[MS-EUMR]: Routing to Exchange Unified Messaging Extensions

documentation for past or current releases as specifically noted in the document, as applicable; it is preliminary documentation for the pre-release (beta) versions. Microsoft will release final documentation in connection with the commercial release of the updated or new version of this technology. As the documentation may change between this preliminary version and the final version of this technology, there are risks in relying on preliminary documentation. To the extent that you incur additional development obligations or any other costs as a result of relying on this preliminary documentation, you do so at your own risk.

Revision Summary

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03/13/2009	1.02		Edited the technical content
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1 Introduction

This document specifies the Routing to Exchange Unified Messaging Extensions, which consist of proprietary application extensions for routing calls to Exchange Unified Messaging (UM) voice mail and generating user notifications based on the **Session Initiation Protocol (SIP)** INFO method.

Sections 1.8, 2, and 3 of this specification are normative and contain RFC 2119 language. Sections 1.5 and 1.9 are also normative but cannot contain RFC 2119 language. All other sections and examples in this specification are informative.

1.1 Glossary

The following terms are defined in [MS-GLOS]:

Augmented Backus-Naur Form (ABNF)
authentication
Coordinated Universal Time (UTC)
fully qualified domain name (FQDN)
server
XML

The following terms are defined in [MS-OFCGLOS]:

200 OK
address-of-record
Audio/Video Edge Server (A/V Edge Server)
caller
dial plan
dialog
endpoint
Globally Routable User Agent URI (GRUU)
in-band provisioning
INVITE
long-term credentials
Multipurpose Internet Mail Extensions (MIME)
proxy
QoE Monitoring Server
Quality of Experience (QoE)
Session Description Protocol (SDP)
Session Initiation Protocol (SIP)
token
Uniform Resource Identifier (URI)
user agent client (UAC)
XML schema

The following terms are specific to this document:

MAY, SHOULD, MUST, SHOULD NOT, MUST NOT: These terms (in all caps) are used as described in [RFC2119]. All statements of optional behavior use either MAY, SHOULD, or SHOULD NOT.

1.2 References

References to Microsoft Open Specification documents do not include a publishing year because links are to the latest version of the documents, which are updated frequently. References to other documents include a publishing year when one is available.

1.2.1 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. We will assist you in finding the relevant information. Please check the archive site, http://msdn2.microsoft.com/en-us/library/E4BD6494-06AD-4aed-9823-445E921C9624, as an additional source.

[IETFDRAFT-DIISIP-08] Levy, S. and Yang, J. R., "Diversion Indication in Session Initiation Protocol (SIP)", draft-levey-sip-diversion-08, February 2005, http://tools.ietf.org/id/draft-levy-sip-diversion-08.txt

[MS-AVEDGEA] Microsoft Corporation, "Audio Video Edge Authentication Protocol Specification".

[MS-QoE] Microsoft Corporation, "Quality of Experience Monitoring Server Protocol Specification".

[MS-SIPRE] Microsoft Corporation, "Session Initiation Protocol (SIP) Routing Extensions".

[MS-SIPREGE] Microsoft Corporation, "Session Initiation Protocol (SIP) Registration Extensions".

[MS-TURN] Microsoft Corporation, "Traversal Using Relay NAT (TURN) Extensions".

[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

[RFC2327] Handley, M., and Jacobson, V., "SDP: Session Description Protocol", RFC 2327, April 1998, http://www.ietf.org/rfc/rfc2327.txt

[RFC2976] Donovan, S., "The SIP INFO Method", RFC 2976, October 2000, http://www.rfc-editor.org/rfc/rfc2976.txt

[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

[RFC3892] Sparks, R., "The Session Initiation Protocol (SIP) Referred-By Mechanism", RFC 3892, September 2004, http://www.rfc-editor.org/rfc/rfc3892.txt

1.2.2 Informative References

[MS-GLOS] Microsoft Corporation, "Windows Protocols Master Glossary".

[MS-OFCGLOS] Microsoft Corporation, "Microsoft Office Master Glossary".

[RFC5234] Crocker, D., Ed., and Overell, P., "Augmented BNF for Syntax Specifications: ABNF", STD 68, RFC 5234, January 2008, http://www.rfc-editor.org/rfc/rfc5234.txt

[XML10] World Wide Web Consortium, "Extensible Markup Language (XML) 1.0 (Third Edition)", February 2004, http://www.w3.org/TR/REC-xml

[XMLNS] Bray, T., Hollander, D., Layman, A., et al., Eds., "Namespaces in XML 1.0 (Third Edition)", W3C Recommendation, December 2009, http://www.w3.org/TR/2009/REC-xml-names-20091208/

[XMLSCHEMA0] Fallside, D., Ed. and Walmsley, P., Ed., "XML Schema Part 0: Primer, Second Edition", W3C Recommendation, October 2004, http://www.w3.org/TR/2004/REC-xmlschema-0-20041028/

1.3 Protocol Overview (Synopsis)

This protocol specifies Session Initiation Protocol (SIP) extensions that are used to route calls to Exchange Unified Messaging (UM) and to generate user notification email messages on call events.

These voice mail routing extensions have been designed to route calls to Exchange UM servers based on SIP. They provide mechanisms to identify the voice mail box for deposit and provide means for communicating the **Audio/Video Edge Server (A/V Edge Server)** information that can be used by UM servers while talking to external **callers**. These extensions are described in detail in section 3.1.

This protocol provides a way for a SIP **proxy** that implements the protocol described in [MS-SIPRE] to use UM servers to generate call notification email messages if one of the following events occurs during the processing of a call:

- The user missed the call because the caller hung up before the call could be routed to voice mail or otherwise answered.
- The call was answered by one of the targets specified in the user's routing script preamble. Routing script preamble and call processing extensions are described in [MS-SIPRE].
- The proxy was unable to route to one of the targets specified in the routing script preamble because of the lack of authorization.

These extensions are described in section 3.2.

1.4 Relationship to Other Protocols

This protocol defines an **XML schema** that supports various extensions specified in this protocol. For more information about **XML**, see [XML10], [XMLNS], and [XMLSCHEMA0].

The voice mail routing extensions specified in this protocol augment the routing extensions for routing calls based on the Routing Script Preamble as described in [MS-SIPRE].

This protocol incorporates the Session Initiation Protocol (SIP) protocols.

1.5 Prerequisites/Preconditions

This protocol assumes that the Session Initiation Protocol (SIP) server implements the Session Initiation Protocol Routing Extensions as described in [MS-SIPRE]. This protocol also assumes that the server has knowledge of each voice mail **dial plan** in the enterprise, the Unified Messaging (UM) servers associated with each dial plan, and which dial plan each user is associated with.

The prerequisites for SIP are also applicable for this protocol.

1.6 Applicability Statement

This protocol is applicable when both the SIP protocol **server** and voicemail server support SIP and intend to use the enhancements offered by this protocol.

1.7 Versioning and Capability Negotiation

None.

1.8 Vendor-Extensible Fields

None.

1.9 Standards Assignments

None.

2 Messages

2.1 Transport

This protocol does not introduce a new transport to exchange messages, but is capable of being used with any transport used by Session Initiation Protocol (SIP).

2.2 Message Syntax

2.2.1 Ms-Mras-Address Header Field

This protocol defines a new header field called **Ms-Mras-Address**. The **Ms-Mras-Address** header field identifies the Audio/Video Edge Server (A/V Edge Server) that can be used by the voice mail server when the caller **endpoint (5)** is external to the enterprise.

The following example is the **Augmented Backus-Naur Form (ABNF)**, as defined in [RFC5234], for the **Ms-Mras-Address** header field:

```
Ms-Mras-Address = "Ms-Mras-Address" HCOLON LWS LAQOUT SIP-URI RAQUOT
```

The voice mail server can use the A/V Edge Server information to obtain **authentication (2) tokens**, as specified in [MS-AVEDGEA].<1>

2.2.2 Request-URI Header Field

This protocol defines certain restrictions on the **Request-URI** field value while routing an **INVITE** to a Unified Messaging (UM) server for a user. The **user** portion of the **URI** MUST be the name of the dial plan to which the user belongs. The **host** portion MUST be the UM server **fully qualified domain name (FQDN) (1)** and the **port** portion MUST be "5061". The URI MUST have a **transport parameter** with a value "tls". The URI MUST have a **maddr** parameter with the UM server FQDN (1) as its value. URI parameters are specified in [RFC3261] section 19.1.1.

2.2.3 User Notification Extensions

This section describes the User Notification Extensions used to notify the Unified Messaging (UM) server about user call events. The UM servers use this information to generate the following types of email messages:

- Missed call email messages
- Call Answered email messages
- Call Forbidden email messages

The User Notification events are delivered as **application/ms-rtc-usernotification+xml** content in the body of Session Initiation Protocol (SIP) INFO messages, as specified in [RFC2976]. The complete schema is defined in section 6. Each notification message is generated as a result of some notification-worthy call event occurring while processing an audio INVITE request.

2.2.3.1 User Notification Description Element

Each INFO message contains a description of one User Notification Event. The **User** element MUST be a valid Session Initiation Protocol (SIP) URI that identifies the user that will receive the email notification. If the **EumProxyAddress** element is present, it MUST be the address string used by

Exchange Unified Messaging (UM) to uniquely identify the user. The **Time** element MUST be a string that corresponds to the time the event occurred in **Coordinated Universal Time (UTC)**.

The following example is a **user-notification-type** element:

```
<xs:complexType name="user-notification-type">
  <xs:sequence>
      <xs:element name="User" type="xs:string" />
      <xs:element name="EumProxyAddress" type="xs:string" minOccurs="0" />
      <xs:element name="Time" type="xs:string" />
      <xs:element name="Template" type="xs:string" fixed="RtcDefault" />
      <xs:element name="Event" type="event-type" />
  </xs:sequence>
</xs:complexType>
```

2.2.3.2 User Event Description Element

Each **Event** element MUST include **CallId** and **From** elements. The **CallId** element MUST contain the **Call-ID** header field value of the INVITE request associated with the event. The **From** element MUST contain the **From** header field value of the INVITE request associated with the event. If the **Subject** element is present, it MUST contain the **Subject** header field value in the corresponding INVITE. If the **Priority** element is present, it MUST contain the **Priority** header field value in the corresponding INVITE. If the **ConversationID** element is present, it MUST contain the **Ms-Conversation-ID** header field value in the corresponding INVITE. If the **ReferredBy** element is present, it MUST contain the **referrer-uri** of the **Referred-By** header, as specified in [RFC3892], in the corresponding INVITE. If the **Target** element is present, it MUST contain the **Request-URI** used when routing the INVITE associated with the event.

If the **AnsweredBy** element is present, it MUST contain the URI present in the **P-Asserted-Identity** header found in the response to the INVITE.

The following example is an **event-type** element:

```
<xs:complexType name="event-type">
  <xs:sequence>
      <xs:element name="CallId" type="xs:string" />
      <xs:element name="From" type="xs:string" />
      <xs:element name="Subject" type="xs:string" minOccurs="0" />
      <xs:element name="Priority" type="xs:string" minOccurs="0" />
      <xs:element name="ConversationID" type="xs:string" minOccurs="0" />
      <xs:element name="ReferredBy" type="xs:string" minOccurs="0" />
      <xs:element name="Target" type="xs:string" minOccurs="0" />
      <xs:element name="TargetClass" type="target-class-type" minOccurs="0" />
      <xs:element name="AnsweredBy" type="xs:string" minOccurs="0" />
      <xs:element name="MissedReason" type="missed-reason-type" minOccurs="0" />
  </xs:sequence>
  <xs:attribute name="type" type="event-type-attribute-type" use="required" />
</xs:complexType>
```

2.2.3.3 Event Type Attribute

Each **Event** element MUST contain a **type** attribute. If the **type** attribute has the value "answered", the **Target**, **TargetClass**, and **AnsweredBy** elements MUST be present in the parent **Event**. If the **type** attribute has the value "forbidden", the **Target** and **TargetClass** elements MUST be present in the parent **Event**.

The following example is an **event-type-attribute-type** attribute:

```
<xs:simpleType name = "event-type-attribute-type">
  <xs:restriction base="xs:string">
      <xs:enumeration value="missed"/>
      <xs:enumeration value="answered"/>
      <xs:enumeration value="forbidden"/>
  </xs:restriction>
</xs:simpleType>
```

2.2.4 User Notification INVITE Request

2.2.4.1 From Header Field

This protocol specifies additional restrictions on the **From** header field syntax, as specified in [RFC3261], for the INVITE **dialog** established to send user event notifications to Unified Messaging (UM) servers.

The original Augmented Backus-Naur Form (ABNF), as defined in [RFC5234], for **from-spec**, as specified in [RFC3261] section 25, is replaced with the following:

```
from-spec = spl-user-identity *(SEMI from-param)
spl-user-identity = LAQUOT "sip" HCOLON "A410AA79-D874-4e56-9B46-709BDD0EB850" RAQUOT
```

2.2.4.2 Request-URI Header Field

This protocol specifies additional restrictions on the **Request-URI** header field syntax, as specified in [RFC3261], for the INVITE dialog established to send user event notifications to Unified Messaging (UM) servers. The **user** portion of the **Request-URI** header field MUST be empty. The **host** portion MUST be the fully qualified domain name (FQDN) (1) of the UM server and the **transport parameter** MUST have the value "tls". In addition, the **Request-URI** MUST have an **opaque parameter** with a **value** of "app:rtcevent". URI parameters are specified in [RFC3261] section 19.1.1.

2.2.4.3 SDP Content

The INVITE dialog that is established to send user event notifications to Unified Messaging (UM) servers MUST contain a **Session Description Protocol (SDP)** body, as specified in [RFC2327].

The SDP body MUST include a **media-field** with the following values:

- The value of **media** MUST be "application".
- The value of **port** MUST be "9".
- The value of **proto** MUST be "SIP".
- Only one **fmt** is present, and its value is "*".

The SDP body MUST also have the following attribute values:

- An **attribute** line with "sendonly" as the value of **att-field.**

- An **attribute** line with "accept-types" as the value of **att-field** and "application/ms-rtc-usernotification+xml" as the value of **att-value**.

2.2.5 User Notification INVITE Response

The **200 OK** response from the Unified Messaging (UM) server MUST also contain a Session Description Protocol (SDP) body.

The SDP body MUST include a **media-field** with the following values:

- The value of **media** MUST be "application".
- The value of **port** MUST be "9".
- The value of **proto** MUST be "SIP".
- Only one **fmt** is present, and its value is "*".

The SDP body MUST also have the following attribute values:

- An **attribute** line with "recvonly" as the value of **att-field**.
- An **attribute** line with "accept-types" as the value of **att-field** and "application/ms-rtc-usernotification+xml" as the value of **att-value**.
- An **attribute** line with "ms-rtc-accept-eventtemplates" as the value of **att-field** and "RtcDefault" as the value of **att-value**.

2.2.6 Option Tag Extensions

This protocol defines one new option tag for use in the **Supported** header field. The new tag extends the set of option tags as specified in [RFC3261].

Ms-Fe: This option tag is for supporting the routing extensions described in this protocol. The inclusion of this tag in the **Supported** header field of the INVITE request routed to the voice mail server indicates that the Session Initiation Protocol (SIP) proxy adheres to the specifications of this protocol.<2>

3 Protocol Details

3.1 Extensions for Routing to Exchange Unified Messaging Details

An incoming audio INVITE is processed based on a routing script preamble published by the user, as specified in [MS-SIPRE]. Under various circumstances, the proxy can decide to route the call to the user's voice mail, as specified in [MS-SIPRE] section 3.8.5. The extensions for routing to Exchange Unified Messaging (UM) are also applicable when the **Request-URI** matches the Augmented Backus-Naur Form (ABNF), as defined in [RFC5234], rules for **voice-mail-gruu** syntax, as specified in [MS-SIPRE] section 2.2.2. This protocol is not applicable if the user is not UM-enabled.

This protocol provides a mechanism for routing such calls to Exchange UM if the user is UM-enabled. This protocol also defines a mechanism for providing the UM server with the Session Initiation Protocol (SIP) URI of anAudio/Video Edge Server (A/V Edge Server), if available. The UM server uses this URI to obtain authentication (2) tokens when needed, as specified in [MS-AVEDGEA].

3.1.1 Abstract Data Model

This section describes a conceptual model of possible data organization that an implementation maintains to participate in this protocol. The described organization is provided to facilitate the explanation of how the protocol behaves. This document does not mandate that implementations adhere to this model as long as their external behavior is consistent with that described in this document.

A Session Initiation Protocol (SIP) proxy compliant with this protocol maintains a database of all the dial plans in the enterprise and all the Unified Messaging (UM) servers in each dial plan. In addition, the proxy also maintains a mapping of all UM-enabled users and their corresponding dial plans. In addition, if any Audio/Video Edge Servers (A/V Edge Servers) are present in the deployment, the SIP proxy stores the **Globally Routable User Agent URI (GRUU)** of the A/V Edge Server in memory.

3.1.2 Timers

A **Unified Messaging server** timer starts when the INVITE is routed to a Unified Messaging (UM) server. The amount of time to wait MUST be less than 180 seconds. The recommended wait time is 5 seconds.

3.1.3 Initialization

None.

3.1.4 Higher-Layer Triggered Events

None.

3.1.5 Message Processing Events and Sequencing Rules

When the Session Initiation Protocol (SIP) proxy routes an INVITE to voice mail based on rules as specified in [MS-SIPRE] or when an INVITE with a **voice-mail-gruu** (as specified in [MS-SIPRE] section 2.2.2) arrives, and the user is enabled for Exchange Unified Messaging (UM), the proxy MUST process the request as follows:

1. If the INVITE already contains any **Diversion** headers, as specified in [IETFDRAFT-DIISIP-08], these headers MUST be removed before forwarding the request to Exchange UM.

2. If the **user** and **host** portions of the **Request-URI** field are not the same as those of the **From** header field, the proxy MUST add a **Diversion** header with **name-addr** equal to the SIP URI in the **Request-URI** field value without any **uri-parameters** or **headers**, as specified in [RFC3261].

3. The UM server can request the provisioning information from the SIP proxy to detect the Globally Routable User Agent URI (GRUU) of the Audio/Video Edge Server (A/V Edge Server).<3> The **QoE Monitoring Server** protocol and the message format for **in-band provisioning** requests to obtain provisioning information are specified in [MS-SIPREGE] section 3.3.

4. If an A/V Edge Server is configured, the proxy MUST add an **Ms-Mras-Address** header with the value of the A/V Edge Server GRUU, as specified in [MS-SIPRE]. The **Ms-Mras-address** header in the incoming INVITE can be the secondary source to detect the A/V Edge Server GRUU in the absence of provisioning information.<4>

5. The proxy SHOULD<5> include the **Ms-Fe** option tag in the **Supported** header field of the request if one is not already present.

6. The proxy MUST decide on an ordering of the UM servers in the user's dial plan and route the INVITE to the first UM server. The **Request-URI** MUST be constructed as specified in section 2.2.2.

7. If there are multiple UM servers in the user's dial plan with different versions, the proxy must restrict the ordering to UM servers with the highest version. Similarly, if at least one UM server in the user's dial plan is a front end or front end array, then the proxy must use only the front ends or arrays when determining the ordering<6>.

8. The proxy MUST start the **Unified Messaging server** timer.

9. Outgoing messages from Exchange UM can include an **Ms-fe** header parameter containing its specific fully qualified domain name (FQDN) (1) value. The proxy MUST be able to handle the contact header with the **Ms-fe** parameter. The syntax and handling for the **Ms-Fe** parameter is specified in [MS-SIPRE].

3.1.5.1 Interacting with an Audio/Video Edge Server

The Unified Messaging (UM) server can request **long-term credentials** from the Audio/Video Edge Server (A/V Edge Server).<7> The protocol and message format for requesting an authentication (2) token is specified in [MS-AVEDGEA] section 2. When the caller endpoint (5) is external to the enterprise, the UM server can use the long-term credentials obtained from the A/V Edge Server to communicate with the A/V Edge Server, as specified in [MS-TURN] section 3.

3.1.5.2 Publishing a Quality of Experience Report

If a QoE Monitoring Server is detected from provisioning data, the Unified Messaging (UM) server can publish a **Quality of Experience (QoE)** report at the end of every audio call to the QoE Monitoring Server.<8> The format and protocol for publishing a QoE report is specified in [MS-QoE] section 3.

3.1.5.3 Processing a 302 Response

The Unified Messaging (UM) server can return a 302 Redirect response to the INVITE.<9> The Session Initiation Protocol (SIP) proxy implementing this protocol MUST process the 302 response without sending it back to the caller. The INVITE MUST be redirected to the UM server identified in the **contact** header field value of the 302 response if the following conditions are met:

- The 302 response came from the UM server that was selected.
- There is only one **contact** listed in the 302 response.
- The **contact** header field value identifies a UM server that is in the user's dial plan.
- No more than four previous 302 responses were processed in relation to routing this INVITE to this particular UM server.

If any one of the preceding conditions is not met, the proxy MUST retry the next UM server, as specified in section 3.1.5.7.

3.1.5.4 Generating a 101 Progress Report

Any time the INVITE is routed to a new Unified Messaging (UM) server, the Session Initiation Protocol (SIP) proxy SHOULD generate a 101 Progress response and send it back to the calling **user agent client (UAC)**.

3.1.5.5 Processing a 415 Response

If a Unified Messaging (UM) server returns a 415 response and the INVITE has multipart **Multipurpose Internet Mail Extensions (MIME)** content, the proxy MUST clear the UM server's ordering list and process the 415 response as specified in [MS-SIPRE] section 3.9.5.5. If a fresh INVITE is sent as a result of this processing, the proxy MUST redo the steps listed in section 3.1.5.

3.1.5.6 Processing Other Responses

Any response with a status code from 100 through 299, inclusive, MUST be processed as specified in [RFC3261] section 13.2.2.

If the Unified Messaging (UM) server returns any other response, the proxy MUST retry the request with the next server, as specified in section 3.1.5.7.

3.1.5.7 Retrying a Request

If the Unified Messaging (UM) server returns any failure response, as described in section 3.1.5.6, or if the **Unified Messaging server** timer expires, the next UM server in the list MUST be selected and the request MUST be routed to that server.

When selecting the next UM server, the UM server version and CAFÉ restrictions specified in step 7 of section 3.1.5 MUST be followed<10>.

If all servers in the list have been attempted, the call SHOULD be terminated with a response code 480.<11>

If the request was routed to a new UM server, the **Unified Messaging server** timer MUST be restarted.

3.1.6 Timer Events

When the **Unified Messaging server** timer expires, the request MUST be routed to the next Unified Messaging (UM) server in the list, as specified in section 3.1.5.7.

3.1.7 Other Local Events

None.

3.2 User Notification Extensions Details

This protocol specifies a mechanism for sending call event email notifications to users through Exchange Unified Messaging (UM). The event information is sent to the UM server by using the Session Initiation Protocol (SIP) INFO method, as specified in [RFC2976], over an already established SIP INVITE dialog. The body of the INFO message MUST adhere to the syntax specified in section 2.2.3. Call event notifications MUST NOT be sent for users who are not UM-enabled. The following events can trigger the User Notification extensions<12>:

- **Missed call:** The caller hung up before the call was answered or routed to voice mail.
- **Team/Delegate pick up:** The call was answered by a team member or delegate. Team and delegate ringing is specified in [MS-SIPRE].
- **Call Forwarded:** The call was answered by the target defined in the user's routing preamble as the forwarding destination.
- **Call Forwarding Failed:** An attempt to forward the call was made, but it failed because the configured destination was invalid or not permitted by administrative policy.

3.2.1 Abstract Data Model

This section describes a conceptual model of possible data organization that an implementation maintains to participate in this protocol. The described organization is provided to facilitate the explanation of how the protocol behaves. This document does not mandate that implementations adhere to this model as long as their external behavior is consistent with that described in this document.

A Session Initiation Protocol (SIP) proxy that implements this protocol MUST maintain a database of all dial plans in the enterprise and all the Unified Messaging (UM) servers in each dial plan. When a call event occurs that requires an email notification to be sent to the user, the proxy MUST establish a SIP INVITE dialog with one of the UM servers in the user's dial plan if one does not already exist. The information about the event is then sent to the UM server using the INFO method on this dialog.

3.2.2 Timers

A **User Notification Inactivity** timer MUST start when an INVITE dialog is established with a Unified Messaging (UM) server. The wait time for this timer is 10 minutes.

3.2.3 Initialization

A Session Initiation Protocol (SIP) proxy implementing this protocol MUST establish a SIP INVITE dialog with one of the Unified Messaging (UM) servers in the user's dial plan before sending the event information. The proxy can randomly pick a UM server from the list<13>. The proxy MUST establish a Mutual-TLS (MTLS) connection with the selected UM server and construct the INVITE as specified in section 2.2.4. After the INVITE dialog is established, the **User Notification Inactivity** timer MUST be started.

If a dialog already exists with a UM server in the user's dial plan, that dialog MUST be reused.

3.2.4 Higher-Layer Triggered Events

3.2.4.1 Missed Call Event

When processing an audio call targeted at a user, as specified in [MS-SIPRE], if the caller hangs up before the call was answered or routed to voice mail, information about the missed call event MUST be sent to the Unified Messaging (UM) server if the user is UM-enabled. A missed call event MUST also be raised if all destinations, as listed in the routing script preamble, returned a negative response code unless the INVITE has an **Ms-Sensitivity** header with the value "private-no-diversion", as specified in [MS-SIPRE].

3.2.4.1.1 SIP Proxy Operation

When a missed call event occurs for a user who is enabled for Exchange Unified Messaging (UM) on a Session Initiation Protocol (SIP) proxy compliant with this protocol, the proxy MUST send information about the missed call event to a UM server on the user's dial plan by sending an INFO request on an already existing INVITE dialog. If a dialog does not already exist, the proxy MUST establish one, as described in section 3.2.3.

The INFO request MUST have a **Content-Type** header with "application/ms-rtc-usernotification+xml" as its value. The body of the request MUST conform to the User Notification format, as specified in section 2.2.3. If the caller hung up the call, the **MissedReason** element MUST contain "CallerReleased". If all destinations returned a negative response code, the **MissedReason** element MUST contain "Declined".

The **Subject**, **Priority**, **ConversationID**, and **ReferredBy** elements MUST be present if their corresponding headers are present in the INVITE associated with the event.

The **type** attribute of the **Event** element MUST be set to "missed".

After the INFO request has been sent, the **User Notification Inactivity** timer for that UM server MUST be restarted.

3.2.4.2 Call Answered Event

When processing an audio call targeted at a user, as specified in [MS-SIPRE], if the call was answered by a destination other than the registered endpoint (5) associated with the **address-of-record** in the **Request-URI** header field or the simultaneous ring destination, information about the call answered event MUST be sent to the Unified Messaging (UM) server if the user is UM-enabled. A call is considered to be answered if one of the destinations responded with a 200, 303, or 605 response code.

3.2.4.2.1 SIP Proxy Operation

When a call answered event occurs for a user who is enabled for Exchange Unified Messaging (UM) on a Session Initiation Protocol (SIP) proxy compliant with this protocol, the proxy MUST send information about the call answered event to a UM server on the user's dial plan by sending an INFO request on an already existing INVITE dialog. If a dialog does not already exist, the proxy MUST establish one, as described in section 3.2.3.

The INFO request MUST have a **Content-Type** header with "application/ms-rtc-usernotification+xml" as its value. The body of the request MUST conform to the User Notification format, as specified in section 2.2.3.

The **Target** and **TargetClass** elements MUST be present. The **Target** element MUST contain the **Request-URI** used when routing the INVITE. If the destination that answered the call was rung as a part of team or delegate ringing, the **TargetClass** MUST be "secondary". Otherwise, the **TargetClass** MUST be "primary".

If the response that triggered the call answered event has a **P-Asserted-Identity** header, the **AnsweredBy** element MUST contain the URI present in the **P-Asserted-Identity** header.

The **Subject**, **Priority**, **ConversationID**, and **ReferredBy** elements MUST be present if their corresponding headers are present in the INVITE associated with the event.

The **type** attribute of the **Event** element MUST be set to "answered".

After the INFO request has been sent, the **User Notification Inactivity** timer for that UM server MUST be restarted.

3.2.4.3 Call Forbidden Event

When processing an audio call targeted at a user, as specified in [MS-SIPRE], if the call was routed to a destination other than the registered endpoints (5) associated with the address-of-record in the **Request-URI** header field and a 403 response was received on that client transaction, information about the call forbidden event MUST be sent to a Unified Messaging (UM) server if the user is UM-enabled.

3.2.4.3.1 SIP Proxy Operation

When a call forbidden event occurs for a user who is enabled for Exchange Unified Messaging (UM) on a Session Initiation Protocol (SIP) proxy compliant with this protocol, the proxy MUST send information about the call forbidden event to a UM server on the user's dial plan by sending an INFO request on an already existing INVITE dialog. If a dialog does not already exist, the proxy MUST establish one, as described in section 3.1.3.

The INFO request MUST have a **Content-Type** header with "application/ms-rtc-usernotification+xml" as its value. The body of the request MUST conform to the User Notification format, as specified in section 2.2.3.

The **Target** and **TargetClass** elements MUST be present. The **Target** element MUST contain the **Request-URI** used when routing the INVITE. If that destination was rung as a part of team or delegate ringing, the SIP proxy SHOULD NOT send a call forbidden event. If such an event is sent, the value of **TargetClass** MUST be "secondary". If the destination was not a part of team or delegate ringing, the value of **TargetClass** MUST be "primary".

The **Subject**, **Priority**, **ConversationID**, and **ReferredBy** elements MUST be present if their corresponding headers are present in the INVITE associated with the event.

The **type** attribute of the **Event** element MUST be set to "forbidden".

After the INFO request has been sent, the **User Notification Inactivity** timer for that UM server MUST be restarted.

3.2.5 Message Processing Events and Sequencing Rules

None.

3.2.6 Timer Events

When the **User Notification Inactivity** timer associated with a Unified Messaging (UM) server expires, the INVITE dialog SHOULD be terminated. If a new notification needs to be sent to this UM server later, a new INVITE dialog SHOULD be established with it.

3.2.7 Other Local Events

None.

4 Protocol Examples

4.1 Missed Call Event

The following example is an INFO dialog that includes a user notification of a missed call event. For more information, see section 3.2.4.1.

```
INFO sip:exchange-um1.contoso.com:5061;transport=Tls SIP/2.0
FROM: <sip:A410AA79-D874-4e56-9B46-709BDD0EB850>;epid=12E34CB0DB;tag=806bdbd128
TO: <sip:exchange-
um1.contoso.com;opaque=app:rtcevent;transport=tls>;epid=9EEC660CCD;tag=826c5fb8f
CSEQ: 7 INFO
CALL-ID: 63536088-9a24-4b36-a671-82c5de77de9c
CONTENT-TYPE: application/ms-rtc-usernotification+xml

<?xml version="1.0" encoding="us-ascii" ?>
  <UserNotification>
    <User>sip:user@example.com</User>
    <EumProxyAddress>EUM:user@contoso.com;phone-context=dp1.contoso.com</EumProxyAddress>
    <Time>2006-05-12 01:33:32Z</Time>
    <Template>RtcDefault</Template>
    <Event type="missed">
    <CallId>234d82934091df92034ad3e329fae03234</CallId>
      <From>sip:someuser@contoso.com</From>
      <Subject>RE: Quote for widgets</Subject>
      <ConversationID>Aca6SdRQ/SvHLJIHDHoWAEvg==</ConversationID>
      <MissedReason>Declined</MissedReason>
    </Event>
  </UserNotification>
```

4.2 Call Answered Event

The following example is an INFO dialog that includes a user notification of an answered call event. For more information, see section 3.2.4.2.

```
INFO sip:exchange-um1.contoso.com:5061;transport=Tls SIP/2.0
FROM: <sip:A410AA79-D874-4e56-9B46-709BDD0EB850>;epid=12E34CB0DB;tag=806bdbd128
TO: <sip:exchange-
um1.contoso.com;opaque=app:rtcevent;transport=tls>;epid=9EEC660CCD;tag=826c5fb8f
CSEQ: 6 INFO
CALL-ID: a7e36088-9a24-4b36-a671-82c5de77de9c
CONTENT-TYPE: application/ms-rtc-usernotification+xml

<?xml version="1.0" encoding="us-ascii" ?>
  <UserNotification>
    <User>sip:user@contoso.com</User>
    <EumProxyAddress>EUM:user@contoso.com;phone-context=dp1.contoso.com</EumProxyAddress>
    <Time>2006-05-02 12:53:32Z</Time>
    <Template>RtcDefault</Template>
    <Event type="answered">
    <CallId>234d82934091df92034ad3e329fae03234</CallId>
      <From>sip:someuser@contoso.com</From>
      <Subject>RE: Car pool</Subject>
      <ConversationID>Aca6SdRQ/SvHLJIHDHoWAEvg==</ConversationID>
      <Target>sip:teammember@contoso.com</Target>
      <TargetClass>secondary</TargetClass>
```

```
    <AnsweredBy>sip:teammember@contoso.com</AnsweredBy>
  </Event>
</UserNotification>
```

4.3 Call Forbidden Event

The following example is an INFO dialog that includes a user notification of a forbidden call event. For more information, see section 3.2.4.3.

```
INFO sip:exchange-um1.contoso.com:5061;transport=Tls SIP/2.0
FROM: <sip:A410AA79-D874-4e56-9B46-709BDD0EB850>;epid=12E34CB0DB;tag=806bdbd128
TO: <sip:exchange-
um1.contoso.com;opaque=app:rtcevent;transport=tls>;epid=9EEC660CCD;tag=826c5fb8f
CSEQ: 10 INFO
CALL-ID: d5b36088-9a24-4b36-a671-82c5de77de9c
CONTENT-TYPE: application/ms-rtc-usernotification+xml

<?xml version="1.0" encoding="us-ascii" ?>
  <UserNotification>
    <User>sip:user@contoso.com</User>
    <EumProxyAddress>EUM:user@contoso.com;phone-context=dp1.contoso.com</EumProxyAddress>
    <Time>2006-05-02 18:53:32Z</Time>
    <Template>RtcDefault</Template>
    <Event type="forbidden">
    <CallId>234d82934091df92034ad3e329fae03234</CallId>
      <From>sip:someuser@contoso.com</From>
      <Subject>Pricing figures</Subject>
      <Priority>High</Priority>
      <ConversationID>Aca6SdRQ/SvHLJIHDHoWAEvg==</ConversationID>
      <Target>sip:+145532290933@contoso.com;user=phone</Target>
      <TargetClass>primary</TargetClass>
    </Event>
  </UserNotification>
```

5 Security

5.1 Security Considerations for Implementers

None.

5.2 Index of Security Parameters

None.

6 Appendix A: Full User Notification Format

For ease of implementation, the full user notification format is provided as follows:

```
<?xml version="1.0" encoding="utf-8"?>
<xs:schema xmlns:xs="http://www.w3.org/2001/XMLSchema" elementFormDefault="unqualified"
attributeFormDefault="unqualified">

  <xs:simpleType name = "missed-reason-type">
    <xs:restriction base="xs:string">
        <xs:enumeration value="CallerReleased"/>
        <xs:enumeration value="Declined"/>
    </xs:restriction>
  </xs:simpleType>

  <xs:simpleType name = "target-class-type">
    <xs:restriction base="xs:string">
        <xs:enumeration value="primary"/>
        <xs:enumeration value="secondary"/>
    </xs:restriction>
  </xs:simpleType>

  <xs:simpleType name = "event-type-attribute-type">
    <xs:restriction base="xs:string">
        <xs:enumeration value="missed"/>
        <xs:enumeration value="answered"/>
        <xs:enumeration value="forbidden"/>
    </xs:restriction>
  </xs:simpleType>

  <xs:complexType name="event-type">
    <xs:sequence>
        <xs:element name="CallId" type="xs:string" />
        <xs:element name="From" type="xs:string" />
        <xs:element name="Subject" type="xs:string" minOccurs="0" />
        <xs:element name="Priority" type="xs:string" minOccurs="0" />
        <xs:element name="ConversationID" type="xs:string" minOccurs="0" />
        <xs:element name="ReferredBy" type="xs:string" minOccurs="0" />
        <xs:element name="Target" type="xs:string" minOccurs="0" />
        <xs:element name="TargetClass" type="target-class-type" minOccurs="0" />
        <xs:element name="AnsweredBy" type="xs:string" minOccurs="0" />
        <xs:element name="MissedReason" type="missed-reason-type" minOccurs="0" />
    </xs:sequence>
    <xs:attribute name="type" type="event-type-attribute-type" use="required" />
  </xs:complexType>

  <!-- Root document defintion -->
  <xs:complexType name="user-notification-type">
    <xs:sequence>
        <xs:element name="User" type="xs:string" />
        <xs:element name="EumProxyAddress" type="xs:string" minOccurs="0" />
        <xs:element name="Time" type="xs:string" />
        <xs:element name="Template" type="xs:string" fixed="RtcDefault" />
        <xs:element name="Event" type="event-type" />
    </xs:sequence>
  </xs:complexType>

  <xs:element name="UserNotification" type="user-notification-type" />
```

```
</xs:schema>
```

7 Appendix B: 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® Exchange Server 2007 Service Pack 1 (SP1)
- Microsoft® Exchange Server 2010
- Microsoft® Office Communications Server 2007
- Microsoft® Office Communications Server 2007 R2
- Microsoft® Office Communicator 2007
- Microsoft® Office Communicator 2007 R2
- Microsoft® Lync™ 2010
- Microsoft® Lync™ Server 2010
- Microsoft® Lync 15 Technical Preview
- Microsoft® Lync Server 15 Technical Preview
- Microsoft® Exchange Server 15 Technical Preview

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 2.2.1: Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

<2> Section 2.2.6: Office Communications Server 2007, Office Communicator 2007: This behavior is not supported. This behavior was added in relation to Knowledge Base Article 972700, July 2009 hotfix. This hotfix applies to Office Communications Server 2007 R2 and Office Communicator 2007 R2.

<3> Section 3.1.5: Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

<4> Section 3.1.5: Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

<5> Section 3.1.5: Office Communications Server 2007, Office Communicator 2007: This behavior is not supported. This behavior was added in relation to Knowledge Base Article 972700, July 2009 hotfix. This hotfix applies to Office Communications Server 2007 R2 and Office Communicator 2007 R2.

<6> Section 3.1.5: Office Communications Server 2007, Office Communicator 2007: This behavior is not supported.

<7> Section 3.1.5.1: Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

<8> Section 3.1.5.2: Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

<9> Section 3.1.5.3: Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

<10> Section 3.1.5.7: Office Communications Server 2007, Office Communicator 2007: This behavior is not supported.

<11> Section 3.1.5.7: Office Communications Server 2007, Office Communications Server 2007 R2: In these releases, the call was terminated with the lowest numbered response among all the final responses received.

<12> Section 3.2: Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

<13> Section 3.2.3: Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

8 Change Tracking

This section identifies changes that were made to the [MS-EUMR] protocol document between the June 2011 and January 2012 releases. Changes are classified as New, Major, Minor, Editorial, or No change.

The revision class **New** means that a new document is being released.

The revision class **Major** means that the technical content in the document was significantly revised. Major changes affect protocol interoperability or implementation. Examples of major changes are:

- A document revision that incorporates changes to interoperability requirements or functionality.
- An extensive rewrite, addition, or deletion of major portions of content.
- The removal of a document from the documentation set.
- Changes made for template compliance.

The revision class **Minor** means that the meaning of the technical content was clarified. Minor changes do not affect protocol interoperability or implementation. Examples of minor changes are updates to clarify ambiguity at the sentence, paragraph, or table level.

The revision class **Editorial** means that the language and formatting in the technical content was changed. Editorial changes apply to grammatical, formatting, and style issues.

The revision class **No change** means that no new technical or language changes were introduced. The technical content of the document is identical to the last released version, but minor editorial and formatting changes, as well as updates to the header and footer information, and to the revision summary, may have been made.

Major and minor changes can be described further using the following change types:

- New content added.
- Content updated.
- Content removed.
- New product behavior note added.
- Product behavior note updated.
- Product behavior note removed.
- New protocol syntax added.
- Protocol syntax updated.
- Protocol syntax removed.
- New content added due to protocol revision.
- Content updated due to protocol revision.
- Content removed due to protocol revision.
- New protocol syntax added due to protocol revision.

- Protocol syntax updated due to protocol revision.
- Protocol syntax removed due to protocol revision.
- New content added for template compliance.
- Content updated for template compliance.
- Content removed for template compliance.
- Obsolete document removed.

Editorial changes are always classified with the change type **Editorially updated.**

Some important terms used in the change type descriptions are defined as follows:

- **Protocol syntax** refers to data elements (such as packets, structures, enumerations, and methods) as well as interfaces.
- **Protocol revision** refers to changes made to a protocol that affect the bits that are sent over the wire.

The changes made to this document are listed in the following table. For more information, please contact protocol@microsoft.com.

Section	Tracking number (if applicable) and description	Major change (Y or N)	Change type
1 Introduction	Stated that sections 1.8, 2, and 3 of this specification are normative and contain RFC 2119 language. Sections 1.5 and 1.9 are also normative but cannot contain RFC 2119 language.	N	New content added.
1 Introduction	Stated that all sections and examples in this specification, other than sections 1.8, 2, 3, 1.5, and 1.9, are informative.	N	New content added.
1.1 Glossary	Added the glossary terms "Coordinated Universal Time (UTC)", "server", "caller", "Multipurpose Internet Mail Extensions (MIME)", and "XML schema".	N	New content added.
1.2.2 Informative References	Added the [RFC5234] reference.	N	New content added.
1.6 Applicability Statement	Stated that this protocol is applicable when both the SIP protocol server and voicemail server support SIP and intend to use the enhancements offered by this protocol.	N	New content added.
3.1 Extensions for Routing to Exchange Unified Messaging Details	Updated the reference for the proxy deciding to route the call to the user's voice mail.	N	Content updated.
3.1.1 Abstract Data Model	Specified that this section describes a conceptual model of possible data organization that an	N	New content added.

Section	Tracking number (if applicable) and description	Major change (Y or N)	Change type
	implementation maintains to participate in this protocol.		
3.1.2 Timers	Moved the description of the Unified Messaging server timer here from the deleted section 3.1.2.1.	N	New content added.
3.1.5 Message Processing Events and Sequencing Rules	Added a reference for voice-mail-gruu.	N	New content added.
3.1.5 Message Processing Events and Sequencing Rules	Added how the proxy MUST process the request if there are multiple UM servers in the user's dial plan with different versions.	N	New content added.
3.1.5 Message Processing Events and Sequencing Rules	Added how the proxy MUST process the request if at least one UM server in the user's dial plan is a front end or front-end array.	N	New content added.
3.1.5 Message Processing Events and Sequencing Rules	Aded a note regarding having multiple UM servers in the user's dial plan with different versions or having at least one UM server in the user's dial plan that is a front end or front-end array.	N	New product behavior note added.
3.1.5.7 Retrying a Request	Specified that when selecting the next UM server, the UM server version and CAFÉ restrictions specified in step 7 of section 3.1.5 MUST be followed.	N	New content added.
3.1.6 Timer Events	Moved the description of the Unified Messaging server timer expiration here from the deleted section 3.1.6.1.	N	New content added.
3.2.1 Abstract Data Model	Specified that this section describes a conceptual model of possible data organization that an implementation maintains to participate in this protocol..	N	New content added.
3.2.2 Timers	Moved the information about the User Notification Inactivity timer here from the deleted section 3.2.2.1 (A User Notification Inactivity Timer).	N	New content added.
3.2.6 Timer Events	Moved the information about the expiration of the User Notification Inactivity timer here from the deleted section 3.2.6.1 (User Notification Inactivity Timer Expiry).	N	New content added.
7 Appendix B: Product Behavior	Added Microsoft® Exchange Server 15 Technical Preview, Microsoft® Lync 15 Technical Preview, and Microsoft® Lync Server 15 Technical Preview to the list of applicable product versions.	N	New content added.
	Moved the description of the Unified Messaging server timer to Timers (section 3.1.2.).	N	New content added.

Section	Tracking number (if applicable) and description	Major change (Y or N)	Change type
	Deleted section 3.1.6.1 (Unified Messaging Server Timer Expiry), and moved the information to section 3.1.6 (Timer Events).	N	Content removed.
	Deleted section 3.2.2.1 (A User Notification Inactivity Timer), and moved the information to section 3.2.2 (Timers).	N	Content removed.
	Deleted section 3.2.6.1 (User Notification Inactivity Timer Expiry), and moved the information to section 3.2.6 (Timer Events).	N	New content added.

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