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Diversion Indication in SIP
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Abstract

This RFC, which contains the text of an Internet Draft that was submitted originally to the SIP Working Group, is being published now for the historical record and to provide a reference for later Informational RFCs. The original Abstract follows.

This document proposes an extension to the Session Initiation Protocol (SIP). This extension provides the ability for the called SIP user agent to identify from whom the call was diverted and why the call was diverted.
The extension defines a general header, Diversion, which conveys the diversion information from other SIP user agents and proxies to the called user agent.
This extension allows enhanced support for various features, including Unified Messaging, Third-Party Voicemail, and Automatic Call Distribution (ACD). SIP user agents and SIP proxies which receive diversion information may use this as supplemental information for feature invocation decisions.

Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

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Table of Contents

1. Introduction 5
2. Definitions 5
3. Abbreviations 5
4. Overview 6
4.1. When is the Diversion header used? 6
5. Extension syntax 6
6. Detailed semantics 7
6.1. UAS Behavior 7
6.2. UAC Behavior 7
6.3. Redirect Server Behavior 7
6.4. Proxy Server Behavior 8
6.4.1. Proxy Logic for Diversion header 8
7. Examples using Diversion header 8
7.1. Call Forward Unconditional 9
7.1.1. Network Call Forward Unconditional, P2 recursing . . . 9
7.1.2. Network Call Forward Unconditional, P1 non-recursing P2 non-recursing 9
7.1.3. Network Call Forward Unconditional, P1 recursing P2 non-recursing 10
7.1.4. Endpoint Call Forward Unconditional, P1 recursing P2 non-recursing 11
7.2. Call Forward on Busy 12
7.2.1. Network Call Forward on Busy, P2 recursing 12
7.2.2. Network Call Forward on Busy, P1 non-recursing P2 non-recursing 13
7.2.3. Network Call Forward on Busy, P1 recursing P2 non-recursing 14
7.2.4. Endpoint Call Forward on Busy, P1 recursing P2 non-recursing 15
7.3. Call Forward on No-Answer 16
7.3.1. Network Call Forward on No-Answer, P2 recursing . . . 17
7.3.2. Network Call Forward on No-Answer, P1 non-recursing P2 non-recursing 17
7.3.3. Network Call Forward on No Answer, P1 recursing P2 non-recursing 18
7.3.4. Endpoint Call Forward on No-Answer, P1 recursing, P2 non-recursing B non-recursing 19
7.4. Call Forward on Unavailable 20
7.4.1. Network Call Forward on Unavailable, P2 recursing . . 21
7.4.2. Network Call Forward on Unavailable, P1 non-recursing P2 non-recursing 21
7.4.3. Network Call Forward on Unavailable, P1 recursing P2 non-recursing 23
7.5. Multiple Diversions 23
7.5.1. Call Forward Unconditional and Call Forward Busy . . . 24
7.5.2. Call Forward Unconditional and Call Forward No

Answer 26
8. Security Considerations 27
9. Further Examples 27
9.1. Night Service/Automatic Call Distribution (ACD) using Diversion header 27
9.2. Voicemail Service using Diversion header 35
9.3. Q&A on alternative approaches 39
10. Mapping ISUP/ISDN Redirection information to SIP Diversion header 39
10.1. Mapping ISUP/ISDN Diversion Reason codes 40
10.2. Mapping ISUP Redirection information to SIP Diversion header 40
10.2.1. ISUP Definitions 41
10.2.2. ISUP parameters 41
10.2.3. ISUP to SIP translation 42
10.2.4. SIP to ISUP translation 42
10.2.5. Example of ISUP to SIP translation 42
10.2.6. Example of SIP to ISUP translation 43
10.3. Mapping ISDN Redirection information to SIP Diversion header 44
10.3.1. ISDN Definitions 44
10.3.2. ISDN parameters 44
10.3.3. ISDN to SIP translation 45
10.3.4. SIP to ISDN translation 46
10.3.5. Example of ISDN to SIP translation 46
10.3.6. Example of SIP to ISDN translation 47
10.4. Information loss in SIP to ISUP/ISDN translation 48
10.4.1. Loss of diversion URI information 48
10.4.2. Loss of diversion reason information 48
10.4.3. Loss of diversion counter information 49
11. Contributors 49
12. Acknowlegements 49
13. Normative References 49
Authors' Addresses 49

1. Introduction

This RFC, which contains the text of an Internet Draft that was submitted originally to the SIP Working Group, is being published now for the historical record and to provide a reference for later Informational RFCs.

In the legacy telephony network, redirection information is passed through the network in ISDN/ISUP signaling messages. This information is used by various service providers and business applications to support enhanced features for the end user.
An analogous mechanism of providing redirection information would enable such enhanced features for SIP users.
The Diversion header allows implementation of feature logic based on from whom the call was diverted.
Version 06 (and forward) of the draft adds support for privacy and screening information.

2. Definitions

diversion:
A change to the ultimate destination endpoint of a request. A change in the Request-URI of a request that was not caused by a routing decision. This is also sometimes called a deflection or redirection.
A diversion can occur when the "user" portion of the Request-URI is changed for a reason other than expansion or translation.
A diversion can occur when only the "host" portion of the Request-URI has changed if the change was due to a non-routing decision.

divertor:
The entity which diverted the call.

recursing:
A SIP proxy or user agent which handles a received or internally generated 3xx response by forking new request(s) itself.

non-recursing:
A SIP proxy or user agent which handles a received or internally generated 3xx response by forwarding it upstream.

3. Abbreviations

CFUNC: Call Forward Unconditional
CFTOD: Call Forward Time-of-Day
CFB: Call Forward on Busy
CFNA: Call Forward on No Answer

CFUNV: Call Forward Unavailable
ACD: Automatic Call Distribution

4. Overview

In order to implement certain third-party features such as third-party voicemail and Automatic Call Distribution (ACD) applications, diversion information needs to be given to the called third-party so that he may respond to the caller intelligently.
In these situations, the party receiving a diverted call needs answers for two questions:
Question 1: From whom was the request diverted?
Question 2: Why was the request diverted?

This document proposes usage of the Diversion header to answer these questions for the party receiving the diverted call.
Insertion of the previous Request-URI (before the diversion occurred) into the Diversion header answers question 1.
Insertion of the "reason" tag into the Diversion header (by the divertor) answers question 2.

4.1. When is the Diversion header used?

The Diversion header SHOULD be added when a SIP proxy server, SIP redirect server, or SIP user agent changes the ultimate endpoint which will receive the call.
Diversion information SHOULD NOT be added for normal call routing changes to the Request-URI. Thus, the Diversion header is not added when features such as speed dial change the Request-URI.
When a diversion occurs, a Diversion header SHOULD be added to the forwarded request or forwarded 3xx response. The Diversion header MUST contain the Request-URI of the request prior to the diversion.
The Diversion header SHOULD contain a reason that the diversion occurred.
Existing Diversion headers received in an incoming request MUST NOT be removed or changed in forwarded requests.
Existing Diversion headers received in an incoming response MUST NOT be removed or changed in the forwarded response.
A Diversion header is added when features such as call forwarding or call deflection change the Request-URI.

5. Extension syntax

The syntax of the Diversion header is:

```
Diversion = "Diversion" ":" 1# (name-addr *( ";" diversion_params ))
    diversion-params = diversion-reason | diversion-counter |
                       diversion-limit | diversion-privacy |
                       diversion-screen | diversion-extension
    diversion-reason = "reason" "="
                    ( "unknown" | "user-busy" | "no-answer" |
                      "unavailable" | "unconditional" |
                      "time-of-day" | "do-not-disturb" |
                      "deflection" | "follow-me" |
                      "out-of-service" | "away" |
                      token | quoted-string )
    diversion-counter = "counter" "=" 1*2DIGIT
    diversion-limit = "limit" "=" 1*2DIGIT
    diversion-privacy = "privacy" "=" ( "full" | "name" |
                        "uri" | "off" | token | quoted-string )
    diversion-screen = "screen" "=" ( "yes" | "no" | token |
                                      quoted-string )
    diversion-extension = token ["=" (token | quoted-string)]
```

The following is an extension of tables 4 and 5 in [RFC3261]for the Diversion header:

	where	enc.	e-e	ACK	BYE	CAN	INV	OPT	REG
Diversion	R		h	-	-	-	o	-	-
Diversion	3xx		h	-	-	-	o	-	-

6. Detailed semantics

6.1. UAS Behavior

A SIP UAS which receives a request and returns a 3xx SHOULD add a Diversion header containing the previous Request-URI and the reason for the diversion.

6.2. UAC Behavior

A SIP UAC which receives a 3xx containing a Diversion header SHOULD copy the Diversion header into each downstream forked request which resulted from the 3xx.

6.3. Redirect Server Behavior

A SIP redirect server which receives a request and returns a 3xx containing a Contact which diverts the request to a different endpoint SHOULD add a Diversion header containing the Request-URI from the incoming request and the reason for the diversion.

6.4. Proxy Server Behavior

A non-recursing SIP proxy which receives a 3xx containing a Diversion header SHOULD forward the 3xx containing the Diversion header upstream unchanged.
A SIP proxy which receives a request and invokes a feature which changes the Request-URI of the forwarded request in order to divert the request to a different endpoint SHOULD add a Diversion header containing the Request-URI from the incoming request and the reason for the diversion.
A SIP proxy which receives a request and returns a 3xx containing a Contact which diverts the request to a different endpoint SHOULD add a Diversion header containing the Request-URI from the incoming request and the reason for the diversion.

6.4.1. Proxy Logic for Diversion header

```
if (pdu.is_request()) {
    if (request-URI is changed due to a called feature) {
        if (proxy.is_recursing()) {
            Add the Diversion header (indicating the reason
              that the call has been diverted) to
              the downstream forwarded request(s).
        } else {
            Add the Diversion header (indicating the reason
              that the call has been diverted) to
              the upstream forwarded 3xx response.
        }
    }
} else if (pdu.is_response()) {
    if (pdu.is_3xx()) {
        if (proxy.is_recursing()) {
          Copy Diversion header into forwarded INVITE(s).
        } else {
          Forward response upstream.
        }
    }
}
```

7. Examples using Diversion header

There are several implementations of call forwarding features that can be implemented by either recursing or non-recursing SIP proxies or SIP user agents.
A SIP proxy or user agent which generates or forwards 3xx's upstream is non-recursing. A SIP proxy or user agent which handles received (or internally generated) 3xx's itself is recursing.
The following examples illustrate usage of the Diversion header for

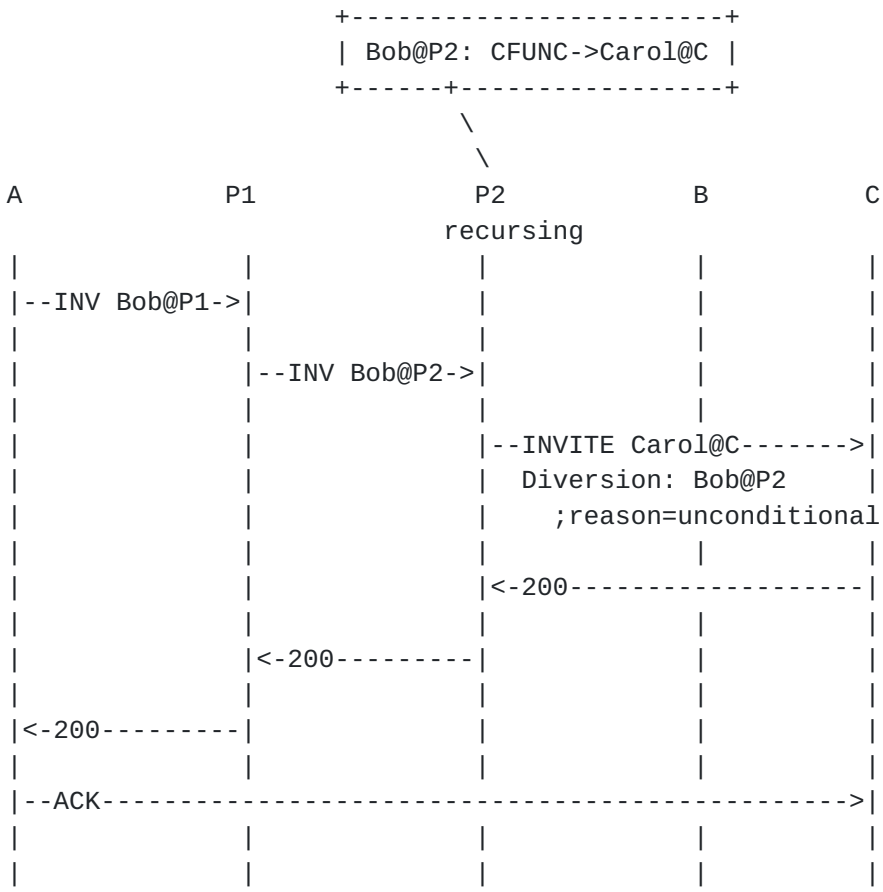
some of the variants of recursing and non-recursing proxies and user agents.

7.1. Call Forward Unconditional

Usage of the Diversion header is shown below for several variant implementations of Call Forward Unconditional.

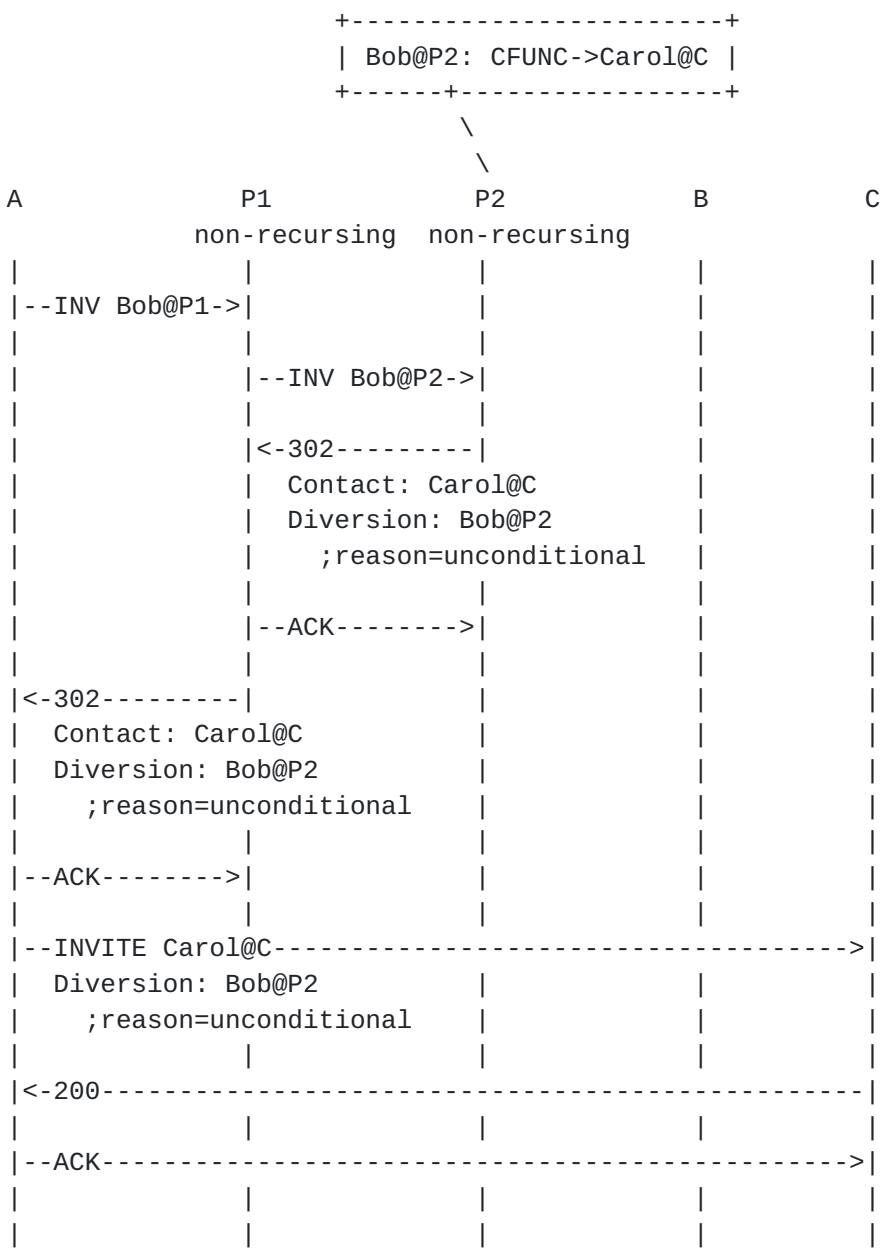
7.1.1. Network Call Forward Unconditional, P2 recursing

In this message flow, the call would normally be routed to Bob@B. However, Proxy 2 (P2) recursively implements Call Forward Unconditional (CFUNC) to Carol@C.



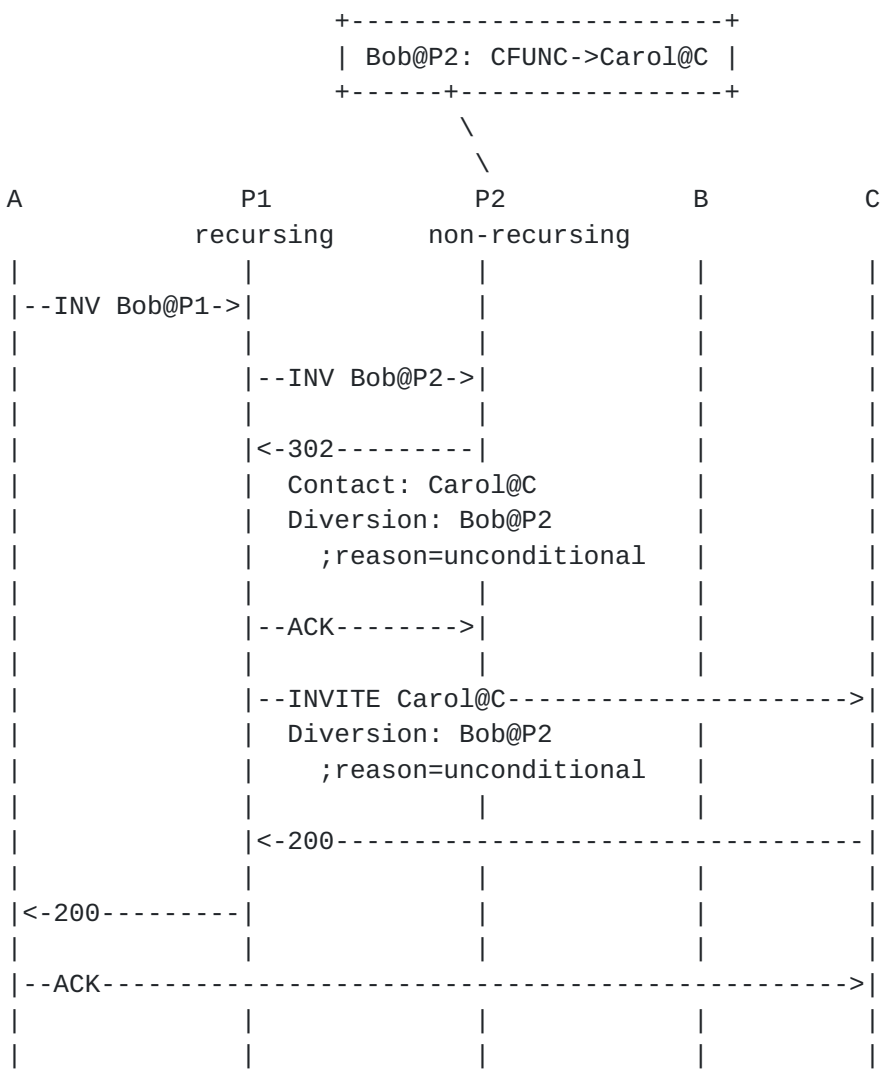
7.1.2. Network Call Forward Unconditional, P1 non-recursing P2 non-recursing

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward Unconditional (CFUNC) to Carol@C. Proxy 1 (P1) is non-recursing.



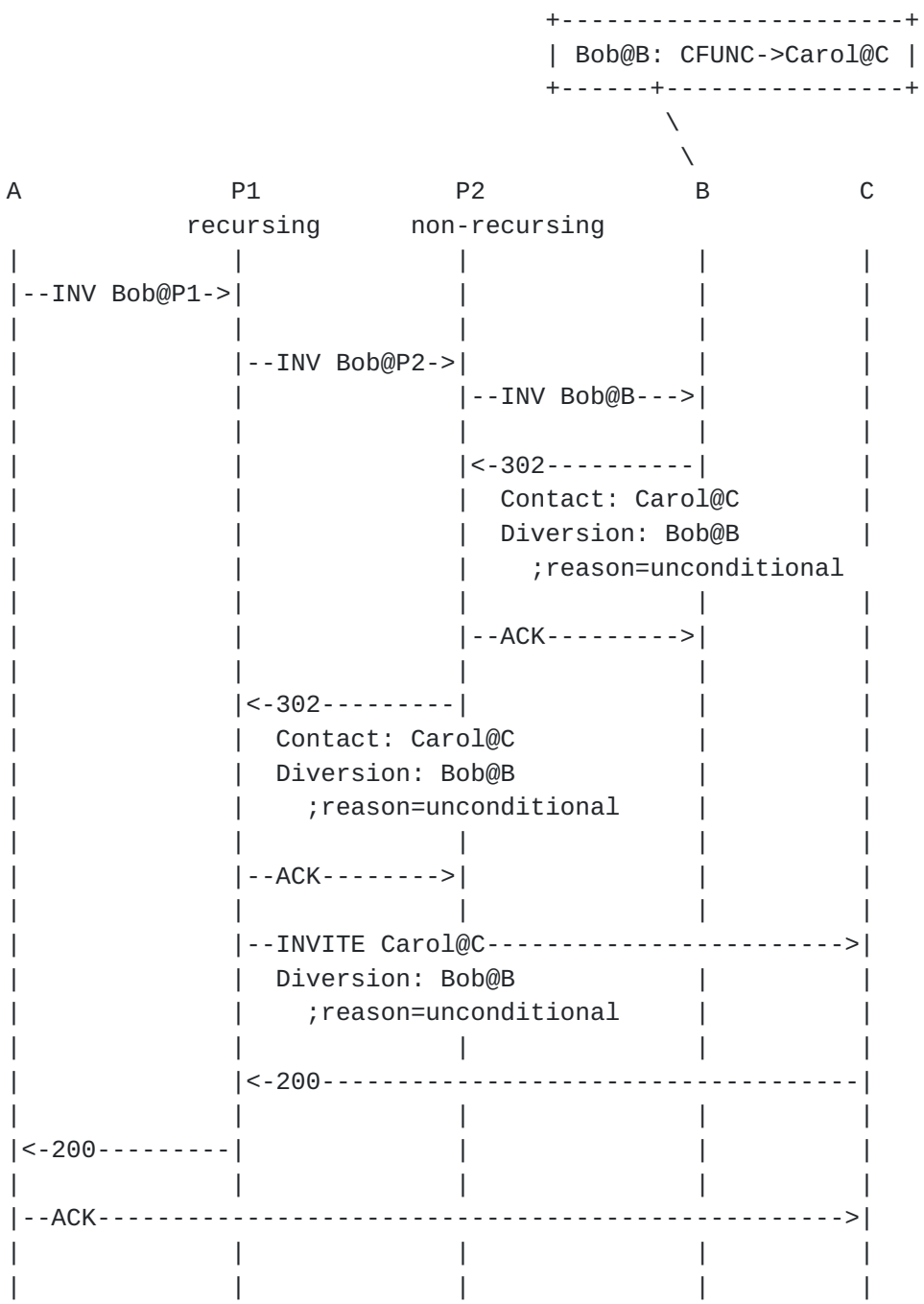
7.1.3. Network Call Forward Unconditional, P1 recursing P2 non-recursing

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward Unconditional (CFUNC) to Carol@C. Proxy 1 (P1) is recursing.



7.1.4. Endpoint Call Forward Unconditional, P1 recursing P2 non-recursing

In this message flow, user agent server B (B) non-recursively implements Call Forward Unconditional (CFUNC) to Carol@C. Proxy 2 (P2) is non-recursing. Proxy 1 (P1) is recursing.

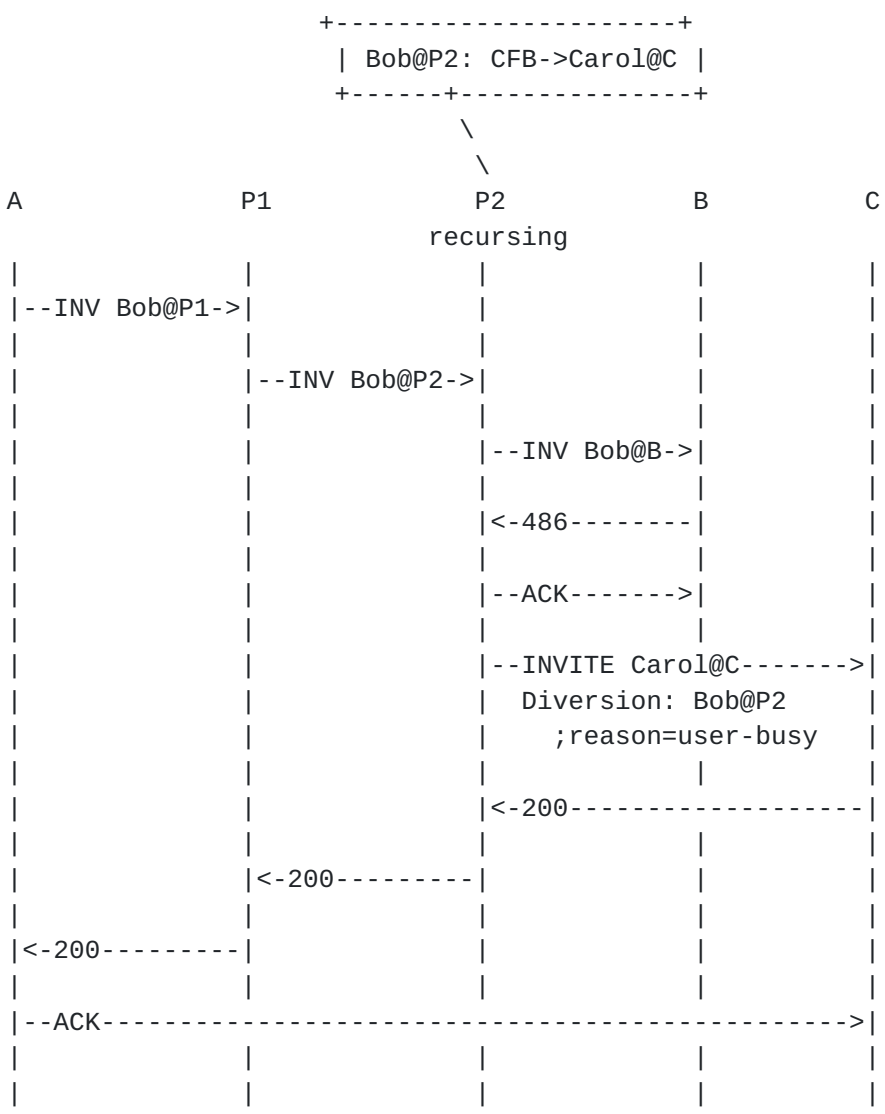


7.2. Call Forward on Busy

Usage of the Diversion header is shown below for several variant implementations of Call Forward on Busy.

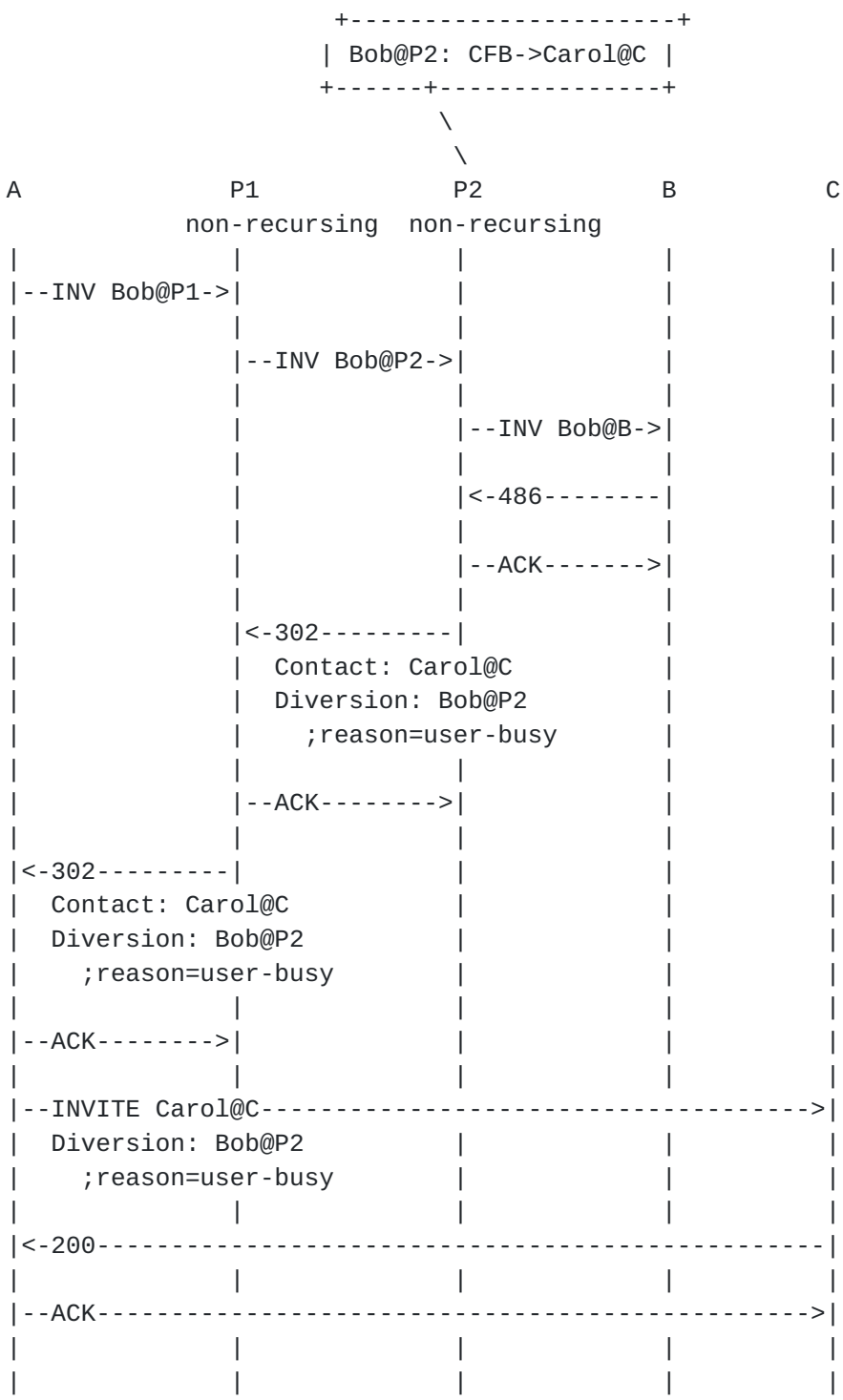
7.2.1. Network Call Forward on Busy, P2 recursing

In this message flow, Proxy 2 (P2) recursively implements Call Forward on Busy (CFB) to Carol@C.



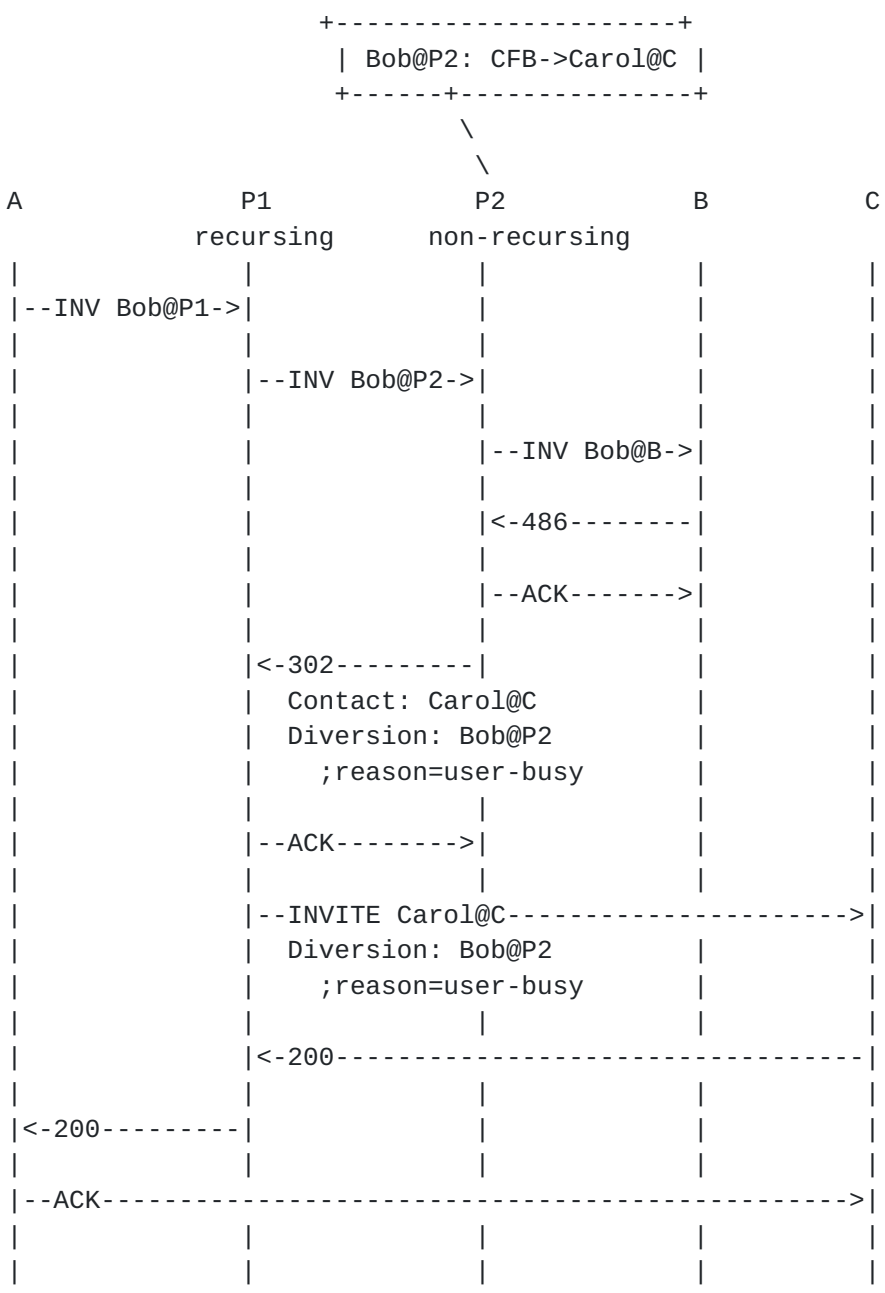
7.2.2. Network Call Forward on Busy, P1 non-recursing P2 non-recursing

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on Busy (CFB) to Carol@C. Proxy 1 (P1) is non-recursing.



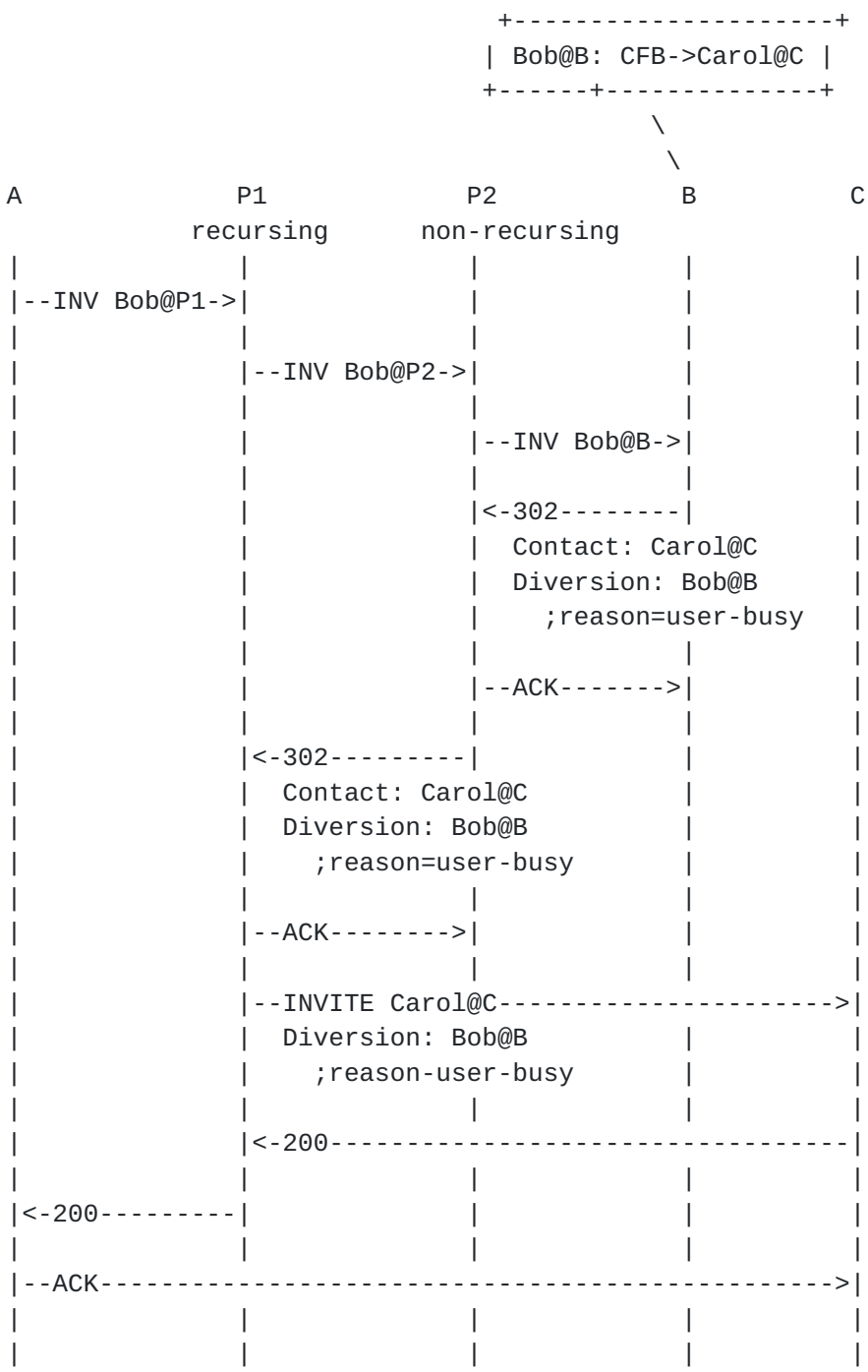
7.2.3. Network Call Forward on Busy, P1 recursing P2 non-recursing

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on Busy (CFB) to Carol@C. Proxy 1 (P1) is recursing.



7.2.4. Endpoint Call Forward on Busy, P1 recursing P2 non-recursing

In this message flow, user agent server B (B) non-recursively implements Call Forward on Busy (CFB) to Carol@C. Proxy 2 (P2) is non-recursing. Proxy 1 (P1) is recursing.

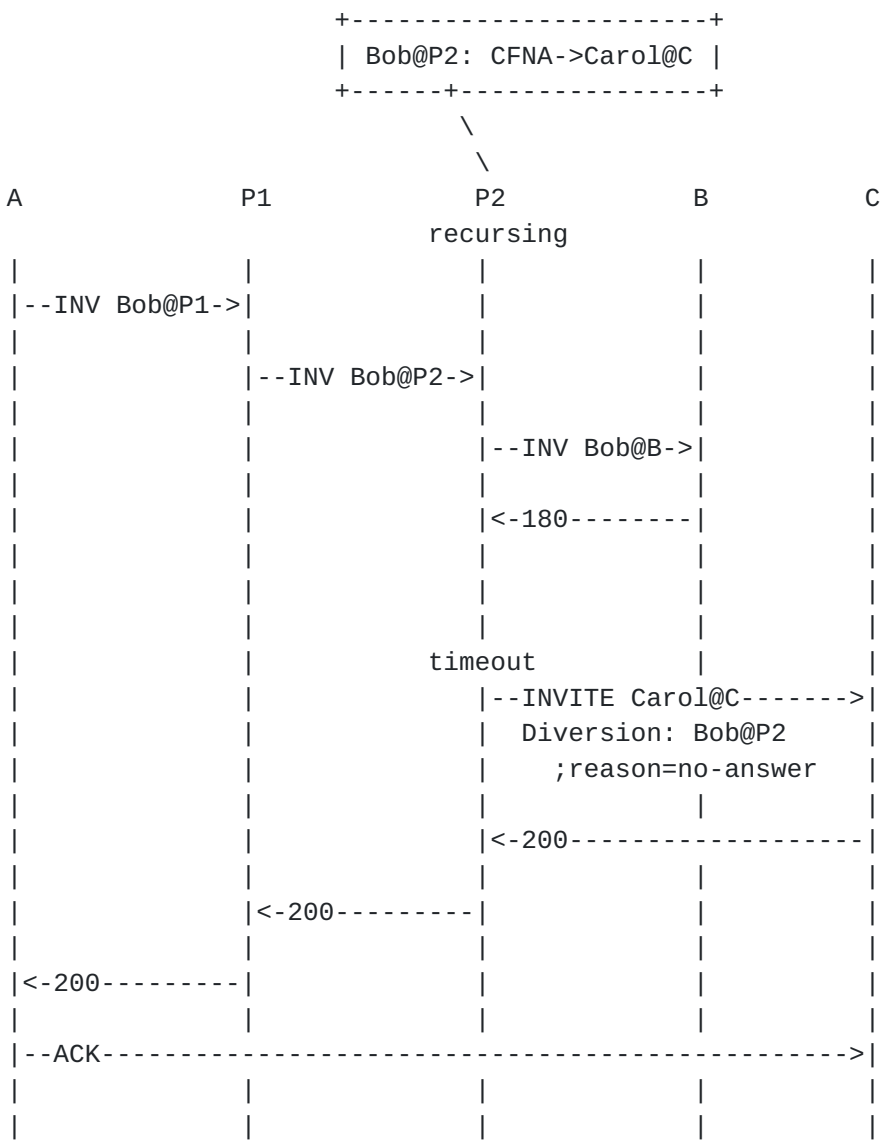


7.3. Call Forward on No-Answer

Usage of the Diversion header is shown below for several variant implementations of Call Forward on No-Answer.

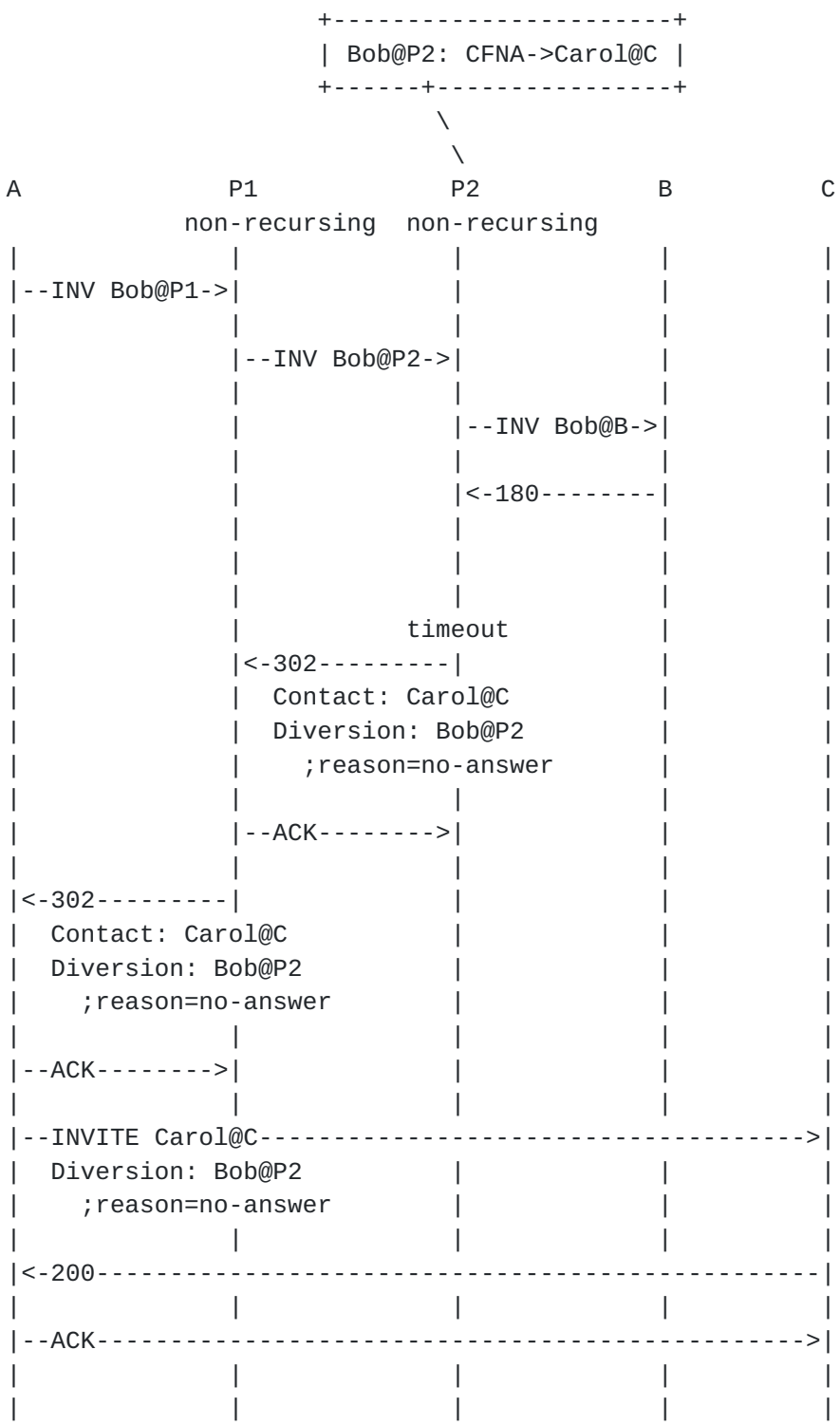
7.3.1. Network Call Forward on No-Answer, P2 recursing

In this message flow, Proxy 2 (P2) recursively implements Call Forward on No Answer (CFNA) to Carol@C.



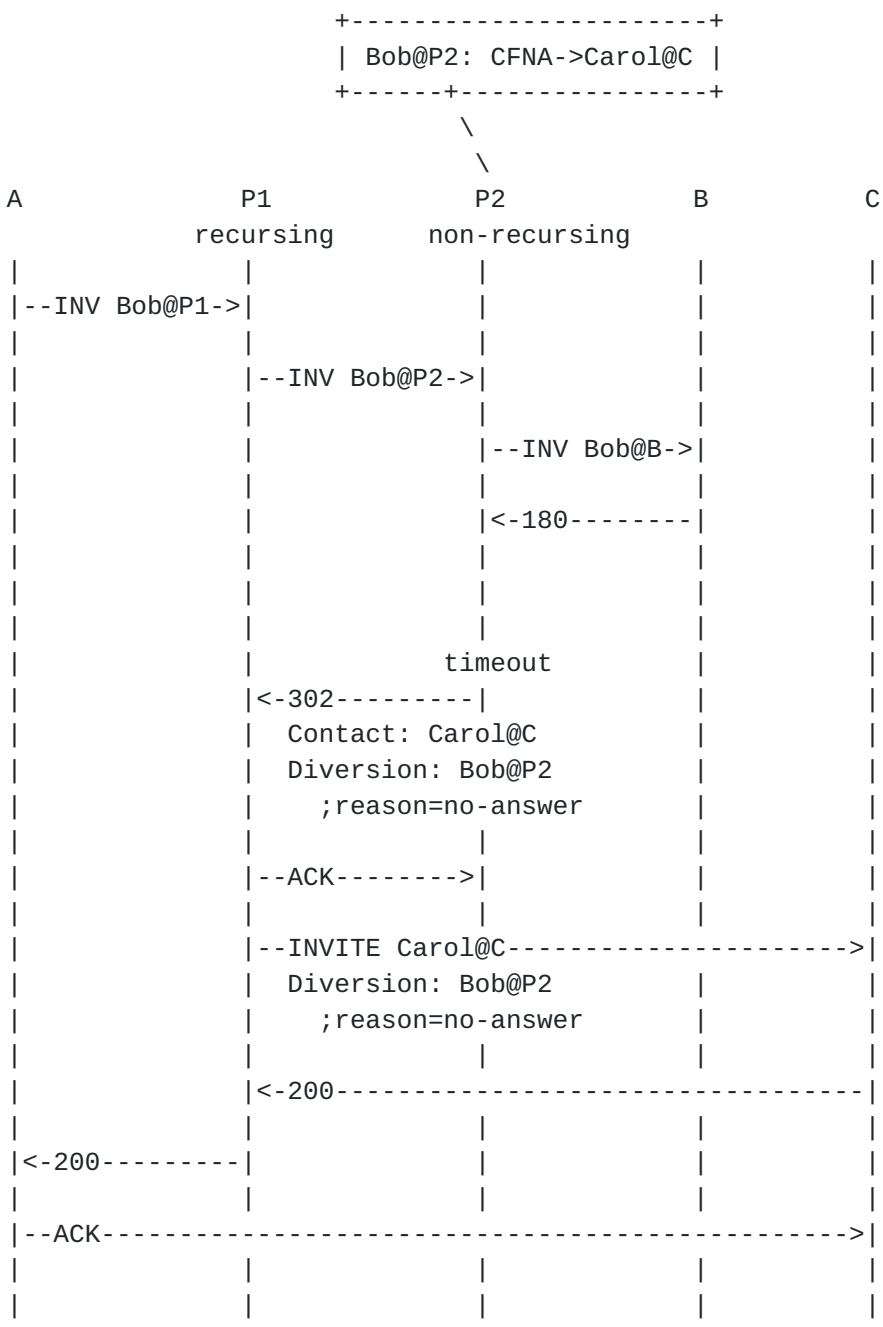
7.3.2. Network Call Forward on No-Answer, P1 non-recursing P2 non-recursing

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on No Answer (CFNA) to Carol@C. Proxy 1 (P1) is non-recursing.



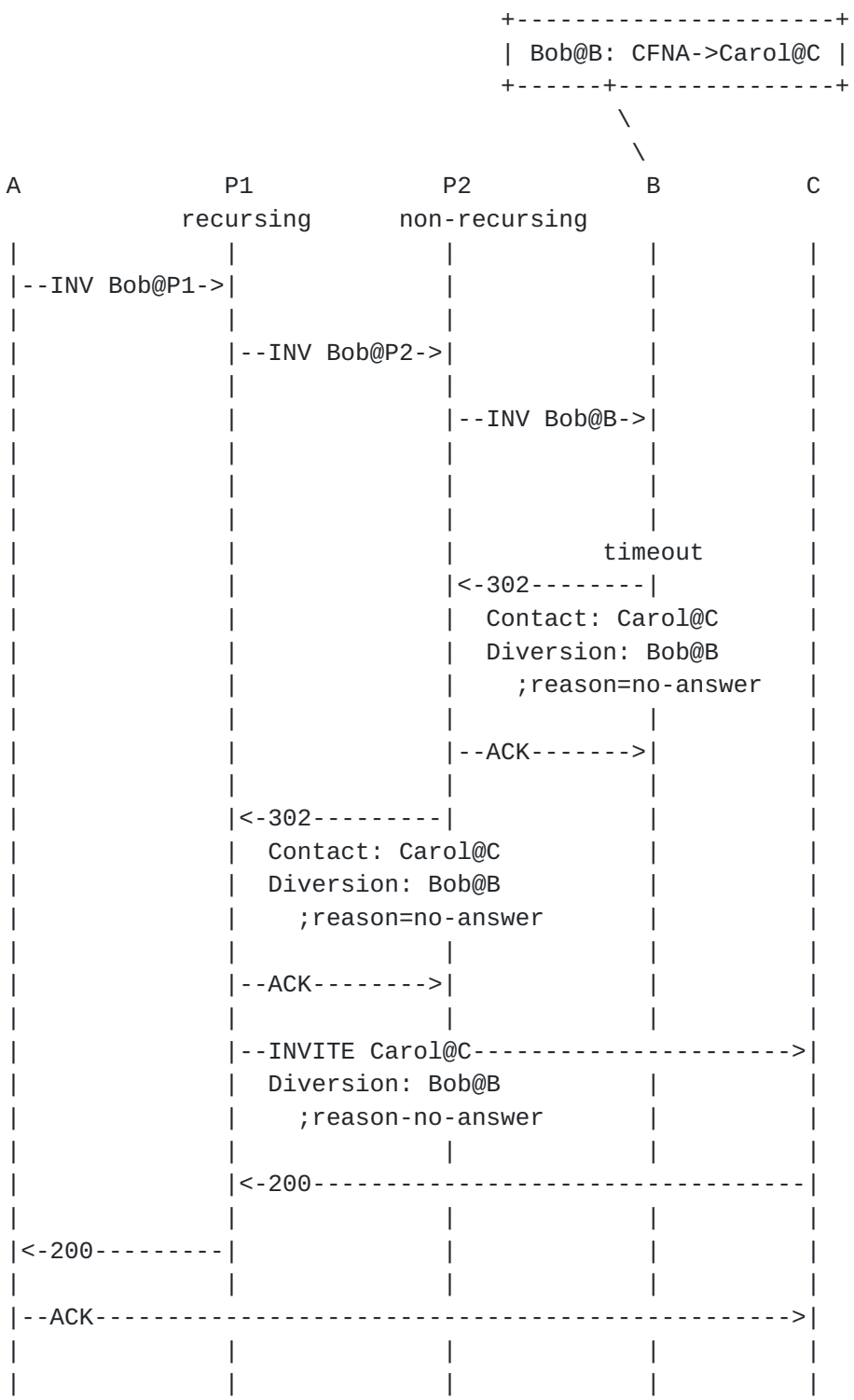
7.3.3. Network Call Forward on No Answer, P1 recursing P2 non-recursing

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on No Answer (CFNA) to Carol@C. Proxy 1 (P1) is recursing.



7.3.4. Endpoint Call Forward on No-Answer, P1 recursing, P2 non-recursing B non-recursing

In this message flow, user agent server B (B) non-recursively implements Call Forward on No Answer (CFNA) to Carol@C. Proxy 2 (P2) is non-recursing. Proxy 1 (P1) is recursing.

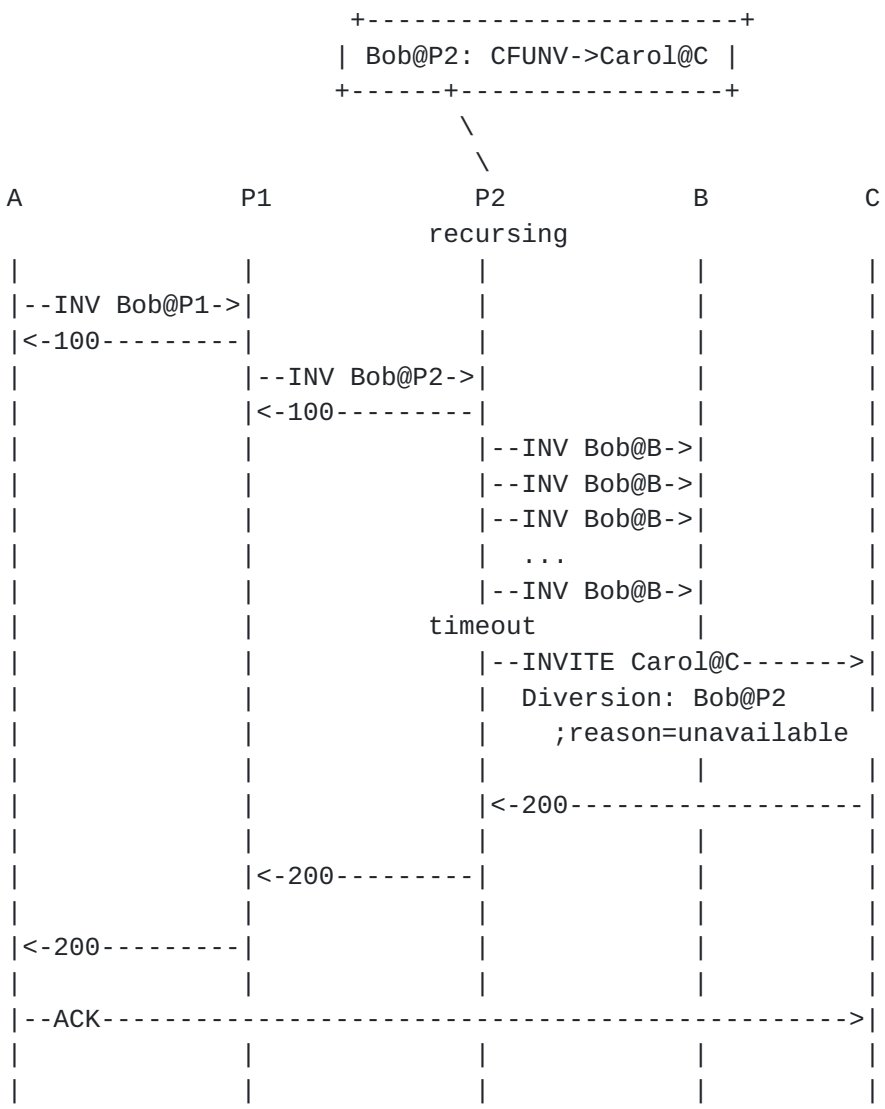


7.4. Call Forward on Unavailable

Usage of the Diversion header is shown below for several variant implementations of Call Forward on Unavailable.

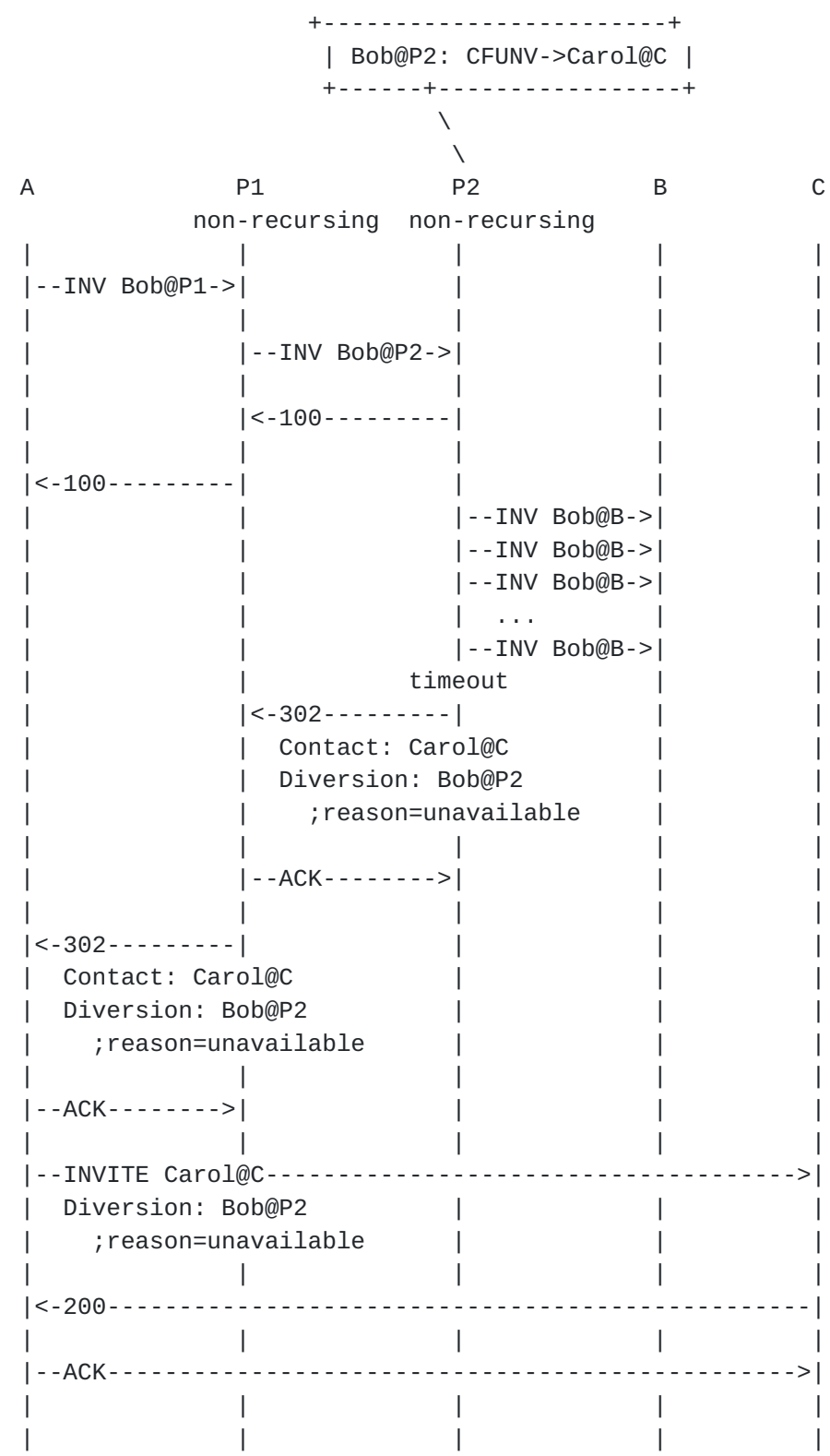
7.4.1. Network Call Forward on Unavailable, P2 recursing

In this message flow, Proxy 2 (P2) recursively implements Call Forward on Unavailable (CFUNV) to Carol@C.



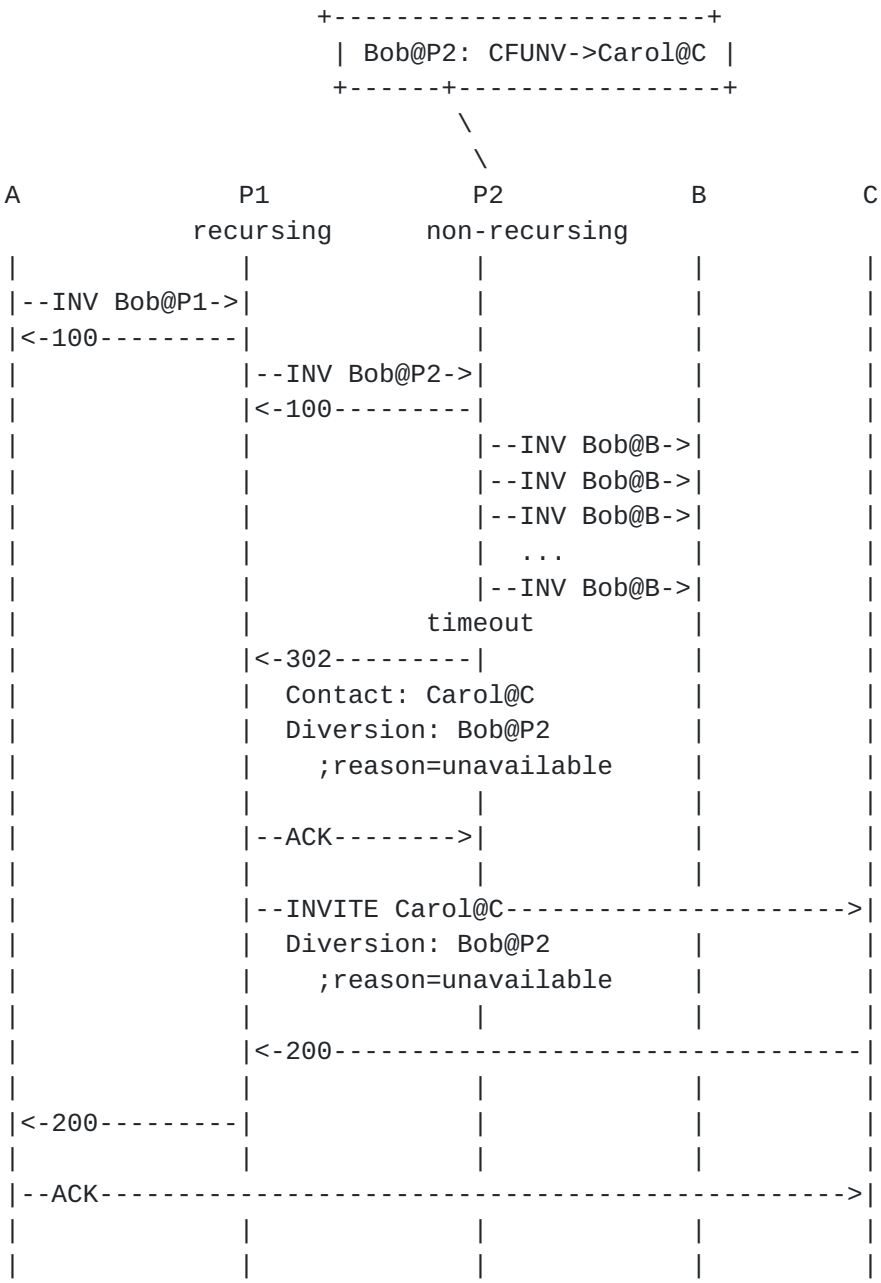
7.4.2. Network Call Forward on Unavailable, P1 non-recursing P2 non-recursing

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on Unavailable (CFUNV) to Carol@C. Proxy 1 (P1) is non-recursing.


+------------------------+
| Bob@P2: CFUNV->Carol@C |
+------+-----------------+
A P1 P2 B C
non-recursing non-recursing
--INV Bob@P1->
--INV Bob@P2->
<-100---------
<-100---------
--INV Bob@B->
--INV Bob@B->
--INV Bob@B->
...
--INV Bob@B->
timeout
<-302---------
Contact: Carol@C
Diversion: Bob@P2
;reason=unavailable
--ACK-------->
<-302---------
Contact: Carol@C
Diversion: Bob@P2
;reason=unavailable
--ACK-------->
--INVITE Carol@C
Diversion: Bob@P2
;reason=unavailable
<-200
--ACK

7.4.3. Network Call Forward on Unavailable, P1 recursing P2 non-recursing

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on Unavailable (CFUNV) to Carol@C. Proxy 1 (P1) is recursing.

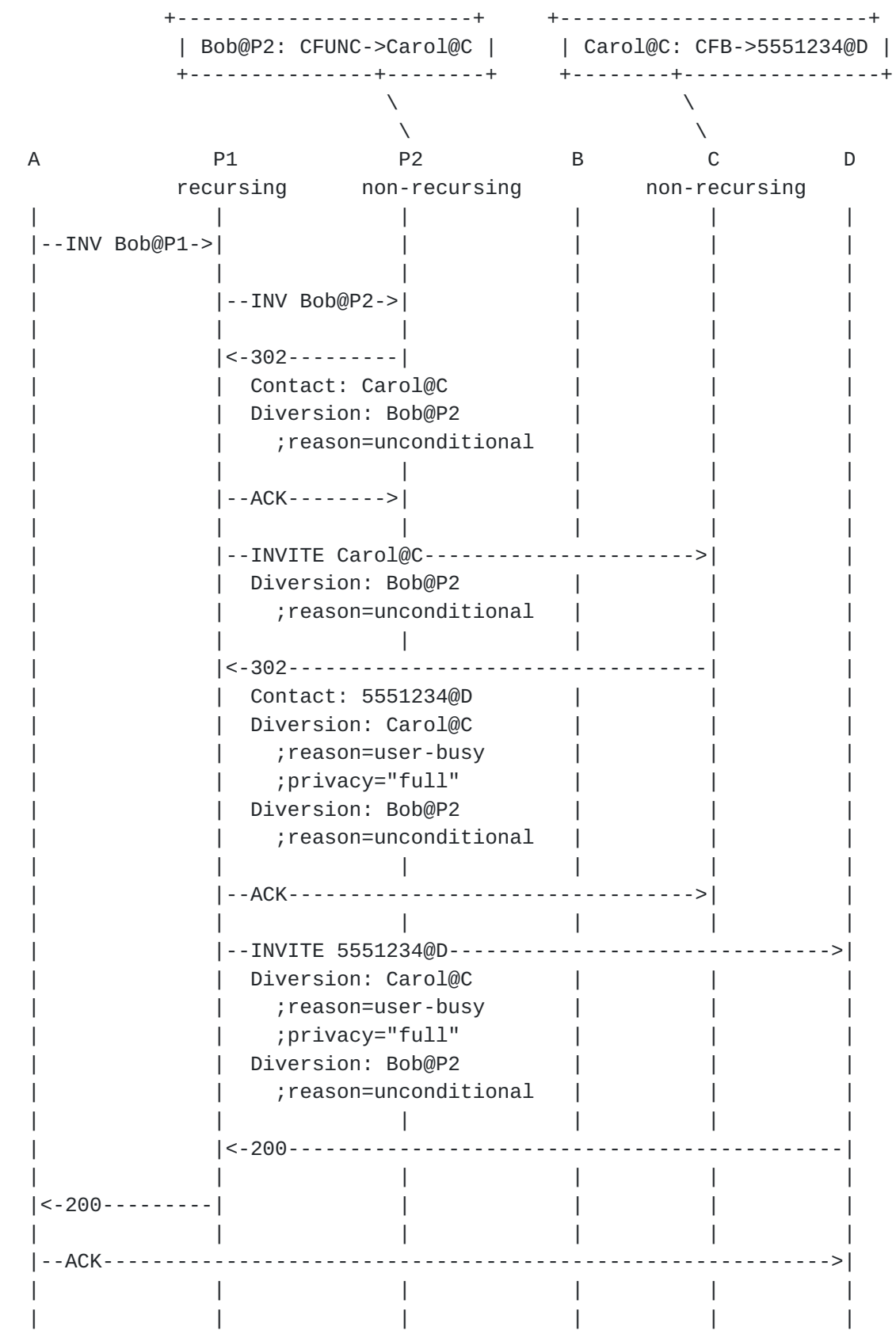


7.5. Multiple Diversions

Usage of the Diversion header when multiple diversions occur are shown the following two examples.

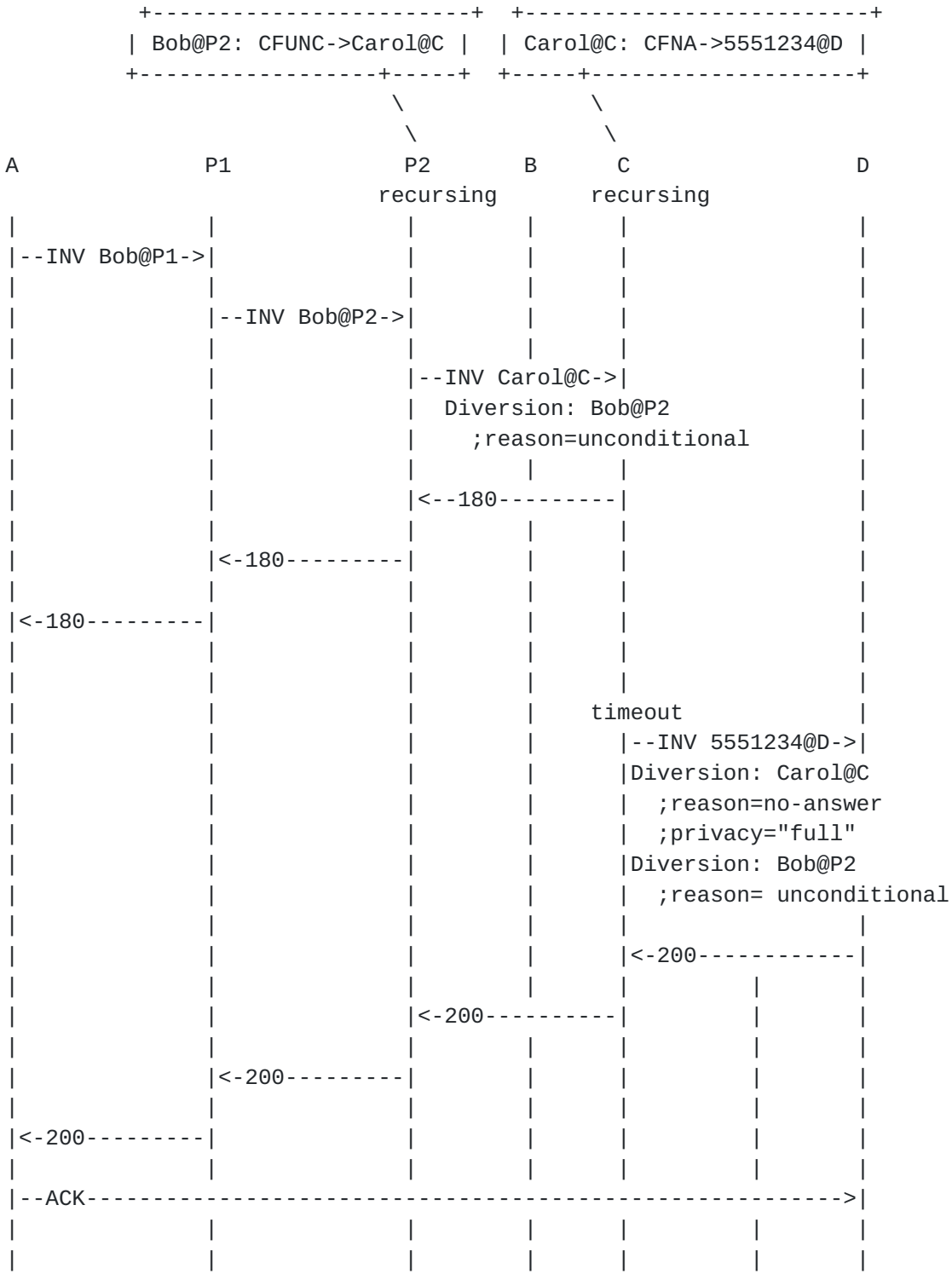
7.5.1. Call Forward Unconditional and Call Forward Busy

In this message flow, Proxy 2 (P2) implements Call Forward Unconditional (CFUNC) to Carol@C. C then implements Call Forward on Busy (CFB) to 5551234@D. P2 is non-recursing. P1 is recursing. C is non-recursing.


| Bob@P2: CFUNC->Carol@C |
| Carol@C: CFB->5551234@D |
A
P1
recursing
P2
non-recursing
B
C
non-recursing
D
--INV Bob@P1->
--INV Bob@P2->
<-302---------
Contact: Carol@C
Diversion: Bob@P2
;reason=unconditional
--ACK-------->
--INVITE Carol@C---------->
Diversion: Bob@P2
;reason=unconditional
<-302----------
Contact: 5551234@D
Diversion: Carol@C
;reason=user-busy
;privacy="full"
Diversion: Bob@P2
;reason=unconditional
--ACK---------->
--INVITE 5551234@D---------->
Diversion: Carol@C
;reason=user-busy
;privacy="full"
Diversion: Bob@P2
;reason=unconditional
<-200----------
<-200---------
--ACK---------->
Diversion: Bob@P2
;reason=unconditional

7.5.2. Call Forward Unconditional and Call Forward No Answer

In this message flow, Proxy 2 (P2) implements Call Forward Unconditional (CFUNC) to Carol@C. (P2 would normally have routed the call to B). C then implements Call Forward on No Answer (CFNA) to 5551234@D. P2 is recursing. C is recursing.



8. Security Considerations

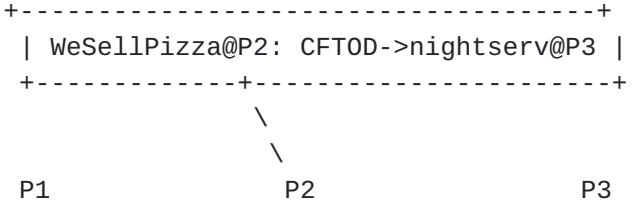
There are some privacy considerations when using the Diversion header.
Usage of the Diversion header implies that the diverting UAS trusts the diverted-to UAS.
Usage of the Diversion header by SIP proxies or SIP user agents can cause information leakage of route information and called information to untrusted SIP proxies and untrusted callers in upstream 3xx's.
Leakage of this information can be mitigated by having a recursing trusted upstream proxy server.
For a SIP network architecture where all proxies are required to be non-recursive, Diversion header hiding may be considered necessary in order to prevent leakage of route information to the caller. To accomplish Diversion header hiding, a trusted upstream proxy would add a Record-Route header and use a secret key to encrypt the contents of the Diversion header in 3xx's which are forwarded upstream. On receipt of re-INVITEs the proxy would decrypt the contents of the Diversion header (using its secret key) and forward the INVITE.
There is no currently defined interaction of the Diversion and Hide headers.
Question: Should there be?

9. Further Examples

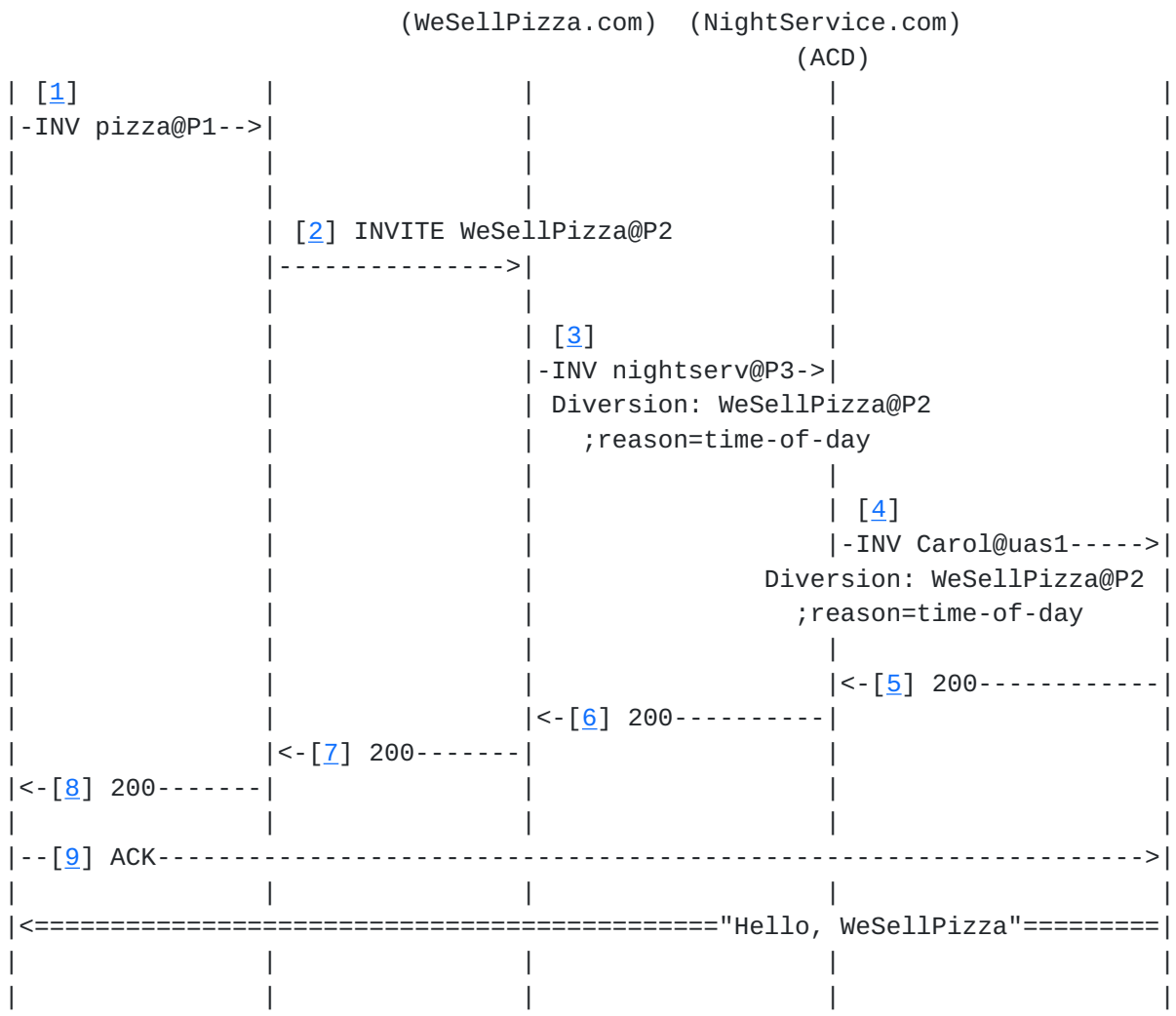
Only the relevant headers have been included in the following examples. The contents of the SDPs has also been omitted.

9.1. Night Service/Automatic Call Distribution (ACD) using Diversion header

In the following 2 message flows two separate companies, WeSellPizza.com and WeSellFlowers.com, have contracted with a third company, NightService.com to provide nighttime support for their incoming voice calls.
In the first flow, Alice calls out for pizza. In the second flow, Alice calls for roses. In both instances, the same night service company (and receptionist, Carol) answers the call. However, because the Diversion header is used, Carol is able to customize her greeting to the caller.



UAC P1 P2 P3 UAS1



Alice calls for pizza.

[1] SIP UAC to SIP proxy server 1:

```
INVITE sip:pizza@p1.isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:pizza@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

The ISP's originating proxy translated the keyword pizza to the company WeSellPizza.com

[2] SIP proxy server 1 to SIP proxy server 2 (WeSellPizza.com):

```
INVITE sip:WeSellPizza@p2.isp.com SIP/2.0
```

```
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:pizza@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

It's after midnight and the pizza people are in bed. Fortunately, WeSellPizza.com has contracted with NightService.com to answer their nighttime calls. Thus, P2 implements CFTOD to to NightService.com.

[3] SIP proxy server 2 (WeSellPizza.com) to SIP proxy server 3 (NightService.com):

```
INVITE sip:NightService@p3.isp.com SIP/2.0
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:pizza@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:WeSellPizza@p2.isp.com>
  ;reason=time-of-day
Content-Type: application/sdp
```

Carol is available to receive the incoming call.

[4] SIP proxy server 3 (NightService.com) to UAS1 (Carol):

```
INVITE sip:carol@uas1.nightservice.com SIP/2.0
Via: SIP/2.0/UDP p3.isp.com
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:pizza@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:WeSellPizza@p2.isp.com>
  ;reason=time-of-day
Content-Type: application/sdp
```

The ACD keys off the Diversion header to pull up the WeSellPizza FAQ on Carol's web browser.

[5] UAS1 to SIP proxy server 3:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p3.isp.com
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:pizza@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[6] SIP proxy server 3 to SIP proxy server 2:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:pizza@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[7] SIP proxy server 2 to SIP proxy server 1:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:pizza@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[8] SIP proxy server 1 to UAC

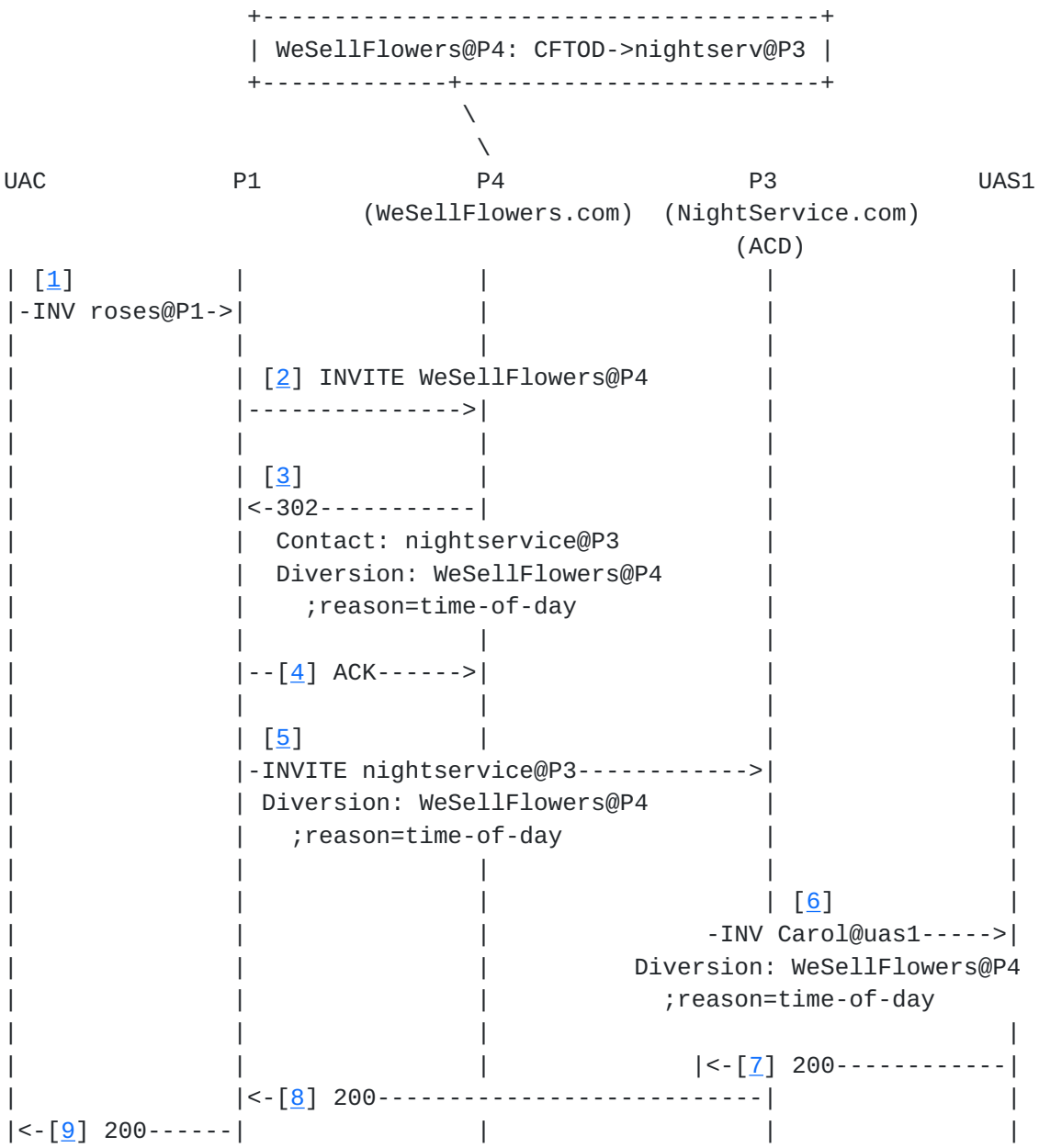
```
SIP/2.0 200 OK
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:pizza@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

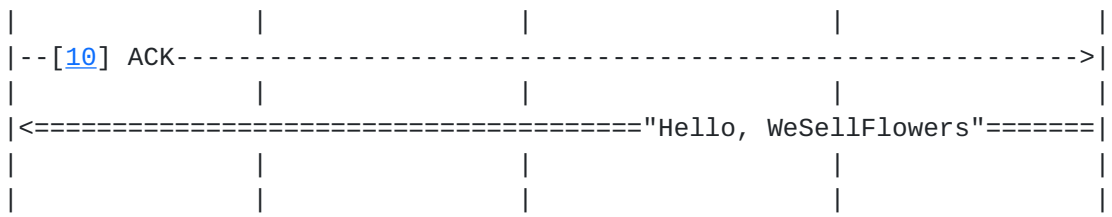
```
          Content-Type: application/sdp

     [9] SIP UAC to UAS1:

          ACK sip:uas1.nightservice.com SIP/2.0
          Via: SIP/2.0/UDP alice-pc.isp.com
          From: sip:alice@isp.com
          To: <sip:pizza@p1.isp.com>;tag=uas1
          Call-ID: 12345600@alice-pc.isp.com
          CSeq: 1 INVITE

     The RTP flows begin and Carol answers "Hello, WeSellPizza.
     How may I help you?"
```





Alice calls for roses.

[1] SIP UAC to SIP proxy server 1:

```
INVITE sip:roses@p1.isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:roses@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

The ISP's originating proxy translated the keyword roses to the company WeSellFlowers.com

[2] SIP proxy server 1 to SIP proxy server 4 (WeSellFlowers.com):

```
INVITE sip:WeSellFlowers@p4.isp.com SIP/2.0
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:roses@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

It's now 1am and the florists are also in bed. Fortunately, WeSellFlowers.com has contracted with NightService.com to answer their nighttime calls, too. Thus, P4 implements CFTOD to to NightService.com.

[3] SIP proxy server 4 (WeSellFlowers.com) to SIP proxy server 1 (NightService.com):

```
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: NightService@p3.isp.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=p4
Call-ID: 12345600@alice-pc.isp.com
```

```
          CSeq: 1 INVITE
          Diversion: <sip:WeSellFlowers@p4.isp.com>
            ;reason=time-of-day
```

[4] SIP proxy server 1 to SIP proxy server 4 (WeSellFlowers.com):

```
          ACK sip:uas1.nightservice.com SIP/2.0
          Via: SIP/2.0/UDP alice-pc.isp.com
          From: sip:alice@isp.com
          To: <sip:roses@p1.isp.com>;tag=p4
          Call-ID: 12345600@alice-pc.isp.com
          CSeq: 1 INVITE
```

[5] SIP proxy server 1 (WeSellFlowers.com) to
SIP proxy server 3 (NightService.com):

```
          INVITE sip:NightService@p3.isp.com SIP/2.0
          Via: SIP/2.0/UDP p1.isp.com
          Via: SIP/2.0/UDP alice-pc.isp.com
          From: sip:alice@isp.com
          To: sip:roses@p1.isp.com
          Call-ID: 12345600@alice-pc.isp.com
          CSeq: 1 INVITE
          Diversion: <sip:WeSellFlowers@p4.isp.com>
            ;reason=time-of-day
          Content-Type: application/sdp
```

Carol is available to receive the incoming call.

[6] SIP proxy server 3 (NightService.com) to UAS1 (Carol):

```
          INVITE sip:carol@uas1.nightservice.com SIP/2.0
          Via: SIP/2.0/UDP p3.isp.com
          Via: SIP/2.0/UDP p1.isp.com
          Via: SIP/2.0/UDP alice-pc.isp.com
          From: sip:alice@isp.com
          To: sip:roses@p1.isp.com
          Call-ID: 12345600@alice-pc.isp.com
          CSeq: 1 INVITE
          Diversion: <sip:WeSellFlowers@p4.isp.com>
            ;reason=time-of-day
          Content-Type: application/sdp
```

The ACD keys off the Diversion header to pull up
the WeSellFlowers FAQ on Carol's web browser.

[7] SIP UAS1 to SIP proxy server 3:

```
          SIP/2.0 200 OK
```

```
Via: SIP/2.0/UDP p3.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[8] SIP proxy server 3 to SIP proxy server 1:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[9] SIP proxy server 1 to UAC

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

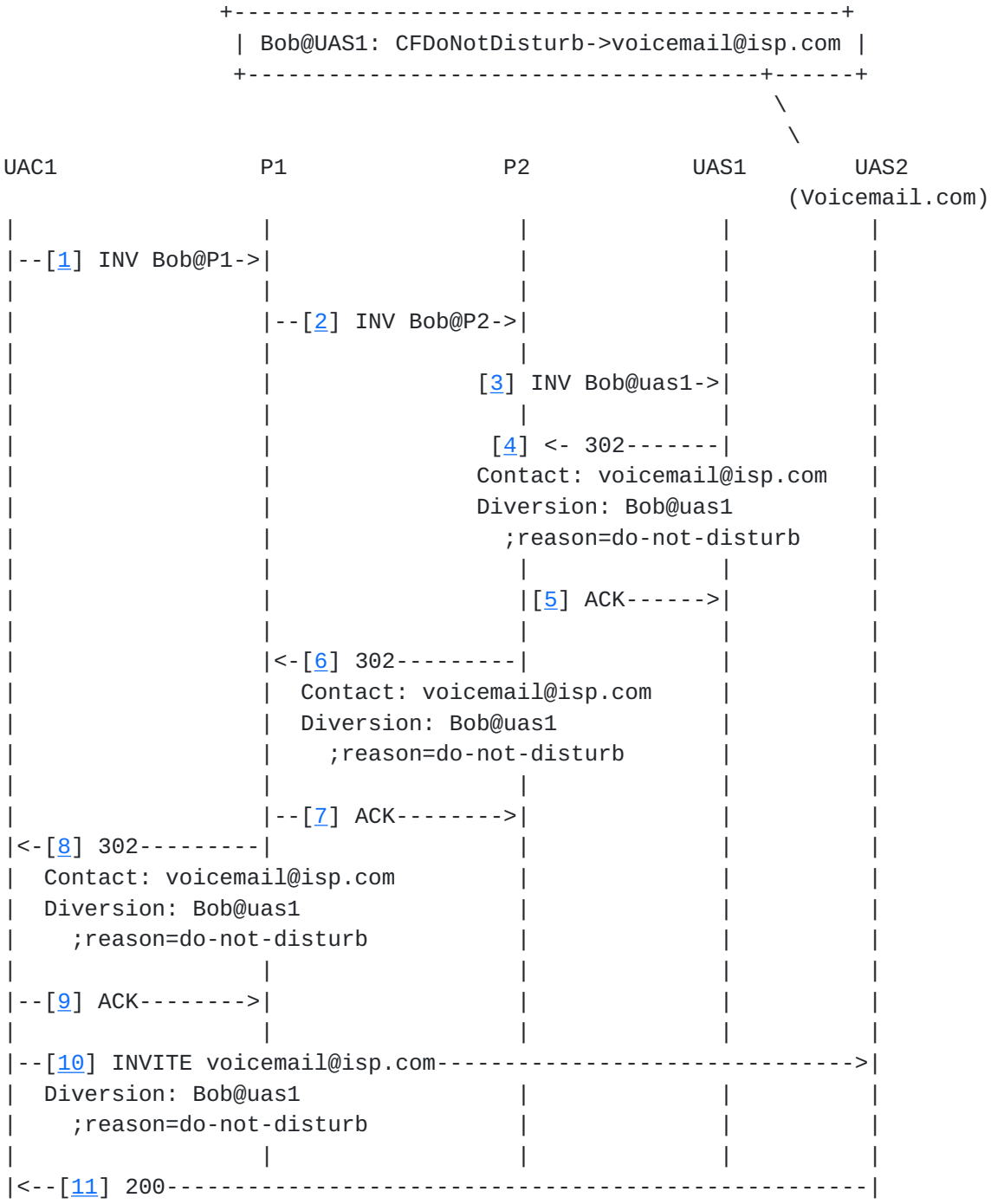
[10] SIP UAC to SIP UAS1:

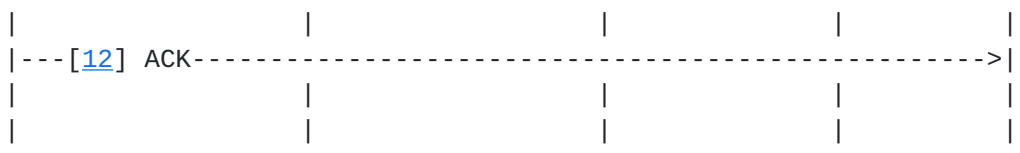
```
ACK sip:uas1.nightservice.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

The RTP flows begin and Carol answers "Hello, WeSellFlowers. How may I help you?"

9.2. Voicemail Service using Diversion header

Bob has contracted his Voicemail to a third-party company, Voicemail.com. In this message flow, Bob has hit the Do-Not-Disturb button on his phone. The Do-Not-Disturb functionality of Bob's phone is configured to CFUNC (Call Forward Unconditional) to voicemail@isp.com. Because the Diversion header is used, Voicemail.com is able to place the incoming call into Bob's voice mailbox.





Alice calls Bob.

[1] SIP UAC to SIP proxy server 1:

```
INVITE sip:Bob@p1.isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:Bob@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

The ISP's originating proxy routes the request to proxy 2 (P2)

[2] SIP proxy server 1 to SIP proxy server 2:

```
INVITE sip:Bob@p2.isp.com SIP/2.0
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:Bob@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[3] SIP proxy server 2 to UAS1 (Bob's SIP phone):

```
INVITE sip:Bob@uas1.isp.com SIP/2.0
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:Bob@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

Since Bob had hit the Do-Not-Disturb button on his SIP phone,
Bob's phone forwards the call to his voicemail service.

[4] User agent server 1 (UAS1) to SIP proxy server 2 (P2)

```
SIP/2.0 302 Moved Temporarily
```

```
          Via: SIP/2.0/UDP p2.isp.com
          Via: SIP/2.0/UDP p1.isp.com
          Via: SIP/2.0/UDP alice-pc.isp.com
          Contact: Voicemail@isp.com
          From: sip:alice@isp.com
          To: <sip:Bob@p1.isp.com>;tag=uas1
          Call-ID: 12345600@alice-pc.isp.com
          CSeq: 1 INVITE
          Diversion: <sip:Bob@uas1.isp.com>
            ;reason=do-not-disturb
```

[5] SIP proxy server 2 to UAS1 (Bob's SIP phone):

```
          ACK sip:Bob@uas1.isp.com SIP/2.0
          Via: SIP/2.0/UDP p2.isp.com
          From: sip:alice@isp.com
          To: <sip:Bob@p1.isp.com>;tag=uas1
          Call-ID: 12345600@alice-pc.isp.com
          CSeq: 1 INVITE
```

[6] SIP proxy server 2 (P2) to SIP proxy server 1 (P1):

```
          SIP/2.0 302 Moved Temporarily
          Via: SIP/2.0/UDP p1.isp.com
          Via: SIP/2.0/UDP alice-pc.isp.com
          Contact: Voicemail@isp.com
          From: sip:alice@isp.com
          To: <sip:Bob@p1.isp.com>;tag=uas1
          Call-ID: 12345600@alice-pc.isp.com
          CSeq: 1 INVITE
          Diversion: <sip:Bob@uas1.isp.com>
            ;reason=do-not-disturb
```

[7] SIP proxy server 1 to SIP proxy server 2:

```
          ACK sip:Bob@p2.isp.com SIP/2.0
          Via: SIP/2.0/UDP p1.isp.com
          From: sip:alice@isp.com
          To: <sip:Bob@p1.isp.com>;tag=uas1
          Call-ID: 12345600@alice-pc.isp.com
          CSeq: 1 INVITE
```

[8] SIP proxy server 1 (P1) to UAC (alice-pc):

```
          SIP/2.0 302 Moved Temporarily
          Via: SIP/2.0/UDP alice-pc.isp.com
          Contact: Voicemail@isp.com
          From: sip:alice@isp.com
          To: <sip:Bob@p1.isp.com>;tag=uas1
```

```
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:Bob@uas1.isp.com>
  ;reason=do-not-disturb
```

[9] SIP UAC to SIP proxy server 1:

```
ACK sip:Bob@p1.isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

[10] SIP UAC (alice-pc) to Voicemail server.

```
INVITE sip:Voicemail@isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:Bob@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:Bob@uas1.isp.com>
  ;reason=do-not-disturb
Content-Type: application/sdp
```

[11] Voicemail server to SIP UAC (alice-pc):

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: Voicemail@isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas2
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[12] SIP UAC to Voicemail server:

```
ACK sip:Voicemail@isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas2
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

Because the Diversion header is present, the Voicemail server is able to place Alice's message into Bob's voice

mailbox.

9.3. Q&A on alternative approaches

Question 1:
Why do we need the Diversion header when we can see the To: header?

Answer:
a) The To: header is not guaranteed to have significance to the called party.
For example, the To: header may contain a locally significant URL (to the caller) such as a private numbering plan, speed dial digits, telephony escape digits, or telephony prefix digits.
Without a Diversion header, enumerating all possible locally significant To: headers that anyone might use to contact Bob@uas1.isp.com becomes a configuration problem at Voicemail@isp.com and is prone to namespace collision.
Support for Diversion headers enables Bob to contract a third-party service (Voicemail@isp.com) with a single globally significant URL for his voice mailbox (Bob@uas1.isp.com).
b) Given a set of multiple diversions, there is a policy decision of which Diversion header takes precedence for service logic. Different services (or even different users for the same service) may want to configure this policy differently (first, last,second to last, etc).

Question 2:
Why do we need the Diversion header when we can see the Via: header?

Answer:
The Via header does not contain information about servers whom have deflected the call (using a 3xx).

10. Mapping ISUP/ISDN Redirection information to SIP Diversion header

The discussions below regarding ISUP/ISDN reflect generic elements in ISUP/ISDN. In some variations of ISUP/ISDN, the information elements are represented differently. Regardless of the ISUP/ISDN variant, translation should be performed for the "first redirecting number" and the "last redirecting number".
In order to prevent ambiguity, it is important to highlight a terminology mismatch between ISUP/ISDN and SIP. In SIP, a "redirect" indicates the act of returning a 3xx response. In ISUP/ISDN a "redirection" is diversion of a call by a network entity. In ISUP/ISDN a call may also be deflected (by an endpoint). Diversion is the more generic term which refers to either the act of an network redirection or endpoint deflection.

In SIP, Diversion can be implemented as either an upstream 3xx (non-recursive) or an additionally forked downstream request (recursive). In the following text, a lower-case "redirect" indicates the SIP usage, while an uppercase "Redirect" indicates ISUP usage.

10.1. Mapping ISUP/ISDN Diversion Reason codes

ISUP and ISDN define the following diversion reasons:

0000 = Unknown
0001 = Call forwarding busy or called DTE busy
0010 = Call forwarding no reply
1111 = Call forwarding unconditional or systematic call redirection
1010 = Call deflection or call forwarding by the called DTE
1001 = Call forwarding DTE out of order

Mapping of ISUP/ISDN reason codes to Diversion reason codes is performed as follows:

ISUP/ISDN reason code	Diversion reason code
0001	"user-busy"
0010	"no-answer"
1111	"unconditional"
1010	"deflection"
1001	"unavailable"
0000	all others

10.2. Mapping ISUP Redirection information to SIP Diversion header

This section describes how generic ISUP diversion information elements may be translated across an ISUP/SIP gateway.

10.2.1. ISUP Definitions

Called party number	**The number of the party to which the call is currently being routed.**
Redirecting number	The number to which the call was being routed when the last diversion occurred.
Redirecting reason	The reason that the last diversion occurred.
Original called number	The number to which the call was being routed when the first diversion occurred.
Original redirecting reason	The reason that the first diversion occurred.
Redirection counter	The count of the total number of diversions that have occurred.
Address Presentation	Indication of whether presentation is allowed or restricted.

10.2.2. ISUP parameters

When a SIP call transits a SIP/ISUP gateway, the following information in the ISUP message should be examined/set when translating SIP Diversion headers to ISUP diversion information:

1) Redirecting Number
2) Redirecting reason
3) Redirecting address presentation
4) Original called number
5) Original redirecting reason
6) Original address presentation
7) Redirection counter.

An ISUP message contains information on the first and last diversions that occurred. The Redirection Number is to which the call was being routed when the last diversion occurred. The Redirecting Reason is the reason that the last diversion occurred.
The Original Called Number is the number to which the call was being routed when the first diversion occurred. The Original Redirecting Reason is the reason that the first diversion occurred.
When only one Diversion has occurred, the number to which the call was being routed when the diversion occurred is in the Redirecting

Number and the reason for that diversion is carried in the Redirect Reason.

10.2.3. ISUP to SIP translation

The ISUP Redirecting Number SHOULD be used to set the value of the name-addr of the top-most Diversion header. The ISUP Redirecting Number address presentation SHOULD be used to set the value of the diversion-privacy of the top-most Diversion header. The ISUP Redirecting Reason SHOULD be used to set the value of the diversion-reason of the top-most Diversion header. When present, the Original Called Number SHOULD be used to set the name-addr of the bottom-most Diversion header. When present, the Original Redirecting Reason SHOULD be used to set the diversion-reason of the bottom-most Diversion header.When present, the Original address presentation SHOULD be used to set the diversion-privacy of the bottom-most Diversion header.
The Redirect Counter value minus 1 SHOULD be stored in the diversion-counter associated with the top-most Diversion header. Presence of the diversion-counter for the bottom-most Diversion header is optional. If present the diversion-counter of the bottom-most Diversion header SHOULD be 1.

10.2.4. SIP to ISUP translation

The name-addr of the top-most Diversion header SHOULD be used to set the ISUP Redirecting Number. The diversion-reason of the top-most Diversion header SHOULD be used to set the ISUP Redirecting Reason.The diversion-privacy of the top-most Diversion header SHOULD be used to set the ISUP Redirecting Address Presentation. When multiple Diversion headers are present, the name-addr of the bottom-most Diversion header SHOULD be used to set the ISUP Original Redirecting Number.
When multiple Diversion headers are present, the diversion-reason of the bottom-most Diversion header SHOULD be used to set the ISUP Original Redirecting Reason. When multiple Diversion headers are present, the diversion-privacy of the bottom-most Diversion header SHOULD be used to set the ISUP Original Redirecting Address Presentation.
The ISUP Redirection counter SHOULD be set equal to the sum of the counters of all Diversion headers in the SIP message. A Diversion header which does not explicitly specify a diversion-counter tag counts as 1.

10.2.5. Example of ISUP to SIP translation

```
                                    ISUP/SIP GW
                                        |
 --IAM--------------------------------->|
   Called Party Number    =+19195551004 |
   Redirecting Number     =+19195551002 |
     Address Presentation =presentation restricted
   Original Called Number =+19195551001 |
   RedirectionInformation:              |
     Original redirecting reason = Unconditional (1111)
     Redirecting Reason = User busy (0001)
     Redirection counter = 5            |
                                        |
                                        |--INVITE +19195551004------>
                                        |  Diversion: <tel:+19195551002>
                                        |    ;reason=user-busy
                                        |    ;privacy="full"
                                        |    ;counter=4
                                        |  Diversion: <tel:+19195551001>
                                        |    ;reason=unconditional
                                        |    ;counter=1
                                        |
                                        |
```

10.2.6. Example of SIP to ISUP translation

```
                                    ISUP/SIP GW
                                        |
                                        |<--INVITE +19195551004------
                                        |  Diversion: <tel:+19195551002>
                                        |    ;reason=user-busy
                                        |    ;privacy="full"
                                        |    ;counter=4
                                        |  Diversion: <tel:+19195551001>
                                        |    ;reason=unconditional
                                        |    ;counter=1
                                        |
                                        |
                                        |
 <--IAM---------------------------------|
   Called Party Number    =+19195551004 |
   Redirecting Number     =+19195551002 |
     Address Presentation =presentation restricted
   Original Called Number =+19195551001 |
   RedirectionInformation:              |
     Original redirecting reason = Unconditional (1111)
     Redirecting Reason = User busy (0001)
     Redirection counter = 5            |
```

10.3. Mapping ISDN Redirection information to SIP Diversion header

An ISDN message can contain up to two instances of a Redirecting number information element. When a diversion occurs, an additional Redirection number information element is added. When a third (or greater) diversion occurs, the new Redirecting number information element replaces the bottom-most Redirection number information element.

10.3.1. ISDN Definitions

Called party number	**The number of the party to which the** call is currently being routed.
Redirecting number information element	Aggregate information element which contains Redirecting number and Reason for diversion.
Redirecting number	The number to which the call was being routed when the last diversion occurred.
Reason for diversion	The reason that the last diversion occurred.
Origin of Number	Indicates whether number is user provided and screened or network provided
Presentation Status	Indicates if presentation is allowed or prohibited

10.3.2. ISDN parameters

When a SIP call transits a SIP/ISDN gateway, the following information in the ISDN message should be examined/set when translating SIP Diversion headers to ISDN diversion information:

1) Redirecting number of the top-most Redirecting number information element
2) Reason for diversion of the top-most Redirection number information element
3) Origin of Number and Presentation Status of the top-most Redirection number information element
4) Redirection number of the bottom-most Redirection number information element
5) Reason for diversion of the bottom-most Redirection number information element.
6) Origin of Number and Presentation Status of the bottom-most Redirection number information element

An ISDN message contains information on the first and last diversions that occurred. The top-most Redirection number information element contains information (including the Redirecting number, Origin of Number, Presentation Status, and Reason for diversion) about the last diversion that occurred. The bottom-most Redirection number information element contains information (including the Redirecting number, Origin of Number, Presentation Status,and Reason for diversion) about the first diversion that occurred.
If only one Diversion has occurred, only one Redirection number information element is present.
The Redirecting Number information element has the same Type of Number/Numbering Plan, and Digits as the Calling Party Number information element.
There is no Redirection counter associated with this ISDN information element.
Notice that the order of the Redirection number information elements in an ISDN message (top=first, bottom=last) is reversed from the order of Diversion headers in a SIP message (top=last, bottom=first).

10.3.3. ISDN to SIP translation

The Redirecting number of the top-most ISDN Redirecting number information element SHOULD be used to set the value of the name-addr of the bottom-most Diversion header. The Reason for Diversion of the top-most ISDN Redirecting number information element SHOULD be used to set the value of the diversion-reason of the bottom-most Diversion header.
The Origin of Number of the top-most ISDN Redirecting number information element SHOULD be used to set the value of the diversion-screen of the bottom-most Diversion header. The Presentation Status of the top-most ISDN Redirecting number information element SHOULD be used to set the value of the diversion-privacy of the bottom-most Diversion header.

The Redirecting number of the bottom-most ISDN Redirecting number information element SHOULD be used to set the value of the name-addr

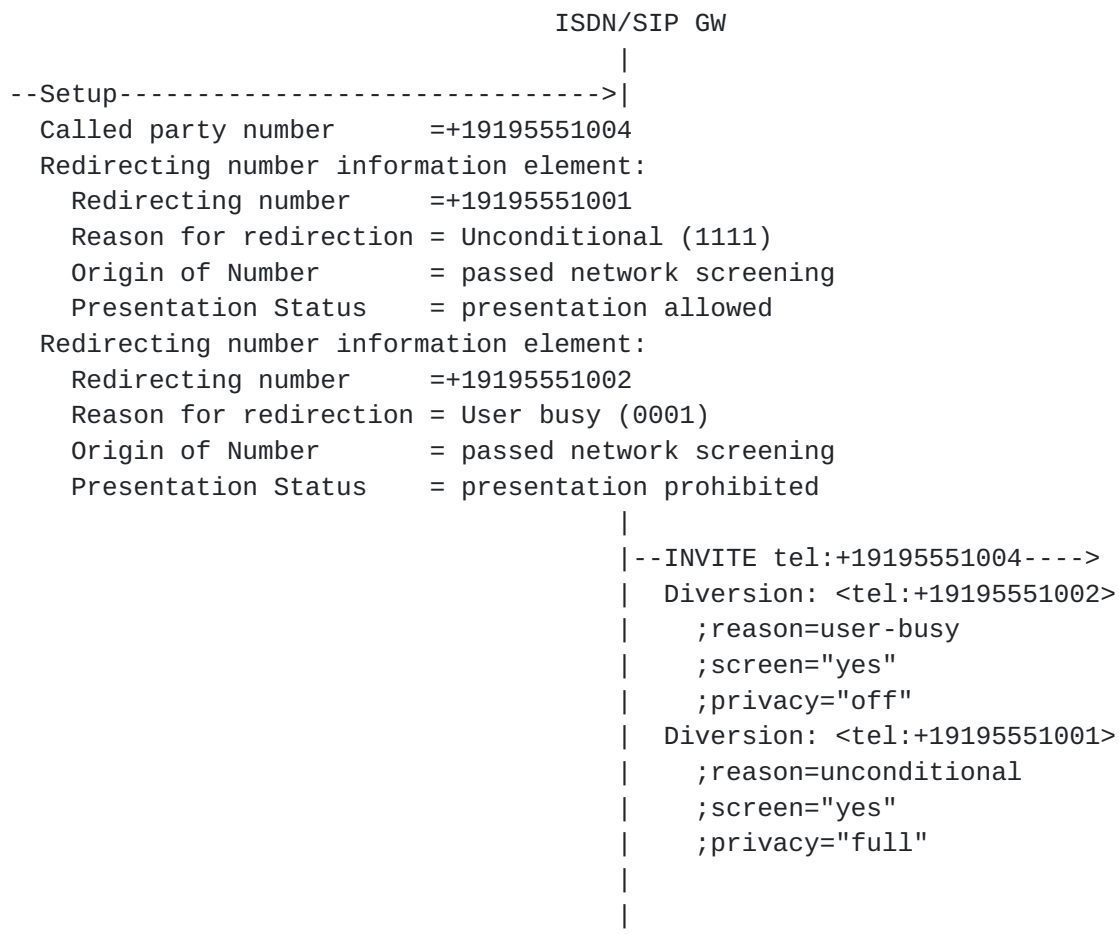
of the top-most Diversion header. The Reason for Diversion of the bottom-most ISDN Redirecting number information element SHOULD be used to set the value of the diversion-reason of the top-most Diversion header.
The Origin of Number of the bottom-most ISDN Redirecting number information element SHOULD be used to set the value of the diversion-screen of the top-most Diversion header. The Presentation Status of the bottom-most ISDN Redirecting number information element SHOULD be used to set the value of the diversion-privacy of the top-most Diversion header.
Presence of the diversion-counter in each of the Diversion headers is optional. If present, the diversion-counter of each Diversion header SHOULD be 1.

10.3.4. SIP to ISDN translation

The name-addr of the top-most Diversion header SHOULD be used to set the Redirecting number of the bottom-most ISDN Redirecting number information element.
The diversion-reason of the top-most Diversion header SHOULD be used to set the Reason for Diversion of the bottom-most ISDN Redirecting number information element.
The diversion-screen of the top-most Diversion header SHOULD be used to set the Origin of Number of the bottom-most ISDN Redirecting number information element.
The diversion-privacy of the top-most Diversion header SHOULD be used to set the Presentation Status of the bottom-most ISDN Redirecting number information element.

The name-addr of the bottom-most Diversion header SHOULD be used to set the Redirecting number of the top-most ISDN Redirecting number information element.
The diversion-reason of the bottom-most Diversion header SHOULD be used to set the Reason for Diversion of the top-most ISDN Redirecting number information element.
The diversion-screen of the bottom-most Diversion header SHOULD be used to set the Origin of Number of the top-most ISDN Redirecting number information element.
The diversion-privacy of the bottom-most Diversion header SHOULD be used to set the Presentation Status of the top-most ISDN Redirecting number information element.

10.3.5. Example of ISDN to SIP translation



10.3.6. Example of SIP to ISDN translation

```
                                    ISDN/SIP GW
                                        |
 <--Setup-------------------------------|
   Called party number      =+19195551004
   Redirecting number information element:
     Redirecting number     =+19195551001
     Reason for redirection = Unconditional (1111)
     Origin of Number       = passed network screening
     Presentation Status    = presentation allowed
   Redirecting number information element:
     Redirecting number     =+19195551002
     Reason for redirection = User busy (0001)
     Origin of Number       = passed network screening
     Presentation Status    = presentation prhibited
                                        |
                                        |<--INVITE tel:+19195551004----
                                        |  Diversion: <tel:+19195551002>
                                        |    ;reason=user-busy
                                        |    ;screen="yes"
                                        |    ;privacy="off
                                        |  Diversion: <tel:+19195551001>
                                        |    ;reason=unconditional
                                        |    ;screen="yes"
                                        |    ;privacy="full"
                                        |
```

10.4. Information loss in SIP to ISUP/ISDN translation

Because ISUP and ISDN only support a subset of the information in a SIP Diversion header, information loss occurs during translation at a SIP/ISUP or SIP/ISDN boundary.

10.4.1. Loss of diversion URI information

Because ISUP and ISDN only support a subset of URI types, (specifically tel: URIs and sip:x@y;user=phone URIs) Diversion information occurring for other URI types may be lost when crossing from SIP to ISDN or ISUP.

10.4.2. Loss of diversion reason information

Because ISUP and ISDN only support a subset of the reason codes supported by the Diversion header, specific reason code information may be lost when crossing from SIP to ISDN or ISUP.

10.4.3. Loss of diversion counter information

Because ISDN does not support a counter field (indicating the number of diversions that have occurred), counter information may be lost when crossing from SIP to ISDN.

11. Contributors

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13. Normative References

[RFC2119] "Key words for use in RFCs to Indicate Requirement Levels", RFC 2119, March 1997.

[RFC3261] "SIP: Session Initiation Protocol", RFC 3261, June 2002.

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