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Session Initiation Protocol Call Control - Transfer
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Abstract

This document describes providing Call Transfer capabilities in the Session Initiation Protocol (SIP). 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 [1]. This work is part of the Multiparty Call Control Framework [5].

The mechanisms discussed here are most closely related to traditional basic and consultation hold transfers. This document does not discuss transfer scenarios involving ad-hoc conferences (where all parties involved are briefly in a conference until this transferor drops out).

This document details the use of REFER method [2] and Replaces [3] header field to achieve call transfer.

A user agent that fully supports the transfer mechanisms described in this document MUST support REFER[2] and Replaces[3] in addition to [RFC 3261](#) [1].

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.
- Screener - receives a call ultimately intended for the Recipient and transfers the calling party to the Recipient if appropriate.

Recipient - the party the Originator is ultimately connected to.

3. 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 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 (this is significantly different from the requirements of the earlier BYE-Also approach to transfer).
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.

4. Using REFER to achieve Call Transfer

A REFER [2] 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 SDP body it would have if he 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 [1] or the conferencing extensions

provided by this framework.

5. 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.

The diagrams below show indicate the first line of each message. The first column of the figure shows the Call-ID 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".

5.1 Successful Transfer

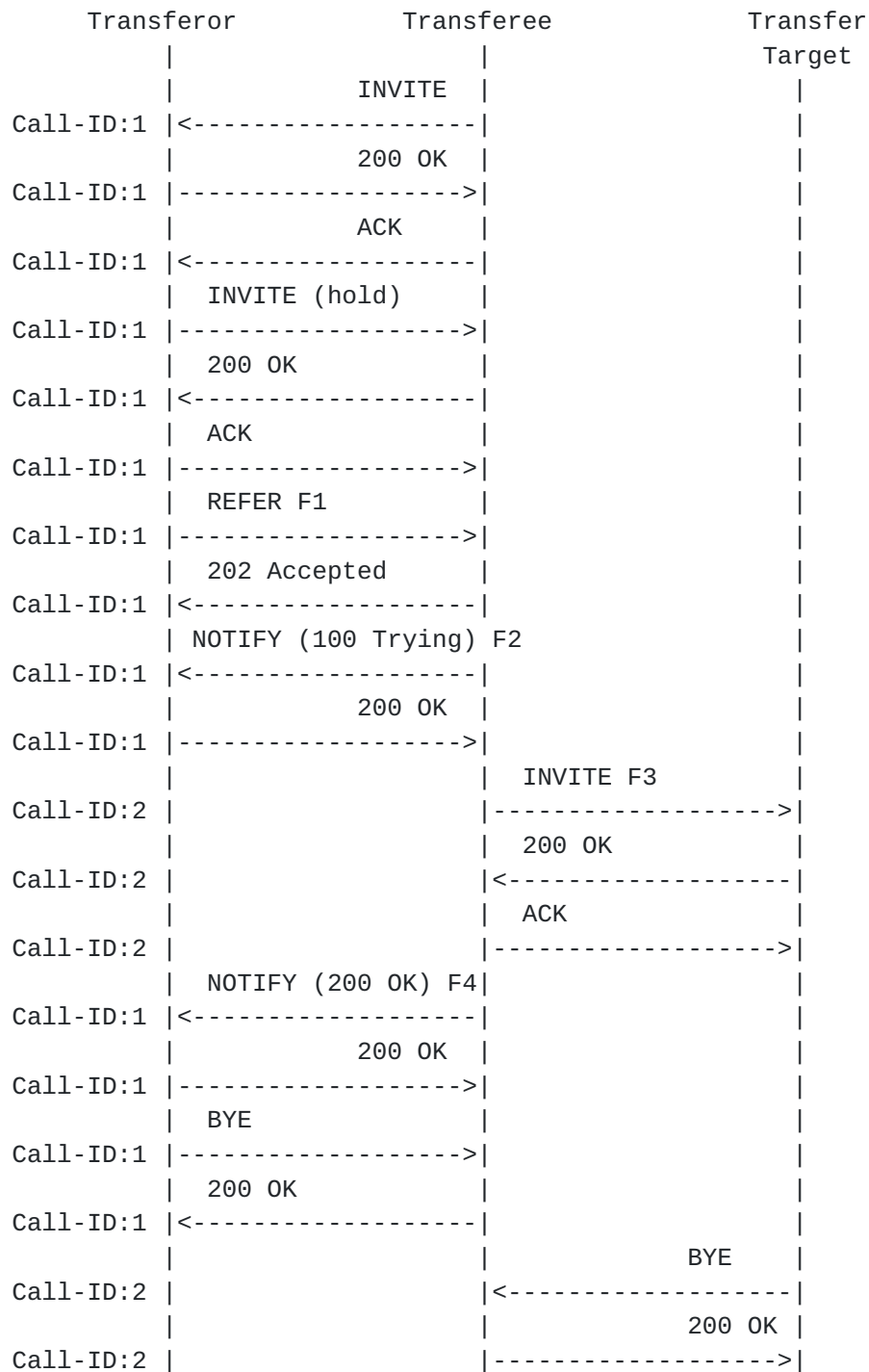


Figure 1. Basic Transfer Call Flow.

F1 REFER Transferor -> Transferee

REFER sip:transferee@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.example.com;branch=z9hG4bKna9
Max-Forwards: 70
To: <sip:transferee@biloxi.example.com>;tag=a6c85cf
From: <sip:transferor@atlanta.example.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 REFER
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Refer-To: <sip:transfertarget@chicago.example.com>
Contact: <sip:transferor@pc33.atlanta.example.com>
Content-Length: 0

F2 NOTIFY Transferee -> Transferor

NOTIFY sip:transferor@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sip:transferor@atlanta.example.com>;tag=1928301774
From: <sip:transferee@biloxi.example.com>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 73 NOTIFY
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

F3 INVITE Transferee -> Transfer Target

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

F4 NOTIFY Transferee -> Transferor

NOTIFY sip:transferor@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sip:transferor@atlanta.example.com>;tag=1928301774
From: <sip:transferee@biloxi.example.com>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 74 NOTIFY
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

5.2 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.

5.2.1 Target Busy

Transferor	Transferee	Transfer Target
	INVITE	
Call-ID:1	<-----	
	200 OK	
Call-ID:1	----->	
	ACK	
Call-ID:1	<-----	
	INVITE (hold)	
Call-ID:1	----->	
	200 OK	
Call-ID:1	<-----	
	ACK	
Call-ID:1	----->	
	REFER	
Call-ID:1	----->	
	202 Accepted	
Call-ID:1	<-----	
	NOTIFY (100 Trying)	
Call-ID:1	<-----	
	200 OK	
Call-ID:1	----->	
	INVITE	
Call-ID:2	----->	
	486 Busy Here	
Call-ID:2	<-----	
	ACK	
Call-ID:2	----->	
	NOTIFY (503 Service Unavailable) or NOTIFY (486 Busy Here)	
Call-ID:1	<-----	
	200 OK	
Call-ID:1	----->	
	INVITE (unhold)	
Call-ID:1	----->	
	200 OK	
Call-ID:1	<-----	
	ACK	


```

Call-ID:1 |----->|
          | BYE      |
Call-ID:1 |----->|
          | 200 OK   |
Call-ID:1 |<-----|

```

Figure 2. Failed Transfer - Target Busy

5.2.2 Transfer Target does not answer

Transferor	Transferee	Transfer Target
	INVITE	
Call-ID:1	<-----	
	200 OK	
Call-ID:1	----->	
	ACK	
Call-ID:1	<-----	
	INVITE (hold)	
Call-ID:1	----->	
	200 OK	
Call-ID:1	<-----	
	ACK	
Call-ID:1	----->	
	REFER	
Call-ID:1	----->	
	202 Accepted	
Call-ID:1	<-----	
	NOTIFY (100 Trying)	
Call-ID:1	<-----	
	200 OK	
Call-ID:1	----->	
	INVITE	
Call-ID:2	----->	
	180 Ringing	
Call-ID:2	<-----	
	(Transferee gets tired of waiting)	
	CANCEL	
Call-ID:2	----->	
	200 OK (CANCEL)	
Call-ID:2	<-----	
	487 Request Cancelled (INVITE)	
Call-ID:2	<-----	
	ACK	
Call-ID:2	----->	
	NOTIFY (487 Request Cancelled)	
Call-ID:1	<-----	

		200 OK		
Call-ID:1		----->		
		INVITE (unhold)		
Call-ID:1		----->		
		200 OK		
Call-ID:1		<-----		
		ACK		
Call-ID:1		----->		
		BYE		
Call-ID:1		----->		
		200 OK		
Call-ID:1		<-----		

Figure 3. Failed Transfer - Target Does Not Answer.

6. 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.

6.1 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.

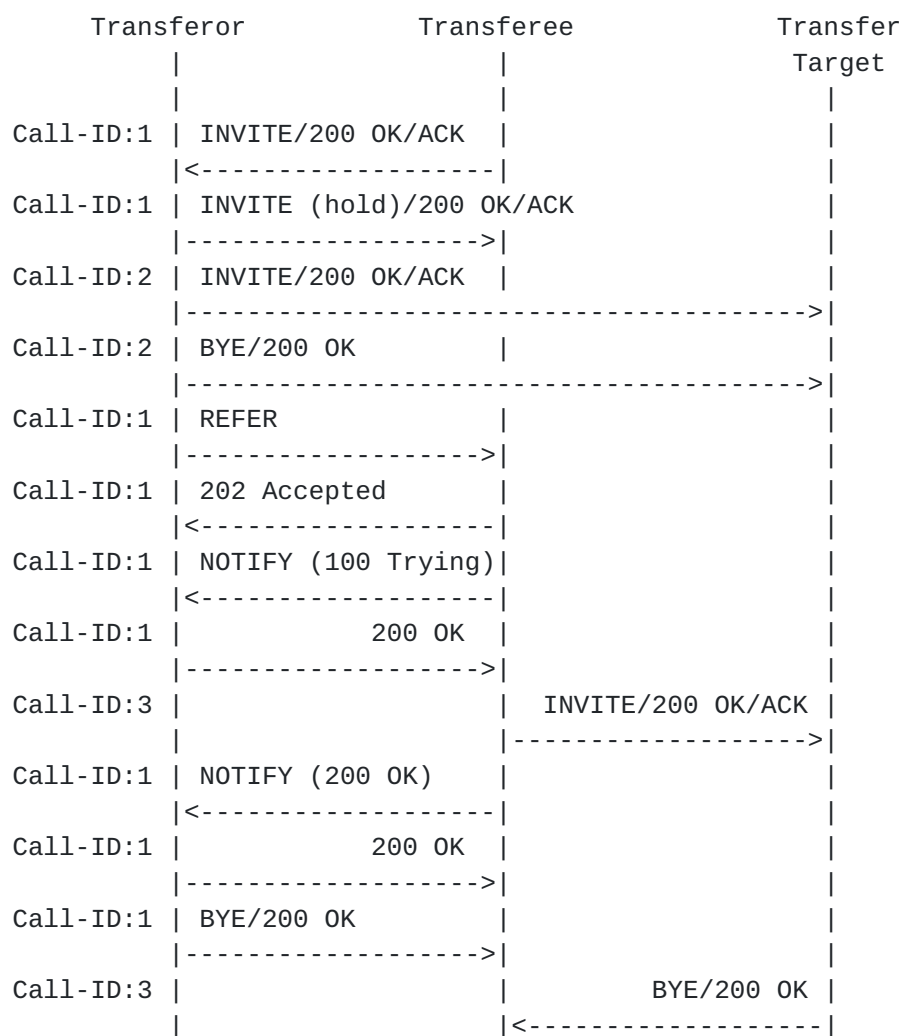


Figure 4. Transfer with Consultation Hold - Exposing Transfer Target.

[6.2](#) 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 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 [\[3\]](#) 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 [\[3\]](#) 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.

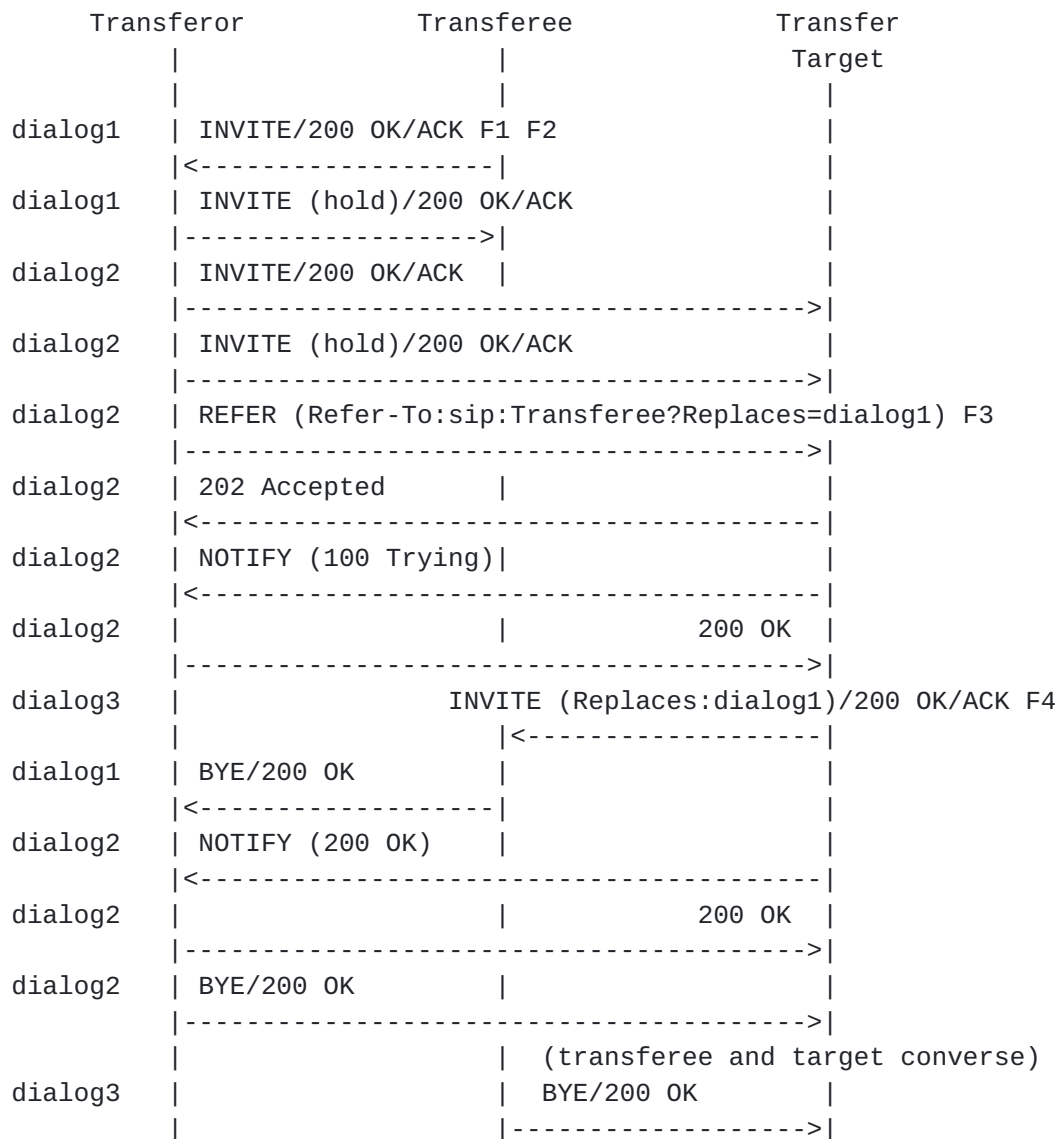


Figure 5. Transfer Protecting Transfer Target.

F1 INVITE Transferee -> Transferor

```

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

```


Contact: <sip:transferee@92.0.2.4>
Content-Type: application/sdp
Content-Length: ...

F2 200 OK Transferor -> Transferee

SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnas432
To: <sip:transferor@atlanta.example.com>;tag=31431
From: <sip:transferee@biloxi.example.com>;tag=7553452
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sip:transferor@pc33.atlanta.example.com>
Content-Type: application/sdp
Content-Length: ...

F3 REFER Transferor -> Transfer Target

REFER sip:transfertarget@client.chicago.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.example.com;branch=z9hG4bKnashds9
Max-Forwards: 70
To: <sip:transfertarget@chicago.example.com>;tag=a6c85cf
From: <sip:transferor@atlanta.example.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 REFER
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Refer-To: <sip:transferee@192.0.2.4;Replaces=
090459243588173445%3Bto-tag%3D31431%3Bfrom-tag%3D7553452>
Contact: <sip:transferor@pc33.atlanta.example.com>
Content-Length: 0

F4 INVITE Transfer Target -> Transferee

INVITE sip:transferee@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP client.chicago.com;branch=z9hG4bKnaslu84
Max-Forwards: 70
To: <sip:transferee@biloxi.example.com>
From: <sip:transfertarget@chicago.example.com>;tag=341234
Call-ID: kmzwdle3dl3d08
CSeq: 41 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY

Supported: replaces
Contact: <sip:transfertarget@client.chicago.com>
Replaces: 090459243588173445;to-tag=31431;from-tag=7553452
Content-Type: application/sdp
Content-Length: ...

6.3 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.

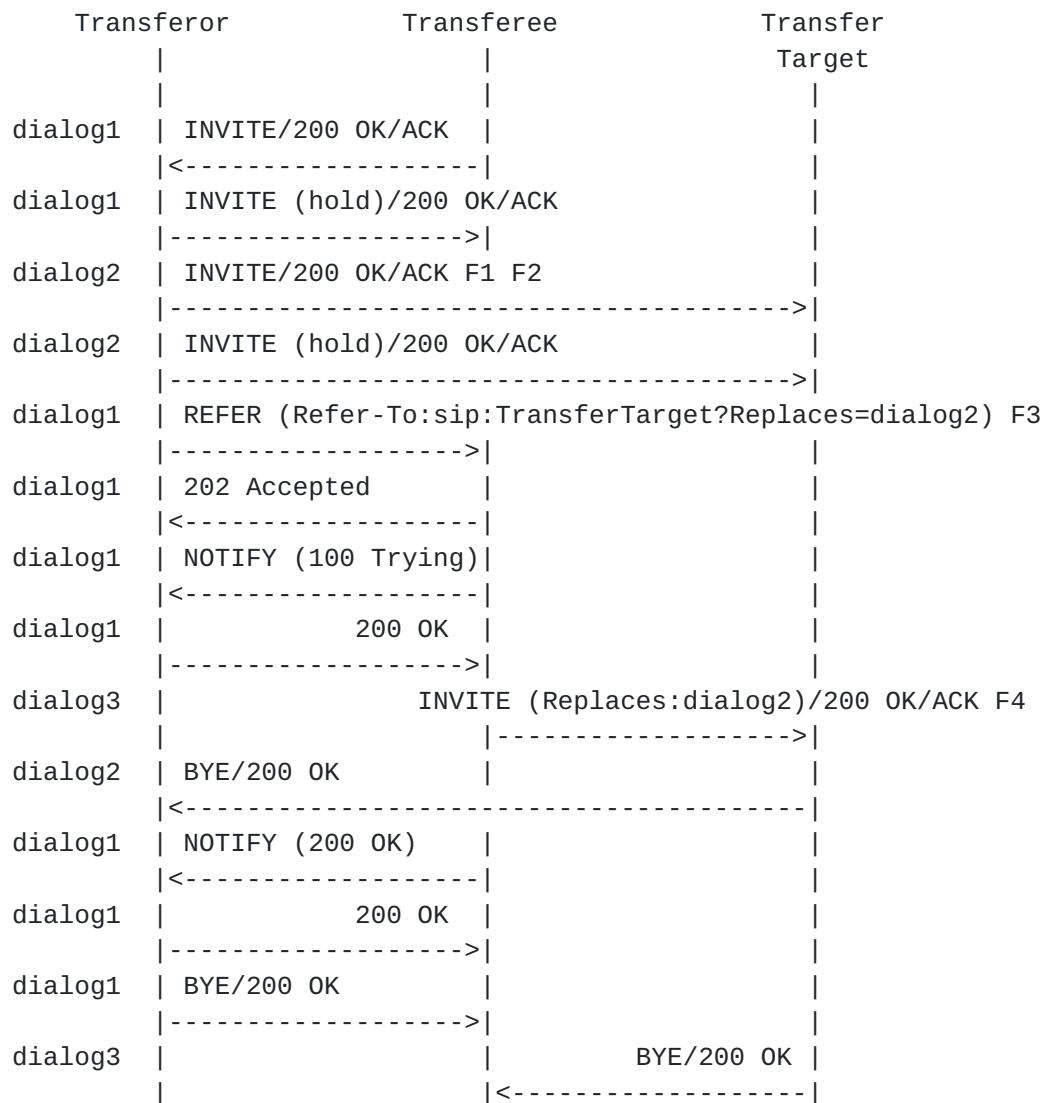


Figure 6. Attended Transfer Call Flow.

F1 INVITE Transferor -> Transfer Target

```

INVITE sip:transfertarget@chicago.example.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.example.com;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sip:transfertarget@chicago.example.com>
From: <sip:transferor@atlanta.example.com>;tag=763231
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sip: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/UDP pc33.atlanta.example.com;branch=z9hG4bKnas432
;received=192.0.2.1
To: <sip:transfertarget@chicago.example.com>;tag=9m2n3wq
From: <sip:transferor@atlanta.example.com>;tag=763231
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sip:transfertarget@client.chicago.example.com>
Content-Type: application/sdp
Content-Length: ...

F3 REFER Transferor -> Transferee

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

F4 INVITE Transferee -> Transfer Target

INVITE sip:transfertarget@client.chicago.example.com SIP/2.0
Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnaslu82
Max-Forwards: 70
To: <sip:transfertarget@chicago.example.com>
From: <sip:transferee@biloxi.example.com>;tag=954
Call-ID: kmzwdle3dl3d08
CSeq: 41 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sip:transferee@192.0.2.4>
Replaces: 090459243588173445;to-tag=9m2n3wq;from-tag=763231
Content-Type: application/sdp

Content-Length: ...

6.4 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.

	Transferor	Transferee	Transfer Target
dialog1	INVITE/200 OK/ACK		
	<-----		
dialog1	INVITE (hold)/200 OK/ACK		
	----->		
dialog2	INVITE/200 OK/ACK		
	----->		
dialog2	INVITE (hold)/200 OK/ACK		
	----->		
dialog1	REFER (Refer-To:sip:TransferTarget?Replaces=dialog2)		
	----->		
dialog1	501 Not Implemented		
	<-----		
dialog2	REFER (Refer-To:sip:Transferee?Replaces=dialog1)		
	----->		
dialog2	202 Accepted		
	<-----		
dialog2	NOTIFY (100 Trying)		
	<-----		
dialog2		200 OK	
	----->		
dialog3		INVITE (Replaces:dialog1)/200 OK/ACK	
		<-----	
dialog2	NOTIFY (200 OK)		
	<-----		
		200 OK	
	----->		
dialog1	BYE/200 OK		
	<-----		
dialog2	BYE/200 OK		
	----->		
dialog3		BYE/200 OK	
		----->	

Figure 7. Recovery when one party does not support REFER.

6.5 Attended Transfer when Contact URI is Not Globally Routable

It is a requirement of [RFC3261](#) that a Contact URI be globally routable even outside the dialog. However, due to [RFC2543](#) User Agents and some architectures (NAT/Firewall traversal, screening proxies, ALGs, etc.) this will not always be the case. As a result, the method of Attended transfer shown in Figures 6 and 7 may fail since they use the Contact URI in the Refer-To header field. Figure 8 shows such a scenario involving a Screening Proxy in which the transfer initially fails but succeeds on a second try. The failure (403 Forbidden, 404 Not Found, or a timeout after no response) response is communicated back to the Transferor. Since this may be caused by routing problems with the Contact URI, the Transferor retries the REFER this time with Refer-To containing the Address of Record (AOR) of the Target (the same URI the Transferor used to reach the Target). However, the use of the AOR URI may result in routing features being activated such as forking or sequential searching which may result in the triggered INVITE reaching the wrong UA. To prevent an incorrect UA answering the INVITE, a Require: replaces header field is included in the Refer-To. This ensures that only the UA which matches the Replaces dialog will answer the INVITE, since any incorrect UA which supports Replaces will reply with a 481 and a UA which does not support Replaces will reply with a 420.

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:sip:TargetContact?Replaces=dialog2) F3			
	----->			
dialog1	202 Accepted			

dialog1	<-----		
dialog1	NOTIFY (100 Trying)		
dialog1	<-----		
dialog1	200 OK		
dialog1	----->		
dialog3	INVITE (Replaces:dialog2)/403/ACK		
dialog1	<-----		
dialog1	NOTIFY (403 Forbidden) F4		
dialog1	<-----		
dialog1	200 OK		
dialog1	----->		
dialog1	REFER(Refer-To:sip:TargetAOR?Replaces=dialog2&Require=replaces) F5		
dialog1	<-----		
dialog1	202 Accepted		
dialog1	<-----		
dialog1	NOTIFY (100 Trying)		
dialog1	<-----		
dialog1	200 OK		
dialog1	----->		
dialog4	INVITE (Replaces:dialog2, Require=replaces)/200 OK/ACK F6		
dialog2	<-----		
dialog2	BYE/200 OK		
dialog1	<-----		
dialog1	NOTIFY (200 OK) F7		
dialog1	<-----		
dialog1	200 OK		
dialog1	----->		
dialog1	BYE/200 OK		
dialog1	----->		
dialog3		BYE/200 OK	
	<-----		

Figure 8. Attended Transfer Call Flow with non-routable Contact URI

F1 INVITE Transferor -> Transfer Target

```

INVITE sip:transfertarget@chicago.example.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.example.com;branch=z9hG4bK76
Max-Forwards: 70
To: <sip:transfertarget@chicago.example.com>
From: <sip:transferor@atlanta.example.com>;tag=763231
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sip: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/UDP pc33.atlanta.example.com;branch=z9hG4bKnas432
;received=192.0.2.1
To: <sip:transfertarget@chicago.example.com>;tag=9m2n3wq
From: <sip:transferor@atlanta.example.com>;tag=763231
Call-ID: 090459243588173445
CSeq: 29887 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Contact: <sip:transfertarget@client.chicago.example.com>
Content-Type: application/sdp
Content-Length: ...

F3 REFER Transferor -> Transferee

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

F4 NOTIFY Transferee -> Transferor

NOTIFY sip:transferor@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sip:transferor@atlanta.example.com>;tag=1928301774
From: <sip:transferee@biloxi.example.com>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 74 NOTIFY
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Event: refer;id=3112
Subscription-State: terminated;reason=noresource
Content-Type: message/sipfrag

Content-Length: ...

SIP/2.0 403 Forbidden

F5 REFER Transferor -> Transferee

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

F6 INVITE Transferee -> Transfer Target

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

F7 NOTIFY Transferee -> Transferor

NOTIFY sip:transferor@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sip:transferor@atlanta.example.com>;tag=1928301774
From: <sip:transferee@biloxi.example.com>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 76 NOTIFY
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

For a UA which requires all request to be routed through a proxy (such as for NAT/firewall traversal or screening/feature reasons), special care must be taken in constructing a globally routable Contact URI. One approach is to construct a URI which is unique for the device which resolves to the proxy. A registration would then be required to bind this URI to a URI which resolves directly to the device.

For example, consider a UA with a username carol and a hostname server51.example.com. A normal Contact would be automatically generated in the form:

Contact: sip:carol@serv51.example.com

However, if this UA requires that all requests come through a proxy server at p1.chicago.com then this Contact will not work as the proxy will be bypassed.

Consider instead a Contact of the form:

Contact: sip:serv51@example.com

in which this sip:serv51@example.com URI would be registered by the UA against a Contact:

Contact: sip:carol@serv51.example.com

which resolves directly to the UA.

This means that a UA would first register a URI that corresponds to the device. Then, it would register a users URI (AOR) and use the device URI that it registered as the Contact URI.

Other approaches may also be used to generate a globally routable Contact URI.

6.6 Aborting a Consultation Hold

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, he may wish to complete the transfer knowing that the transferee will end up eventually talking to the transfer-target's voice-mail 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 true implementation of this feature requires a short ad-hoc conference between all parties, which ensures that no media clipping occurs. This flow is outside the scope of this document.

For flows that expose the transfer target, this simply becomes a basic transfer.

This scenario is far more complicated for flows that protect the transfer target. Since no session is established between the transferor and the transfer target, the transfer target's agent would have to honor out-of-session REFERs, and somehow indicate what's happening via its user interface (this scenario is most likely to occur when the transfer-target is away from his agent).

6.7 Attended Transfer Fallback to Basic Transfer

In this flow, an attempted attended transfer fails so the transferor falls back to basic transfer. The use of OPTIONS is shown 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. 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. (Note that the same information could have been determined by including a Require: replaces in the initial INVITE in dialog2, which would have failed with a 421 response.) The Transferor then ends dialog2 by sending a BYE then sends a REFER to the Transferee using the AOR URI of the Transfer Target.

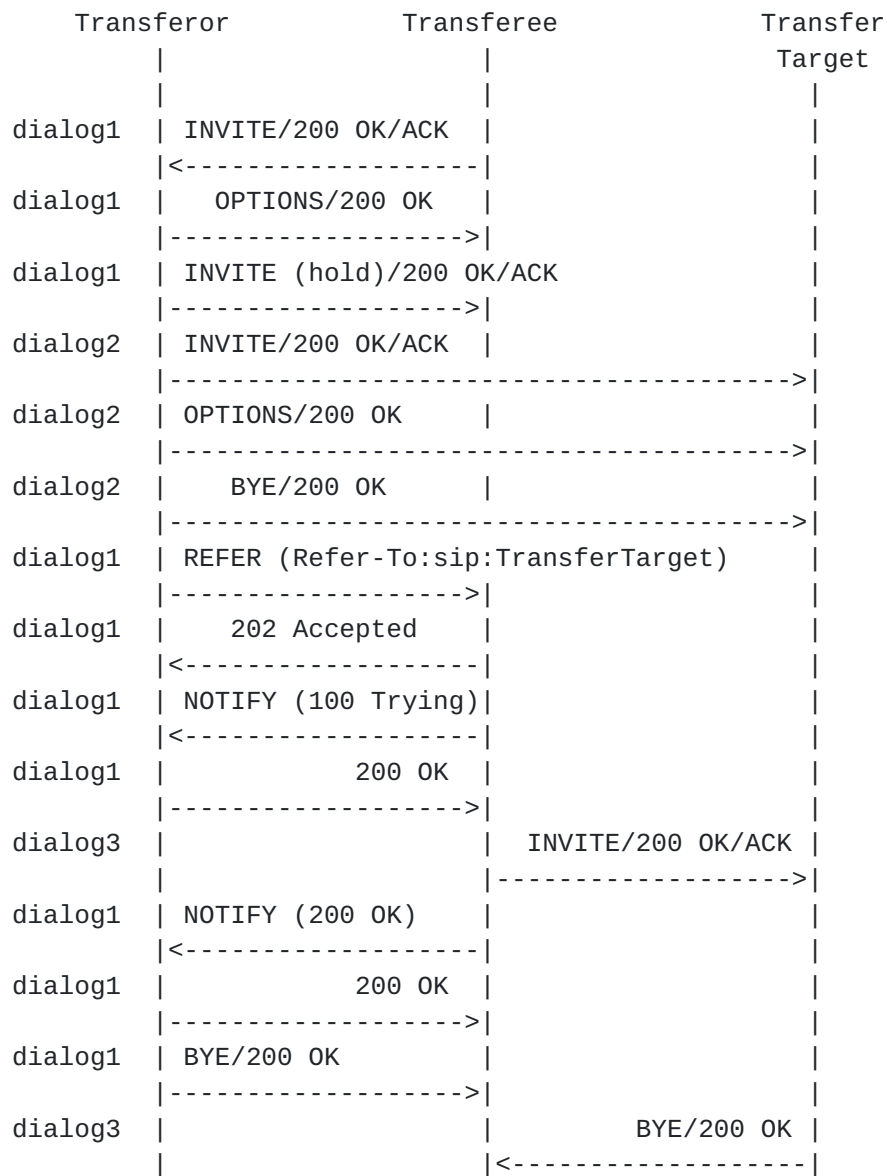


Figure 9. Attended Transfer Fallback to Basic Transfer.

7. 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 [4] 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. A call flow showing the request is in [\[4\]](#)

8. Transfer with multiple parties

In this example the Originator places 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 should not be used as the design of an implementation.

Call-ID	Originator	Facilitator	Screener	Recipient
1	INVITE/200 OK/ACK			"Get Fred for me!"
	----->			"Right away!"
1	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	
			----->	
2		REFER		
		<-----		
2		202 Accepted		
		----->		
2		NOTIFY (100 Trying)		
		----->		
2		200 OK		
		<-----		
2		INVITE/200 OK/ACK		
		----->		"This is Fred"
2		NOTIFY (200 OK)		"Please hold for
		----->		Mary"
2		200 OK		
		<-----		
2		BYE/200 OK		
		<-----		
3			BYE/200 OK	
			----->	
2		INVITE (hold)/200 OK/ACK		


```

1  | | |----->|
   | REFER | | |
   |<-----| | |
1  | 202 Accepted | | |
   |----->| | |
1  | NOTIFY (100 Trying) | | |
   |----->| | |
1  | 200 OK | | |
   |<-----| | |
1  | INVITE/200 OK/ACK | | |
   |----->| "Hey Fred"
1  | NOTIFY (200 OK) | | | "Hello Mary"
   |----->| | |
1  | 200 OK | | |
   |<-----| | |
1  | BYE/200 OK | | |
   |<-----| | |
2  | | BYE/200 OK | | |
   | | |----->|
1  | BYE/200 OK | | |
   |<-----| "See you later"

```

Figure 10. Transfer with Multiple Parties Example.

9. Changes from [draft-sipping-cc-transfer-00](#)

- o Added section on use of Referred-By header.
- o Added selected message details.
- o Added flow for attended transfer with non-globally routable Contact URI.
- o Added flow for attended transfer fallback to unattended transfer.
- o Added Security Considerations Section.

10. IANA Considerations

None.

11. 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 attacks and security approaches are those detailed in the REFER and Replaces documents 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 [6.1](#) and [6.2](#).

Any REFER request must be appropriately authenticated and authorized using standard SIP mechanisms or 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.

[12. Acknowledgments](#)

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

Normative References

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