**[MS-OCPSTN]:**

**Session Initiation Protocol (SIP) for PSTN Calls Extensions**

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Table of Contents

[1 Introduction 7](#_Toc462894585)

[1.1 Glossary 7](#_Toc462894586)

[1.2 References 9](#_Toc462894587)

[1.2.1 Normative References 9](#_Toc462894588)

[1.2.2 Informative References 9](#_Toc462894589)

[1.3 Overview 10](#_Toc462894590)

[1.4 Relationship to Other Protocols 11](#_Toc462894591)

[1.5 Prerequisites/Preconditions 11](#_Toc462894592)

[1.6 Applicability Statement 11](#_Toc462894593)

[1.7 Versioning and Capability Negotiation 11](#_Toc462894594)

[1.8 Vendor-Extensible Fields 12](#_Toc462894595)

[1.9 Standards Assignments 12](#_Toc462894596)

[2 Messages 13](#_Toc462894597)

[2.1 Transport 13](#_Toc462894598)

[2.2 Message Syntax 13](#_Toc462894599)

[2.2.1 isGateway 13](#_Toc462894600)

[2.2.2 phone-context 13](#_Toc462894601)

[2.2.3 ms-call-source 14](#_Toc462894602)

[2.2.4 ms-early-media 14](#_Toc462894603)

[2.2.5 Anonymous Phone URI 14](#_Toc462894604)

[2.2.6 ms-bypass 14](#_Toc462894605)

[2.2.7 ms-accepted-content-id 14](#_Toc462894606)

[2.2.8 ms-trunking-peer 15](#_Toc462894607)

[2.2.9 ms-mediation-generated 15](#_Toc462894608)

[3 Protocol Details 16](#_Toc462894609)

[3.1 isGateway Details 16](#_Toc462894610)

[3.1.1 Abstract Data Model 16](#_Toc462894611)

[3.1.2 Timers 16](#_Toc462894612)

[3.1.3 Initialization 16](#_Toc462894613)

[3.1.4 Higher-Layer Triggered Events 16](#_Toc462894614)

[3.1.5 Message Processing Events and Sequencing Rules 16](#_Toc462894615)

[3.1.6 Timer Events 16](#_Toc462894616)

[3.1.7 Other Local Events 16](#_Toc462894617)

[3.2 phone-context Details 16](#_Toc462894618)

[3.2.1 Abstract Data Model 17](#_Toc462894619)

[3.2.2 Timers 17](#_Toc462894620)

[3.2.3 Initialization 17](#_Toc462894621)

[3.2.4 Higher-Layer Triggered Events 17](#_Toc462894622)

[3.2.5 Message Processing Events and Sequencing Rules 18](#_Toc462894623)

[3.2.6 Timer Events 18](#_Toc462894624)

[3.2.7 Other Local Events 18](#_Toc462894625)

[3.3 ms-call-source Details 18](#_Toc462894626)

[3.3.1 Abstract Data Model 18](#_Toc462894627)

[3.3.2 Timers 18](#_Toc462894628)

[3.3.3 Initialization 18](#_Toc462894629)

[3.3.4 Higher-Layer Triggered Events 18](#_Toc462894630)

[3.3.5 Message Processing Events and Sequencing Rules 19](#_Toc462894631)

[3.3.6 Timer Events 19](#_Toc462894632)

[3.3.7 Other Local Events 19](#_Toc462894633)

[3.4 ms-early-media Details 19](#_Toc462894634)

[3.4.1 Abstract Data Model 19](#_Toc462894635)

[3.4.2 Timers 19](#_Toc462894636)

[3.4.3 Initialization 19](#_Toc462894637)

[3.4.4 Higher-Layer Triggered Events 19](#_Toc462894638)

[3.4.5 Message Processing Events and Sequencing Rules 19](#_Toc462894639)

[3.4.6 Timer Events 20](#_Toc462894640)

[3.4.7 Other Local Events 20](#_Toc462894641)

[3.5 Anonymous Phone URI Details 20](#_Toc462894642)

[3.5.1 Abstract Data Model 20](#_Toc462894643)

[3.5.2 Timers 20](#_Toc462894644)

[3.5.3 Initialization 20](#_Toc462894645)

[3.5.4 Higher-Layer Triggered Events 20](#_Toc462894646)

[3.5.5 Message Processing Events and Sequencing Rules 20](#_Toc462894647)

[3.5.6 Timer Events 20](#_Toc462894648)

[3.5.7 Other Local Events 20](#_Toc462894649)

[3.6 ms-bypass Details 20](#_Toc462894650)

[3.6.1 Abstract Data Model 21](#_Toc462894651)

[3.6.2 Timers 21](#_Toc462894652)

[3.6.3 Initialization 21](#_Toc462894653)

[3.6.4 Higher-Layer Triggered Events 21](#_Toc462894654)

[3.6.5 Message Processing Events and Sequencing Rules 21](#_Toc462894655)

[3.6.6 Timer Events 21](#_Toc462894656)

[3.6.7 Other Local Events 21](#_Toc462894657)

[3.7 ms-accepted-content-id Details 21](#_Toc462894658)

[3.7.1 Abstract Data Model 21](#_Toc462894659)

[3.7.2 Timers 22](#_Toc462894660)

[3.7.3 Initialization 22](#_Toc462894661)

[3.7.4 Higher-Layer Triggered Events 22](#_Toc462894662)

[3.7.5 Message Processing Events and Sequencing Rules 22](#_Toc462894663)

[3.7.6 Timer Events 22](#_Toc462894664)

[3.7.7 Other Local Events 22](#_Toc462894665)

[3.8 ms-trunking-peer Details 22](#_Toc462894666)

[3.8.1 Abstract Data Model 22](#_Toc462894667)

[3.8.2 Timers 22](#_Toc462894668)

[3.8.3 Initialization 22](#_Toc462894669)

[3.8.4 Higher-Layer Triggered Events 22](#_Toc462894670)

[3.8.5 Message Processing Events and Sequencing Rules 22](#_Toc462894671)

[3.8.6 Timer Events 22](#_Toc462894672)

[3.8.7 Other Local Events 23](#_Toc462894673)

[3.9 ms-mediation-generated Details 23](#_Toc462894674)

[3.9.1 Abstract Data Model 23](#_Toc462894675)

[3.9.2 Timers 23](#_Toc462894676)

[3.9.3 Initialization 23](#_Toc462894677)

[3.9.4 Higher-Layer Triggered Events 23](#_Toc462894678)

[3.9.5 Message Processing Events and Sequencing Rules 23](#_Toc462894679)

[3.9.6 Timer Events 23](#_Toc462894680)

[3.9.7 Other Local Events 23](#_Toc462894681)

[4 Protocol Examples 24](#_Toc462894682)

[4.1 isGateway SIP Contact Header Parameter 24](#_Toc462894683)

[4.1.1 Inbound Call 24](#_Toc462894684)

[4.1.1.1 Step 3: INVITE Message Is Received by the UAC 24](#_Toc462894685)

[4.1.1.2 Step 7: 200 Message Is Sent from the UAC 25](#_Toc462894686)

[4.1.2 Outbound Call 26](#_Toc462894687)

[4.1.2.1 Step 1: INVITE Message Is Sent from the UAC 26](#_Toc462894688)

[4.1.2.2 Step 13: 200 Message Is Received by the UAC 27](#_Toc462894689)

[4.2 phone-context SIP URI Parameter 28](#_Toc462894690)

[4.2.1 Inbound Call 28](#_Toc462894691)

[4.2.1.1 Step 3: INVITE Message Is Received by the UAC 28](#_Toc462894692)

[4.2.1.2 Step 7: 200 Message Is Sent from the UAC 29](#_Toc462894693)

[4.2.2 Outbound Call 30](#_Toc462894694)

[4.2.2.1 Step 1: INVITE Message Is Sent from the UAC 30](#_Toc462894695)

[4.2.2.2 Step 13: 200 Message Is Received by the UAC 31](#_Toc462894696)

[4.3 ms-call-source SIP Header 32](#_Toc462894697)

[4.3.1 Inbound Call 32](#_Toc462894698)

[4.3.1.1 Step 2: INVITE Message Is Received by the UAC 32](#_Toc462894699)

[4.3.1.2 Step 8: INVITE Message Is Received by the UAC 33](#_Toc462894700)

[4.3.1.3 Step 9: 605 Message Is Sent from the UAC 33](#_Toc462894701)

[4.3.1.4 Step 12: 200 Message Is Sent from the UAC 34](#_Toc462894702)

[4.3.2 Outbound Call 34](#_Toc462894703)

[4.4 ms-early-media SIP Supported Header Option Tag 34](#_Toc462894704)

[4.4.1 Inbound Call 34](#_Toc462894705)

[4.4.2 Outbound Call 35](#_Toc462894706)

[4.4.2.1 Step 1: INVITE Is Sent from the UAC 35](#_Toc462894707)

[4.4.2.2 Step 7: 183 Message Is Received by the UAC 36](#_Toc462894708)

[4.4.2.3 Step 13: 200 Message Is Received by the UAC 37](#_Toc462894709)

[4.5 ms-bypass SIP Supported Header Option Tag 38](#_Toc462894710)

[4.5.1 Inbound Call 38](#_Toc462894711)

[4.5.1.1 Step 6: INVITE Message Is Received by the Protocol Client 39](#_Toc462894712)

[4.5.1.2 Step 17: 200 Message Is Sent by the Protocol Client 40](#_Toc462894713)

[4.5.2 Outbound Call 41](#_Toc462894714)

[4.5.2.1 Step 1: INVITE Message Is Sent by the Protocol Client 42](#_Toc462894715)

[4.5.2.2 Step 13: 200 OK Message Is Received by the Protocol Client 44](#_Toc462894716)

[4.6 ms-accepted-content-id SIP Header 45](#_Toc462894717)

[4.6.1 Inbound Call 45](#_Toc462894718)

[4.6.1.1 Step 6: INVITE Message Is Received by the Protocol Client 45](#_Toc462894719)

[4.6.1.2 Step 17: 200 Message Is Sent by the Protocol Client 47](#_Toc462894720)

[4.6.2 Outbound Call 48](#_Toc462894721)

[4.6.2.1 Step 1: INVITE Message Is Sent by the Protocol Client 48](#_Toc462894722)

[4.6.2.2 Step 13: 200 Message Is Received by the Protocol Client 51](#_Toc462894723)

[4.7 ms-trunking-peer SIP Header 51](#_Toc462894724)

[4.7.1 Inbound Call 52](#_Toc462894725)

[4.7.1.1 Step 6: INVITE Message Is Received by the Protocol Client 52](#_Toc462894726)

[4.7.2 Outbound Call 53](#_Toc462894727)

[4.7.2.1 Step 13: 200 Message Is Received by the Protocol Client 54](#_Toc462894728)

[4.8 ms-mediation-generated SIP Header 54](#_Toc462894729)

[4.8.1 Outbound Call 55](#_Toc462894730)

[4.8.1.1 Step 5: 183 Message Is Received by the Protocol Client 55](#_Toc462894731)

[4.8.1.2 Step 10: 180 Message Is Received by the Protocol Client 56](#_Toc462894732)

[5 Security 57](#_Toc462894733)

[5.1 Security Considerations for Implementers 57](#_Toc462894734)

[5.2 Index of Security Parameters 57](#_Toc462894735)

[6 Appendix A: Product Behavior 58](#_Toc462894736)

[7 Change Tracking 60](#_Toc462894737)

[8 Index 61](#_Toc462894738)

# Introduction

The [**Session Initiation Protocol (SIP)**](#gt_586971aa-3b65-4de3-be93-1a9756777d89) for public switched telephone network (PSTN) Calls Extensions protocol consists of proprietary extensions applicable for interfacing a protocol client with other traditional telephony networks, such as the public switched telephone network (PSTN) and an enterprise private branch exchange (PBX) or IP-PBX.

Sections 1.5, 1.8, 1.9, 2, and 3 of this specification are normative. All other sections and examples in this specification are informative.

## Glossary

This document uses the following terms:

**200 OK**: A response to indicate that the request has succeeded.

**answer**: A message that is sent in response to an [**offer**](#gt_83dc4e88-7966-41f5-a35d-61bb5daee80b) that is received from an offerer.

**Augmented Backus-Naur Form (ABNF)**: A modified version of Backus-Naur Form (BNF), commonly used by Internet specifications. ABNF notation balances compactness and simplicity with reasonable representational power. ABNF differs from standard BNF in its definitions and uses of naming rules, repetition, alternatives, order-independence, and value ranges. For more information, see [[RFC5234]](http://go.microsoft.com/fwlink/?LinkId=123096).

**call**: A communication between peers that is configured for a multimedia conversation.

**dial plan**: The rules that govern the translation of dial strings into [**SIP**](#gt_586971aa-3b65-4de3-be93-1a9756777d89) and tel [**URIs**](#gt_e18af8e8-01d7-4f91-8a1e-0fb21b191f95), either global or local, as described in [[RFC3966]](http://go.microsoft.com/fwlink/?LinkId=114246).

**dial string**: The numbers, symbols, and pauses that users enter to place a phone call. It is consumed by one or more network entities and understood in the context of the configuration of those entities. It is used to generate an address-of-record or identifier to route a [**call**](#gt_9c30971d-7054-4d6b-90a7-c7410283f71d).

**dialog**: A peer-to-peer [**Session Initiation Protocol (SIP)**](#gt_586971aa-3b65-4de3-be93-1a9756777d89) relationship that exists between two user agents and persists for a period of time. A dialog is established by [**SIP messages**](#gt_2690e796-e281-48f3-ba0e-1f9acdb3ba8c), such as a 2xx response to an INVITE request, and is identified by a call identifier, a local tag, and a remote tag.

**domain**: A set of users and computers sharing a common namespace and management infrastructure. At least one computer member of the set must act as a domain controller (DC) and host a member list that identifies all members of the domain, as well as optionally hosting the Active Directory service. The domain controller provides authentication (2) of members, creating a unit of trust for its members. Each domain has an identifier that is shared among its members. For more information, see [[MS-AUTHSOD]](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-AUTHSOD%5d.pdf#Section_953d700a57cb4cf7b0c3a64f34581cc9) section 1.1.1.5 and [[MS-ADTS]](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-ADTS%5d.pdf#Section_d243592709994c628c6d13ba31a52e1a).

**E.164**: An international public telecommunication numbering plan that is used in the public switched telephone network (PSTN) and some data networks. It defines the format of telephone numbers. E.164 numbers can have a maximum of 15 digits and typically are written with a plus sign (+) prefix.

**early media**: Media, such as audio and video, that is exchanged before a specific session is accepted by the called user. During a dialog, early media occurs when the initial INVITE is sent, until the [**user agent server (UAS)**](#gt_6f39aa0f-2438-4c06-8ccc-5d36b6e50a28) generates a final response.

**fully qualified domain name (FQDN)**: An unambiguous domain name (2) that gives an absolute location in the Domain Name System's (DNS) hierarchy tree, as defined in [[RFC1035]](http://go.microsoft.com/fwlink/?LinkId=90264) section 3.1 and [[RFC2181]](http://go.microsoft.com/fwlink/?LinkId=127732) section 11.

**gateway**: A network edge device that bridges Microsoft Office Communications Server protocols with legacy telephony networks protocols.

**INVITE**: A [**Session Initiation Protocol (SIP)**](#gt_586971aa-3b65-4de3-be93-1a9756777d89) method that is used to invite a user or a service to participate in a session.

**IP-PBX**: A [**PBX**](#gt_10d7622e-7d99-49c9-89ba-0a732e4d2374) that supports Voice over IP (VoIP).

**Multipurpose Internet Mail Extensions (MIME)**: A set of extensions that redefines and expands support for various types of content in email messages, as described in [[RFC2045]](http://go.microsoft.com/fwlink/?LinkId=90307), [[RFC2046]](http://go.microsoft.com/fwlink/?LinkId=90308), and [[RFC2047]](http://go.microsoft.com/fwlink/?LinkId=90309).

**offer**: A message that is sent by an offerer.

**P-Asserted-Identity (PAI)**: A Session Initiation Protocol (SIP) header field, as described in [[RFC3325]](http://go.microsoft.com/fwlink/?LinkId=114232), that is used by trusted entities to carry the identity of the user who is sending an SIP message as it was verified by authentication (2).

**private branch exchange (PBX)**: A server-based telephony solution that services a specific organization or office.

**proxy**: A computer, or the software that runs on it, that acts as a barrier between a network and the Internet by presenting only a single network address to external sites. By acting as a go-between that represents all internal computers, the proxy helps protects network identities while also providing access to the Internet.

**public switched telephone network (PSTN)**: Public switched telephone network is the voice-oriented public switched telephone network. It is circuit-switched, as opposed to the packet-switched networks.

**SDP answer**: A [**Session Description Protocol (SDP)**](#gt_5ecff0fe-93f3-480a-aa69-57586d46967b) message that is sent by an answerer in response to an offer that is received from an offerer.

**SDP offer**: A [**Session Description Protocol (SDP)**](#gt_5ecff0fe-93f3-480a-aa69-57586d46967b) message that is sent by an offerer.

**Session Description Protocol (SDP)**: A protocol that is used for session announcement, session invitation, and other forms of multimedia session initiation. For more information see [[MS-SDP]](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SDP%5d.pdf#Section_697845ff53574eb78bcb162a0bc84deb) and [RFC3264].

**Session Initiation Protocol (SIP)**: An application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. [**SIP**](#gt_586971aa-3b65-4de3-be93-1a9756777d89) is defined in [[RFC3261]](http://go.microsoft.com/fwlink/?LinkId=90410).

**SIP message**: The data that is exchanged between [**Session Initiation Protocol (SIP)**](#gt_586971aa-3b65-4de3-be93-1a9756777d89) elements as part of the protocol. An SIP message is either a request or a response.

**SIP transaction**: A [**SIP transaction**](#gt_f45dabe6-9287-40c0-835b-43c83011b943) occurs between a [**UAC**](#gt_e5f72a3f-9df4-47e1-b4ee-eda52237bafb) and a [**UAS**](#gt_6f39aa0f-2438-4c06-8ccc-5d36b6e50a28). The [**SIP transaction**](#gt_f45dabe6-9287-40c0-835b-43c83011b943) comprises all messages from the first request sent from the [**UAC**](#gt_e5f72a3f-9df4-47e1-b4ee-eda52237bafb) to the [**UAS**](#gt_6f39aa0f-2438-4c06-8ccc-5d36b6e50a28) up to a final response (non-1xx) sent from the [**UAS**](#gt_6f39aa0f-2438-4c06-8ccc-5d36b6e50a28) to the [**UAC**](#gt_e5f72a3f-9df4-47e1-b4ee-eda52237bafb). If the request is [**INVITE**](#gt_d4b1b9b3-4b41-4686-aae0-afcd932693da), and the final response is a non-2xx, the [**SIP transaction**](#gt_f45dabe6-9287-40c0-835b-43c83011b943) also includes an ACK to the response. The ACK for a 2xx response to an [**INVITE**](#gt_d4b1b9b3-4b41-4686-aae0-afcd932693da) request is a separate [**SIP transaction**](#gt_f45dabe6-9287-40c0-835b-43c83011b943).

**Uniform Resource Identifier (URI)**: A string that identifies a resource. The URI is an addressing mechanism defined in Internet Engineering Task Force (IETF) Uniform Resource Identifier (URI): Generic Syntax [[RFC3986]](http://go.microsoft.com/fwlink/?LinkId=90453).

**user agent client (UAC)**: A logical entity that creates a new request, and then uses the client transaction state machinery to send it. The role of [**UAC**](#gt_e5f72a3f-9df4-47e1-b4ee-eda52237bafb) lasts only for the duration of that transaction. In other words, if a piece of software initiates a request, it acts as a [**UAC**](#gt_e5f72a3f-9df4-47e1-b4ee-eda52237bafb) for the duration of that transaction. If it receives a request later, it assumes the role of a [**user agent server (UAS)**](#gt_6f39aa0f-2438-4c06-8ccc-5d36b6e50a28) for the processing of that transaction.

**user agent server (UAS)**: A logical entity that generates a response to a [**Session Initiation Protocol (SIP)**](#gt_586971aa-3b65-4de3-be93-1a9756777d89) request. The response either accepts, rejects, or redirects the request. The role of the UAS lasts only for the duration of that transaction. If a process responds to a request, it acts as a UAS for that transaction. If it initiates a request later, it assumes the role of a [**user agent client (UAC)**](#gt_e5f72a3f-9df4-47e1-b4ee-eda52237bafb) for that transaction.

**MAY, SHOULD, MUST, SHOULD NOT, MUST NOT:** These terms (in all caps) are used as defined in [[RFC2119]](http://go.microsoft.com/fwlink/?LinkId=90317). All statements of optional behavior use either MAY, SHOULD, or SHOULD NOT.

## References

Links to a document in the Microsoft Open Specifications library point to the correct section in the most recently published version of the referenced document. However, because individual documents in the library are not updated at the same time, the section numbers in the documents may not match. You can confirm the correct section numbering by checking the [Errata](http://msdn.microsoft.com/en-us/library/dn781092.aspx).

### Normative References

We conduct frequent surveys of the normative references to assure their continued availability. If you have any issue with finding a normative reference, please contact [dochelp@microsoft.com](mailto:dochelp@microsoft.com). We will assist you in finding the relevant information.

[MS-SDPEXT] Microsoft Corporation, "[Session Description Protocol (SDP) Version 2.0 Extensions](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SDPEXT%5d.pdf#Section_cd17a549b94842a6aa6bfa707710faac)".

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[RFC2111] Levinson, E., "Content-ID and Message-ID Uniform Resource Locators", RFC 2111, March 1997, [http://www.rfc-editor.org/rfc/rfc2111.txt](http://go.microsoft.com/fwlink/?LinkId=185257)

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997, [http://www.rfc-editor.org/rfc/rfc2119.txt](http://go.microsoft.com/fwlink/?LinkId=90317)

[RFC3261] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and Schooler, E., "SIP: Session Initiation Protocol", RFC 3261, June 2002, [http://www.ietf.org/rfc/rfc3261.txt](http://go.microsoft.com/fwlink/?LinkId=90410)

[RFC3966] Schulzrinne, H., "The tel URI for Telephone Numbers", RFC 3966, December 2004, [http://www.rfc-editor.org/rfc/rfc3966.txt](http://go.microsoft.com/fwlink/?LinkId=114246)

### Informative References

[MS-SIPAE] Microsoft Corporation, "[Session Initiation Protocol (SIP) Authentication Extensions](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SIPAE%5d.pdf#Section_ba3e9821fa854e0fa80c5a4c720a00bd)".

[MS-SIPREGE] Microsoft Corporation, "[Session Initiation Protocol (SIP) Registration Extensions](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SIPREGE%5d.pdf#Section_21acf797984c48ce97e3c7df5c776b3d)".

[MS-SIPRE] Microsoft Corporation, "[Session Initiation Protocol (SIP) Routing Extensions](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SIPRE%5d.pdf#Section_ab4ab24937964ed18cecf496d81a1a83)".

[RFC3262] Rosenberg, J., and Schulzrinne, H., "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)", RFC 3262, June 2002, [http://www.ietf.org/rfc/rfc3262.txt](http://go.microsoft.com/fwlink/?LinkId=90411)

[RFC3263] Rosenberg, J., and Schulzrinne, H., "Session Initiation Protocol (SIP): Locating SIP Servers", RFC 3263, June 2002, [http://www.ietf.org/rfc/rfc3263.txt](http://go.microsoft.com/fwlink/?LinkId=90412)

[RFC3264] Rosenberg, J., and Schulzrinne, H., "An Offer/Answer Model with the Session Description Protocol (SDP)", RFC 3264, June 2002, [http://www.rfc-editor.org/rfc/rfc3264.txt](http://go.microsoft.com/fwlink/?LinkId=114231)

[RFC3325] Jennings, C., Peterson, J., and Watson, M., "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks", RFC 3325, November 2002, [http://www.rfc-editor.org/rfc/rfc3325.txt](http://go.microsoft.com/fwlink/?LinkId=114232)

[RFC3515] Sparks, R., "The Session Initiation Protocol (SIP) Refer Method", RFC 3515, April 2003, [http://www.ietf.org/rfc/rfc3515.txt](http://go.microsoft.com/fwlink/?LinkId=90428)

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## Overview

This protocol adds extensions to the Session Initiation Protocol (SIP), for interfacing a protocol client with other traditional telephony networks, such as a [**public switched telephone network (PSTN)**](#gt_475991aa-abb2-4521-a0a5-6993d27dba9f) and an enterprise [**private branch exchange (PBX)**](#gt_10d7622e-7d99-49c9-89ba-0a732e4d2374) or [**IP-PBX**](#gt_ec6bdfb2-dec6-4bae-b6af-14b75ac8bfd3).

The logical entities that are affected by these extensions are protocol client, server ([**proxy**](#gt_2b529701-3e64-4bf8-97ec-15afbba18b73)), and [**gateway**](#gt_6d5e3aba-23e2-4c54-a87c-6c8034872681) entities. The protocol client and the gateway can function as a [**user agent client (UAC)**](#gt_e5f72a3f-9df4-47e1-b4ee-eda52237bafb) or [**user agent server (UAS)**](#gt_6f39aa0f-2438-4c06-8ccc-5d36b6e50a28), depending on their role in the [**SIP transaction**](#gt_f45dabe6-9287-40c0-835b-43c83011b943), as illustrated in the following diagram.

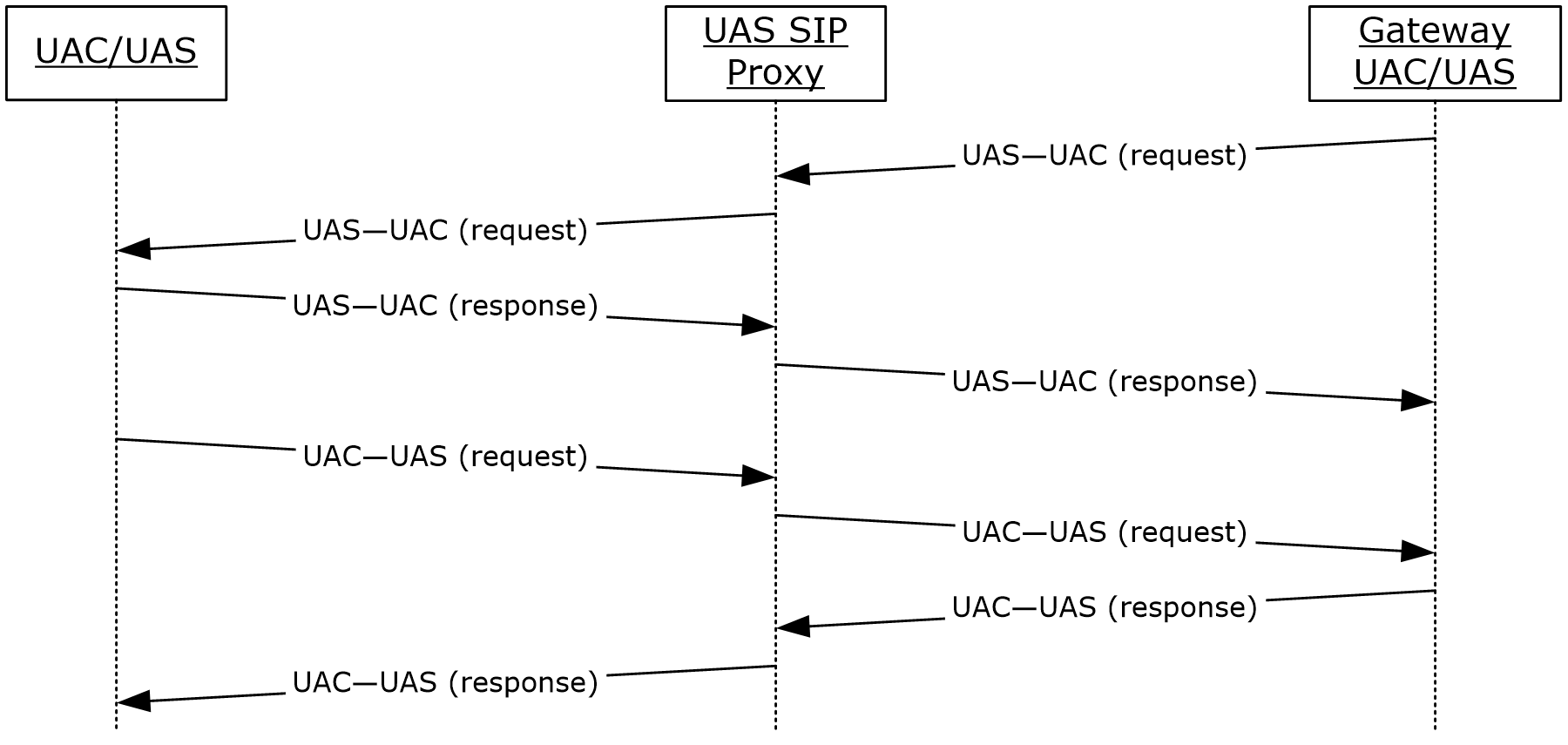


Figure 1: SIP transaction

The extensions do the following:

* Enable a SIP user agent (SIP UA) to be aware that a remote SIP UA in a SIP [**dialog**](#gt_71ad645f-db5b-4e9f-9b3d-887039ada331) is a gateway, as described in section [2.2.1](#Section_5d20970ca734403da106dba6e4852da1) and section [3.1](#Section_cb23f3721c144009ba48a6cb6a4e4e5c). This information can be rendered to the user interface (UI) to provide a better user experience (UX).
* Enable a SIP [**URI**](#gt_e18af8e8-01d7-4f91-8a1e-0fb21b191f95) to hold an address of a [**dial string**](#gt_c49bd875-352a-4c84-9b18-d8531e635421) that is given by a user, as described in section [2.2.2](#Section_8cf63c8841b4462aab7f7ff079064a9f) and section [3.2](#Section_353bd27a65a04376b5d97a255c69d19a).
* Enable a SIP UAS to detect a redundant [**call**](#gt_9c30971d-7054-4d6b-90a7-c7410283f71d) that is triggered as a result of a loop, as described in section [2.2.3](#Section_296a4f84357046008ddaadacd2e5ae98) and section [3.3](#Section_29248f89c3eb4995b88db6c853ade5f8). A loop occurs when a call is forked to a PBX that forks the call back, using a new SIP dialog.
* Enable a SIP UA to indicate that it is willing to receive an [**SDP answer**](#gt_3c11031c-654a-4df7-be46-0affb0ac03f5) through a non-reliable 183 provisional response to an [**INVITE**](#gt_d4b1b9b3-4b41-4686-aae0-afcd932693da) message, as described in section [2.2.4](#Section_f74c2d71f1c44495af47f7620b4c493f) and section [3.4](#Section_88069d5c3ab24db88aa616f24954e9d2). Note that the standard recommends sending an SDP answer for [**early media**](#gt_91185b14-74c7-4885-8da2-ce925e079fd8) only through a reliable provisional response, as described in [[RFC3262]](http://go.microsoft.com/fwlink/?LinkId=90411).
* Define an anonymous phone URI, as described in section [2.2.5](#Section_840750ba15ee46cea22b46087b365395) and section [3.5](#Section_8743bdaa822447f7a6a172d084e959b2), as an alternative to the standard anonymous SIP URI, as described in [[RFC3261]](http://go.microsoft.com/fwlink/?LinkId=90410). Note that the standard anonymous SIP URI is not supported.
* Enable a SIP UA in the protocol network to indicate that it supports media bypass functionality, as described in section [2.2.6](#Section_a473fa9ac8b941aca1c6193d0f4fb1b3) and section [3.6](#Section_7fcf5e4e1c264eb4b8b9cccc1f650971). Media bypass has the media from the protocol network entity involved in a PSTN call going directly to the gateway used to interface with the PSTN for that call, without traversing any intermediate element in the protocol network.
* Enable a SIP UA in the protocol network to reference the appropriate [**Session Description Protocol (SDP)**](#gt_5ecff0fe-93f3-480a-aa69-57586d46967b) that was selected from a received [**offer**](#gt_83dc4e88-7966-41f5-a35d-61bb5daee80b) when sending a [**SIP message**](#gt_2690e796-e281-48f3-ba0e-1f9acdb3ba8c) with an [**answer**](#gt_0435df97-8344-48f6-9ffe-4d8a57b84366) to the offer, as described in section [2.2.7](#Section_8af44c27fad24065a1d0cfafc686b22f) and section [3.7](#Section_79781c0675814622a6078eed53a7c686).
* Identify the specific gateway used to interface with the PSTN for a PSTN call, as described in section [2.2.8](#Section_311b0ac4f595464fae3f5d195c03bce5) and section [3.8](#Section_8f79c3449f9e4401990b0fcba764b8a1).

## Relationship to Other Protocols

This protocol uses the protocols as described in [[MS-SIPAE]](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SIPAE%5d.pdf#Section_ba3e9821fa854e0fa80c5a4c720a00bd), [[MS-SIPREGE]](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SIPREGE%5d.pdf#Section_21acf797984c48ce97e3c7df5c776b3d), [[MS-SIPRE]](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SIPRE%5d.pdf#Section_ab4ab24937964ed18cecf496d81a1a83), and [[MS-SDPEXT]](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SDPEXT%5d.pdf#Section_cd17a549b94842a6aa6bfa707710faac) as well as the following Internet Engineering Task Force (IETF) Session Initiation Protocol (SIP) specifications:

* SIP: Session Initiation Protocol, as described in [[RFC3261]](http://go.microsoft.com/fwlink/?LinkId=90410).
* Session Initiation Protocol (SIP): Locating SIP Servers, as described in [[RFC3263]](http://go.microsoft.com/fwlink/?LinkId=90412).
* the Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks, as described in [[RFC3325]](http://go.microsoft.com/fwlink/?LinkId=114232).
* the Session Initiation Protocol (SIP) Refer Method, as described in [[RFC3515]](http://go.microsoft.com/fwlink/?LinkId=90428).
* the Session Initiation Protocol (SIP) "Replaces" Header, as described in [[RFC3891]](http://go.microsoft.com/fwlink/?LinkId=114233).
* the Session Initiation Protocol (SIP) Referred-By Mechanism, as described in [[RFC3892]](http://go.microsoft.com/fwlink/?LinkId=114234).
* an Offer/Answer Model with the Session Description Protocol (SDP), as described in [[RFC3264]](http://go.microsoft.com/fwlink/?LinkId=114231).

## Prerequisites/Preconditions

None.

## Applicability Statement

This protocol is applicable for interfacing a protocol client with other traditional telephony networks, such as a PSTN and an enterprise PBX or IP-PBX.

## Versioning and Capability Negotiation

This protocol does not have protocol versioning. Instead, explicit capability negotiation is done by using the **Supported** header to indicate support of various features. The **Supported** header is the standard SIP mechanism for doing capability negotiation.

## Vendor-Extensible Fields

None.

## Standards Assignments

None.

# Messages

## Transport

This protocol relies on SIP transport.

## Message Syntax

This protocol uses the SIP messageformat, as specified in [[RFC3261]](http://go.microsoft.com/fwlink/?LinkId=90410) section 7, and extends definitions of URI parameters and headers by adding new values for parameter and header names as well as their corresponding values.

### isGateway

The **isGateway** parameter is defined by this protocol as a new **Contact** header field parameter. The original [**Augmented Backus-Naur Form (ABNF)**](#gt_24ddbbb4-b79e-4419-96ec-0fdd229c9ebf), as defined in [[RFC5234]](http://go.microsoft.com/fwlink/?LinkId=123096), for the **Contact** header field, as specified in [[RFC3261]](http://go.microsoft.com/fwlink/?LinkId=90410) section 25, is extended as follows. The SIP **Contact** header field extension is the second line.

contact-params = c-p-q / c-p-expires

/ c-p-gw

/ contact-extension

c-p-gw = "isGateway"

The syntax of the **Contact** header field with the SIP **Contact** header field extension is illustrated as follows. The extension is the final ;isGateway.

CONTACT: <sip:a@example.com;gruu;opaque=srvr:MediationServer:xxx;grid=yyy>;**isGateway**

### phone-context

This protocol extends the semantics of the *phone-context* parameter but does not change its syntax, as specified in [[RFC3966]](http://go.microsoft.com/fwlink/?LinkId=114246). The *phone-context* value for a dial string is the provisioned location profile name of the user.

The *phone-context* extension defines the following two *phone-context* names:

* "dialstring"
* "enterprise"

The first one is used if a user location profile name is not provisioned, and the second is used if a SIP URI holds a phone number in a non-[**E.164**](#gt_b7bdb287-a9d3-470f-8a94-f49b7afa8cc6) format that is a result of applying enterprise [**dial plan**](#gt_58830cca-0c22-42a8-8fa0-e5ba6ae0af3e) rules.

The syntax of a SIP URI with a *phone-context* parameter is illustrated as follows:

1. sip:12345;phone-context=lp1@example.com;user=phone

In the previous line, example.com is the host part of the SIP URI. It is not affected by the *phone-context* parameter.

### ms-call-source

The ABNF, as defined in [[RFC5234]](http://go.microsoft.com/fwlink/?LinkId=123096), for the **ms-call-source** SIP header is as follows:

Ms-Call-Source = "Ms-Call-Source" HCOLON ("ms-rtc" / "non-ms-rtc")

The supported tokens for the **ms-call-source** header are "ms-rtc" and "non-ms-rtc". The first token designates that the call originated from a protocol server network, and the latter means that the call originated from a non–protocol server network, such as a PSTN or IP-PBX.

The syntax of this header is illustrated as follows:

Ms-Call-Source: ms-rtc

### ms-early-media

The **ms-early-media** option tag is a proprietary option tag for the SIP **Supported** header, as specified in [[RFC3261]](http://go.microsoft.com/fwlink/?LinkId=90410) section 20.37.

The syntax of the **ms-early-media** tag in the **Supported** header is illustrated as follows:

supported: ms-early-media

### Anonymous Phone URI

The anonymous phone URI is an alternative to the standard anonymous SIP URI, as specified in [[RFC3261]](http://go.microsoft.com/fwlink/?LinkId=90410). The user part of the SIP URI is set with the value "anonymous" and the parameter *user=phone* is added to the URI.

The syntax of a SIP URI with these settings is illustrated as follows:

1. sip:anonymous@contoso.com;user=phone

### ms-bypass

The **ms-bypass** option tag is a proprietary option tag for the SIP **Supported** header, as specified in [[RFC3261]](http://go.microsoft.com/fwlink/?LinkId=90410) section 20.37.

### ms-accepted-content-id

The ABNF, as defined in [[RFC5234]](http://go.microsoft.com/fwlink/?LinkId=123096), for the **ms-accepted-content-id** SIP header is as follows:

1. ms-accepted-content-id = "ms-accepted-content-id" HCOLON content-id

The **content-id** element is specified in [[RFC2045]](http://go.microsoft.com/fwlink/?LinkId=90307) section 7 and [[RFC2111]](http://go.microsoft.com/fwlink/?LinkId=185257) section 2. Note that the **ms-accepted-content-id** header includes the value of the **Content-ID MIME** header associated with the selected [**SDP offer**](#gt_22225c03-4a17-4279-8385-8150ac9f8317).

The syntax of the **ms-accepted-content-id** header is illustrated as follows:

1. ms-accepted-content-id: <da6e05c91d6b4132afa14d8b528732e6>

### ms-trunking-peer

The ABNF, as defined in [[RFC5234]](http://go.microsoft.com/fwlink/?LinkId=123096), for the **ms-trunking-peer** SIP header is as follows:

1. ms-trunking-peer = "ms-trunking-peer" HCOLON host \*1(SEMI trunkname) \*1(SEMI User-Agent)
2. trunkname = "trunk" EQUAL hostname
3. User-Agent = "User-Agent" EQUAL quoted-string

The **host**, **hostname**, and **quoted-string** elements are specified in [[RFC3261]](http://go.microsoft.com/fwlink/?LinkId=90410) section 25.

The syntax of the **ms-trunking-peer** header is illustrated as follows:

ms-trunking-peer: gateway.company1.com;trunk=trunk1;User-Agent="Gateway 1.0"

### ms-mediation-generated

The ABNF, as defined in [[RFC5234]](http://go.microsoft.com/fwlink/?LinkId=123096), for the **ms-mediation-generated** SIP header[<1>](#Appendix_A_1" \o "Product behavior note 1) is as follows:

ms-mediation-generated = "ms-mediation-generated" HCOLON "yes"

The syntax of the **ms-mediation-generated** header is illustrated as follows:

ms-mediation-generated: yes

# Protocol Details

## isGateway Details

**UAC Behavior**

If a UAC has a gateway role, it MUST insert the **isGateway** parameter in the **Contact** header. Otherwise, it MUST NOT insert this parameter in the **Contact** header.

**UAS Behavior**

If a UAS receives a SIP message with a **Contact** header that includes an **isGateway** parameter, it SHOULD render it to the application. The application MAY display this information to the user.

**Proxy Behavior**

This parameter does not affect a SIP proxy.

### Abstract Data Model

None.

### Timers

None.

### Initialization

None.

### Higher-Layer Triggered Events

None.

### Message Processing Events and Sequencing Rules

None.

### Timer Events

None.

### Other Local Events

None.

## phone-context Details

**UAC Behavior**

A UAC that is about to send an INVITE message to an address of a dial string MUST use the following logic. If a UAC cannot resolve the dial string to a SIP URI with an E.164 number of a remote party, based on local provisioning, it MUST set the SIP URI of the **Request URI** header with a phone number and a *phone-context* of the phone number in the user part of the SIP URI, as specified in [[RFC3966]](http://go.microsoft.com/fwlink/?LinkId=114246) section 5.1.5. The value of the *phone-context* SIP URI parameter holds the location profile name of the user if the dial string could not be resolved or the resolution of the dial string is to a non-E.164 number. If a user profile name is not provisioned and the dial string is not an E.164 number, the user agent MUST set *phone-context* to the default predefined **dialstring** name.

Callback presents another case where an outgoing call can have a **Request URI** and **To URI** containing a *phone-context* parameter. An incoming call to a UA can have a [**P-Asserted-Identity (PAI)**](#gt_0fb2b437-8b88-4203-93eb-0a99568241c9) header containing a non-E.164 number with a *phone-context* of "dialstring". Alternatively an incoming call to a UA can have a **P-Asserted-Identity** header containing a non-E.164 number with a *phone-context* of "enterprise". If the callback feature is implemented in the UA, the value of the **P-Asserted-Identity** header SHOULD be used to populate the **Request URI** and **To URI** fields in the INVITE triggered by the callback.

A UAC SHOULD[<2>](#Appendix_A_2" \o "Product behavior note 2) set a valid SIP URI in the **From** header. If the SIP URI is in a phone number format, user=phone, the phone number SHOULD[<3>](#Appendix_A_3" \o "Product behavior note 3) be in either E.164 format or a private number with a *phone-context* that is set with the location profile of the UAC.

**UAS Behavior**

If a UAS has a gateway role, it SHOULD include a **P-Asserted-Identity** in the [**200 OK**](#gt_d9c398c0-9009-4dc6-9340-36423671182b) that it sends in response to an initial INVITE used to establish a dialog. This 200 OK is derived by using the **Request URI** from the initial INVITE used to establish the dialog. If this **Request URI** is used, and it contains an E.164 number in the user part, the UAS SHOULD copy it to the PAI. If this **Request URI** is used, and it contains a non-E.164 number in the user part, the UAS SHOULD copy it to the P-Asserted-Identity (PAI) and the *phone-context* SHOULD be set to the location profile of the UAS.

**Proxy Behavior**

A SIP proxy that receives a SIP URI with a *phone-context* in the **Request URI** MUST try to match the *phone-context* name with a list of provisioned dial plan names. If there is a match, it MUST convert the SIP URI based on the rules that are defined in the matched dial plan. Otherwise if there is no match, it SHOULD use other proxy logic to route this URI.

A **P-Preferred-Identity** header is a SIP header field, as described in section 9.2 of [[RFC3325]](http://go.microsoft.com/fwlink/?LinkId=114232), which carries the preferred identity for the user sending the SIP message. This header field is sent by a UA to a trusted proxy to be inserted as the value in the P-Asserted-Identity (PAI) header field.

A SIP proxy that receives an INVITE that contains a *phone-context* in the **From URI** SHOULD[<4>](#Appendix_A_4" \o "Product behavior note 4) try to match the *phone-context* name with a list of provisioned dial plan names. If there is a match, and the INVITE does not contain a P-Asserted-Identity (PAI) or **P-Preferred-Identity** header, the proxy SHOULD[<5>](#Appendix_A_5" \o "Product behavior note 5) apply the rules defined in the matched dial plan and add a P-Asserted-Identity (PAI) with the result of the translation. If the result of the translation is not an E.164 number, the proxy SHOULD[<6>](#Appendix_A_6" \o "Product behavior note 6) insert a *phone-context* with the value "enterprise".

### Abstract Data Model

None.

### Timers

None.

### Initialization

None.

### Higher-Layer Triggered Events

None.

### Message Processing Events and Sequencing Rules

None.

### Timer Events

None.

### Other Local Events

None.

## ms-call-source Details

**UAC Behavior**

If the UAC is a gateway, it MUST insert an **ms-call-source** header in the SIP INVITE message that is sent to a UAS with the value "ms-rtc".

If the UAC is a gateway, it MUST insert an **ms-call-source** header in the SIP INVITE message that is sent to the PSTN with the value "non-ms-rtc".

If the UAC is not a gateway, it MUST NOT insert an **ms-call-source** header in an INVITE message.

**UAS Behavior**

If a UAS receives an INVITE message with an **ms-call-source** header while in alerting state in another SIP dialog, it MUST reject the INVITE message with a 605 error code. Otherwise, it processes the INVITE as a regular INVITE message.

**Proxy Behavior**

If a SIP proxy receives an INVITE with an **ms-call-source** header and the call is forwarded to a different user based on the forwarding rules of the original user, it MUST strip the header.

If a SIP proxy receives an INVITE with an **ms-call-source** header and it forwards the INVITE to a user who is not provisioned to receive loop calls, as described in section [1.3](#Section_f9b74451b4234b2b9bda05bba6b66a8f), it MUST strip this header.

If a SIP proxy receives a 605 message to an INVITE message that includes the **ms-call-source header**, it MUST send back a 480 message to the previous hop.

### Abstract Data Model

None.

### Timers

None.

### Initialization

None.

### Higher-Layer Triggered Events

None.

### Message Processing Events and Sequencing Rules

None.

### Timer Events

None.

### Other Local Events

None.

## ms-early-media Details

**UAC Behavior**

If a UAC supports an SDP answer in a non-reliable 183 provisional response to an INVITE message, it MUST send a SIP **Supported** header with the **ms-early-media** option tag. The SDP content and the procedure for starting early media are specified in [[MS-SDPEXT]](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SDPEXT%5d.pdf#Section_cd17a549b94842a6aa6bfa707710faac) section 3.1.5.12.

**UAS Behavior**

A UAS with a gateway role that receives an INVITE with a **Supported** header that includes an **ms-early-media** option tag MUST send an unreliable 183 provisional response with an SDP answer. The SDP content and the procedure for starting early media are specified in [MS-SDPEXT] section 3.1.5.12.

A UAS with a non-gateway role that receives an INVITE with a **Supported** header that includes an **ms-early-media** option tag MAY send an unreliable 183 provisional response with an SDP answer. The SDP content and the procedure for starting early media are specified in [MS-SDPEXT] section 3.1.5.12.

**Proxy Behavior**

There is no special handling for this extension for a SIP proxy.

### Abstract Data Model

None.

### Timers

None.

### Initialization

None.

### Higher-Layer Triggered Events

None.

### Message Processing Events and Sequencing Rules

None.

### Timer Events

None.

### Other Local Events

None.

## Anonymous Phone URI Details

The special URI "anonymous@*host*;user=phone" in the **From** header field MUST be used to denote an anonymous user. [<7>](#Appendix_A_7" \o "Product behavior note 7) The *host* portion contains the IP address, [**fully qualified domain name (FQDN)**](#gt_1769aec9-237e-44ed-9014-1abb3ec6de6e), or [**domain**](#gt_b0276eb2-4e65-4cf1-a718-e0920a614aca) of the user. The encoding for an anonymous user that uses "anonymous.invalid" in the *host* portion MUST NOT be used.

### Abstract Data Model

None.

### Timers

None.

### Initialization

None.

### Higher-Layer Triggered Events

None.

### Message Processing Events and Sequencing Rules

None.

### Timer Events

None.

### Other Local Events

None.

## ms-bypass Details

A user agent (UA) supporting media bypass SHOULD[<8>](#Appendix_A_8" \o "Product behavior note 8) include a Session Initiation Protocol (SIP) **Supported** header with the **ms-bypass** option tag whenever it advertises the options it supports. Media bypass has the media from the protocol network entity involved in a public switched telephone network (PSTN) call going directly to the gateway used to interface with the PSTN for that call, without traversing any intermediate element in the protocol network.

### Abstract Data Model

None.

### Timers

None.

### Initialization

None.

### Higher-Layer Triggered Events

None.

### Message Processing Events and Sequencing Rules

None.

### Timer Events

None.

### Other Local Events

None.

## ms-accepted-content-id Details

This section describes the **ms-accepted-content-id** SIP header.[<9>](#Appendix_A_9" \o "Product behavior note 9)

**UAC Behavior**

A UAC MUST include a **Content-ID MIME** header with each [**Multipurpose Internet Mail Extensions (MIME)**](#gt_af6ba277-34c1-493d-8103-71d2af36ce30) type of "application/SDP" and "application/gw-sdp" that it sends in an offer. The SDP content is specified in [[MS-SDPEXT]](file:///E:\Target\Office\Published\Books\MS-OCPSTN\%5bMS-SDPEXT%5d.pdf#Section_cd17a549b94842a6aa6bfa707710faac) section 3.

**UAS Behavior**

A UAS MUST include an **ms-accepted-content-id** SIP header in a SIP message containing an SDP answer if the selected SDP from the offer with which the answer is associated contained a **Content-ID MIME** header. The value of the **Content-ID MIME** header from the selected SDP in the offer MUST be copied as the value for the **ms-accepted-content-id** header.

**Proxy Behavior**

This parameter does not affect a SIP proxy.

### Abstract Data Model

None.

### Timers

None.

### Initialization

None.

### Higher-Layer Triggered Events

None.

### Message Processing Events and Sequencing Rules

None.

### Timer Events

None.

### Other Local Events

None.

## ms-trunking-peer Details

The **ms-trunking-peer** Session Initiation Protocol (SIP) header is included by a SIP UA that has a gateway role. It is used to identify the specific gateway used to interface with the public switched telephone network (PSTN) for a PSTN call.[<10>](#Appendix_A_10" \o "Product behavior note 10)

### Abstract Data Model

None.

### Timers

None.

### Initialization

None.

### Higher-Layer Triggered Events

None.

### Message Processing Events and Sequencing Rules

None.

### Timer Events

None.

### Other Local Events

None.

## ms-mediation-generated Details

The **ms-mediation-generated** Session Initiation Protocol (SIP) header is included by a SIP UA that has a gateway role.[<11>](#Appendix_A_11" \o "Product behavior note 11) It is used in provisional responses to indicate that the response was auto-generated by the UA and is not forwarded from a gateway used to interface with the public switched telephone network (PSTN) for a PSTN call.

### Abstract Data Model

None.

### Timers

None.

### Initialization

None.

### Higher-Layer Triggered Events

None.

### Message Processing Events and Sequencing Rules

None.

### Timer Events

None.

### Other Local Events

None.

# Protocol Examples

## isGateway SIP Contact Header Parameter

### Inbound Call

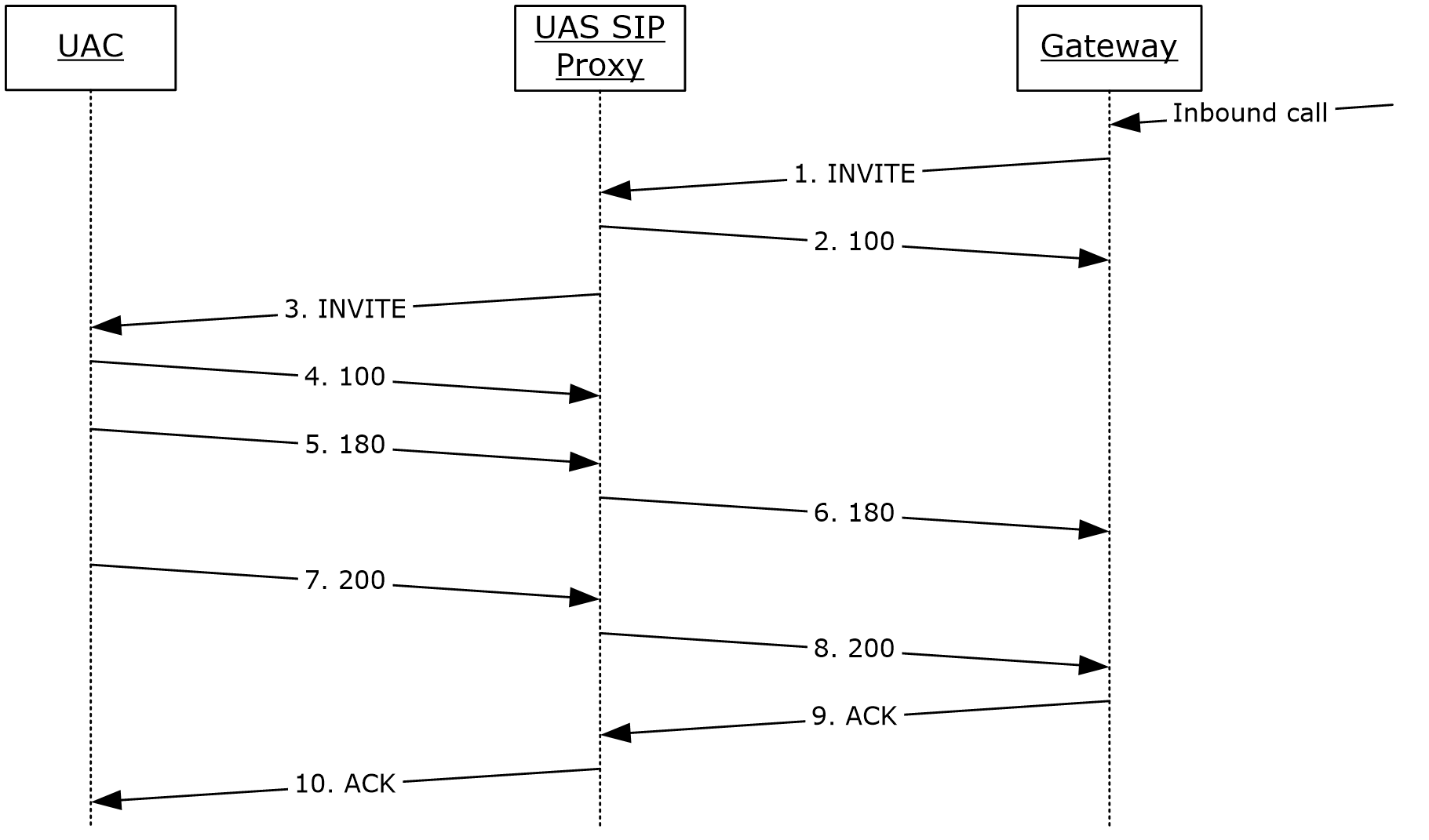


Figure 2: Inbound call

The preceding figure includes only one message in each direction because other messages repeat the same values.

The messages in the following subsections illustrate the use of the **Contact** header **isGateway** parameter in messages that are sent from and received by a protocol client.

#### Step 3: INVITE Message Is Received by the UAC

INVITE sip:10.56.66.167:1501;transport=tls;ms-opaque=56d3073f52;ms-received-cid=8000 SIP/2.0

Record-Route: <sip:server1.example.com:5061;transport=tls;ms-role-rs-from;lr;ms-identity=C8ybl0ausk5JrrJOeabpGevnl7YoohctFBsEB3Oy33pmWwR9xH\_oTAlgAA;ms-route-sig=ea0m1vIX8ijETotqsV9nVQESDR\_2qwR9xH\_oTAlgAA>;tag=D78DE2B2FF72EB24FDA98B88DCC879B2

Via: SIP/2.0/TLS 10.56.64.202:5061;branch=z9hG4bKD262F853.B047DC47;branched=TRUE;ms-internal-info="daqI8a1fcNQkUHDJyMoUxdQudrDTCwR9xH7\_OEdQAA"

Authentication-Info: Kerberos rspauth="602306092A864886F71201020201011100FFFFFFFF1125B31E1322F6E6A4E65212D8DEDCA4", srand="A8085D66", snum="58", opaque="C216B7E9", qop="auth", targetname="sip/media.example.com", realm="SIP Communications Service"

Max-Forwards: 69

Content-Length: 1606

Via: SIP/2.0/TLS 10.56.64.207:2861;branch=z9hG4bK27555a4e;ms-received-port=2861;ms-received-cid=8900

From: <sip:**anonymous**@server1.example.com;**user=phone**>;epid=571F84BB45;tag=ed77bad0f0

To: <sip:7275036;phone-context=normal-loc@server1.example.com;user=phone>;epid=782abb8f70

CSeq: 6 INVITE

Call-ID: 46bac89b-3f5f-4f1f-bb0b-e791706e2401

Contact: <sip:server1.example.com@server1.example.com;gruu;opaque=srvr:MediationServer:ANaNrdcy8EmB-dKmljqX-wAA;grid=9b192c6b829d4373adb88ea9ef4dff03>;**isGateway**

Supported: replaces

Supported: gruu-10

User-Agent: RTCC/3.0.0.0 MediationServer

Content-Type: application/sdp; charset=utf-8

#### Step 7: 200 Message Is Sent from the UAC

SIP/2.0 200 OK

Via: SIP/2.0/TLS 10.56.64.202:5061;branch=z9hG4bKD262F853.B047DC47;branched=TRUE;ms-internal-info="daqI8a1fcNQkUHDJyMoUxdQudrDTCwR9xH7\_OEdQAA"

Via: SIP/2.0/TLS 10.56.64.207:2861;branch=z9hG4bK27555a4e;ms-received-port=2861;ms-received-cid=8900

From: <sip:**anonymous**@server1.example.com;**user=phone**>;epid=571F84BB45;tag=ed77bad0f0

To: "" <sip:7275036;phone-context=normal-loc@server1.example.com;user=phone>;epid=782abb8f70;tag=8827660e0c

Call-ID: 46bac89b-3f5f-4f1f-bb0b-e791706e2401

CSeq: 6 INVITE

Record-Route: <sip:server1.example.com:5061;transport=tls;ms-role-rs-from;lr;ms-identity=C8ybl0ausk5JrrJOeabpGevnl7YoohctFBsEB3Oy33pmWwR9xH\_oTAlgAA;ms-route-sig=ea0m1vIX8ijETotqsV9nVQESDR\_2qwR9xH\_oTAlgAA>;tag=D78DE2B2FF72EB24FDA98B88DCC879B2

Contact: <sip:alice@server1.example.com;opaque=user:epid:reTyjuqAaVmcCIO4qlA4vwAA;gruu>

User-Agent: UCCP/2.0.6362.0 OC/2.0.6362.0 (Client)

P-Preferred-Identity: <sip:alice@server1.example.com>, <tel:+15555550103>

Proxy-Authorization: Kerberos qop="auth", realm="SIP Communications Service", opaque="C216B7E9", crand="dde2ad45", cnum="44", targetname="sip/server1.example.com", response="602306092a864886f71201020201011100ffffffff77de9d7a16f9693a9cc29ed8d6735499"

Content-Type: application/sdp

### Outbound Call

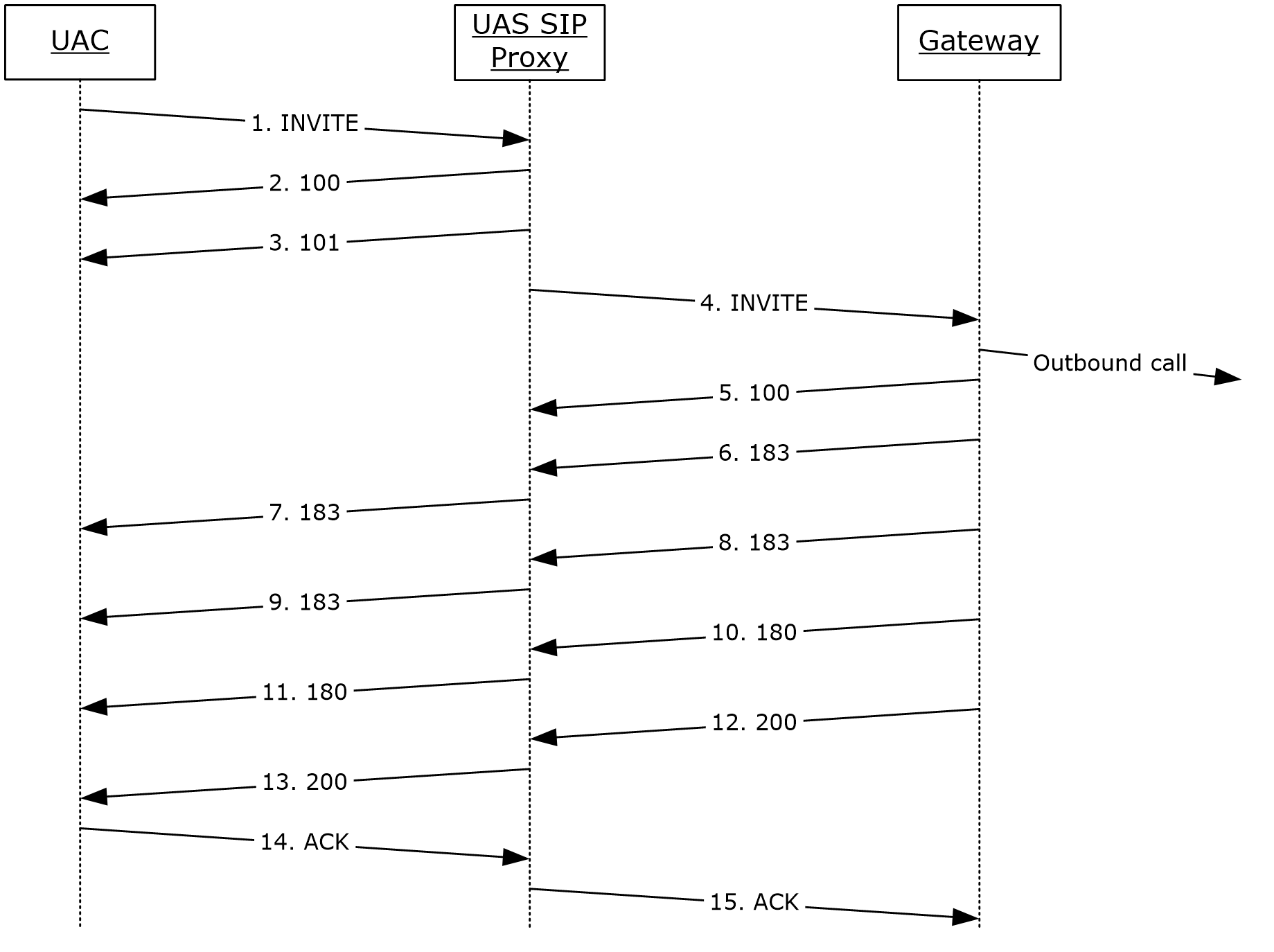


Figure 3: Outbound call

The preceding figure includes only one message in each direction because other messages repeat the same values.

The messages in the following subsections illustrate the use of the **Contact** header **isGateway** parameter in messages that are sent from and received by a UAC.

#### Step 1: INVITE Message Is Sent from the UAC

INVITE sip:+15555550103@server1.example.com;user=phone SIP/2.0

Via: SIP/2.0/TLS 10.56.66.167:1501

Max-Forwards: 70

From: <sip:alice@server1.example.com>;tag=85e83db3c6;epid=782abb8f70

To: <sip:+15555550103@server1.example.com;user=phone>

Call-ID: accd397afad9439d880f45cfce04bd66

CSeq: 1 INVITE

Contact: <sip:alice@server1.example.com;opaque=user:epid:reTyjuqAaVmcCIO4qlA4vwAA;gruu>

User-Agent: UCCP/2.0.6362.0 OC/2.0.6362.0 (Client)

Ms-Conversation-ID: AchdT5LJJvFktNrrSjejQUAy0wgfoA==

Supported: timer

Supported: ms-sender

Supported: ms-early-media

ms-keep-alive: UAC;hop-hop=yes

P-Preferred-Identity: <sip:alice@server1.example.com>, <tel:+15555550103>

Supported: ms-conf-invite

Proxy-Authorization: Kerberos qop="auth", realm="SIP Communications Service", opaque="C216B7E9", crand="aaee0f50", cnum="35", targetname="sip/server1.example.com", response="602306092a864886f71201020201011100ffffffffec51ac48141bf21d6a1487eaca68cca6"

Content-Type: application/sdp

Content-Length: 1076

#### Step 13: 200 Message Is Received by the UAC

SIP/2.0 200 OK

Authentication-Info: Kerberos rspauth="602306092A864886F71201020201011100FFFFFFFF0714800DE2F658803052D07C86052224", srand="B997073B", snum="50", opaque="C216B7E9", qop="auth", targetname="sip/server1.example.com", realm="SIP Communications Service"

Via: SIP/2.0/TLS 10.56.66.167:1501;ms-received-port=1501;ms-received-cid=8000

FROM: "alice"<sip:alice@server1.example.com>;tag=85e83db3c6;epid=782abb8f70

TO: <sip:+15555550103@server1.example.com;user=phone>;epid=571F84BB45;tag=a0f83282b

CSEQ: 1 INVITE

CALL-ID: accd397afad9439d880f45cfce04bd66

RECORD-ROUTE: <sip:server1.example.com:5061;transport=tls;ms-role-rs-from;lr;ms-route-sig=eab5PYPD\_tLMiadtWiQ5tem-72y4vocRve\_oTAlgAA>

CONTACT: <sip:server1.example.com@server1.example.com;gruu;opaque=srvr:MediationServer:ANaNrdcy8EmB-dKmljqX-wAA;grid=439be8c54ef04ce0baa8842286f86c53>;**isGateway**

CONTENT-LENGTH: 1412

SUPPORTED: gruu-10

SUPPORTED: replaces

CONTENT-TYPE: application/sdp; charset=utf-8

ALLOW: UPDATE

P-ASSERTED-IDENTITY: <sip:+15555550103@server1.example.com;user=phone>

SERVER: RTCC/3.0.0.0 MediationServer

ALLOW: Ack, Cancel, Bye,Invite,Refer

## phone-context SIP URI Parameter

### Inbound Call

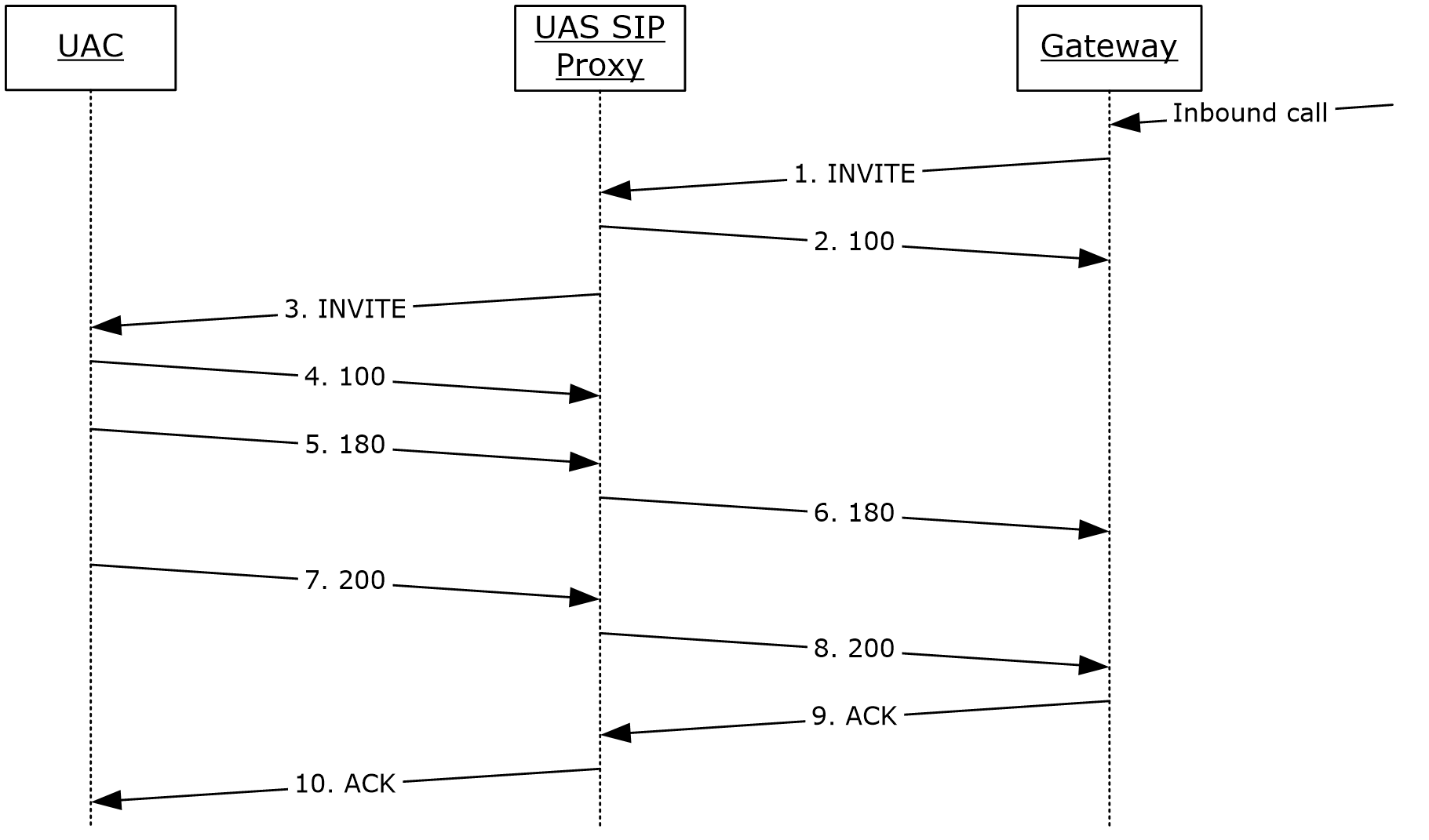


Figure 4: Inbound call

The preceding figure includes one message in each direction because other messages repeat the same values.

The messages in the following subsections illustrate the use of the *phone-context* parameter in messages that are sent from and received by the UAC.

#### Step 3: INVITE Message Is Received by the UAC

A UAS proxy replaces the **Request URI** header with the *phone-context* parameter that is received from the gateway. However, the **To** header is not replaced and holds the SIP URI with the *phone-context* that was inserted by the gateway.

INVITE sip:10.56.66.167:1501;transport=tls;ms-opaque=56d3073f52;ms-received-cid=8000 SIP/2.0

Record-Route: <sip:server1.example.com:5061;transport=tls;ms-role-rs-from;lr;ms-identity=C8ybl0ausk5JrrJOeabpGevnl7YoohctFBsEB3Oy33pmWwR9xH\_oTAlgAA;ms-route-sig=ea0m1vIX8ijETotqsV9nVQESDR\_2qwR9xH\_oTAlgAA>;tag=D78DE2B2FF72EB24FDA98B88DCC879B2

Via: SIP/2.0/TLS 10.56.64.202:5061;branch=z9hG4bKD262F853.B047DC47;branched=TRUE;ms-internal-info="daqI8a1fcNQkUHDJyMoUxdQudrDTCwR9xH7\_OEdQAA"

Authentication-Info: Kerberos rspauth="602306092A864886F71201020201011100FFFFFFFF1125B31E1322F6E6A4E65212D8DEDCA4", srand="A8085D66", snum="58", opaque="C216B7E9", qop="auth", targetname="sip/server1.example.com", realm="SIP Communications Service"

Max-Forwards: 69

Content-Length: 1606

Via: SIP/2.0/TLS 10.56.64.207:2861;branch=z9hG4bK27555a4e;ms-received-port=2861;ms-received-cid=8900

From: <sip:+15555550103@server1.example.com;user=phone>;epid=571F84BB45;tag=ed77bad0f0

To: <sip:7275036;**phone-context=normal-loc**@server1.example.com;user=phone>;epid=782abb8f70

CSeq: 6 INVITE

Call-ID: 46bac89b-3f5f-4f1f-bb0b-e791706e2401

Contact: <sipserver1.example.com@server1.example.com;gruu;opaque=srvr:MediationServer:ANaNrdcy8EmB-dKmljqX-wAA;grid=9b192c6b829d4373adb88ea9ef4dff03>;**isGateway**

Supported: replaces

Supported: gruu-10

User-Agent: RTCC/3.0.0.0 MediationServer

Content-Type: application/sdp; charset=utf-8

#### Step 7: 200 Message Is Sent from the UAC

SIP/2.0 200 OK

Via: SIP/2.0/TLS 10.56.64.202:5061;branch=z9hG4bKD262F853.B047DC47;branched=TRUE;ms-internal-info="daqI8a1fcNQkUHDJyMoUxdQudrDTCwR9xH7\_OEdQAA"

Via: SIP/2.0/TLS 10.56.64.207:2861;branch=z9hG4bK27555a4e;ms-received-port=2861;ms-received-cid=8900

From: <sip:+15555550103@server1.example.com;user=phone>;epid=571F84BB45;tag=ed77bad0f0

To: "" <sip:7275036;phone-context=normal-loc@server1.example.com;user=phone>;epid=782abb8f70;tag=8827660e0c

Call-ID: 46bac89b-3f5f-4f1f-bb0b-e791706e2401

CSeq: 6 INVITE

Record-Route: <sip:server1.example.com:5061;transport=tls;ms-role-rs-from;lr;ms-identity=C8ybl0ausk5JrrJOeabpGevnl7YoohctFBsEB3Oy33pmWwR9xH\_oTAlgAA;ms-route-sig=ea0m1vIX8ijETotqsV9nVQESDR\_2qwR9xH\_oTAlgAA>;tag=D78DE2B2FF72EB24FDA98B88DCC879B2

Contact: <sip:alice@server1.example.com;opaque=user:epid:reTyjuqAaVmcCIO4qlA4vwAA;gruu>

User-Agent: UCCP/2.0.6362.0 OC/2.0.6362.0 (Client)

P-Preferred-Identity: <sip:alice@server1.example.com>, <tel:+15555550106>

Proxy-Authorization: Kerberos qop="auth", realm="SIP Communications Service", opaque="C216B7E9", crand="dde2ad45", cnum="44", targetname="sip/server1.example.com", response="602306092a864886f71201020201011100ffffffff77de9d7a16f9693a9cc29ed8d6735499"

Content-Type: application/sdp

### Outbound Call

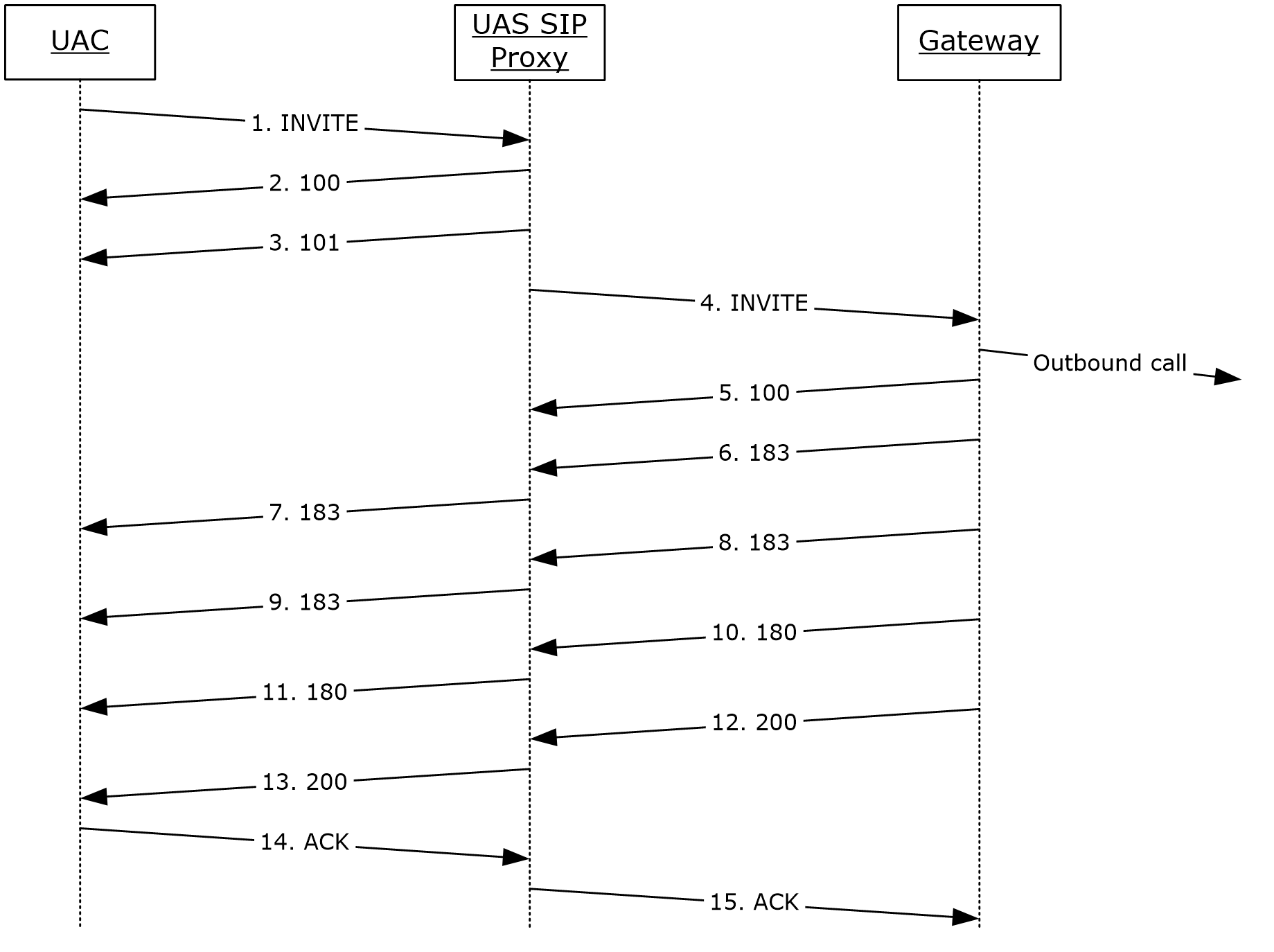


Figure 5: Outbound call

The preceding figure includes one message in each direction because other messages repeat the same values.

The messages in the following subsections illustrate the use of the *phone-context* parameter in messages that are sent from and received by a UAC.

#### Step 1: INVITE Message Is Sent from the UAC

INVITE sip:72181;**phone-context=dialstring**@example.com;user=phone SIP/2.0

Via: SIP/2.0/TLS 10.56.64.148:4031

Max-Forwards: 70

From: <sip:test2@example.com>;tag=cefe741803;epid=7d725e08a1

To: <sip:72181;**phone-context=dialstring**@example.com;user=phone>

Call-ID: a6a53b0e3b7d40a3b445dc4d9249b6fe

CSeq: 1 INVITE

Contact: <sip:test2@example.com;opaque=user:epid:0ONaA0AXIFCRDgr367kcHwAA;gruu>

User-Agent: UCCP/2.0.6362.36 OC/2.0.6362.36 (Client)

Ms-Conversation-ID: Achis3b6kqiLEhnZR/+DMH2N7CO9hg==

Supported: timer

Supported: ms-sender

Supported: ms-early-media

ms-keep-alive: UAC;hop-hop=yes

P-Preferred-Identity: <sip:test2@example.com>, <tel:+15555550100>

Supported: ms-conf-invite

Proxy-Authorization: NTLM qop="auth", realm="SIP Communications Service", opaque="9ACB05CE", crand="05c62674", cnum="10", targetname="server1.example.com", response="01000000b09f0702d4e4e934e25e6f9b"

Content-Type: application/sdp

#### Step 13: 200 Message Is Received by the UAC

SIP/2.0 200 OK

Authentication-Info: NTLM rspauth="01000000653865359728F3C8E25E6F9B", srand="8A3F6211", snum="19", opaque="9ACB05CE", qop="auth", targetname="server1.example.com", realm="SIP Communications Service"

Via: SIP/2.0/TLS 10.56.64.148:4031;ms-received-port=4031;ms-received-cid=500

FROM: "test2"<sip:test2@example.com>;tag=cefe741803;epid=7d725e08a1

TO: <sip:72181;phone-context=dialstring@example.com;user=phone>;epid=6477F45221;tag=a5c53ff9d6

CSEQ: 1 INVITE

CALL-ID: a6a53b0e3b7d40a3b445dc4d9249b6fe

RECORD-ROUTE: <sip:server1.example.com:5061;transport=tls;ms-role-rs-from;lr;ms-route-sig=aaabLHUMznhhFXTzutN9dwpQ-RmwQYZA\_UIeytlQAA>

CONTACT: <sip:SH13-LCT.example.com@example.com;gruu;opaque=srvr:MediationServer:TIRig7bu5kGXhNJb1ZwQfgAA;grid=f1f9379bd9334f65aa1dfb77bed58905>;isGateway

CONTENT-LENGTH: 740

SUPPORTED: gruu-10

SUPPORTED: replaces

CONTENT-TYPE: application/sdp; charset=utf-8

## ms-call-source SIP Header

### Inbound Call

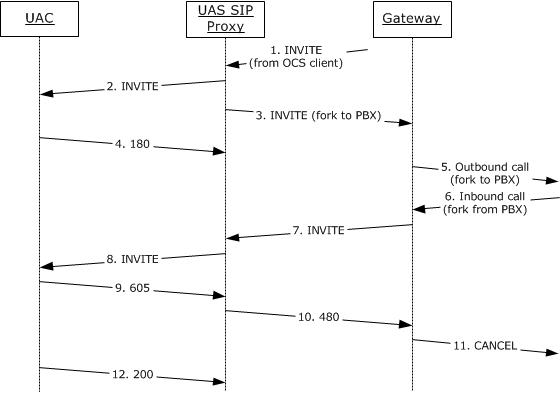


Figure 6: Inbound call

The preceding figure includes only key messages that are described in this section.

The messages in the following subsections illustrate the use of the **ms-call-source** header in messages that are sent from and received by a UAC.

#### Step 2: INVITE Message Is Received by the UAC

This call originates from a protocol client; therefore, it does not include an **ms-call-source** header.

INVITE sip:10.56.66.167:3080;transport=tls;ms-opaque=0e2b3bcc10;ms-received-cid=300 SIP/2.0

Record-Route: <sip:server1.example.com:5061;transport=tls;ms-role-rs-to;ms-role-rs-from;lr;ms-route-sig=aa1PYPSx-ubLAoi5ZiOqMHlGcU3l0xkBBiLt2WZQAA>;tag=8951C70C798E10EA48EB96EAA4B379BC

Via: SIP/2.0/TLS 172.29.106.3:5061;branch=z9hG4bK76F2CFD5.31901062;branched=TRUE;ms-internal-info="aaC8UGYE\_vlAjm36glJ1-v1NQi15UxkBBihbkIPAAA"

Authentication-Info: NTLM rspauth="0100000044415441C22E2F66F9C08F09", srand="4CB6D6F5", snum="31", opaque="FB347BC6", qop="auth", targetname="server1.example.com", realm="SIP Communications Service"

Max-Forwards: 69

Content-Length: 1074

Via: SIP/2.0/TLS 10.56.64.148:3981;ms-received-port=3981;ms-received-cid=200

P-Asserted-Identity: "test2"<sip:test2@example.com>,<tel:+15555550100>

From: "test2"<sip:test2@example.com>;tag=08399379aa;epid=7d725e08a1

To: <sip:test1@example.com>;epid=782abb8f70

Call-ID: ee22d219e9f44441bbac7b304ddc1096

CSeq: 1 INVITE

Contact: <sip:test2@example.com;opaque=user:epid:0ONaA0AXIFCRDgr367kcHwAA;gruu>

User-Agent: UCCP/2.0.6362.36 OC/2.0.6362.36 (Client)

Ms-Conversation-ID: AchisczVZkuxUO7mTZieBaNoXIHJ8g==

Supported: timer

Supported: ms-sender

Supported: ms-early-media

ms-keep-alive: UAC;hop-hop=yes

Supported: ms-conf-invite

Content-Type: application/sdp

#### Step 8: INVITE Message Is Received by the UAC

This call originates from the private branch exchange (PBX); therefore, it includes an **ms-call-source** header.

INVITE sip:10.56.66.167:3080;transport=tls;ms-opaque=0e2b3bcc10;ms-received-cid=300 SIP/2.0

Record-Route: <sip:server1.example.com:5061;transport=tls;ms-role-rs-from;lr;ms-identity=B5buGzyhwo49ocKOaabbgxdAqaQRu\_k9cZxy4WI-659Sq6MHw6Lt2WZQAA;ms-route-sig=aa3Oeh935IYJmaV5AyTn4katb\_Zd-6MHw6Lt2WZQAA>;tag=8951C70C798E10EA48EB96EAA4B379BC

Via: SIP/2.0/TLS 172.29.106.3:5061;branch=z9hG4bKD300E89E.BA307C3A;branched=TRUE;ms-internal-info="aaKlz4lwQeqhL-R5X7wnN8hEhuJwK6MHw6hbkIPAAA"

Authentication-Info: NTLM rspauth="0100000000000000A761D372F9C08F09", srand="51ED7291", snum="32", opaque="FB347BC6", qop="auth", targetname="server1.example.com", realm="SIP Communications Service"

Max-Forwards: 69

Content-Length: 934

Via: SIP/2.0/TLS 10.198.92.126:4757;branch=z9hG4bK557e25e3;ms-received-port=4757;ms-received-cid=700

From: <sip:2160;phone-context=dialstring@example.com;user=phone>;epid=6477F45221;tag=a3a3579bb

To: <sip:+15555550108@example.com;user=phone>;epid=782abb8f70

CSeq: 179 INVITE

Call-ID: 729ab37d-c0f5-4ad7-b7e0-8f3dadb99065

Contact: <sip:SH13-LCT.example.com@example.com;gruu;opaque=srvr:MediationServer:TIRig7bu5kGXhNJb1ZwQfgAA;grid=2cfb52f9fd4b4930a6f0a82dbfcd39e4>;isGateway

Supported: replaces

Supported: gruu-10

User-Agent: RTCC/3.0.0.0 MediationServer

Content-Type: application/sdp; charset=utf-8

Allow: UPDATE

Allow: Ack, Cancel, Bye,Invite,Refer

Ms-Call-Source: non-ms-rtc

#### Step 9: 605 Message Is Sent from the UAC

This INVITE is rejected with a 605 Decline because it originated as a result of a loop in the private branch exchange (PBX).

SIP/2.0 605 Decline Everywhere

Via: SIP/2.0/TLS 172.29.106.3:5061;branch=z9hG4bKD300E89E.BA307C3A;branched=TRUE;ms-internal-info="aaKlz4lwQeqhL-R5X7wnN8hEhuJwK6MHw6hbkIPAAA"

Via: SIP/2.0/TLS 10.198.92.126:4757;branch=z9hG4bK557e25e3;ms-received-port=4757;ms-received-cid=700

From: <sip:2160;phone-context=dialstring@example.com;user=phone>;epid=6477F45221;tag=a3a3579bb

To: "" <sip:+15555550108@example.com;user=phone>;epid=782abb8f70;tag=b9bc5b444c

Call-ID: 729ab37d-c0f5-4ad7-b7e0-8f3dadb99065

CSeq: 179 INVITE

Record-Route: <sip:server1.example.com:5061;transport=tls;ms-role-rs-from;lr;ms-identity=B5buGzyhwo49ocKOaabbgxdAqaQRu\_k9cZxy4WI-659Sq6MHw6Lt2WZQAA;ms-route-sig=aa3Oeh935IYJmaV5AyTn4katb\_Zd-6MHw6Lt2WZQAA>;tag=8951C70C798E10EA48EB96EAA4B379BC

User-Agent: UCCP/2.0.6362.0 OC/2.0.6362.0 (Client)

Ms-client-diagnostics: 52034; reason="Rejected Duplicate call from PBX"

Proxy-Authorization: NTLM qop="auth", realm="SIP Communications Service", opaque="FB347BC6", crand="b062a12e", cnum="29", targetname="server1.example.com", response="01000000730069006642f8a9f9c08f09"

Content-Length: 0

#### Step 12: 200 Message Is Sent from the UAC

The first INVITE from a protocol client is accepted and the following 200 OK is sent.

SIP/2.0 200 OK

Via: SIP/2.0/TLS 172.29.106.3:5061;branch=z9hG4bK76F2CFD5.31901062;branched=TRUE;ms-internal-info="aaC8UGYE\_vlAjm36glJ1-v1NQi15UxkBBihbkIPAAA"

Via: SIP/2.0/TLS 10.56.64.148:3981;ms-received-port=3981;ms-received-cid=200

From: "test2"<sip:test2@example.com>;tag=08399379aa;epid=7d725e08a1

To: "" <sip:test1@example.com>;epid=782abb8f70;tag=281b612cd0

Call-ID: ee22d219e9f44441bbac7b304ddc1096

CSeq: 1 INVITE

Record-Route: <sip:server1.example.com:5061;transport=tls;ms-role-rs-to;ms-role-rs-from;lr;ms-route-sig=aa1PYPSx-ubLAoi5ZiOqMHlGcU3l0xkBBiLt2WZQAA>;tag=8951C70C798E10EA48EB96EAA4B379BC

Contact: <sip:test1@example.com;opaque=user:epid:reTyjuqAaVmcCIO4qlA4vwAA;gruu>

User-Agent: UCCP/2.0.6362.0 OC/2.0.6362.0 (Client)

P-Preferred-Identity: <sip:test1@example.com>, <tel:+15555550100>

Proxy-Authorization: NTLM qop="auth", realm="SIP Communications Service", opaque="FB347BC6", crand="1b602324", cnum="30", targetname="server1.example.com", response="010000006895aa03478d7d34f9c08f09"

Content-Type: application/sdp

### Outbound Call

The **ms-call-source** header is not sent or received by a UAC in this scenario.

## ms-early-media SIP Supported Header Option Tag

### Inbound Call

The **ms-early-media** tag is not sent or received by a UAC in this scenario.

### Outbound Call

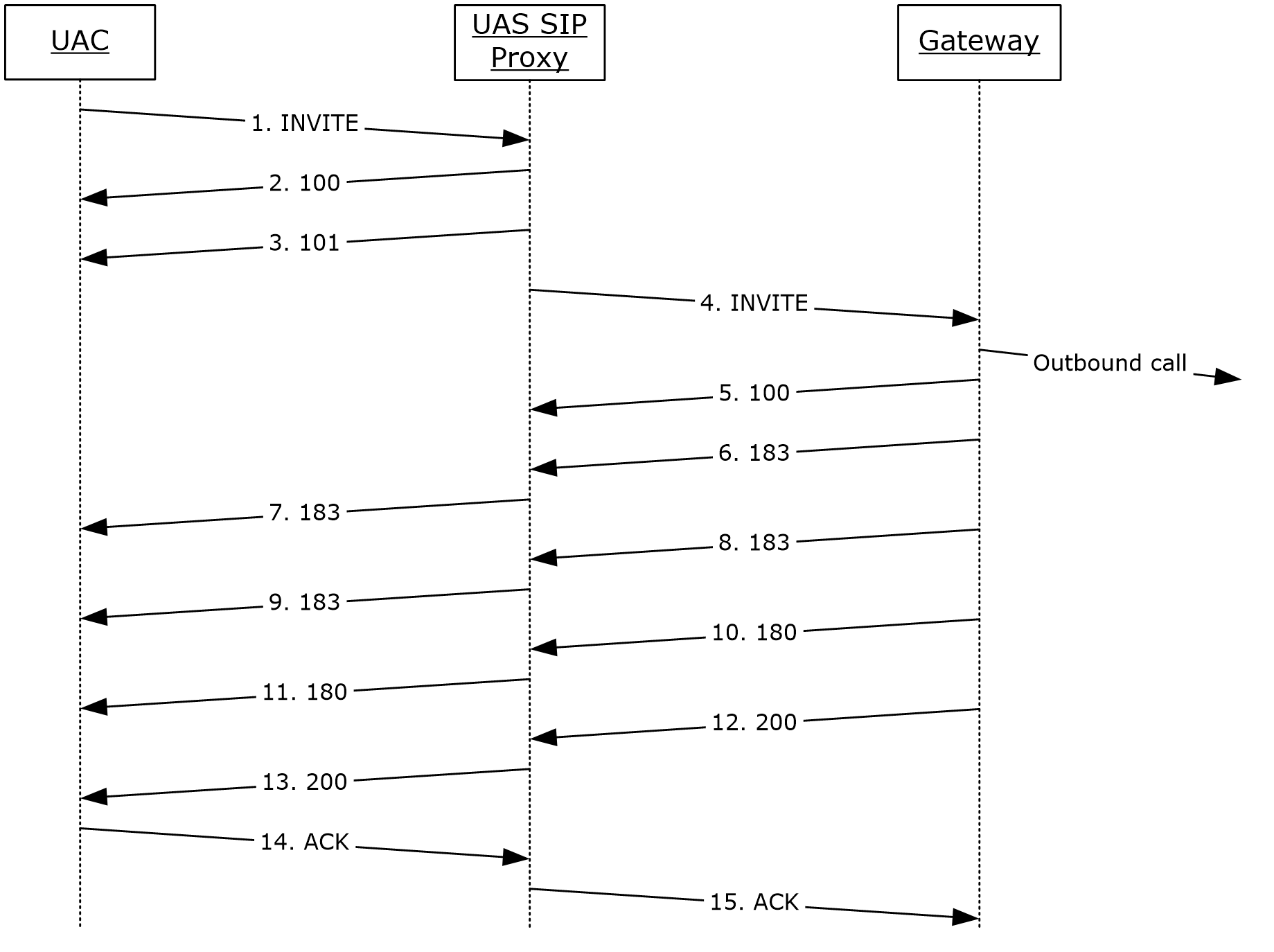


Figure 7: Outbound call

The preceding figure includes only key messages that are described in this section.

The following messages illustrate the use of the **ms-early-media** option tag in messages that are sent from and received by a UAC.

#### Step 1: INVITE Is Sent from the UAC

The following INVITE includes an **ms-early-media** option tag in a **Supported** header and an SDP offer.

INVITE sip:+15555550100@example.com;user=phone SIP/2.0

Via: SIP/2.0/TLS 10.56.66.167:3137

Max-Forwards: 70

From: <sip:test1@example.com>;tag=2b95504d65;epid=782abb8f70

To: <sip:+15555550100@example.com;user=phone>

Call-ID: ca22890914c34bf8a7439dfe1e834420

CSeq: 1 INVITE

Contact: <sip:test1@example.com;opaque=user:epid:reTyjuqAaVmcCIO4qlA4vwAA;gruu>

User-Agent: UCCP/2.0.6362.0 OC/2.0.6362.0 (Client)

Ms-Conversation-ID: Achit1o1q5CCFcXhRKeZABfaZzvWNw==

Supported: timer

Supported: ms-sender

Supported: ms-early-media

ms-keep-alive: UAC;hop-hop=yes

P-Preferred-Identity: <sip:test1@example.com>, <tel:+15555550101>

Supported: ms-conf-invite

Proxy-Authorization: NTLM qop="auth", realm="SIP Communications Service", opaque="B25450B8", crand="620d1d6e", cnum="79", targetname="server1.example.com", response="0100000008aab30387f6e10ef27db686"

Content-Type: application/sdp

Content-Length: 1076

v=0

o=- 0 0 IN IP4 10.56.66.167

s=session

c=IN IP4 10.56.66.167

b=CT:99980

t=0 0

m=audio 50016 RTP/AVP 114 111 112 115 116 4 8 0 97 101

k=base64:Bcw/3c0RQ/ndiix3QiLgO9s3z1ZhEcLU3ZC85C74zuNSmyIrx11eIA4kErwh

a=candidate:Hfb3G/XvuV5G7gXYnDfWjyyZ8aIUbsPUhQRkndBg3vU 1 x4Ykst3u0Cf7mAW0moKnMQ UDP 0.900 10.56.66.167 50016

a=candidate:Hfb3G/XvuV5G7gXYnDfWjyyZ8aIUbsPUhQRkndBg3vU 2 x4Ykst3u0Cf7mAW0moKnMQ UDP 0.900 10.56.66.167 50008

a=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:aAzhJhKx1bOgjuVWNfI8C4f1K9lE5SJIb6vFIAWP|2^31|1:1

a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:VqAkQvuZOMKH1uaXvi+8kjiJlRsiyngtcuh2AA5k|2^31|1:1

a=maxptime:200

a=rtcp:50008

a=rtpmap:114 x-msrta/16000

a=fmtp:114 bitrate=29000

a=rtpmap:111 SIREN/16000

a=fmtp:111 bitrate=16000

a=rtpmap:112 G7221/16000

a=fmtp:112 bitrate=24000

a=rtpmap:115 x-msrta/8000

a=fmtp:115 bitrate=11800

a=rtpmap:116 AAL2-G726-32/8000

a=rtpmap:4 G723/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:0 PCMU/8000

a=rtpmap:97 RED/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=encryption:optional

#### Step 7: 183 Message Is Received by the UAC

The following 183 Session Progress message includes an SDP answer.

SIP/2.0 183 Session Progress

Authentication-Info: NTLM rspauth="010000000000000008AC67ADF27DB686", srand="DF9D53C4", snum="103", opaque="B25450B8", qop="auth", targetname="server1.example.com", realm="SIP Communications Service"

Via: SIP/2.0/TLS 10.56.66.167:3137;ms-received-port=3137;ms-received-cid=100

FROM: "test1"<sip:test1@example.com>;tag=2b95504d65;epid=782abb8f70

TO: <sip:+15555550100@example.com;user=phone>;epid=6477F45221;tag=b5bb1243e3

CSEQ: 1 INVITE

CALL-ID: ca22890914c34bf8a7439dfe1e834420

CONTENT-LENGTH: 740

CONTENT-TYPE: application/sdp; charset=utf-8

SERVER: RTCC/3.0.0.0 MediationServer

v=0

o=- 0 0 IN IP4 10.198.92.126

s=session

c=IN IP4 10.198.92.126

b=CT:1000

t=0 0

m=audio 60625 RTP/SAVP 111 115 8 97 101

c=IN IP4 10.198.92.126

a=rtcp:60532

a=candidate:ZHqwSbPvIZyDX24RjvIW4lryUx/QbdAiP7FyQ0yvTGo 1 Bx2Is+Wi/HJbdQKM3FIBKg UDP 0.900 10.198.92.126 60625

a=candidate:ZHqwSbPvIZyDX24RjvIW4lryUx/QbdAiP7FyQ0yvTGo 2 Bx2Is+Wi/HJbdQKM3FIBKg UDP 0.900 10.198.92.126 60532

a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:Pb+rI3y4U1xd47P8USsgDc/znOiBIv5s0Ev2abRT|2^31|1:1

a=label:main-audio

a=encryption:rejected

a=rtpmap:111 SIREN/16000

a=fmtp:111 bitrate=16000

a=rtpmap:115 x-msrta/8000

a=fmtp:115 bitrate=11800

a=rtpmap:8 PCMA/8000

a=rtpmap:97 RED/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=ptime:20

#### Step 13: 200 Message Is Received by the UAC

The following 200 OK message repeats the SDP answer that was sent in the preceding 183 Session Progress message.

SIP/2.0 200 OK

Authentication-Info: NTLM rspauth="010000003240756E24DFD336F27DB686", srand="C1DF9895", snum="105", opaque="B25450B8", qop="auth", targetname="server1.example.com", realm="SIP Communications Service"

Via: SIP/2.0/TLS 10.56.66.167:3137;ms-received-port=3137;ms-received-cid=100

FROM: "test1"<sip:test1@example.com>;tag=2b95504d65;epid=782abb8f70

TO: <sip:+15555550100@example.com;user=phone>;epid=6477F45221;tag=b5bb1243e3

CSEQ: 1 INVITE

CALL-ID: ca22890914c34bf8a7439dfe1e834420

RECORD-ROUTE: <sip:server1.example.com:5061;transport=tls;ms-role-rs-from;lr;ms-route-sig=aaelhWZJsyQvUcPVgXY5rCBgN5MnVHAhdiIeytlQAA>

CONTACT: <sip:SH13-LCT.example.com@example.com;gruu;opaque=srvr:MediationServer:TIRig7bu5kGXhNJb1ZwQfgAA;grid=b6796217d6ea465cbe261a778c10d5c0>;isGateway

CONTENT-LENGTH: 740

SUPPORTED: gruu-10

SUPPORTED: replaces

CONTENT-TYPE: application/sdp; charset=utf-8

ALLOW: UPDATE

P-ASSERTED-IDENTITY: <sip:+17036508897@example.com;user=phone>

SERVER: RTCC/3.0.0.0 MediationServer

ALLOW: Ack, Cancel, Bye,Invite,Refer

v=0

o=- 0 0 IN IP4 10.198.92.126

s=session

c=IN IP4 10.198.92.126

b=CT:1000

t=0 0

m=audio 60625 RTP/SAVP 111 115 8 97 101

c=IN IP4 10.198.92.126

a=rtcp:60532

a=candidate:ZHqwSbPvIZyDX24RjvIW4lryUx/QbdAiP7FyQ0yvTGo 1 Bx2Is+Wi/HJbdQKM3FIBKg UDP 0.900 10.198.92.126 60625

a=candidate:ZHqwSbPvIZyDX24RjvIW4lryUx/QbdAiP7FyQ0yvTGo 2 Bx2Is+Wi/HJbdQKM3FIBKg UDP 0.900 10.198.92.126 60532

a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:Pb+rI3y4U1xd47P8USsgDc/znOiBIv5s0Ev2abRT|2^31|1:1

a=label:main-audio

a=encryption:rejected

a=rtpmap:111 SIREN/16000

a=fmtp:111 bitrate=16000

a=rtpmap:115 x-msrta/8000

a=fmtp:115 bitrate=11800

a=rtpmap:8 PCMA/8000

a=rtpmap:97 RED/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=ptime:20

## ms-bypass SIP Supported Header Option Tag

### Inbound Call

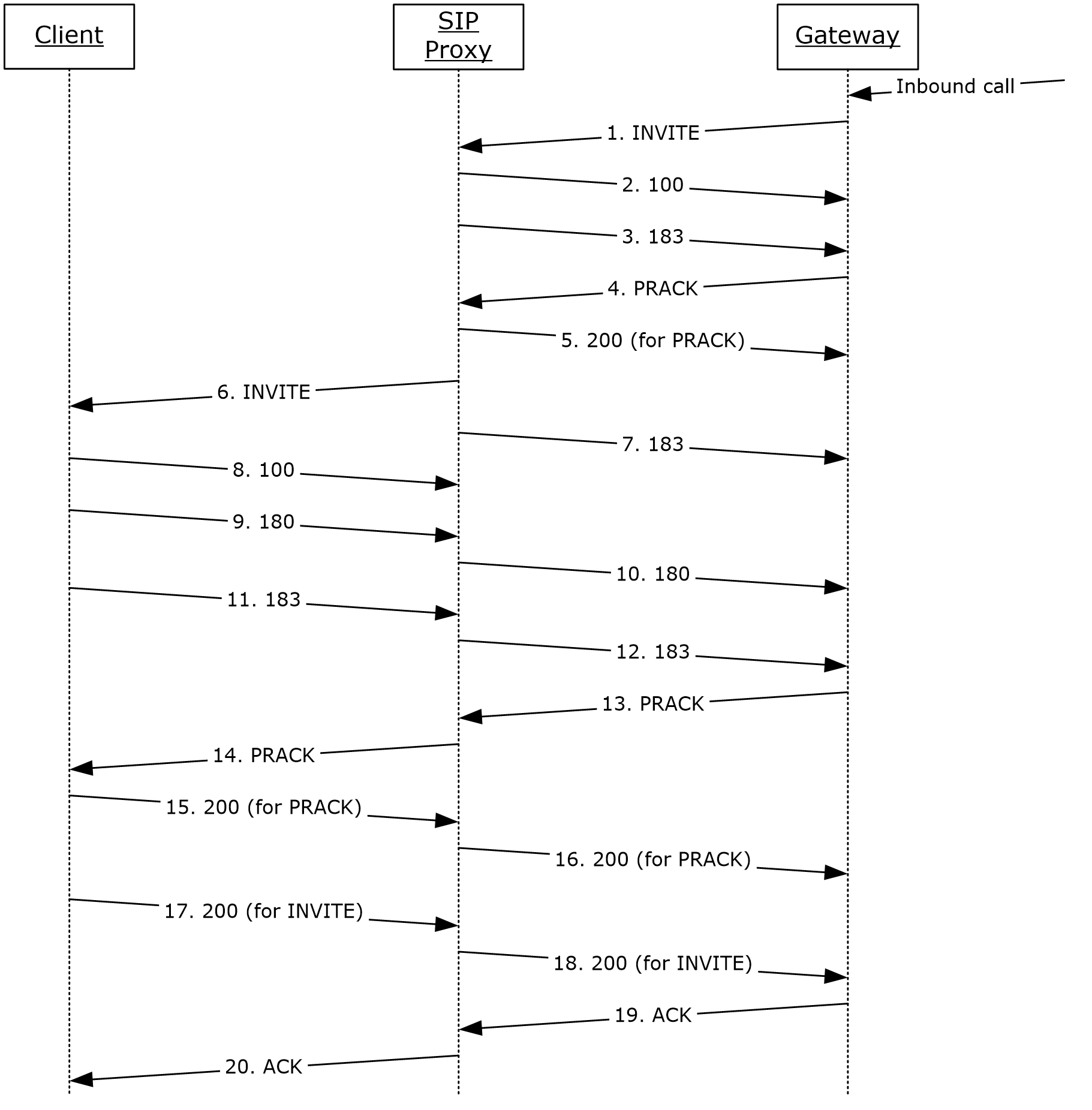


Figure 8: Inbound call

The messages in the following subsections illustrate the use of the **ms-bypass** option tag in a **Supported** header in messages that are sent from and received by a protocol client.

#### Step 6: INVITE Message Is Received by the Protocol Client

1. INVITE sip:192.168.1.114:4535;transport=tls;ms-opaque=acee5f6d3a;ms-received-cid=475300 SIP/2.0
2. Record-Route: <sip:PROXY.company1:5061;transport=tls;opaque=state:F:Ci.R475300:Ieh.gU65xODvwq\_j78KvdcC-dRxH7lEBsv8oECfECdswTe7QW4niMEtoTOi\_iwBgjHnKsZgY3jngAA;lr;ms-route-sig=dcrEzxvkq3iKgc2ApWyiXbCYC7NNwE-pYCMYxgjFJ3kxfHnKsZgY3jngAA>;tag=45F7A969AE33112CB9877940D7F56D40
3. Via: SIP/2.0/TLS 10.1.1.54:5061;branch=z9hG4bK1C7C8A0E.19AB9CC7A4B7C3D3;branched=TRUE;ms-internal-info="cehce-xXzqcRs3A\_ZSAwy8D4JLgyqxDKREgfIVFt6noRjHnKsZUY47CgAA"
4. Authentication-Info: TLS-DSK qop="auth", opaque="F755045D", srand="CC46B5FD", snum="26", rspauth="d6179291f72761e057a67adb7288fd256c2b1e4d", targetname="PROXY.company1", realm="SIP Communications Service", version=4
5. Max-Forwards: 69
6. Content-Length: 3161
7. Via: SIP/2.0/TLS 10.1.1.102:57350;branch=z9hG4bKe82f3c;ms-received-port=57350;ms-received-cid=475900
8. From: <sip:4259876543;phone-context=Location1@company1;user=phone>;epid=CDCFEF8F18;tag=3d965223ea
9. To: <sip:+14251234567@company1;user=phone>;epid=54dd5867e8
10. CSeq: 35 INVITE
11. Call-ID: df601b2d-e42e-4677-b921-c9dbf4e25940
12. Contact: <sip:ms5.company1@company1;gruu;opaque=srvr:MediationServer:XzRY6u68aVipMcYVfYs0hQAA;grid=bd9c42fc618147d0af4d8f84f718910b>;isGateway
13. Supported: replaces
14. Supported: ms-safe-transfer
15. Supported: ms-bypass
16. Supported: ms-dialog-route-set-update
17. Supported: timer
18. Supported: 100rel
19. Supported: gruu-10
20. User-Agent: Mediation Server
21. Content-Type: multipart/alternative; boundary=9dvaKhfhPJxCOyObvB70o0f2xfgiXN3J
22. Allow: ACK
23. ms-trunking-peer: gateway.company1.com;trunk=trunk1;User-Agent="Gateway 1.0"
24. Session-Expires: 1800
25. Min-SE: 90
26. Allow: CANCEL,BYE,INVITE,REFER,NOTIFY,PRACK,UPDATE
27. P-Asserted-Identity: <sip:+4259876543@company1;user=phone>
28. History-Info: <sip:user112@company1>;index=1
29. --9dvaKhfhPJxCOyObvB70o0f2xfgiXN3J
30. Content-Type: application/sdp
31. Content-ID: <72e03bb9-6acc-453b-ae09-4b8671344d83>
32. Content-Disposition: Session;handling=optional;ms-proxy-2007fallback
33. v=0
34. o=- 1 0 IN IP4 10.1.1.102
35. s=session
36. c=IN IP4 10.1.1.102
37. b=CT:1000000
38. t=0 0
39. m=audio 56568 RTP/AVP 0 8 115 13 118 97 101
40. c=IN IP4 10.1.1.102
41. a=rtcp:56569
42. a=candidate:wPBogiU8NLp21GV4/zj/6WviEjTkj55FxhrdRkHiZcc 1 0tKavBj1axiy4rc19atywg UDP 0.830 10.1.1.102 56568
43. a=candidate:wPBogiU8NLp21GV4/zj/6WviEjTkj55FxhrdRkHiZcc 2 0tKavBj1axiy4rc19atywg UDP 0.830 10.1.1.102 56569
44. a=candidate:bgLnsm3DP4aSPQloj2Ak1IUYeGDPsldLRetvScj5izM 1 5VdtqvYZImPIpth0Tx5Mcg TCP 0.150 10.3.0.7 59954
45. a=candidate:bgLnsm3DP4aSPQloj2Ak1IUYeGDPsldLRetvScj5izM 2 5VdtqvYZImPIpth0Tx5Mcg TCP 0.150 10.3.0.7 59954
46. a=candidate:hdj57XrOXJwib/pE8R3lzSwmfWi3trrUtRt4pmcfb5Y 1 RDbzrPzUksHqIX1Aqv0bFA UDP 0.450 10.3.0.7 55690
47. a=candidate:hdj57XrOXJwib/pE8R3lzSwmfWi3trrUtRt4pmcfb5Y 2 RDbzrPzUksHqIX1Aqv0bFA UDP 0.450 10.3.0.7 57652
48. a=candidate:JqHr0VQ3SBc1eDZ+TPZ4wktouOoWH1fag30kyuLWlFQ 1 8yhh8eM+T1Z9w0CbEkzwfA TCP 0.250 10.1.1.102 207 52082
49. a=candidate:JqHr0VQ3SBc1eDZ+TPZ4wktouOoWH1fag30kyuLWlFQ 2 8yhh8eM+T1Z9w0CbEkzwfA TCP 0.250 10.1.1.102 52082
50. a=label:main-audio
51. a=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:f6V1oCUyKIzjLEBRg46FFt7BenyVzlLNEk3EJ6T3|2^31|1:1
52. a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:P46SfAzfbRN3d06tBm787I3Pv/3j+5hOmtM1tvdv|2^31|1:1
53. a=crypto:3 AES\_CM\_128\_HMAC\_SHA1\_80 inline:tbZ34R5hvhfBSsVLMd0/uiQ7AWOCJD5Hj+Q58HQm|2^31
54. a=rtpmap:0 PCMU/8000
55. a=rtpmap:8 PCMA/8000
56. a=rtpmap:115 x-msrta/8000
57. a=fmtp:115 bitrate=11800
58. a=rtpmap:13 CN/8000
59. a=rtpmap:118 CN/16000a=rtpmap:97 RED/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-16,36--9dvaKhfhPJxCOyObvB70o0f2xfgiXN3JContent-Type: application/sdpContent-ID: <d05db498-7556-445d-86e3-bfeb36fd52e9>v=0o=- 2 0 IN IP4 10.1.1.102s=sessionc=IN IP4 10.1.1.102b=CT:1000000t=0 0m=audio 50352 RTP/AVP 0 8 115 13 118 97 101c=IN IP4 10.1.1.102a=rtcp:50353a=ice-ufrag:LxLAa=ice-pwd:3470M/yHdvxSWmMqhs+jJF2Ea=candidate:1 1 UDP 2130706431 10.1.1.102 50352 typ hosta=candidate:1 2 UDP 2130705918 10.1.1.102 50353 typ host
60. a=candidate:2 1 tcp-pass 6555135 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
61. a=candidate:2 2 tcp-pass 6555134 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
62. a=candidate:3 1 UDP 16647679 10.0.3.7 52516 typ relay raddr 10.1.1.102 rport 55636
63. a=candidate:3 2 UDP 16647678 10.0.3.7 58728 typ relay raddr 10.1.1.102 rport 55637
64. a=candidate:4 1 tcp-act 7076863 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
65. a=candidate:4 2 tcp-act 7076350 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
66. a=candidate:5 1 tcp-act 1684798975 10.1.1.102 53970 typ srflx raddr 10.1.1.102 rport 53970a=candidate:5 2 tcp-act 1684798462 10.1.1.102 53970 typ srflx raddr 10.1.1.102 rport 53970a=label:main-audioa=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:f6V1oCUyKIzjLEBRg46FFt7BenyVzlLNEk3EJ6T3|2^31|1:1a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:P46SfAzfbRN3d06tBm787I3Pv/3j+5hOmtM1tvdv|2^31|1:1a=crypto:3 AES\_CM\_128\_HMAC\_SHA1\_80 inline:tbZ34R5hvhfBSsVLMd0/uiQ7AWOCJD5Hj+Q58HQm|2^31a=rtpmap:0 PCMU/8000a=rtpmap:8 PCMA/8000a=rtpmap:115 x-msrta/8000a=fmtp:115 bitrate=11800a=rtpmap:13 CN/8000a=rtpmap:118 CN/16000a=rtpmap:97 RED/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-16,36--9dvaKhfhPJxCOyObvB70o0f2xfgiXN3JContent-Type: application/gw-sdp; x-bypassid=9CD08A01-E998-456a-AC8A-D028930E5933Content-ID: <466ac626-be34-4f8d-ba0d-c7bacf53c0ac>Content-Disposition: Session;handling=optionalv=0o=Gateway 94331345 94331031 IN IP4 10.1.2.12s=sessionc=IN IP4 10.1.2.12t=0 0m=audio 6430 RTP/SAVP 0 8 4 2 3 13 101c=IN IP4 10.1.2.12a=rtcp:6431a=x-bypassid:9CD08A01-E998-456a-AC8A-D028930E5933a=crypto:1 AES\_CM\_128\_HMAC\_SHA1\_80 inline:uch9eRm5IMoOhC+jNRprVaEuvK2JN0upP2+9ciM9|2^31|129:1a=sendrecva=rtpmap:0 PCMU/8000a=rtpmap:8 PCMA/8000a=rtpmap:4 G723/8000a=fmtp:4 annexa=yesa=rtpmap:2 G726-32/8000a=rtpmap:3 GSM/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-15a=ptime:20a=x-mediasettings:signalboostunsupported
67. --9dvaKhfhPJxCOyObvB70o0f2xfgiXN3J--

#### Step 17: 200 Message Is Sent by the Protocol Client

SIP/2.0 200 OKVia: SIP/2.0/TLS 10.1.1.54:5061;branch=z9hG4bK1C7C8A0E.19AB9CC7A4B7C3D3;branched=TRUE;ms-internal-info="cehce-xXzqcRs3A\_ZSAwy8D4JLgyqxDKREgfIVFt6noRjHnKsZUY47CgAA"Via: SIP/2.0/TLS 10.1.1.102:57350;branch=z9hG4bKe82f3c;ms-received-port=57350;ms-received-cid=475900From: <sip:4259876543;phone-context=Location1@company1;user=phone>;epid=CDCFEF8F18;tag=3d965223eaTo: <sip:+14251234567@company1;user=phone>;epid=54dd5867e8;tag=c608fec21aCall-ID: df601b2d-e42e-4677-b921-c9dbf4e25940CSeq: 35 INVITERecord-Route: <sip:PROXY.company1:5061;transport=tls;opaque=state:F:Ci.R475300:Ieh.gU65xODvwq\_j78KvdcC-dRxH7lEBsv8oECfECdswTe7QW4niMEtoTOi\_iwBgjHnKsZgY3jngAA;lr;ms-route-sig=dcrEzxvkq3iKgc2ApWyiXbCYC7NNwE-pYCMYxgjFJ3kxfHnKsZgY3jngAA>;tag=45F7A969AE33112CB9877940D7F56D40Contact: <sip:user112@company1;opaque=user:epid:jVxLXKl9l12yFm93r\_ArNgAA;gruu>User-Agent: Client 1.0Supported: histinfoSupported: ms-safe-transferAllow: INVITE, BYE, ACK, CANCEL, INFO, UPDATE, REFER, NOTIFY, BENOTIFY, OPTIONSSession-Expires: 720;refresher=uacms-accepted-content-id: <466ac626-be34-4f8d-ba0d-c7bacf53c0ac>P-Preferred-Identity: <sip:user112@company1>, <tel:+14251234567>Supported: ms-bypassSupported: replacesProxy-Authorization: TLS-DSK qop="auth", realm="SIP Communications Service", opaque="F755045D", targetname="PROXY.company1", crand="bdaff021", cnum="26", response="ec06b619fdde8d00dae6a5e3ef008db607f08538"Content-Type: application/sdpContent-Length: 362v=0o=- 0 0 IN IP4 192.168.1.114s=sessionc=IN IP4 192.168.1.114b=CT:99980t=0 0m=audio 10228 RTP/SAVP 0 8 4 101a=crypto:1 AES\_CM\_128\_HMAC\_SHA1\_80 inline:coOkWf6dIPsrYkRXi7QkjHb4n1ZwOQyZn0wWwBZR|2^31|1:1a=maxptime:200a=rtpmap:0 PCMU/8000a=rtpmap:8 PCMA/8000a=rtpmap:4 G723/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-16a=x-bypass

### Outbound Call

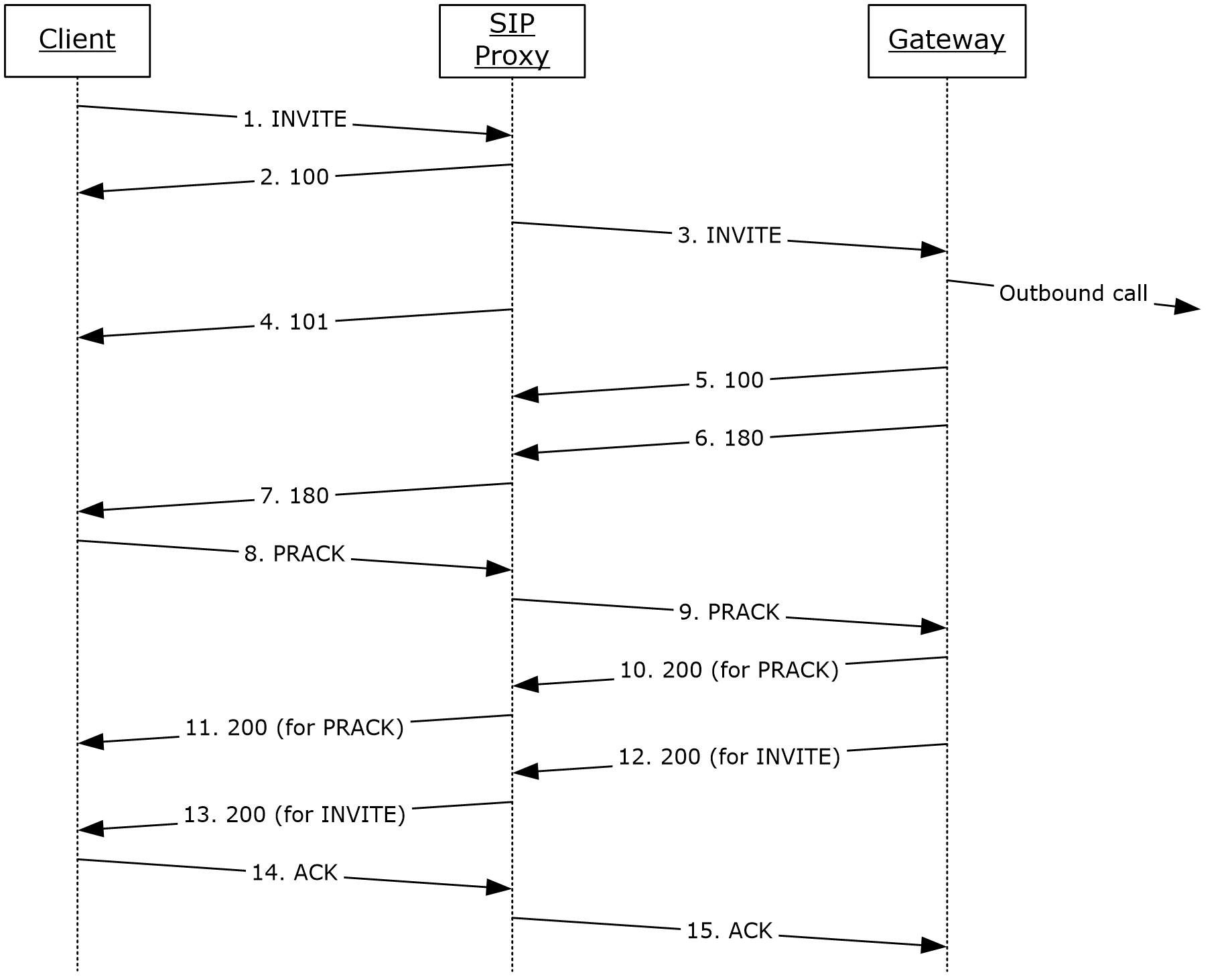


Figure 9: Outbound call

The messages in the following subsections illustrate the use of the **ms-bypass** option tag in a **Supported** header in messages that are sent from and received by a protocol client.

#### Step 1: INVITE Message Is Sent by the Protocol Client

1. INVITE sip:+14258901234@company1;user=phone SIP/2.0
2. Via: SIP/2.0/TLS 192.168.1.114:4535
3. Max-Forwards: 70
4. From: <sip:user112@company1>;tag=ed04066c4a;epid=54dd5867e8
5. To: <sip:+14258901234@company1;user=phone>
6. Call-ID: e571df11a45947f1a5b90da8d957b8ae
7. CSeq: 1 INVITE
8. Contact: <sip:user112@company1;opaque=user:epid:jVxLXKl9l12yFm93r\_ArNgAA;gruu>
9. User-Agent: Client 1.0
10. Ms-Conversation-ID: AcrCQkQ2CGV+fQQpS5OprWuDL+KaYQ==
11. Supported: timer
12. Supported: histinfo
13. Supported: ms-safe-transfer
14. Supported: ms-sender
15. Supported: ms-early-media
16. Supported: 100rel
17. ms-keep-alive: UAC;hop-hop=yes
18. Allow: INVITE, BYE, ACK, CANCEL, INFO, UPDATE, REFER, NOTIFY, BENOTIFY, OPTIONS
19. P-Preferred-Identity: <sip:user112@company1>, <tel:+14251234567>
20. Supported: ms-bypass
21. Supported: replaces
22. Supported: ms-conf-invite
23. Proxy-Authorization: TLS-DSK qop="auth", realm="SIP Communications Service", opaque="F755045D", targetname="PROXY.company1", crand="738839d3", cnum="12", response="2b5e54b5d29a1493e07894772e5ce0dcca06bdf3"
24. Content-Type: multipart/alternative;boundary="----=\_NextPart\_000\_0003\_01CAC1FF.366488E0"
25. Content-Length: 3052
26. ------=\_NextPart\_000\_0003\_01CAC1FF.366488E0
27. Content-Type: application/sdp
28. Content-Transfer-Encoding: 7bit
29. Content-ID: <2dd1547f1a2043c2a622586b444229e2>
30. Content-Disposition: session; handling=optional; ms-proxy-2007fallback
31. v=0
32. o=- 0 0 IN IP4 192.168.1.114
33. s=session
34. c=IN IP4 192.168.1.114
35. b=CT:99980
36. t=0 0
37. m=audio 25486 RTP/AVP 114 9 112 111 0 8 116 115 4 97 13 118 101
38. a=candidate:XhpPtyjMgVxDIhWFgBIMhdLFIVXLwt+YRBaobG+X43A 1 4Q/jKJde54nbJ5sfchXniA UDP 0.830 192.168.1.114 25486
39. a=candidate:XhpPtyjMgVxDIhWFgBIMhdLFIVXLwt+YRBaobG+X43A 2 4Q/jKJde54nbJ5sfchXniA UDP 0.830 192.168.1.114 25487
40. a=candidate:+oWYSe96HnD9j7GRgjAf47ImvcM2GeooLhFH8L6sN1M 1 wiGTb6hg53yn1/Keu8TGSg TCP 0.190 10.3.0.7 57587
41. a=candidate:+oWYSe96HnD9j7GRgjAf47ImvcM2GeooLhFH8L6sN1M 2 wiGTb6hg53yn1/Keu8TGSg TCP 0.190 10.3.0.7 57587
42. a=candidate:+LqcUBIcwTUej3u0lhJq7UET5SYTrNNWvpIzn7S4lho 1 X3SHHBGYzFqLK8TzSd5vNQ UDP 0.490 10.3.0.7 51247
43. a=candidate:+LqcUBIcwTUej3u0lhJq7UET5SYTrNNWvpIzn7S4lho 2 X3SHHBGYzFqLK8TzSd5vNQ UDP 0.490 10.3.0.7 50976
44. a=candidate:DzxkqWh6pd3wMmObq9itqTbhQ6yI4DLm1I8ZRbI3J6c 1 AUw+lgvF2GlnnLiF4otDhg TCP 0.250 192.168.1.114 50007
45. a=candidate:DzxkqWh6pd3wMmObq9itqTbhQ6yI4DLm1I8ZRbI3J6c 2 AUw+lgvF2GlnnLiF4otDhg TCP 0.250 192.168.1.114 50007
46. a=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:OY1qCCFx84fwIkrR39XpPDA2HuNdtAB+6ekK1y5a|2^31|1:1
47. a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:b28SzCBSdH7eBr13AhecN34gKh8OeCYQG6IMwxbC|2^31|1:1
48. a=crypto:3 AES\_CM\_128\_HMAC\_SHA1\_80 inline:8LUckwDobd31ORi6KGZLYf+My7wvCwftc5Nw7G79|2^31
49. a=maxptime:200
50. a=rtpmap:114 x-msrta/16000
51. a=fmtp:114 bitrate=29000
52. a=rtpmap:9 G722/8000
53. a=rtpmap:112 G7221/16000
54. a=fmtp:112 bitrate=24000
55. a=rtpmap:111 SIREN/16000
56. a=fmtp:111 bitrate=16000
57. a=rtpmap:0 PCMU/8000
58. a=rtpmap:8 PCMA/8000
59. a=rtpmap:116 AAL2-G726-32/8000
60. a=rtpmap:115 x-msrta/8000
61. a=fmtp:115 bitrate=11800
62. a=rtpmap:4 G723/8000
63. a=rtpmap:97 RED/8000
64. a=rtpmap:13 CN/8000
65. a=rtpmap:118 CN/16000
66. a=rtpmap:101 telephone-event/8000
67. a=fmtp:101 0-16
68. a=encryption:optional
69. a=x-bypassid:9CD08A01-E998-456a-AC8A-D028930E5933
70. ------=\_NextPart\_000\_0003\_01CAC1FF.366488E0
71. Content-Type: application/sdp
72. Content-Transfer-Encoding: 7bit
73. Content-ID: <3d45476919eb4c81be0c4e19c730c655>
74. Content-Disposition: session; handling=optional
75. v=0
76. o=- 0 0 IN IP4 192.168.1.114
77. s=session
78. c=IN IP4 192.168.1.114
79. b=CT:99980
80. t=0 0
81. m=audio 28238 RTP/AVP 114 9 112 111 0 8 116 115 4 97 13 118 101
82. a=ice-ufrag:ayqK
83. a=ice-pwd:ckRbkR22lv38PhlmqvzmVe5n
84. a=candidate:1 1 UDP 2130706431 192.168.1.114 28238 typ host
85. a=candidate:1 2 UDP 2130705918 192.168.1.114 28239 typ host
86. a=candidate:2 1 TCP-PASS 6556159 10.3.0.7 59752 typ relay raddr 192.168.1.114 rport 50031
87. a=candidate:2 2 TCP-PASS 6556158 10.3.0.7 59752 typ relay raddr 192.168.1.114 rport 50031
88. a=candidate:3 1 UDP 16648703 10.3.0.7 50217 typ relay raddr 192.168.1.114 rport 50006
89. a=candidate:3 2 UDP 16648702 10.3.0.7 58942 typ relay raddr 192.168.1.114 rport 50007
90. a=candidate:4 1 TCP-ACT 7076863 10.3.0.7 59752 typ relay raddr 192.168.1.114 rport 50031
91. a=candidate:4 2 TCP-ACT 7076350 10.3.0.7 59752 typ relay raddr 192.168.1.114 rport 50031
92. a=candidate:5 1 TCP-ACT 1684798975 192.168.1.114 50031 typ srflx raddr 192.168.1.114 rport 50031
93. a=candidate:5 2 TCP-ACT 1684798462 192.168.1.114 50031 typ srflx raddr 192.168.1.114 rport 50031
94. a=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:OY1qCCFx84fwIkrR39XpPDA2HuNdtAB+6ekK1y5a|2^31|1:1
95. a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:b28SzCBSdH7eBr13AhecN34gKh8OeCYQG6IMwxbC|2^31|1:1
96. a=crypto:3 AES\_CM\_128\_HMAC\_SHA1\_80 inline:8LUckwDobd31ORi6KGZLYf+My7wvCwftc5Nw7G79|2^31
97. a=maxptime:200
98. a=rtpmap:114 x-msrta/16000
99. a=fmtp:114 bitrate=29000
100. a=rtpmap:9 G722/8000
101. a=rtpmap:112 G7221/16000
102. a=fmtp:112 bitrate=24000
103. a=rtpmap:111 SIREN/16000
104. a=fmtp:111 bitrate=16000
105. a=rtpmap:0 PCMU/8000
106. a=rtpmap:8 PCMA/8000
107. a=rtpmap:116 AAL2-G726-32/8000
108. a=rtpmap:115 x-msrta/8000
109. a=fmtp:115 bitrate=11800
110. a=rtpmap:4 G723/8000
111. a=rtpmap:97 RED/8000
112. a=rtpmap:13 CN/8000
113. a=rtpmap:118 CN/16000
114. a=rtpmap:101 telephone-event/8000
115. a=fmtp:101 0-16
116. a=encryption:optional
117. a=x-bypassid:9CD08A01-E998-456a-AC8A-D028930E5933
118. ------=\_NextPart\_000\_0003\_01CAC1FF.366488E0--

#### Step 13: 200 OK Message Is Received by the Protocol Client

1. SIP/2.0 200 OK
2. Authentication-Info: TLS-DSK qop="auth", opaque="F755045D", srand="1D9666D9", snum="17", rspauth="3359c8ac2e6229b2eb9738ac707dc8c3e54f65f0", targetname="PROXY.company1", realm="SIP Communications Service", version=4
3. Via: SIP/2.0/TLS 192.168.1.114:4535;ms-received-port=4535;ms-received-cid=475300
4. FROM: "user112"<sip:user112@company1>;tag=ed04066c4a;epid=54dd5867e8
5. TO: <sip:+14258901234@company1;user=phone>;tag=201fec487e;epid=CDCFEF8F18
6. CSEQ: 1 INVITE
7. CALL-ID: e571df11a45947f1a5b90da8d957b8ae
8. RECORD-ROUTE: <sip:PROXY.company1:5061;transport=tls;opaque=state:F:Ci.R475300;lr;ms-route-sig=dcw0SbeehYaHu9dRxfcQNPNLaiGM-c5DzikYU7AfKG2hHch3QtgY3jngAA>
9. CONTACT: <sip:ms5.company1@company1;gruu;opaque=srvr:MediationServer:XzRY6u68aVipMcYVfYs0hQAA;grid=46236573d0ae4a339d83726b2bf7f7ab>;isGateway
10. CONTENT-LENGTH: 422
11. SUPPORTED: replaces
12. SUPPORTED: ms-safe-transfer
13. SUPPORTED: ms-bypass
14. SUPPORTED: ms-dialog-route-set-update
15. SUPPORTED: gruu-10
16. SUPPORTED: timer
17. SUPPORTED: 100rel
18. CONTENT-TYPE: application/gw-sdp
19. ALLOW: ACK
20. P-ASSERTED-IDENTITY: <sip:+14258901234@company1;user=phone>
21. SERVER: Mediation Server
22. Ms-Accepted-Content-ID: <3d45476919eb4c81be0c4e19c730c655>
23. ms-trunking-peer: gateway.company1.com;trunk=trunk1;User-Agent="Gateway 1.0"
24. Allow: CANCEL,BYE,INVITE,REFER,NOTIFY,PRACK,UPDATE
25. Session-Expires: 1800;refresher=uas
26. Min-SE: 90
27. v=0
28. o=Gateway 1303417666 1303417345 IN IP4 10.1.2.12
29. s=session
30. c=IN IP4 10.1.2.12
31. t=0 0
32. m=audio 6390 RTP/SAVP 0 13 101
33. c=IN IP4 10.1.2.12
34. a=rtcp:6391
35. a=x-bypass
36. a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:bN1zDJ0LC8QYNvMIdohDtGkWD/rCastpGbz5ObNo|2^31|244:1
37. a=sendrecv
38. a=rtpmap:0 PCMU/8000
39. a=rtpmap:101 telephone-event/8000
40. a=fmtp:101 0-15
41. a=ptime:20
42. a=x-mediasettings:signalboostunsupported

## ms-accepted-content-id SIP Header

### Inbound Call

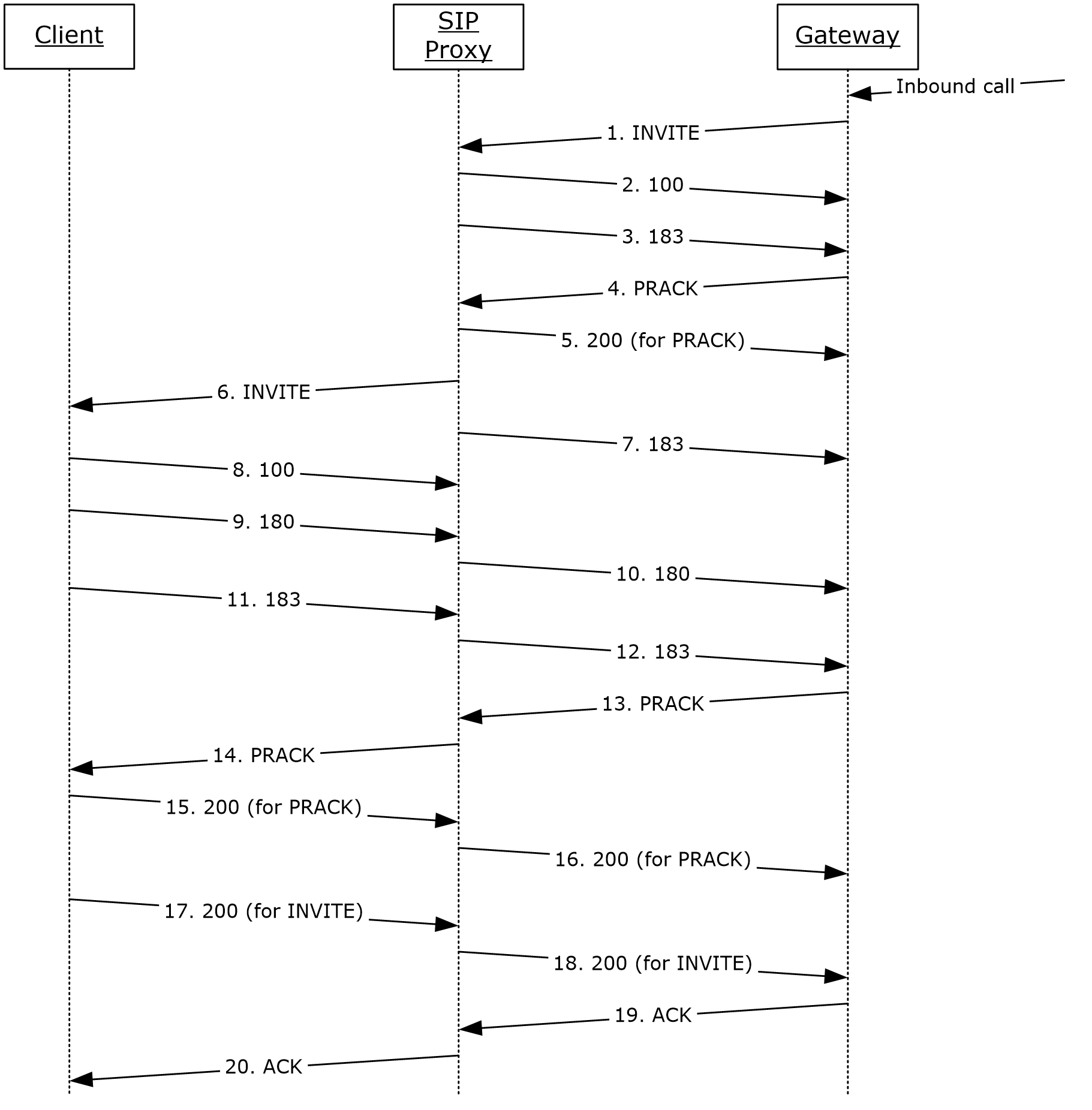


Figure 10: Inbound call

The following messages illustrate the use of the **ms-accepted-content-id** Session Initiation Protocol (SIP) header in messages that are sent from and received by a protocol client.

#### Step 6: INVITE Message Is Received by the Protocol Client

1. INVITE sip:192.168.1.114:4535;transport=tls;ms-opaque=acee5f6d3a;ms-received-cid=475300 SIP/2.0
2. Record-Route: <sip:PROXY.company1:5061;transport=tls;opaque=state:F:Ci.R475300:Ieh.gU65xODvwq\_j78KvdcC-dRxH7lEBsv8oECfECdswTe7QW4niMEtoTOi\_iwBgjHnKsZgY3jngAA;lr;ms-route-sig=dcrEzxvkq3iKgc2ApWyiXbCYC7NNwE-pYCMYxgjFJ3kxfHnKsZgY3jngAA>;tag=45F7A969AE33112CB9877940D7F56D40
3. Via: SIP/2.0/TLS 10.1.1.54:5061;branch=z9hG4bK1C7C8A0E.19AB9CC7A4B7C3D3;branched=TRUE;ms-internal-info="cehce-xXzqcRs3A\_ZSAwy8D4JLgyqxDKREgfIVFt6noRjHnKsZUY47CgAA"
4. Authentication-Info: TLS-DSK qop="auth", opaque="F755045D", srand="CC46B5FD", snum="26", rspauth="d6179291f72761e057a67adb7288fd256c2b1e4d", targetname="PROXY.company1", realm="SIP Communications Service", version=4
5. Max-Forwards: 69
6. Content-Length: 3161
7. Via: SIP/2.0/TLS 10.1.1.102:57350;branch=z9hG4bKe82f3c;ms-received-port=57350;ms-received-cid=475900
8. From: <sip:4259876543;phone-context=Location1@company1;user=phone>;epid=CDCFEF8F18;tag=3d965223ea
9. To: <sip:+14251234567@company1;user=phone>;epid=54dd5867e8
10. CSeq: 35 INVITE
11. Call-ID: df601b2d-e42e-4677-b921-c9dbf4e25940
12. Contact: <sip:ms5.company1@company1;gruu;opaque=srvr:MediationServer:XzRY6u68aVipMcYVfYs0hQAA;grid=bd9c42fc618147d0af4d8f84f718910b>;isGateway
13. Supported: replaces
14. Supported: ms-safe-transfer
15. Supported: ms-bypass
16. Supported: ms-dialog-route-set-update
17. Supported: timer
18. Supported: 100rel
19. Supported: gruu-10
20. User-Agent: Mediation Server
21. Content-Type: multipart/alternative; boundary=9dvaKhfhPJxCOyObvB70o0f2xfgiXN3J
22. Allow: ACK
23. ms-trunking-peer: gateway.company1.com;trunk=trunk1;User-Agent="Gateway 1.0"
24. Session-Expires: 1800
25. Min-SE: 90
26. Allow: CANCEL,BYE,INVITE,REFER,NOTIFY,PRACK,UPDATE
27. P-Asserted-Identity: <sip:+4259876543@company1;user=phone>
28. History-Info: <sip:user112@company1>;index=1
29. --9dvaKhfhPJxCOyObvB70o0f2xfgiXN3J
30. Content-Type: application/sdp
31. Content-ID: <72e03bb9-6acc-453b-ae09-4b8671344d83>
32. Content-Disposition: Session;handling=optional;ms-proxy-2007fallback
33. v=0
34. o=- 1 0 IN IP4 10.1.1.102
35. s=session
36. c=IN IP4 10.1.1.102
37. b=CT:1000000
38. t=0 0
39. m=audio 56568 RTP/AVP 0 8 115 13 118 97 101
40. c=IN IP4 10.1.1.102
41. a=rtcp:56569
42. a=candidate:wPBogiU8NLp21GV4/zj/6WviEjTkj55FxhrdRkHiZcc 1 0tKavBj1axiy4rc19atywg UDP 0.830 10.1.1.102 56568
43. a=candidate:wPBogiU8NLp21GV4/zj/6WviEjTkj55FxhrdRkHiZcc 2 0tKavBj1axiy4rc19atywg UDP 0.830 10.1.1.102 56569
44. a=candidate:bgLnsm3DP4aSPQloj2Ak1IUYeGDPsldLRetvScj5izM 1 5VdtqvYZImPIpth0Tx5Mcg TCP 0.150 10.3.0.7 59954
45. a=candidate:bgLnsm3DP4aSPQloj2Ak1IUYeGDPsldLRetvScj5izM 2 5VdtqvYZImPIpth0Tx5Mcg TCP 0.150 10.3.0.7 59954
46. a=candidate:hdj57XrOXJwib/pE8R3lzSwmfWi3trrUtRt4pmcfb5Y 1 RDbzrPzUksHqIX1Aqv0bFA UDP 0.450 10.3.0.7 55690
47. a=candidate:hdj57XrOXJwib/pE8R3lzSwmfWi3trrUtRt4pmcfb5Y 2 RDbzrPzUksHqIX1Aqv0bFA UDP 0.450 10.3.0.7 57652
48. a=candidate:JqHr0VQ3SBc1eDZ+TPZ4wktouOoWH1fag30kyuLWlFQ 1 8yhh8eM+T1Z9w0CbEkzwfA TCP 0.250 10.1.1.102 207 52082
49. a=candidate:JqHr0VQ3SBc1eDZ+TPZ4wktouOoWH1fag30kyuLWlFQ 2 8yhh8eM+T1Z9w0CbEkzwfA TCP 0.250 10.1.1.102 52082
50. a=label:main-audio
51. a=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:f6V1oCUyKIzjLEBRg46FFt7BenyVzlLNEk3EJ6T3|2^31|1:1
52. a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:P46SfAzfbRN3d06tBm787I3Pv/3j+5hOmtM1tvdv|2^31|1:1
53. a=crypto:3 AES\_CM\_128\_HMAC\_SHA1\_80 inline:tbZ34R5hvhfBSsVLMd0/uiQ7AWOCJD5Hj+Q58HQm|2^31
54. a=rtpmap:0 PCMU/8000
55. a=rtpmap:8 PCMA/8000
56. a=rtpmap:115 x-msrta/8000
57. a=fmtp:115 bitrate=11800
58. a=rtpmap:13 CN/8000
59. a=rtpmap:118 CN/16000a=rtpmap:97 RED/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-16,36--9dvaKhfhPJxCOyObvB70o0f2xfgiXN3JContent-Type: application/sdpContent-ID: <d05db498-7556-445d-86e3-bfeb36fd52e9>v=0o=- 2 0 IN IP4 10.1.1.102s=sessionc=IN IP4 10.1.1.102b=CT:1000000t=0 0m=audio 50352 RTP/AVP 0 8 115 13 118 97 101c=IN IP4 10.1.1.102a=rtcp:50353a=ice-ufrag:LxLAa=ice-pwd:3470M/yHdvxSWmMqhs+jJF2Ea=candidate:1 1 UDP 2130706431 10.1.1.102 50352 typ hosta=candidate:1 2 UDP 2130705918 10.1.1.102 50353 typ host
60. a=candidate:2 1 tcp-pass 6555135 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
61. a=candidate:2 2 tcp-pass 6555134 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
62. a=candidate:3 1 UDP 16647679 10.0.3.7 52516 typ relay raddr 10.1.1.102 rport 55636
63. a=candidate:3 2 UDP 16647678 10.0.3.7 58728 typ relay raddr 10.1.1.102 rport 55637
64. a=candidate:4 1 tcp-act 7076863 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
65. a=candidate:4 2 tcp-act 7076350 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
66. a=candidate:5 1 tcp-act 1684798975 10.1.1.102 53970 typ srflx raddr 10.1.1.102 rport 53970a=candidate:5 2 tcp-act 1684798462 10.1.1.102 53970 typ srflx raddr 10.1.1.102 rport 53970a=label:main-audioa=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:f6V1oCUyKIzjLEBRg46FFt7BenyVzlLNEk3EJ6T3|2^31|1:1a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:P46SfAzfbRN3d06tBm787I3Pv/3j+5hOmtM1tvdv|2^31|1:1a=crypto:3 AES\_CM\_128\_HMAC\_SHA1\_80 inline:tbZ34R5hvhfBSsVLMd0/uiQ7AWOCJD5Hj+Q58HQm|2^31a=rtpmap:0 PCMU/8000a=rtpmap:8 PCMA/8000a=rtpmap:115 x-msrta/8000a=fmtp:115 bitrate=11800a=rtpmap:13 CN/8000a=rtpmap:118 CN/16000a=rtpmap:97 RED/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-16,36--9dvaKhfhPJxCOyObvB70o0f2xfgiXN3JContent-Type: application/gw-sdp; x-bypassid=9CD08A01-E998-456a-AC8A-D028930E5933Content-ID: <466ac626-be34-4f8d-ba0d-c7bacf53c0ac>Content-Disposition: Session;handling=optionalv=0o=Gateway 94331345 94331031 IN IP4 10.1.2.12s=sessionc=IN IP4 10.1.2.12t=0 0m=audio 6430 RTP/SAVP 0 8 4 2 3 13 101c=IN IP4 10.1.2.12a=rtcp:6431a=x-bypassid:9CD08A01-E998-456a-AC8A-D028930E5933a=crypto:1 AES\_CM\_128\_HMAC\_SHA1\_80 inline:uch9eRm5IMoOhC+jNRprVaEuvK2JN0upP2+9ciM9|2^31|129:1a=sendrecva=rtpmap:0 PCMU/8000a=rtpmap:8 PCMA/8000a=rtpmap:4 G723/8000a=fmtp:4 annexa=yesa=rtpmap:2 G726-32/8000a=rtpmap:3 GSM/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-15a=ptime:20a=x-mediasettings:signalboostunsupported
67. --9dvaKhfhPJxCOyObvB70o0f2xfgiXN3J—

#### Step 17: 200 Message Is Sent by the Protocol Client

1. SIP/2.0 200 OKVia: SIP/2.0/TLS 10.1.1.54:5061;branch=z9hG4bK1C7C8A0E.19AB9CC7A4B7C3D3;branched=TRUE;ms-internal-info="cehce-xXzqcRs3A\_ZSAwy8D4JLgyqxDKREgfIVFt6noRjHnKsZUY47CgAA"Via: SIP/2.0/TLS 10.1.1.102:57350;branch=z9hG4bKe82f3c;ms-received-port=57350;ms-received-cid=475900From: <sip:4259876543;phone-context=Location1@company1;user=phone>;epid=CDCFEF8F18;tag=3d965223eaTo: <sip:+14251234567@company1;user=phone>;epid=54dd5867e8;tag=c608fec21aCall-ID: df601b2d-e42e-4677-b921-c9dbf4e25940CSeq: 35 INVITERecord-Route: <sip:PROXY.company1:5061;transport=tls;opaque=state:F:Ci.R475300:Ieh.gU65xODvwq\_j78KvdcC-dRxH7lEBsv8oECfECdswTe7QW4niMEtoTOi\_iwBgjHnKsZgY3jngAA;lr;ms-route-sig=dcrEzxvkq3iKgc2ApWyiXbCYC7NNwE-pYCMYxgjFJ3kxfHnKsZgY3jngAA>;tag=45F7A969AE33112CB9877940D7F56D40Contact: <sip:user112@company1;opaque=user:epid:jVxLXKl9l12yFm93r\_ArNgAA;gruu>User-Agent: Client 1.0Supported: histinfoSupported: ms-safe-transferAllow: INVITE, BYE, ACK, CANCEL, INFO, UPDATE, REFER, NOTIFY, BENOTIFY, OPTIONSSession-Expires: 720;refresher=uacms-accepted-content-id: <466ac626-be34-4f8d-ba0d-c7bacf53c0ac>P-Preferred-Identity: <sip:user112@company1>, <tel:+14251234567>Supported: ms-bypassSupported: replacesProxy-Authorization: TLS-DSK qop="auth", realm="SIP Communications Service", opaque="F755045D", targetname="PROXY.company1", crand="bdaff021", cnum="26", response="ec06b619fdde8d00dae6a5e3ef008db607f08538"Content-Type: application/sdpContent-Length: 362v=0o=- 0 0 IN IP4 192.168.1.114s=sessionc=IN IP4 192.168.1.114b=CT:99980t=0 0m=audio 10228 RTP/SAVP 0 8 4 101a=crypto:1 AES\_CM\_128\_HMAC\_SHA1\_80 inline:coOkWf6dIPsrYkRXi7QkjHb4n1ZwOQyZn0wWwBZR|2^31|1:1a=maxptime:200a=rtpmap:0 PCMU/8000a=rtpmap:8 PCMA/8000a=rtpmap:4 G723/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-16a=x-bypass

### Outbound Call

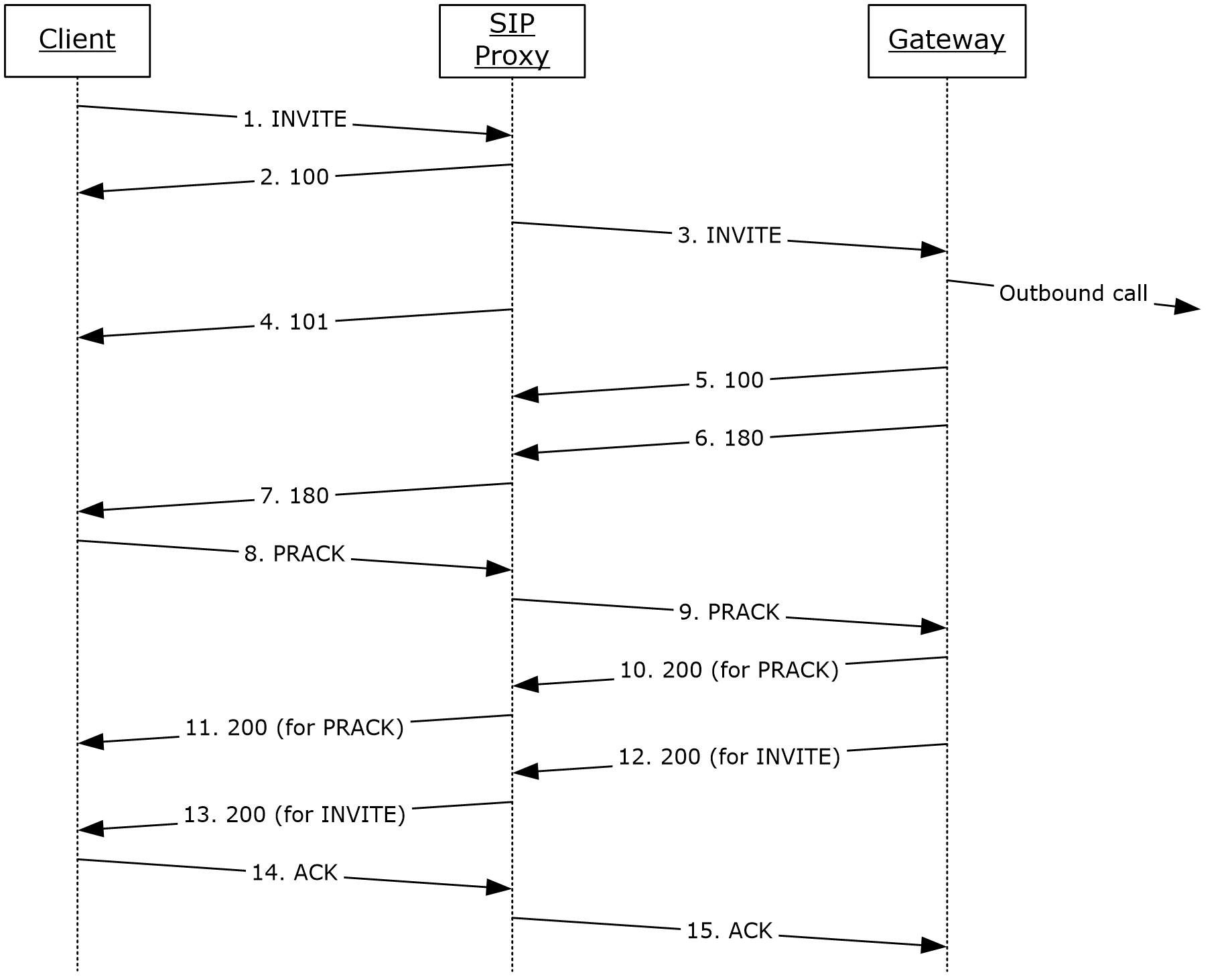


Figure 11: Outbound call

The messages in the following subsections illustrate the use of the **ms-accepted-content-id** Session Initiation Protocol (SIP) header in messages that are sent from and received by a protocol client.

#### Step 1: INVITE Message Is Sent by the Protocol Client

1. INVITE sip:+14258901234@company1;user=phone SIP/2.0
2. Via: SIP/2.0/TLS 192.168.1.114:4535
3. Max-Forwards: 70
4. From: <sip:user112@company1>;tag=ed04066c4a;epid=54dd5867e8
5. To: <sip:+14258901234@company1;user=phone>
6. Call-ID: e571df11a45947f1a5b90da8d957b8ae
7. CSeq: 1 INVITE
8. Contact: <sip:user112@company1;opaque=user:epid:jVxLXKl9l12yFm93r\_ArNgAA;gruu>
9. User-Agent: Client 1.0
10. Ms-Conversation-ID: AcrCQkQ2CGV+fQQpS5OprWuDL+KaYQ==
11. Supported: timer
12. Supported: histinfo
13. Supported: ms-safe-transfer
14. Supported: ms-sender
15. Supported: ms-early-media
16. Supported: 100rel
17. ms-keep-alive: UAC;hop-hop=yes
18. Allow: INVITE, BYE, ACK, CANCEL, INFO, UPDATE, REFER, NOTIFY, BENOTIFY, OPTIONS
19. P-Preferred-Identity: <sip:user112@company1>, <tel:+14251234567>
20. Supported: ms-bypass
21. Supported: replaces
22. Supported: ms-conf-invite
23. Proxy-Authorization: TLS-DSK qop="auth", realm="SIP Communications Service", opaque="F755045D", targetname="PROXY.company1", crand="738839d3", cnum="12", response="2b5e54b5d29a1493e07894772e5ce0dcca06bdf3"
24. Content-Type: multipart/alternative;boundary="----=\_NextPart\_000\_0003\_01CAC1FF.366488E0"
25. Content-Length: 3052
26. ------=\_NextPart\_000\_0003\_01CAC1FF.366488E0
27. Content-Type: application/sdp
28. Content-Transfer-Encoding: 7bit
29. Content-ID: <2dd1547f1a2043c2a622586b444229e2>
30. Content-Disposition: session; handling=optional; ms-proxy-2007fallback
31. v=0
32. o=- 0 0 IN IP4 192.168.1.114
33. s=session
34. c=IN IP4 192.168.1.114
35. b=CT:99980
36. t=0 0
37. m=audio 25486 RTP/AVP 114 9 112 111 0 8 116 115 4 97 13 118 101
38. a=candidate:XhpPtyjMgVxDIhWFgBIMhdLFIVXLwt+YRBaobG+X43A 1 4Q/jKJde54nbJ5sfchXniA UDP 0.830 192.168.1.114 25486
39. a=candidate:XhpPtyjMgVxDIhWFgBIMhdLFIVXLwt+YRBaobG+X43A 2 4Q/jKJde54nbJ5sfchXniA UDP 0.830 192.168.1.114 25487
40. a=candidate:+oWYSe96HnD9j7GRgjAf47ImvcM2GeooLhFH8L6sN1M 1 wiGTb6hg53yn1/Keu8TGSg TCP 0.190 10.3.0.7 57587
41. a=candidate:+oWYSe96HnD9j7GRgjAf47ImvcM2GeooLhFH8L6sN1M 2 wiGTb6hg53yn1/Keu8TGSg TCP 0.190 10.3.0.7 57587
42. a=candidate:+LqcUBIcwTUej3u0lhJq7UET5SYTrNNWvpIzn7S4lho 1 X3SHHBGYzFqLK8TzSd5vNQ UDP 0.490 10.3.0.7 51247
43. a=candidate:+LqcUBIcwTUej3u0lhJq7UET5SYTrNNWvpIzn7S4lho 2 X3SHHBGYzFqLK8TzSd5vNQ UDP 0.490 10.3.0.7 50976
44. a=candidate:DzxkqWh6pd3wMmObq9itqTbhQ6yI4DLm1I8ZRbI3J6c 1 AUw+lgvF2GlnnLiF4otDhg TCP 0.250 192.168.1.114 50007
45. a=candidate:DzxkqWh6pd3wMmObq9itqTbhQ6yI4DLm1I8ZRbI3J6c 2 AUw+lgvF2GlnnLiF4otDhg TCP 0.250 192.168.1.114 50007
46. a=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:OY1qCCFx84fwIkrR39XpPDA2HuNdtAB+6ekK1y5a|2^31|1:1
47. a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:b28SzCBSdH7eBr13AhecN34gKh8OeCYQG6IMwxbC|2^31|1:1
48. a=crypto:3 AES\_CM\_128\_HMAC\_SHA1\_80 inline:8LUckwDobd31ORi6KGZLYf+My7wvCwftc5Nw7G79|2^31
49. a=maxptime:200
50. a=rtpmap:114 x-msrta/16000
51. a=fmtp:114 bitrate=29000
52. a=rtpmap:9 G722/8000
53. a=rtpmap:112 G7221/16000
54. a=fmtp:112 bitrate=24000
55. a=rtpmap:111 SIREN/16000
56. a=fmtp:111 bitrate=16000
57. a=rtpmap:0 PCMU/8000
58. a=rtpmap:8 PCMA/8000
59. a=rtpmap:116 AAL2-G726-32/8000
60. a=rtpmap:115 x-msrta/8000
61. a=fmtp:115 bitrate=11800
62. a=rtpmap:4 G723/8000
63. a=rtpmap:97 RED/8000
64. a=rtpmap:13 CN/8000
65. a=rtpmap:118 CN/16000
66. a=rtpmap:101 telephone-event/8000
67. a=fmtp:101 0-16
68. a=encryption:optional
69. a=x-bypassid:9CD08A01-E998-456a-AC8A-D028930E5933
70. ------=\_NextPart\_000\_0003\_01CAC1FF.366488E0
71. Content-Type: application/sdp
72. Content-Transfer-Encoding: 7bit
73. Content-ID: <3d45476919eb4c81be0c4e19c730c655>
74. Content-Disposition: session; handling=optional
75. v=0
76. o=- 0 0 IN IP4 192.168.1.114
77. s=session
78. c=IN IP4 192.168.1.114
79. b=CT:99980
80. t=0 0
81. m=audio 28238 RTP/AVP 114 9 112 111 0 8 116 115 4 97 13 118 101
82. a=ice-ufrag:ayqK
83. a=ice-pwd:ckRbkR22lv38PhlmqvzmVe5n
84. a=candidate:1 1 UDP 2130706431 192.168.1.114 28238 typ host
85. a=candidate:1 2 UDP 2130705918 192.168.1.114 28239 typ host
86. a=candidate:2 1 TCP-PASS 6556159 10.3.0.7 59752 typ relay raddr 192.168.1.114 rport 50031
87. a=candidate:2 2 TCP-PASS 6556158 10.3.0.7 59752 typ relay raddr 192.168.1.114 rport 50031
88. a=candidate:3 1 UDP 16648703 10.3.0.7 50217 typ relay raddr 192.168.1.114 rport 50006
89. a=candidate:3 2 UDP 16648702 10.3.0.7 58942 typ relay raddr 192.168.1.114 rport 50007
90. a=candidate:4 1 TCP-ACT 7076863 10.3.0.7 59752 typ relay raddr 192.168.1.114 rport 50031
91. a=candidate:4 2 TCP-ACT 7076350 10.3.0.7 59752 typ relay raddr 192.168.1.114 rport 50031
92. a=candidate:5 1 TCP-ACT 1684798975 192.168.1.114 50031 typ srflx raddr 192.168.1.114 rport 50031
93. a=candidate:5 2 TCP-ACT 1684798462 192.168.1.114 50031 typ srflx raddr 192.168.1.114 rport 50031
94. a=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:OY1qCCFx84fwIkrR39XpPDA2HuNdtAB+6ekK1y5a|2^31|1:1
95. a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:b28SzCBSdH7eBr13AhecN34gKh8OeCYQG6IMwxbC|2^31|1:1
96. a=crypto:3 AES\_CM\_128\_HMAC\_SHA1\_80 inline:8LUckwDobd31ORi6KGZLYf+My7wvCwftc5Nw7G79|2^31
97. a=maxptime:200
98. a=rtpmap:114 x-msrta/16000
99. a=fmtp:114 bitrate=29000
100. a=rtpmap:9 G722/8000
101. a=rtpmap:112 G7221/16000
102. a=fmtp:112 bitrate=24000
103. a=rtpmap:111 SIREN/16000
104. a=fmtp:111 bitrate=16000
105. a=rtpmap:0 PCMU/8000
106. a=rtpmap:8 PCMA/8000
107. a=rtpmap:116 AAL2-G726-32/8000
108. a=rtpmap:115 x-msrta/8000
109. a=fmtp:115 bitrate=11800
110. a=rtpmap:4 G723/8000
111. a=rtpmap:97 RED/8000
112. a=rtpmap:13 CN/8000
113. a=rtpmap:118 CN/16000
114. a=rtpmap:101 telephone-event/8000
115. a=fmtp:101 0-16
116. a=encryption:optional
117. a=x-bypassid:9CD08A01-E998-456a-AC8A-D028930E5933
118. ------=\_NextPart\_000\_0003\_01CAC1FF.366488E0--

#### Step 13: 200 Message Is Received by the Protocol Client

1. SIP/2.0 200 OK
2. Authentication-Info: TLS-DSK qop="auth", opaque="F755045D", srand="1D9666D9", snum="17", rspauth="3359c8ac2e6229b2eb9738ac707dc8c3e54f65f0", targetname="PROXY.company1", realm="SIP Communications Service", version=4
3. Via: SIP/2.0/TLS 192.168.1.114:4535;ms-received-port=4535;ms-received-cid=475300
4. FROM: "user112"<sip:user112@company1>;tag=ed04066c4a;epid=54dd5867e8
5. TO: <sip:+14258901234@company1;user=phone>;tag=201fec487e;epid=CDCFEF8F18
6. CSEQ: 1 INVITE
7. CALL-ID: e571df11a45947f1a5b90da8d957b8ae
8. RECORD-ROUTE: <sip:PROXY.company1:5061;transport=tls;opaque=state:F:Ci.R475300;lr;ms-route-sig=dcw0SbeehYaHu9dRxfcQNPNLaiGM-c5DzikYU7AfKG2hHch3QtgY3jngAA>
9. CONTACT: <sip:ms5.company1@company1;gruu;opaque=srvr:MediationServer:XzRY6u68aVipMcYVfYs0hQAA;grid=46236573d0ae4a339d83726b2bf7f7ab>;isGateway
10. CONTENT-LENGTH: 422
11. SUPPORTED: replaces
12. SUPPORTED: ms-safe-transfer
13. SUPPORTED: ms-bypass
14. SUPPORTED: ms-dialog-route-set-update
15. SUPPORTED: gruu-10
16. SUPPORTED: timer
17. SUPPORTED: 100rel
18. CONTENT-TYPE: application/gw-sdp
19. ALLOW: ACK
20. P-ASSERTED-IDENTITY: <sip:+14258901234@company1;user=phone>
21. SERVER: Mediation Server
22. Ms-Accepted-Content-ID: <3d45476919eb4c81be0c4e19c730c655>
23. ms-trunking-peer: gateway.company1.com;trunk=trunk1;User-Agent="Gateway 1.0"
24. Allow: CANCEL,BYE,INVITE,REFER,NOTIFY,PRACK,UPDATE
25. Session-Expires: 1800;refresher=uas
26. Min-SE: 90
27. v=0
28. o=Gateway 1303417666 1303417345 IN IP4 10.1.2.12
29. s=session
30. c=IN IP4 10.1.2.12
31. t=0 0
32. m=audio 6390 RTP/SAVP 0 13 101
33. c=IN IP4 10.1.2.12
34. a=rtcp:6391
35. a=x-bypass
36. a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:bN1zDJ0LC8QYNvMIdohDtGkWD/rCastpGbz5ObNo|2^31|244:1
37. a=sendrecv
38. a=rtpmap:0 PCMU/8000
39. a=rtpmap:101 telephone-event/8000
40. a=fmtp:101 0-15
41. a=ptime:20
42. a=x-mediasettings:signalboostunsupported

## ms-trunking-peer SIP Header

The messages in the following subsections illustrate the use of the **ms-trunking-peer** Session Initiation Protocol (SIP) header in messages that are sent from and received by a protocol client.

### Inbound Call

The message in the following subsection illustrates the use of the **ms-trunking-peer** SIP header for inbound calls. For a diagram of the inbound call, see the figure in section [4.6.1](#Section_27829eab0bc549cab5dafe214709ed36).

#### Step 6: INVITE Message Is Received by the Protocol Client

1. INVITE sip:192.168.1.114:4535;transport=tls;ms-opaque=acee5f6d3a;ms-received-cid=475300 SIP/2.0
2. Record-Route: <sip:PROXY.company1:5061;transport=tls;opaque=state:F:Ci.R475300:Ieh.gU65xODvwq\_j78KvdcC-dRxH7lEBsv8oECfECdswTe7QW4niMEtoTOi\_iwBgjHnKsZgY3jngAA;lr;ms-route-sig=dcrEzxvkq3iKgc2ApWyiXbCYC7NNwE-pYCMYxgjFJ3kxfHnKsZgY3jngAA>;tag=45F7A969AE33112CB9877940D7F56D40
3. Via: SIP/2.0/TLS 10.1.1.54:5061;branch=z9hG4bK1C7C8A0E.19AB9CC7A4B7C3D3;branched=TRUE;ms-internal-info="cehce-xXzqcRs3A\_ZSAwy8D4JLgyqxDKREgfIVFt6noRjHnKsZUY47CgAA"
4. Authentication-Info: TLS-DSK qop="auth", opaque="F755045D", srand="CC46B5FD", snum="26", rspauth="d6179291f72761e057a67adb7288fd256c2b1e4d", targetname="PROXY.company1", realm="SIP Communications Service", version=4
5. Max-Forwards: 69
6. Content-Length: 3161
7. Via: SIP/2.0/TLS 10.1.1.102:57350;branch=z9hG4bKe82f3c;ms-received-port=57350;ms-received-cid=475900
8. From: <sip:4259876543;phone-context=Location1@company1;user=phone>;epid=CDCFEF8F18;tag=3d965223ea
9. To: <sip:+14251234567@company1;user=phone>;epid=54dd5867e8
10. CSeq: 35 INVITE
11. Call-ID: df601b2d-e42e-4677-b921-c9dbf4e25940
12. Contact: <sip:ms5.company1@company1;gruu;opaque=srvr:MediationServer:XzRY6u68aVipMcYVfYs0hQAA;grid=bd9c42fc618147d0af4d8f84f718910b>;isGateway
13. Supported: replaces
14. Supported: ms-safe-transfer
15. Supported: ms-bypass
16. Supported: ms-dialog-route-set-update
17. Supported: timer
18. Supported: 100rel
19. Supported: gruu-10
20. User-Agent: Mediation Server
21. Content-Type: multipart/alternative; boundary=9dvaKhfhPJxCOyObvB70o0f2xfgiXN3J
22. Allow: ACK
23. ms-trunking-peer: gateway.company1.com;trunk=trunk1;User-Agent="Gateway 1.0"
24. Session-Expires: 1800
25. Min-SE: 90
26. Allow: CANCEL,BYE,INVITE,REFER,NOTIFY,PRACK,UPDATE
27. P-Asserted-Identity: <sip:+4259876543@company1;user=phone>
28. History-Info: <sip:user112@company1>;index=1
29. --9dvaKhfhPJxCOyObvB70o0f2xfgiXN3J
30. Content-Type: application/sdp
31. Content-ID: <72e03bb9-6acc-453b-ae09-4b8671344d83>
32. Content-Disposition: Session;handling=optional;ms-proxy-2007fallback
33. v=0
34. o=- 1 0 IN IP4 10.1.1.102
35. s=session
36. c=IN IP4 10.1.1.102
37. b=CT:1000000
38. t=0 0
39. m=audio 56568 RTP/AVP 0 8 115 13 118 97 101
40. c=IN IP4 10.1.1.102
41. a=rtcp:56569
42. a=candidate:wPBogiU8NLp21GV4/zj/6WviEjTkj55FxhrdRkHiZcc 1 0tKavBj1axiy4rc19atywg UDP 0.830 10.1.1.102 56568
43. a=candidate:wPBogiU8NLp21GV4/zj/6WviEjTkj55FxhrdRkHiZcc 2 0tKavBj1axiy4rc19atywg UDP 0.830 10.1.1.102 56569
44. a=candidate:bgLnsm3DP4aSPQloj2Ak1IUYeGDPsldLRetvScj5izM 1 5VdtqvYZImPIpth0Tx5Mcg TCP 0.150 10.3.0.7 59954
45. a=candidate:bgLnsm3DP4aSPQloj2Ak1IUYeGDPsldLRetvScj5izM 2 5VdtqvYZImPIpth0Tx5Mcg TCP 0.150 10.3.0.7 59954
46. a=candidate:hdj57XrOXJwib/pE8R3lzSwmfWi3trrUtRt4pmcfb5Y 1 RDbzrPzUksHqIX1Aqv0bFA UDP 0.450 10.3.0.7 55690
47. a=candidate:hdj57XrOXJwib/pE8R3lzSwmfWi3trrUtRt4pmcfb5Y 2 RDbzrPzUksHqIX1Aqv0bFA UDP 0.450 10.3.0.7 57652
48. a=candidate:JqHr0VQ3SBc1eDZ+TPZ4wktouOoWH1fag30kyuLWlFQ 1 8yhh8eM+T1Z9w0CbEkzwfA TCP 0.250 10.1.1.102 207 52082
49. a=candidate:JqHr0VQ3SBc1eDZ+TPZ4wktouOoWH1fag30kyuLWlFQ 2 8yhh8eM+T1Z9w0CbEkzwfA TCP 0.250 10.1.1.102 52082
50. a=label:main-audio
51. a=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:f6V1oCUyKIzjLEBRg46FFt7BenyVzlLNEk3EJ6T3|2^31|1:1
52. a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:P46SfAzfbRN3d06tBm787I3Pv/3j+5hOmtM1tvdv|2^31|1:1
53. a=crypto:3 AES\_CM\_128\_HMAC\_SHA1\_80 inline:tbZ34R5hvhfBSsVLMd0/uiQ7AWOCJD5Hj+Q58HQm|2^31
54. a=rtpmap:0 PCMU/8000
55. a=rtpmap:8 PCMA/8000
56. a=rtpmap:115 x-msrta/8000
57. a=fmtp:115 bitrate=11800
58. a=rtpmap:13 CN/8000
59. a=rtpmap:118 CN/16000a=rtpmap:97 RED/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-16,36--9dvaKhfhPJxCOyObvB70o0f2xfgiXN3JContent-Type: application/sdpContent-ID: <d05db498-7556-445d-86e3-bfeb36fd52e9>v=0o=- 2 0 IN IP4 10.1.1.102s=sessionc=IN IP4 10.1.1.102b=CT:1000000t=0 0m=audio 50352 RTP/AVP 0 8 115 13 118 97 101c=IN IP4 10.1.1.102a=rtcp:50353a=ice-ufrag:LxLAa=ice-pwd:3470M/yHdvxSWmMqhs+jJF2Ea=candidate:1 1 UDP 2130706431 10.1.1.102 50352 typ hosta=candidate:1 2 UDP 2130705918 10.1.1.102 50353 typ host
60. a=candidate:2 1 tcp-pass 6555135 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
61. a=candidate:2 2 tcp-pass 6555134 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
62. a=candidate:3 1 UDP 16647679 10.0.3.7 52516 typ relay raddr 10.1.1.102 rport 55636
63. a=candidate:3 2 UDP 16647678 10.0.3.7 58728 typ relay raddr 10.1.1.102 rport 55637
64. a=candidate:4 1 tcp-act 7076863 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
65. a=candidate:4 2 tcp-act 7076350 10.0.3.7 55634 typ relay raddr 10.1.1.102 rport 53970
66. a=candidate:5 1 tcp-act 1684798975 10.1.1.102 53970 typ srflx raddr 10.1.1.102 rport 53970a=candidate:5 2 tcp-act 1684798462 10.1.1.102 53970 typ srflx raddr 10.1.1.102 rport 53970a=label:main-audioa=cryptoscale:1 client AES\_CM\_128\_HMAC\_SHA1\_80 inline:f6V1oCUyKIzjLEBRg46FFt7BenyVzlLNEk3EJ6T3|2^31|1:1a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:P46SfAzfbRN3d06tBm787I3Pv/3j+5hOmtM1tvdv|2^31|1:1a=crypto:3 AES\_CM\_128\_HMAC\_SHA1\_80 inline:tbZ34R5hvhfBSsVLMd0/uiQ7AWOCJD5Hj+Q58HQm|2^31a=rtpmap:0 PCMU/8000a=rtpmap:8 PCMA/8000a=rtpmap:115 x-msrta/8000a=fmtp:115 bitrate=11800a=rtpmap:13 CN/8000a=rtpmap:118 CN/16000a=rtpmap:97 RED/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-16,36--9dvaKhfhPJxCOyObvB70o0f2xfgiXN3JContent-Type: application/gw-sdp; x-bypassid=9CD08A01-E998-456a-AC8A-D028930E5933Content-ID: <466ac626-be34-4f8d-ba0d-c7bacf53c0ac>Content-Disposition: Session;handling=optionalv=0o=Gateway 94331345 94331031 IN IP4 10.1.2.12s=sessionc=IN IP4 10.1.2.12t=0 0m=audio 6430 RTP/SAVP 0 8 4 2 3 13 101c=IN IP4 10.1.2.12a=rtcp:6431a=x-bypassid:9CD08A01-E998-456a-AC8A-D028930E5933a=crypto:1 AES\_CM\_128\_HMAC\_SHA1\_80 inline:uch9eRm5IMoOhC+jNRprVaEuvK2JN0upP2+9ciM9|2^31|129:1a=sendrecva=rtpmap:0 PCMU/8000a=rtpmap:8 PCMA/8000a=rtpmap:4 G723/8000a=fmtp:4 annexa=yesa=rtpmap:2 G726-32/8000a=rtpmap:3 GSM/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-15a=ptime:20a=x-mediasettings:signalboostunsupported
67. --9dvaKhfhPJxCOyObvB70o0f2xfgiXN3J—

### Outbound Call

The message in the following subsection illustrates the use of the **ms-trunking-peer** SIP header for outbound calls. For a diagram of the outbound call, see the figure in section [4.6.2](#Section_4c37e48677164a2f86b219388e20c775).

#### Step 13: 200 Message Is Received by the Protocol Client

1. SIP/2.0 200 OK
2. Authentication-Info: TLS-DSK qop="auth", opaque="F755045D", srand="1D9666D9", snum="17", rspauth="3359c8ac2e6229b2eb9738ac707dc8c3e54f65f0", targetname="PROXY.company1", realm="SIP Communications Service", version=4
3. Via: SIP/2.0/TLS 192.168.1.114:4535;ms-received-port=4535;ms-received-cid=475300
4. FROM: "user112"<sip:user112@company1>;tag=ed04066c4a;epid=54dd5867e8
5. TO: <sip:+14258901234@company1;user=phone>;tag=201fec487e;epid=CDCFEF8F18
6. CSEQ: 1 INVITE
7. CALL-ID: e571df11a45947f1a5b90da8d957b8ae
8. RECORD-ROUTE: <sip:PROXY.company1:5061;transport=tls;opaque=state:F:Ci.R475300;lr;ms-route-sig=dcw0SbeehYaHu9dRxfcQNPNLaiGM-c5DzikYU7AfKG2hHch3QtgY3jngAA>
9. CONTACT: <sip:ms5.company1@company1;gruu;opaque=srvr:MediationServer:XzRY6u68aVipMcYVfYs0hQAA;grid=46236573d0ae4a339d83726b2bf7f7ab>;isGateway
10. CONTENT-LENGTH: 422
11. SUPPORTED: replaces
12. SUPPORTED: ms-safe-transfer
13. SUPPORTED: ms-bypass
14. SUPPORTED: ms-dialog-route-set-update
15. SUPPORTED: gruu-10
16. SUPPORTED: timer
17. SUPPORTED: 100rel
18. CONTENT-TYPE: application/gw-sdp
19. ALLOW: ACK
20. P-ASSERTED-IDENTITY: <sip:+14258901234@company1;user=phone>
21. SERVER: Mediation Server
22. Ms-Accepted-Content-ID: <3d45476919eb4c81be0c4e19c730c655>
23. ms-trunking-peer: gateway.company1.com;trunk=trunk1;User-Agent="Gateway 1.0"
24. Allow: CANCEL,BYE,INVITE,REFER,NOTIFY,PRACK,UPDATE
25. Session-Expires: 1800;refresher=uas
26. Min-SE: 90
27. v=0
28. o=Gateway 1303417666 1303417345 IN IP4 10.1.2.12
29. s=session
30. c=IN IP4 10.1.2.12
31. t=0 0
32. m=audio 6390 RTP/SAVP 0 13 101
33. c=IN IP4 10.1.2.12
34. a=rtcp:6391
35. a=x-bypass
36. a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:bN1zDJ0LC8QYNvMIdohDtGkWD/rCastpGbz5ObNo|2^31|244:1
37. a=sendrecv
38. a=rtpmap:0 PCMU/8000
39. a=rtpmap:101 telephone-event/8000
40. a=fmtp:101 0-15
41. a=ptime:20
42. a=x-mediasettings:signalboostunsupported

## ms-mediation-generated SIP Header

The messages in the following subsections illustrate the use of the ms-mediation-generated Session Initiation Protocol (SIP) header in messages that are sent from and received by a protocol client.

### Outbound Call

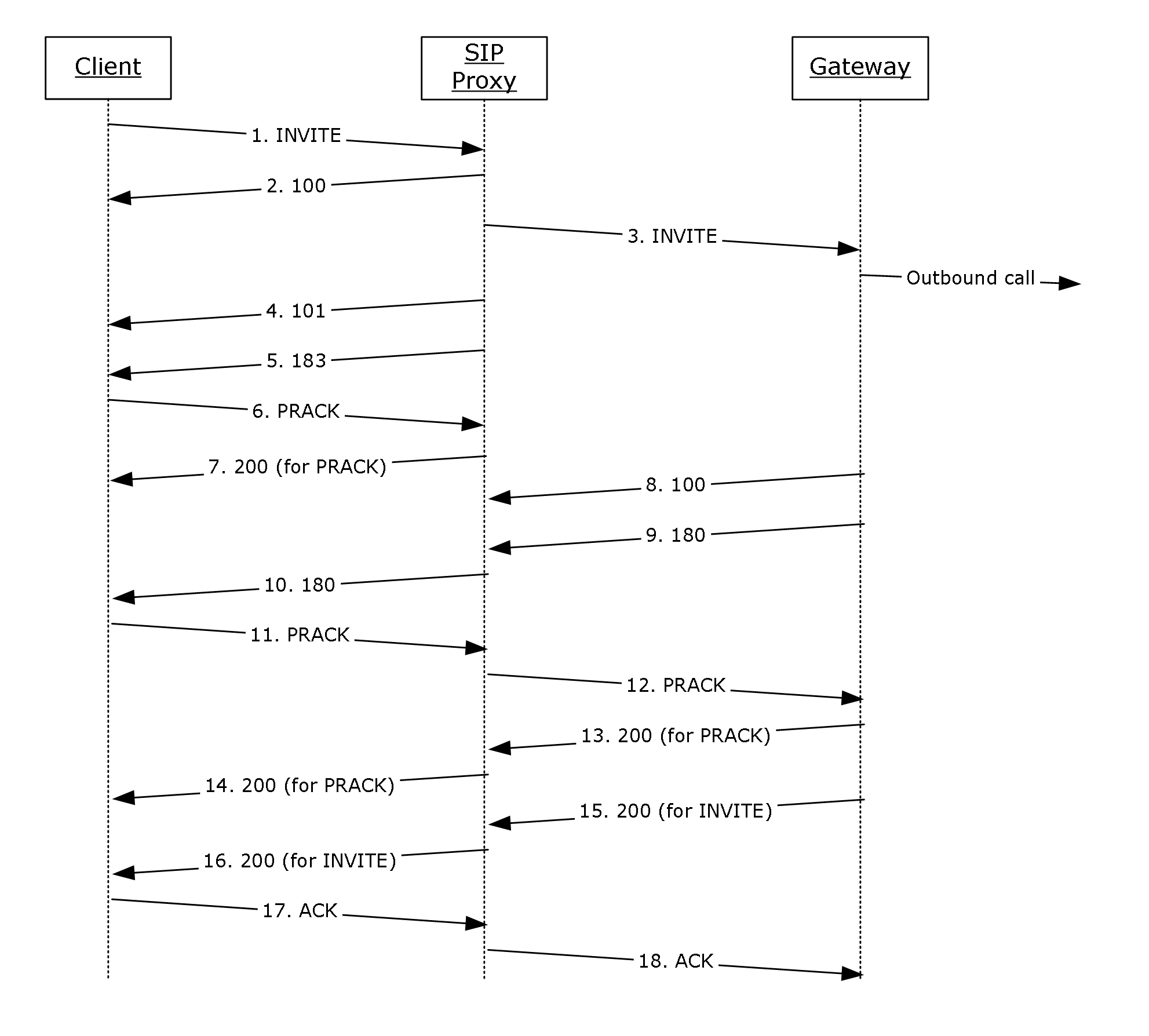


Figure 12: Outbound Call

The message in the following subsection illustrates the use of the ms-mediation-generated SIP header for outbound calls. For a diagram of the outbound call, see the preceding figure.

#### Step 5: 183 Message Is Received by the Protocol Client

SIP/2.0 183 Session Progress

Authentication-Info: NTLM rspauth="010000000000000008AC67ADF27DB686", srand="DF9D53C4", snum="103", opaque="B25450B8", qop="auth", targetname="server1.example.com", realm="SIP Communications Service"

Via: SIP/2.0/TLS 10.56.66.167:3137;ms-received-port=3137;ms-received-cid=100

FROM: "test1"<sip:test1@example.com>;tag=2b95504d65;epid=782abb8f70

TO: <sip:+15555550100@example.com;user=phone>;epid=6477F45221;tag=b5bb1243e3

CSEQ: 1 INVITE

CALL-ID: ca22890914c34bf8a7439dfe1e834420

ms-mediation-generated: yes

CONTENT-LENGTH: 740

CONTENT-TYPE: application/sdp; charset=utf-8

SERVER: RTCC/3.0.0.0 MediationServer

v=0

o=- 0 0 IN IP4 10.198.92.126

s=session

c=IN IP4 10.198.92.126

b=CT:1000

t=0 0

m=audio 60625 RTP/SAVP 111 115 8 97 101

c=IN IP4 10.198.92.126

a=rtcp:60532

a=candidate:ZHqwSbPvIZyDX24RjvIW4lryUx/QbdAiP7FyQ0yvTGo 1 Bx2Is+Wi/HJbdQKM3FIBKg UDP 0.900 10.198.92.126 60625

a=candidate:ZHqwSbPvIZyDX24RjvIW4lryUx/QbdAiP7FyQ0yvTGo 2 Bx2Is+Wi/HJbdQKM3FIBKg UDP 0.900 10.198.92.126 60532

a=crypto:2 AES\_CM\_128\_HMAC\_SHA1\_80 inline:Pb+rI3y4U1xd47P8USsgDc/znOiBIv5s0Ev2abRT|2^31|1:1

a=label:main-audio

a=encryption:rejected

a=rtpmap:111 SIREN/16000

a=fmtp:111 bitrate=16000

a=rtpmap:115 x-msrta/8000

a=fmtp:115 bitrate=11800

a=rtpmap:8 PCMA/8000

a=rtpmap:97 RED/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=ptime:20

#### Step 10: 180 Message Is Received by the Protocol Client

SIP/2.0 180 Ringing

Authentication-Info: NTLM rspauth="010000000000000008AC67ADF27DB686", srand="DF9D53C4", snum="103", opaque="B25450B8", qop="auth", targetname="server1.example.com", realm="SIP Communications Service"

Via: SIP/2.0/TLS 10.56.66.167:3137;ms-received-port=3137;ms-received-cid=100

FROM: "test1"<sip:test1@example.com>;tag=2b95504d65;epid=782abb8f70

TO: <sip:+15555550100@example.com;user=phone>;epid=6477F45221;tag=b5bb1243e3

CSEQ: 1 INVITE

CALL-ID: ca22890914c34bf8a7439dfe1e834420

CONTENT-LENGTH: 0

# Security

## Security Considerations for Implementers

None.

## Index of Security Parameters

None.

# Appendix A: Product Behavior

The information in this specification is applicable to the following Microsoft products or supplemental software. References to product versions include released service packs.

* Microsoft Office Communicator 2007
* Microsoft Office Communicator 2007 R2
* Microsoft Office Communications Server 2007
* Microsoft Office Communications Server 2007 R2
* Microsoft Lync 2010
* Microsoft Lync Server 2010
* Microsoft Lync Client 2013/Skype for Business
* Microsoft Lync Server 2013
* Microsoft Skype for Business 2016
* Microsoft Skype for Business Server 2015

Exceptions, if any, are noted below. If a service pack or Quick Fix Engineering (QFE) number appears with the product version, behavior changed in that service pack or QFE. The new behavior also applies to subsequent service packs of the product unless otherwise specified. If a product edition appears with the product version, behavior is different in that product edition.

Unless otherwise specified, any statement of optional behavior in this specification that is prescribed using the terms SHOULD or SHOULD NOT implies product behavior in accordance with the SHOULD or SHOULD NOT prescription. Unless otherwise specified, the term MAY implies that the product does not follow the prescription.

[<1> Section 2.2.9](#Appendix_A_Target_1): This header is not available in Office Communicator 2007, Office Communicator 2007 R2, Office Communications Server 2007, Office Communications Server 2007 R2, Lync 2010, or Lync Server 2010.

[<2> Section 3.2](#Appendix_A_Target_2): Office Communications Server 2007, Office Communicator 2007: This behavior is not supported.

[<3> Section 3.2](#Appendix_A_Target_3): Office Communications Server 2007, Office Communicator 2007: This behavior is not supported.

[<4> Section 3.2](#Appendix_A_Target_4): Office Communications Server 2007, Office Communicator 2007: This behavior is not supported.

[<5> Section 3.2](#Appendix_A_Target_5): Office Communications Server 2007, Office Communicator 2007: This behavior is not supported.

[<6> Section 3.2](#Appendix_A_Target_6): Office Communications Server 2007, Office Communicator 2007: This behavior is not supported.

[<7> Section 3.5](#Appendix_A_Target_7): Office Communications Server 2007, Office Communicator 2007: This behavior is not supported.

[<8> Section 3.6](#Appendix_A_Target_8): Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

[<9> Section 3.7](#Appendix_A_Target_9): Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

[<10> Section 3.8](#Appendix_A_Target_10): Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

[<11> Section 3.9](#Appendix_A_Target_11): This behavior is not supported in Office Communicator 2007, Office Communicator 2007 R2, Lync 2010, Office Communications Server 2007, Office Communications Server 2007 R2, or Lync Server 2010.

# Change Tracking

No table of changes is available. The document is either new or has had no changes since its last release.

# Index

A

Abstract data model

[anonymous phone URI](#section_78cd1dcfa99949ed9e46329893dc0103) 20

[isGateway parameter](#section_7792893f48854ea79e4db6d62b9e461b) 16

[ms-accepted-content-id header](#section_0c850188b8df410484531bf195dab75e) 21

[ms-bypass option tag](#section_19b8bdd26d12478fa7f2b1447d7a59bf) 21

[ms-call-source header](#section_00bbc9f7a7ef46d98ce994bbc59a1a33) 18

[ms-early-media option tag](#section_0a2c6c9e0453441ba3b095a39dad9bcb) 19

[ms-mediation-generated header](#section_f08a5e810b8948e0a587ae196d5a373d) 23

[ms-trunking-peer header](#section_963ebf9886b74489b4e0871f5da00df7) 22

[phone-context parameter](#section_bf504c98d2644ca2b72b644ba21db82d) 17

[Anonymous phone URI](#section_8743bdaa822447f7a6a172d084e959b2) 20

[abstract data model](#section_78cd1dcfa99949ed9e46329893dc0103) 20

[higher-layer triggered events](#section_e475d43c5e07488eba2a795f82f12a0a) 20

[initialization](#section_7b0f10ceba274b108c2812d3a497e955) 20

[local events](#section_62cd5384b37641af98b9660af6f5c75c) 20

[message processing](#section_31b874a13aec41b3a9047679bda29747) 20

[timer events](#section_34e3b93505d64217b180e2f35a7e15a4) 20

[timers](#section_7815c4c36cc741f899bd3de124e66776) 20

[Anonymous Phone URI message](#section_840750ba15ee46cea22b46087b365395) 14

[Applicability](#section_14068174021945498346ede1763454b9) 11

C

[Capability negotiation](#section_43ecd83a15c34d3a9a55561bdb1bd210) 11

[Change tracking](#section_82bfdf094823451cbe4eb2f7f76e2664) 60

Contact header

[isGateway parameter](#section_cb23f3721c144009ba48a6cb6a4e4e5c) 16

E

[Examples](#section_c735a734f31940a98144338515ca3884) 24

[isGateway SIP contact header parameter](#section_9ecf40c8efad458fb1735737865e3458) 24

[ms-accepted-content-id SIP header](#section_59255a66a6b640899e011b9d883b7d53) 45

ms-bypass option tag

[inbound call](#section_65d54a2f32714b0a96f11888d971ea5d) 38

[outbound call](#section_1a74ed83df8f481597003109591c05d6) 41

[ms-bypass SIP supported header option tag](#section_064a9102278849d1927934745f5e87c0) 38

[ms-call-source SIP header](#section_b0d1585a5f874e768add85c38339c30b) 32

[ms-early-media SIP supported header option tag](#section_c58057f6c710412d81af1a8206cb7269) 34

[ms-mediation-generated SIP header](#section_8e8f7506dc70421e9e6ab40b9d70e9fc) 54

[ms-trunking-peer SIP header](#section_2b9c0ff25c2d4a4a9268e0121e1f55d4) 51

[phone-context SIP URI parameter](#section_b2e54ea2067b40e3950fc19a60064b20) 28

Supported header

ms-bypass option tag

[inbound call](#section_65d54a2f32714b0a96f11888d971ea5d) 38

[outbound call](#section_1a74ed83df8f481597003109591c05d6) 41

F

[Fields - vendor-extensible](#section_31dbe11784be4135968d616df7044316) 12

G

[Glossary](#section_0d389928c99f4860b0d50a696a97b398) 7

H

Higher-layer triggered events

[anonymous phone URI](#section_e475d43c5e07488eba2a795f82f12a0a) 20

[isGateway parameter](#section_85364068b59f42d788ee48e75201177c) 16

[ms-accepted-content-id header](#section_caeb5c043e1f40118b1c3a13d8538ae1) 22

[ms-bypass option tag](#section_bd183546c66645ce9b4ec596adf20394) 21

[ms-call-source header](#section_985562fa97e84db8b70a319b7285c41a) 18

[ms-early-media option tag](#section_b3abfab9400b48b1999cfddbd3937661) 19

[ms-mediation-generated header](#section_6ef9bf3bb58d4c01a2818a9d2b086b5e) 23

[ms-trunking-peer header](#section_97a864729a7949ee9cf014a07ff78d6c) 22

[phone-context parameter](#section_0a993d3089254153ad3f29aa1e71d576) 17

I

[Implementer - security considerations](#section_45bd4a4ab2964a6d95233c3d52a44796) 57

inbound call examples

isGateway SIP contact header parameter

[200 message sent from UAC](#section_d98325bc6fb8468b943aa8b07a16f9cc) 25

[INVITE message received by UAC](#section_bac2210fa07740018323909df3f07739) 24

ms-accepted-content-id SIP header

[200 message sent by protocol client](#section_86ae74f9372b49e5abd0c85d5f5d7df6) 47

[INVITE message received by protocol client](#section_b2fd4b065a794b648aa47e8f0319587b) 45

ms-bypass SIP supported header option tag

[200 message sent by protocol client](#section_d4b8207cad3a4a21a8b2692058419543) 40

[INVITE message received by protocol client](#section_76fabb44542b4fbaac85d88e11eb347d) 39

ms-call-source SIP header

[200 message sent from UAC](#section_7b27c12c133e40919263657b111ddcc3) 34

[605 message sent from UAC](#section_2dcbf90553754f2db89b21474b8658b8) 33

[INVITE message received by UAC - step 2](#section_48ab9eb12deb4c6db0ce3049975aada1) 32

[INVITE message received by UAC – step 8](#section_7b9123ac733f469c91cac359ed67f21a) 33

ms-trunking-peer SIP header

[INVITE message received by protocol client](#section_9573d6ad5d1046a5833620a4aa3d20dc) 52

phone-context SIP URI parameter

[200 message sent from UAC](#section_c117f42fd83c4f3b9f40d67bade6cf76) 29

[INVITE message received by UAC](#section_07c90d62bb2746218a316a1232509938) 28

[Index of security parameters](#section_1ac030e46a594245bdeee76f21dd2c28) 57

[Informative references](#section_0f3a8e613dee4e7f82aeeebe66f31939) 9

Initialization

[anonymous phone URI](#section_7b0f10ceba274b108c2812d3a497e955) 20

[isGateway parameter](#section_a442e653335e4eeb8d540f04372abc39) 16

[ms-accepted-content-id](#section_bcd5ae46fa7245ea88bdb14239e364ab) 22

[ms-bypass option tag](#section_7f71b39511e748349a992a705cfd42fc) 21

[ms-call-source header](#section_687a3c94a45b4a898af5cb9744080163) 18

[ms-early-media option tag](#section_c1e20a75a84444c29510f8cc87cd9110) 19

ms-mediation-generated header ([section 3.9.3](#section_0c99e87674a0452a8abaaea2ef603567) 23, [section 6](#section_424edea6cd60437daeb41f04c72f7cc3) 58)

[ms-trunking-peer](#section_7f0dfea3c64e4784b47133c0dd826f22) 22

[phone-context parameter](#section_055b7827a6b4423385c275c3db8c1102) 17

[Introduction](#section_684e7fa75ce5482780b8b2def76aa2fd) 7

[isGateway](#section_cb23f3721c144009ba48a6cb6a4e4e5c) 16

[abstract data model](#section_7792893f48854ea79e4db6d62b9e461b) 16

[higher-layer triggered events](#section_85364068b59f42d788ee48e75201177c) 16

[initialization](#section_a442e653335e4eeb8d540f04372abc39) 16

[local events](#section_a0ffd09ab3664c55846cb291b4db88e4) 16

[message processing](#section_14450577785240089ac4de6b6c69bcdc) 16

[timer events](#section_8fd4bb7df8ce42a99bb310e395820bdb) 16

[timers](#section_ca614467b6084925a5063b3c2b9ac93f) 16

[isGateway message](#section_5d20970ca734403da106dba6e4852da1) 13

[isGateway SIP contact header parameter example](#section_9ecf40c8efad458fb1735737865e3458) 24

[inbound call](#section_d6bbe72cc30140c1b42ec265290f76b6) 24

[200 message sent from UAC](#section_d98325bc6fb8468b943aa8b07a16f9cc) 25

[INVITE message received by UAC](#section_bac2210fa07740018323909df3f07739) 24

[outbound call](#section_1baef6c8749045418bbf9966b8edf307) 26

[200 message received by UAC](#section_9f565c1f48ae4daa8064f8882b41da08) 27

[INVITE message sent from UAC](#section_e11a7a237d254c14a96ca974e43240e0) 26

L

Local events

[anonymous phone URI](#section_62cd5384b37641af98b9660af6f5c75c) 20

[isGateway parameter](#section_a0ffd09ab3664c55846cb291b4db88e4) 16

[ms-accepted-content-id header](#section_1f2a5c4c48684b75a789652607f813e5) 22

[ms-bypass option tag](#section_aeaa75a20eb94e46bb9af8dd54753835) 21

[ms-call-source header](#section_733553ae202c4d1f98c6175641931b1f) 19

[ms-early-media option tag](#section_4ef20dc1d2b94e618caec7a7f7df53dc) 20

[ms-trunking-peer header](#section_487beeae31d34a5a9c681dbd0e12e78a) 23

[phone-context parameter](#section_7bc9a88f6cfb44b289ab7c6ddf3bb8bf) 18

M

Message processing

[anonymous phone URI](#section_31b874a13aec41b3a9047679bda29747) 20

[isGateway parameter](#section_14450577785240089ac4de6b6c69bcdc) 16

[ms-accepted-content-id header](#section_1fd9121c354640bb8c53eb02b7201ab6) 22

[ms-bypass option tag](#section_acfd72d8732f4baf9652b67f888dce31) 21

[ms-call-source header](#section_ea765ba1d07b476db12621e2fd7f6a84) 19

[ms-early-media option tag](#section_51500079cf7340d78b887583cff086eb) 19

[ms-mediation-generated header](#section_bd5eb85351414e9f8ca6835098db4a2d) 23

[ms-trunking-peer header](#section_1845804717cf4b07877bd3f24a437caa) 22

[phone-context parameter](#section_9cb0d09dbba54d8a866ce180d25a0047) 18

[Message syntax](#section_0d11ed60988349f1934fbb7d3e2498ed) 13

Messages

[Anonymous Phone URI](#section_840750ba15ee46cea22b46087b365395) 14

[isGateway](#section_5d20970ca734403da106dba6e4852da1) 13

[ms-accepted-content-id](#section_8af44c27fad24065a1d0cfafc686b22f) 14

[ms-bypass](#section_a473fa9ac8b941aca1c6193d0f4fb1b3) 14

[ms-call-source](#section_296a4f84357046008ddaadacd2e5ae98) 14

[ms-early-media](#section_f74c2d71f1c44495af47f7620b4c493f) 14

[ms-mediation-generated](#section_8ddaf85743a349c4a321ab428e2a10da) 15

[ms-trunking-peer](#section_311b0ac4f595464fae3f5d195c03bce5) 15

[phone-context](#section_8cf63c8841b4462aab7f7ff079064a9f) 13

[syntax](#section_0d11ed60988349f1934fbb7d3e2498ed) 13

[transport](#section_0ba65e2b7bd4427d83bce4f7eb664198) 13

[ms-accepted-content-id](#section_79781c0675814622a6078eed53a7c686) 21

[abstract data model](#section_0c850188b8df410484531bf195dab75e) 21

[higher-layer triggered events](#section_caeb5c043e1f40118b1c3a13d8538ae1) 22

[initialization](#section_bcd5ae46fa7245ea88bdb14239e364ab) 22

[local events](#section_1f2a5c4c48684b75a789652607f813e5) 22

[message processing](#section_1fd9121c354640bb8c53eb02b7201ab6) 22

[timer events](#section_04048deebe494ebc927092ace949bee4) 22

[timers](#section_184d746a89144b4da4517b325d5097b7) 22

[ms-accepted-content-id message](#section_8af44c27fad24065a1d0cfafc686b22f) 14

[ms-accepted-content-id SIP header example](#section_59255a66a6b640899e011b9d883b7d53) 45

[inbound call](#section_27829eab0bc549cab5dafe214709ed36) 45

[200 message sent by protocol client](#section_86ae74f9372b49e5abd0c85d5f5d7df6) 47

[INVITE message received by protocol client](#section_b2fd4b065a794b648aa47e8f0319587b) 45

[outbound call](#section_4c37e48677164a2f86b219388e20c775) 48

[200 message received by protocol client](#section_4920f815af474411a4b78ddc1bdb033a) 51

[INVITE message sent by protocol client](#section_aa448a8d12804441bb9cb47b0e3ed3c7) 48

[ms-bypass](#section_7fcf5e4e1c264eb4b8b9cccc1f650971) 20

[abstract data model](#section_19b8bdd26d12478fa7f2b1447d7a59bf) 21

example

[inbound call](#section_65d54a2f32714b0a96f11888d971ea5d) 38

[outbound call](#section_1a74ed83df8f481597003109591c05d6) 41

[higher-layer triggered events](#section_bd183546c66645ce9b4ec596adf20394) 21

[initialization](#section_7f71b39511e748349a992a705cfd42fc) 21

[local events](#section_aeaa75a20eb94e46bb9af8dd54753835) 21

[message processing](#section_acfd72d8732f4baf9652b67f888dce31) 21

[timer events](#section_b205ebf0e6ad456fa8c36498be163daf) 21

[timers](#section_b03b8bde624e4011add989d4d0c29b00) 21

[ms-bypass message](#section_a473fa9ac8b941aca1c6193d0f4fb1b3) 14

[ms-bypass SIP supported header option tag example](#section_064a9102278849d1927934745f5e87c0) 38

[inbound call](#section_65d54a2f32714b0a96f11888d971ea5d) 38

[200 message sent by protocol client](#section_d4b8207cad3a4a21a8b2692058419543) 40

[INVITE message received by protocol client](#section_76fabb44542b4fbaac85d88e11eb347d) 39

[outbound call](#section_1a74ed83df8f481597003109591c05d6) 41

[200 OK message received by protocol client](#section_5348a6e578fc49f6a5782cf3a88f629b) 44

[INVITE message sent by protocol client](#section_8af9ca0cf0cf4301a47dffdaafe92480) 42

ms-call-source ([section 3.3](#section_29248f89c3eb4995b88db6c853ade5f8) 18, [section 3.4](#section_88069d5c3ab24db88aa616f24954e9d2) 19)

[abstract data model](#section_00bbc9f7a7ef46d98ce994bbc59a1a33) 18

[higher-layer triggered events](#section_985562fa97e84db8b70a319b7285c41a) 18

[initialization](#section_687a3c94a45b4a898af5cb9744080163) 18

[local events](#section_733553ae202c4d1f98c6175641931b1f) 19

[message processing](#section_ea765ba1d07b476db12621e2fd7f6a84) 19

[timer events](#section_ecf14b17830743c6b059595a6a916dbf) 19

[timers](#section_e41187d7abf34f528b549d7de063c2ce) 18

[ms-call-source message](#section_296a4f84357046008ddaadacd2e5ae98) 14

[ms-call-source SIP header example](#section_b0d1585a5f874e768add85c38339c30b) 32

[inbound call](#section_a9bbf2005fa44bc5b715d1143c6dc34f) 32

[605 message sent from UAC](#section_2dcbf90553754f2db89b21474b8658b8) 33

[I200 message sent from UAC](#section_7b27c12c133e40919263657b111ddcc3) 34

[INVITE message received by UAC – step 2](#section_48ab9eb12deb4c6db0ce3049975aada1) 32

[INVITE message received by UAC – step 8](#section_7b9123ac733f469c91cac359ed67f21a) 33

[outbound call](#section_7627f66c908a47ad99c1a0eb44123cd3) 34

ms-early-media

[abstract data model](#section_0a2c6c9e0453441ba3b095a39dad9bcb) 19

[higher-layer triggered events](#section_b3abfab9400b48b1999cfddbd3937661) 19

[initialization](#section_c1e20a75a84444c29510f8cc87cd9110) 19

[local events](#section_4ef20dc1d2b94e618caec7a7f7df53dc) 20

[message processing](#section_51500079cf7340d78b887583cff086eb) 19

[timer events](#section_47c37df4567c4b08992b328cd5a20a50) 20

[timers](#section_480dde53a34b4e86a73c08f00983e85b) 19

[ms-early-media message](#section_f74c2d71f1c44495af47f7620b4c493f) 14

[ms-early-media SIP supported header option tag example](#section_c58057f6c710412d81af1a8206cb7269) 34

[inbound call](#section_33aa03ab683b4f3ba339053101146e14) 34

[outbound call](#section_7428b593ae4848e397467a5889cc849f) 35

[183 message received by UAC](#section_a5505d33b36246c9995b3f15d0034164) 36

[200 message received by UAC](#section_4295706f18be43cab849a3675db76f37) 37

[INVITE sent from UAC](#section_0949d4674dfa41febccba2fea2a0edd0) 35

[ms-mediation-generated](#section_9d6bdaef45dd4178bcced29f11dfc185) 23

[abstract data model](#section_f08a5e810b8948e0a587ae196d5a373d) 23

[higher-layer triggered events](#section_6ef9bf3bb58d4c01a2818a9d2b086b5e) 23

initialization ([section 3.9.3](#section_0c99e87674a0452a8abaaea2ef603567) 23, [section 6](#section_424edea6cd60437daeb41f04c72f7cc3) 58)

[message processing](#section_bd5eb85351414e9f8ca6835098db4a2d) 23

[other local events](#section_def566e8548c4ba88cde0b4f3be60651) 23

[timer events](#section_8693439b35ed4e06816f17157a17b502) 23

[timers](#section_ba4baf35c44f45f48e994f64a3992438) 23

[ms-mediation-generated message](#section_8ddaf85743a349c4a321ab428e2a10da) 15

[ms-mediation-generated SIP header example](#section_8e8f7506dc70421e9e6ab40b9d70e9fc) 54

[outbound call](#section_9803911efb7748bd8dfd8d82e563a6b1) 55

[180 message received by protocol client](#section_ff9c8978606b4a519289adddfd15426a) 56

[183 message received by protocol client](#section_909c930ab731465c95a7a41a7cf1a67f) 55

[ms-trunking-peer](#section_8f79c3449f9e4401990b0fcba764b8a1) 22

[abstract data model](#section_963ebf9886b74489b4e0871f5da00df7) 22

[higher-layer triggered events](#section_97a864729a7949ee9cf014a07ff78d6c) 22

[initialization](#section_7f0dfea3c64e4784b47133c0dd826f22) 22

[local events](#section_487beeae31d34a5a9c681dbd0e12e78a) 23

[message processing](#section_1845804717cf4b07877bd3f24a437caa) 22

[timer events](#section_1c59e80e77ed4956b6594ea3fea49bb9) 22

[timers](#section_8f364b1b4c334abeaaa1cab9d0a807d7) 22

[ms-trunking-peer message](#section_311b0ac4f595464fae3f5d195c03bce5) 15

[ms-trunking-peer SIP header example](#section_2b9c0ff25c2d4a4a9268e0121e1f55d4) 51

[inbound call](#section_dcf376e091ac434695ab9152df44bf2c) 52

[INVITE message received by protocol client](#section_9573d6ad5d1046a5833620a4aa3d20dc) 52

[outbound call](#section_b930b10cccd34fa088ec7e2e827ae38f) 53

[200 message received by protocol client](#section_df164817d2984db38daf8696058f5ee4) 54

N

[Normative references](#section_db5f4eb9ff094c1caf805f1c7f4c72e5) 9

O

Other local events

[ms-mediation-generated header](#section_def566e8548c4ba88cde0b4f3be60651) 23

outbound call examples

isGateway SIP contact header parameter

[200 message received by UAC](#section_9f565c1f48ae4daa8064f8882b41da08) 27

[INVITE message sent from UAC](#section_e11a7a237d254c14a96ca974e43240e0) 26

ms-accepted-content-id SIP header

[200 message received by protocol client](#section_4920f815af474411a4b78ddc1bdb033a) 51

[INVITE message sent by protocol client](#section_aa448a8d12804441bb9cb47b0e3ed3c7) 48

ms-bypass SIP supported header option tag

[200 OK message received by protocol client](#section_5348a6e578fc49f6a5782cf3a88f629b) 44

[INVITE message sent by protocol client](#section_8af9ca0cf0cf4301a47dffdaafe92480) 42

ms-early-media SIP supported header option tag

[183 message received by UAC](#section_a5505d33b36246c9995b3f15d0034164) 36

[200 message received by UAC](#section_4295706f18be43cab849a3675db76f37) 37

[INVITE sent from UAC](#section_0949d4674dfa41febccba2fea2a0edd0) 35

ms-mediation-generated SIP header

[180 message received by protocol client](#section_ff9c8978606b4a519289adddfd15426a) 56

[183 message received by protocol client](#section_909c930ab731465c95a7a41a7cf1a67f) 55

ms-trunking-peer SIP header

[200 message received by protocol client](#section_df164817d2984db38daf8696058f5ee4) 54

phone-context SIP URI parameter

[200 message received by UAC](#section_38f5a242160e4ef59dae31a039172282) 31

[INVITE message sent from UAC](#section_42362d8f7132467590d0231f0a15fb35) 30

[Overview (synopsis)](#section_f9b74451b4234b2b9bda05bba6b66a8f) 10

P

[Parameters - security index](#section_1ac030e46a594245bdeee76f21dd2c28) 57

[phone-context](#section_353bd27a65a04376b5d97a255c69d19a) 16

[abstract data model](#section_bf504c98d2644ca2b72b644ba21db82d) 17

[higher-layer triggered events](#section_0a993d3089254153ad3f29aa1e71d576) 17

[initialization](#section_055b7827a6b4423385c275c3db8c1102) 17

[local events](#section_7bc9a88f6cfb44b289ab7c6ddf3bb8bf) 18

[message processing](#section_9cb0d09dbba54d8a866ce180d25a0047) 18

[timer events](#section_3a15bab93828480185d1dd334eb43f38) 18

[timers](#section_89fe75807cb849d688e1605677880b3b) 17

[phone-context message](#section_8cf63c8841b4462aab7f7ff079064a9f) 13

[phone-context SIP URI parameter example](#section_b2e54ea2067b40e3950fc19a60064b20) 28

[inbound call](#section_751fe6b3184846a2868e6506fa099c88) 28

[200 message sent from UAC](#section_c117f42fd83c4f3b9f40d67bade6cf76) 29

[INVITE message received by UAC](#section_07c90d62bb2746218a316a1232509938) 28

[outbound call](#section_a94e714c0a8d42bf9d9aae94255098c7) 30

[200 message received by UAC](#section_38f5a242160e4ef59dae31a039172282) 31

[INVITE message sent from UAC](#section_42362d8f7132467590d0231f0a15fb35) 30

[Preconditions](#section_b28c4de6e2fc4cddbc248a883ec78051) 11

[Prerequisites](#section_b28c4de6e2fc4cddbc248a883ec78051) 11

[Product behavior](#section_424edea6cd60437daeb41f04c72f7cc3) 58

R

[References](#section_3ead3f786a3941498c72353ed92afd34) 9

[informative](#section_0f3a8e613dee4e7f82aeeebe66f31939) 9

[normative](#section_db5f4eb9ff094c1caf805f1c7f4c72e5) 9

[Relationship to other protocols](#section_5185499610f749cdaddafcbcd87089bd) 11

S

Security

[implementer considerations](#section_45bd4a4ab2964a6d95233c3d52a44796) 57

[parameter index](#section_1ac030e46a594245bdeee76f21dd2c28) 57

SIP headers

[ms-accepted-content-id](#section_79781c0675814622a6078eed53a7c686) 21

[ms-bypass option tag](#section_7fcf5e4e1c264eb4b8b9cccc1f650971) 20

[ms-call-source](#section_29248f89c3eb4995b88db6c853ade5f8) 18

[ms-early-media option tag](#section_88069d5c3ab24db88aa616f24954e9d2) 19

[ms-mediation-generated](#section_9d6bdaef45dd4178bcced29f11dfc185) 23

[ms-trunking-peer](#section_8f79c3449f9e4401990b0fcba764b8a1) 22

SIP URI

[phone-context parameter](#section_353bd27a65a04376b5d97a255c69d19a) 16

[Standards assignments](#section_68738de54c674ace99380d572c32c29b) 12

Supported header

ms-bypass option tag

example

[inbound call](#section_65d54a2f32714b0a96f11888d971ea5d) 38

[outbound call](#section_1a74ed83df8f481597003109591c05d6) 41

T

Timer events

[anonymous phone URI](#section_34e3b93505d64217b180e2f35a7e15a4) 20

[isGateway parameter](#section_8fd4bb7df8ce42a99bb310e395820bdb) 16

[ms-accepted-content-id header](#section_04048deebe494ebc927092ace949bee4) 22

[ms-bypass option tag](#section_b205ebf0e6ad456fa8c36498be163daf) 21

[ms-call-source header](#section_ecf14b17830743c6b059595a6a916dbf) 19

[ms-early-media option tag](#section_47c37df4567c4b08992b328cd5a20a50) 20

[ms-mediation-generated header](#section_8693439b35ed4e06816f17157a17b502) 23

[ms-trunking-peer header](#section_1c59e80e77ed4956b6594ea3fea49bb9) 22

[phone-context parameter](#section_3a15bab93828480185d1dd334eb43f38) 18

Timers

[anonymous phone URI](#section_7815c4c36cc741f899bd3de124e66776) 20

[isGateway parameter](#section_ca614467b6084925a5063b3c2b9ac93f) 16

[ms-accepted-content-id header](#section_184d746a89144b4da4517b325d5097b7) 22

[ms-bypass option tag](#section_b03b8bde624e4011add989d4d0c29b00) 21

[ms-call-source header](#section_e41187d7abf34f528b549d7de063c2ce) 18

[ms-early-media option tag](#section_480dde53a34b4e86a73c08f00983e85b) 19

[ms-mediation-generated header](#section_ba4baf35c44f45f48e994f64a3992438) 23

[ms-trunking-peer header](#section_8f364b1b4c334abeaaa1cab9d0a807d7) 22

[phone-context parameter](#section_89fe75807cb849d688e1605677880b3b) 17

[Tracking changes](#section_82bfdf094823451cbe4eb2f7f76e2664) 60

[Transport](#section_0ba65e2b7bd4427d83bce4f7eb664198) 13

V

[Vendor-extensible fields](#section_31dbe11784be4135968d616df7044316) 12

[Versioning](#section_43ecd83a15c34d3a9a55561bdb1bd210) 11