NICC ND 1021 V1.1.1 (2009-03)

NICC Document

Voice Line Control for UK Interconnect using TISPAN IMS-based PSTN/ISDN Emulation

NICC

Ofcom,

2a Southwark Bridge Road,

London SE1 9HA.

Normative Information

If you have any comments concerning the accuracy of the contents of this document, please write to:

The Technical Secretary, Network Interoperability Consultative Committee,

Ofcom,

2a Southwark Bridge Road,

London SE1 9HA.

Contents

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to NICC.

Pursuant to the NICC IPR Policy, no investigation, including IPR searches, has been carried out by NICC. No guarantee can be given as to the existence of other IPRs which are, or may be, or may become, essential to the present document.

Foreword

This NICC Document (ND) has been produced by NICC TSG AP WP

Introduction

This UK specification of the Session Initiation Protocol (SIP) and the Session Description Protocol (SDP) for Voice Line Control has been produced by the Technical Steering Group (TSG) of the Network Interoperability Consultative Committee (NICC). This specification is intended for use in the architectural environment described in the Interconnect Architecture for Voice Line Control Service between UK Next Generation Networks document – ND 1620 [9]. This document specifies the SIP and SDP required within public electronic communications networks (PECNs) in the UK to support services in a VLC User Network and a VLC Provider Network. The minimum service set supported SHALL be those currently supported by the PSTN but the design is such that it should not constrain development of future services.
“In the UK” is defined as the UK network up to and including the National component of an International Switching Centre. The actions at an International Gateway are beyond the scope of this document. Text describing any such actions is included for information only.

This issue of the specification contains the functionality of ETSI TS 183 043 “Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation Stage 3 specification” [1] modified to include additional functionality which is required for the UK VLC service.

This specification is written as an endorsement to ETSI documents. This is done by endorsing the ETSI documents and listing those sections of the ETSI documents that require a UK exception or addition.

Additionally the interaction between the Z interface and the Ic interface is described by means of flow diagrams and mapping tables. They are not a complete set of call flows covering every supplementary service in full detail, but should be sufficient for the detailed working of any other supplementary services to be derived from them. Some supplementary services (e.g. CFB, CFU, etc) are implemented entirely and some (e.g. Ring Back When Free, CFNR etc) are implemented almost entirely within the VLC User Network (possibly except for using the basic incoming call – see E.2.2).

See diagram below which is included to show the functions as described in the ETSI TISPAN Release 1 documents and their relationships with each other. However it should be noted that the actual implementations in the VLC User Network and VLC Provider Network are not constrained by the inclusion of the signalling model shown in Figure 0-1.

The functions shown above have equivalent functions in the VLC Architecture document ND 1620 [9] as shown below:

Function shown in Figure 0-1	Function shown in VLC Architecture
A-MGW	Access Gateway Func (fB4)
AGCF	Access Gateway Control Func (fC3)
IBCF	Edge Session Control Func (fC1) & Signalling Border Func (fB2)
I-BGF	IP Media Border Func (fB3)
I/S-CSCF	User Session Control Func (fC4)
AS	User Session Control Func (fC4)
MRFC	Not shown
MRFP	Not shown
RACS	BW Mang Func (fC2)

1 Scope

The present document endorses those parts of ETSI TS 183 043 "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation Stage 3 specification" [1] which refer to the Ic & Z reference points, the contents of which apply together with the addition of the modifications being covered herein. This document also endorses ETSI ES 183 028 "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Common basic communication procedures; protocol specification" [5], the contents of which apply together with the addition of the modifications being covered herein.

Note: Underlining and/or strike-out are used to highlight detailed modifications where necessary.

2 References

2.1 Normative references

[1] ETSI TS 183 043 Telecommunications and Internet converged Services and Protocols for Advanced Services (TISPAN); IMS-based PSTN/ISDN Emulation; Stage 3 Specification.

[2] ETSI TISPAN TR 183 056 –Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Feasibility study on new methods for overlap sending

[3] 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

[4] ND 1016 Requirements on communications providers in relation to Customer Line Identification Display and other related services

[5] ETSI TS 183 028 Telecommunications and Internet converged Services and Protocols for Advanced Services (TISPAN); Common basic communication procedures; Protocol specification.

[6] ETSI TS 182 012 Telecommunications and Internet converged Services and Protocols for Advanced Services (TISPAN); IMS-based PSTN/ISDN Emulation; Functional architecture

[7] ND 1019 IP Multimedia Call Control based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) for UK Interconnect

[8] RFC 3842 A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol

[9] ND 1620 Interconnect Architecture for Voice Line Control Service between UK Next Generation Networks

[10] RFC 3455 Private Header (P-Header) Extensions to the Session InitiationProtocol (SIP) for the 3rd-Generation Partnership Project (3GPP)

[11] RFC 3323 A Privacy Mechanism for the Session Initiation Protocol (SIP)

[12] RFC 3325 Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks

[13] ES 200 659-3 Access and terminals (AT); Analogue access to the Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services; Part 3: Data link message and parameter codings

[14] BT SIN 227 CDS Calling Line Identification Service – Service Description

[15] RFC 3261 Session Initiation Protocol

[16] ND 1615 NGN Interconnect; Voice Line Control Service; general connectivity management

Note: Documents with a "ND" reference may be obtained from http://www.nicc.org.uk

Documents with a "BT SIN" reference can be obtained from http://www.sinet.bt.com/

2.2 Informative references

[i.1] ETSI TS 124 229 Digital cellular telecommunications system (phase 2+);Universal Mobile Telecommunications System (UMTS); Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229)

3 Definitions and abbreviations

3.1 Definitions

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

The following definitions are contained in ND 1016 [4]:

calling line identity (CLI)
calling line identity presentation (CLIP) service

network number (NN)

presentation number (PN)

originating network

The network to which the customer who originates a call is directly connected.

Publicly Available Telephone Service

A service available to the public for originating and receiving national and international telephone calls and access to Emergency Organisations through a Telephone Number or Numbers in the National Telephone Numbering Plan or an international telephone numbering plan, and in addition may, where relevant, include one or more of the following services: the provision of operator assistance services, Directory Enquiry Facilities, Directories, provision of Public Pay Telephones, provision of service under special terms, provision of specific facilities for end-users with disabilities or with special social needs and/or the provision of non-geographic services.

required

Where a service/feature/message/parameter is qualified as “required” it SHALL be fully supported by the implementation concerned.
Note: The term may be applied independently to an interface protocol and/or the underlying functionality.

terminating network

The network to which the customer who receives a call is directly connected.

VLC User Network

The network which is providing the services to the VLC line and has the responsibility of collecting billing information for the VLC line. This network **uses** the functionality provided by the VLC Provider Network.

VLC Provider Network

This is the network that physically hosts the VLC line and the A-MGW to which it is connected. This network also has functionality which converts access line signalling to/from the SIP/SDP signalling between VLC User and Provider networks.

3.2 Abbreviations

3GPP 3rd Generation Partnership Project
3PTY Three Party Service

AGCF Access Gateway Control Function
A-MGW Access Media Gateway
AS Application Server
ASCII American Standard Code for Information Interchange

AVP	Audio/Video profile
BCD	Binary Coded Decimal
CLI	Calling line identity
CLIP	Calling line identification presentation
CPC	Calling party's category
CSCF	Call Session Control Function
CSH	Called Subscriber Hold
CW	Call Waiting
DTMF	Dual Tone Multi-Frequency
ETSI	European Telecommunications Standards Institute
FSK	Frequency Shift Keying
HOLD	Call hold
IBCF	Interconnect Border Control Function
I-BGF	Interconnect Border Gateway Function
I-CSCF	Interrogating Call Session Control Function
IMS	IP Multimedia core network Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network
MGW	Media Gateway (May be A-MGW or R-MGW)
MRF	Media Resource Function
MRFC	Media Resource Function Controller
MRFP	Media Resource Function Processor
ND	NICC Document
NGN	Next Generation Network
NICC	Network Interoperability Consultative Committee
Ofcom	Office of Communications (The Regulator for the UK Communications Industries)
OOR	Operator OverRide
PATS	Publicly Available Telephone Service
P-CSCF	Proxy Call Session Control Function
PECN	Public Electronic Communications Network
PSTN	Public Switched Telephone Network
RACS	Resource and Admission Control Subsystem
R-MGW	Residential Media Gateway
RFC	Request For Comments
S-CSCF	Serving Call Session Control Function
SOC	Switching Order Command
SDP	Session Description Protocol
SIN	Suppliers Information Note (BT interface description)
SIP	Session Initiation Protocol
TISPAN	Telecommunications and Internet converged Services and Protocols for Advanced Networking
TSG	Technical Steering Group
UE	User Equipment
UK	United Kingdom of Great Britain and Northern Ireland
URI	Uniform Resource Identifier
VGW	Voice over IP GateWay
VLC	Voice Line Control

4 Global modifications to TISPAN TS 183 043 [1]

Replace references as listed in the left hand column of the table below with those listed in the right hand column.

Reference in TS 183 043	Modified reference
ETSI ES 182 012: “Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation Subsystem; Functional architecture” [6]	ND 1620 Voice Line Control Service for UK Next Generation Networks [9]
ETSI ES 283 003 [3]	ND 1019 IP Multimedia Call Control based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) for UK Interconnect [7]

5 Exceptions and Additions to TS 183 043 [1]

5.1 Exceptions

TS 183 043 Clause/Sub-clause	Title	Comment
1	Scope	Replace the second Paragraph with the following: The present document is applicable to: • The reference point (Ic) between two IBCF's (one in the VLC Provider Network and the other in the VLC User Network); • The interaction between the analogue line (Z reference point) and the Ic reference point. Although the functionality in this document is described in terms of an IMS PES, this does not constrain the actual implementation to any particular physical entities, as long as the signalling over the Ic & Z reference points is not affected.
4.1	General	Replace fourth paragraph with the following: Analogue terminals are connected to access media gateways using standard analogue interfaces (e.g. SIN 242 & 351 to 354). These types of gateway are known as call control agnostic voice over IP media gateways (A-MGWs).
4.2	URI and address assignments	Replace the first paragraph with the following: When implicit (group) registration is used a private user identity SHALL only be provided for a group of subscribers connected to the same access media gateway (A-MGW). i.e. private identities SHALL NOT be provided on a per subscriber basis. Add the following sentence at the end of the second paragraph: The provisioning of the AGCF local database SHALL NOT be possible over the Ic reference point.
5.2.1	User Equipment	Replace the entire sub-clause text with the following: SIP based Voice over IP Gateways (VGW's) are not required to be supported for this issue of the specification, therefore the only type of UE in scope is analogue UE.
5.2.5	Media Gateway Controller Function (MGCF)	Delete the text in this sub-clause and replace the heading with "Void" (Out of scope of the present document).
5.3.1	PES Endpoint	Delete the text in this sub-clause and replace the heading with "Void".
5.3.2.2	Subscription for Dial Tone management	Add the following to the end of the first sentence: ",and the functionality described in RFC 3842 [8]." Add a new sentence at the end of the first paragraph: "The subscription SHALL be implicit." Delete the second paragraph ("The subscription may be …… profile delivery server."). In NOTE 2, replace the text "may as an option subscribe" with the text "SHALL implicitly subscribe" and delete the final sentence.

<table>
<tr>
<td>5.3.2.3</td>
<td>Registration Procedures</td>
<td>
Add the following to the beginning of this sub-clause (immediately after the heading):
REGISTER messages shall be sent by the VLC Provider network whenever one of the following events occurs:

When the first line or a new line is added to a group - as a result of an indication from the A-MGW that the line has been added to the VLC part of the Access Media Gateway (e.g. H.248 Service Change);
When the “expires” period ends (1800 - 600,000 seconds);
When it is detected that the signalling path from AGCF to the appropriate point in the user network has been lost (by use of OPTIONS messages);
When the AGCF data relating to a particular A-MGW termination is changed such that the termination is now included in a different group.

Add the following text to the second bullet of the second paragraph:
This approach implies explicit (single line) registration and, in order to prevent unacceptable delay in getting lines back in service, its use SHALL be limited such that a maximum number* of REGISTER messages are sent by the AGCF to any one of the associated VLC User Networks when an A-MGW comes into service (e.g. when a H.248 Service Change is received by the AGCF).
* This threshold SHALL be agreed bilaterally.
In paragraph 4, Item f) – Replace the NOTE with the following:
NOTE: The VLC User Network MAY respond to a REGISTER message with an Expires value less than 600,000 seconds but SHOULD NOT respond with an Expires value less than 1800 seconds. If the VLC Provider Network receives a response to a REGISTER message with an Expires value less than 1800 seconds it SHALL assume the requested Expires value to be 1800 seconds. It is the VLC User Network’s decision on the value (between 1800 and 600,000 seconds) the Expires interval will have.
In paragraph 4, Item k) - Replace “ES 282 010 [28]” with “RFC 3455 [10]” and ND 1615 [16].
In paragraph 4, Item l) - Replace the text with “the P-Access-Network-Info header SHALL be omitted.”
In paragraph 4, Item m) - Replace the text with “the P-Visited-Network-ID header SHALL be omitted.”
Replace the final paragraph (“When group….is ignored”) with the following:
When group (implicit) registration is used, the Public Identities contained in the P-Associated-URI header (in the 200 OK response to the REGISTER message) shall be used to control the provision of Dial Tone to the individual lines within the group. i.e. Dial Tone shall only be connected to a line if its Public Identity is included in the P-Associated-URI header. If Dial Tone is not connected to the line a calling customer SHALL NOT receive any substitute tone or announcement. Note: It is also possible for the VLC User Network to take the line temporarily out of service (TOS) without denial of Dial Tone by rejecting INVITE requests received from the VLC Provider Network (maybe with the exception of calls to 999 etc).
Add the following note:
Note: The format of Public (equivalent to VLC_Line_ID), Temporary Public (equivalent to VLC Line Group) & Private (equivalent to Authorisation Group) identities are described in ND 1620 [9]
Add the following text to the end of this subsection:
If topology hiding is being used in the VLC User Network the 200 OK Response
</td>
</tr>
</table>

		to the REGISTER message generated by the VLC User Network SHALL include a “Path Reference” in the Service-Route header. This Path Reference SHALL be used by the AGCF in conjunction with the unencrypted part of the Service Route to determine the different S-CSCF / path combinations to which it will send “heartbeat” OPTION messages at a rate of one every 10 seconds per unique S-CSCF / path. The format of the Service-Route header when encryption has taken place is shown in ND 1620 [9] section 11.6.
5.3.2.4	Outgoing Call Control procedures	Replace the second sentence with the following: The P-Asserted-Identity header SHALL be populated with the same URI as the From header but with the addition of the “cpc” parameter set in accordance with ND 1019 [7]. However the only allowed values shall be "ordinary" and "priority".
5.3.3.2	Basic call procedures	Replace the last paragraph with the following: When handling an outgoing call the PES application server MAY modify the contents of the P-Asserted-Identity header. How the PES application server determines the contents of this header is out of scope of the present document.
5.3.3.3	Announcement procedures	Replace the contents of this sub-clause with the following: The A-MGW will play some announcements as a direct result of receiving a SIP failure response (see E.1.1). Alternatively the full range of A-MGW announcements can be accessed by including an Error-Info header in the failure response (see F.4). There will be no indication to the VLC User Network when the announcement has finished as the SIP dialogue will already have ended. However at the end of the announcement the Number Unobtainable tone SHALL be provided to the calling customer and the A-MGW will await an “On Hook” indication from the analogue UE. Incoming Call attempts (INVITE’s) for this analogue line received by the AGCF during this period SHALL cause a “486 Busy Here” failure response. Announcements may also be provided by the VLC User Network using a MRF within that network.
5.3.3.4	Dial Tone Management	Replace this sub-clause with the following: The PES application server MAY notify the AGCF of changes to the type of Dial Tone to be applied to the analogue line using the Dial Tone Management documents as described in Annex A or using the method described in RFC 3842 [8].
5.3.3.5	Transport of ISUP Information	Delete the text of this sub-clause and replace the heading with “Void”.
5.3.5	PES Interworking Application	Delete the text in this sub-clause and replace the heading with “Void” (Out of scope of the present document).
5.3.6.2	Procedures related to NSS message bodies	Delete the text of this sub-clause and replace the heading with “Void”.
6.2.1	User Equipment (UE)	Replace the entire sub-clause text with the following: SIP based Voice over IP Gateways (VGW’s) are not required to be supported for this issue of the specification, therefore the only type of UE in scope is analogue UE.
6.3.1	PES Endpoint	Delete the text in this sub-clause and replace the heading with “Void” (Out of scope of the current version of this document).
6.3.2.2	Originating calls	Add the following to end of section 6.3.2.2: “In line with TSG Green Release requirements for UK VLC Tranche 1, G.711 A-law SHALL be the only supported codec.” This gives rise to the following SDP contents:

<table>
<tr><td></td><td></td><td>
<table>
<tr><td colspan="3">m=line</td><td>b=line
(Optional)</td><td colspan="2">a=line</td></tr>
<tr><td><media></td><td><transport></td><td><fmt-list></td><td><modifier>:
<bandwidth-value></td><td>rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>]</td><td>ptime: <packet time></td></tr>
<tr><td>audio</td><td>RTP/AVP</td><td>8</td><td>AS:64</td><td>rtpmap:8 PCMA/8000</td><td>ptime:10</td></tr>
</table>
</td></tr>
<tr><td>6.3.2.3</td><td>Terminating Calls</td><td>Add the following to end of section 6.3.2.3: “In line with TSG Green Release requirements for UK VLC Tranche 1, G.711 A-law SHALL be the only supported codec. This gives rise to the following SDP contents:
<table>
<tr><td colspan="3">m=line</td><td>b=line
(Optional)</td><td colspan="2">a=line</td></tr>
<tr><td><media></td><td><transport></td><td><fmt-list></td><td><modifier>:
<bandwidth-value></td><td>rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>]</td><td>ptime: <packet time></td></tr>
<tr><td>Audio</td><td>RTP/AVP</td><td>8</td><td>AS:64</td><td>rtpmap:8 PCMA/8000</td><td>ptime:10</td></tr>
</table>
This places a responsibility for transcoding on the VLC User Network if it receives an INVITE (destined for a VLC line) which does not include a G.711 A-law codec in the SDP.”</td></tr>
<tr><td>7</td><td>Protocol using H.248 for PES</td><td>Delete the text in this Clause and replace the heading with “Void”</td></tr>
<tr><td>Annex B</td><td>AGCF Internal communication</td><td>Delete the contents of this annex and replace the heading with “Void”.

Note: the internal communications within an AGCF are out scope of this NICC specification.</td></tr>
<tr><td>Annex C (Informative)</td><td>Implementation of Supplementary Services</td><td>Replace “(Informative)” in the title with “(Normative)”</td></tr>
<tr><td>C.1.1</td><td>Introduction</td><td>SIP based Voice over IP Gateways (VGW’s) are not required to be supported for this issue of the specification, therefore:

Delete the final sentence of the second paragraph of C.1.1 (“Similar procedures…..P-CSCF in the PES”)

Delete the third paragraph of C.1.1 and replace with the following:

AGCF involvement in the execution of these services is limited to the generic capabilities described in the call flows in annex E.2.

In this annex some supplementary services are described for both loose and tight AGCF/AS coupling. For this specification only tight coupling SHALL be implemented.

At the end of sub-clause C.1.1 add the following:

Note: Some supplementary services are described in Annex E.2 of the present document by means of Call Flows. In the event of any discrepancy between annex C of TS 183 043 and the Call Flows in Annex E.2, the Call Flows in Annex E.2 SHALL take precedence.</td></tr>
</table>

C.1.2	Supplementary service control	Replace the first paragraph with: This annex assumes that subscribers can control their supplementary services using service code commands and switching order commands. These commands may be as defined in ETS 300 738 [12] or may use other coding schemes as determined by the VLC User Network. Add the following to the end of the existing text: Depending on the subscriber profile, service code commands and switching order commands may be dialled after a RECALL during an existing call. Processing of commands in this circumstance follows the same principles as call flow E.2.6. Processing of the INVITE (flash) at the AS depends on the call configuration and often involves sending a 484 back to the AGCF to request the access media gateway to deliver Dial Tone and collect digits. (Note: in some cases the AS will send a 200 OK back to the AGCF when Dial Tone and collection of digits is not required – see E.2.4.1).
C.1.2.1.2	Generic procedure at the AGCF side	Replace the bullet point beginning “A Request-URI structured as follows” with the following: A Request-URI as for an outgoing call (see steps 7 to 18 of call flow E.2.1 or steps 7 to 8 of call flow E.2.5). Delete the contents of Note 1.
C.1.2.1.3	Generic procedure at the AS side	Add the following text to the end of the last paragraph: Alternatively an announcement may be generated directly by the A-MGW by inclusion of an Error-Info header in the failure response to the INVITE message (see F.4).
C.1.2.2	Switching order commands	Replace the second paragraph with: The format of switching order commands (SOC) as defined in ETS 300 738 [12] is reproduced below: Delete the last paragraph.
C1.4	Supplementary services using ISUP information	Delete the contents of sub-clause C.1.4 and replace the title with “Void”.
C.2	Advice of Charge	Replace the contents of clause C.2 with the following: The Advice of Charge service MAY be provided by the VLC User Network by the inclusion of a SIP MIME body of type “application/X-Display-Data-Block” (see Annex F.1) in an unsolicited NOTIFY message (see E.2.14) or INVITE sent to the VLC Provider Network.
C.3.3	Actions at the Terminating AS	Replace the first paragraph with the following If the service is activated, the terminating AS rejects the call if the Privacy header field indicates "id", "header" or "user" as specified in RFC3325. In all other cases the communication proceeds normally.
C.5.1	Actions at the originating AGCF	Replace the text of this sub-clause with the following: The AGCF shall forward the service code command (if received from the A-MGW) and subsequent digits exactly as dialled to the VLC User Network, in an INVITE message or INVITE messages.
C.5.2 second bullet	Actions at the Originating AS	Replace the second bullet of C.5.2 with: • Include a Privacy header field set to "header" & "id" in accordance with RFC 3323 [11] and RFC 3325 [12].
C.5.3	Actions at the Terminating AS	Replace the text in this sub-clause with the following: The CLI information received in the From, P-Asserted-Identity and Privacy header fields SHALL be passed on unchanged to the terminating AGCF. If the

		called user has subscribed to the CLIP/CND service this information MAY also be used to derive the CLI information to be included in an optional “X-Display-Data-Block” MIME type formatted in accordance with ETSI ES 200 659-3 [13] / BT SIN 227 [14] or some other scheme chosen by the VLC User Network (see F.1 of this document for encoding of this information). Note that other information (e.g. the number of messages waiting, Advice of Charge etc) MAY also be included in the “X-Display-Data-Block”.
C.5.4	Actions at the Terminating AGCF	Replace the text in this sub-clause with the following: If the received INVITE contains a MIME body of type “application/X-Display-Data-Block” its contents SHALL be copied unchanged to the A-MGW (e.g. using the H.248 andisp package). If the INVITE does not contain a MIME body of type “application/X-Display-Data-Block” and the line is subscribed to CLI delivery then the mapping described in Annex D.2 as modified by this specification SHALL apply. In all other cases, no action is taken by the AGCF in relation to this service.
C.6.3	Actions at the Terminating Application Server	Replace the text in this sub-clause with the following: The information received in the From, P-Asserted-Identity and Privacy header fields SHALL be passed on unchanged to the terminating AGCF. If the called user has subscribed to the Calling Name Delivery service this information MAY also be used to derive the Calling Party Name information to be included in an optional “X-Display-Data-Block” MIME type formatted in accordance with ETSI ES 200 659-3 [13] / BT SIN 227 [14] or some other scheme chosen by the VLC User Network (see F.1 of this document for encoding of this information). Note that other information (e.g. the number of messages waiting, Advice of Charge etc) MAY also be included in the “X-Display-Data-Block”.
C.6.4	Actions at the Terminating AGCF	Replace the text in this sub-clause with the following: If the received INVITE contains a MIME body of type “application/X-Display-Data-Block” its contents SHALL be copied unchanged to the A-MGW (e.g. using the H.248 andisp package). In all other cases, no action is taken by the AGCF in relation to this service.
C.7.1.1	Actions at the AGCF	Replace the text of this sub-clause with the following: The AGCF shall forward the service code commands exactly as dialled to the VLC User Network, in an INVITE message or messages.
C.7.2.6	Actions at the terminating AGCF	Replace the text of this sub-clause with the following: No specific action is performed by the terminating AGCF in relation to this service. However if the received INVITE contains a MIME body of type “application/X-Display-Data-Block” its contents SHALL be copied unchanged to the A-MGW (e.g. using the H.248 andisp package). Note: this MIME body may include a Redirecting Number parameter.
C.8.3	Actions at the Terminating Application Server	Add the following to the end of the existing text: See Annex F.2 for the coding of the Alert-Info header.
C.8.4	Actions at the Terminating AGCF	Add the following to the end of the existing text: See Annex F.2 for the coding of the Alert-Info header.
C9.1.1	Actions at the AGCF at the terminating side	In C.9.1.1 first paragraph add “(D2)” immediately after “INVITE” and replace the third bullet of the first paragraph with: • “Send a 182 (Queued) towards the AS in the VLC User Network” Delete the fourth bullet of the first paragraph. Delete the second paragraph.

		Replace the third paragraph, its bullet item list and Note 1 with the following: If a flash-hook event is reported by the media gateway (as shown in E.2.4.1), the AGCF requests the media gateway to set the stream mode of the ephemeral termination to inactive. Delete the fourth paragraph, and replace with the following: The AGCF SHALL send an INVITE (flash) on dialogue D4 to the AS in the VLC User Network and await receipt of 200 OK (Invite) and when this is received it SHALL send an ACK. The AGCF SHALL then send a 200 OK (Invite) on dialogue D2 and await receipt of the ACK. The AGCF then awaits receipt of a Re-INVITE on dialogue D2, (which allows re-negotiation of the SDP between user A/B and user C) responds with a 200 OK (Invite) and awaits receipt of an ACK and a BYE (D4). On receipt of this BYE it sends a 200 OK (Bye). Delete the fifth paragraph ("Processing of the……the tight coupling case."). Delete the text of NOTE 2. Add the following after NOTE 2 (now deleted) as a new paragraph: The call flows in Annex E.2.4.1 (RECALL to accept waiting call) & E.2.4.2 (ON HOOK to accept waiting call) replace Figure C.2 of TS 183 043.
C.9.1.2	Actions at the AS at the terminating side	Modify the fourth paragraph by adding "(D2)" immediately after "INVITE" Modify the fifth paragraph by replacing "re-INVITE request with a SDP "sendonly" attribute" with "182 Queued response", by replacing "held" with "calling" and by replacing "in accordance with TS183 028 [5]" with "and starts a timer (which is stopped on receipt of an INVITE (flash)) to determine the overall call waiting active period (if this timer expires a CANCEL (D2) is sent to the VLC Provider Network)". Add the following new paragraphs immediately after the fifth paragraph: When (if) an INVITE (flash) (D4) is received the AS in the VLC User Network SHALL send a 200 OK response (as this is Simplified Call Waiting) and await receipt of the ACK. The AS in the VLC User Network shall await receipt of a 200 OK (Invite) on dialogue D2, and respond with an ACK. The AS in the VLC User Network SHALL then re-arrange the bearers to connect the calling party (C) to the called party (A/B) by sending a Re-INVITE (D2) with no SDP to the AGCF in the VLC Provider Network. It then awaits receipt of the 200 OK (Invite) containing the SDP of the A/B party and when received it sends an ACK to the AGCF in the VLC Provider Network containing the SDP of the C party. The AS MAY also arrange for an announcement to be played to the held party (B/A) via an MRFC/MRFP in the VLC User Network and SHALL arrange for a BYE (D4) to be sent to the AGCF in the VLC Provider Network. The call flow for this service (when accepting the waiting call with a RECALL) is shown in E.2.4.1 and is described in the previous paragraphs of this sub-clause. The call flow for this service (when accepting the waiting call by going ON HOOK) is shown in E.2.4.2.
C.9.2	Option 1 (Loose Coupling)	Delete the contents of sub-clause C.9.2 and replace the heading with "Void"
C.9.3	Option 2 (Tight Coupling)	Delete the contents of sub-clause C.9.3 and replace the heading with "Void"
C.10.1.1	Actions at the AGCF	Replace the text of this sub-clause with the following: The AGCF shall forward the service code commands exactly as dialled to the

<table>
<tr><td></td><td></td><td>VLC User Network, in an INVITE message, or messages.</td></tr>
<tr><td>C.11.4</td><td>Actions at the Terminating AGCF</td><td>Replace the text of this sub-clause with the following:

The AGCF shall forward the service code command exactly as dialled to the VLC User Network, in an INVITE message or INVITE messages. The AGCF will also send an INVITE (flash@domain) to the VLC User Network when a RECALL is received from the analogue UE, as this may be used during the call to invoke the service. (The method of invocation is determined by the VLC User Network).</td></tr>
<tr><td>C.12</td><td>Message Waiting Indicator</td><td>Replace the contents of clause C.12 with the following:

The Message Waiting Indicator service MAY be provided by the VLC User Network by the inclusion of a SIP MIME body of type “application/X-Display-Data-Block” (see Annex F.1) in an unsolicited NOTIFY(see E.2.14) or in the next INVITE message sent to the VLC Provider Network.</td></tr>
<tr><td>C.14.1.1</td><td>Actions at the AGCF at the service invocation side</td><td>Replace the text “Figure C.3” in the second sentence of the first paragraph with “Call Flow E.2.6” and delete “between the AGCF and the AS”.

Replace the text of the two bullets of the second paragraph with the following:

• Send an INVITE (flash) request on dialogue D2 to the VLC User Network and await a 484 (Address Incomplete) response

• On receipt of a 484 (Address Incomplete) it SHALL instruct the A-MGW to play Dial Tone and notify when a match to the digit map has occurred

• On receipt of a notification of the dialled digits, send an INVITE (D2) message or messages containing these digits to the AS in the VLC User Network and awaits a provisional response (183 or 180).

• Receipt of a non-200 final response other than 484 (Address Incomplete) SHALL be treated in the same way as if the analogue RECALL signal had not been received.

• Receipt of a 200 OK (invite) response to the INVITE (flash) indicates that the VLC User Network has acted on the INVITE (flash) and the AGCF does not need to play Dial Tone or collect further digits.

Delete the third paragraph.

Modify the beginning of the fourth paragraph “On receipt of the 183 (Session Progress)…” by adding the text “or 180 (Ringing)”, and delete the text of the second bullet in this paragraph.

Modify the text of the fifth paragraph by replacing “re-INVITE” with “INVITE (flash)”.</td></tr>
<tr><td>C.14.1.2</td><td>Actions at the AS at the service invocation side</td><td>In the second and third paragraphs, replace “re-INVITE” with “INVITE”.

In the third paragraph delete the text “with an SDP “a=sendonly” attribute”

Delete Figure C.3.</td></tr>
<tr><td>C.14.2</td><td>Option 1 (Loose Coupling)</td><td>Delete the contents of sub-clause C.14.2 and replace the heading with “Void”.</td></tr>
<tr><td>C.14.3</td><td>Option 2 (Tight Coupling)</td><td>Replace the title “Option 2 (Tight Coupling)” with “Tight Coupling”.

Add the following text to beginning of this clause:

The following description identifies the actions required when the service user wishes to establish a three party call (by using Register RECALL and dialling a switching order code SOC), i.e. from step 26 in E.2.6.</td></tr>
</table>

C.14.3.1	Actions at the AGCF at the originating side	Replace the text of this sub-clause with the following: On receipt of a notification of Register RECALL from the A-MGW, the AGCF opens a new dialogue (D3) and sends an INVITE (flash) to the AS in the VLC User Network. This INVITE includes the following: • The Request-URI is structured as follows: -A user part containing “flash” -A domain name that together with the user part provides sufficient information for the VLC User Network to forward the request to the appropriate AS, based on Initial Filter Criteria stored in the user profile, e.g. “flash@cs21.vlc.<CP name>.uktel.org.uk“ • A From header containing the VLC_Line_ID of the line on which the RECALL occurred. • An SDP offer for a speech call The AGCF now awaits receipt of a 484 Address Incomplete from the originating AS in the VLC User Network, and when received the AGCF takes the following actions: • Requests the A-MGW to play Dial Tone and collect one digit. • Sends an INVITE (D3) containing this single digit (as this is a Recall sequence with more than one active dialogue) and await receipt of 200 OK (Invite) or a failure response code. This INVITE is built in the same way as the previous INVITE except that the dialled digit replaces “flash”. The AGCF then awaits a re-INVITE (D2) with the SDP of a Media Server in the VLC User Network (acting as a 3 party bridge)and when received it takes the following actions: • Sends an instruction to the A-MGW to change the address to which RTP packets are sent and from which they are received (e.g. it modifies the H.248 Remote Descriptor) • Sends a 200 OK (Invite) to the AS in the VLC User Network, awaits receipt of a BYE to end dialogue D3, and when this is received it sends a 200 OK (Bye).
C.14.3.2	Actions at the Originating AS at the originating side	Replace the text of this sub-clause with the following: On receipt of an INVITE (flash) (D3) the AS in the VLC User Network takes the following actions: • Sends a 484 Address Incomplete response to the AGCF in the VLC Provider Network and awaits receipt of an ACK. • Awaits receipt of an INVITE (D3) with a single digit and when this is received sends a 200 OK (Invite) to the AGCF in the VLC Provider Network and awaits receipt of the ACK. • Sends a Re-INVITE (D2) with the SDP of a Media Server (acting as a 3 party bridge) to the AGCF in the VLC Provider Network, awaits a 200 OK (Invite) and when this is received it sends an ACK followed by a BYE (D3). • It then awaits a 200 OK (Bye).
C.15.1	AGCF at the served user side	Replace the text of this sub-clause with the following: The AGCF shall forward the service code commands exactly as dialled to the VLC User Network, in an INVITE message or INVITE messages.
Annex D	Mapping between SIP and the subscriber	Add the following text to the beginning of sub-clause “D.2 Call Setup Message”: This mapping SHALL be used only if the received INVITE does not contain a

	line protocol	SIP MIME body of type “application/X-Display-Data-Block” and it is known (by a service mark in the AGCF) that the subscriber indicated by the called public identity has subscribed to the display service covered in this sub-clause, Note: Alternatively, this mapping MAY be done at the terminating AS in the VLC User Network (rather than in the AGCF as described in this Annex) and conveyed over the Ic reference point using a SIP MIME body “application/X-Display-Data-Block” (as described in Annex F.1) Replace Table D.1 with the replacement table that follows. Delete sub-clauses D.3 & D.4.
Annex E	Bibliography	Renumber as Annex H

Table D.1 Call set-up message parameters (replacement table)

Parameter type	Does Privacy header contain "id", "user" or "header"?	Does From contain an E.164 number?	Does P-Asserted-Identity contain an E.164 number?	Does "cpc" in P-Asserted-Identity header contain "payphone" or "operator"	Populating Rules
Date and Time	X	X	X	X	Set from local clock
Calling Line Identity	Y	X	X	X	Omit this parameter
	N	Y	X	X	Set according to contents of "From" header. Note 1
		N	Y	X	Set according to contents of "P-Asserted-Identity" header. Note 1
		N	N	X	Omit this parameter
Reason for absence of Calling Line Identity	Y	X	X	X	Set to "Private" (0101 0000)
	N	N	N	X	Set to "unavailable" (0100 1111)
		N	Y	X	Omit this parameter
		Y	X	X	
Called Line Identity	X	X	X	X	Set from "P-Called-Party-Identity" header. Note 1
Calling Party Name	Y	X	X	X	Omit this parameter
	N	X	X	Y	Set to "Payphone" or "Operator" respectively.
		N	N	N	Omit this parameter
		X	Y	N	If country code of E.164 number in P-Asserted-Identity is not "44" then set to "International", else omit this parameter
		Y	N	N	Omit this parameter
Reason for absence of Calling Party Name	Y	X	X	X	Set to "Private" (0101 0000)
	N	X	X	X	Omit this parameter
Call type	X	X	X	X	Set to "Normal (voice) call" (0000 0001)

Note 1: If the country code is "44" remove the "+44" from the beginning of the userinfo portion of the URI and replace with "0" before mapping to the parameter indicated above. Otherwise remove the "+" from the beginning of the userinfo portion of the URI and replace with "00" before mapping to the parameter indicated above. In either case the maximum length of the parameter shall be 22 characters and strings longer than this shall be truncated.

5.2 Additions

The additions to TS 183 043 [1] are shown in Annexes E, F and G

6 Exceptions and Additions to TS 183 028 [5]

6.1 Exceptions

TS 183 028 Clause/Sub-clause	Title	Comment
4.3	Alternative ringtone	Add the following to the end of the existing text: If either the VLC Provider or VLC User Network wishes an alternative ringtone to be played it SHALL arrange for this entirely in its own network. An Alert-Info header SHALL NOT be included in the 180 Ringing sent between the VLC User and Provider Networks.
A.1.2	Including Alert-Info header field in the 180 (Ringing) response	Delete this section. Note: If the VLC User or Provider Network wishes an alternative ringing tone to be played it SHALL arrange for this entirely in its own network i.e. the Alert-Info header SHALL NOT be included in the 180 Ringing sent between the VLC User and Provider Networks.

6.2 Additions

None.

Annex E (normative): Interactions between Z interface & Ic interface

E.1 Mapping Tables

E.1.1 Mapping of SIP Response Codes to Audible Tones/Announcements in VLC Provider Network

These are the default mappings that shall be used if the response code does not contain an Error-Info header with an "Announcement Indicator" (See F.4 for full details).

The default mappings are not applicable to non-200 final responses received following an INVITE (flash). See E.2.6 for this case.

SIP Response Code	**Audible Tone or Announcement**
18x except 180	None
180	None Note: see the outgoing call flows, e.g. E.2.1 flow 19, for the use of the P-Early-Media header.
400 Bad Request	Connection Not Admitted Indication (NU tone)
401 Unauthorized	Call cannot be connected announcement
402 Payment Required	Call cannot be connected announcement
403 Forbidden	Call cannot be connected announcement
404 Not Found	Unrecognised number announcement
405 Method Not Allowed	Call cannot be connected announcement
406 Not Acceptable	Call cannot be connected announcement
407 Proxy Authentication Required	Call cannot be connected announcement
408 Request Time-out	Sorry there is no reply announcement
410 Gone	Unrecognised number announcement
413 Request Entity Too Large	Connection Not Admitted Indication (NU tone)
414 Request-URI Too Long	Connection Not Admitted Indication (NU tone)
415 Unsupported Media Type	Call cannot be connected announcement
416 Unsupported URI Scheme	Connection Not Admitted Indication (NU tone
420 Bad Extension	Call cannot be connected announcement
421 Extension Required	Call cannot be connected announcement
423 Interval Too Brief	Call cannot be connected announcement

433 Anonymity Disallowed	Call cannot be connected announcement. The VLC User Network SHOULD NOT send this response code to the VLC Provider Network but SHOULD provide an announcement and indicate this to the VLC Provider Network. The announcement may be generated at the VLC User Network or switched through from the far end network.
480 Temporarily Unavailable	Temporary Out of Order announcement.
481 Call Leg/Transaction Does Not Exist	Connection Not Admitted Indication (NU tone)
482 Loop Detected	Connection Not Admitted Indication (NU tone)
483 Too Many Hops	Connection Not Admitted Indication (NU tone)
484 Address Incomplete	None (VLC Provider Network awaits further digits). A timer SHALL be run by the VLC Provider Network awaiting further digits (which may already have been received by the time the 484 is received). If the timer expires the procedure shown in E.2.10.2 SHALL be followed.
485 Ambiguous	Unrecognised number announcement
486 Busy Here	Number Busy Tone
487 Request Terminated	Connection Not Admitted Indication (NU tone)
488 Not Acceptable Here	Call cannot be connected announcement
491 Request Pending	None
493 Undecipherable	Connection Not Admitted Indication (NU tone)
500 Server Internal Error	Connection Not Admitted Indication (NU tone)
501 Not Implemented	Connection Not Admitted Indication (NU tone)
502 Bad Gateway	Connection Not Admitted Indication (NU tone)
503 Service Unavailable	Path Engaged Tone
504 Server Time-out	Fault announcement
505 Version Not Supported	Connection Not Admitted Indication (NU tone)
513 Message Too Large	Connection Not Admitted Indication (NU tone)
580 Precondition Failure	All lines busy announcement
600 Busy Everywhere	Number Busy Tone
603 Decline	Connection Not Admitted Indication (NU tone)
604 Does Not Exist Anywhere	Connection Not Admitted Indication (NU tone)
606 Not Acceptable	Call cannot be connected announcement

Any Other Response Code	Connection Not Admitted Indication (NU tone)

Note: The announcement “Sorry, a technical problem has occurred, there is no need to report this problem as we are already aware of it – please try again later” SHALL be autonomously played to the calling customer by the A-MGW in the event of its isolation from the rest of the network.

E.2 Call Flow Diagrams

The call flow diagrams that follow have been produced either:

- to replace call flow diagrams shown in TS 183 043 [1] (e.g. Figures C.2 & C.3) and are annotated as doing this where this is the case, or
- because there are no equivalent call flows provided in TS 183 043 [1] or the underlying base specifications (i.e. ES 283 003 [3]) to cover Basic Call Setup and Clear-down (with UK specific clearing programmes) or a particular UK specific service (e.g. Operator Override).

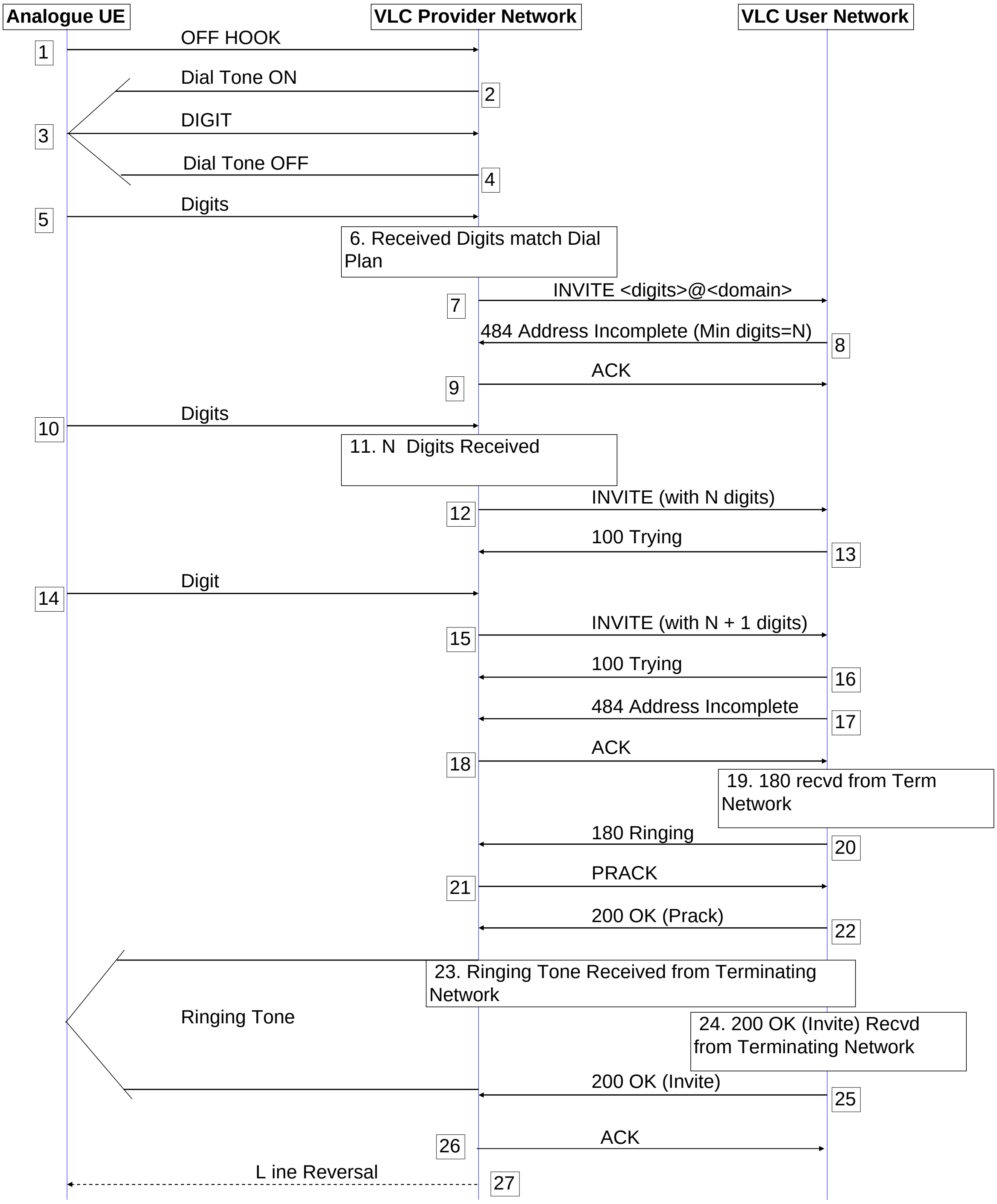
They are not a complete set covering every supplementary service in full detail, but should be sufficient for the detailed working of any other supplementary services to be derived from them. The minimum set of services SHALL be those currently deployed in the UK PSTN.

It is envisaged that there may be breaks in the media path whilst a service scenario is being set-up. The implementations should ensure that the break in the media stream should be as short as possible and no longer than 400ms.

Note 1: Some supplementary services (e.g. CFB, CFU, etc) are implemented entirely and some (e.g. Ring Back When Free, CFNR etc) are implemented almost entirely within the VLC User Network (possibly except for using the basic incoming call – E.2.2).

Note 2: 100 Trying shown in some call flows is only required if a response or provisional response cannot be returned within 200 ms.

E.2.1 Outgoing Call Attempt (Z interface idle) – Overlap digit sending



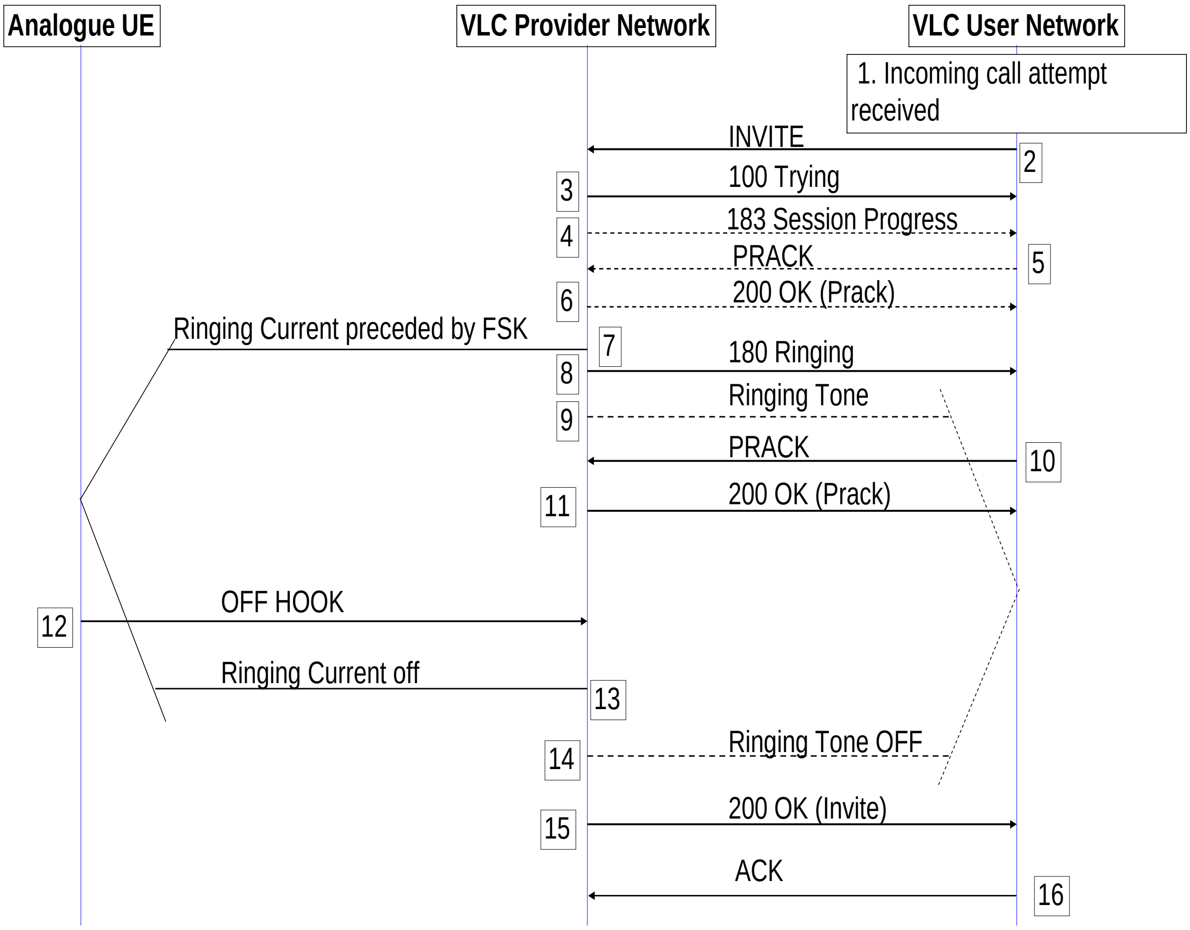
Note: "Preconditions" which optionally may be included have not been shown on this Call Flow

Flow Number	Action	Additional Comments
1	VLC Provider network identifies correct profile for this line and applies correct digit map (including timers). Note: An OFF HOOK is either: • A Loop (for a DEL or Loop Calling PBX); or • An earth on the B leg (for an Earth calling PBX)	Once the OFF HOOK is detected any subsequent INVITE received by the VLC Provider Network is rejected with a 486 Busy Here.
2	VLC Provider network applies correct Dial Tone (Ordinary Dial Tone or Message Waiting Dial Tone)	
3	Calling customer sends the first digit (DTMF or Loop Disconnect). VLC Provider network detects digit and removes Dial Tone	
4	Dial Tone removed	
5	VLC Provider Network starts digit analysis against applied dial plan	
6	The received digits match the dial plan	The VLC Provider Network checks to see if there is any available bandwidth for the call. If there is no available bandwidth to the VLC User Network for the type of call (ordinary / priority) the VLC Provider Network SHALL connect the caller to the All Lines Busy announcement.
7	The VLC Provider Network sends an INVITE to the VLC User Network including the following contents: Request-URI – contains digits received from calling customer (sip:<digits>@<domain> - see ND 1620 [9]) To: – contains the same as the Request-URI From: - contains the VLC_Line_ID of calling line Route: - contains the same as the Service-Route header in the 200 OK response to the associated previous REGISTER message. P-Asserted-Identity: - contains the VLC_Line_ID of calling line and the cpc parameter set in accordance with 5.1/5.3.2.4 of the present document. P-Charging-Vector: - Contents in accordance with RFC 3455 [10] and ND 1615 [16]. Require: 100rel Supported: preconditions SDP Offer The VLC User Network marks the line as “busy on an outgoing call”. (This status will be maintained until the calling customer goes On Hook and a BYE is received by the VLC User Network - see E.2.8)	For lines with PATS the following rules apply: If there is a PN associated with the calling line the VLC USER Network SHALL remove the URI from the From header and replace it with the PN. Otherwise it SHALL remove the URI from the From: header and replace it with the NN before forwarding the INVITE. The PN or NN shall be a Tel or SIP URI containing an E.164 number beginning with “+44”. The VLC User Network SHALL also modify the P-Asserted-Identity header so that it contains the NN (Tel or SIP URI containing an E.164 number beginning with “+44”) before forwarding the INVITE towards the terminating network. The VLC User Network SHALL, when appropriate, add

		the SIP Privacy header (or equivalent CLI restricted/withheld indication if interworking directly to another signalling system) before forwarding the call towards the terminating network. The addition of the Privacy header (with priv-value = "id") or equivalent indication should be determined by a combination of receipt of a prefix (e.g.141 or 1470) and the subscribed privacy service for the calling line. For definitions of PN & NN see ND 1016 [4]
8	VLC User Network analyses the digits received in the INVITE and determines that the minimum number of digits required (N digits) has not been received. So a 484 (optionally with an indication that N digits are required) SHALL be sent to the VLC Provider Network. Note: The case when the complete number has been received is shown in E.2.5. The VLC Provider Network SHALL start a timer and await further digits (which may already have been received by the time the 484 is received). If the timer expires (no / insufficient further digits) the actions shown in E.2.10.2 shall be followed.	**Note: Work is currently being progressed within ETSI TISPAN that will standardise the inclusion of a "Min Digits = N" indication in the 484, however it is not expected that this work will be completed until after the required issue date of this specification. The method of including "Min Digits = N" is therefore described in section F.3 of this specification. Optionally the VLC User Network MAY send a 484 without a "Min Digits = N" indication, but this will mean that each further digit received by the VLC Provider Network will result in a new INVITE (and in many cases a 484 & ACK)**
9	ACK to 484	
10	Further digit(s) are sent.	
11	A total of N digits have been received so far	
12	The VLC Provider Network sends a new INVITE with N digits (but otherwise the same contents as the first INVITE)	
13	VLC User Network sends 100 Trying	
14	A further digit is received from the calling customer	
15	A new INVITE is sent containing N+1 digits (but otherwise the same contents as the previous INVITEs)	
16	VLC User Network sends 100 Trying	
17	VLC User Network sends 484 Address Incomplete (to terminate the dialogue started by the second INVITE at 12)	
18	ACK to 484	

19	When the VLC User Network receives a 180 RINGING from the terminating network it SHALL arrange for the bearer to be connected to a suitable audible tone and SHALL include a P-Early-Media header with parameter "sendrecv" or "sendonly" in the 180 RINGING sent to the VLC Provider Network. The audible tone may be generated at the VLC User Network or switched through from the far end network. The latter would normally be indicated by the presence of a suitable P-Early-Media header in the 180 RINGING from the terminating network.	
20	A 180 Ringing which shall include a P-Early_media header (with SDP Answer) is sent to the VLC Provider Network. The VLC Provider Network SHALL switch-through the forward and backward bearer paths and provide an o/g half ECD if not already done.	
21	PRACK	
22	200 OK (Prack)	
23	Ringing Tone (either from the terminating network or from the VLC User Network) is sent to the calling customer. This tone is removed (either by the terminating network or the VLC User Network as appropriate) when the called customer answers.	
24	When the VLC User Network receives a 200 OK (Invite) (or other Answer indication) from the terminating network it sends a 200 OK (Invite) to the VLC Provider network. If the VLC User Network is providing Ringing Tone it SHALL now re-arrange the bearers to connect the calling customer through to the called customer. This is achieved by sending a Re-INVITE (with no SDP) to the VLC Provider Network.	
25	On receipt of 200 OK (Invite) the VLC Provider Network MAY, depending on the service mark for the line, apply a line reversal at the Z reference point.	
26	ACK	
27	Optional Line Reversal	

E.2.2 Incoming Call attempt (Z interface IDLE)

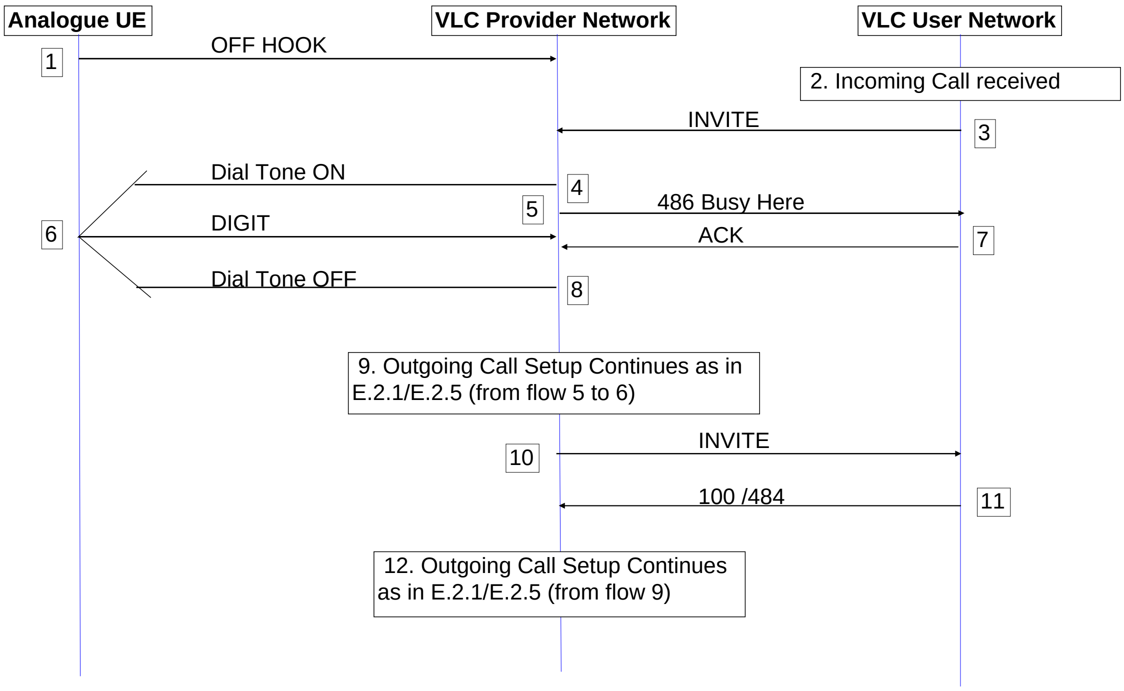


Flow Number	Action	Additional Comments
1	An incoming call attempt is received by the VLC User Network which then marks the line as busy on an incoming call.	The VLC User Network checks to see if there is any available bandwidth for the call. If there is no available bandwidth to the VLC Provider Network for the type of call (ordinary / priority) the VLC User Network SHALL reject the call attempt by sending a SIP 580 Response (or equivalent) to the preceding network.
2	INVITE sent which includes the following: Request-URI – contains the VLC_Line_ID as defined in	If the called customer has subscribed to the CND or other equivalent service, the VLC User

	ND1620 [9]. Note that the VLC_Line_ID is equivalent to the IMS Public Identity To: – contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. From: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be a PN if there is one else the NN. P-Asserted-Identity: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be the NN. P-Charging-Vector: - Contents as in the received INVITE or if call is received in another signalling system generated in accordance with RFC 3455 [10] and ND 1615 [16] Require: 100rel Supported: preconditions SDP Offer and may include: Alert-Info: <data:,RCxx> “X-Display-Data-Block”	Network MAY map the From: header in the sent INVITE (which may have been derived from an equivalent CLI parameter of another signalling system e.g. UK ISUP according to ND 1020 etc) to the “Calling Line Directory Number” parameter of the “Display Data Block” (application/X-Display-Data-Block) The mapping is described in section F.1 of this document.
3	100 Trying	
4	Optional 183 Session Progress	
5	PRACK (only if 4 occurs)	
6	200 OK (Prack) (only if 5 occurs)	
7	Ringing current (cadence as indicated in Alert-Info header or default cadence if header is not present) which MAY be preceded by FSK (derived from contents of “X-Display-Data-Block” if present)	
8	180 Ringing (with SDP Answer). The P-Early-Media header with parameter value "sendrecv" SHALL be included.	
9	Ringing Tone sent in the bearer from the A-MGW towards the I-BGF in VLC Provider network	
10	PRACK	
11	200 OK (Prack)	
12	Called Customer answers which automatically stops ringing current (13) & ringing tone (14) and causes a 200 OK (Invite) to be sent to the VLC User Network (15)	
13	Ringing current off & forward bearer switch through	
14	Ringing Tone off & backward bearer switch through	

15	200 OK (Invite)	
16	ACK	

E.2.3 Incoming Call attempt (Z interface busy - setting up outgoing call)



Note: "Preconditions" which may optionally be included have not been shown on this Call Flow

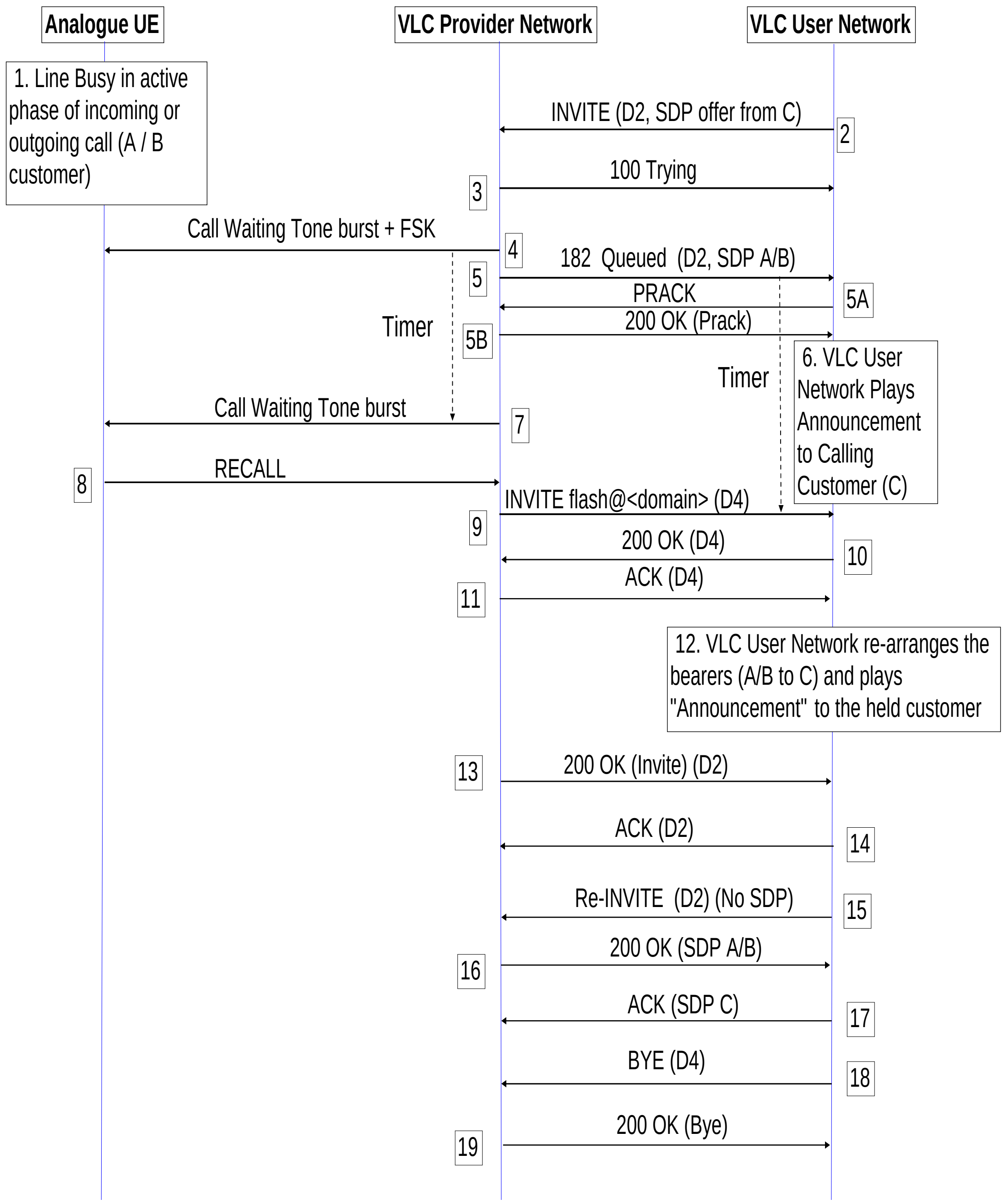
Flow Number	Action	Additional Comments
1	VLC Provider network identifies correct profile for this line and applies correct digit map (including timers). Note: An OFF HOOK is either: • A Loop (for a DEL or Loop Calling PBX); or • An earth on the B leg (for an Earth calling PBX)	
2	An incoming call attempt is received by the VLC User Network which then marks the line as busy on an incoming call.	The VLC User Network checks to see if there is any available bandwidth for the call. If there is no available bandwidth to the VLC Provider Network for the type of call (ordinary / priority) the VLC User Network SHALL reject the call attempt by sending a SIP 580 Response (or equivalent) to the preceding network.
3	INVITE sent which includes the following: Request-URI – contains the VLC_Line_ID as defined in ND1620 [9]. Note that the VLC_Line_ID is equivalent to the IMS Public Identity To: – contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP;	If the called customer has subscribed to the CND or other equivalent service, the VLC User Network MAY map the From: header in the sent INVITE (which may have been derived from an equivalent CLI parameter of another signalling system e.g. UK ISUP according to

	subject to any processing by the VLC User Network. From: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be a PN if there is one else the NN. P-Asserted-Identity: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be the NN. P-Charging-Vector: - Contents as in the received INVITE or if call is received in another signalling system generated in accordance with RFC 3455 [10] and ND 1615 [16] Require: 100rel Supported: preconditions SDP Offer and may include: Alert-Info: <data:,RCxx> “X-Display-Data-Block”	ND 1020 etc) to the “Calling Line Directory Number” parameter of the “Display Data Block” (application/X-Display-Data-Block) The mapping is described in section F.1 of this document.
4	VLC Provider network applies correct Dial Tone (Ordinary Dial Tone or Message Waiting Dial Tone)	
5	486 Busy Here to reject incoming INVITE (3)	
6	Calling customer sends the first digit (DTMF or Loop Disconnect). VLC Provider network detects digit and removes Dial Tone	
7	ACK	
8	Dial Tone removed	
9	Outgoing Call setup continues as in E.2.1/E.2.5 from flow 5 to 6.	
10	INVITE (as in E.2.1/E.2.5 flow 7). The VLC User Network marks the line as “busy on an outgoing call”. (This status will be maintained until the calling customer goes On Hook and a BYE is received by the VLC User Network - see E.2.8)	
11	100 Trying or 484 Address Incomplete depending on whether the minimum number of digits to route has been received.	
12	Outgoing Call setup continues as in E.2.1 (from flow 9) if the message at flow 14 was “484” OR continues as in E.2.5 (from flow 9) if message at flow 14 was “100”.	

E.2.4 Incoming Call (Z interface busy on incoming or outgoing call) Simplified Call Waiting Service

(The call flows in E.2.4 replace Figure C.2 in TS 183 043 [1])

E.2.4.1 Customer Presses Recall

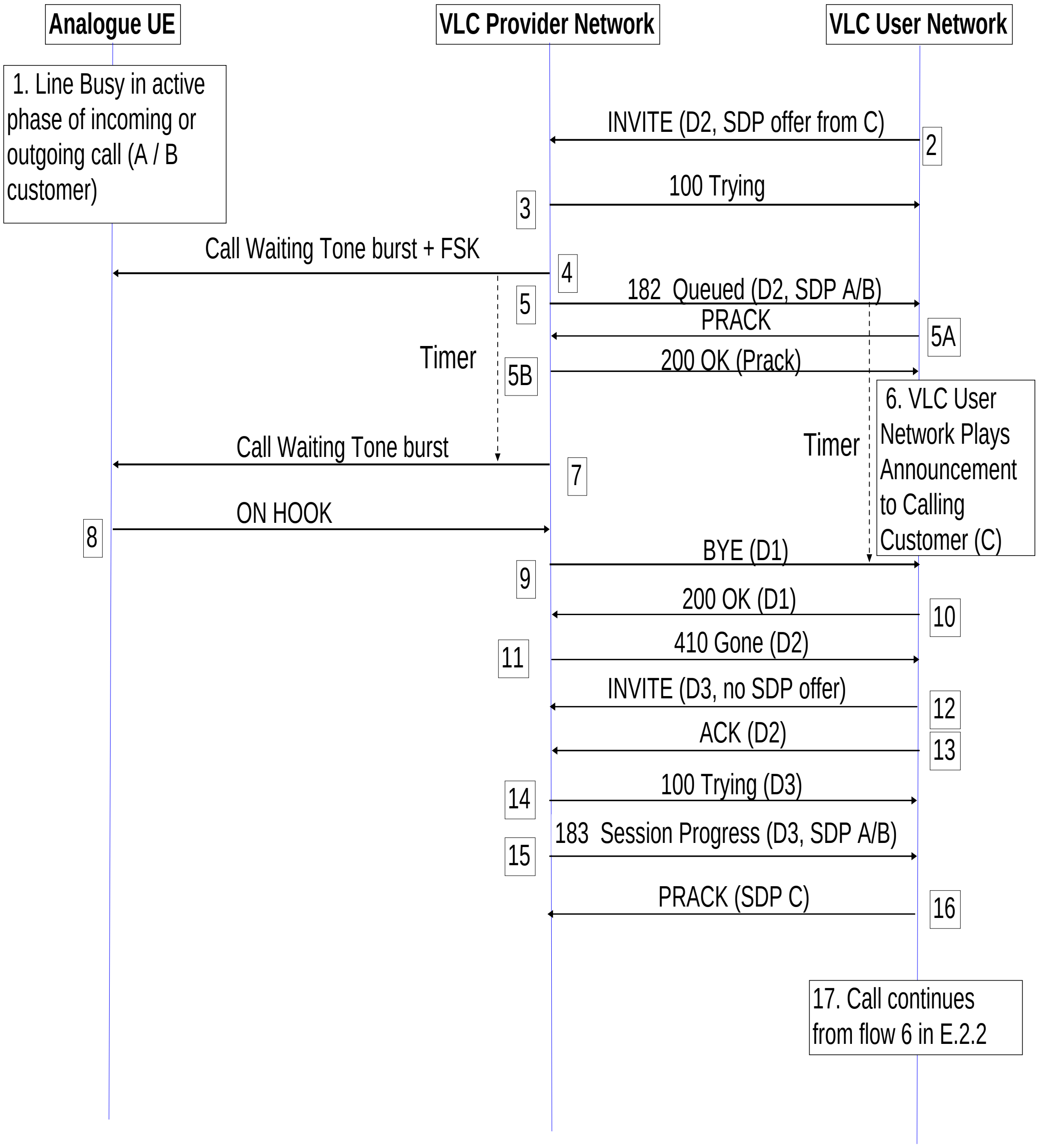


Note: "Preconditions" which may optionally be included are not shown on this Call Flow

Flow Number	Action	Additional Comments
1	The Line is busy in active phase of an incoming or outgoing call	
2	INVITE (D2) sent which includes the following: Request-URI – contains the VLC_Line_ID as defined in ND1620 [9]. Note that the VLC_Line_ID is equivalent to the IMS Public Identity. To: – contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. From: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be a PN if there is one else the NN. P-Asserted-Identity: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be the NN. P-Charging-Vector: - Contents as in the received INVITE or if call is received in another signalling system generated in accordance with RFC 3455 [10] and ND 1615 [16] Require: 100rel Supported: preconditions Alert-Info: <data:,CWTxx> SDP Offer and may include: "X-Display-Data-Block"	If the called customer has subscribed to the CND or other display service, the VLC User Network MAY map the From: header in the sent INVITE (which may have been derived from an equivalent CLI parameter of another signalling system e.g. UK ISUP according to ND 1020 etc) to the "Calling Line Directory Number" parameter of the "Display Data Block" (application/X-Display-Data-Block) The mapping is described in section F.1 of this document.
3	100 Trying	
4	Call waiting tone burst + FSK (derived from "X-Display-Data-Block") sent from VLC Provider Network to Analogue UE. The VLC Provider Network starts a TIMER with a value of the required interval between Call Waiting tone bursts (derived from the Alerting Cadence Indicator in the Alert-Info header).	
5	182 Queued VLC User Network also runs an overall CW active timer. (This timer SHALL be greater than the time taken for the maximum number of CWT cycles).	If this overall CW active timer expires or the VLC User Network abandons the Call Waiting attempt the VLC User Network SHALL send a CANCEL (D2) to the VLC Provider Network
5A	PRACK	
5B	200 OK (Prack)	

6	On receipt of 182 Queued (at (5)) the VLC User Network may play an Announcement to the calling customer.	
7	When the TIMER (started at [4]) expires a Call Waiting tone burst is sent from the VLC Provider Network to the Analogue UE. The VLC Provider Network re-starts the TIMER.	
8	Called customer uses RECALL to accept the waiting call. (The TIMER re-started at [7] is cancelled).	
9	INVITE flash@domain(D4)	The VLC User Network cancels the overall CW active timer.
10	200 OK (Invite) (D4) It is possible that a 416 Unsupported URI Scheme or other failure response may be returned. The action taken by the VLC Provider Network SHALL be the same as if the RECALL had not been received. This action is rather than applying a tone or announcement as shown in the mapping table in section E.1.1.	If the Service required is “Traditional” Call Waiting a 484 Address Incomplete SHALL be sent to the VLC Provider Network (instead of the 200 OK). This SHALL result in Dial Tone being connected to the Analogue line and further actions are similar to that shown in E.2.6 from (31).
11	ACK (D4)	
12	VLC User Network re-arranges the bearers so that the Called User (A/B) is connected to the Calling User (C) (flows (15) to (17)) and the other User (B/A) is connected to a “Held Announcement”.	
13	200 OK (Invite) (D2)	
14	ACK (D2)	
15	Re-INVITE to the called customer (A/B) on dialogue D2 (no SDP)	
16	200 OK (Invite) (SDP A/B)	
17	ACK (SDP C)	
18	BYE (D4)	VLC Provider Network re-arms the A-MGW for Recall detection. Note: On receipt of another RECALL the sequence from (9) would be followed except that the A & B customers are reconnected and the C customer is connected to a “Held Announcement”.
19	200 OK (Bye)	

E.2.4.2 Called Customer goes ON HOOK

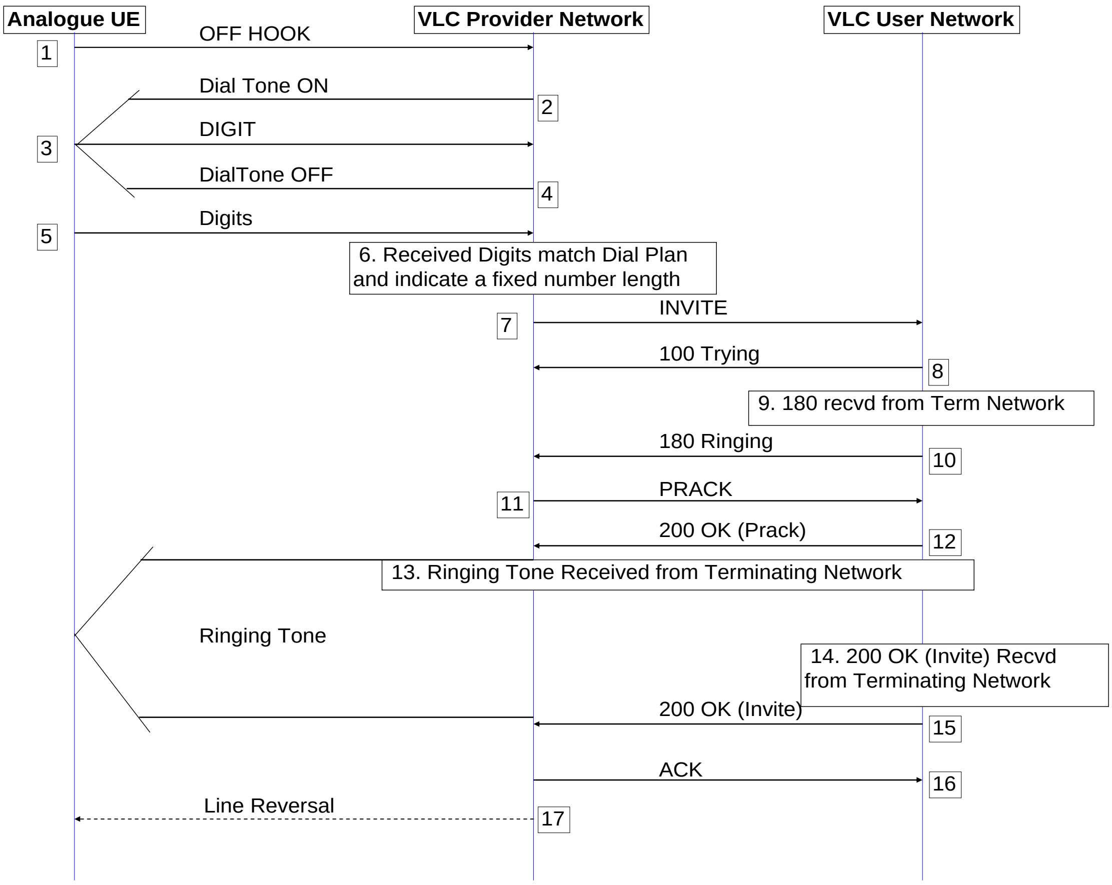


Note: "Preconditions" which may optionally be included are not shown on this Call Flow

Flow Number	Action	Additional Comments
1	The Line is busy in active phase of an incoming or outgoing call	
2	INVITE (D2) sent which includes the following: Request-URI – contains the VLC_Line_ID as defined in ND1620 [9]. Note that the VLC_Line_ID is equivalent to the IMS Public Identity. To: – contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. From: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be a PN if there is one else the NN. P-Asserted-Identity: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be the NN. P-Charging-Vector: - Contents as in the received INVITE or if call is received in another signalling system generated in accordance with RFC 3455 [10] and ND 1615 [16] Require: 100rel Supported: preconditions Alert-Info: <data:,CWTxx> SDP Offer and may include: “X-Display-Data-Block”	If the called customer has subscribed to the CND or other display service, the VLC User Network MAY map the From: header in the sent INVITE (which may have been derived from an equivalent CLI parameter of another signalling system e.g. UK ISUP according to ND 1020 etc) to the “Calling Line Directory Number” parameter of the “Display Data Block” (application/X-Display-Data-Block) The mapping is described in section F.1 of this document.
3	100 Trying	
4	Call waiting tone burst + FSK (derived from “X-Display-Data-Block”) sent from VLC Provider Network to Analogue UE. The VLC Provider Network starts a TIMER with a value of the required interval between Call Waiting tone bursts (derived from the Alerting Cadence Indicator in the Alert-Info header).	
5	182 Queued VLC User Network also runs an overall CW active timer. (This timer SHALL be greater than the time taken for the maximum number of CWT cycles).	If this overall CW active timer expires the VLC User Network SHALL send a CANCEL (D2) to the VLC Provider Network
5A	PRACK	

5B	200 OK (Prack)	
6	On receipt of 182 Queued (at (5)) the VLC User Network Plays an Announcement to the calling customer.	
7	When the TIMER (started at flow 4) expires a Call Waiting tone burst is sent from the VLC Provider Network to the Analogue UE. The VLC Provider Network re-starts the TIMER.	
8	Called customer accepts new call by going ON HOOK (The TIMER re-started at flow 7 is cancelled).	VLC Provider Network SHALL take no action on receipt of any subsequent OFF HOOKs and ON HOOKs from the calling customer until flow 13 below, but shall maintain a record of the current HOOK status. The VLC Provider Network implicitly enters the “access resources held” condition as a result of its procedures for handling the CW supplementary service.
9	BYE (D1)	The VLC User Network cancels the overall CW active timer.
10	200 OK (Bye) (D1)	
11	410 Gone (D2)	
12	INVITE(D3) The INVITE may also include an X-service-indicator header with a service-identifier parameter value of "use-held-resource".	VLC User Network removes the announcement from the Calling Party (C)
13	ACK (D2)	On receipt of the ACK the VLC Provider Network shall act according to the current HOOK status of the line (i.e. if the line is ON HOOK take no action, but if the line is OFF HOOK it shall send a 200 OK (Invite) (D3) to the VLC User Network, containing SDP A/B.)
14	100 Trying	
15	183 Session Progress (D3, SDP A/B)	The VLC User Network SHOULD send a Re-INVITE or UPDATE towards the originating network containing SDP A/B to obtain an SDP answer containing SDP C. This SDP is used to populate the PRACK in step 16 below.
16	PRACK (SDP C)	
17	Call Flow continues from flow (6) in E.2.2	

E.2.5 Outgoing Call Attempt (Z interface idle) – En-block digit sending



Note: "Preconditions" which optionally may be included are not shown on this Call Flow

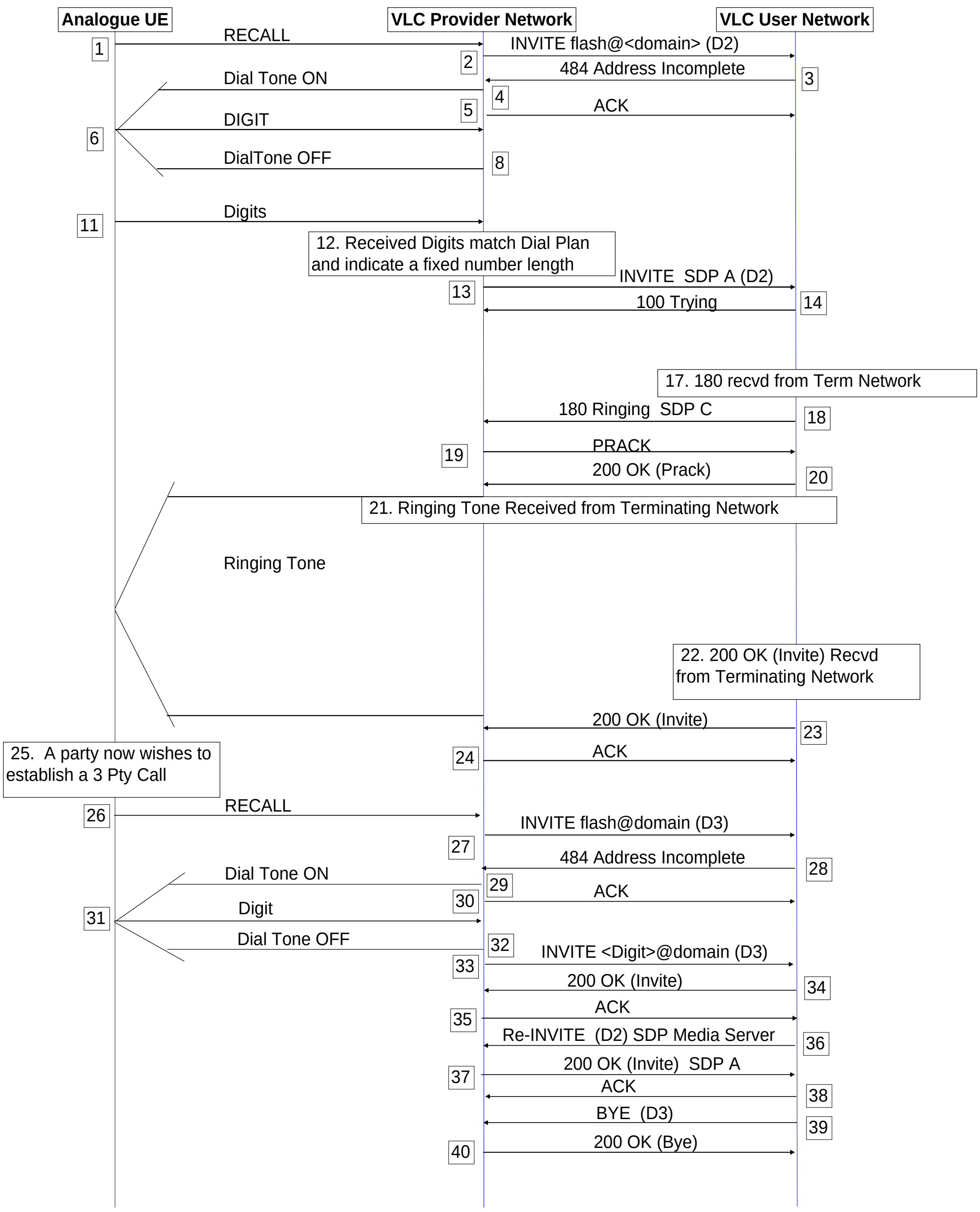
Flow Number	Action	Additional Comments
1	VLC Provider network identifies correct profile for this line and applies correct digit map (including timers). Note: An OFF HOOK is either: • A Loop (for a DEL or Loop Calling PBX); or • An earth on the B leg (for an Earth calling PBX)	Once the OFF HOOK is detected any subsequent INVITE received by the VLC Provider Network is rejected with a 486 Busy Here.
2	VLC Provider network applies correct Dial Tone (Ordinary Dial Tone or Message Waiting Dial Tone)	
3	Calling customer sends the first digit (DTMF or Loop Disconnect). VLC Provider network detects digit and removes Dial Tone	
4	Dial Tone removed	
5	VLC Provider Network starts digit analysis against applied dial plan	

6	The received digits match the dial plan and indicate a fixed number length. Note: Any further digits received from the calling customer shall be discarded.	The VLC Provider Network checks to see if there is any available bandwidth for the call. If there is no available bandwidth to the VLC User Network for the type of call (ordinary / priority) the VLC Provider Network SHALL connect the caller to the All Lines Busy announcement.
7	The VLC Provider Network sends an INVITE to the VLC User Network including the following contents: Request-URI – contains digits received from calling customer (sip:<digits>@<domain> - see ND 1620 [9]) To: – contains the same as the Request-URI From: - contains the VLC_Line_ID of calling line Route: - contains the same as the Service-Route header in the 200 OK response to the associated previous REGISTER message. P-Asserted-Identity: - contains the VLC_Line_ID of calling line and the cpc parameter set in accordance with ND 1019 [7] sections 3.1 & 4.1. P-Charging-Vector: - Contents in accordance with RFC 3455 [10] and ND 1615 [16]. Require: 100rel Supported: preconditions SDP Offer The VLC User Network marks the line as “busy on an outgoing call”. (This status will be maintained until the calling customer goes On Hook and a BYE is received by the VLC User Network - see E.2.8)	For lines with PATS the following rules apply: If there is a PN associated with the calling line the VLC USER Network SHALL remove the URI from the From header and replace it with the PN. Otherwise it SHALL remove the URI from the From header and replace it with the NN before forwarding the INVITE. The PN or NN shall be a Tel or SIP URI containing an E.164 number beginning with “+44”. The VLC User Network SHALL also modify the P-Asserted-Identity header so that it contains the NN (Tel or SIP URI containing an E.164 number prefixed with “+44”) before forwarding the INVITE towards the terminating network. The VLC User Network shall, when appropriate, add the SIP Privacy header (or equivalent CLI restricted/withheld indication if interworking directly to another signalling system). The addition of the Privacy header (with priv-value = "id") or equivalent indication should be determined by a combination of receipt of a prefix (e.g. 141 or 1470) and the subscribed privacy service for the calling line. For definitions of PN & NN see ND 1016 [4]
8	VLC User Network sends 100 Trying	
9	When the VLC User Network receives a 180 RINGING from the terminating network it SHALL arrange for the bearer to be connected to a suitable audible tone and SHALL include a P-Early-Media header with parameter "sendrecv" or "sendonly" in the 180 RINGING sent to the VLC Provider Network in step 10. The audible tone may be generated at the VLC User Network or switched through from the far end network. The latter would normally be indicated by the presence of a suitable P-Early-Media header in the 180 RINGING	

	from the terminating network.	
10	A 180 Ringing which shall include a P-Early_Media header (with SDP Answer) is sent to the VLC Provider Network. The VLC Provider Network SHALL switch-through the forward and backward bearer paths and provide an o/g half ECD if not already done, and start a timer awaiting receipt of 200 OK (Invite).	If the timer expires the VLC Provider Network SHALL play the “No Reply” announcement.
11	PRACK	
12	200 OK (Prack)	
13	Ringing Tone (either from the terminating network or from the VLC User Network) is sent to the calling customer. This tone is removed when the called customer answers (by the terminating network or the VLC User Network as appropriate)	
14	When the VLC User Network receives a 200 OK (Invite) (or other Answer indication) from the terminating network it sends a 200 OK (Invite) to the VLC Provider network. If the VLC User Network is providing Ringing Tone it SHALL now re-arrange the bearers to connect the calling customer through to the called customer. This is achieved by sending a Re-INVITE (with no SDP) to the VLC Provider Network.	
15	On receipt of 200 OK (Invite) the VLC Provider Network MAY, depending on the service mark for the line, apply a line reversal on the Z interface.	
16	ACK	
17	Optional Line Reversal	

E.2.6 Enquiry Call (A call in the active conversation phase exists between the A & B parties)

The call flow in E.2.6 replaces Figure C.3 in TS 183 043



Note: "Preconditions" which optionally may be included are not shown on this Call Flow

Flow Number	Action	Additional Comments
1	On receipt of RECALL the VLC Provider network identifies the correct profile for this line and applies a digit map to collect the digits (including timers)	
2	INVITE used to notify VLC User Network that customer has pressed Recall. The contents of the INVITE include the following: Request-URI – sip:flash@<domain> To: – contains the same as the Request-URI From: - contains the VLC_Line_ID of calling line Route: - contains the same as the Service-Route header in the 200 OK response to the associated previous REGISTER message. P-Asserted-Identity: - contains the VLC_Line_ID of calling line and the cpc parameter set in accordance with ND 1019 [7] sections 3.1 & 4.1. P-Charging-Vector: - Contents in accordance with RFC 3455 [10] and ND 1615 [16]. Require: 100rel Supported: preconditions SDP Offer The VLC User Network marks the line as "busy on an enquiry call".	When the INVITE is received by the VLC User Network it puts the B party on hold, and MAY connect them to a recorded announcement.
3	484 indicating insufficient (no) digits were received	It is possible that a 404 not found or other failure response may be returned. The action taken by the VLC Provider Network SHALL be the same as if the RECALL had not been received. This action is rather than applying a tone or announcement as shown in the mapping table in section E.1.1. Note: If there is a requirement not to return Dial Tone (e.g. for a fixed destination call) at this point, a 200 OK (Invite) SHALL be sent.
4	VLC Provider network applies correct Dial Tone (Ordinary Dial Tone or Message Waiting Dial Tone) At this point the VLC Provider Network will re-arm the A-MGW for detection of RECALL.	
5	ACK	
6	Calling customer sends the first digit (DTMF or Loop Disconnect). VLC Provider network detects digit and removes Dial Tone	

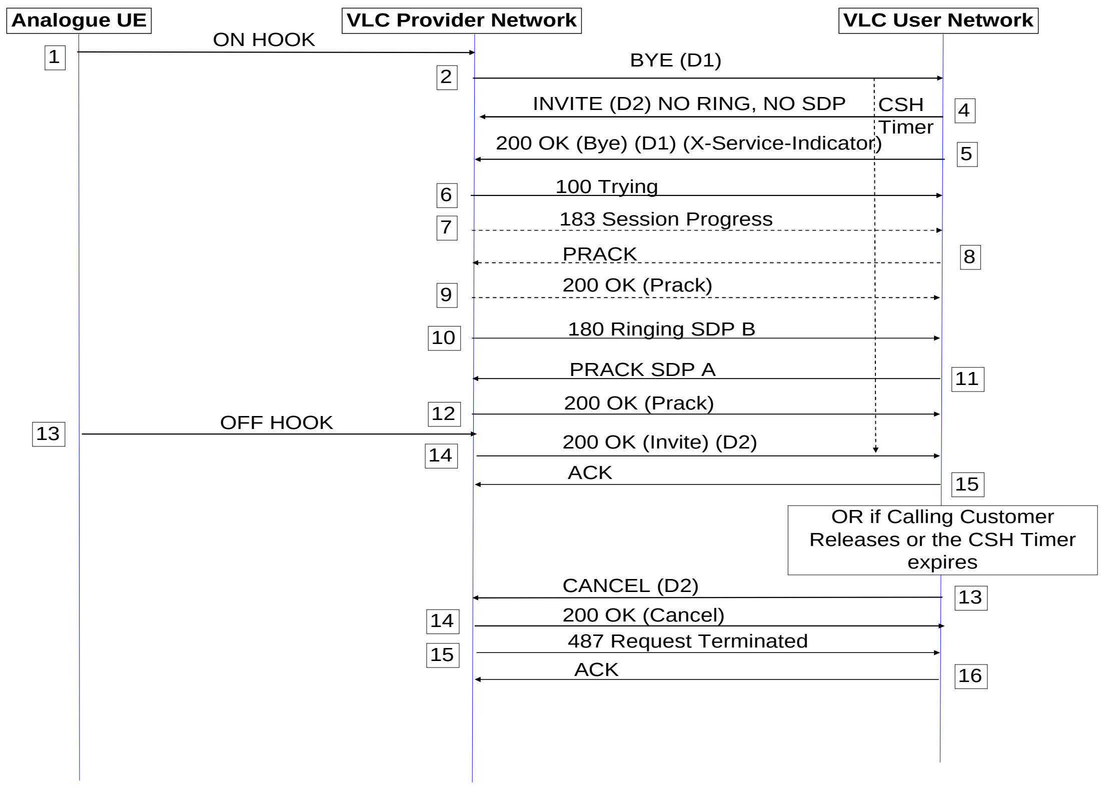
7	Not used	
8	Dial Tone removed	
9	Not used	
10	Not used	
11	VLC Provider Network starts digit analysis against applied dial plan. Note: This action is taken because the AGCF is aware that there was only one active dialogue when the RECALL was received.	
12	The received digits match the dial plan and indicate a fixed number length.	
13	The VLC Provider Network sends an INVITE to the VLC User Network including the following contents: Request-URI – contains digits received from calling customer (sip:<digits>@<domain> - see ND 1620 [9]) To: – contains the same as the Request-URI From: - contains the VLC_Line_ID of calling line P-Asserted-Identity: - contains the VLC_Line_ID of calling line and the cpc parameter set in accordance with ND 1019 [7] sections 3.1 & 4.1. P-Charging-Vector: - Contents in accordance with RFC 3455 [10] and ND 1615 [16]. Require: 100rel Supported: preconditions SDP Offer	For lines with PATS the following rules apply: If there is a PN associated with the calling line the VLC USER Network SHALL remove the URI from the From header and replace it with the PN. Otherwise it SHALL remove the URI from the From header and replace it with the NN before forwarding the INVITE. The PN or NN shall be a Tel or SIP URI containing an E.164 number beginning with “+44”. The VLC User Network SHALL also modify the P-Asserted-Identity header so that it contains the NN (Tel or SIP URI containing an E.164 number prefixed with “+44”) before forwarding the INVITE towards the terminating network. The VLC User Network shall, when appropriate, add the SIP Privacy header (or equivalent CLI restricted/withheld indication if interworking directly to another signalling system). The addition of the Privacy header (with priv-value = "id") or equivalent indication should be determined by a combination of receipt of a prefix (e.g. 141 or 1470) and the subscribed privacy service for the calling line. For definitions of PN & NN see ND 1016 [4]
14	VLC User Network sends 100 Trying	
15	Not used	
16	Not used	
17	When the VLC User Network receives a 180 RINGING	

	from the terminating network it SHALL arrange for the bearer to be connected to a suitable audible tone and SHALL include a P-Early-Media header with parameter "sendrecv" or "sendonly" in the 180 RINGING sent to the VLC Provider Network in step 18. The audible tone may be generated at the VLC User Network or switched through from the far end network. The latter would normally be indicated by the presence of a suitable P-Early-Media header in the 180 RINGING from the terminating network.	
18	A 180 Ringing which shall include a P-Early_Media header (with SDP Answer) is sent to the VLC Provider Network. The VLC Provider Network SHALL switch-through the forward and backward bearer paths and provide an o/g half ECD if not already done, and start a timer awaiting receipt of 200 OK (Invite)..	If the timer expires the VLC Provider Network SHALL play the “No Reply” announcement.
19	PRACK	
20	200 OK (Prack)	
21	Ringing Tone (either from the terminating network or from the VLC User Network) is sent to the calling customer. This tone is removed when the called customer answers (by the terminating network or the VLC User Network as appropriate)	
22	When the VLC User Network receives a 200 OK (Invite) (or other Answer indication) from the terminating network it sends a 200 OK (Invite) to the VLC Provider network. If the VLC User Network is providing Ringing Tone it SHALL now re-arrange the bearers to connect the calling customer through to the called customer. This is achieved by sending a Re-INVITE (with no SDP) to the VLC Provider Network.	
23	200 OK (Invite)	
24	ACK	
25	The customer now decides to make the call into a 3 Party Call. Note: this call flow assumes the method of RECALL, Dial Tone & SOC.	
26	Customer presses RECALL	
27	The VLC Provider Network sends an INVITE “flash” (D3) to the VLC User Network	
28	484 Address Incomplete	Alternatively the VLC User Network MAY send a 200 OK because the “flash” is sufficient to indicate the service action required.
29	The VLC Provider Network applies Dial Tone (as a result of receiving 484)	
30	ACK	
31	Customer sends a single digit	

32	VLC Provider Network removes Dial Tone	
33	VLC Provider Network sends an INVITE (D3) to the VLC User Network containing the digit dialled. Note: this action is taken because the AGCF is aware that there was more than one active dialogue when it received the RECALL.	
34	200 OK (Invite)	
35	ACK	
36	Re-Invite (D2) with the SDP of the Media Server providing the 3 Party Bridge.	
37	200 OK (Invite) with the SDP of customer A’s Access Media Gateway	
38	ACK	
39	BYE (D3)	
40	200 OK (Bye)	

E.2.7 Call Release from Incoming Call

E.2.7.1 Called Party goes ON HOOK (Called Subscriber Held)

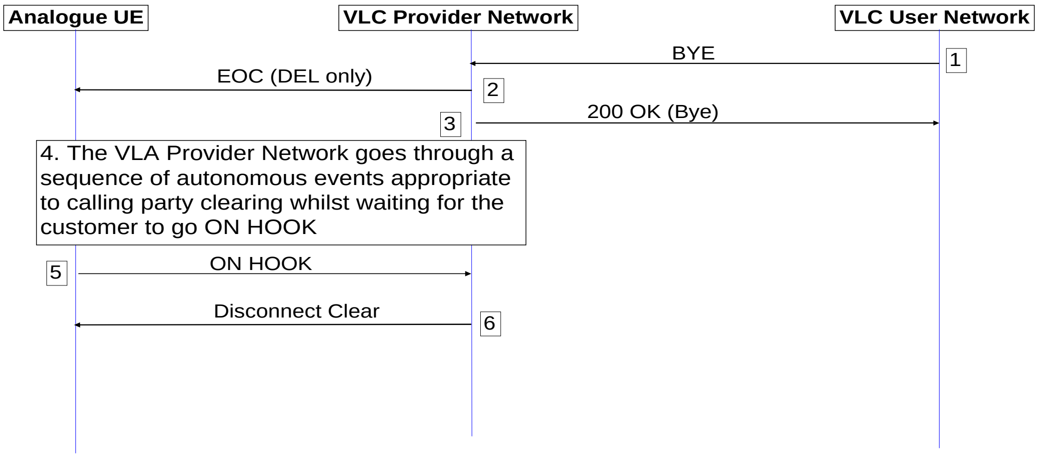


Flow Number	Action	Additional Comments
1	Called customer goes ON HOOK	VLC Provider Network SHALL take no action on receipt of any subsequent OFF HOOKs and ON HOOKs from the calling customer until flow (5) below, but shall maintain a record of the current HOOK status.
2	VLC Provider Network sends BYE to VLC User Network. NOTE 1: This is required so that first party release can be implemented by the VLC User Network if it chooses to do so – in which case it would not start the CSH timer or send the INVITE (NO RING) to the VLC Provider Network. NOTE 2: It is recognised that there is a slim chance that during busy periods there may be no spare bandwidth to allow the connection to be re-established. For Non-PBX lines the VLC User Network starts a Called Subscriber Held timer waiting for the customer to go OFF HOOK.	If the timer expires the VLC User Network SHALL send a BYE or other equivalent Release message towards the originating network, continue from OR (13) below and mark the line as free.

	For PBX Lines the VLC User Network SHALL send a BYE or other equivalent Release message towards the originating network and mark the line as free.	
3	Not used	
4	For lines that are not first party clear the VLC User Network sends an INVITE (D2) with an Alert Info header indicating NO RING CALL (RC07). The VLC Provider Network MAY start an “access held” guard timer (default 10 minutes) which would be cancelled on sending of a 200 OK (Invite) to the VLC User Network. The INVITE may also include an X-service-indicator header with a service-identifier parameter value of "use-held-resource".	If the “access held” guard timer expires the VLC Provider Network would send a 408 response to the VLC User Network. If the optional X-service-indicator functionality is supported and used here then it SHALL be used also in step 5 and vice versa
5	200 OK (Bye) may include an X-service-indicator header with a service-identifier header set to "hold-resource". (see F.5 for the syntax) The X-service-indicator header instructs the VLC Provider Network not to release the Access should 200 OK (Bye) be received before the Invite at flow 4. If the 200 OK (Bye) is received before the Invite at flow 4 the VLC Provider Network will run a short timer awaiting the Invite.	On receipt of the 200 OK (Bye) the VLC Provider Network shall act according to the current HOOK status of the line (i.e. if the line is ON HOOK take no action, but if the line is OFF HOOK it shall send a 200 OK (Invite) to the VLC User Network, containing SDP B and continue from step 15 below)
6	The VLC Provider Network SHALL send a 100 Trying if the autonomous clearing phase is still in progress, otherwise it moves straight to (7) below.	
7	Optional 183	
8	PRACK (optional)	
9	200 OK (Prack) (optional)	
10	The VLC Provider Network sends a 180 Ringing (containing SDP B) to the VLC User Network.	The VLC User Network SHOULD send a Re-INVITE or UPDATE towards the originating network containing SDP B to obtain an SDP answer containing SDP A. This SDP is used to populate the PRACK in step 11 below.
11	PRACK (containing SDP A)	
12	200 OK (Prack)	
13	OFF HOOK – Called Customer Re-Answers	
14	The VLC Provider Network sends a 200 OK (Invite) to the VLC User Network. The VLC User Network cancels the CSH timer.	
15	ACK	Conversation phase of original call now resumes.
OR 13	If the calling customer clears the call or the CSH Timer expires the VLC User Network SHALL send a CANCEL to the VLC Provider Network.	

14	200 OK (Cancel)	
15	487 Request Terminated	
16	ACK	

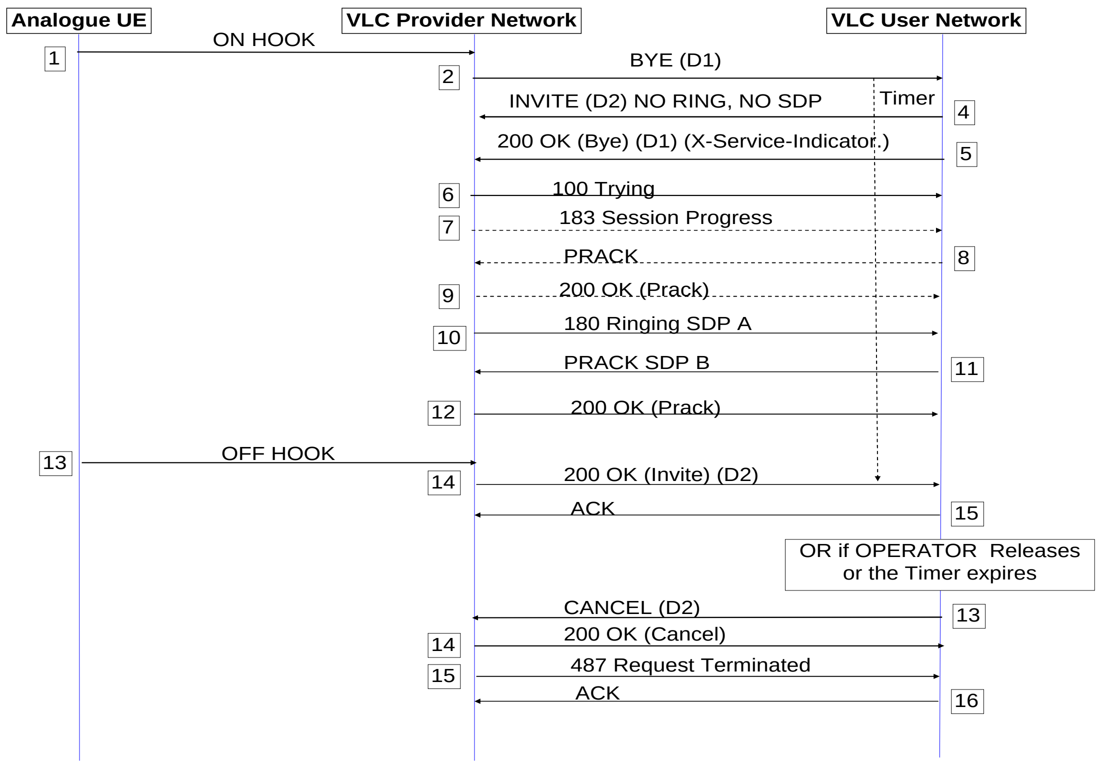
E.2.7.2 Calling Party Clears first



Flow Number	Action	Additional Comments
1	The VLC User Network sends a BYE to the VLC Provider Network	
2	If the line is a DEL the VLC Provider Network SHALL send an End of Call (EOC) indication to the Analogue UE.	
3	200 OK (Bye)	
4	The VLC Provider Network goes through a sequence of autonomous events appropriate to calling party clearing e.g. “Please hang up” announcement for a certain time, followed by Parked Line feed for a certain time, followed by Howler with Normal or Reverse line feed as appropriate for a certain time, and finally back to the Parked line feed. This sequence can be interrupted at any point by the called party going ON HOOK.	
5	Called party goes ON HOOK.	
6	The VLC Provider Network sends a Disconnect Clear indication to the Analogue UE.	

E.2.8 Call release from Outgoing Call

E.2.8.1 Calling Party goes ON HOOK (Call to Operator – calling subscriber held)

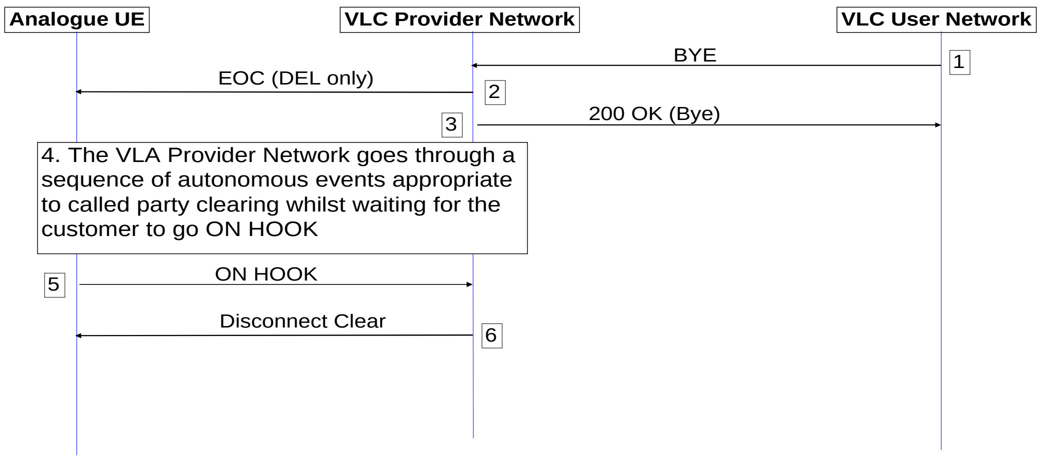


Flow Number	Action	Additional Comments
1	Calling customer goes ON HOOK	VLC Provider Network SHALL take no action on receipt of any subsequent OFF HOOKs and ON HOOKs from the calling customer until flow (5) below, but shall maintain a record of the current HOOK status.
2	VLC Provider Network sends BYE to VLC User Network. NOTE 1: This is required so that first party release can be implemented by the VLC User Network if it chooses to do so – in which case it would not start the timer or send the INVITE (NO RING) to the VLC Provider Network. NOTE 2: It is recognised that there is a slim chance that during busy periods there may be no spare bandwidth to allow the connection to be re-established. VLC User Network either: • If the call is not to the Operator (e.g. not an emergency or other assistance call) or is from an Earth or Loop Calling PBX it SHALL send a BYE or equivalent Release message towards the	

	terminating network and marks the line as free; • OR if the call is to the Operator from a DEL it SHALL start a 5 minute duration CSH timer awaiting the calling customer to go OFF HOOK.	If the timer expires the VLC User Network SHALL send a BYE or other equivalent Release message towards the terminating network and continue from “OR 13” below.
3	Not used	
4	For calls to the Operator from lines that are not first party clear (e.g. DEL’s) the VLC User Network sends an INVITE (D2) with an Alert Info header indicating NO RING CALL (RC07). The VLC Provider Network MAY start an “access held” guard timer (default 10 minutes) which would be cancelled on sending of a 200 OK (Invite) to the VLC User Network. The INVITE may also include an X-service-indicator header with a service-identifier parameter value of "use-held-resource".	If the “access held” guard timer expires the VLC Provider Network would send a 408 response to the VLC User Network. If the optional X-service-indicator functionality is supported and used here then it SHALL be used also in step 5 and vice versa
5	200 OK (Bye) may include an X-service-indicator header with a service-identifier header set to "hold-resource". (see F.5 for the syntax) The X-service-indicator header instructs the VLC Provider Network not to release the Access should 200 OK (Bye) be received before the Invite at flow 4. If the 200 OK (Bye) is received before the Invite at flow 4 the VLC Provider Network will run a short timer awaiting the Invite.	On receipt of the 200 OK (Bye) the VLC Provider Network shall act according to the current HOOK status of the line (i.e. if the line is ON HOOK take no action, but if the line is OFF HOOK it shall send a 200 OK (Invite) to the VLC User Network, containing SDP A and continue from step 15 below.)
6	The VLC Provider Network SHALL send a 100 Trying if the autonomous clearing phase is still in progress, otherwise it moves straight to (7) below.	
7	Optional 183	
8	PRACK (optional)	
9	200 OK (Prack) (optional)	
10	The VLC Provider Network sends a 180 Ringing containing SDP A to the VLC User Network.	The VLC User Network SHOULD send a Re-INVITE or UPDATE towards the terminating network containing SDP A to obtain an SDP answer containing SDP B. This SDP is used to populate the PRACK in step 11 below.
11	PRACK (containing SDP B)	
12	200 OK (Prack)	
13	Calling Customer goes OFF HOOK.	
14	VLC Provider Network sends a 200 OK (Invite) (D2) to the VLC User Network The VLC User Network cancels the 5 minute duration CSH timer.	

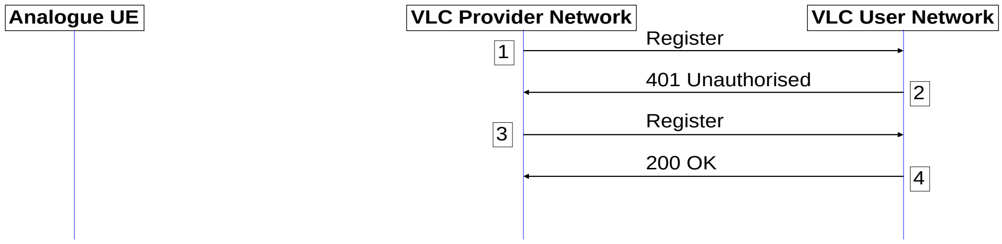
15	ACK	Conversation phase of original call now resumes.
OR 13	If the Operator clears the call, or the Timer expires, the VLC User Network SHALL send a CANCEL (D2) to the VLC Provider Network.	
14	200 OK (Cancel)	
15	487 Request Terminated	
16	ACK	

E.2.8.2 Called Party Clears First



Flow Number	Action	Additional Comments
1	The VLC User Network sends a BYE to the VLC Provider Network	
2	If the line is a DEL the VLC Provider Network SHALL send an End of Call (EOC) indication to the Analogue UE.	
3	200 OK (Bye)	
4	The VLC Provider Network shall go through a sequence of autonomous events appropriate to called party clearing e.g. “The other party has hung up” announcement or NU tone (if a call has not yet been established) for a certain time, followed by Parked Line feed for a certain time, followed by Howler with Normal or Reverse line feed as appropriate for a certain time, and finally back to the Parked line feed. This sequence can be interrupted at any point by the calling party going ON HOOK.	
5	Calling Customer goes ON HOOK.	
6	The VLC Provider Network sends a Disconnect Clear indication to the Analogue UE.	

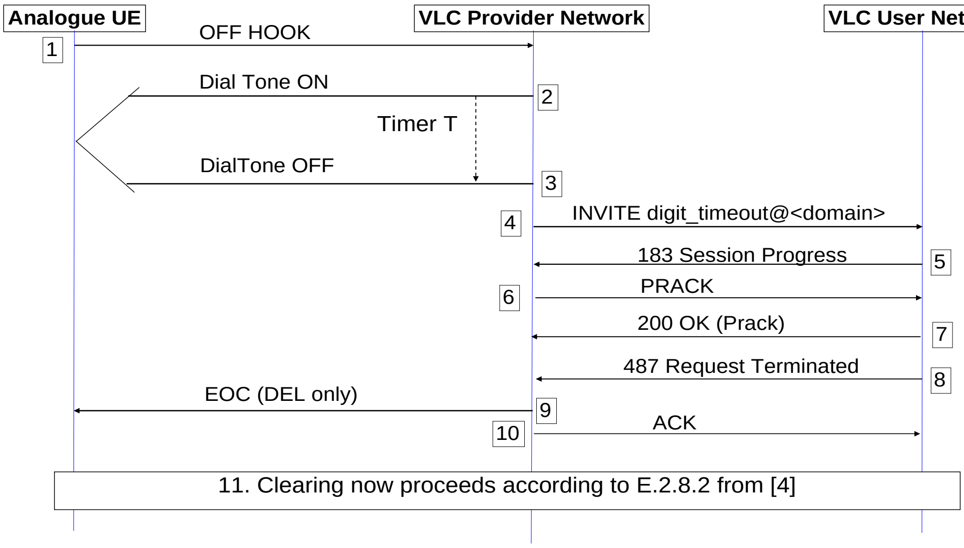
E.2.9 Registration & Re-Registrations



Flow Number	Action	Additional Comments
1	Initial REGISTER without authentication credentials – contents as defined in section 5.3.2.3 of TS 183 043 [1] as modified by section 2.1 of this document. Note: Registration is carried out by the VLC Provider Network when: • It receives a new registration from an A-MGW (e.g. Service Change); • The existing signalling path between the AGCF and S-CSCF is lost (as determined by a mixture of failure to get a response to INVITE or OPTION messages); • The “expires” after period has been reached.	Implicit (Group) Registrations SHALL be used whenever there is a large* number of lines on an A-MGW being controlled by the same VLC User Network. When the number of lines on an A-MGW being controlled by the same VLC User Network is below a certain threshold* Explicit (single line) registration MAY be used. * The threshold above which large applies is agreed bilaterally.
2	401 Unauthorised containing the “nonce” in a WWW-Authenticate header.	
3	REGISTER with authentication credentials – contents the same as [1] above except for the new authentication credentials	Implicit (Group) Registrations SHALL be used whenever there is a large* number of lines on an A-MGW being controlled by the same VLC User Network. When the number of lines on an A-MGW being controlled by the same VLC User Network is below a certain threshold* Explicit (single line) registration MAY be used. * The threshold above which large applies is agreed bilaterally.
4	200 OK – This will include a list of all lines which have been registered as a result of the implicit REGISTER message in the P-Associated-URI header.	Note: No action is taken by the VLC Provider Network on the contents of the P-Associated-URI header.

E.2.10 Digit Collection Timeout

E.2.10.1 No Digits Dialled

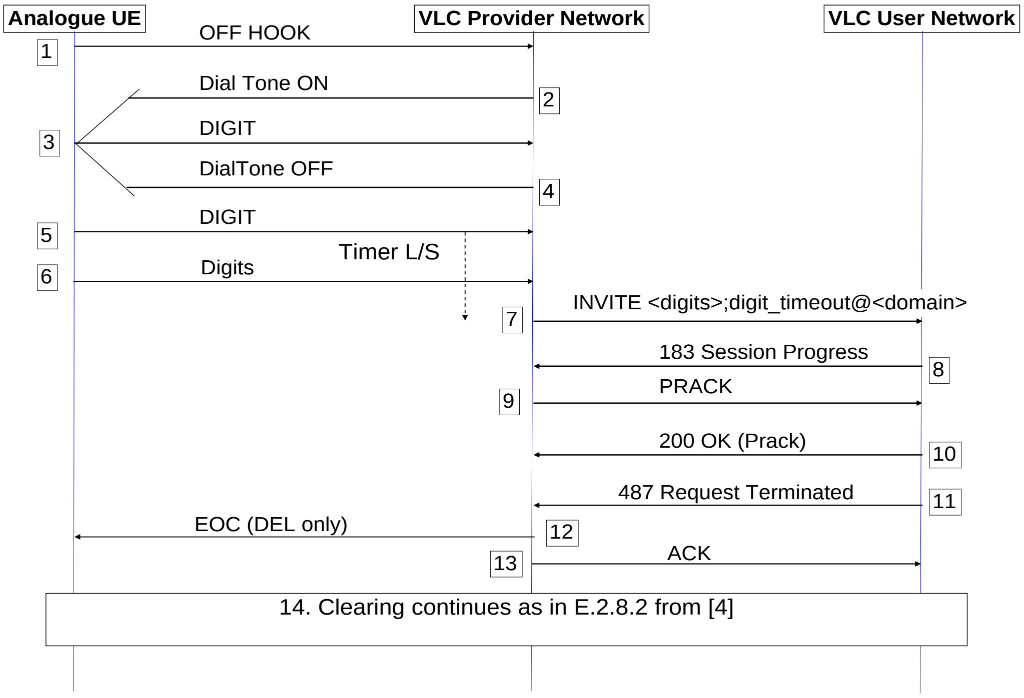


Note: "Preconditions" which optionally may be included are not shown on this Call Flow

Flow Number	**Action**	**Additional Comments**
1	VLC Provider network identifies correct profile for this line and applies correct digit map (including timers). Note: An OFF HOOK is either: • A Loop (for a DEL or Loop Calling PBX); or • An earth on the B leg (for an Earth calling PBX)	Once the OFF HOOK is detected any subsequent INVITE received by the VLC Provider Network is rejected with a 486 Busy Here.
2	VLC Provider network applies correct Dial Tone (Ordinary Dial Tone or Message Waiting Dial Tone)	
3	Initial digit timer T expires and Dial Tone is removed	The VLC Provider Network checks to see if there is any available bandwidth for the call. If there is no available bandwidth to the VLC User Network for the type of call (ordinary / priority) the VLC Provider Network SHALL connect the caller to the No Digits announcement.
4	The VLC Provider Network sends an INVITE digit_timeout@domain with SDP A to the VLC User Network. The contents of the INVITE include the following: Request-URI – sip:digit_timeout@<domain> To: – contains the same as the Request-URI From: - contains the VLC_Line_ID of calling line Route: - contains the same as the Service-Route header in the 200 OK response to the associated previous	This allows the VLC User Network to choose whether to play a tone or an announcement to the Calling Customer.

	REGISTER message. P-Charging-Vector: - Contents in accordance with RFC 3455 [10] and ND 1615 [16]. Require: 100rel Supported: preconditions SDP Offer The VLC User Network marks the line as “busy on an outgoing call”. (This status will be maintained until the calling customer goes On Hook and a BYE is received by the VLC User Network - see E.2.8 or until the announcement finishes and a BYE is sent by the VLC User Network to the VLC Provider Network as in (8) below.	
5	The VLC User Network chooses to play an announcement to the calling customer so it sends a 183 Session Progress with SDP of the Media Server to the VLC Provider Network. Note: If the VLC User Network wants to play NU tone to the calling customer it MAY send a 487 to the VLC Provider Network. But if the VLC User Network wishes the A-MGW to play an announcement it may send a suitable failure response message (e.g. 487) with an Error-Info header indicating the particular announcement it wants (See F.4).	
6	PRACK	
7	200 OK (Prack)	
8	When the announcement has ended the VLC User Network sends a 487 to the VLC Provider Network.	
9	EOC (DEL only)	
10	ACK	
11	The clearing sequence now continues as in E.2.8.2 from flow no.4	

E.2.10.2 Insufficient digits dialled

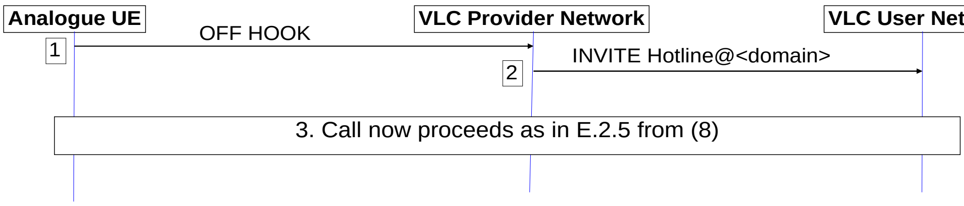


Note: "Preconditions" which optionally may be included are not shown on this Call Flow

Flow Number	Action	Additional Comments
1	VLC Provider network identifies correct profile for this line and applies correct digit map (including timers). Note: An OFF HOOK is either: • A Loop (for a DEL or Loop Calling PBX); or • An earth on the B leg (for an Earth calling PBX)	Once the OFF HOOK is detected any subsequent INVITE received by the VLC Provider Network is rejected with a 486 Busy Here.
2	VLC Provider network applies correct Dial Tone (Ordinary Dial Tone or Message Waiting Dial Tone)	
3	Calling customer sends the first digit (DTMF or Loop Disconnect). VLC Provider network detects digit and removes Dial Tone	
4	Dial Tone removed	
5	VLC Provider Network starts digit analysis against applied dial plan and starts Timer L.	
6	Further digits are received and the Timer L or S is restarted or started on receipt of each digit according to the digit map in the Access Media Gateway.	
7	Timer L or Timer S expires and the VLC Provider Network sends an INVITE with SDP A to the VLC User Network including the following contents: Request-URI – contains digits received from calling customer (sip:<digits>;digit_timeout@<domain>)	The VLC Provider Network checks to see if there is any available bandwidth for the call. If there is no available bandwidth to the VLC User Network for the type of call (ordinary / priority) the VLC Provider Network SHALL connect

	To: – contains the same as the Request-URI From: - contains the VLC_Line_ID of calling line Route: - contains the same as the Service-Route header in the 200 OK response to the associated previous REGISTER message. P-Charging-Vector: - Contents in accordance with RFC 3455 [10] and ND 1615 [16]. Require: 100rel Supported: preconditions SDP Offer The VLC User Network marks the line as "busy on an outgoing call". (This status will be maintained until the calling customer goes On Hook and a BYE is received by the VLC User Network - see E.2.8 or until the announcement finishes and a BYE is sent by the VLC User Network to the VLC Provider Network as in (11) below.)	the caller to the All Lines Busy announcement.
8	The VLC User Network chooses to play an announcement to the calling customer so it sends a 183 Session Progress with SDP of the Media Server to the VLC Provider Network. Note: If the VLC User Network wants to play NU tone to the calling customer it MAY send a 487 to the VLC Provider Network. But if the VLC User Network wishes the A-MGW to play an announcement it may send a suitable failure response message (e.g. 487) with an Error-Info header indicating the particular announcement it wants (See F.4).	
9	PRACK	
10	200 OK (Prack)	
11	When the announcement has ended the VLC User Network sends a 487 to the VLC Provider Network.	
12	EOC (DEL only)	
13	ACK	
14	The clearing sequence now continues as in E.2.8.2 from flow no.4	

E.2.11 HOTLINE

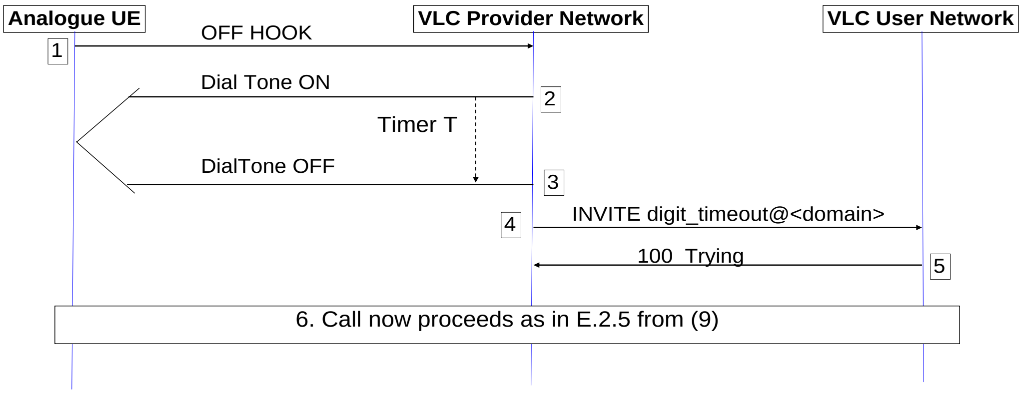


Note: "Preconditions" which optionally may be included are not shown on this Call Flow

Flow Number	Action	Additional Comments
1	VLC Provider network identifies that this line is configured for the HOTLINE service	Once the OFF HOOK is detected any subsequent INVITE received by the VLC Provider Network is rejected with a 486 Busy Here. The VLC Provider Network checks to see if there is any available bandwidth for the call. If there is no available bandwidth to the VLC User Network for the type of call (ordinary / priority) the VLC Provider Network SHALL connect the caller to the All Lines Busy announcement.
2	The VLC Provider Network sends an INVITE with SDP A to the VLC User Network including the following contents: Request-URI – “sip:Hotline@<domain>” To: – contains the same as the Request-URI From: - contains the VLC_Line_ID of calling line Route: - contains the same as the Service-Route header in the 200 OK response to the associated previous REGISTER message. P-Asserted-Identity: - contains the VLC_Line_ID of calling line and the cpc parameter set in accordance with ND 1019 [7] sections 3.1 & 4.1. P-Charging-Vector: - Contents in accordance with RFC 3455 [10] and ND 1615 [16]. Require: 100rel Supported: preconditions SDP Offer The VLC User Network marks the line as “busy on an outgoing call” (this status will be maintained until the calling customer goes On Hook and a BYE is received by the VLC User Network - see E.2.8) and proceeds to set up	For lines with PATS the following rules apply: If there is a PN associated with the calling line the VLC USER Network SHALL remove the URI from the From header and replace it with the PN. Otherwise it SHALL remove the URI from the From header and replace it with the NN before forwarding the INVITE. The PN or NN shall be a Tel or SIP URI containing an E.164 number beginning with “+44”. The VLC User Network SHALL also modify the P-Asserted-Identity header so that it contains the NN (Tel or SIP URI containing an E.164 number prefixed with “+44”) before forwarding the INVITE towards the terminating network. The VLC User Network shall, when appropriate, add the SIP Privacy header (or equivalent CLI restricted/withheld indication if interworking directly to another signalling system). The addition of the Privacy header (with priv-value =

	the call to the pre-determined destination (e.g. sends an INVITE towards the terminating network).	"id") or equivalent indication should be determined by the subscribed privacy service for the calling line. For definitions of PN & NN see ND 1016 [4]
3	The call continues as in E.2.5 from (8).	

E.2.12 WARMLINE (No Digits Dialled)



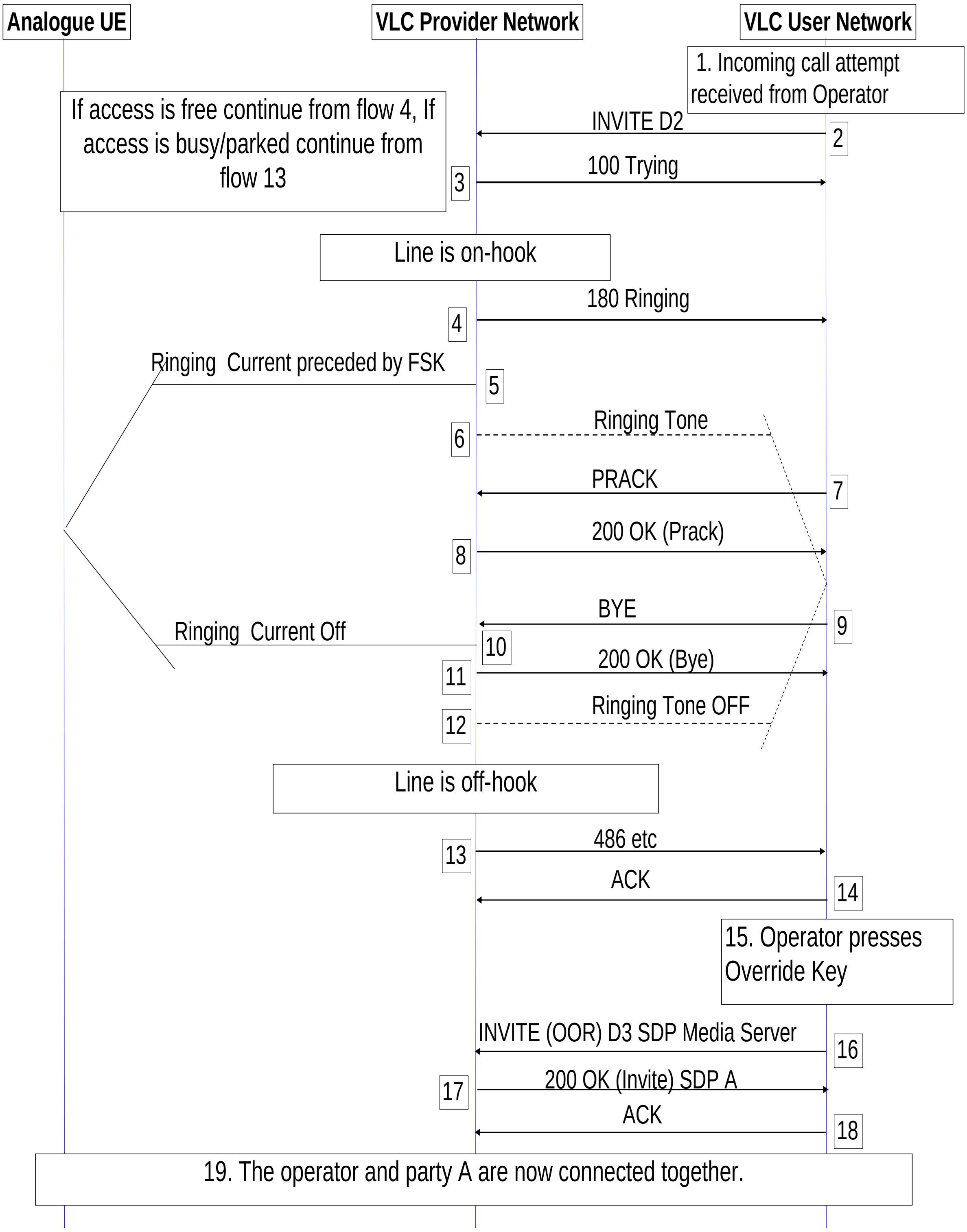
Note: "Preconditions" which optionally may be included are not shown on this Call Flow

Flow Number	Action	Additional Comments
1	VLC Provider network identifies correct profile for this line (WARMLINE) and applies correct digit map (including timers). Note: WARMLINE requires that the initial digit timer duration is significantly shorter than for a normal line (i.e. 5 seconds).	Once the OFF HOOK is detected any subsequent INVITE received by the VLC Provider Network is rejected with a 486 Busy Here.
2	VLC Provider network applies correct Dial Tone (Ordinary Dial Tone or Message Waiting Dial Tone)	
3	Initial digit timer T expires and Dial Tone is removed	The VLC Provider Network checks to see if there is any available bandwidth for the call. If there is no available bandwidth to the VLC User Network for the type of call (ordinary / priority) the VLC Provider Network SHALL connect the caller to the All Lines Busy announcement.
4	The VLC Provider Network sends an INVITE digit_timeout@domain with SDP A to the VLC User Network. The contents of the INVITE include the following: Request-URI – sip:digit_timeout@<domain> To: – contains the same as the Request-URI From: - contains the VLC_Line_ID of calling line Route: - contains the same as the Service-Route header in the 200 OK response to the associated previous REGISTER message. P-Asserted-Identity: - contains the VLC_Line_ID of calling line and the cpc parameter set in accordance with ND 1019 [7] sections 3.1 & 4.1. P-Charging-Vector: - Contents in accordance with RFC 3455 [10] and ND 1615 [16].	For lines with PATS the following rules apply: If there is a PN associated with the calling line the VLC USER Network SHALL remove the URI from the From header and replace it with the PN. Otherwise it SHALL remove the URI from the From header and replace it with the NN before forwarding the INVITE. The PN or NN shall be a Tel or SIP URI containing an E.164 number beginning with "+44". The VLC User Network SHALL also modify the P-Asserted-Identity header so that it contains the NN (Tel or SIP URI containing an E.164 number prefixed with "+44") before forwarding the INVTE towards the

	Require: 100rel Supported: preconditions SDP Offer The VLC User Network marks the line as “busy on an outgoing call”. (This status will be maintained until the calling customer goes On Hook and a BYE is received by the VLC User Network - see E.2.8) and proceeds to set up the call to the pre-determined destination (e.g. sends an INVITE towards the terminating network).	terminating network. The VLC User Network shall, when appropriate, add the SIP Privacy header (or equivalent CLI restricted/withheld indication if interworking directly to another signalling system). The addition of the Privacy header (with priv-value = "id") or equivalent indication should be determined by the subscribed privacy service for the calling line. For definitions of PN & NN see ND 1016 [4]
5	100 Trying	
6	Call continues in E.2.5 from flow no.9	

E.2.13 Operator Override (OOR)

This flow covers the case where the VLC User Network believes the line is on-hook but the VLC Provider Network knows the line is off-hook. OOR calls to lines that the VLC User Network knows to be busy are handled within the VLC User Network.

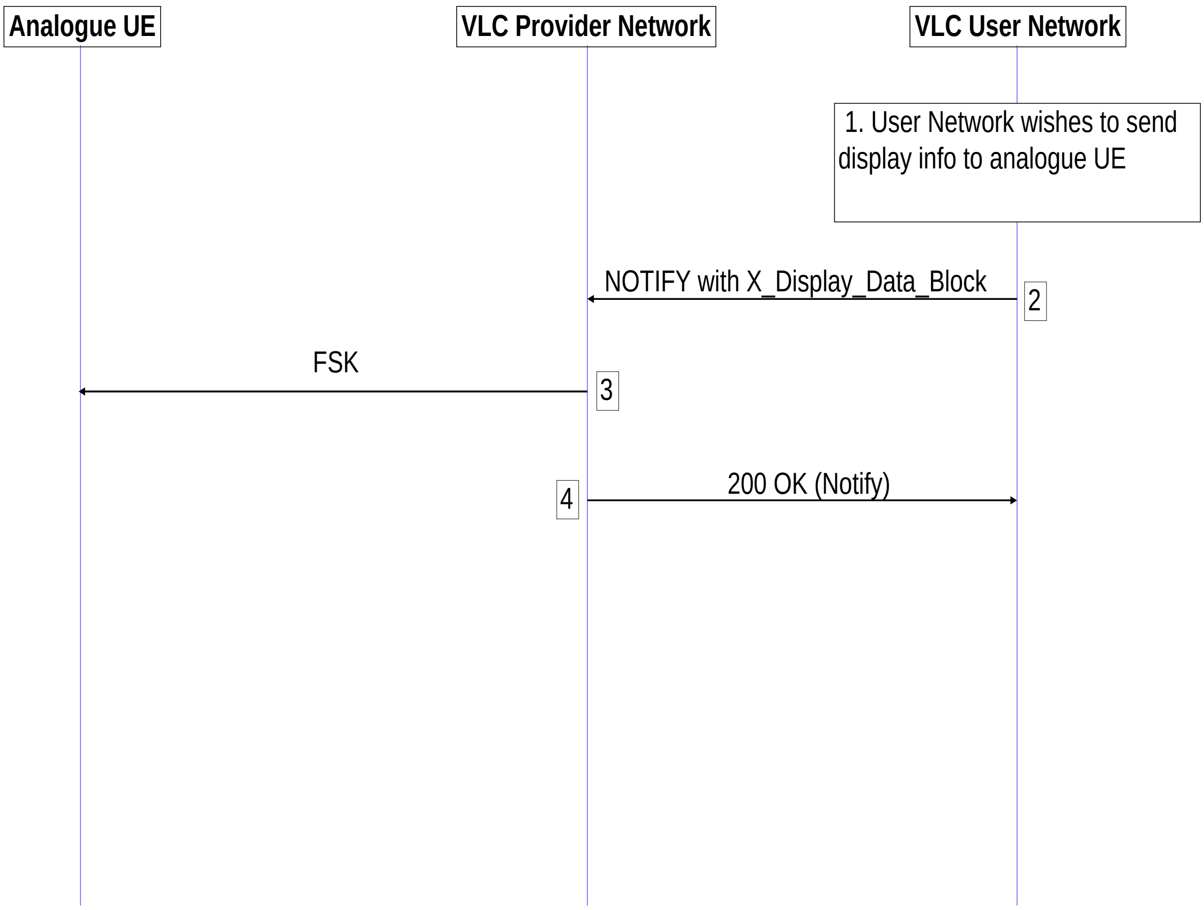


Flow Number	Action	Additional Comments
1	An incoming call attempt from an Operator (CPC= Operator) is received by the VLC User Network which then marks the line as busy on an incoming call (if not already marked as busy).	The VLC User Network checks to see if there is any available bandwidth for the call. If there is no available bandwidth to the VLC Provider Network for the type of call the VLC User Network SHALL reject the call attempt by sending a SIP 580 Response (or equivalent) to the preceding network.
2	Assuming the VLC User Network thinks the line was Free it sends an INVITE which includes the following: Request-URI – contains the VLC_Line_ID as defined in ND1620 [9]. Note that the VLC_Line_ID is equivalent to the IMS Public Identity. To: – contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. From: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be a PN if there is one else the NN. P-Asserted-Identity: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be the NN. P-Charging-Vector: - Contents as in the received INVITE or if call is received in another signalling system generated in accordance with RFC 3455 [10] and ND 1615 [16] Require: 100rel Supported: preconditions SDP Offer and may include: Alert-Info: <data:,RCxx> “X-Display-Data-Block”	If the called customer has subscribed to the CND or other equivalent service, the VLC User Network MAY map the From: header in the sent INVITE (which may have been derived from an equivalent CLI parameter of another signalling system e.g. UK ISUP according to ND 1020 etc) to the “Calling Line Directory Number” parameter of the “Display Data Block” (application/X-Display-Data-Block) The mapping is described in section F.1 of this document.
3	100 Trying	
4	180 Ringing (with SDP Answer). The P-Early-Media header with parameter value "sendrecv" SHALL be included.	Alternatively if the line is Busy on an outgoing call setup or in the clearing phase, the call then continues from flow 13
5	Ringing current (cadence as indicated in Alert-Info header or default cadence if header is not present) which MAY be preceded by FSK (derived from contents of “X-Display-Data-Block” if present)	

6	Ringing Tone sent in the bearer from the A-MGW towards the I-BGF in VLC Provider network	
7	PRACK	
8	200 OK (Prack)	
9	The Operator releases the call resulting in a BYE from the VLC User Network to the VLC Provider Network. Note: Alternatively the Operator may wait for the called customer to answer the call.	
10	Ringing current off	
11	200 OK (Bye)	
12	Ringing Tone Off	
13	486 etc	
14	ACK	
15	Operator Presses Override Key	The VLC User Network checks to see if there is any available bandwidth for the call. If there is no available bandwidth to the VLC Provider Network for the type of call the VLC User Network SHALL reject the Override request according to the OOR service implemented by the VLC User Network.
16	The VLC User Network sends an INVITE including the following contents: Request-URI – contains the VLC_Line_ID as defined in ND1620 [9]. Note that the VLC_Line_ID is equivalent to the IMS Public Identity. To: – contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. From: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be a PN if there is one else the NN. P-Asserted-Identity: - contents as in the received INVITE or, if call is received in another signalling system, as mapped to SIP; subject to any processing by the VLC User Network. For calls from a line with PATS this should be the NN. P-Charging-Vector: - Contents as in the received INVITE or if call is received in another signalling system generated in accordance with RFC 3455 [10] and ND 1615 [16] Require: 100rel Supported: preconditions Alert-Info: <data:,OORxx>	

	SDP Offer and may include: “X-Display-Data-Block”	
17	The VLC Provider Network connects to the called line irrespective of the current conditions on the line and sends a 200 OK (Invite) to the VLC User Network. Note: If the line had become free then ringing current would be applied to the analogue line and a 180 Ringing would be returned to the VLC User Network and the call would continue as in E.2.2 flow 8.	
18	ACK	
19	The Operator and Party A are connected together in the VLC User Network.	

E.2.14 Handling of NOTIFY containing X-display data

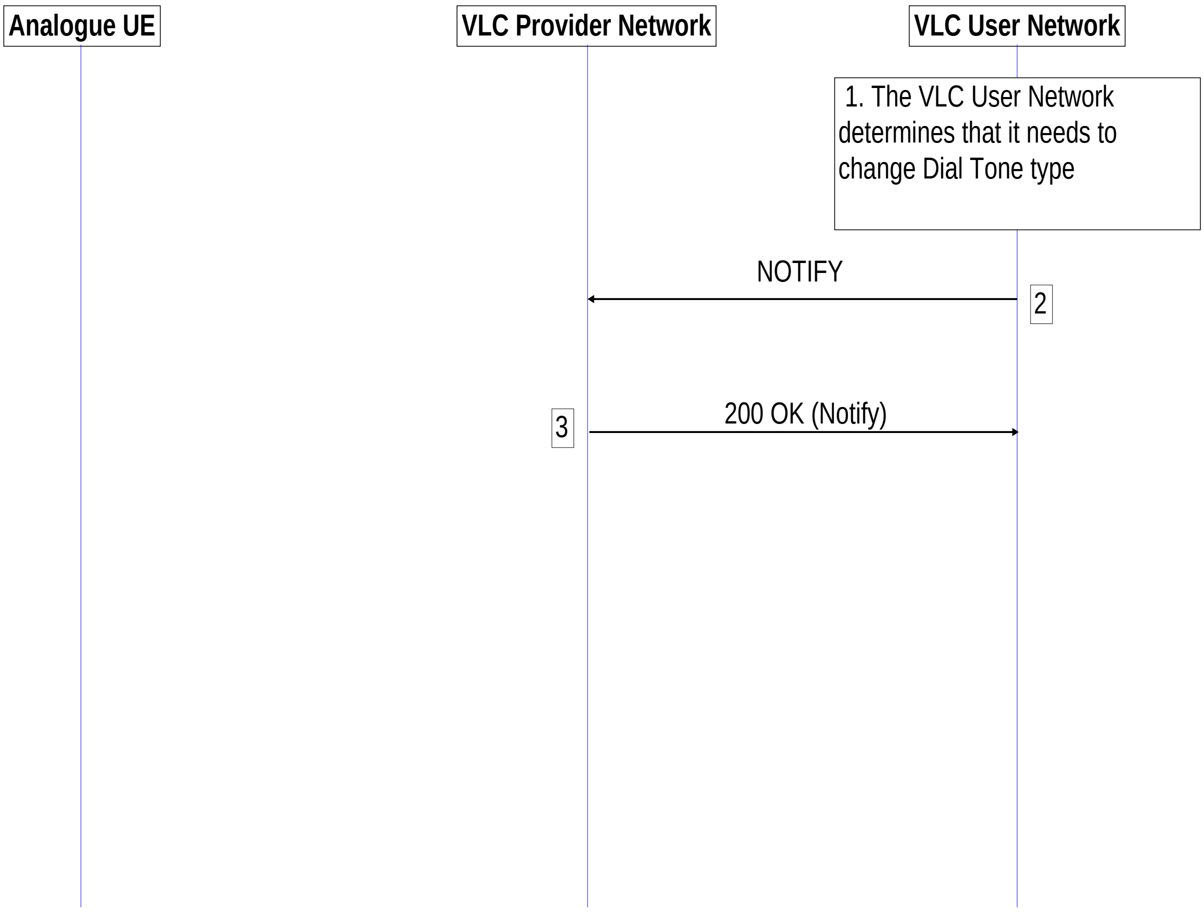


Note: the NOTIFY handling described in this section can be combined with any other appropriate NOTIFY.

Flow Number	Action	Additional Comments
1	User Network wishes to send display info to analogue UE and an INVITE is not appropriate.	
2	A NOTIFY is sent containing a MIME type "X-Display-Data-Block" from the VLC User Network to the VLC Provider Network	
3	The content of the "X-Display-Data-Block" SHALL be used to send FSK (either in the idle state or the active call state) by copying the contents of this block to the appropriate signalling protocol between the AGCF and the A-MGW (e.g. using the H.248 andisp/data signal). Note: If the line is busy in the call set-up phase then the FSK SHALL be delayed until the line is either idle or in the active	

	call phase.	
4	200 OK (Notify)	

E.2.15 Handling of NOTIFY to change Dial Tone (e.g. Message Waiting Service)



Note: the NOTIFY handling described in this section can be combined with any other appropriate NOTIFY.

Flow Number	Action	Additional Comments
1	The VLC User Network determines that it needs to change Dial Tone type	
2	A NOTIFY is sent from the VLC User Network to the VLC Provider Network. The VLC Provider Network SHALL store the request and subsequently, for future call originations, instruct the A-MGW to play the requested Dial Tone. Note: the methods used to request Dial Tone change can be found in section 5.3.2.2.	
3	200 OK (Notify)	

E.3 VLC Provider Network procedures to handle unexpected INVITEs

E.3.1 Introduction

The treatment of a new SIP session INVITE received by the VLC Provider Network will depend on its perception of:

- the state of the VLC line addressed by that INVITE; and
- the state of any existing SIP session(s) associated with that VLC line.

Under normal operation a new SIP session INVITE will be treated as a terminating call attempt, CW call offer, etc. as shown by the example call flow diagrams in E.2. However, under some anomalous operation conditions (e.g. following loss of call/session records in the VLC User Network or VLC Provider Network) it is possible that a new SIP session INVITE will have to be treated as “unexpected”.

Further information on identifying unexpected INVITEs may be found in Annex G.

Note: re-INVITEs within existing SIP dialogues are not covered by these procedures. SIP re-INVITEs are not regarded as unexpected unless they break underlying SIP protocol in which case they are handled by standard SIP procedures.

E.3.2 Handling unexpected INVITEs

On receiving an unexpected INVITE the VLC Provided Network shall attempt to recover alignment between the VLC Provider and VLC User Networks by applying the following principles:

a) Reject the unexpected INVITE by using response code 486 “Busy here”.

b) Fail (by appropriate means) any existing session establishment attempt which has not yet reached a stable state (200 OK (Invite) sent or received), e.g.:

  i) by sending a CANCEL to the VLC User Network in the case of an outgoing session establishment attempt; or

  ii) by returning response code 487 “Request terminated” in the case of an incoming session establishment attempt.

c) Maintain any established session which appears (at least to the VLC Provider Network) to be in a stable state (200 OK (Invite) sent or received). The session will be maintained until either:

  - the VLC line initiates clearing by going on-hook; or
  - the SIP session refresh procedures discover an inconsistency between the VLC Provider Network and the VLC User Network.

The purpose of this approach in the VLC Provider Network is to allow an established communication to continue (possibly to a natural conclusion) as long as the bearer connection has not been impaired elsewhere.

Note: In some supplementary service cases, if an unexpected INVITE is received, there can be more than one existing session in a stable state.

d) Initiate an autonomous analogue clearing sequence (including in-band indications) if, at the end of the SIP recovery action, the VLC line is no longer associated with a session, but is still off-hook.

Annex F (normative):Coding of SIP headers and MIME bodies specifically for UK Voice Line Control

F.1 Derivation of FSK data block when sent using MIME body (Optional)

Data to be transmitted to the analogue UE in the FSK data block MAY be sent from the VLC User Network to the VLC Provider Network in an INVITE or NOTIFY message. If so, it SHALL be in a body part with MIME type and sub-type "application/X-display-Data-Block". The contents SHALL be ASCII-hexadecimal coded data and shall consist of the Message type, Message length, Presentation layer message plus the Checksum of the FSK Data link message (packet) as defined in ETSI ES 200 659-3 [13] / BT SIN 227 [14] (this data MAY include the calling number and other information e.g. the number of messages waiting etc as defined in ETSI ES 200 659-3 [13] / BT SIN 227 [14] or it may use some other coding scheme defined by the VLC User Network, and therefore the VLC Provider Network SHALL NOT attempt to verify any checksum);

MIME X-Display-Data-Block	Derivation
FSK Data Block i.e. consisting of the Message type, Message length, Presentation layer message plus the Checksum of the FSK Data link message (packet) as defined in ETSI ES 200 659-3 [13] / BT SIN 227 [14]	Data Block to be sent as FSK (e.g. formatted in accordance with ETSI ES 200 659-3 [13] / BT SIN 227 [14] or some other coding scheme defined by the VLC User Network) but coded as ASCII-Hexadecimal. Note: For the PSTN CND service if the Calling Line Identity (or Calling Line Directory Number) parameter is included it SHOULD be derived from a CLI parameter containing the Presentation Number (if received), or if no Presentation Number has been received, from a CLI parameter containing the Network Number (e.g. from the SIP "From" header which contains a PN if there is one otherwise it contains a NN). See ND 1016 [4] for definitions of Presentation Number and Network Number.

F.2 Format and Coding of Alert-Info header

An Alert-Info header MAY be included in an INVITE message sent from the VLC User Network to the VLC Provider Network to control/indicate one of the following:

- the Ringing Current Cadence of the alerting signal sent on the analogue line; or
- the Frequency & Cadence of the Call Waiting Tone bursts sent on the analogue line that are used to alert the customer that a new call has arrived.
- The fact that this is an Operator Override Call (OOR)

If the Alert-Info header is not included and not mandated by the signalling in the INVITE message then the default value of the Cadence Code SHALL be 01. (See table below)

The Alert-Info header shall conform to the following ABNF syntax:

```
Alert-Info = "Alert-Info" HCOLON  "<data:," cadence ">"
 cadence = %x52.43 code /              ; RCxx
    %x43.57.54 code "-" interval /     ; CWTxx-yyyy
    %x4F.4F.52 code                    ; OORxx
 code = 2HEXDIG
 interval = NZDIGIT 3DIGIT
 NZDIGIT = %x31-39                     ; non-zero digit
```

The interpretation of the cadence is shown in the table below.

Example:

Alert-Info: <data:,CWT02-7500>

<table>
<tr>
<td>Alerting Cadence Indicator</td>
<td>RCxx or CWTxx-yyyy or OORxx

Where xx is the Cadence Code (ASCII-Hexadecimal coded) and

where yyyy is the interval (ms) between successive Call Waiting Tone pulses (allowable range 1000 – 9999, default 5000)

The coding of the Cadence Code is shown below:
<table>
<tr><td>Cadence Code (xx)</td><td>Ringing Current / Operator OverRide Cadence</td></tr>
<tr><td>00</td><td>Not Used – treat as value 01</td></tr>
<tr><td>01 (default)</td><td>0.4sec On, 0.2sec Off, 0.4sec On, 2.0sec Off</td></tr>
<tr><td>02</td><td>0.4sec On, 0.8 sec Off</td></tr>
<tr><td>03</td><td>0.25sec On, 0.25sec Off, 0.25sec On, 0.25sec Off, 0.25sec On, 1.75sec Off</td></tr>
<tr><td>04</td><td>2.0sec On, 4.0sec Off</td></tr>
<tr><td>05</td><td>Continuous ringing</td></tr>
<tr><td>06</td><td>1.0sec On, 2.0sec Off</td></tr>
<tr><td>07</td><td>No ringing current.</td></tr>
<tr><td>08 to FF</td><td>Not Used – treat as value 01</td></tr>
<tr><td>Cadence Code (xx)</td><td>Call Waiting Tone - Frequency & Duration of Pulses (interval between successive pulses determined by yyyy field)</td></tr>
<tr><td>00</td><td>Not used – treat as value 01</td></tr>
<tr><td>01 (default)</td><td>400Hz for 0.1 sec (this timing is controlled by the access media gateway)</td></tr>
<tr><td>02</td><td>400Hz for 0.03sec On, 0.01sec Off, 0.03sec On (these timings are controlled by the access media gateway)</td></tr>
<tr><td>03 to FF</td><td>Not used – treat as value 01</td></tr>
</table>
</td>
</tr>
</table>

According to RFC 3261 [15] the inclusion of an Alert-Info header in a 180 Ringing response is also allowed, but its use over the Ic reference point between the VLC User Network and the VLC Provider Network is not supported in this release of the specification.

F.3 Error-Info header in 484 Response with “Min. Digits = N” Indication

See: ETSI TISPAN TR 183 056 [2]

F.4 Error-Info header in SIP Failure Response Messages

The Error-Info header MAY be included in an appropriate SIP Response message in order to control the actual announcement played to the calling customer. The SIP response code used SHOULD still align as closely as possible with the reason for failure of the call.

The Error-Info header shall conform to the following ABNF syntax:

```
Error-Info = "Error-info" HCOLON "<data:," announcement ">"
 announcement = %x41 announcement-name           ; Annnn
 announcement-name = 1*LOWER
 LOWER = %x41-5A                                 ; lowercase letters
```

The interpretation of the cadence is shown in the table below.

Example:
Error-Info: <data:,Aicban>

<table>
<tr>
<td>Announcement Indicator</td>
<td>Astring

Where string is the Announcement Name string

The coding of the Announcement Name string is shown below:
<table>
<tr><th>Announcement Name string</th><th>Announcement</th></tr>
<tr><td>nuan</td><td>“Sorry, the number you have called is not available”</td></tr>
<tr><td>icban</td><td>“This number does not receive incoming calls”</td></tr>
<tr><td>callgapan</td><td>“The telephone network is busy at the moment – please try again later – you have not been charged for this call” .</td></tr>
<tr><td>unrecnuman</td><td>“The number you have dialled has not been recognised, please check and try again”</td></tr>
<tr><td>fltan</td><td>“Sorry there is a fault, please try again”</td></tr>
<tr><td>numtooan</td><td>“This number is temporarily out of order, we are sorry for any inconvenience”</td></tr>
<tr><td>noreplyan</td><td>“Sorry there is no reply”</td></tr>
<tr><td>linesbusyan</td><td>“Sorry lines are busy, please try later”</td></tr>
<tr><td>opcan</td><td>“The other person has hung up”</td></tr>
<tr><td>callnotconan</td><td>“Sorry your call cannot be connected at present, please try again”</td></tr>
<tr><td>nodigitsan</td><td>“Please hang up and try again”</td></tr>
</table>
</td>
</tr>
</table>

	servterman	“Please hang up”
	Any other string	Play tone or announcement according to SIP Response code without Error-Info header (see E.1.1).

F.5 X-Service-Indicator header

The X-Service-Indicator header MAY be included in appropriate INVITEs and 200 OK (Bye) responses in order to instruct the AGCF to hold the access resource.

The X-Service-Indicator header shall conform to the following ABNF syntax:

```
Service-Indicator = "X-service-indicator" HCOLON service-identifier
    *(COMMA service-identifier)
service-identifier = "hold-resource" / "use-held-resource"
```

Annex G (normative):VLC Provider Network: identifying unexpected INVITEs

In the following Table:

- the rows identify a number of possible conditions which the VLC Provider Network might perceive in relation to a particular VLC line.
- the columns identify different values of Alerting cadence indicator and "use-held-resource" indication which might be included in a newly received INVITE for the same VLC line.
- the elements identify for each perceived condition and INVITE contents combination whether the received INVITE should be treated as a normal occurrence (as described in E.2) or as an unexpected occurrence (as described in E.3).

VLC Provider Network view of the current condition of the VLC line and any associated session(s)		**indications in newly received INVITE**		
		Alerting cadence indicator = "RC"” (or absent)		**Alerting cadence indicator = "CW"**
		"use-held-resource" absent	**"use-held-resource" present**	
Basic call cases	**Idle / VLC line on hook**	**normal** (terminating call case)	treat as **normal** (terminating call case)	treat as **normal** (terminating call case)
	outgoing session requested (no response yet received other than 100 Trying)	**normal** (i/c & o/g calls cross)	treat as **normal** (i/c & o/g calls cross)	treat as **normal** (i/c & o/g calls cross)
	outgoing or incoming session proceeding (200 OK not yet received or sent)	**unexpected**	**unexpected**	**unexpected**
	outgoing or incoming session established (200 OK received or sent)	**unexpected**	**unexpected**	**normal** (CW call offer case)
	session released, but VLC line not yet free (still off-hook)	**normal** (terminating call case)	treat as **normal** (terminating call case)	treat as **normal** (terminating call case)
Multi-party service cases	**e.g. CW being progressed held call / active call 3-way call etc.**	**unexpected**	**unexpected**	**unexpected**
Access resources explicitly held cases	**e.g. in support of: CSH feature on-hook to resume held call etc.**	**unexpected**	**normal** (use of held resource case)	**unexpected**
Access resources implicitly held cases	**on-hook to take waiting call**	treat as **normal** (use of held resource case)	**normal** (use of held resource case)	**unexpected**

History

Document history		
Version	**Date**	**Milestone**
Issue 1.0 Draft 1	January 2007	Initial draft
Issue 1.0 Draft 2	February 2007	Further content added in particular section 4.2.
Issue 1.0 Draft 2b	February 2007	Changes to section 2.2.1 and further content added in particular to section 4.2
Issue 1.0 Draft 2c	February 2007	Changes to Call flows 4.2.1, 4.2.4 & 4.2.5
Issue 1.0 Draft 3a	March 2007	Changes agreed at AP working party meeting 01/03/07
Issue 1.0 Draft 3b	April 2007	Changed and additional Call Flows in section 4.2. Changes to X-Display-Data-Block. Description of Alerting Cadence Indicator in section 2.2.2.
Issue 1.0 Draft 3c	April 2007	Modified Call Flows 4.2.1 & 4.2.5
Issue 1.0 Draft 4a	April 2007	Changes agreed at AP working party meeting 20/04/07. Changes to and additional Call Flows in section 4.2. Also some of the comments received from Mark Ashworth (Nortel) have been incorporated.
Issue 1.0 Draft 4b	May 2007	Editorial corrections and technical corrections to various call flow descriptions in section 4.2
Issue 1.0 Draft 5a	June 2007	Changes agreed at AP working Party 24/05/07. Additional Call Flows in section 4.2
Issue 1.0 Draft 5b	June 2007	A significant change to outgoing call and release sequences in section 4.2 Call Flows. Call flows for Hotline, Warmline, Operator Override (OOR) and digit collection timeouts added. Annex A & B added
Issue 1.0 Draft 5c	June 2007	Changes to mapping of SIP response codes to tones and announcements. Nodal actions mapping to announcements included in various Call Flows.
Issue 1.0 Draft 5d	July 2007	Restructure of document layout. Various technical changes.
V0.6.1	August 2007	Changes agreed at AP working party meeting 27/07/07. Correction of Call Flow E.2.6, Changes to the Announcement Indicator defined in F.4, and other minor technical changes.
V0.6.2	September 2007	Change to registration process – P-Associated-URI header contents no longer ignored.
V0.6.3	September 2007	Clarification added regarding setting of Privacy header on outgoing calls.
V0.7.1	November 2007	Updated after review at AP working party on 23rd October and as a result of further comments received after the meeting.
V0.7.2	November 2007	Further changes as a result of comments received after the last meeting (23rd October)
V0.8.1	December 2007	Updated after review at AP working party on 29th November.
V0.8.2	December 2007	Further editorial changes.

V0.8.3	January 2008	Changes received by e-mail after last meeting (29th November 07)
V0.9.1	February 2008	Updated after review at AP working party on 1st February 2008. Document number changed to ND 1021 (from ND 1619).
V0.10.1	March 2008	Updated after review at AP working party on 11th March 2008.
V0.11.1	May 2008	Updated after review at AP working party on 28th April 2008.
V0.11.2	June 2008	Further updates
V0.12.1	June 2008	Updated after review at AP working party on 13th June 2008. Moved to new NICC template.
V0.12.2	July 2008	Updated to include agreed CP04(08)22
V0.12.3	July 2008	Further updates and editorials
V0.13.1	Sept 2008	Updates following resolution meeting on 8th Sept.
V0.14.1	Oct 2008	Updates following meeting on 9th Oct and follow up audio on 22nd Oct
V0.14.2	Oct 2008	Completion of updates following meeting on 9th Oct and follow up audio on 22nd Oct.
V0.15.1	Nov 2008	Updated following AP WP meeting on 7th Nov 2008
V0.15.2	Nov 2008	Further editorials
V0.16.1	Jan 2009	Updated to include Nortels comments (amended with the AP WP Chairman proposals) included. Note that these comments were received during the TSG 28 day approval process.
V0.16.2	Jan 2009	Updated with minor changes to Notrels comments at the AP WP meeting on 28th Jan
V0.16.3	Feb 2009	Minor editorials
V1.1.1	Mar 2009	Formal issue