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Session Initiation Protocol (SIP) Call Control - Transfer

Status of This Memo

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Abstract

This document describes providing Call Transfer capabilities in the Session Initiation Protocol (SIP). SIP extensions such as REFER and Replaces are used to provide a number of transfer services including blind transfer, consultative transfer, and attended transfer. This work is part of the SIP multiparty call control framework.

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1. Overview

This document describes providing Call Transfer capabilities and requirements in SIP [RFC3261]. This work is part of the multiparty call control framework [CC-FRMWRK].

The mechanisms discussed here are most closely related to traditional, basic, and consultation hold transfers.

This document details the use of the REFER method [RFC3515] and Replaces [RFC3891] header field to achieve call transfer.

A User Agent (UA) that fully supports the transfer mechanisms described in this document supports REFER [RFC3515] and Replaces [RFC3891] in addition to RFC 3261 [RFC3261]. A User Agent should use a Contact URI that meets the requirements in Section 8.1.1.8 of RFC 3261. A compliant User Agent supports the Target-Dialog header field [RFC4538].

2. Actors and Roles

There are three actors in a given transfer event, each playing one of the following roles:

Transferee: the party being transferred to the Transfer Target.

Transferor: the party initiating the transfer.

Transfer Target: the new party being introduced into a call with the Transferee.

The following roles are used to describe transfer requirements and scenarios:

Originator: wishes to place a call to the Recipient. This actor is the source of the first INVITE in a session, to either a Facilitator or a Screener.

Facilitator: receives a call or out-of-band request from the Originator, establishes a call to the Recipient through the Screener, and connects the Originator to the Recipient. Typically, a Facilitator acts on behalf of the Originator.

Screener: receives a call ultimately intended for the Recipient and transfers the calling party to the Recipient if appropriate. Typically, a Screener acts on behalf of the Recipient.

Recipient: the party to which the Originator is ultimately connected.

3. Terminology

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 BCP 14, RFC 2119 [RFC2119].

4. Requirements

1. Any party in a SIP session must be able to transfer any other party in that session at any point in that session.

2. The Transferor and the Transferee must not be removed from a session as part of a transfer transaction.

   At first glance, requirement 2 may seem to indicate that the user experience in a transfer must be significantly different from what a current Private Branch Exchange (PBX) or Centrex user expects. As the call flows in this document show, this is not the case. A client may preserve the current experience. In fact, without this requirement, some forms of the current experience (ringback on transfer failure, for instance) will be lost.

3. The Transferor must know whether or not the transfer was successful.

4. The Transferee must be able to replace an existing dialog with a new dialog.

5. The Transferor and Transferee should indicate their support for the primitives required to achieve transfer.

6. The Transferor should provide the Transfer Target and Transferee with information about the nature and progress of the transfer operation being attempted.

To meet this requirement, the transfer operation can be modeled as an ad hoc conference between three parties, as discussed in Section 9.

5. Using REFER to Achieve Call Transfer

A REFER [RFC3515] can be issued by the Transferor to cause the Transferee to issue an INVITE to the Transfer Target. Note that a successful REFER transaction does not terminate the session between the Transferor and the Transferee. If those parties wish to terminate their session, they must do so with a subsequent BYE request. The media negotiated between the transferee and the Transfer Target is not affected by the media that had been negotiated between the Transferor and the Transferee. In particular, the INVITE issued by the Transferee will have the same Session Description Protocol (SDP) body it would have if the Transferee had initiated that INVITE on its own. Further, the disposition of the media streams between the Transferor and the Transferee is not altered by the REFER method.

Agents may alter a session's media through additional signaling. For example, they may make use of the SIP hold re-INVITE [RFC3261] or conferencing extensions described in the conferencing framework [RFC4353].

To perform the transfer, the Transferor and Transferee could reuse an existing dialog established by an INVITE to send the REFER. This would result in a single dialog shared by two uses -- an invite usage and a subscription usage. The call flows for this are shown in detail in Section 6.2. However, the approach described in this document is to avoid dialog reuse. The issues and difficulties associated with dialog reuse are described in [RFC5057].

Motivations for reusing the existing dialog include:

1. There was no way to ensure that a REFER on a new dialog would reach the particular endpoint involved in a transfer. Many factors, including details of implementations and changes in proxy routing between an INVITE and a REFER could cause the REFER to be sent to the wrong place. Sending the REFER down the existing dialog ensured it got to the endpoint to which we were already talking.

2. It was unclear how to associate an existing invite usage with a REFER arriving on a new dialog, where it was completely obvious what the association was when the REFER came on the INVITE usage's dialog.

3. There were concerns with authorizing out-of-dialog REFERs. The authorization policy for REFER in most implementations piggybacks on the authorization policy for INVITE (which is, in most cases, based simply on "I placed or answered this call").

Globally Routable UA URIs (GRUUs) [SIP-GRUU] can be used to address problem 1. Problem 2 can be addressed using the Target-Dialog header field defined in [RFC4538]. In the immediate term, this solution to problem 2 allows the existing REFER authorization policy to be reused.

As a result, if the Transferee supports the target-dialog extension and the Transferor knows the Contact URI is routable outside the dialog, the REFER SHOULD be sent in a new dialog. If the nature of the Contact URI is not known or if support for the target-dialog extension is not known, the REFER SHOULD be sent inside the existing dialog. A Transferee MUST be prepared to receive a REFER either inside or outside a dialog. One way that a Transferor could know that a Contact URI is routable outside a dialog is by validation (e.g., sending an OPTIONS and receiving a response) or if it satisfies the properties described in the GRUU specification [SIP-GRUU].

This document does not prescribe the flows and examples precisely as they are shown, but rather the flows illustrate the principles for best practice for the transfer feature. The call flows represent well-reviewed examples of SIP usage to implement transfer with REFER, which are Best Common Practice according to IETF consensus.

In most of the following examples, the Transferor is in the atlanta.example.com domain, the Transferee is in the biloxi.example.com, and the Transfer Target is in the chicago.example.com domain.

6. Basic Transfer

Basic Transfer consists of the Transferor providing the Transfer Target’s contact to the Transferee. The Transferee attempts to establish a session using that contact and reports the results of that attempt to the Transferor. The signaling relationship between the Transferor and Transferee is not terminated, so the call is recoverable if the Transfer Target cannot be reached. Note that the Transfer Target’s contact information has been exposed to the Transferee. The provided contact can be used to make new calls in the future.

The participants in a basic transfer SHOULD indicate support for the REFER and NOTIFY methods in Allow header fields in INVITE, 200 OK to INVITE, and OPTIONS messages. Participants SHOULD also indicate support for Target-Dialog in the Supported header field.

The diagrams below show the first line of each message. The first column of the figure shows the dialog used in that particular message. In these diagrams, media is managed through re-INVITE holds, but other mechanisms (mixing multiple media streams at the UA or using the conferencing extensions, for example) are valid. Selected message details are shown labeled as message F1, F2, etc.

Each of the flows below shows the dialog between the Transferor and the Transferee remaining connected (on hold) during the REFER process. While this provides the greatest flexibility for recovery from failure, it is not necessary. If the Transferor's agent does not wish to participate in the remainder of the REFER process and has no intention of assisting with recovery from transfer failure, it could emit a BYE to the Transferee as soon as the REFER transaction completes. This flow is sometimes known as "unattended transfer" or "blind transfer".

Figure 1 shows transfer when the Transferee utilizes a GRUU and supports the target-dialog extension and indicates this to the Transferor. As a result, the Transferor sends the REFER outside the INVITE dialog. The Transferee is able to match this REFER to the existing dialog using the Target-Dialog header field in the refer which references the existing dialog.

6.1. Successful Transfer

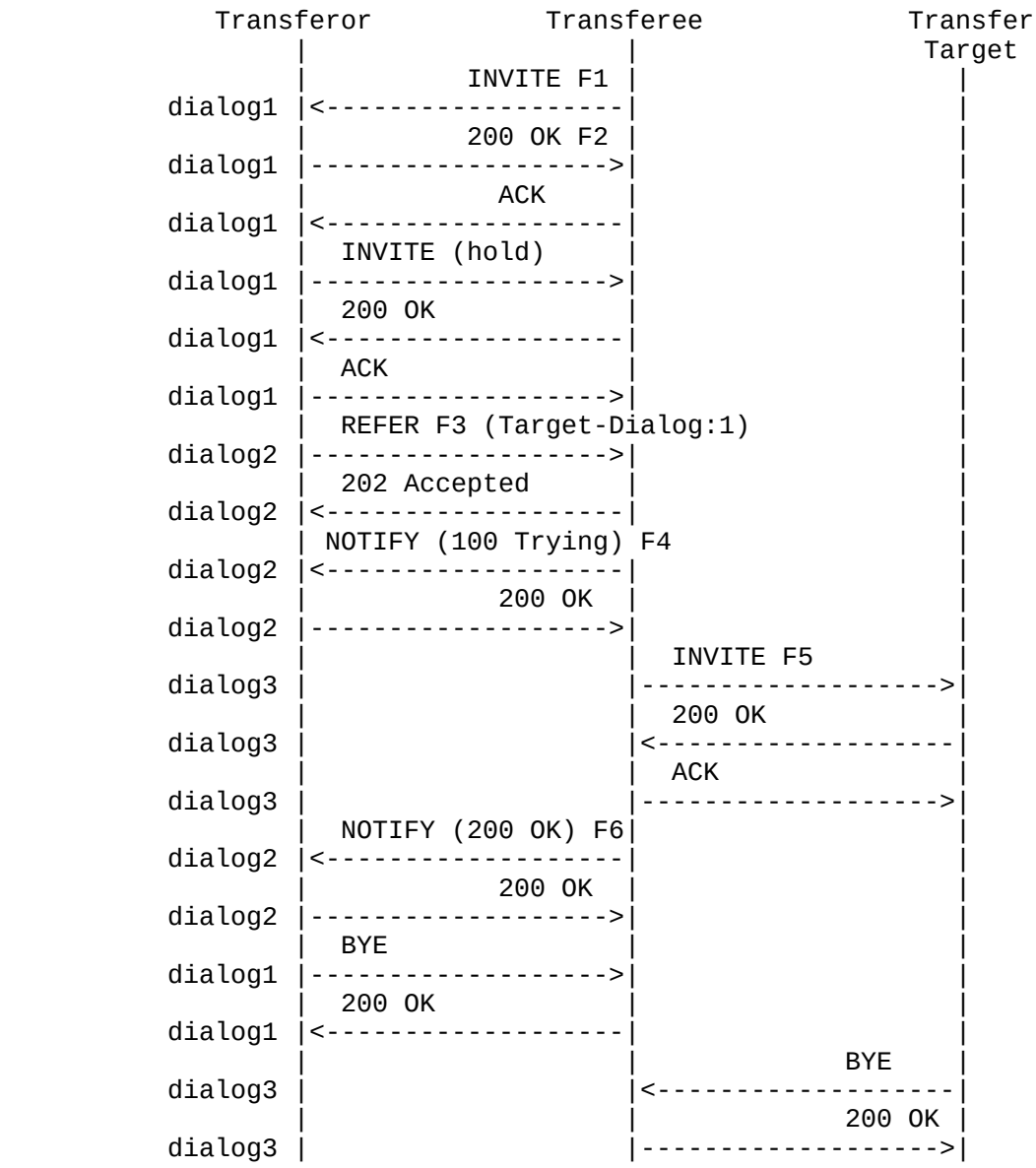


Figure 1: Basic Transfer Call Flow

F1 INVITE Transferee -> Transferor

```
INVITE sips:transferor@atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>
From: <sips:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu, tdialog
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Content-Type: application/sdp
Content-Length: ...
```

F2 200 OK Transferor -> Transferee

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
To: <sips:transferor@atlanta.example.com>;tag=31kdl4i3k
From: <sips:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu, tdialog
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d>
Content-Type: application/sdp
Content-Length: ...
```

F3 REFER Transferor -> Transferee

```
REFER sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKna9
Max-Forwards: 70
To: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
From: <sips:transferor@atlanta.example.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 REFER
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: gruu, replaces, tdialog
Require: tdialog
Refer-To: <sips:transfertarget@chicago.example.com>
Target-Dialog: 090459243588173445;local-tag=7553452
 ;remote-tag=31kdl4i3k
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d>
Content-Length: 0
```

F4 NOTIFY Transferee -> Transferor

```
NOTIFY sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>;tag=1928301774
From: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
 ;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 73 NOTIFY
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, tdialog
Event: refer
Subscription-State: active;expires=60
Content-Type: message/sipfrag
Content-Length: ...

SIP/2.0 100 Trying
```

F5 INVITE Transferee -> Transfer Target

```
INVITE sips:transfertarget@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas41234
Max-Forwards: 70
To: <sips:transfertarget@chicago.example.com>
From: <sips:transferee@biloxi.example.com>;tag=j3kso3iqhq
Call-ID: 90422f3sd23m4g56832034
CSeq: 521 REFER
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu, tdialog
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Content-Type: application/sdp
Content-Length: ...
```

F6 NOTIFY Transferee -> Transferor

```
NOTIFY sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>;tag=1928301774
From: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
 ;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 74 NOTIFY
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, tdialog
Event: refer
Subscription-State: terminated;reason=noresource
Content-Type: message/sipfrag
Content-Length: ...

SIP/2.0 200 OK
```

6.2. Transfer with Dialog Reuse

In this scenario, the Transferor does not know the properties of the Transferee's Contact URI or does not know that the Transferee supports the Target-Dialog header field. As a result, the REFER is sent inside the INVITE dialog.

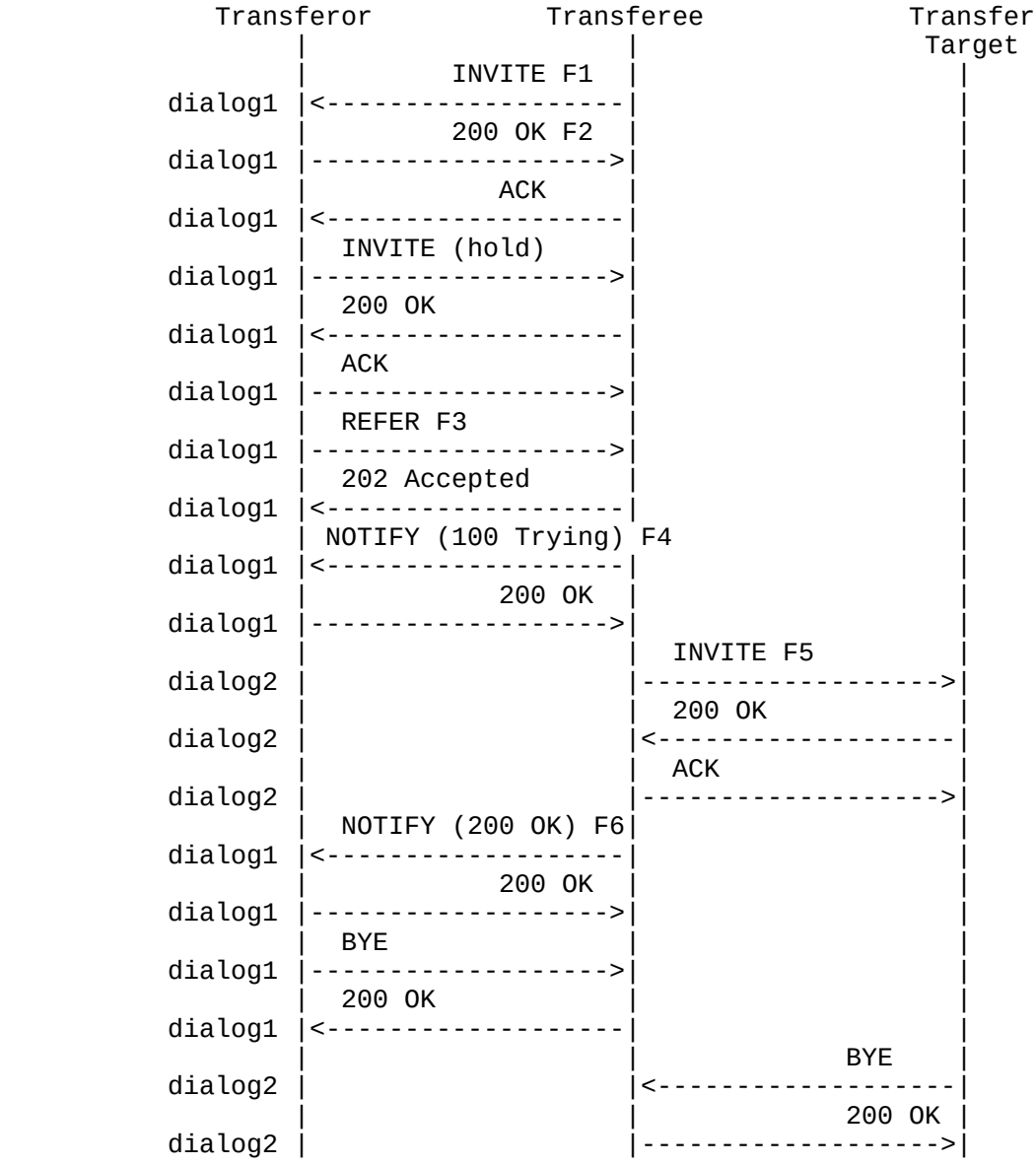


Figure 2: Transfer with Dialog Reuse

F1 INVITE Transferee -> Transferor

```
INVITE sips:transferor@atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>
From: <sips:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sips:transferee@192.0.2.4>
Content-Type: application/sdp
Content-Length: ...
```

F2 200 OK Transferor -> Transferee

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
To: <sips:transferor@atlanta.example.com>;tag=31kdl4i3k
From: <sips:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: gruu, replaces
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d>
Content-Type: application/sdp
Content-Length: ...
```

F3 REFER Transferor -> Transferee

```
REFER sips:transferee@192.0.2.4 SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKna9
Max-Forwards: 70
To: <sips:transferee@biloxi.example.com>;tag=7553452
From: <sips:transferor@atlanta.example.com>;tag=31kdl4i3k
Call-ID: 090459243588173445
CSeq: 314159 REFER
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Refer-To: <sips:transfertarget@chicago.example.com>
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d>
Content-Length: 0
```

F4 NOTIFY Transferee -> Transferor

```
NOTIFY sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>;tag=31kdl4i3k
From: <sips:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29888 INVITE
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Event: refer
Subscription-State: active;expires=60
Content-Type: message/sipfrag
Content-Length: ...

SIP/2.0 100 Trying
```

F5 INVITE Transferee -> Transfer Target

```
INVITE sips:transfertarget@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas41234
Max-Forwards: 70
To: <sips:transfertarget@chicago.example.com>
From: <sips:transferee@biloxi.example.com>;tag=j3kso3iqhq
Call-ID: 90422f3sd23m4g56832034
CSeq: 521 REFER
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sips:transferee@192.0.2.4>
Content-Type: application/sdp
Content-Length: ...
```

F6 NOTIFY Transferee -> Transferor

```
NOTIFY sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>;tag=31kdl4i3k
From: <sips:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29889 INVITE
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Event: refer
Subscription-State: terminated;reason=noresource
Content-Type: message/sipfrag
Content-Length: ...

SIP/2.0 200 OK
```

6.3. Failed Transfer

This section shows examples of failed transfer attempts. After the transfer failure occurs, the Transferor takes the Transferee off hold and resumes the session.

6.3.1. Target Busy

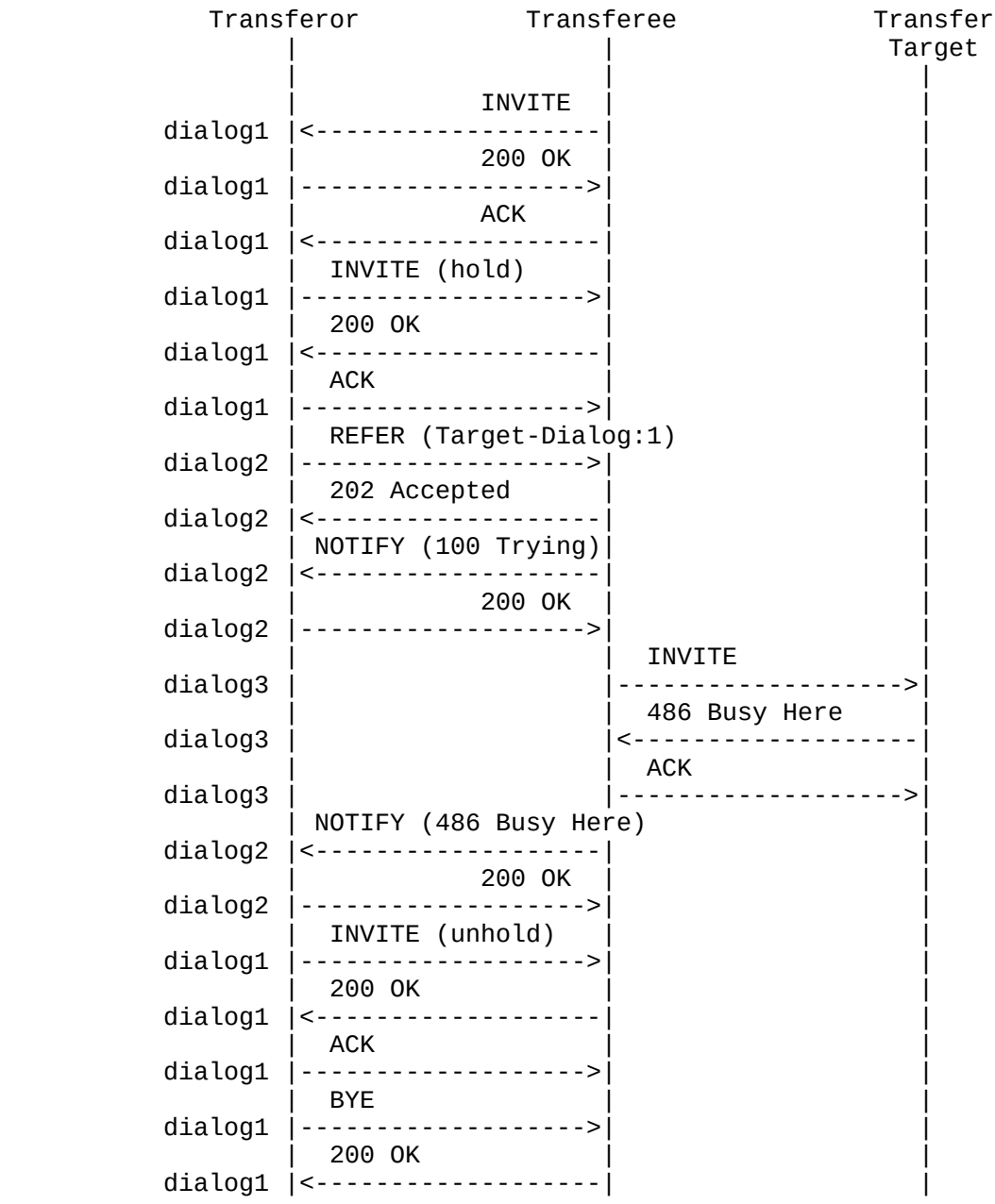
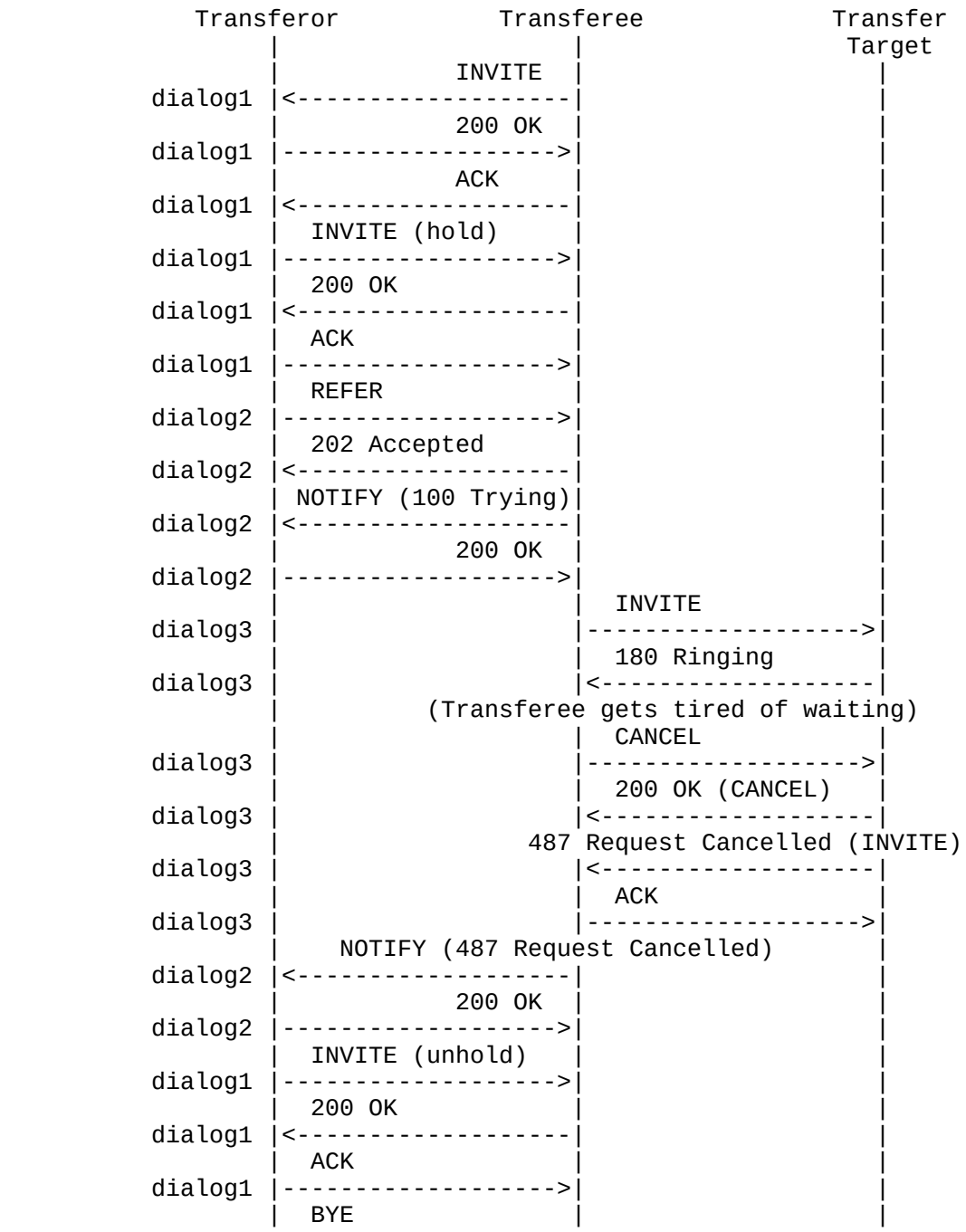


Figure 3: Failed Transfer - Target Busy

6.3.2. Transfer Target Does Not Answer



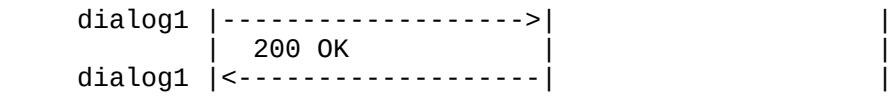


Figure 4: Failed Transfer - Target Does Not Answer

7. Transfer with Consultation Hold

Transfer with consultation hold involves a session between the Transferor and the Transfer Target before the transfer actually takes place. This is implemented with SIP Hold and Transfer as described above.

A nice feature is for the Transferor to let the target know that the session relates to an intended transfer. Since many UAs render the display name in the From header field to the user, a consultation INVITE could contain a string such as "Incoming consultation from Transferor with intent to transfer Transferee", where the display names of the transferor and transferee are included in the string.

7.1. Exposing Transfer Target

The Transferor places the Transferee on hold, establishes a call with the Transfer Target to alert them to the impending transfer, terminates the connection with the Transfer Target, then proceeds with transfer as above. This variation can be used to provide an experience similar to that expected by current PBX and Centrex users.

To (hopefully) improve clarity, non-REFER transactions have been collapsed into one indicator with the arrow showing the direction of the request.

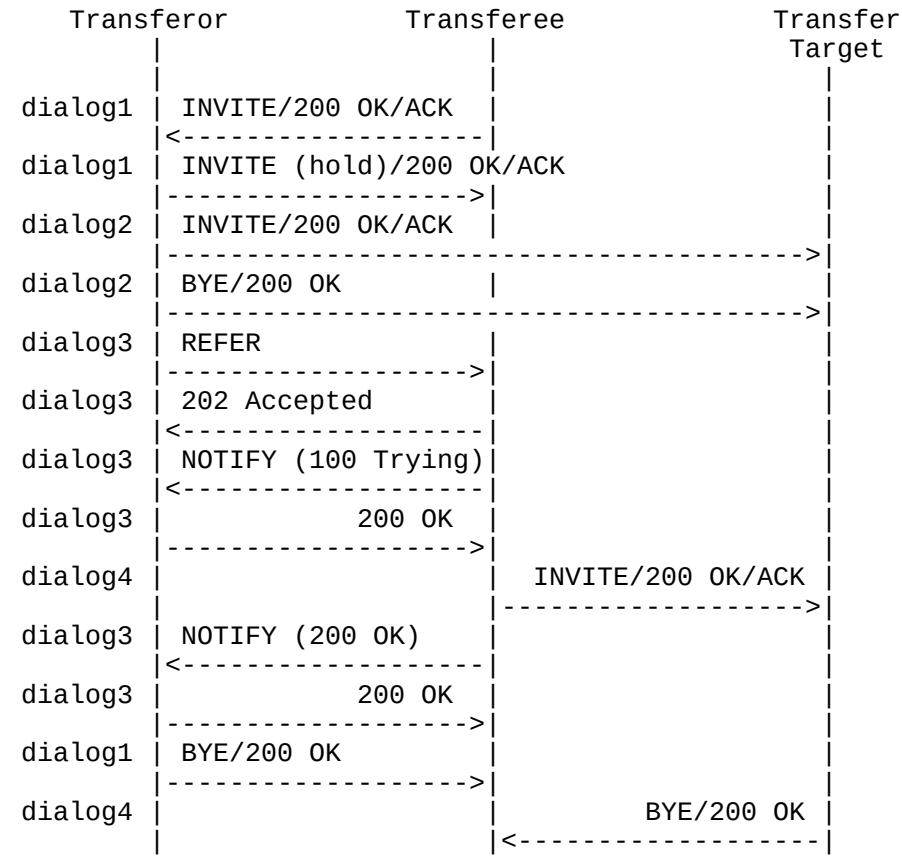


Figure 5: Transfer with Consultation Hold - Exposing Transfer Target

7.2. Protecting Transfer Target

The Transferor places the Transferee on hold, establishes a call with the Transfer Target and then reverses their roles, transferring the original Transfer Target to the original Transferee. This has the advantage of hiding information about the original Transfer Target from the original Transferee. On the other hand, the Transferee's experience is different than in current systems. The Transferee is effectively "called back" by the Transfer Target.

One of the problems with this simplest implementation of a target protecting transfer is that the Transferee is receiving a new call from the Transfer Target. Unless the Transferee's agent has a reliable way to associate this new call with the call it already has with the Transferor, it will have to alert the new call on another appearance. If this, or some other call-waiting-like UI were not

available, the Transferee might be stuck returning a Busy-Here to the Transfer Target, effectively preventing the transfer. There are many ways that correlation could be provided. The dialog parameters could be provided directly as header parameters in the Refer-To URI, for example. The Replaces mechanism [RFC3891] uses this approach and solves this problem nicely.

For the flow below, dialog1 means dialog identifier 1, and consists of the parameters of the Replaces header for dialog 1. In [RFC3891], this is the Call-ID, To-tag, and From-tag.

Note that the Transferee's agent emits a BYE to the Transferor's agent as an immediate consequence of processing the Replaces header.

The Transferor knows that both the Transferee and the Transfer Target support the Replaces header from the Supported: replaces header contained in the 200 OK responses from both.

In this scenario, the Transferee utilizes a GRUU as a Contact URI for reasons discussed in Section 6.3.

Note that the conventions used in the SIP Torture Test Messages [RFC4475] document are reused, specifically the <allOneLine> tag.

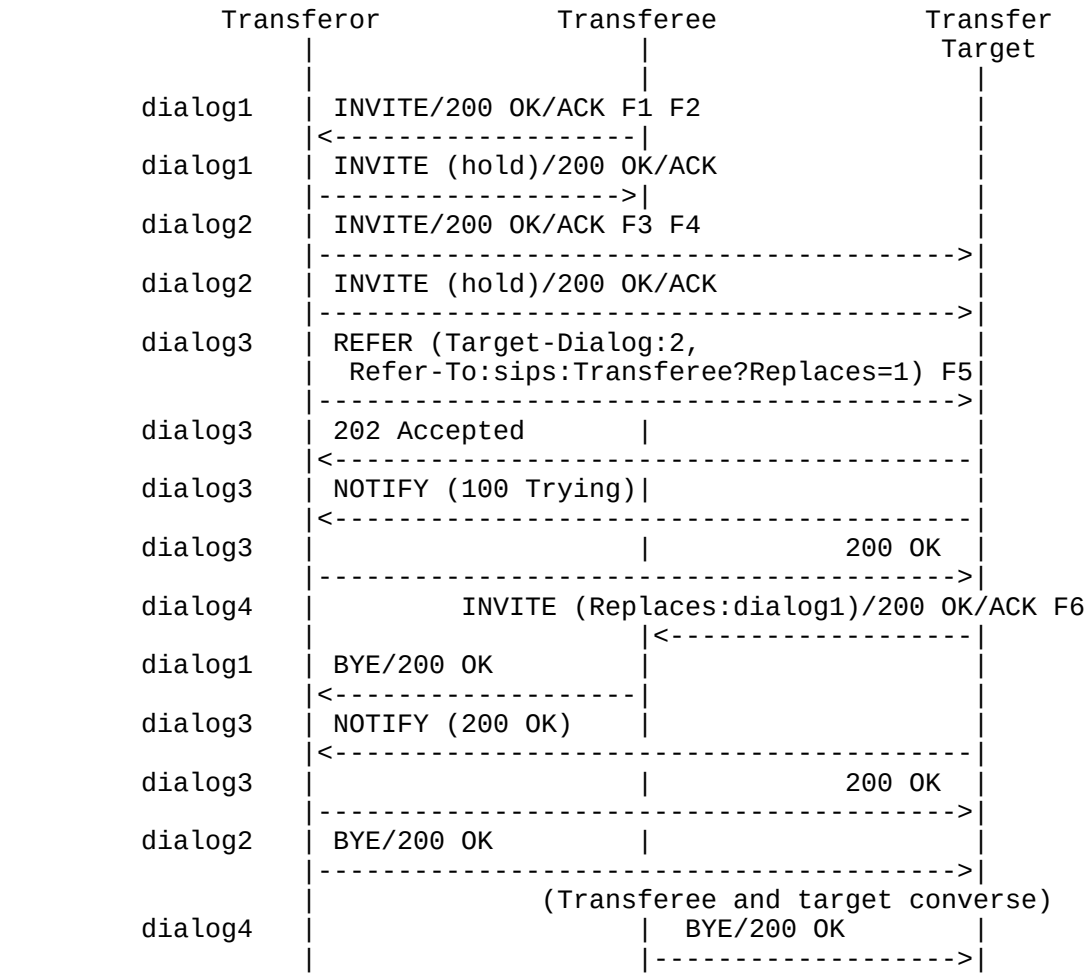


Figure 6: Transfer Protecting Transfer Target

F1 INVITE Transferee -> Transferor

```
INVITE sips:transferor@atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>
From: <sips:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Content-Type: application/sdp
Content-Length: ...
```

F2 200 OK Transferor -> Transferee

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
To: <sips:transferor@atlanta.example.com>;tag=31431
From: <sips:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu, tdialog
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d>
Content-Type: application/sdp
Content-Length: ...
```

F3 INVITE Transferor -> Transfer Target

```
INVITE sips:transfertarget@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transfertarget@chicago.example.com>
From: <sips:transferor@atlanta.example.com>;tag=763231
Call-ID: 592435881734450904
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: gruu, replaces, tdialog
Require: replaces
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=384i32lw3>
Content-Type: application/sdp
Content-Length: ...
```

F4 200 OK Transfer Target -> Transferor

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnas432
 ;received=192.0.2.1
To: <sips:transfertarget@chicago.example.com>;tag=9m2n3wq
From: <sips:transferor@atlanta.example.com>;tag=763231
Call-ID: 592435881734450904
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu, tdialog
Contact: <sips:482n4z24kdg@chicago.example.com;gr=8594958>
Content-Type: application/sdp
Content-Length: ...
```

F5 REFER Transferor -> Transfer Target

```
REFER sips:482n4z24kdg@chicago.example.com;gr=8594958 SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnashds9
Max-Forwards: 70
To: <sips:482n4z24kdg@chicago.example.com;gr=8594958>
From: <sips:transferor@atlanta.example.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 REFER
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: gruu, replaces, tdialog
Require: tdialog
<allOneLine>
Refer-To: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha
?Replaces=090459243588173445%3Bto-tag%3D7553452%3Bfrom-tag%3D31431>
</allOneLine>
Target-Dialog: 592435881734450904;local-tag=9m2n3wq
 ;remote-tag=763231
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d>
Content-Length: 0
```

F6 INVITE Transfer Target -> Transferee

```
INVITE sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha SIP/2.0
Via: SIP/2.0/TLS client.chicago.example.com;branch=z9hG4bKnaslu84
Max-Forwards: 70
To: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
From: <sips:transfertarget@chicago.example.com>;tag=341234
Call-ID: kmzwdle3dl3d08
CSeq: 41 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: gruu, replaces, tdialog
Contact: <sips:482n4z24kdg@chicago.example.com;gr=8594958>
Replaces: 090459243588173445;to-tag=7553452;from-tag=31431
Content-Type: application/sdp
Content-Length: ...
```

7.3. Attended Transfer

The Transferor places the Transferee on hold, establishes a call with the Transfer Target to alert them to the impending transfer, places the target on hold, then proceeds with transfer using an escaped Replaces header field in the Refer-To header. This is another common service expected by current PBX and Centrex users.

The Contact URI of the Transfer Target SHOULD be used by the Transferor as the Refer-To URI, unless the URI is suspected or known to not be routable outside the dialog. Otherwise, the Address of Record (AOR) of the Transfer Target SHOULD be used. That is, the same URI that the Transferor used to establish the session with the Transfer Target should be used. In case the triggered INVITE is routed to a different User Agent than the Transfer Target, the Require: replaces header field SHOULD be used in the triggered INVITE. (This is to prevent an incorrect User Agent that does not support Replaces from ignoring the Replaces and answering the INVITE without a dialog match.)

It is possible that proxy/service routing may prevent the triggered INVITE from reaching the same User Agent. If this occurs, the triggered invite will fail with a timeout, 403, 404, etc. error. The Transferee MAY then retry the transfer with the Refer-To URI set to the Contact URI.

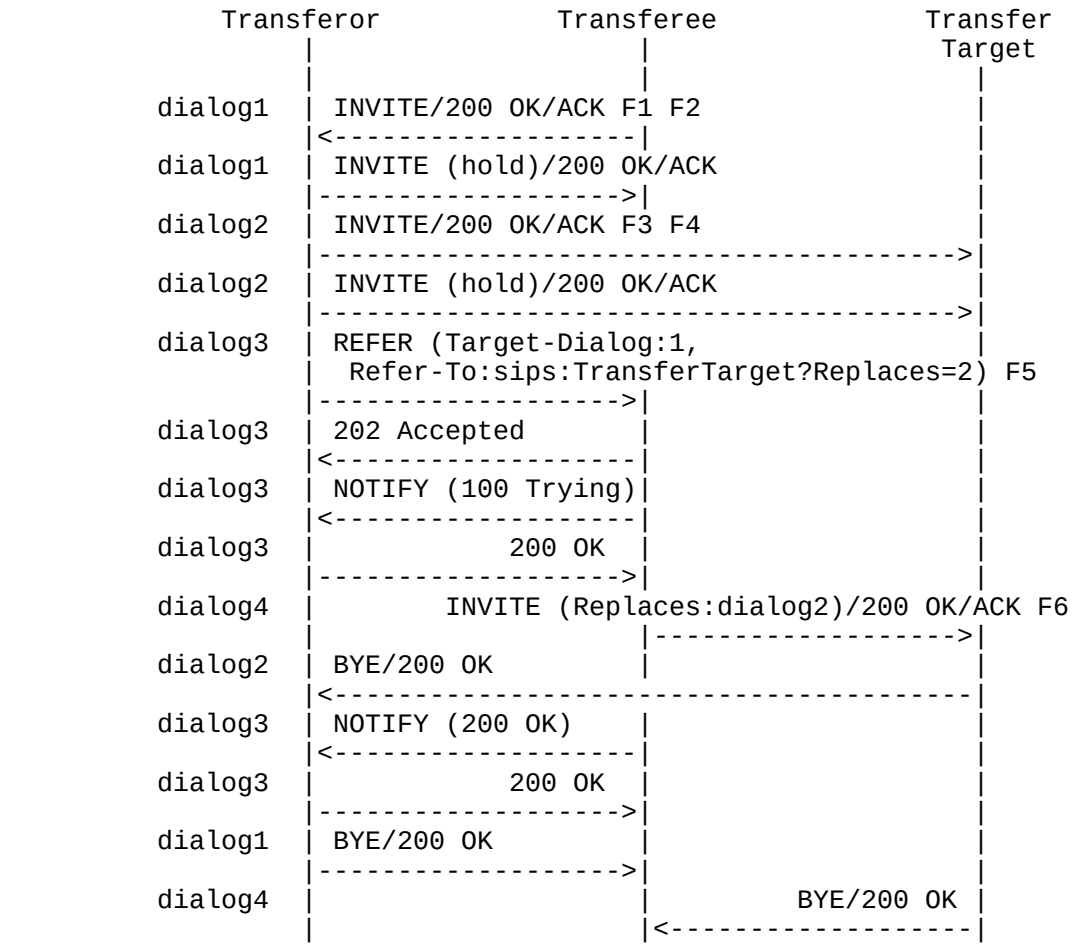


Figure 7: Attended Transfer Call Flow

F1 INVITE Transferee -> Transferor

```
INVITE sips:transferor@atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>
From: <sips:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu, tdialog
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Content-Type: application/sdp
Content-Length: ...
```

F2 200 OK Transferor -> Transferee

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
To: <sips:transferor@atlanta.example.com>;tag=31431
From: <sips:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu, tdialog
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d>
Content-Type: application/sdp
Content-Length: ...
```

F3 INVITE Transferor -> Transfer Target

```
INVITE sips:transfertarget@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transfertarget@chicago.example.com>
From: <sips:transferor@atlanta.example.com>;tag=763231
Call-ID: 592435881734450904
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: gruu, replaces, tdialog
Require: replaces
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=384i32lw3>
Content-Type: application/sdp
Content-Length: ...
```

F4 200 OK Transfer Target -> Transferor

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnas432
 ;received=192.0.2.1
To: <sips:transfertarget@chicago.example.com>;tag=9m2n3wq
From: <sips:transferor@atlanta.example.com>;tag=763231
Call-ID: 592435881734450904
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Contact: <sips:482n4z24kdg@chicago.example.com;gr=8594958>
Content-Type: application/sdp
Content-Length: ...
```

F5 REFER Transferor -> Transferee

```
REFER sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnashds9
Max-Forwards: 70
To: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
From: <sips:transferor@atlanta.example.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 REFER
Require: tdialog
<allOneLine>
Refer-To: <sips:482n4z24kdg@chicago.example.com;gr=8594958?
Replaces=592435881734450904%3Bto-tag%3D9m2n3wq%3Bfrom-tag3D763231>
</allOneLine>
Target-Dialog: 592435881734450904;local-tag=9m2n3wq
 ;remote-tag=763231
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d>
Content-Length: 0
```

F6 INVITE Transferee -> Transfer Target

```
INVITE sips:482n4z24kdg@chicago.example.com;gr=8594958 SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnaslu82
Max-Forwards: 70
To: <sips:482n4z24kdg@chicago.example.com;gr=8594958>
From: <sips:transferee@biloxi.example.com>;tag=954
Call-ID: kmzwdle3dl3d08
CSeq: 41 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: gruu, replaces, tdialog
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Replaces: 592435881734450904;to-tag=9m2n3wq;from-tag=763231
Content-Type: application/sdp
Content-Length: ...
```

7.4. Recovery When One Party Does Not Support REFER

If protecting or exposing the Transfer Target is not a concern, it is possible to complete a transfer with consultation hold when only the transferor and one other party support REFER. Note that a 405 Method Not Allowed might be returned instead of the 501 Not Implemented response.

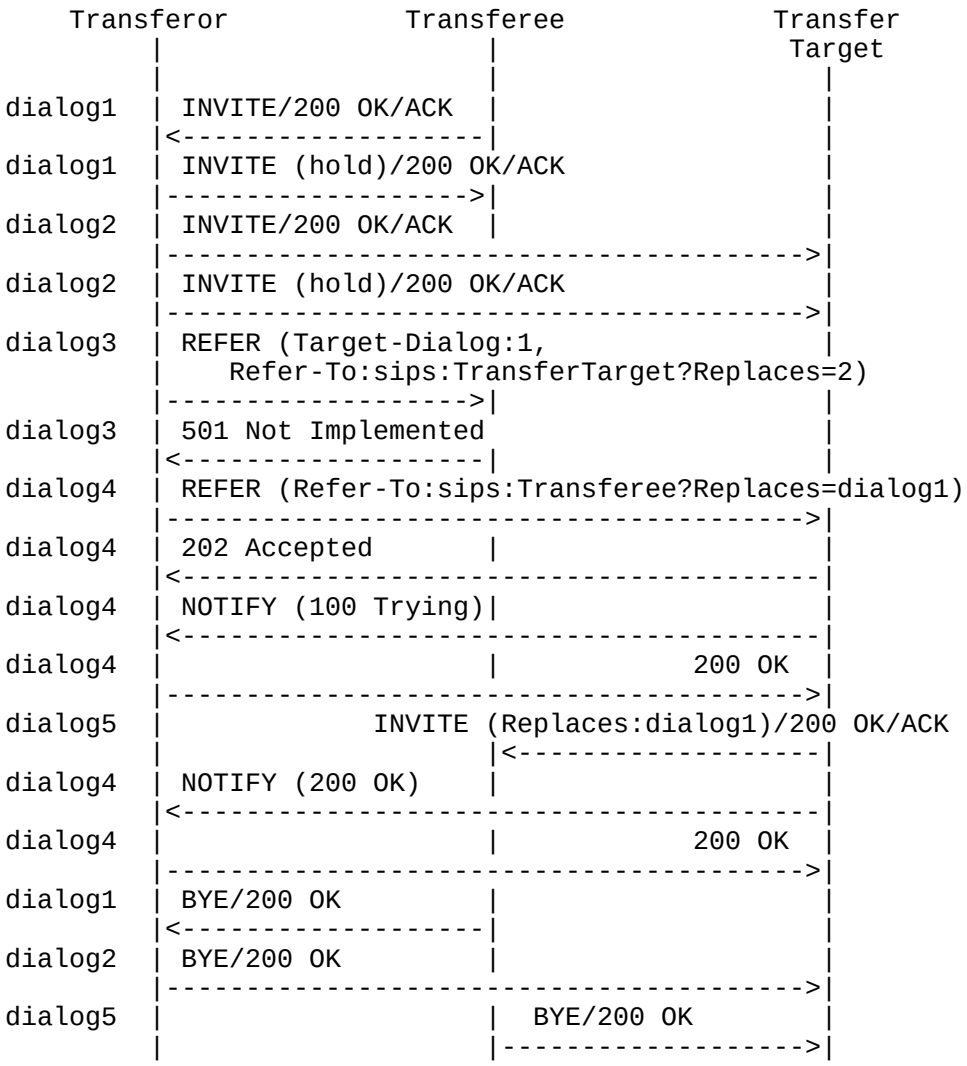


Figure 8: Recovery When One Party Does Not Support REFER

7.5. Attended Transfer When Contact URI Is Not Known to Route to a Unique User Agent

It is a requirement of RFC 3261 that a Contact URI be globally routable even outside the dialog. However, due to RFC 2543 User Agents and some architectures (NAT/Firewall traversal, screening proxies, Application Layer Gateways (ALGs), etc.) this will not

always be the case. As a result, the method of attended transfer shown in Figures 6, 7, and 8 SHOULD only be used if the Contact URI is known to be routable outside the dialog.

Figure 9 shows such a scenario where the Transfer Target Contact URI is not routable outside the dialog, so the triggered INVITE is sent to the AOR of the Transfer Target.

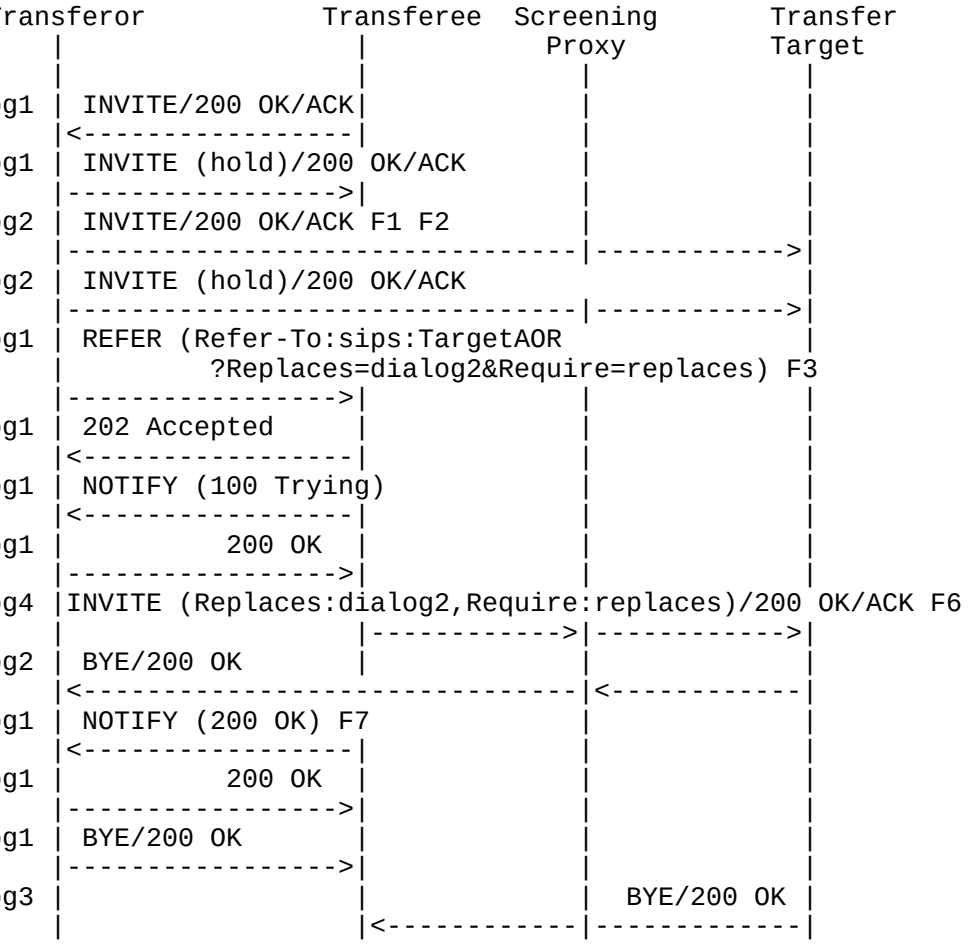


Figure 9: Attended Transfer Call Flow with a Contact URI Not Known to Be Globally Routable

F1 INVITE Transferor -> Transfer Target

```
INVITE sips:transfertarget@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bK76
Max-Forwards: 70
To: <sips:transfertarget@chicago.example.com>
From: <sips:transferor@atlanta.example.com>;tag=763231
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sips:transferor@pc33.atlanta.example.com>
Content-Type: application/sdp
Content-Length: ...
```

F2 200 OK Transfer Target -> Transferee

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnas432
 ;received=192.0.2.1
To: <sips:transfertarget@chicago.example.com>;tag=9m2n3wq
From: <sips:transferor@atlanta.example.com>;tag=763231
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sips:transfertarget@client.chicago.example.com>
Content-Type: application/sdp
Content-Length: ...
```

F3 REFER Transferor -> Transferee

```
REFER sips:transferee@192.0.2.4 SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnashds9
Max-Forwards: 70
To: <sips:transferee@biloxi.example.com>;tag=a6c85cf
From: <sips:transferor@atlanta.example.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 REFER
<allOneLine>
Refer-To: <sips:transfertarget@chicago.example.com?Replaces=
090459243588173445%3Bto-tag%3D9m2n3wq%3Bfrom-tag%3D763231
&Require=replaces>
<allOneLine>
Contact: <sips:transferor@pc33.atlanta.example.com>
Content-Length: 0
```

F4 INVITE Transferee -> Transfer Target

```
INVITE sips:transfertarget@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnaslu82
Max-Forwards: 70
To: <sips:transfertarget@chicago.example.com>
From: <sips:transferee@biloxi.example.com>;tag=954
Call-ID: 20482817324945934422930
CSeq: 42 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sips:transferee@192.0.2.4>
Replaces: 090459243588173445;to-tag=9m2n3wq;from-tag=763231
Require: replaces
Content-Type: application/sdp
Content-Length: ...
```

F5 NOTIFY Transferee -> Transferor

```
NOTIFY sips:transferor@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>;tag=1928301774
From: <sips:transferee@biloxi.example.com>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 76 NOTIFY
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Event: refer;id=98873867
Subscription-State: terminated;reason=noresource
Content-Type: message/sipfrag
Content-Length: ...

SIP/2.0 200 OK
```

Figure 10 shows a failure case in which the AOR URI fails to reach the Transfer Target. As a result, the transfer is retried with the Contact URI, at which point it succeeds.

Note that there is still no guarantee that the correct endpoint will be reached, and the result of this second REFER may also be a failure. In that case, the Transferor could fall back to unattended transfer or give up on the transfer entirely. Since two REFERs are sent within the dialog creating two distinct subscriptions, the Transferee uses the 'id' parameter in the Event header field to distinguish notifications for the two subscriptions.

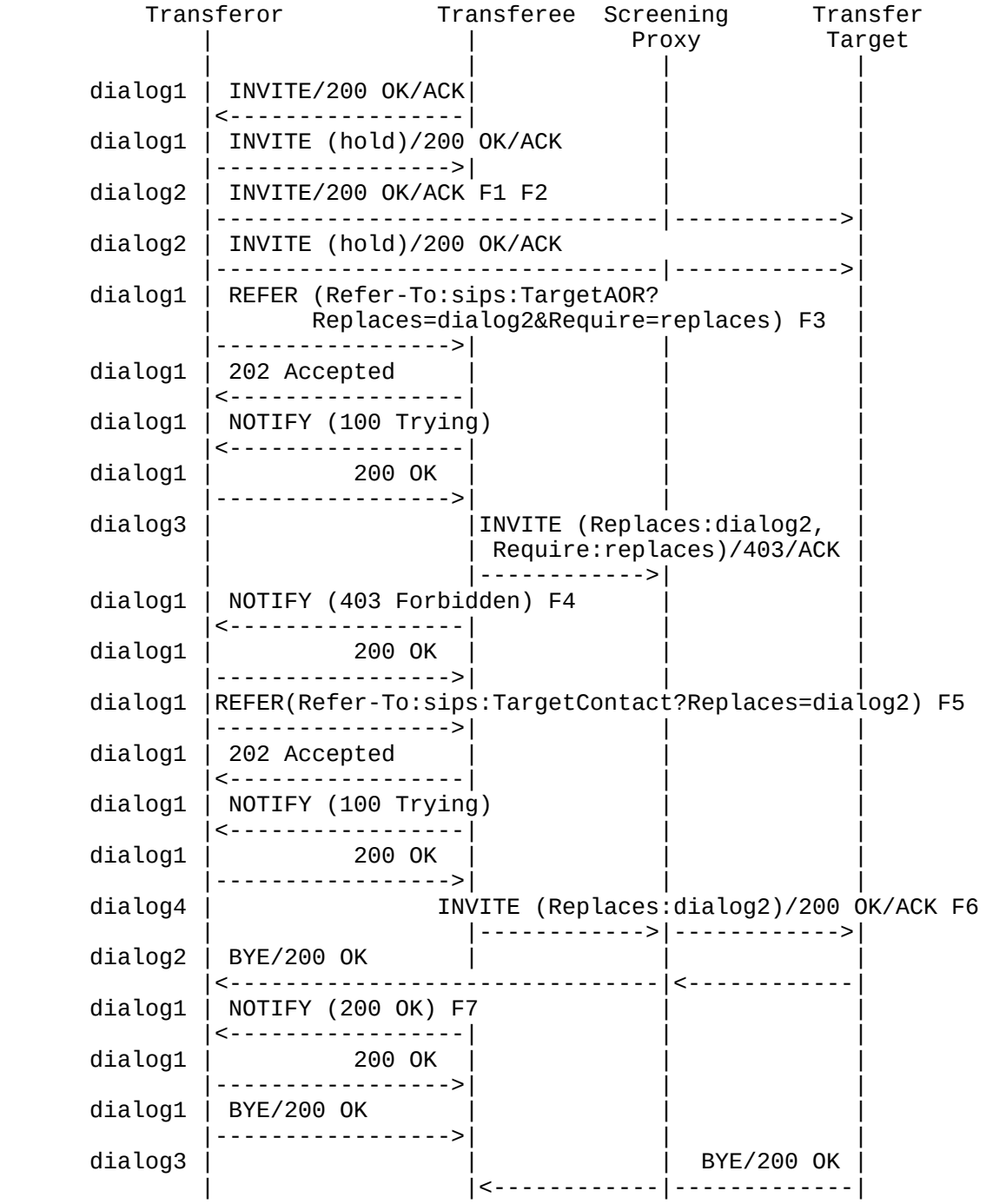

Transferor
Transferee
Screening Proxy
Transfer Target
dialog1 INVITE/200 OK/ACK
dialog1 INVITE (hold)/200 OK/ACK
dialog2 INVITE/200 OK/ACK F1 F2
dialog2 INVITE (hold)/200 OK/ACK
dialog1 REFER (Refer-To:sips:TargetAOR?
Replaces=dialog2&Require=replaces) F3
dialog1 202 Accepted
dialog1 NOTIFY (100 Trying)
dialog1 200 OK
dialog3 INVITE (Replaces:dialog2,
Require:replaces)/403/ACK
dialog1 NOTIFY (403 Forbidden) F4
dialog1 200 OK
dialog1 REFER(Refer-To:sips:TargetContact?Replaces=dialog2) F5
dialog1 202 Accepted
dialog1 NOTIFY (100 Trying)
dialog1 200 OK
dialog4 INVITE (Replaces:dialog2)/200 OK/ACK F6
dialog2 BYE/200 OK
dialog1 NOTIFY (200 OK) F7
dialog1 200 OK
dialog1 BYE/200 OK
dialog3 BYE/200 OK

Figure 10: Attended Transfer Call Flow with Non-Routable Contact URI and AOR Failure

F1 INVITE Transferor -> Transfer Target

```
INVITE sips:transfertarget@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bK76
Max-Forwards: 70
To: <sips:transfertarget@chicago.example.com>
From: <sips:transferor@atlanta.example.com>;tag=763231
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sips:transferor@pc33.atlanta.example.com>
Content-Type: application/sdp
Content-Length: ...
```

F2 200 OK Transfer Target -> Transferee

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnas432
 ;received=192.0.2.1
To: <sips:transfertarget@chicago.example.com>;tag=9m2n3wq
From: <sips:transferor@atlanta.example.com>;tag=763231
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sips:transfertarget@client.chicago.example.com>
Content-Type: application/sdp
Content-Length: ...
```

F3 REFER Transferor -> Transferee

```
REFER sips:transferee@192.0.2.4 SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnashds9
Max-Forwards: 70
To: <sips:transferee@biloxi.example.com>;tag=a6c85cf
From: <sips:transferor@atlanta.example.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 REFER
<allOneLine>
Refer-To: <sips:transfertarget@chicago.example.com?Replaces=
090459243588173445%3Bto-tag%3D9m2n3wq%3Bfrom-tag%3D763231
&Require=replaces>
</allOneLine>
Contact: <sips:transferor@pc33.atlanta.example.com>
Content-Length: 0
```

F4 NOTIFY Transferee -> Transferor

```
NOTIFY sips:transferor@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>;tag=1928301774
From: <sips:transferee@biloxi.example.com>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 74 NOTIFY
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Event: refer;id=314159
Subscription-State: terminated;reason=noresource
Content-Type: message/sipfrag
Content-Length: ...

SIP/2.0 403 Forbidden
```

F5 REFER Transferor -> Transferee

```
REFER sips:transferee@192.0.2.4 SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bKnashds9
Max-Forwards: 70
To: <sips:transferee@biloxi.example.com>;tag=a6c85cf
From: <sips:transferor@atlanta.example.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 REFER
<allOneLine>
Refer-To: <sips:transfertarget@client.chicago.example.com
?Replaces=09045924358817344 5%3Bto-tag%3D9m2n3wq
%3Bfrom-tag%3D763231>
</allOneLine>
Contact: <sips:transferor@pc33.atlanta.example.com>
Content-Length: 0
```

F6 INVITE Transferee -> Transfer Target

```
INVITE sips:transfertarget@client.chicago.example.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnaslu82
Max-Forwards: 70
To: <sips:transfertarget@chicago.example.com>
From: <sips:transferee@biloxi.example.com>;tag=954
Call-ID: 20482817324945934422930
CSeq: 42 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sips:transferee@192.0.2.4>
Replaces: 090459243588173445;to-tag=9m2n3wq;from-tag=763231
Content-Type: application/sdp
Content-Length: ...
```

F7 NOTIFY Transferee -> Transferor

```
NOTIFY sips:transferor@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>;tag=1928301774
From: <sips:transferee@biloxi.example.com>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 76 NOTIFY
Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Event: refer;id=314160
Subscription-State: terminated;reason=noresource
Content-Type: message/sipfrag
Content-Length: ...

SIP/2.0 200 OK
```

To prevent this scenario from happening, the Transfer Target SHOULD use a Contact URI that is routable outside the dialog, which will result in the call flow of Figure 7.

7.6. Semi-Attended Transfer

In any of the consultation hold flows above, the Transferor may decide to terminate its attempt to contact the Transfer Target before that session is established. Most frequently, that will be the end of the scenario, but in some circumstances, the Transferor may wish to proceed with the transfer action. For example, the Transferor may wish to complete the transfer knowing that the Transferee will end up eventually talking to the Transfer Target's voicemail service. Some PBX systems support this feature, sometimes called "semi-attended transfer", that is effectively a hybrid between a fully attended transfer and an unattended transfer. A call flow is shown in Figure 11. In this flow, the Transferor's User Agent continues the transfer as an attended transfer even after the Transferor hangs up. Note that media must be played to the Transfer Target upon answer -- otherwise, the Target may hang up and the resulting transfer operation will fail.

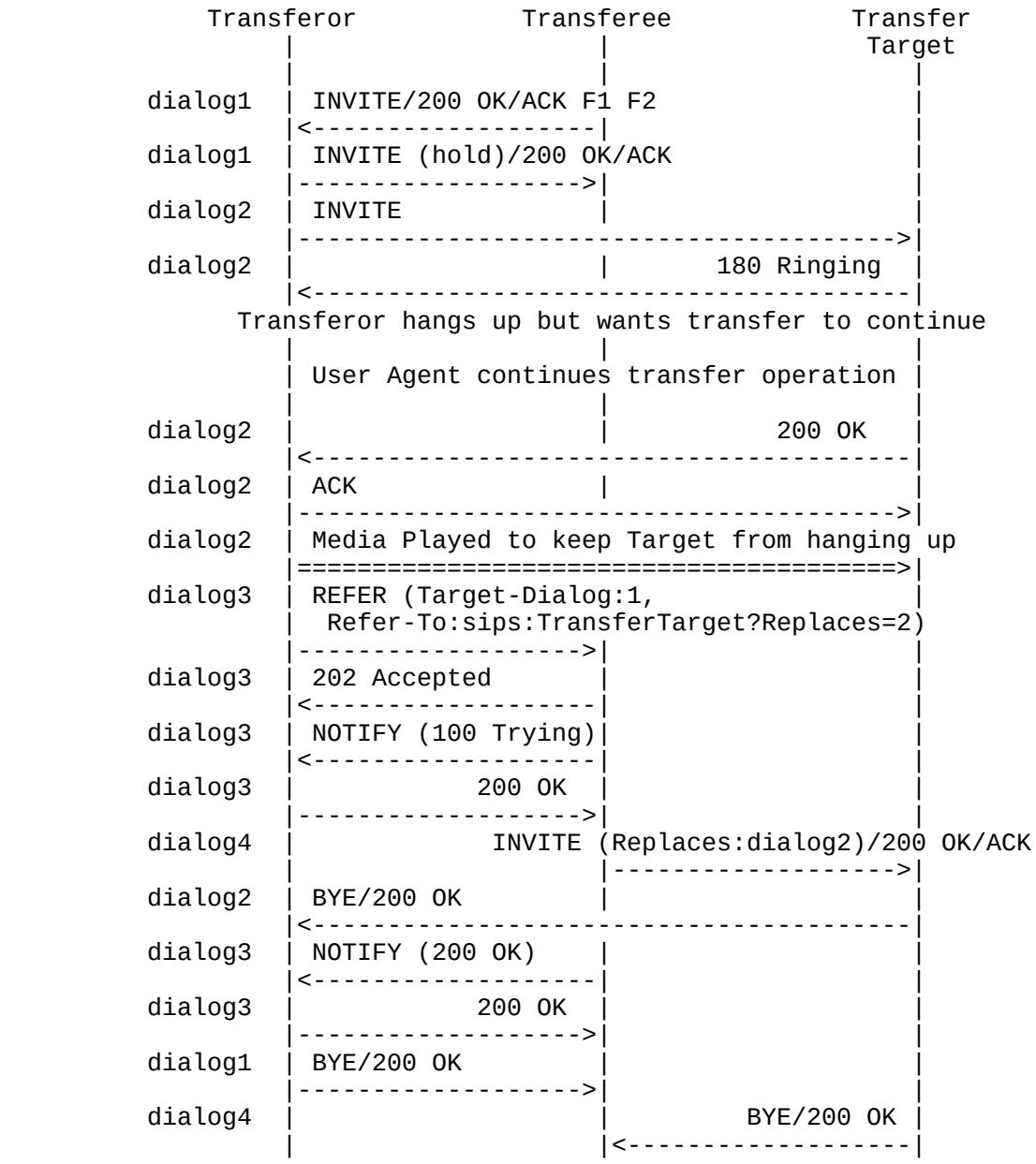


Figure 11: Recommended Semi-Attended Transfer Call Flow

Two other possible semi-attended transfer call flows are shown in Figures 12 and 13. However, these call flows are NOT RECOMMENDED due to race conditions. In both of these flows, when the Transferor

hangs up, the Transferor attempts to revert to unattended transfer by sending a CANCEL to the target. This can result in two race conditions. One is that the target answers despite the CANCEL and the resulting unattended transfer fails. This race condition can be eliminated by the Transferor waiting to send the REFER until the 487 response from the target is returned. Instead of a 487, a 200 OK may be returned indicating that the target has answered the consultation call. In this case, the call flow in Figure 13 must be followed. In this flow, the Transferor must play some kind of media to the Target to prevent the Target from hanging up, or the transfer will fail. That is, the human at the Transfer Target will hear silence from when they answer (message F1) until the transfer completes (F3 and they are talking to the Transferee unless some media is played (F2)).

The second race condition occurs in Figure 12 if the Transfer Target goes "off hook" after the CANCEL is received and the 487 returned. This may result in a 486 Busy Here response to the unattended transfer.

The recommended call flow of Figure 11 does not utilize a CANCEL and does not suffer from these race conditions.

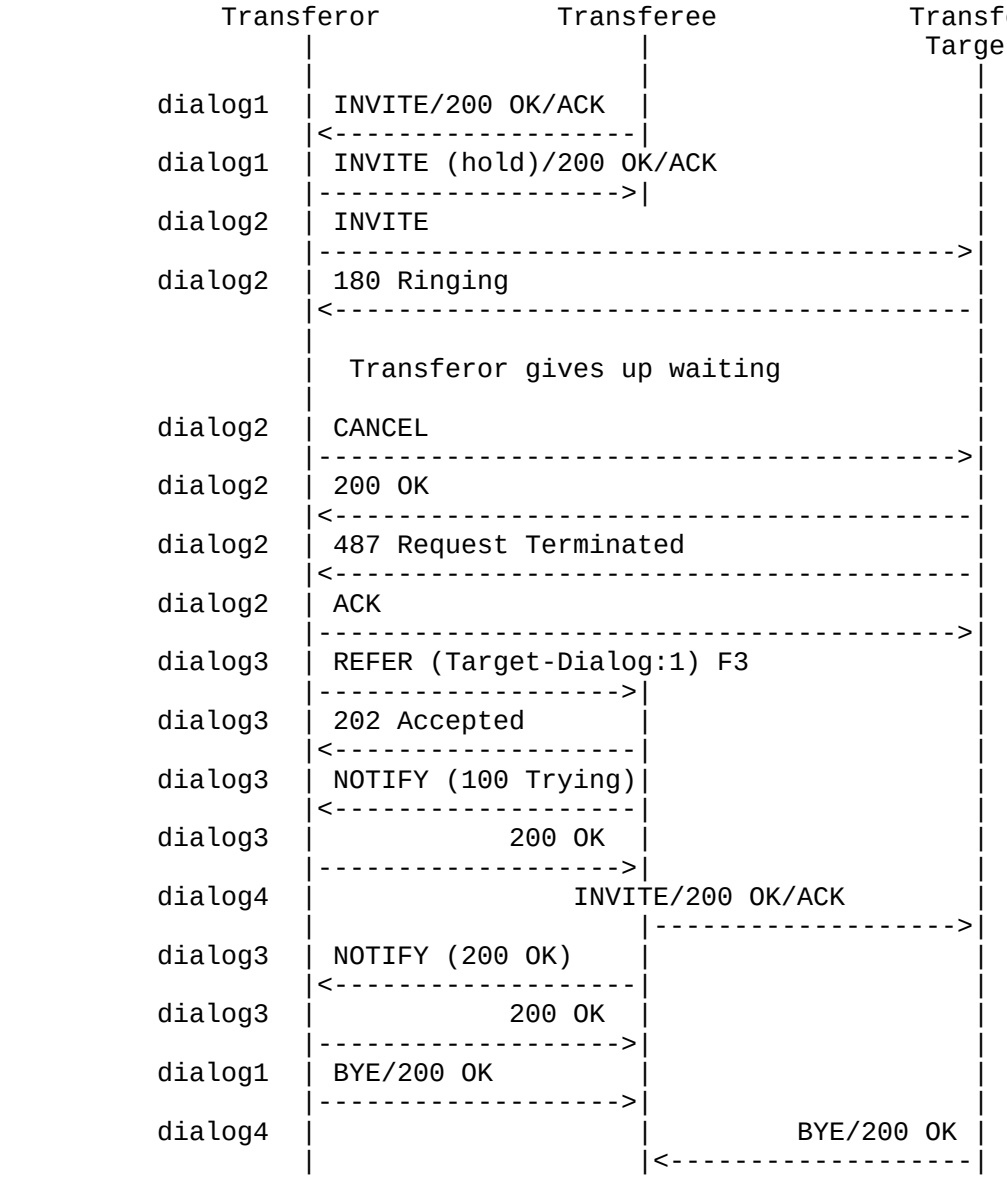


Figure 12: Semi-Attended Transfer as Blind Transfer Call Flow (Not Recommended)

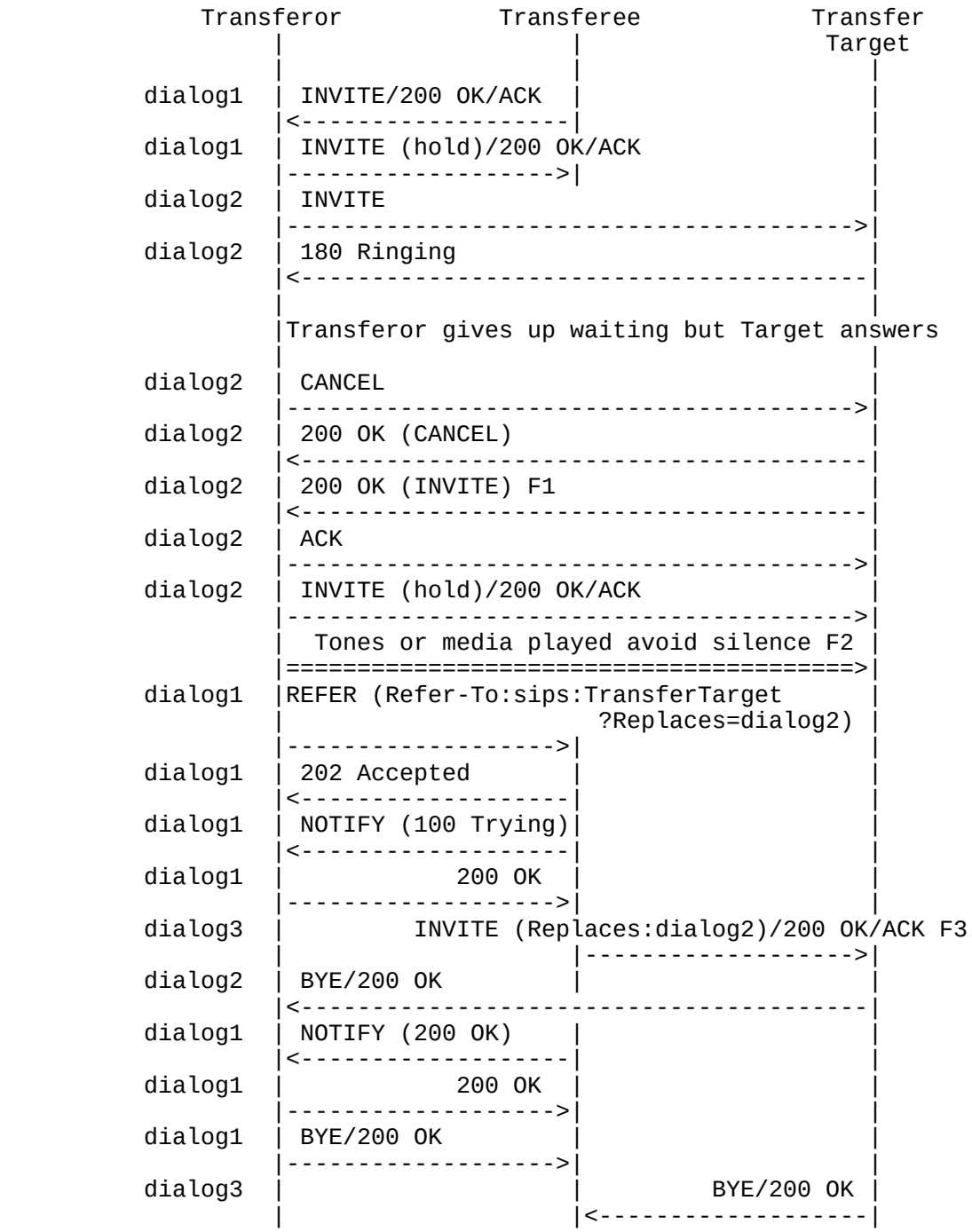

Transferor
Transferee
Transfer Target
dialog1 INVITE/200 OK/ACK
dialog1 INVITE (hold)/200 OK/ACK
dialog2 INVITE
dialog2 180 Ringing
Transferor gives up waiting but Target answers
dialog2 CANCEL
dialog2 200 OK (CANCEL)
dialog2 200 OK (INVITE) F1
dialog2 ACK
dialog2 INVITE (hold)/200 OK/ACK
Tones or media played avoid silence F2
dialog1 REFER (Refer-To:sips:TransferTarget ?Replaces=dialog2)
dialog1 202 Accepted
dialog1 NOTIFY (100 Trying)
dialog1 200 OK
dialog3 INVITE (Replaces:dialog2)/200 OK/ACK F3
dialog2 BYE/200 OK
dialog1 NOTIFY (200 OK)
dialog1 200 OK
dialog1 BYE/200 OK
dialog3 BYE/200 OK

Figure 13: Semi-Attended Transfer as Attended Transfer Call Flow (Not Recommended)

7.7. Attended Transfer Fallback to Basic Transfer

In this flow, an attempted attended transfer fails so the Transferor falls back to basic transfer.

The call flow in Figure 14 shows the use of Require: replaces in the INVITE sent by the Transferor to the Transfer Target in which the Transferor's intention at the time of sending the INVITE to the Transfer Target was known to be to complete an attended transfer. Since the Target does not support Replaces, the INVITE is rejected with a 420 Bad Extension response, and the Transferor switches from attended transfer to basic transfer immediately.

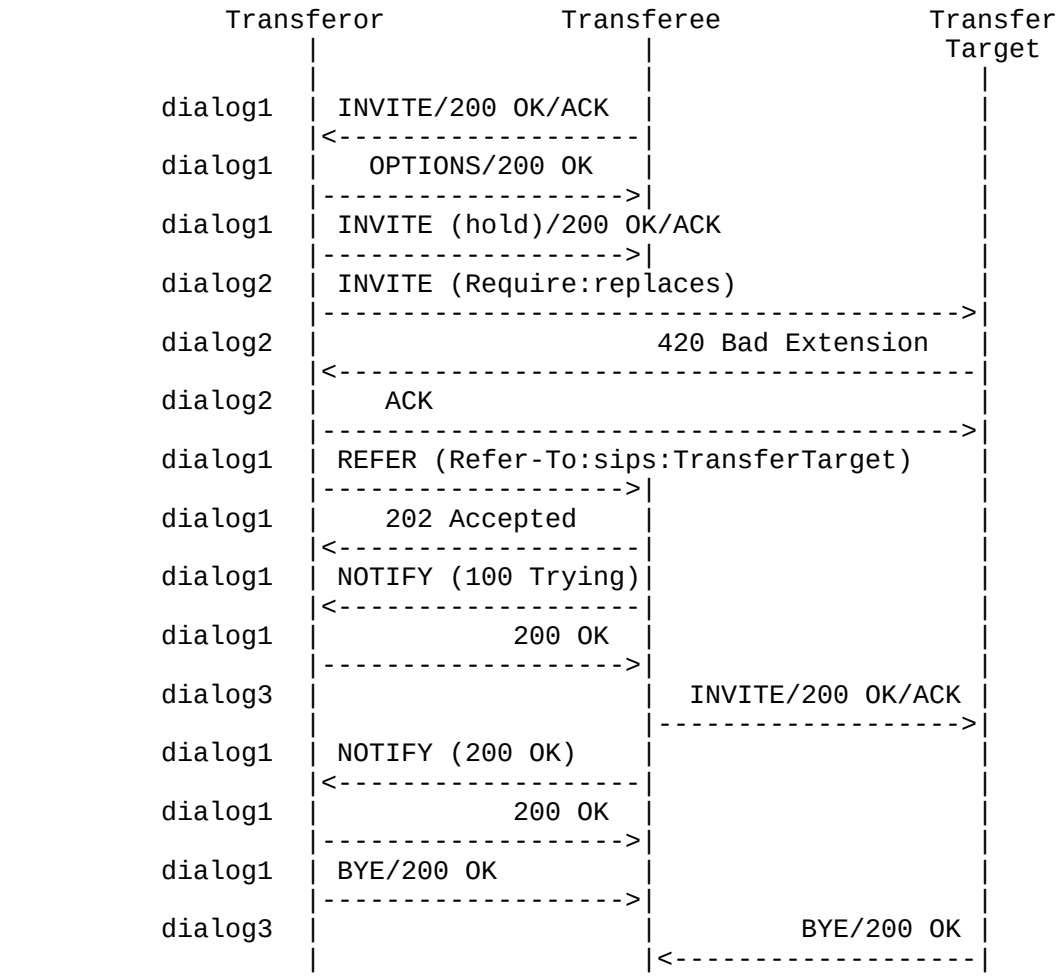


Figure 14: Attended Transfer Fallback to Basic Transfer Using Require:replaces

Figure 15 shows the use of OPTIONS when the Transferee and Transfer Target do not explicitly indicate support for the REFER method and Replaces header fields in Allow and Supported header fields and the Transferor did not have the intention of performing an attended transfer when the INVITE to the Target was sent. In dialog1, the Transferor determines, using OPTIONS, that the Transferee does support REFER and Replaces. As a result, the Transferor begins the attended transfer by placing the Transferee on hold and calling the Transfer Target. Using an OPTIONS in dialog2, the Transferor determines that the target does not support either REFER or Replaces,

making attended transfer impossible. The Transferor then ends dialog2 by sending a BYE then sends a REFER to the Transferee using the AOR URI of the Transfer Target.

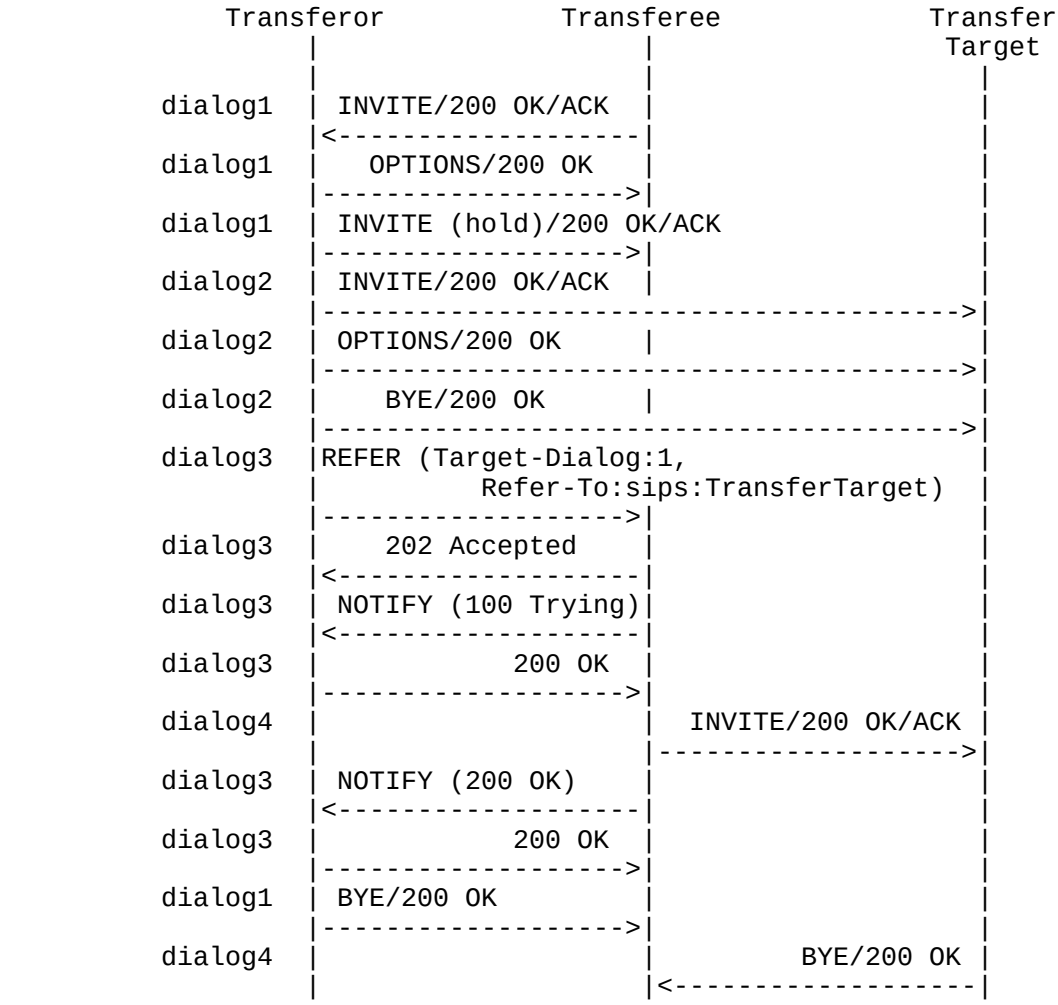


Figure 15: Attended Transfer Fallback to Basic Transfer

8. Transfer with Referred-By

In the previous examples, the Transfer Target does not have definitive information about what party initiated the transfer, or, in some cases, even that transfer is taking place. The Referred-By mechanism [RFC3892] provides a way for the Transferor to provide the Transferee with a way to let the Transfer Target know what party initiated the transfer.

The simplest and least secure approach just involves the inclusion of the Referred-By header field in the REFER, which is then copied into the triggered INVITE. However, a more secure mechanism involving the Referred-By security token, which is generated and signed by the Transferor and passed in a message body to the Transferee then to the Transfer Target.

The call flow in Figure 16 shows the Referred-By header field and body in the REFER F5 and triggered INVITE F6. Note that the Secure/Multipurpose Internet Mail Extensions (S/MIME) signature is not shown in the example below. The conventions used in the SIP Torture Test Messages [RFC4475] document are reused, specifically the <hex> and <allOneLine> tags.

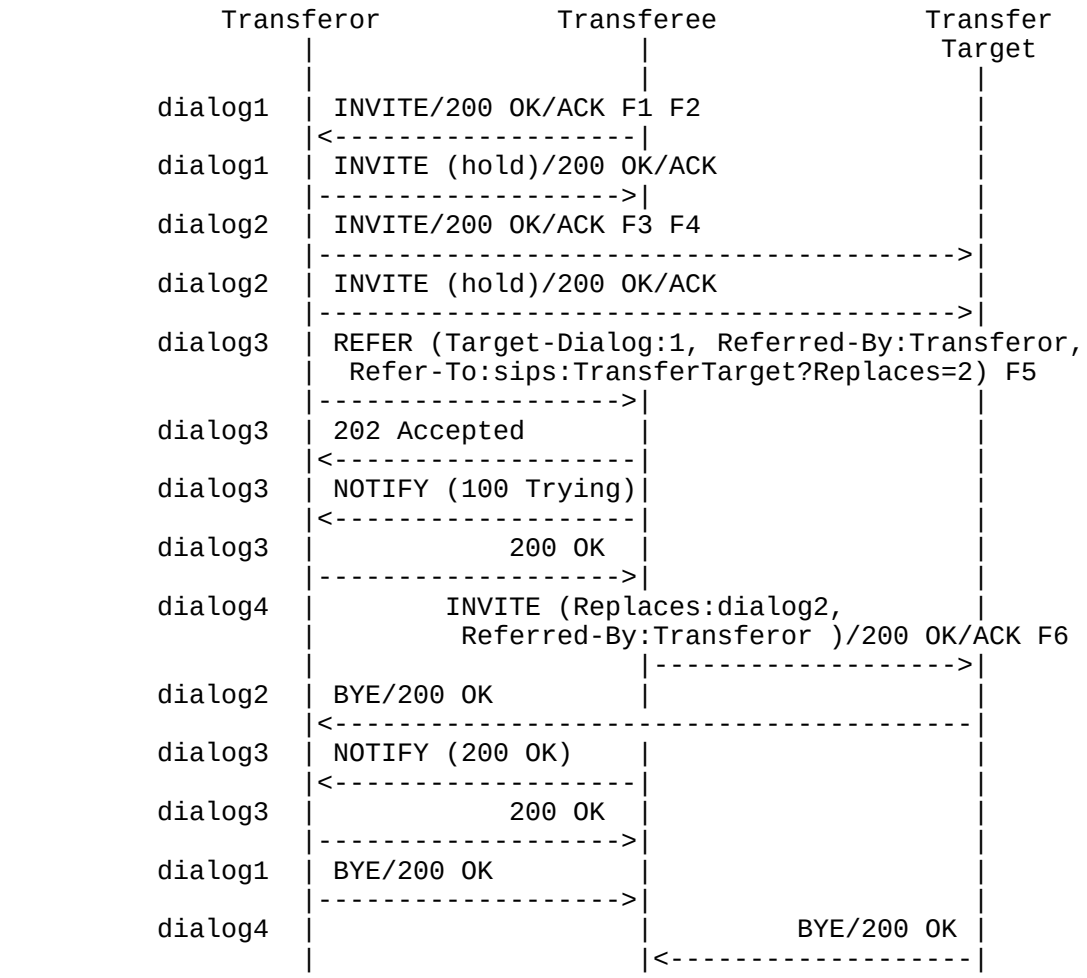


Figure 16: Attended Transfer Call Flow with Referred-By

F5 REFER Transferor -> Transferee

```
REFER sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha SIP/2.0
Via: SIP/2.0/TLS pc33.atlanta.example.com;branch=z9hG4bK392039842
Max-Forwards: 70
To: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
From: <sips:transferor@atlanta.example.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 REFER
<allOneLine>
Refer-To: <sips:482n4z24kdg@chicago.example.com;gr=8594958
?Replaces=090459243588173445%3Bto-tag%3D9m2n3wq%3Bfrom-tag
%3D763231&Require=replaces>
</allOneLine>
Supported: gruu, replaces, tdialog
Require: tdialog
Referred-By: <sips:transferor@atlanta.example.com>
 ;cid="20398823.2UWQFN309shb3@atlanta.example.com"
Target-Dialog: 592435881734450904;local-tag=9m2n3wq;remote-tag=763231
Contact: <sips:4889445d8kjtk3@atlanta.example.com;gr=723jd2d>
Content-Type: multipart/mixed; boundary=unique-boundary-1
Content-Length: ...

--unique-boundary-1
Content-ID: <20398823.2UWQFN309shb3@atlanta.example.com>

Content-Length: 2961
Content-Type: multipart/signed;
              protocol="application/pkcs-7-signature";
              micalg=sha1;
              boundary="----590F24D439B31E08745DEF0CD9397189"

------590F24D439B31E08745DEF0CD9397189
Content-Type: message/sipfrag

Date: Thu, 18 Sep 2003 13:07:43 GMT
<allOneLine>
Refer-To: <sips:482n4z24kdg@chicago.example.com;gr=8594958
?Replaces=090459243588173445%3B
to-tag%3D9m2n3wq%3Bfrom-tag%3D763231&Require=replaces>
</allOneLine>
Referred-By: <sips:transferor@atlanta.example.com>
 ;cid="20398823.2UWQFN309shb3@atlanta.example.com"

------590F24D439B31E08745DEF0CD9397189
Content-Type: application/pkcs-7-signature; name="smime.p7s"
```

```
   Content-Transfer-Encoding: binary
   Content-Disposition: attachment; filename="smime.p7s"

   <hex>3082088806092A86
   4886F70D010702A0820879308208750201013 10B300906052B0E03021A050030

   . . . (Signature not shown)

   8E63D306487A740A197A3970594CF47DD385643B1DC49FF767A3D2B428388966
   79089AAD95767F</hex>

   ------590F24D439B31E08745DEF0CD9397189--

   --unique_boundary-1


   F6 INVITE Transferee -> Transfer Target

   INVITE sips:482n4z24kdg@chicago.example.com;gr=8594958 SIP/2.0
   Via: SIP/2.0/TLS referee.example;branch=z9hG4bKffe209934aac
   To: <sips:482n4z24kdg@chicago.example.com;gr=8594958>
   From: <sips:transferee@biloxi.example.com>;tag=2909034023
   Call-ID: fe9023940-a3465@referee.example
   CSeq: 889823409 INVITE
   Max-Forwards: 70
   Contact: <sips:3ld812adkjw@biloxi.example.com;gr=3413kj2ha>
   Referred-By: <sips:transferor@atlanta.example.com>
      ;cid="20398823.2UWQFN309shb3@atlanta.example.com"
   Replaces:090459243588173445;to-tag=9m2n3wq;from-
    tag=76323
   Require: replaces
   Supported: gruu, replaces, tdialog
   Content-Type: multipart/mixed; boundary=my-boundary-9
   Content-Length: ...

   --my-boundary-9
   Content-Type: application/sdp

   Content-Length: 156

   v=0
   o=referee 2890844526 2890844526 IN IP4 referee.example
   s=Session SDP
   c=IN IP4 referee.example
   t=0 0
   m=audio 49172 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
```

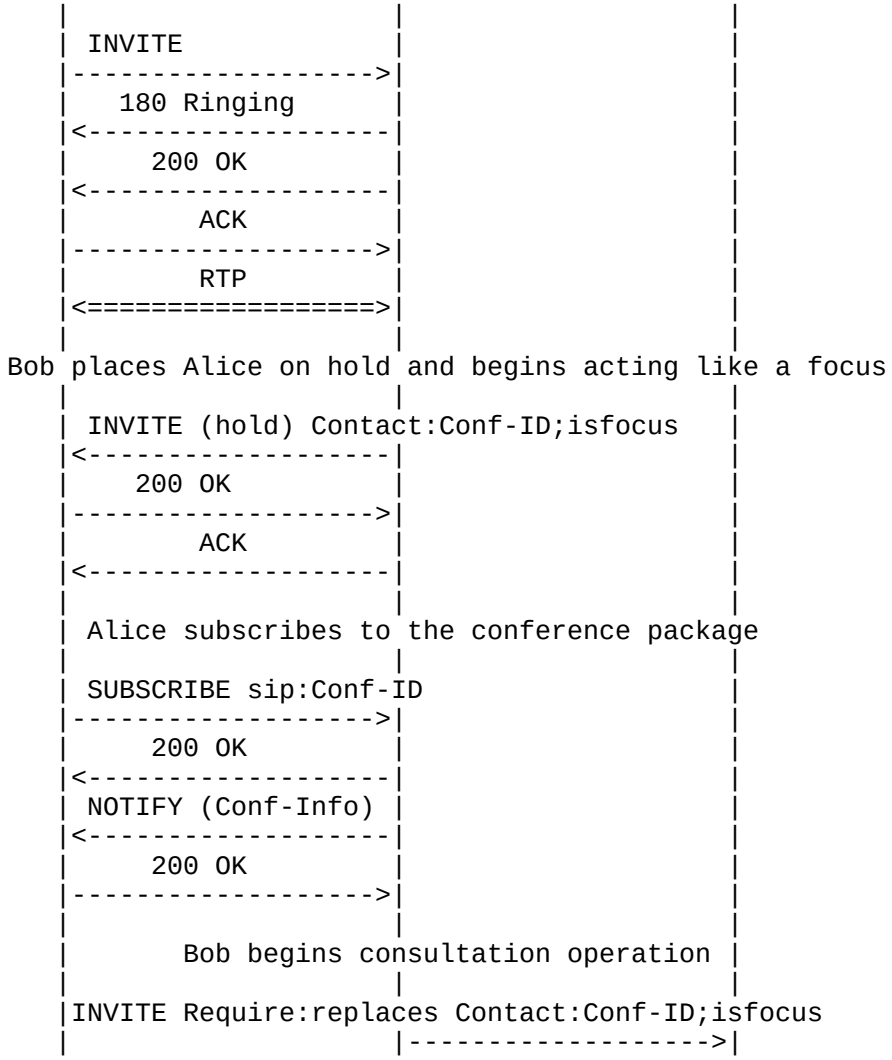
```
--my-boundary-9
Content-Length: 2961
Content-Type: multipart/signed;
             protocol="application/pkcs-7-signature";
             micalg=sha1;
             boundary="----590F24D439B31E08745DEF0CD9397189"

------590F24D439B31E08745DEF0CD9397189
Content-Type: message/sipfrag

Date: Thu, 18 Sep 2003 13:07:43 GMT

<allOneLine>
Refer-To: <sips:transfertarget@chicago.example.com;
Replaces=090459243588173445%3B
to-tag%3D9m2n3wq%3Bfrom-tag%3D763231&Require=replaces>
</allOneLine>
Referred-By: <sips:transferor@atlanta.example.com>
 ;cid="20398823.2UWQFN309shb3@atlanta.example.com"

------590F24D439B31E08745DEF0CD9397189
Content-Type: application/pkcs-7-signature; name="smime.p7s"
Content-Transfer-Encoding: binary
Content-Disposition: attachment; filename="smime.p7s"

<hex>3082088806092A86
4886F70D010702A0820879308208750201013l0B300906052B0E03021A050030

. . .  (Signature not shown)

8E63D306487A740A197A3970594CF47DD385643B1DC49FF767A3D2B428388966
79089AAD95767F</hex>

------590F24D439B31E08745DEF0CD9397189--

--my-boundary-9--
```

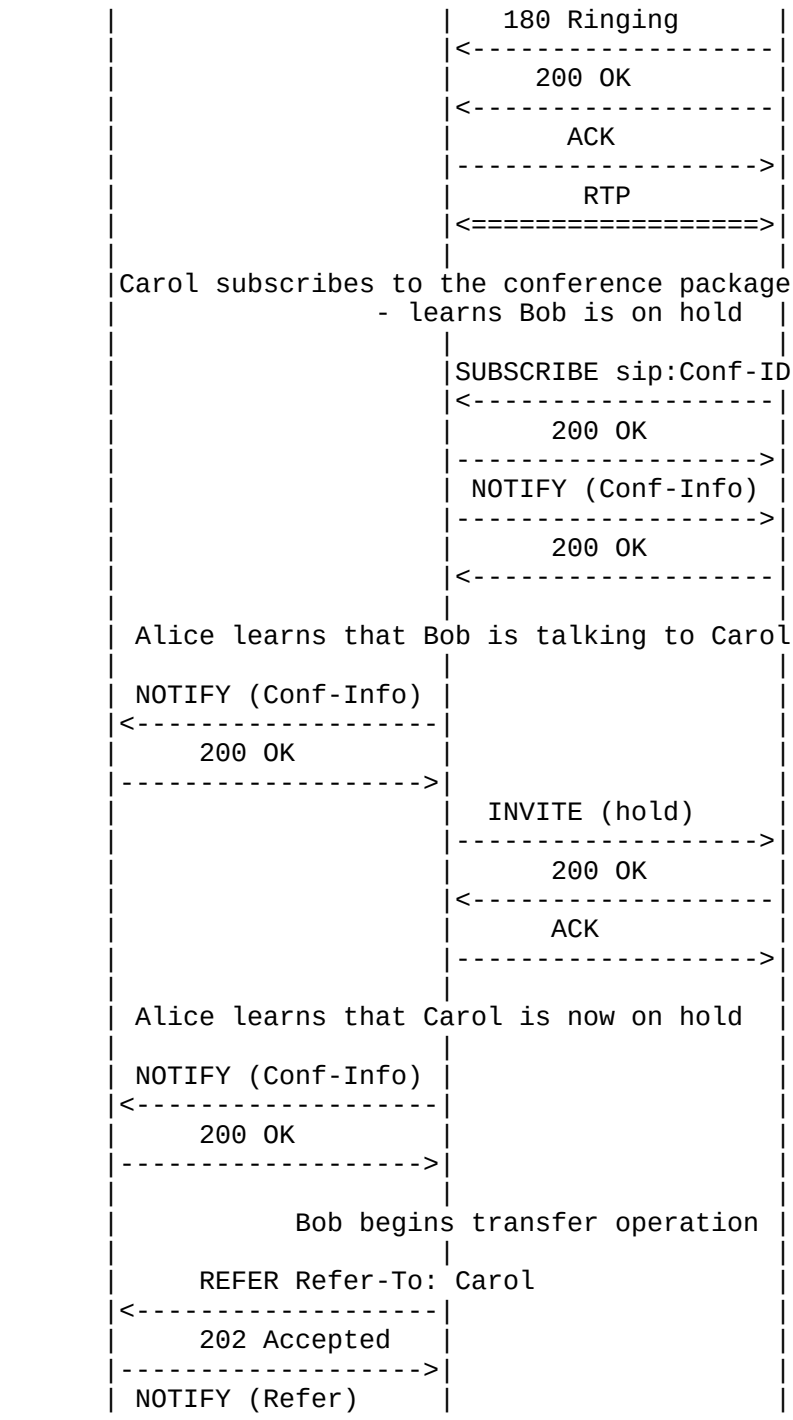
9. Transfer as an Ad Hoc Conference

In this flow, shown in Figure 17, Bob does an attended transfer of Alice to Carol. In order to keep both Alice and Carol fully informed of the nature and state of the transfer operation, Bob acts as a focus [RFC4579] and hosts an ad hoc conference involving Alice, Bob, and Carol. Alice and Carol subscribe to the conference package [RFC4575] of Bob’s focus, which allows them to know the exact status of the operation. After the transfer operation is complete, Bob deletes the conference.

This call flow meets requirement 6 of Section 4. NOTIFY messages related to the refer package are indicated as NOTIFY (refer), while NOTIFYs related to the Conference Info package are indicated as NOTIFY (Conf-Info).

Note that any type of semi-attended transfer in which media mixing or relaying could be implemented using this model. In addition to simply mixing, the focus could introduce additional media signals such as simulated ring tone or on hold announcements to improve the user experience.




180 Ringing
200 OK
ACK
RTP
Carol subscribes to the conference package
- learns Bob is on hold
SUBSCRIBE sip:Conf-ID
200 OK
NOTIFY (Conf-Info)
200 OK
Alice learns that Bob is talking to Carol
NOTIFY (Conf-Info)
200 OK
INVITE (hold)
200 OK
ACK
Alice learns that Carol is now on hold
NOTIFY (Conf-Info)
200 OK
Bob begins transfer operation
REFER Refer-To: Carol
202 Accepted
NOTIFY (Refer)

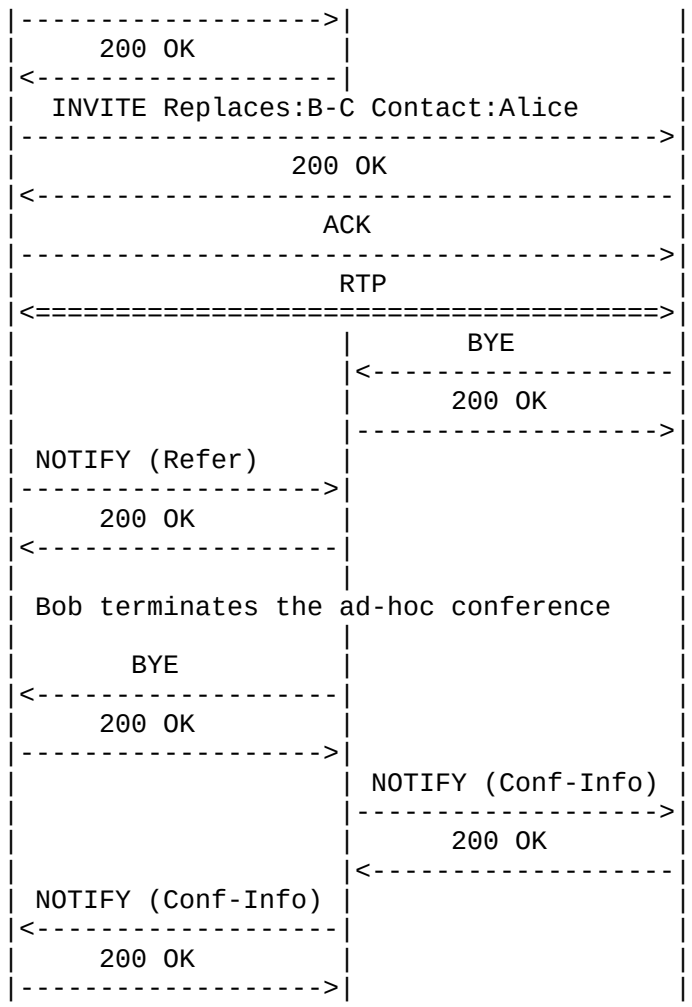
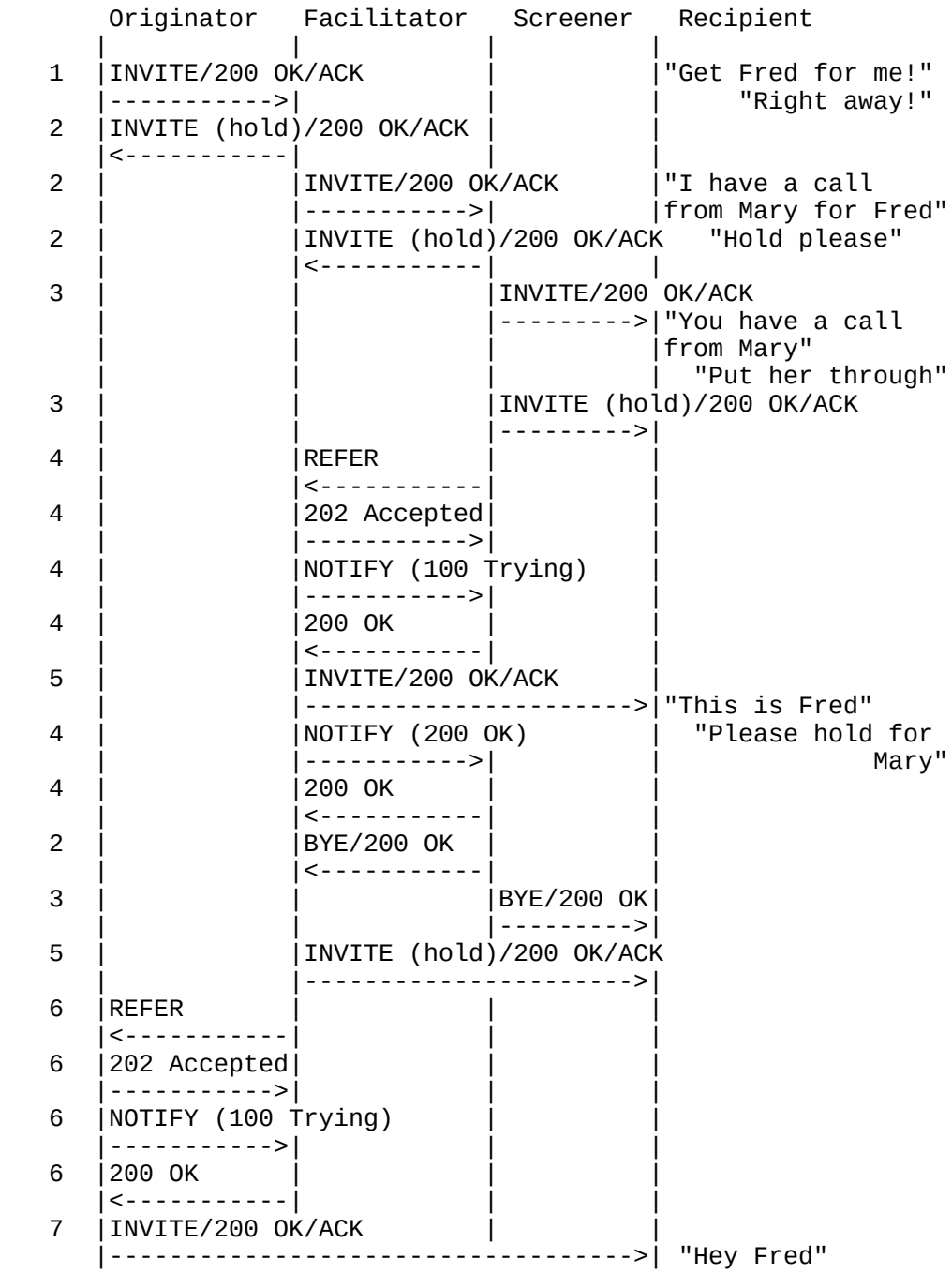


Figure 17: Attended Transfer as an Ad Hoc Conference

10. Transfer with Multiple Parties

In this example, shown in Figure 18, the Originator places a call to the Facilitator who reaches the Recipient through the Screener. The Recipient's contact information is exposed to the Facilitator and the Originator. This example is provided for clarification of the semantics of the REFER method only, and it should not be used as the design of an implementation.


Originator
Facilitator
Screener
Recipient
1 INVITE/200 OK/ACK
"Get Fred for me!"
"Right away!"
2 INVITE (hold)/200 OK/ACK
2 INVITE/200 OK/ACK
"I have a call from Mary for Fred"
2 INVITE (hold)/200 OK/ACK
"Hold please"
3 INVITE/200 OK/ACK
"You have a call from Mary"
"Put her through"
3 INVITE (hold)/200 OK/ACK
4 REFER
4 202 Accepted
4 NOTIFY (100 Trying)
4 200 OK
5 INVITE/200 OK/ACK
"This is Fred"
4 NOTIFY (200 OK)
"Please hold for Mary"
4 200 OK
2 BYE/200 OK
3 BYE/200 OK
5 INVITE (hold)/200 OK/ACK
6 REFER
6 202 Accepted
6 NOTIFY (100 Trying)
6 200 OK
7 INVITE/200 OK/ACK
"Hey Fred"

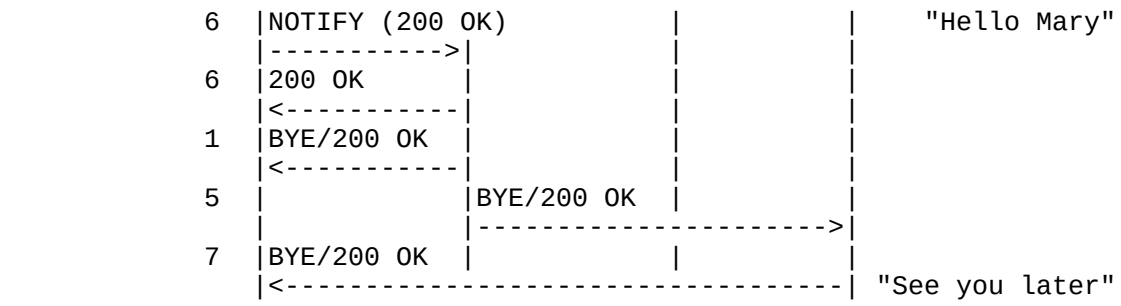


Figure 18: Transfer with Multiple Parties Example

11. Gateway Transfer Issues

A gateway in SIP acts as a User Agent. As a result, the entire preceding discussion and call flows apply equally well to gateways as native SIP endpoints. However, there are some gateway-specific issues that are documented in this section. While this discussion focuses on the common cases involving Public Switched Telephone Network (PSTN) gateways, similar situations exist for other gateways, such as H.323/SIP gateways.

11.1. Coerce Gateway Hairpins to the Same Gateway

To illustrate how a hairpin situation can occur in transfer, consider this example. The original call dialog is setup with the Transferee residing on the PSTN side of a SIP gateway. The Transferor is a SIP phone purely in the IP space. The Transfer Target is on the PSTN side of a SIP gateway as well. After completing the transfer, (regardless of consultative or blind) the Transferee is in a call with the Transfer Target (both on the PSTN side of a gateway). It is often desirable to remove the gateway(s) out of the loop. This is likely to only be possible if both legs of the target call are on the same gateway. With both legs on the same gateway, it may be able to invoke the analogous transfer on the PSTN side. Then the target call would not involve the gateway.

So the problem is how to give the proxy enough information so that it knows to route the call to the same gateway. With a simple single call that hairpins, the incoming and outgoing leg have the same dialog. The proxy should have enough information to optimize the routing.

In the consultative transfer scenario, it is desirable to coerce the consultative INVITE out the same gateway as the original call to be transferred. However, there is no way to relate the consultation with the original call. In the consultative case, the target call

INVITE includes the Replaces header, which contains dialog information that can be used to relate it to the consultation. However, there is no information that relates the target call to the original.

In the blind transfer scenario, it is desirable to coerce the target call onto the same gateway as the original call. However, the same problem exists in that the target-dialog cannot be related to the original dialog.

In either transfer scenario, it may be desirable to push the transfer operation onto the non-SIP side of the gateway. Presumably, this is not possible unless all of the legs go out the same gateway. If the gateway supports more than one trunk group, it might also be necessary to get all of the legs on the same trunk group in order to perform the transfer on the non-SIP side of the gateway.

Solutions to these gateway specific issues may involve new extensions to SIP in the future.

11.2. Consultative Turned Blind Gateway Glare

In the consultative transfer case turned blind, there is a glare-like problem. The Transferor initiates the consultation INVITE, the Transferor gets impatient and hangs up, transitioning this to a blind transfer. The Transfer Target on the gateway (connected through a PSTN switch to a single line or dumb analog phone) rings. The user answers the phone just after the CANCEL is received by the Transfer Target. The REFER and INVITE for the target call are sent. The Transferee attempts to set up the call on the PSTN side, but gets either a busy response or lands in the users voicemail as the user has the handset in hand and off hook.

This is another example of a race condition that this call flow can cause. The recommended behavior is to use the approach described in Section 7.6.

12. Security Considerations

The call transfer flows shown in this document are implemented using the REFER and Replaces call control primitives in SIP. As such, the security considerations detailed in the REFER [RFC3515] and Replaces [RFC3891] documents MUST be followed, which are briefly summarized in the following paragraphs. This document addresses the issue of protecting the Address of Record URI of a Transfer Target in Sections 7.1 and 7.2.

Any REFER request MUST be appropriately authenticated and authorized using standard SIP mechanisms or else calls may be hijacked. A User Agent may use local policy or human intervention in deciding whether or not to accept a REFER. In generating NOTIFY responses based on the outcome of the triggered request, care should be taken in constructing the message/sipfrag body to ensure that no private information is leaked.

An INVITE containing a Replaces header field SHOULD only be accepted if it has been properly authenticated and authorized using standard SIP mechanisms, and the requestor is authorized to perform dialog replacement. Special care is needed if the replaced dialog utilizes additional media streams compared to the original dialog. In this case, the user MUST authorize the addition of new media streams in a dialog replacement. For example, the same mechanism used to authorize the addition of a media stream in a re-INVITE could be used.

13. Acknowledgments

This document is a collaborative product of the SIP working group. Thanks to Rohan Mahy for his input on the use of Replaces in transfer.

14. References

14.1. Normative References

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[RFC3261] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, June 2002.

[RFC3515] Sparks, R., "The Session Initiation Protocol (SIP) Refer Method", RFC 3515, April 2003.

[RFC3891] Mahy, R., Biggs, B., and R. Dean, "The Session Initiation Protocol (SIP) "Replaces" Header", RFC 3891, September 2004.

[RFC3892] Sparks, R., "The Session Initiation Protocol (SIP) Referred-By Mechanism", RFC 3892, September 2004.

[RFC4538] Rosenberg, J., "Request Authorization through Dialog Identification in the Session Initiation Protocol (SIP)", RFC 4538, June 2006.

14.2. Informative References

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[RFC4579] Johnston, A. and O. Levin, "Session Initiation Protocol (SIP) Call Control - Conferencing for User Agents", BCP 119, RFC 4579, August 2006.

[RFC5057] Sparks, R., "Multiple Dialog Usages in the Session Initiation Protocol", RFC 5057, November 2007.

[SIP-GRUU] Rosenberg, J., "Obtaining and Using Globally Routable User Agent (UA) URIs (GRUU) in the Session Initiation Protocol (SIP)", Work in Progress, October 2007.

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