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R. Mahy
Airespace
D. Petrie
Pingtel

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The Session Initiation Protocol (SIP) "Join" Header

Status of this Memo

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Abstract

This document defines a new header for use with SIP multi-party applications and call control. The Join header is used to logically join an existing SIP dialog with a new SIP dialog. This primitive can be used to enable a variety of features, for example: "Barge-In", answering-machine-style "Message Screening" and "Call Center Monitoring". Note that definition of these example features is non-normative.

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1. Introduction

This document describes a SIP [1] extension header field as part of the SIP multiparty applications architecture framework [12]. The Join header is used to logically join an existing SIP dialog with a new SIP dialog. This is especially useful in peer-to-peer call control environments.

One use of the "Join" header is to insert a new participant into a multimedia conversation (which may be a two-party call or a SIP conference [15]). While this functionality is already available using 3rd party call control [17], style call control, the 3pcc model requires a central point of control which may not be desirable in many environments. As such, a method of performing these same call control primitives in a distributed, peer-to-peer fashion is very desirable.

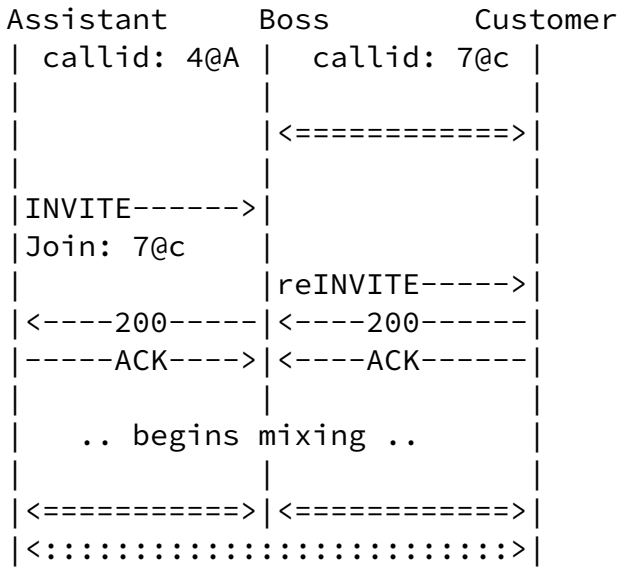
Use of an explicit Join header is needed in some cases instead of addressing an INVITE to a conference URI for the following reasons:

- A conference may not yet exist--the new invitation may be trying to join an ordinary two-party call.
- The party joining may not know if the dialog it wants to join is part of a conference.
- The party joining may not know the conference URI.

The Join header enables services such as barge-in, real-time message screening, and call center monitoring in a distributed peer-to-peer way. This list of services is not exhaustive.

For example, the Boss has an established 2-party conversation with a Customer, and using some out-of-band mechanism (e.g., voice, gestures, or email) asks an Assistant to join the conversation. The

Assistant sends an INVITE with a Join header to the Boss with the dialog information for the established dialog. The Assistant obtained this information from some other mechanism, for example a web-page, an instant message, or from the SIP session dialog package [13].



Note that this operation effectively creates a new conference. The Boss needs to cause a new conference to start (and consequently create or obtain a new conference URI). In our example, the Boss mixes all media locally, so it needs to generate a new conference URI, return the conference URI as the Contact to the Join INVITE (with the "isfocus" Contact header field parameter as defined in [6], and reINVITE or UPDATE [22] the Customer with the conference URI as the new Contact. This scenario is also discussed in more detail in [16].

2. Conventions

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 RFC 2119 [2].

This document refers frequently to the terms "confirmed dialog" and "early dialog". These are defined in Section 12 of SIP [1].

3. Applicability of RFC 2804 ("Raven")

This primitive can be used to create services which are used for monitoring purposes, however these services do not meet the definition of a wiretap according to RFC 2804 [14]. The definition from RFC 2804 is included here:

> Wiretapping is what occurs when information passed across the Internet from one party to one or more other parties is delivered to a third party:
>
> 1. Without the sending party knowing about the third party

> 2. Without any of the recipient parties knowing about the delivery to the third party
>
> 3. When the normal expectation of the sender is that the transmitted information will only be seen by the recipient parties or parties obliged to keep the information in confidence
>
> 4. When the third party acts deliberately to target the transmission of the first party, either because he is of interest, or because the second party's reception is of interest.

Specifically, item 2 of this definition does not apply to this extension, as one party is always aware of a Join request and can even decline such requests. In addition, in many applications of this primitive, some or all of the other items may not apply. For example, in many call centers which handle financial transactions, all conversations are recorded with the full knowledge and expectation of all parties involved.

4. User Agent Server Behavior: Receiving a Join Header

The Join header contains information used to match an existing SIP dialog (call-id, to-tag, and from-tag). Upon receiving an INVITE with a Join header, the UA attempts to match this information with a confirmed or early dialog. The to-tag and from-tag parameters are matched as if they were tags present in an incoming request. In other words the to-tag parameter is compared to the local tag, and

the from-tag parameter is compared to the remote tag.

If more than one Join header field is present in an INVITE, or if a Join header field is present in a request other than INVITE, the UAS MUST reject the request with a 400 Bad Request response.

The Join header has specific call control semantics. If both a Join header field and another header field with contradictory semantics (for example a Replaces [8] header field) are present in a request, the request MUST be rejected with a 400 "Bad Request" response.

If the Join header field matches more than one dialog, the UA MUST act as if no match is found.

If no match is found, but the Request-URI in the INVITE corresponds to a conference URI, the UAS MUST ignore the Join header and continue processing the INVITE as if the Join header did not exist. This allows User Agents which receive an INVITE with Join to redirect the request directly to a conference URI.

Otherwise if no match is found, the UAS rejects the INVITE and returns a 481 Call/Transaction Does Not Exist response. Likewise, if the Join header field matches a dialog which was not created with an INVITE, the UAS MUST reject the request with a 481 response.

If the Join header field matches a dialog which has already terminated, the UA SHOULD decline the request with a 603 Declined response.

If the Join header field matches an active dialog (n.b. unlike the Replaces header, the Join header has no limitation on its use with early dialogs), the UA MUST verify that the initiator of the new INVITE is authorized to join the matched dialog. If the initiator of the new INVITE has authenticated successfully as equivalent to the user who is being joined, then the join is authorized. For example, if the user being joined and the initiator of the joining dialog share the same credentials for Digest authentication [4], or they sign the join request with S/MIME [5] with the same private key and present the (same) corresponding certificate used in the original dialog, then the join is authorized.

Alternatively, the Referred-By mechanism [9] defines a mechanism that

the UAS can use to verify that a join request was sent on behalf of the other participant in the matched dialog (in this case, triggered by a REFER request). If the join request contains a Referred-By header which corresponds to the user being joined, the UA SHOULD treat the join as if it was authorized by the joined party. The Referred-By header MUST reference a corresponding, valid Refererred-By Authenticated Identity Body [10]. The UA MAY apply other local policy to authorize the remainder of the request. In other words, the UAS may apply different policy to the joined dialog than was applied to the target dialog.

The UA MAY also maintain a list of authorized entities who are allowed to join any dialog with certain characteristics (for example, all dialogs placed in the call center context of the UA). In addition, the UA MAY use other authorization mechanisms defined for this purpose in standards track extensions. For example, an extension could define a mechanism for transitively asserting authorization of a join.

If authorization is successful, the UA attempts to accept the new INVITE, and assign any mixing or conferencing resources necessary to complete the join. If the UA cannot accept the new INVITE (for example: it cannot establish required QoS or keying, or it has incompatible media), the UA MUST return an appropriate error response and MUST leave the matched dialog unchanged.

A User Agent that accepts a Join header needs to setup dialogs or conferences such that the requesting UAC is logically added to the conversation space associated with the matched dialog. Any dialogs which are already logically associated with the matched dialog in the same conversation space are included as well. For a detailed description of various conferencing mechanisms that could be used to handle a Join, please consult the SIP conferencing framework [15].

If the UAS has sufficient resources to locally handle the Join request, the UAS SHOULD accept the Join request and perform the appropriate media mixing or combining. The UAS MAY rearrange appropriate dialogs instead as described below, based on some local policy.

If the UAS does not have sufficient resources locally to handle the

request, or does not wish to use these local resources, but is aware of other resources which could be used to satisfy the request (e.g., a centralized conference server), the UA SHOULD create a conference using this resource (e.g., INVITE the conference server to obtain a conference URI), redirect the requestor to this resource, and request other participants in the same conversation space to use this resource. The UA MAY use any appropriate mechanism to transition participants to the new resource (e.g., 3xx response, 3rd-party call control reinvitiations, REFER requests, or reinvitations to a multicast group). The UA SHOULD only use mechanisms which are expected to be acceptable to the other participants. For example, the UA SHOULD NOT attempt to transition the participants to a multicast group unless the UA can reasonably expect that all the participants can support multicast.

If the UAS is incapable of satisfying the Join request, it MUST return a 488 "Not Acceptable Here" response.

5. User Agent Client Behavior: Sending a Join header

A User Agent that wishes to add a new dialog of its own to a single existing early or confirmed dialog and any associated dialogs or conferences, MAY send the target User Agent an INVITE request containing a Join header field. The UAC places the Call-ID, to-tag, and from-tag information for the target dialog in a single Join header field and sends the new INVITE to the target.

If the User Agent receives a 300-class response, and acts on this response by sending an INVITE to a Contact in the response, this redirected INVITE MUST contain the same Join header which was present in the original request. Although this is unusual, this allows INVITE requests with a Join header to be redirected before reaching the target UAS.

Note that use of the Join mechanism does not provide a way to match multiple dialogs, nor does it provide a way to match an entire call, an entire transaction, or to follow a chain of proxy forking logic.

6. Proxy behavior

Proxy Servers do not require any new behavior to support this extension. They simply pass the Join header field transparently as

described in the SIP specification.

Note that it is possible for a proxy (especially when forking based on some application layer logic, such as caller screening or time-of-day routing) to forward an INVITE request containing a Join header field to a completely orthogonal set of Contacts than the original request it was intended to replace. In this case, the INVITE request with the Join header field will fail.

7. Syntax

7.1. The Join Header

The Join header field indicates that a new dialog (created by the INVITE in which the Join header field in contained) should be joined with a dialog identified by the header field, and any associated dialogs or conferences. It is a request header only, and defined only for INVITE requests. The Join header field MAY be encrypted as part of end-to-end encryption. Only a single Join header field value may be present in a SIP request

This document adds the following entry to Table 3 of [1]. Additions to this table are also provided for extension methods defined at the time of publication of this document. This is provided as a courtesy to the reader and is not normative in any way. MESSAGE, SUBSCRIBE and NOTIFY, REFER, INFO, UPDATE, PRACK, and PUBLISH are defined respectively in [19], [20], [7], [21], [22], [23], and [24].

Header field	where	proxy	ACK	BYE	CAN	INV	OPT	REG	MSG
Join	R		-	-	-	o	-	-	-

Header field	where	SUB	NOT	REF	INF	UPD	PRA	PUB
Join	R	-	-	-	-	-	-	-

The following syntax specification uses the augmented Backus-Naur

Form (BNF) as described in RFC 2234 [3].

```
Join            = "Join" HCOLON callid *(SEMI join-param)
join-param      = to-tag / from-tag / generic-param
to-tag          = "to-tag" EQUAL token
from-tag        = "from-tag" EQUAL token
```

A Join header MUST contain exactly one to-tag and exactly one from-tag, as they are required for unique dialog matching. For compatibility with dialogs initiated by RFC 2543 [11] compliant UAs, a to-tag of zero matches both a to-tag value of zero and a null to-tag. Likewise, a from-tag of zero matches both a to-tag value of zero and a null from-tag.

Examples:

```
Join: 98732@sip.example.com
       ;from-tag=r33th4x0r
       ;to-tag=ff87ff

Join: 12adf2f34456gs5;to-tag=12345;from-tag=54321

Join: 87134@192.0.2.23;to-tag=24796;from-tag=0
```

7.2. New option tag for Require and Supported headers

This specification defines a new Require/Supported header option tag "join". UAs which support the Join header MUST include the "join" option tag in a Supported header field. UAs that want explicit failure notification if Join is not supported MAY include the "join" option in a Require header field.

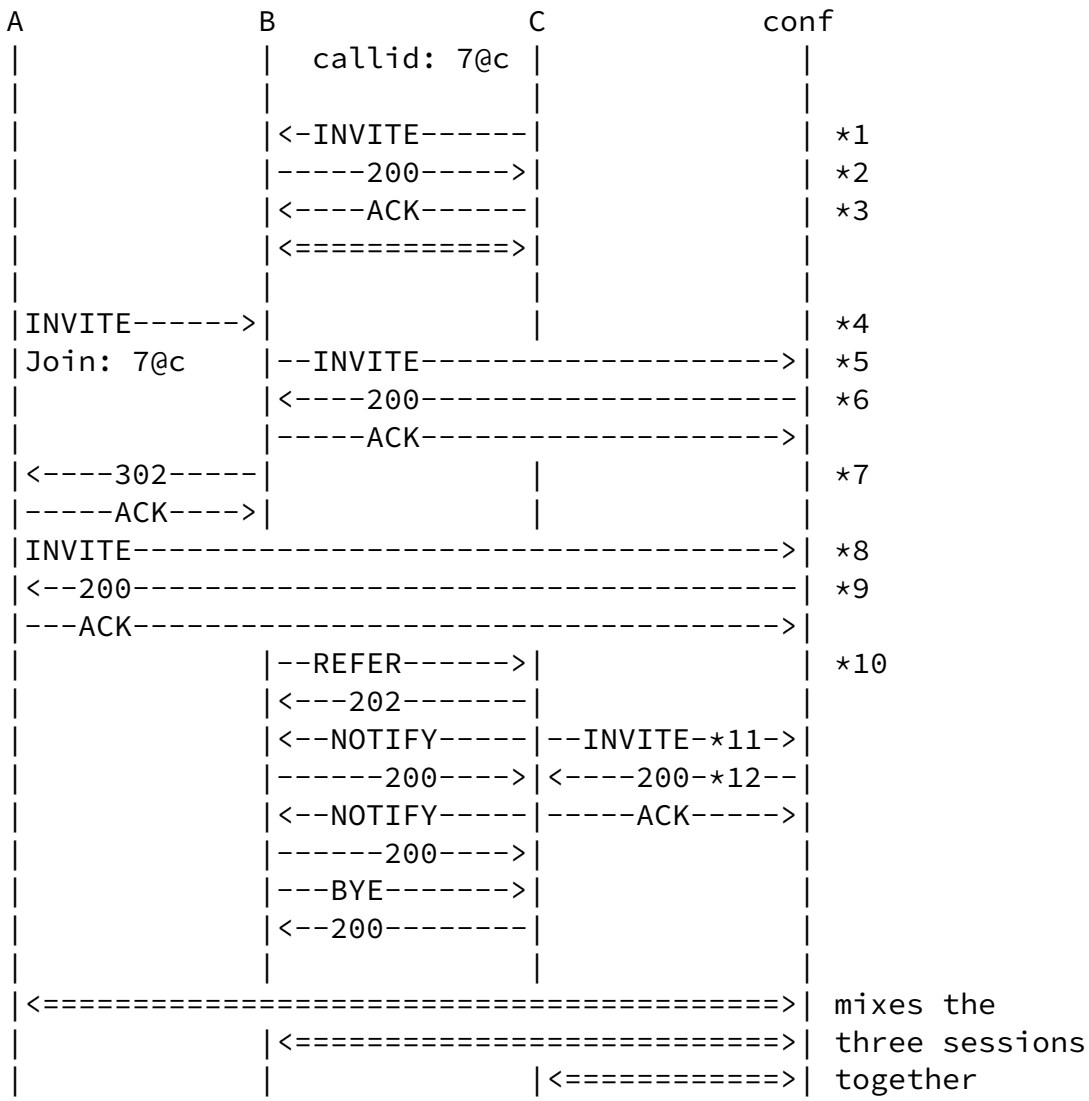
Example:

```
Require: join, 100rel
```

8. Usage Examples

The following non-normative examples are not intended to enumerate all the possibilities for the usage of this extension, but rather to provide examples or ideas only. For more examples, please see service-examples [18].

8.1. Join accepted and transitioned to central conference



The conversation now appears identical to the locally mixed one from the example in the Introduction. Details of how the Join are implemented are transparent to A. B could have used 3rd party call control instead to move the necessary sessions.

Message *1: C -> B

```
INVITE sip:bob@example.org SIP/2.0
To: <bob@example.org>
From: <carol@example.org>;tag=xyz
Call-Id: 7@c.example.org
CSeq 1 INVITE
Contact: <sip:carol@c.example.org>
```

```
Message *2: B -> C

SIP/2.0 200 OK
To: <bob@example.org>;tag=pdq
From: <carol@example.org>;tag=xyz
Call-Id: 7@c.example.org
CSeq 1 INVITE
Contact: <sip:bob@b.example.org>


Message *3: C -> B

ACK sip:carol@c.example.org SIP/2.0
To: <bob@example.org>;tag=pdq
From: <carol@example.org>;tag=xyz
Call-Id: 7@c.example.org
CSeq 1 INVITE


Message *4: A ->  B

INVITE sip:bob@b.example.org SIP/2.0
To: <sip:bob@example.org>
From: <sip:alice@example.org>;tag=iii
Call-Id: 777@a.example.org
CSeq: 1 INVITE
Contact: <sip:alice@a.example.org>
Join: 7@c.example.org;to-tag=xyz;from-tag=pdq


Message *5: B -> conf

INVITE sip:conf-factory@example.org SIP/2.0
To: <sip:conf-factory@example.org>
From: <sip:bob@example.org>;tag=abc
Call-Id: 999@b.example.org
CSeq: 1INVITE
Contact: <sip:bob@b.example.org>
```

```
Message *6: conf -> B

SIP/2.0 200 OK
To: <sip:conf-factory@example.org>;tag=def
From: <sip:bob@example.org>;tag=abc
Call-Id: 999@b.example.org
CSeq: 1INVITE
Contact: <sip:conf456@conf-srv2.example.org>;isfocus
```



```
Message *7: B -> A

SIP/2.0 302 Moved Temporarily
To: <sip:bob@example.org>
From: <sip:alice@example.org>;tag=iii
Call-Id: 777@a.example.org
CSeq: 1 INVITE
Contact: <sip:conf456@conf-srv2.example.org>;isfocus


Message *8: A -> conf

INVITE sip:conf456@conf-srv2.example.org SIP/2.0
To: <sip:bob@example.org>
From: <sip:alice@example.org>;tag=iii
Call-Id: 777@a.example.org
CSeq: 2 INVITE
Contact: <sip:alice@a.example.org>
Join: 7@c.example.org;to-tag=xyz;from-tag=pdq


Message *9: conf ->A

SIP/2.0 200 OK
To: <sip:bob@example.org>;tag=jjj
From: <sip:alice@example.org>;tag=iii
Call-Id: 777@a.example.org
CSeq: 2 INVITE
Contact: <sip:conf456@conf-srv2.example.org>;isfocus


Message *10: B -> C
```

```
REFER sip:carol@c.example.org SIP/2.0
To: <carol@example.org>;tag=xyz
From: <bob@example.org>;tag=pdq
Call-Id: 7@c.example.org
CSeq: 1 REFER
Contact: <sip:bob@b.example.org>
Refer-To: <sip:conf456@conf-srv2.example.org>
Referred-By: <sip:bob@b.example.org>
```

Message *11: C -> conf

```
INVITE sip:conf456@conf-srv2.example.org SIP/2.0
To: <sip:conf456@conf-srv2.example.org>
From: <carol@example.org>;tag=mmm
```

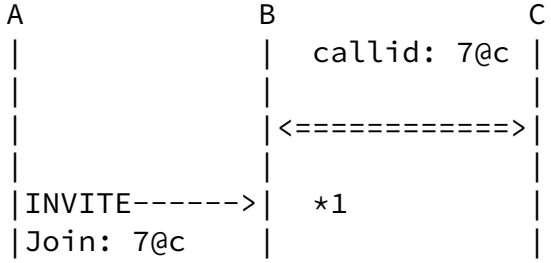


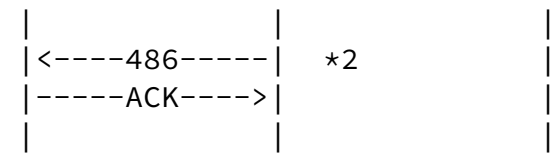
```
Call-Id: 34343@c.example.com
CSeq: 1 INVITE
Contact: <sip:carol@c.example.com>
Referred-By: <sip:bob@b.example.org>
```

Message *12: C -> conf

```
SIP/2.0 200 OK
To: <sip:conf456@conf-srv2.example.org>
From: <carol@example.org>;tag=mmm
Call-Id: 34343@c.example.com
CSeq: 1 INVITE
Contact: <sip:conf456@conf-srv2.example.org>;isfocus
Referred-By: <sip:bob@b.example.org>
```

8.2. Join rejected





In this example B is Busy (does not want to be disturbed), and therefore does not wish to add A. B could also decline the request with a 603 response.

Message *1: A -> B

```
INVITE sip:bob@b.example.org SIP/2.0
To: <sip:bob@example.org>
From: <sip:alice@example.org>;tag=iii
Call-Id: 777@a.example.org
CSeq: 1 INVITE
Contact: <sip:alice@a.example.org>
Join: 7@c.example.org;to-tag=xyz;from-tag=pdq
```


Message *2: B -> A

```
SIP/2.0 486 Busy
To: <sip:bob@example.org>
From: <sip:alice@example.org>;tag=iii
Call-Id: 777@a.example.org
CSeq: 1 INVITE
```

9. Security Considerations

The extension specified in this document significantly changes the relative security of SIP devices. Currently in SIP, even if an eavesdropper learns the Call-ID, To, and From headers of a dialog, they cannot easily modify or destroy that dialog if Digest authentication or end-to-end message integrity are used.

This extension can be used to insert or monitor potentially sensitive content in a multimedia conversation. As such, invitations with the

Join header MUST only be accepted if the peer requesting replacement has been properly authenticated using a standard SIP mechanism (Digest or S/MIME), and authorized to be joined with the target dialog. (All SIP implementations are already required to support Digest Authentication.) Generally authorization for joins are configured as a matter of local policy as long-duration persistent relationships.

For example, the UAs used by call center agents might be configured with a list of identities who could join their calls (supervisors and any call center monitoring User Agents). Alternatively the call center agents might rely on transitive authorization assertions from a (shorter) list of authorized hosts (e.g., a certificate authority). For answering-machine-style message screening this is even easier. Presumably the user screening their messages already has some credentials with their messaging server.

Some mechanisms for obtaining the dialog information needed by the Join header (Call-ID, to-tag, and from-tag) include URIs on a web page, subscriptions to an appropriate event package, and notifications after a REFER request. Use of end-to-end security mechanisms to integrity protect and encrypt this information is also RECOMMENDED.

This extension was designed to take advantage of future signature or authorization schemes defined by standards track extensions. In general, call control features would benefit considerably from such work.

Section 4 describes specific mechanisms for authorization using Digest Authentication and S/MIME (RFC 3261) and Referred-by [9], the currently available capabilities in SIP.

10. IANA Considerations

10.1. Registration of "Join" SIP header

Name of Header: Join

Short form: none

Normative description: section 7.1 of this document

10.2. Registration of "join" SIP Option-tag

Name of option: join

Description: Support for the SIP Join header

SIP headers defined: Join

Normative description: This document

11. Acknowledgments

Thanks to Robert Sparks, Alan Johnston, and Ben Campbell and many other members of the SIP WG for their continued support of the cause of distributed call control in SIP.

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13. Authors' Addresses

Rohan Mahy
Airespace
110 Nortech Parkway
San Jose, CA 95134
USA

EMail: rohan@airespace.com

Dan Petrie
Pingtel
400 West Cummings Park, Suite 2200
Woburn, MA 01801
USA

EMail: dpetrie@pingtel.com

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