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

Revision Summary

Date	Revision History	Revision Class	Comments
12/12/2008	1.0		Initial version
02/13/2009	1.01		Revised and edited the technical content
03/13/2009	1.02		Edited the technical content
07/13/2009	1.03	Major	Revised and edited the technical content
08/28/2009	1.04	Editorial	Revised and edited the technical content
11/06/2009	1.05	Editorial	Revised and edited the technical content
02/19/2010	1.06	Editorial	Revised and edited the technical content
03/31/2010	1.07	Major	Updated and revised the technical content
04/30/2010	1.08	Editorial	Revised and edited the technical content
06/07/2010	1.09	Minor	Updated the technical content
06/29/2010	1.10	Editorial	Changed language and formatting in the technical content.
07/23/2010	1.10	No change	No changes to the meaning, language, or formatting of the technical content.
09/27/2010	2.0	Major	Significantly changed the technical content.
11/15/2010	2.0	No change	No changes to the meaning, language, or formatting of the technical content.
12/17/2010	2.0	No change	No changes to the meaning, language, or formatting of the technical content.

Table of Contents

1 Introduction

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

1.1 Glossary

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

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

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

200 OK
address-of-record
Audio/Video Edge Server (A/V Edge Server)
dial plan
dialog
endpoint
Globally Routable User Agent URI (GRUU)
in-band provisioning
INVITE
long-term credentials
proxy
QoE Monitoring Server
Quality of Experience (QoE)
Session Description Protocol (SDP)
Session Initiation Protocol (SIP)
token
URI (Uniform Resource Identifier)
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

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", June 2008.

[MS-QoE] Microsoft Corporation, "Quality of Experience Monitoring Server Protocol Specification", June 2008.

[MS-SIPRE] Microsoft Corporation, "Session Initiation Protocol (SIP) Routing Extensions", June 2008.

[MS-SIPREGE] Microsoft Corporation, "Session Initiation Protocol (SIP) Registration Extensions", June 2008.

[MS-TURN] Microsoft Corporation, "Traversal Using Relay NAT (TURN) Extensions", June 2008.

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997, http://www.ietf.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.ietf.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.ietf.org/rfc/rfc3892.txt

1.2.2 Informative References

[MS-GLOS] Microsoft Corporation, "Windows Protocols Master Glossary", March 2007.

[MS-OFCGLOS] Microsoft Corporation, "Microsoft Office Master Glossary", June 2008.

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

[XMLNS] World Wide Web Consortium, "Namespaces in XML 1.0 (Third Edition)", W3C Recommendation 8 December 2009, http://www.w3.org/TR/REC-xml-names/

[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 discusses **Session Initiation Protocol (SIP)** extensions that are used to route calls to Exchange Unified Messaging and to generate user notification emails on call events.

These voice mail routing extensions have been designed to route calls to Exchange Unified Messaging (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 [MS-SIPRE] to use UM servers to generate call notification emails 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 described in [MS-SIPRE].

This protocol incorporates the SIP protocols.

1.5 Prerequisites/Preconditions

This protocol assumes that the SIP server implements the Session Initiation Protocol Routing Extensions described in [MS-SIPRE]. This protocol also assumes that the server has knowledge of each voice mail **dial plan** in the enterprise, the 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

None.

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 SIP.

2.2 Message Syntax

All of the message syntax specified in this protocol is described in both prose and an **Augmented Backus-Naur Form (ABNF)**.

2.2.1 Ms-Mras-Address Header Field Syntax

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

The following example is the ABNF 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 Syntax

This protocol defines certain restrictions on the **Request-URI** field value while routing an **INVITE** to a UM server for a user. The **user** portion of the **URI (Uniform Resource Identifier)** 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 UM server about user call events. The UM servers use this information to generate the following types of emails:

- Missed call emails
- Call Answered emails
- Call Forbidden emails

The User Notification events are delivered as **application/ms-rtc-usernotification+xml** content in the body of 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 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 Exchanged Unified Messaging to uniquely identify the user. The **Time** element MUST be a string that corresponds to the time the event occurred in **UTC (Coordinated Universal Time)**.

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 **Referred-By** header in the corresponding INVITE (ABNF for **Referred-By** header defined in [RFC3892]). If the **Target** element is present, it MUST contain the **Request-URI** used when proxying 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 Syntax

2.2.4.1 From Header Field Syntax

This protocol specifies additional restrictions on the **From** header field syntax defined in [RFC3261] for the INVITE **dialog** established to send user event notifications to UM servers.

The original ABNF for **from-spec**, as defined 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 Syntax

This protocol specifies additional restrictions on the **Request-URI** header field syntax defined in [RFC3261] for the INVITE dialog established to send user event notifications to UM servers. The **user** portion of the **Request-URI** header field MUST be empty. The **host** portion MUST be the FQDN 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 Syntax

The INVITE dialog that is established to send user event notifications to UM servers MUST contain a **Session Description Protocol (SDP)** body, as defined 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**, with the value "*".

The SDP body MUST also have the following attribute values:

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

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

2.2.5 User Notification INVITE Response Syntax

The **200 OK** response from the UM server MUST also contain an 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**, with the value "*".

The SDP body MUST also have the following attribute values:

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

2.2.6 Option Tag Extensions

This section follows the product behavior described in footnote <2>.

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

Ms-Fe: This option tag is for support of 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 SIP proxy adheres to the specifications of this protocol.

3 Protocol Details

3.1 Extensions for Routing to Exchange Unified Messaging

[MS-SIPRE] specifies how an incoming audio INVITE is processed based on a routing script preamble published by the user. As described in [MS-SIPRE] section 3.7.5, under various circumstances, the proxy can decide to route the call to the user's voice mail. The extensions for routing to Exchange Unified Messaging (UM) are also applicable when the **Request-URI** matches the ABNF for **voice-mail-gruu** syntax, as defined 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 Unified Messaging if the user is UM-enabled. This protocol also defines a mechanism for providing the UM server with the SIP URI of an A/V Edge Server, if available. The UM server uses this URI to obtain authentication (2) tokens when needed. The protocol for obtaining these tokens is specified in [MS-AVEDGEA].

3.1.1 Abstract Data Model

A SIP proxy compliant with this protocol maintains a database of all the dial plans in the enterprise and all the 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 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

3.1.2.1 Unified Messaging Server Timer

When the INVITE is routed to a UM server, a UM server timer is started. 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 SIP proxy decides to route an INVITE to voice mail based on rules specified in [MS-SIPRE] or when an INVITE with a **voice-mail-gruu** arrives, and the user is Exchange Unified Messaging-enabled, the proxy MUST process the request as follows:

1. If the INVITE already contains any **Diversion** headers, these headers MUST be removed before forwarding the request to Exchange Unified Messaging. The **Diversion** header is defined in [IETFDRAFT-DIISIP-08].

2. If the **user** and **host** portion of the **Request-URI** field is not the same as that 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**, which are defined 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 A/V Edge Server.<3> The **QoE Monitoring Server** protocol and the message format for **in-band provisioning** requests to obtain provisioning information are defined in [MS-SIPREGE] section 3.3.

4. If an A/V Edge Server is configured, the proxy MUST add a **Ms-Mras-Address** header with the value of the A/V Edge Server GRUU. The syntax of the GRUU is defined in [MS-SIPRE]. The **Ms-Mras-address** header on 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 include the **Ms-Fe** option tag in the **Supported** header field of the request if one is not already present.<5>

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. The proxy MUST start the Unified Messaging Server timer.

8. Outgoing messages from Exchange Unified Messaging (UM) can include an **Ms-fe** header parameter containing its specific 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 defined in [MS-SIPRE].

3.1.5.1 Interaction With Audio/Video Edge Server (A/V Edge Server)

This section follows the product behavior specified in footnote <6>.

The UM server can request **long-term credentials** from the A/V Edge Server. The protocol and message format for requesting an authentication (2) token is defined in [MS-AVEDGEA] section 2. When the caller's 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 Quality of Experience (QoE) report

This section follows the product behavior described in footnote <7>.

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

3.1.5.3 Processing 302 Response

This section follows the product behavior described in footnote <8>.

The UM server can return a 302 Redirect response to the INVITE. The 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 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 101 Progress Report

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

3.1.5.5 Processing 415 Response

If a UM server returns a 415 response and the INVITE has **multipart/MIME** content, the proxy MUST clear the UM server's ordering list and process the 415 as specified in [MS-SIPRE] section 3.8.5.5. If a fresh INVITE is sent as a result of this processing, the proxy MUST re-do the steps listed in section 3.1.5.

3.1.5.6 Processing Other Responses

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

If the 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 Request

If the 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.

If all servers in the list have been attempted, the call SHOULD be terminated with a response code 480.<9>

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

3.1.6 Timer Events

3.1.6.1 Unified Messaging Server Timer Expiry

When the Unified Messaging Server timer expires, the request MUST be routed to the next 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

This protocol specifies a mechanism for sending call event notification emails to users through Exchange Unified Messaging (UM). The event information is sent to the UM server using the SIP INFO method over an already-established SIP INVITE dialog. The INFO method is specified in [RFC2976]. 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<10>:

- **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

A SIP proxy that implements this protocol MUST maintain a database of all dial plans in the enterprise and all the UM servers in each dial plan. When a call event occurs that requires a notification email 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

3.2.2.1 User Notification Inactivity Timer

When an INVITE dialog is established with an UM server, a User Notification Inactivity timer MUST be started. The wait time for this timer is 10 minutes.

3.2.3 Initialization

A SIP proxy implementing this protocol MUST establish a SIP INVITE dialog with one of the UM servers in the user's dial plan before sending the event information. The proxy can randomly pick a UM server from the list<11>. 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. Once 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 defined 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 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 an Exchange Unified Messaging-enabled user on a 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".

Once 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 defined 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 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 an Exchange Unified Messaging-enabled user on a 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 proxying 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".

Once 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 defined 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 UM server if the user is UM-enabled.

3.2.4.3.1 SIP Proxy Operation

When a call forbidden event occurs for an Exchange Unified Messaging-enabled user on a 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 proxying 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 **TargetClass** MUST be "secondary". If the destination was not a part of team or delegate ringing,, the **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".

Once 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

3.2.6.1 User Notification Inactivity Timer Expiry

When the User Notification Inactivity timer associated with a 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 Example

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 Example

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 Example

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® Office Communications Server 2007
- Microsoft® Office Communicator 2007
- Microsoft® Office Communications Server 2007 R2
- Microsoft® Office Communicator 2007 R2
- Microsoft® Lync™ Server 2010
- Microsoft® Lync™ 2010
- Microsoft® Exchange Server 2010
- Microsoft® Exchange Server 2007 Service Pack 1 (SP1)

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 added in relation to Knowledge Base Article 972700, July 2009 QFE. This QFE 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 added in relation to Knowledge Base Article 972700, July 2009 QFE. This QFE applies to Office Communications Server 2007 R2 and Office Communicator 2007 R2.

<6> 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.

<7> 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.

<8> 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.

<9> 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.

<10> Section 3.2: Office Communications Server 2007, Office Communicator 2007, Office Communications Server 2007 R2, Office Communicator 2007 R2: This behavior is not supported.

<11> 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

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

9 Index

S

T

U

V