some extra signalling load in terminating network
  (However, this can be reduced by mechanisms to reduce signalling load used in the originating network).
- The terminating network may forward the session towards to the PSTN erroneously if the INVITE reaches the terminating S-CSCF.

6.3.2 Digit Collection in originating network and forwarding en-bloc signalling.

6.3.2.1 Pros

- Incomplete numbers are not sent to the terminating network.

6.3.2.2 Cons

- May require complex information about terminating side numbering plan to be configured at originating side, and may otherwise substantially delay setup of every call towards network not supporting overlap.
- Requires dedicated interworking function.

6.4 Impacts of inter-working between In-Dialog and Multi-INVITE overlap methods

When a same overlap signalling method is used on both interfaces, there are no special impacts when continuing the overlap signalling between networks. However, when the method is different between networks, and when it is not acceptable to add extra delay by converting the overlap signalling to en-bloc signalling, there are interworking impacts that must be addressed. The following summarizes the major interworking impacts between the two methods:

1) It may be necessary to repeat SDP offers or answers; this will require the IBCF to remember the latest received SDP.

2) The interworking may result in an imbalance of SDP offer/answer exchanges between networks which may not stabilize.

3) Once a dialog is established to allow the use of the INFO method, it may be necessary to interwork subsequent INVITE requests into both an INFO request and an UPDATE request. If there are cases when UPDATE is not supported, the terminating side may not have the latest and correct SDP information.

NOTE: To support interworking of overlap methods, endpoints performing in-dialog overlap should also support the UPDATE method.

7 Final Conclusion

In Line with the decision reflected in TS 129 163 [i.8] and TS 129 229 [i.7] Release 8, it is recommended to standardize both methods:

- In-Dialog; and
- Multiple-Invites.

The current version of the present document reflects the discussion status of May 2008. The present document will not be maintained.

Annex A: (network option): Call ID extended Overlap Dialling Procedures

These procedures are based upon the overlap dialling procedures described in clause 5.5.

A.1 Actions at the originating VGW/AGCF

A.1.1 Terminating overlap signalling at originating VGW/AGCF

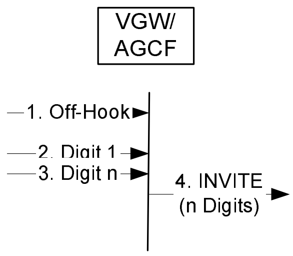

Figure A.1.1-1: Receipt of Digit information at the originating VGW/AGCF

After initiating the normal incoming PSTN call establishment procedures, determining the end of address signalling and selecting to route the call to the IMS domain, the originating VGW/AGCF sends the initial INVITE.

The end of address signalling is determined by the earlier of the following criteria:

a) by receipt of an end-of-pulsing (ST) signal; or

b) by receipt of the maximum number of digits used in the national numbering plan; or

c) by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party; or

d) by observing that timer Ta1 has expired after the receipt of the latest address message and the minimum number of digits required for routing the calls have been received.

If the end of the address signalling is determined in accordance with criteria a), b) or c), the timer Ta2 is started when INVITE is sent.

A.1.2 Sending of INVITE without determining the end of address signalling

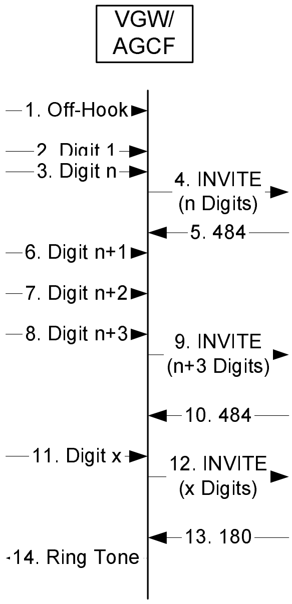

Figure A.1.2-1: Receipt of Digit information at the originating VGW/AGCF

1) As a network option, the originating VGW/AGCF may send INVITE requests without determining the end of address signalling. If the originating VGW/AGCF sends an INVITE request before the end of address signalling is determined, the originating VGW/AGCF:

   - uses the SIP precondition extension within the INVITE request;
   - starts timer Ta2; and
   - is prepared to process further incoming digits as described below;
   - is prepared to handle incoming SIP 404 or 484 error responses as detailed in clause 5.2.1.

2) As a network option on receipt of fresh address information from the originating side, the originating VGW/AGCF:

   - starts timer Ta4;
   - on receipt of further Digit information the timer is restarted;
   - at expiry of timer Ta4 the communication proceeds as described under 3.

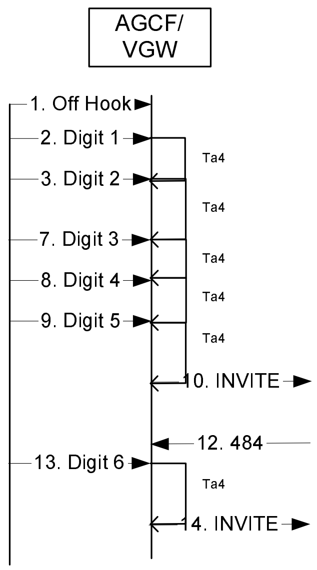

Figure A.1.2-2: Use of Timer Ta4 originating VGW/AGCF

3) On receipt of fresh address information from the originating side, the originating VGW/AGCF:

- stop timer Ta3 (if it is running);
- send an INVITE request complying to the following:
    - The INVITE request uses the SIP preconditions extension.
    - The INVITE request includes all digits received so far for this call in the Request-URI.
- if subsequent address information is received after the SIP 404 or 484 error responses have been received, the INVITE request additionally includes the digits received;
- restarts Ta2 or as a network option proceeds as described under 2.

If timer Ta2 has expired, the originating VGW/AGCF ignores subsequent Digit information received.

On receipt of a 183 Progress message, the originating VGW/AGCF stops Ta2.

A.1.3 Special handling of 404 Not Found and 484 Address Incomplete responses after sending of INVITE without determining the end of address signalling

This clause is only applicable when the network option of Sending of INVITE without determining the end of address signalling is being used (see clause 5.2.1).

On receipt of a 404 Not Found or 484 Address Incomplete response while Ta2 is running, the originating VGW/AGCF starts timer Ta3, if there are no other pending INVITE transactions for the corresponding call.

At the receipt of fresh address information, or a SIP 1xx provisional response, or a SIP 200 OK (INVITE), the originating VGW/AGCF stops Ta2 and Ta3. As a network optionTimer Ta4 may be started (The communication is then proceeding as described in clause A.1.2 Bullet point 2).

The originating VGW/AGCF sends a REL message with Cause Value 28 towards the BICC/ISUP network if Ta3 expires.

A.2 Actions at the terminating VGW/AGCF

No action with regard to overlap is needed.

DDI is out of scope of the present document.

A.3 Timers

Table A.3-1: Timers for interworking

Symbol	Time-out value	Cause for initiation	Normal termination	At expiry
Ta1	4 s to 6 s (default of 4 s)	When last address message is received and the minimum number of digits required for routing the call have been received.	At the receipt of fresh address information.	Send INVITE, send the address complete message
Ta2	4 s to 14 s (default of 4 s)	When INVITE is sent.	On reception of 180 Ringing , or 183 Session Progress and P-Early-Media header authorizing early media, or 404 Not Found or 484 Address Incomplete for an INVITE transaction for which Ta3 is running, or 200 OK (INVITE).	no action
Ta3	4 s to 6 s (default of 4 s)	On receipt of 404 Not Found or 484 Address Incomplete if there is no other pending INVITE transactions for the corresponding call.	At the receipt of SAM	Release call
Ta4	0,5 s to 4 s (default of 0,5 s)	On receipt of new Digit information.	On receipt of fresh address information	Send INVITE
NOTE 1: This timer is used when overlap signalling is received from BICC/ISUP network and converted to en-block signalling at the MGCF. NOTE 2: This timer is used to wait for a 404/484 response. NOTE 3: This timer is known as the "SIP dialog protection timer". This timer is only used where the O-MGCF is configured to send INVITE before end of address signalling is determined.				

Annex B: Example calculation for signalling load reduction as achieved by different proposals

This example calculation tries to compare the signalling load reduction achieved by the proposals in clause 5.2.

An international call to number 004981221875093 is assumed in the first table and a national call to number 081221875093 in the second table. The Callflow is similar to the one depicted in figure 5.2-1 and the same network entities are involved. The call is between two IMS networks supporting overlap signalling.

In the example calculations, the proposals with min length feedback achieved moderate signalling load reductions (43 % for international call and 33 % national call), and the proposal digit collection at AGCF/VGW/MGCF achieved about twice that reduction (83 % / 79 %). A combination of both proposals did only bring a small improvement compared to the digit collection at AGCF/VGW/MGCF (3 % / 1 %).

NOTE: Although the tables show multiple INVITEs, the calculation is also applicable to the in-dialog method.

Table B-1: Example calculation for international call

Digit			Proposals with feedback on min number length (see clauses 5.2.1 or 5.2.2)		Proposal number collection at AGCF/VGW/MGCF (see clause 5.2.3)		Combination of Number Collection at AGCF/VGW/MGCF and feedback on min number length (see clauses 5.2.1 or 5.2.2 and 5.2.3)	
			Number of INVITEs	**Comment**	**Number of INVITEs**	**Comment**	**Number of INVITEs**	**Comment**
1	0	Country City	1	6 is assumed to be provisioned as min length for non-local calls from S-CSCF A (e.g. in Germany short city code plus emergency extension)		6 is assumed to be configured at VGW as min length for non-local calls		6 is assumed to be configured at VGW as min length for non-local calls
2	0		1,5	Race condition between 484 containing min digit and next INVITE in every second call assumed		9 is assumed to be configured at VGW as min length for international calls (4 digit country code plus 2 digit city and 3 digit local extension)		9 is assumed to be configured at VGW as min length for international calls
3	4							
4	9							
5	8							
6	1	City Local	2,5	S-CSCF A has sufficient digits to forward call to terminating network (Germany) I-CSCF B derives that 4 more digits are required as minimum (completion of city code plus emergency extension length) and therefore feeds back min length 10 in 484.				
7	2		3	Race condition between 484 containing min digit and next INVITE in every second call assumed				
8	2							

Digit			**Proposals with feedback on min number length (see clauses 5.2.1 or 5.2.2)**		**Proposal number collection at AGCF/VGW/MGCF (see clause 5.2.3)**		**Combination of Number Collection at AGCF/VGW/MGCF and feedback on min number length (see clauses 5.2.1 or 5.2.2 and 5.2.3)**	
			Number of INVITEs	**Comment**	**Number of INVITEs**	**Comment**	**Number of INVITEs**	**Comment**
9	1				1	S-CSCF has sufficient digits for routeing	1	S-CSCF has sufficient digits for routeing Could be an emergency number, so three digit min length for local extension I-CSCF feeds back min length 11
10	8	Local	4	I-CSCF feeds back min length 12 for 4 digit local extension as this is no emergency call	2		1,5	Race condition between 484 containing min digit and next INVITE in every second call assumed
11	7		4,5	Race condition between 484 containing min digit and next INVITE in every second call assumed	3		2,5	No emergency call, min length 12
12	5		5,5		4		3,5	
13	0		6,5		5		4,5	
14	9		7,5		6		5,5	
15	3		8,5		7		6,5	
Total number of INVITEs with **in addition** short digit collection timer (see note)			-		2,5		2,375	
Total number of INVITEs			8,5		2,5		2,375	
Signalling load Reduction			**43 %**		**83 %**		**84 %**	
NOTE:			The short digit collection timer at AGCF is described in clause 5.2.3. It is assumed that for every second call, the caller uses either a phone book or a call button (e.g. in a DECT phone), and for these calls the short digit collection timer is able to collect all digits and only a single INVITE is required, For other calls, it is assumed that the digit collection timer is able to collect two digits on average. Thus, assuming that a call without digit collection timer requires *n* INVITES, a call with digit collection timer will require *1 + 0,25 * (n-1)* INVITEs.					

Table B-2: Example calculation for national call

Digit			**Proposals with feedback on min number length (see clauses 5.2.1 or 5.2.2)**		**Proposal number collection at AGCF/VGW/MGCF (see clause 5.2.3)**		**Combination of Number Collection at AGCF/VGW/MGCF and feedback on min number length (see clauses 5.2.1 or 5.2.2 and 5.2.3)**	
			Number of INVITEs	**Comment**	**Number of INVITEs**	**Comment**	**Number of INVITEs**	**Comment**
1	0	City	1	6 is assumed to be provisioned as min length for non-local calls from S-CSCF A (e.g. in Germany short city code plus emergency extension)	1	6 is assumed to be configured at AGCF/VGW/MGCF as min length for non-local calls	1	6 is assumed to be configured at AGCF/VGW/MGCF as min length for non-local calls
2	8		1,5	Race condition between 484 containing min digit and next INVITE in every second call assumed				
3	1							
4	2							
5	2							
6	1	Local	2,5	S-CSCF A has sufficient digits to forward call to terminating network (Germany) I-CSCF B derives that 2 more digits are required as minimum (3 digit minimal local length for emergency extension) and therefore feeds back min length 8 in 484.	1		1	S-CSCF A has sufficient digits to forward call to terminating network (Germany) I-CSCF B derives that 2 more digits are required as minimum (3 digit minimal local length for emergency extension) and therefore feeds back min length 8 in 484.
7	8		3	Race condition between 484 containing min digit and next INVITE in every second call assumed	2		2,5	Race condition between 484 containing min digit and next INVITE in every second call assumed
8	7		4	No emergency call, min length 4	3		3,5	
9	5		5		4		4,5	
10	0		6		5		5,5	
11	9		7		6		6,5	
12	3		**8**		**7**		**7,5**	
Total number of INVITEs with **in addition** short digit collection timer (see note)			-		2,5		2,375	
Total number of INVITEs			**8**		**2,5**		**2,375**	
Signalling load Reduction			**33 %**		**79 %**		**80 %**	

NOTE: The short digit collection timer at AGCF/VGW/MGCF is described in clause 5.2.3. It is assumed that for every second call, the caller uses either a phone book or a call button (e.g. in a DECT phone), and for these calls the short digit collection timer is able to collect all digits and only a single INVITE is required, For other calls, it is assumed that the digit collection timer is able to collect two digits on average. Thus, assuming that a call without digit collection timer requires *n* INVITEs, a call with digit collection timer will require *1 + 0,25 * (n-1)* INVITEs.

History

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