International Multimedia Telecommunications Consortium

Universal MultiMedia Addressing Platform (UMMAP)

Recommendation A.1: Acceptable Addressing Formats

Version 1.0.5

Contents

Version number	Modifications	Date
0.1	Original document	April 2, 2004
0.2	Draft sent to UMMAP list	April 12, 2004
0.3	Feedback provided during April 14 UMMAP meeting	April 20, 2004
0.4	Miscellaneous typographical and formatting changes	April 22, 2004
0.5	Clarified TEL: URI and E.164 address issues	April 29, 2004
0.6	Added reference to RFC 3508; minor wording changes	May 7, 2004
0.7	Added a glossary & clarification/examples as noted	June 7, 2004
0.8	Added section numbers, updated glossary, added Appendix A	October 11, 2004
0.8.2	Changed sections 3.0, 3.1.3, 4.1, updated page numbers in Table of Contents, removed Appendix A to section 1 Added reference to Appendix A in section 4.0 Added diagrams and flowcharts to Appendix A	October 21, 2004 December 3, 2004
0.8.7	Changed sections 1.0, 4.1,4.2,4.3	December 17, 2004
0.8.7.1	Replaced 5 figures in Appendix A with 9 new Figures.	January 11,2005
0.9	Replaced reference to 2806bis with current and correct RFC numbering to RFC 3966	March 16, 2005
0.9.1	Added footnote in section 3.1.3 Changed section 3.2 and added section 3.2.1 Updated table in section 4.1 Removed tables and examples from section 4.2 Reformatted Reference section 5.0 Removed 2 figures from section 7	April 4, 2005
1.0	Removed table in section 4.1 Removed 1st paragraph in section 4.2 Changed 2nd bullet point in section 4.2 Removed sentence in section 4.3	April 18, 2005
1.0.1	Removed note and bullet points in section 4.1 Changed it's to it in 3rd paragraph section 4.3	April 20,2005
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1.0.3	Changed misspelling in Section 9 Added Acknowledgements section Adjusted text spacing of figures in Appendix A	May 4, 2005
1.0.4	Reworded section 2 Fixed grammar in section 3.1.3 and section 10 Added company names to author's names in section 5 Added reference to RFC 2119 in section 6 Adjusted formatting in section 6	May 23, 2005
1.0.5	Removed the word "Draft" from title page. Removed the watermark that said "DRAFT"	August 9, 2005

1.0 - Purpose and Scope

The IMTC Universal Multimedia Addressing Platform (UMMAP) Recommendation A.1 defines a set of addressing guidelines which ensure that any multimedia conferencing user or system can be uniquely identified and signaled on the Internet. The overarching goal is that any user can be reached at an address, regardless of the source or destination of the call, or the source, destination or intermediate signaling protocols in use.

UMMAP recommendation A.1 is a designed to have manufacturers and vendors support multiple native dialing techniques that allow end users of any one device (H323, H320, SIP, E164) to dial another device (H323, H320, SIP, E164) without having to use any special characters that are not native to that device. For Example, if a person is using a H323 device to call a SIP device that user can input either the E164 address, SIP URI, or an IP address and the corresponding gateway(s) along the route will route the call to the correct device.

To use OSI terms, UMMAP is similar to working at the Presentation layer between the application and session layers. An end user enters in a dial string on their endpoint (Application Layer). The end point then sends the packets through the system to the destination device based upon one of the core UMMAP addressing schemes. The two endpoints negotiate the session and the information is presented to the destination end point where it is resolved in to a two way or multi person communication. The end users are not aware of the underlying infrastructure to make this happen.

Drawing from existing standards, UMMAP Recommendation A.1 prescribes acceptable addressing formats that can be used by multimedia conferencing systems to enable any user or system to be contacted. In short, using systems conforming to UMMAP Recommendation A.1, any user should be able to say, ‘Here is my address. Call me,’ and be assured that a call placed to that address will connect. UMMAP does not prescribe protocol level or network level functionality, but rather assumes that functionality to be already present.

2.0 - Definitions

In this document, the key words “MUST” and “SHOULD” are to be interpreted as described in BCP 14, RFC 2119, [RFC2119] and indicate requirement levels for complaint implementations. .

3.0 - UMMAP Recommendation

UMMAP allows several core addressing types, including H.323 URL, SIP URI and Tel URI. UMMAP does not favor one of these types, but rather leaves it to the market and individual implementations to choose which types to use and publish. For example, UMMAP does not prescribe that all H.323 Internet systems have E.164 numbers reachable by Tel URI through ENUM. Instead, an H.323 Internet system may have H.323 URLs and no corresponding E.164 address.

UMMAP Recommendation A.1 does not specify the user interface implementation at the terminal device. This allows for devices to use any user input method that a manufacturer develops, and allows for the use of an implied protocol type (e.g. IP phones may insert a TEL: prefix into the call signaling channel, even if the user does not dial it prior to dialing the E.164 address).

It should be noted that a string of numbers, regardless of the format, is meaningless without context. Therefore, all valid addresses MUST include context information. i.e. a TEL:, SIP: or H323: prefix (or other UMMAP-compliant prefix) as defined in RFC 2483 [RFC2483] to be meaningful.

For example, the digit string 19816012813 could be:

- IP address 198.160.128.13
- Telephone number +1-918-601-2813
- a ViDe, Packet8, Level3, or other ‘private’network number

However, if a defined URI is used, the address becomes meaningful. The digit stream,”19816012813” has no meaning, whereas “tel:+19816012813” is defined. The same is true for “me@somewhere.com”, is not clearly defined. Whereas sip:me@somewhere.com”, “h323:me@somewhere.com” or mailto:me@somewhere.com is clearly defined.

3.1 - Core Address Types

3.1.1 - H323 URL

H.323 defines a URL syntax (described in RFC 3508 [RFC3508]) for resolution of H.323 addresses using DNS. When an address is represented in H.323 URL format, it SHOULD be resolved using a DNS lookup as specified by ITU-T Recommendation H.323.

3.1.2 - SIP URI

The Session Initiation Protocol (SIP) defines URL syntax for resolution of SIP addresses using DNS. When an address is represented in SIP URI, or SIPS URI format, it SHOULD be resolved using a DNS lookup as specified by IETF RFC 3261 [RFC3261].

3.1.3 - Tel URL

The Tel URL defines URI syntax for the representation of telephone numbers on the Internet. Telephone numbers SHOULD be represented in Tel URL format as specified in IETF RFC 3699 [RFC3699]. E.164 numbers MUST be fully represented as globally unique numbers per RFC 3699, or must include the appropriate phone-context information, which (in effect) makes the URI globally unique.

ENUM defines a method for translating E.164 telephone numbers into Internet addresses resolvable through DNS. When an address is represented as E.164 number, e.g. in "tel" or "sip" URI format, it SHOULD be resolved using a DNS lookup as specified by IETF RFC 3761 [RFC3761] and related RFCs

The ENUM lookup returns normally a "sip" or "h323" URI. If a different "tel" URI is returned, another ENUM lookup should be made[1]. If ENUM cannot be used to resolve the destination, another database (e.g. Carrier ENUM) may be queried or the call may be routed to the PSTN per default. [1]

Local numbers with phone-context as defined in RFC3699 may be routed to other domains if a common understanding of the phone-context exists between the domains.

It should be noted that the "tel" URI allows for numeric address space beyond E.164 and these addresses are in conformance with UMMAP Recommendation A.1 as long as the "tel" URI is properly formed."

[1] A mechanism must be in place to ensure that only 5 lookups (iterations) are done to avoid loops.

3.2 - Gateway Dialing

It is the responsibility of the system originating the call to present an address to the network in the format of the target address. This is trivial when placing calls between systems using the same protocol. When dialing between protocols, a gateway must be used, thus the system initiating the call is responsible for indicating a gateway is necessary for call completion. For example, if an H.323 system calls a SIP system, the H.323 system must forward the call to a gateway capable of resolving the SIP URI and establishing the signaling connection to the destination.

UMMAP does not specify how that gateway is discovered or signaled. Thus, every system that is in conformance with UMMAP Recommendation A.1 MUST be capable of dialing every UMMAP Core Address Type.

The call controller must know about gateways available to translate between signaling protocols based on the source device capabilities and the dialed destination URI. Additionally, the call controller must be capable of routing the call thru any transcoding systems necessary for content protocol conversion as necessary.

3.2.1 – Gateway Dialing example

In some cases, a network may choose to implement one technology internally (such as SIP) but present one or more protocols externally via local gateways. Thus, even though a destination is a SIP device, it can be reached via SIP, H.323, or TEL via the assigned gateways, and the originating caller (and source network) needs to do nothing to put the call thru. With the gateway at the destination, the URI assigned must be in the source form and the gateway will translate the URI into the correct destination address. For example, a H323 to SIP gateway would have a H323 URL assigned to each SIP destination, and when an incoming call arrived for H323:user@net.com, the gateway could translate it to SIP:user@net.com and process the call.

There is a need for a standard method to locate translational gateways and to learn their capabilities, as none exists today.

4.0 - Conformance

UMMAP defines conformance: Address Representation, Call Signaling and User Input. (See accompanying diagrams and flowcharts in Appendix A)

4.1 - UMMAP Address Representation Conformance

An address is said to be UMMAP Address Representation Conformant when it is represented in one of the Core Address Types. Address representations are instances where an address is stored in a database, displayed on a web page, included in an email, printed on paper letterhead, or the like. A UMMAP conformant address MUST include a universally recognizable scheme, such as a protocol prefix (e.g. 'tel:' or 'h323:')as part of the address.

Dialing non-E.164 numbers from a PSTN device presents a serious challenge. Several options exists, from replacing all PSTN phones with IP phones (not likely in the short term), special prefix digits, IVR/speech recognition system, personal calling list (maintained via the web), or operator-assisted. The actual solution and implementation is outside the scope of this recommendation.

Note: Since legacy ISDN systems are still dominant in the marketplace, a URI scheme for H.320 calls is needed. This could be in the format H320:+17175551212 for multichannel bonded calls and H.320:+17175551212*17175551213 for non-bonded NX56/64 calls. (Call type and bandwidth requirements are transmitted via standard Q.931 call setup messages). Most legacy ISDN systems will support these numbering formats.

4.2 - UMMAP Call Signaling Conformance

A system that is said to be UMMAP Call Signaling Conformant MUST be capable of receiving a call in the format in which its address is represented. For example, an H.323 endpoint whose UMMAP Address Representation is h323:john@gk.domain must be capable of receiving calls in H.323 URL format.

A system receiving a call in one format but that needs to be translated to another signaling format (i.e. H.323 to SIP, H.323 to TEL, SIP to TEL) must be able to:

- Accept the incoming call from the originating terminal
- Accept the incoming call via the appropriate gateway - The gateway continues to translate the call.

A PSTN call to an H.323 and or SIP device is not applicable at this time according to RFC 3699. This poses a severe problem. Endpoints from the SIP and H.323 worlds can access an endpoint in the PSTN world. But an endpoint in the PSTN world cannot access an endpoint in the SIP or H.323 world without the use of a gateway and an assigned E.164 address.

However, a solution *could* take the form of a TEL URI that can be routed to the H.323 and SIP worlds. For example, a SIP endpoint could be 12125551234@somewhere.com, allowing for the SIP users to dial via URI, but PSTN dialers only dial up to the @ sign.

4.3 - UMMAP User Input Conformance

An endpoint that is said to be UMMAP User Input Conformant MUST allow the user to enter and store an address in ALL of the recommended Core Address Types in their native syntax without embedding. UMMAP does not specify the method of user input, but these methods might include keyboard entry, text entry via numeric dial pad, web page, voice recognition, or selection through directory lookup.

For example, to dial tel:+15735551212, the user interface could require:

- Pressing a TEL: button and dialing 15735551212
- Dialing 15735551212 (with the device presuming TEL: as the prefix)
- Dialing 835*15735551212

The user must input, and the device must store, the Core Address in its native format without embedding.

A distinction must be made between the device and what the user does to dial a number. The user deals with the terminal's UI, which merely instructs the terminal what number is being requested. The user may or may not be required to insert the context information, and may dial digits or letters (depending on the destination address and the terminal UI). However, the terminal MUST transmit a full-qualified UMMAP-compliant address on the network.

The manner in which the terminal's call controller handles the request may be based on an explicit address method (i.e. if tel:, send to the PSTN; if SIP or H.323, use the IP network); or may cycle thru one or more directory services (a H.350 LDAP database; ENUM, etc.) looking for a match to determine what to do with the call, returning an unreachable only if all lookups fail.

5.0 – Acknowledgements

This paper could not have been done without the contributions from the following people. David Olson, MOREnet; Christine Perey, Perey Research & Consulting; Tyler Johnson, University of North Carolina; Alan Stillerman, Indiana Higher Education Telecommunication System; Richard Stastny, OFEG/TELEKOM Austria; Jamie Poindexter, University of Wisconsin-Extension; Lawrence Conroy, Siemens Roke Manor Research and Stuart Morris, GlowPoint Inc.

6.0 - References

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- ITU-T Recommendation H.323 (2000), *Packet-based multimedia communications systems.* From http://www.itu.int/rec/recommendation.asp?type=folders&lang=e&parent=T-REC-H.323

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[RFC 3761] Faltstrom, P., Mealling, M., "The E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)", RFC 3761, April 2004. From http://www.ietf.org/rfc/rfc3761.txt

[RFC 3966] Schulzrinne, H., "The tel URI for Telephone Numbers". RFC 3966, December 2004, From http://www.faqs.org/rfcs/rfc3966.html

7.0 - Contact

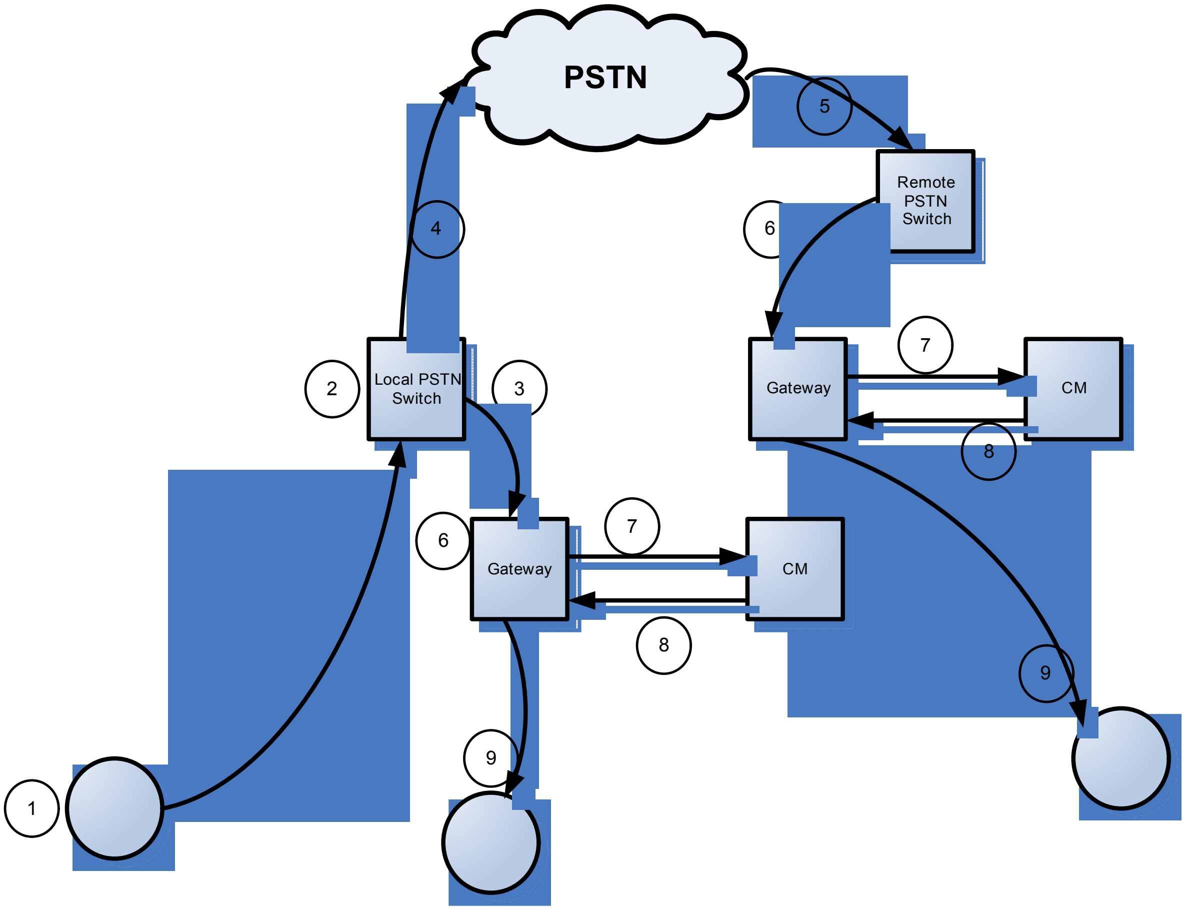
For questions or comments about this recommendation, contact the UMMAP Activity Group (details listed at http://www.imtc.org/activity_groups/UMMAP.asp)

8.0 – Appendix A – Call signaling Flow Charts

8 call types are represented in the following flow charts.

Figure 1 – PSTN – H.323
Figure 2 – PSTN – SIP
Figure 3 – H.323 – SIP_IP_1
Figure 4 – H.323 – SIP_IP_2
Figure 5 – H.323 – PSTN
Figure 6 – H.323 – H.323_IP

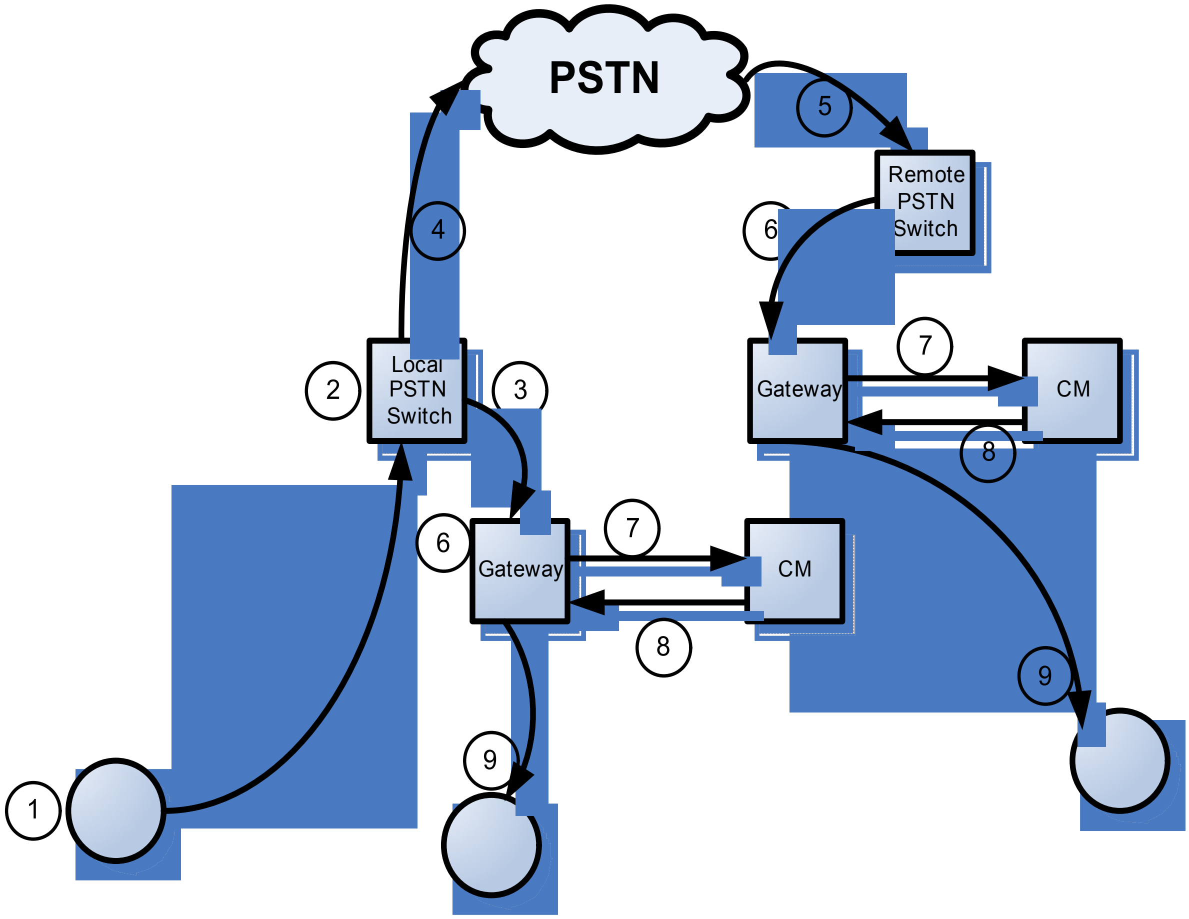
Figure 1



PSTN to H.323 call using E.164 addressing
Destination Number: TEL:+1-444-555-1234

1. Caller picks up phone and dials number (1-444-555-1234).
2. Local CO switch looks at digits dialed to determine route to destination.
3. If a local call, Local PSTN switch connects to the local destination (jump to step 6).
4. If a long-distance call, Local PSTN switch routes to the toll network.
5. Toll network delivers call to remote PSTN switch
6. PSTN connects to the destination number, which is a PSTN<>H.323 gateway.
7. Gateway asks Call Manager where to send the incoming call.
8. Call Manager responds with redirection information.
9. Call is connected to destination device.

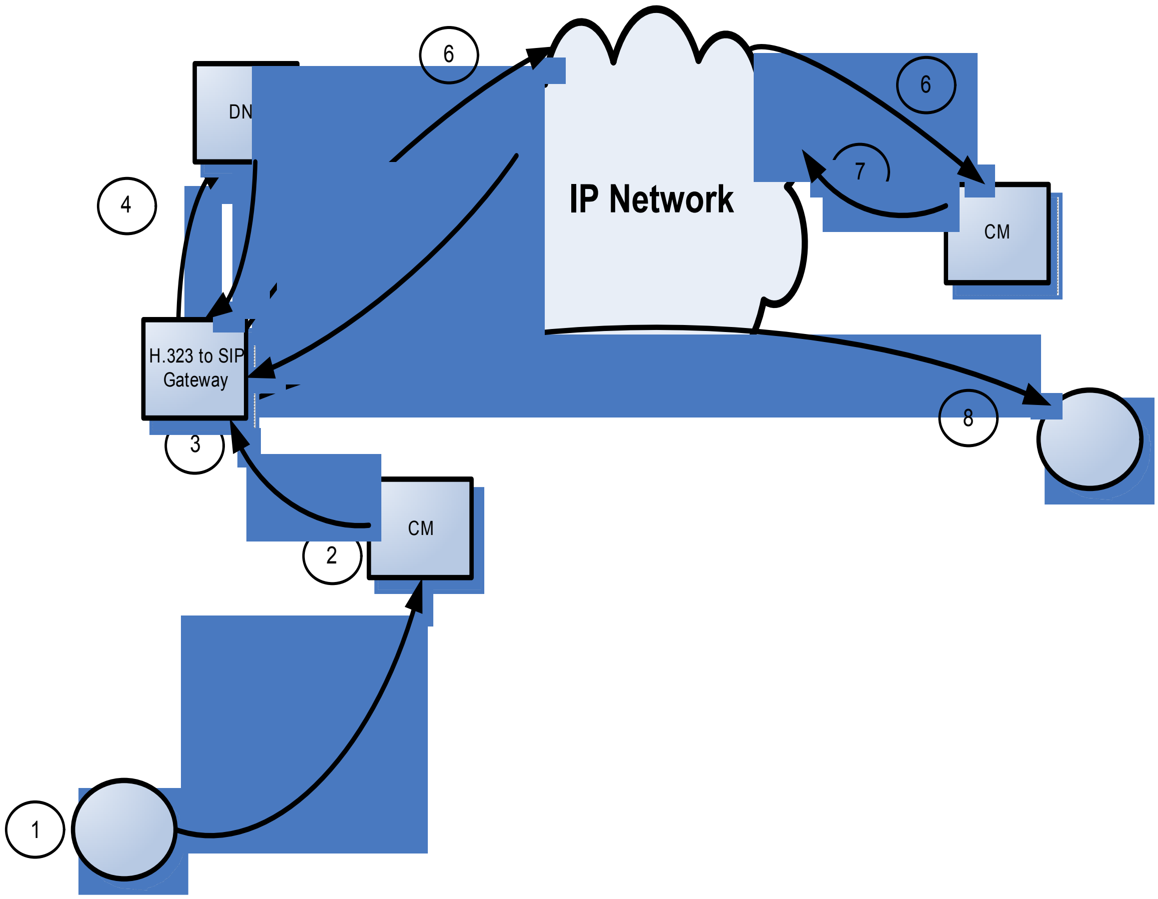
Figure 2



PSTN to SIP call using E.164 addressing
Destination Number: TEL:+1-444-555-1234

1. Caller picks up phone and dials number (1-444-555-1234).
2. Local CO switch looks at digits dialed to determine route to destination.
3. If a local call, Local PSTN switch connects to the local destination (jump to step 6).
4. If a long-distance call, Local PSTN switch routes to the toll network.
5. Toll network delivers call to remote PSTN switch
6. PSTN connects to the destination number, which is a PSTN<>SIP gateway.
7. Gateway asks Call Manager where to send the incoming call.
8. Call Manager responds with redirection information.
9. Call is connected to destination device.

Figure 3

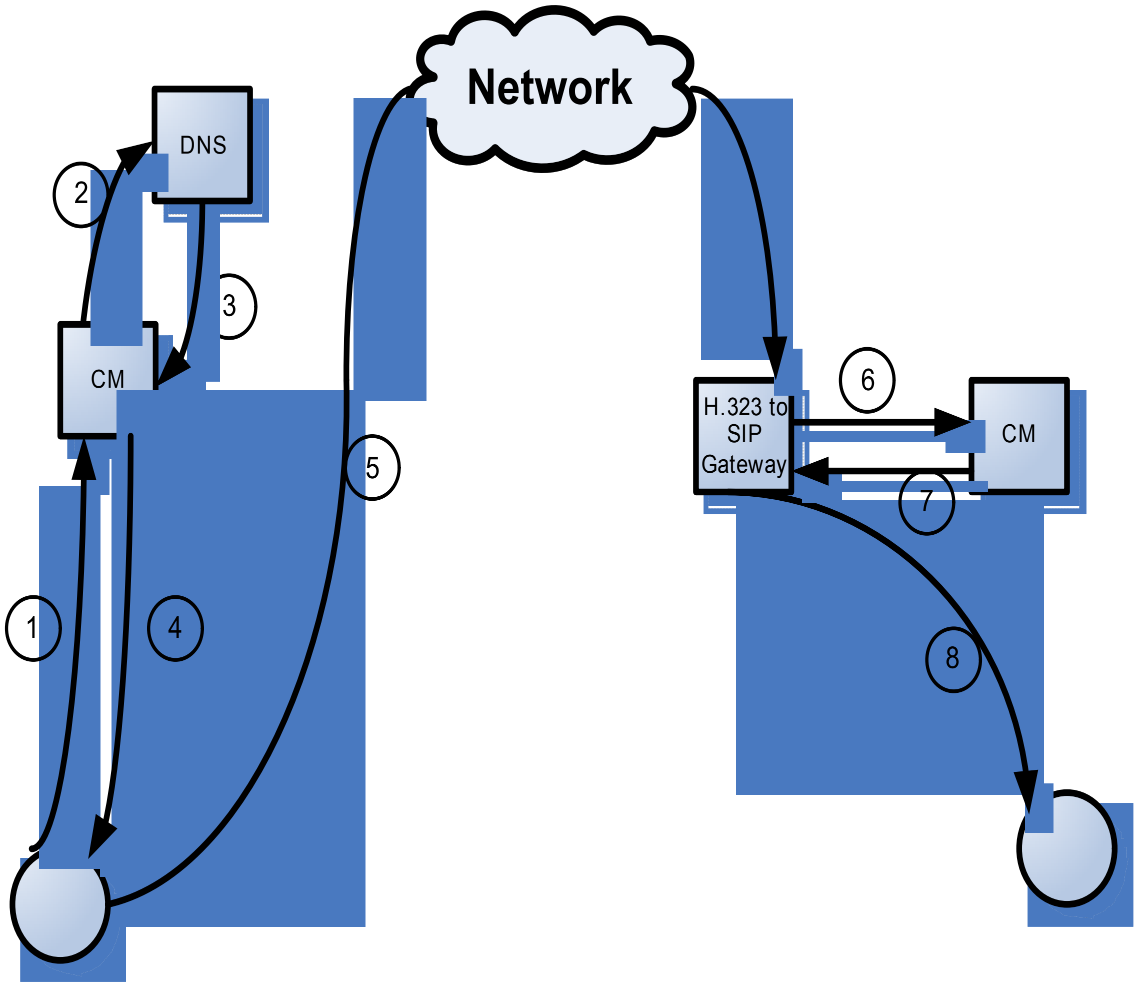


H.323 to SIP call via IP using SIP URI addressing
Destination Number: SIP:user@somewhere.com

1. Caller picks up phone and dials URI (user@somewhere.com).
2. Call Manager parses URI and determines that it needs to go to a H.323-SIP gateway.
3. H.323-SIP gateway accepts H.323 call and originates SIP call to user@somewhere.com.
4. Gateway asks DNS where to send the call.
5. DNS responds with destination CM information.
6. Gatway connects to destination CM and asks for endpoint information.
7. CM provides endpoint information
8. Call is connected to destination device.

NOTE: The presumption is made that the originating CM knows where a H.323 to SIP GW is. This could also be done with a gateway at the destination, answering all the H.323 calls and translating them to the SIP 'internal' address. However, this method requires the destination to actively maintain the gateway and the translations, or to have a directory to look up the H.323 to SIP translations (i.e. H.350).

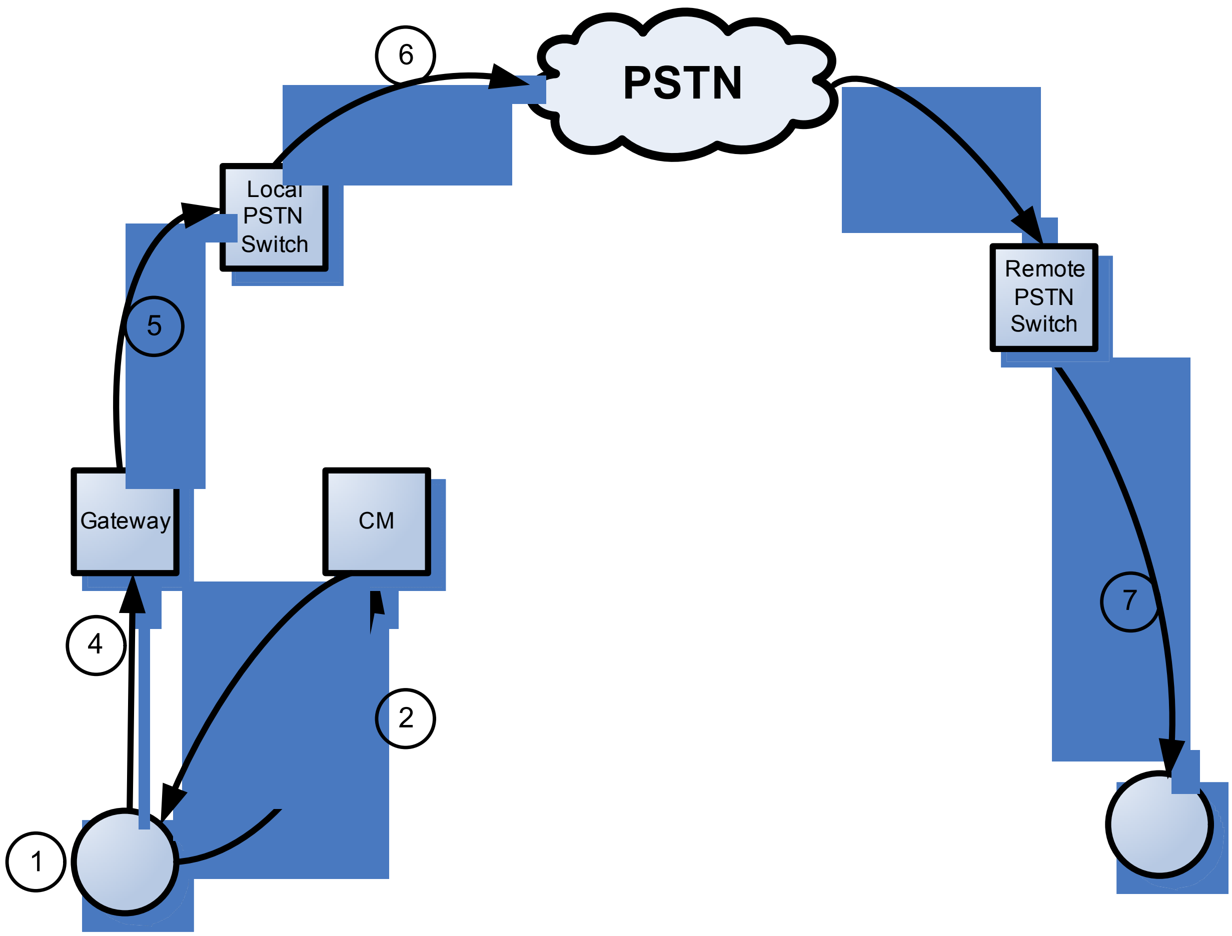
Figure 4.



H.323 VoIP to SIP VoIP call
Destination Number: SIP:dave@some.net

1. Caller picks up phone and dials URI (SIP:dave@some.net).
2. Call Manager requests a lookup of the URI via DNS and determines the destination.
3. DNS returns the destination of the URI.
4. Call Manager issues redirect to the phone based on URI lookup.
5. Call terminates on H.323 to SIP gateway.
6. Gateway asks Call Manager where to send the incoming call.
7. Call Manager responds with redirection information.
8. Call is connected to destination device.

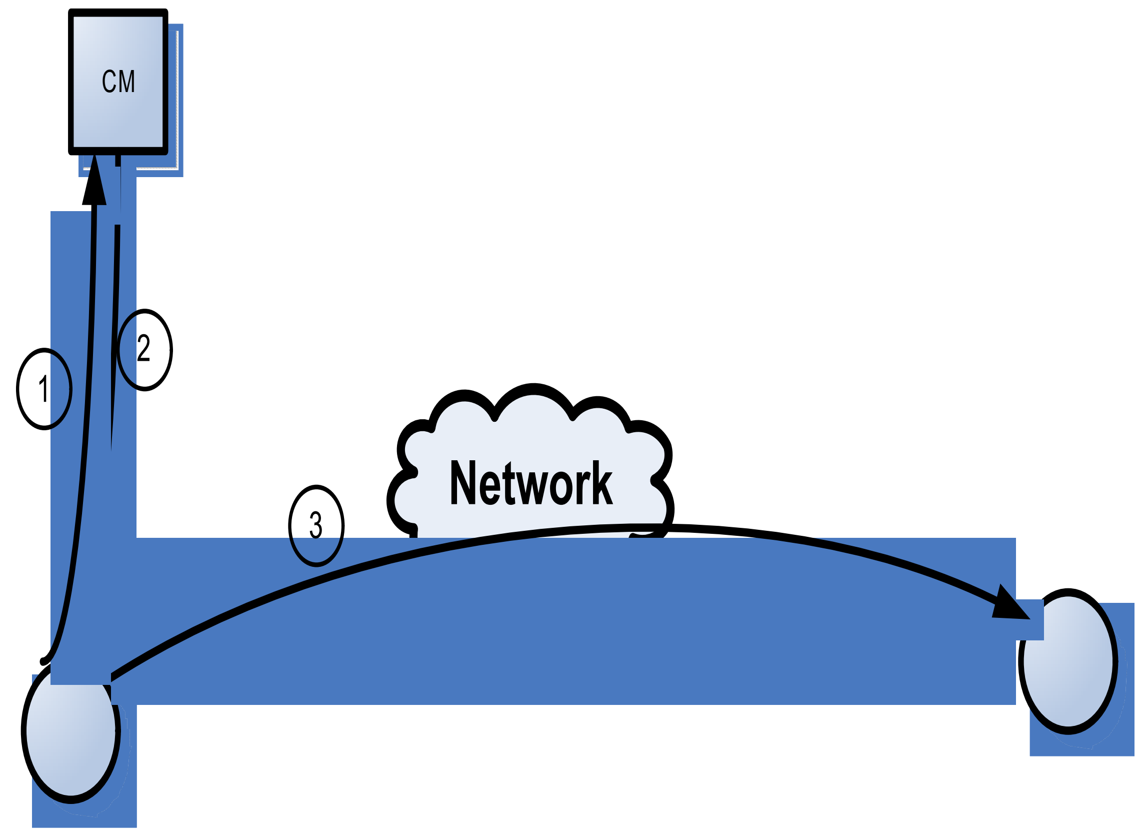
Figure 5



H.323 to PSTN call using E.164 addressing
Destination Number: TEL:+1-444-555-1234

1. Caller picks up phone and dials number (1-444-555-1234).
2. CM looks at dialed digits and determines gateway to use for call.
3. CM redirects call to the gateway.
4. Phone talks to gateway for call.
5. Gateway connects to PSTN and dials number.
6. PSTN routes call to the appropriate end PSTN switch
7. End switch rings the destination device.

Figure 6



H.323 to H.323 call (or SIP to SIP) using URI
Destination Number: H323:user@somewhere.com (or IP:user@somewhere.com

1. Caller picks up phone and dials URI.
2. Call Manager looks up the URI and redirects the phone to the destination device.

9.0 - Glossary

E.164 - An ITU-T standard which defines the international public telecommunication numbering plan. It is a 16 digit numbering scheme that provides a unique telephone number for every subscriber in the world. E.164 addresses can be used in DNS by using ENUM.

ENUM – ENUM (RFC 3761) is a protocol that is the result of work of the Internet Engineering Task Force's (IETF's) Telephone Number Mapping working group. ENUM allocates a specific zone, namely e164.arpa for use with E.164 numbers. Any phone number, such as +1 555 42 42 can be transformed into a hostname by reversing the numbers, separating them with dots and adding the e164.arpa suffix, like so: 2.4.2.4.5.5.5.1.e164.arpa. DNS can then be used to look up internet addresses for services such as SIP VoIP telephony.

Call Controller - A call controller is a software application that provides basic call processing as well as signaling and connection services to configured devices including IP phones and soft-phones, Voice over IP gateways, video terminals, software applications, and other devices on a packet network.

Gateway - A hardware or software set-up that translates between two dissimilar protocols.

H.323 - Is an ITU-T standard for the transmission of real-time audio, video and data information over packet switching-based networks. Such networks include IP-based (including the Internet), Internet packet exchange-based local area networks, enterprise networks and metropolitan and wide area networks. H.323 can also be applied to multipoint-multimedia communications. The technology provides a vast array of services which means it can be used in consumer, business and entertainment applications. H.323 is an essential element in ensuring that the compatibility of the mobile multimedia applications and services will be introduced with the implementation of third generation wireless technologies.

H.323 URL – Defined in H.323 version 4, H.323 URL allows entities to access users through a common syntax. The syntax takes the form: h.323:user@host, where user is a user or service and host is a gatekeeper. Example. H.323:astiller@157.91.29.221.

H.350 - Is a lightweight directory access protocol (LDAP) object class specification designed to store information related to SIP, H.323 and H.320 voice and video endpoints. Such information includes their IP addresses, aliases and other connection related details. Because the information is LDAP-based it can be combined with a traditional corporate directory to give employees a single place to look up co-worker information and connect to them, be it via the telephone or video endpoint.

IMSI - International Mobile Subscriber Identity. An IMSI is required so that any public land mobile network (PLMN) can identify a roaming mobile handset, terminal or user, in order to contact the subscriber's home network for subscription and billing information. This IMSI is used by all mobile operators in the world to be able to identify any mobile handset logged onto their network. An IMSI code therefore has worldwide validity.

IP - Internet Protocol, often referred to as TCP/IP for Transport Control Protocol and Internet Protocol. Designed for non real time delivery of data. Voice and Video applications have required Quality of Service (QoS) protocols to be developed to deliver real time Voice and Video applications.

IP Routing - Routing is the process of deciding the disposition of each packet that a router handles. This applies to incoming packets, outbound packets leaving your network for external destinations, and those packets being routed among your internal networks.

ISDN - (Integrated Services Digital Network) - Digital telephony scheme that allows a user to connect to the Internet over standard phone lines at speeds higher than a 56K modem allows. Integrated Services Digital Network is a system of digital phone connections which allows voice and data to be transmitted simultaneously across the world using end-to-end digital connectivity.

LDAP – (Lightweight Directory Access Protocol) A typical LDAP server is a simple network-accessible database where an organization stores information about its authorized users and what privileges each user has. Thus rather than create a new employee an account on 50 different computers, the new employee is entered into LDAP and granted rights to those 50 systems. If the employee leaves, revoking all privileges is as simple as removing one entry in the LDAP directory.

LERG - (Local Exchange Routing Guide) This is a document that lists all North American Class 5 offices (Central Offices, or end offices) and which describes their relationships to Class 4 offices (Tandem Offices). Carriers use the LERG in the network design process.

NANPA - The North American Numbering Plan Administrator is responsible for day to day administration, assignment and management of area codes in the United States.

PIDF - (Presence Information Document Format) An XML document from the IETF that contains the name, address and current status of an instant messaging user. PIDF documents are exchanged when users subscribe to a buddy list or when users are notified that a buddy has gone online or offline.

SIMPLE – SIP for Instant Messaging and Presence Leveraging Extensions. Is an effort by the SIMPLE WG to make Instant Messaging and Presence indicators, whether they are "off line" or "on line", available to other entities by using SIP as a transport mechanism

SIP - Short for Session Initiation Protocol, SIP is an Internet Engineering Task Force (IETF) standard protocol for initiating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality.

SIP URI – SIP Uniform Resource Indicator. This is the syntax that allows SIP messages to communicate. The general syntax is: sip: user@destination. sip:astiller@ihets.org

SS7 - Signaling System 7: An international high speed signaling backbone for the public switched telephone network (PSTN). The basis for modern methods to route traffic with out-of-brand signaling.

TEL URI – Telephone URI as defined in draft-ietf-iptel-tel-rfc3699-05. Defines telephone number resources identified by telephone numbers. TEL URI takes on the syntax of: tel:telephone-subscriber tel:+1-212-554-4839. Note: dashes added only for readability.

TRIP - TRIP (Telephony Routing over IP, RFC 3219) is a policy driven dynamic routing protocol for advertising the reach ability of telephony destinations, and for advertising attributes of the routes to those destinations. TRIPs operation is independent of any signaling protocol, hence TRIP can serve as the telephony routing protocol for any signaling protocol. TRIP is a routing protocol based on a well-known Internet routing protocol called BGP-4 that forms much of backbone of the Internet today.

URI - Uniform Resource Identifier is the name and address syntax of present and future objects on the Internet. In its most basic form, a URI consists of a scheme name (such as file, http, ftp, news, mailto, gopher) followed by a colon, followed by a path whose nature is determined by the scheme that precedes it (see RFC 1630). URI is the umbrella term for URNs, URLs, and all other Uniform Resource Identifiers.

URL - (Uniform Resource Locator) – A subset of URI. URL takes on the form of http://address for the internet. Example: http://www.ietf.org

XMPP – (Extensible Messaging and Presence Protocol) An open, XML-based protocol for near real-time extensible messaging and presence events.

10.0 - Revision History

A.1v1.0.5: August 9, 2005 – Edits based on correspondence with Jim Polizotto of IMTC
-Removed the word Draft from the title page.
-Removed the watermark "DRAFT" from the document.

A.1v1.0.4: May 23, 2005 – Edits based on Last call period ending on May 20, 2005.
-Reworded section 2 and made reference to RFC 2119.
-Corrected grammar in section 3.1.3 and section 10.
-Added
Company names to author's names in Section 5.
Reference to RFC 2119 in section 6.
-Adjusted formatting in section 6.

A.1v1.0.3: May 4, 2005 – Edits based on conference call and e-mails
-Corrected spelling of Morris in section 9 of last version, section 10 this version.

-Added acknowledgement section and renumbered sections.
-Adjusted text spacing of figures in Appendix A.

A.1v1.0.2: April 28, 2005 – Edit based on contribution by Stuart Morris
-Removed
Bullet points from section 4.1
-Added note to section 4.1

A.1v1.0.1: April 20, 2005 – Edits based on conference call.
-Removed
Note and bullet points in section 4.1
Changed it's to its in 3rd paragraph of section 4.3

A.1v1.0: April 18, 2005 – Edits based on contributions by David Olson.
-Removed
Table from section 4.1
1st paragraph in section 4.2
1st sentence in paragraph 3 of section 4.3

-Revised
2nd bullet point in section 4.2

A.1v0.9.1: April 4, 2005 – Edits based on contributions by David Olson.
-Removed:
All tables from section 4.2
Syntax examples from section 4.2
Figure 1 PSTN-PSTN in section 7.0
Figure 8 VoIP-VoIP in section 7.0

-Revised
Added section 3.2.1
Table in section 4.1
Added additional paragraph in section 4.1
Added additional paragraph in section 4.2
Reformatted references in section 5.0

A.1v0.9: March 16, 2005 – Edits based on e-mails from Richard Stastny and Lawrence Conroy.
-Replaced reference to RFC 2806bis to RFC 3699.

A.1v0.8.7.1: January 11, 2005 – Additional edits based on December 8th, 2004 conference call
-Replaced figures 1 – 5 in Appendix A with 8 new figures.

A.1v08.7: December 17, 2004 – Edits based on December 8th conference call
-Revised
Changed sentence in 3rd paragraph Section 1 to include SIP URI
Fixed Spelling of UMMAP in paragraph 4, section 1.
Section 4.1 examples – Changed Uses to Use and Originates to Originate. Changed Translates to Originate a call to in third example.
Section 4.2 Put a period after Core Address Types in first sentence of paragraph and deleted rest of first sentence. Made changes to example table. Changed (i.e. H.323 to SIP) to (i.e. H.323 to SIP,H.323 to TEL and SIP to TEL). Deleted H.323 from first bullet point. Changed H.323/ SIP to an appropriate in the second bullet. Deleted H.323 to SIP from the third paragraph. Added to the end of fourth paragraph.
Section 4.3 Added + sign to TEL formatted number in 3rd paragraph.

A.1v0.8.2: December 3, 2004,October 21,2004- Edits based on October 15th conference call
-Added
Reference to Appendix A in section 4.0
Appendix A – flowcharts and diagrams of calls between and within protocols
-Revised
Changed reference to RFC2806bis-07 to RFC2806bis-09
Section numbers to reflect accuracy of Table of Contents
Section 3.0 with comments from Richard Stastny
Section 3.1.3 with comments from Richard Stastny
Section 4.1 with comments from Richard Stastny
Section 4.1 with flowchart from Dave Olson
Section 4.2 with new definition of tel format to SIP and H.323 from Alan Stillerman
-Removed
Appendix A and inserted text into section 1. Introduction

A.1v0.8: October 11, 2004 – Edits based on June 9th & 23rd & September 15 conference calls
-Added
Numbers to sections for easier reference
Updated Glossary w/ new version from Alan Stillerman
Appendix A

A.1v0.7: June 7, 2004 – Edits based on feedback and the 5/26/04 conference call
-Added
Examples to the Conformance section
Additional clarification to the Conformance section
A glossary of terms
-Revised reference to RFC2806 to be 2806bis
-Misc. editing and formatting

A.1v0.6: May 7, 2004 – Edits based on feedback
-Added
Reference to RFC3508, H.323 Uniform Resource Locator (URL) Scheme Registration
-Changed
Proscribe to prescribe
Compliant to conform

A.1v0.5: April 29, 2004 – Edits based on UMMAP review & 4/28/04 conference call
-Removed
ENUM as a core address type
-Added
Clarification of user interface vs. address representation
Clarified E.164 addresses must be globally unique or include tags per RFC2806
Clarified Address Representation Conformance section

A.1v0.4: April 22, 2004
-Changed
Contact information updated.
Fixed various spelling and syntax errors

A.1v0.3: April 20, 2004 – Edits based on UMMAP AG conference call 4/14/04 & editorial corrections
-Changed
'Specification' to 'Recommendation'
Fixed various spelling and syntax errors
-Added
Modem and Fax URI to the Tel URI section under Core Address Types
Paragraph at the end of Core Address Types-Tel URI regarding what happens if the ENUM lookup fails
Paragraph to Gateway Dialling regarding signalling and protocol translation
Clarification under User Input Conformance regarding address entry and storage
Clarification under Call Signalling Conformance regarding embedding of URIs
URLs for all references
Revision History

A.1v0.2: April 9, 2004 – Named document 'Specification A.1'

A.1v0.1: April 1, 2004 – Original document submitted by Tyler Johnson, UNC