Basic Network Media Services with SIP

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Abstract

   In SIP-based networks, there is a need to provide basic network media
   services. Such services include network announcements, user
   interaction, and conferencing services. These services are basic
   building blocks, from which one can construct interesting
   applications. In order to have interoperability between servers
   offering these building blocks (also known as Media Servers) and
   application developers, one needs to be able to locate and invoke
   such services in a well defined manner.

   This document describes a mechanism for providing an interoperable
   interface between Application Servers, which provide application
   services to SIP-based networks, and Media Servers, which provide the
   basic media processing building blocks.

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1. Overview

   In SIP-based media networks (RFC 3261 [10]), there is a need to
   provide basic network media services. Such services include playing
   announcements, initiating a media mixing session (conference), and
   prompting and collecting information with a user.

   These services are basic in nature, are few in number, and
   fundamentally have not changed in 25 years of enhanced telephony
   services. Moreover, given their elemental nature, one would not
   expect them to change in the future.

   Multifunction media servers provide network media services to clients
   using server protocols such as SIP, often in conjunction with markup
   languages such as VoiceXML [20] and MSCML [21]. This document
   describes how to identify to a multifunction media server what sort
   of session the client is requesting, without modifying the SIP
   protocol.

   It is critically important to note that the mechanism described here
   in no way modifies the SIP protocol, the meaning, or definition of a
   SIP Request URI, or does it put any restrictions, in any way, on
   devices that do not implement this convention.

Announcements are media played to the user. Announcements can be
static media files, media files generated in real-time, media streams
generated in real-time, multimedia objects, or combinations of the
above.

Media mixing is the act of mixing different RTP streams, as described
in RFC 3550 [13]. Note that the service described here suffices for
simple mixing of media for a basic conferencing service. This
service does not address enhanced conferencing services, such as
floor control, gain control, muting, subconferences, etc. MSCML [21]
addresses enhanced conferencing. However, that is beyond the scope
of this document. Interested readers should read conferencing-
framework [22] for details on the IETF SIP conferencing framework.

Prompt and collect is where the server prompts the user for some
information, as in an announcement, and then collects the user's
response. This can be a one-step interaction, for example by playing
an announcement, "Please enter your pass code", followed by
collecting a string of digits. It can also be a more complex
interaction, specified, for example, by VoiceXML [20] or MSCML [21].

1.1. Conventions Used in This Document

RFC 2119 [6] the interpretations for the key words "MUST", "MUST
NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT",
"RECOMMENDED", "MAY", and "OPTIONAL" found in this document.

2. Mechanism

In the context of SIP control of media servers, we take advantage of
the fact that the standard SIP URI has a user part. Multifunction
media servers do not have users. Thus we use the user address, or
the left-hand-side of the URI, as a service indicator.

The use of the user part of the SIP Request URI has a number of
useful properties:

o There is no change to core SIP.
o Only devices that choose to conform to this standard have to
   implement it.
o This document only applies to multifunction SIP-controlled media
   servers.
o This document has no impact on non-multifunction SIP-controlled
   media servers.
o The mechanism described in this document has absolutely no impact
   on SIP devices other than media servers.

The last bullet point is crucial. In particular, the user part
convention described here places absolutely no restrictions on any
SIP user agent, proxy, back-to-back user agent (B2BUA), or any future
device. The user parts defined here only apply to multifunction
media servers that chose to implement the convention. With the
exception of a conforming media server, these user names and
conventions have no impact on the user part namespace. They do not
restrict the use of these user names at devices other than a
multifunction media server.

Note that the set of services is small, well defined, and well
contained. The section The User Part (Section 7) discusses the
issues with using a fixed set of user-space names.

For per-service security, the media server SHOULD use the security
protocols described in RFC 3261 [10].

The media server MAY issue 401 challenges for authentication. The
media server SHOULD support the sips: scheme for the announcement
service. The media server MUST support the sips: scheme for the
dialog and conference services. The level of authentication to
require for each service is a matter of local policy.

The media server, upon receiving an INVITE, notes the service
indicator. Depending on the service indicator, the media server will
either honor the request or return a failure response code.

The service indicator is the concatenation of the service name and an
optional service instance identifier, separated by an equal sign.

Per RFC 3261 [10], the service indicator is case insensitive. The
service name MUST be from the set alphanumeric characters plus dash
(US-ASCII %2C). The service name MUST NOT include an equal sign
(US-ASCII %3D).

The service name MAY have long- and short-forms, as SIP does for
headers.

A given service indicator MAY have an associated set of parameters.
Such parameters MUST follow the convention set out for SIP URI
parameters. That is, a semi-colon separated list of keyword=value
pairs.

Certain services may have an association with a unique service
instance on the media server. For example, a given media server can
host multiple, separate conference sessions. To identify unique
service instances, a unique identifier modifies the service name.

The unique identifier MUST meet the rules for a legal user part of a
SIP URI. An equal sign, US-ASCII %3D, MUST separate the service
indicator from the unique identifier.

Note that since the service indicator is case insensitive, the
service instance identifier is also case insensitive.

The requesting client issues a SIP INVITE to the media server,
specifying the requested service and any appropriate parameters.

If the media server can perform the requested service, it does so,
following the processing steps described in the service definition
document.

If the media server cannot perform the requested service or does not
recognize the service indicator, it MUST respond with the response
code 488 NOT ACCEPTABLE HERE. This is appropriate, as 488 refers to
a problem with the user part of the URI. Moreover, 606 is not
appropriate, as some other media server may be able to satisfy the
request. RFC 3261 [10] describes the 488 and 606 response codes.

Some services require a unique identifier. Most services
automatically create a service instance upon the first INVITE with
the given identifier. However, if a service requires an existing
service instance, and no such service instance exists on the media
server, the media server MUST respond with the response code 404 NOT
FOUND. This is appropriate as the service itself exists on the media
server, but the particular service instance does not. It is as if
the user was not home.

3. Announcement Service

A network announcement is the delivery of a multimedia resource, such
as a prompt file, to a terminal device. Note the multimedia resource
may be any multimedia object that the media server supports. This
service can play a single object with multiple streams, such as a
video and audio prompt. However, this service cannot play multiple
objects on the same SIP dialog.

There are two types of network announcements. The differentiating
characteristic between the two types is whether the network fully
sets up the SIP dialog before playing the announcement. The analog
in the Public Switched Telephone Network (PSTN) is whether answer
supervision is supplied (i.e., does the announcement server answer
the call prior to delivering the announcement?).

Playing an announcement after call setup is straightforward. First,
the requesting device issues an INVITE to the media server requesting
the announcement service. The media server negotiates the SDP and
responds with a 200 OK. After receiving the ACK from the requesting
device, the media server plays the requested object and issues a BYE
to the requesting device.

If the media server supports announcements, but it cannot find the
referenced URI, it MUST respond with the 404 response code and SHOULD
send the reason phrase "Announcement content not found".

If the media server receives an INVITE for the announcement service
without a "play=" parameter, it MUST respond with the response code
400 and SHOULD send the reason phrase "Mandatory play parameter
missing".

If there is an error retrieving the announcement, the media server
MUST respond with a 400 response code and SHOULD send the reason
phrase "Announcement content could not be retrieved". In addition
the media server SHOULD include a Warning header with appropriate
explanatory text explaining what failed.

The Request URI fully describes the announcement service through the
use of the user part of the address and additional URI parameters.
The user portion of the address, "annc", specifies the announcement
service on the media server. The service has several associated URI
parameters that control the content and delivery of the announcement.

These parameters are described below:

play
   Specifies the resource or announcement sequence to be played.

repeat
   Specifies how many times the media server should repeat the
   announcement or sequence named by the "play=" parameter. The
   value "forever" means the repeat should be effectively unbounded.
   In this case, it is RECOMMENDED the media server implements some
   local policy, such as limiting what "forever" means, to ensure
   errant clients do not create a denial of service attack.

delay
   Specifies a delay interval between announcement repetitions. The
   delay is measured in milliseconds.

duration
   Specifies the maximum duration of the announcement. The media
   server will discontinue the announcement and end the call if the

   maximum duration has been reached. The duration is measured in
   milliseconds.

   locale
      Specifies the language and optionally country variant of the
      announcement sequence named in the "play=" parameter. RFC 3066
      [9] specifies the locale tag. The locale tag is usually a two- or
      three-letter code per ISO 639-1 [11]. The country variant is also
      often a two-letter code per ISO 3166-1 [12]. These elements are
      concatenated with a single under bar (%x5F) character, such as
      "en_CA". If only the language is specified, such as locale=en,
      the choice of country variant is an implementation matter.
      Implementations SHOULD provide the best possible match between the
      requested locale and the available languages in the event the
      media server cannot honor the locale request precisely. For
      example, if the request has locale=ca_FR, but the media server
      only has fr_FR available, the media server should use the fr_FR
      variant. Implementations SHOULD provide a default locale to use
      if no language variants are available.

   param[n]
      Provides a mechanism for passing values that are to be substituted
      into an announcement sequence. Up to 9 parameters ("param1="
      through "param9=") may be specified. The mechanics of
      announcement sequences are beyond the scope of this document.

   extension
      Provides a mechanism for extending the parameter set. If the
      media server receives an extension it does not understand, it MUST
      silently ignore the extension parameter and value.

   The "play=" parameter is mandatory and MUST be present. All other
   parameters are OPTIONAL.

   NOTE: Some encodings are not self-describing. Thus, the
   implementation relies on filename extension conventions for
   determining the media type.

   Note that RFC 3261 [10] implies that proxies are supposed to pass
   parameters through unchanged. However, be aware that non-conforming
   proxies may strip Request-URI parameters. That said, given the
   likely scenarios for the mechanisms presented in this document, this
   should not be an issue. Most likely, the proxy inserting the
   parameters is the last proxy before the media server. If the service
   provider deploys a proxy for load balancing or service location
   purposes, the service provider should ensure that its choice of proxy
   preserves parameters.

The form of the SIP Request URI for announcements is as follows.
Note that the backslash, CRLF, and spacing before the "play=" in the
example is for readability purposes only.

```
sip:annc@ms2.example.net; \
    play=http://audio.example.net/allcircuitsbusy.g711
```

```
sip:annc@ms2.example.net; \
    play=file://fileserver.example.net//geminii/yourHoroscope.wav
```

3.1. Operation

The scenarios below assume there is a SIP Proxy, application server,
or media gateway controller between the caller and the media server.
However, the announcement service works as described below even if
the caller invokes the service directly. We chose to discuss the
proxy case, as it will be the most common case.

The caller issues an INVITE to the serving SIP Proxy. The SIP Proxy
determines what audio prompt to play to the caller. The proxy
responds to the caller with 100 TRYING.

It is important to note that the mechanism described here in no way
modifies the behavior of SIP [10]. In particular, this convention
does not modify SDP negotiation [18].

The proxy issues an INVITE to the media server, requesting the
appropriate prompt to play coded in the play= parameter. The media
server responds with 200 OK. The proxy relays the 200 OK to the
caller. The caller then issues an ACK. The proxy then relays the
ACK to the media server.

With the call established, the media server plays the requested
prompt. When the media server completes the play of the prompt, it
issues a BYE to the proxy. The proxy then issues a BYE to the
caller.

3.2. Protocol Diagram

```
     Caller                   Proxy                 Media Server
        |     INVITE             |                        |
        |----------------------->|   INVITE               |
        |     100 TRYING         |----------------------->|
        |<-----------------------|   200 OK               |
        |     200 OK             |<-----------------------|
        |<-----------------------|                        |
        |     ACK                |                        |
        |----------------------->|   ACK                  |
        |                        |----------------------->|
        |                        |                        |
        |                        |                        |
        |             Play Announcement (RTP)             |
        |<================================================|
        |                        |                        |
        |                        |   BYE                  |
        |     BYE                |<-----------------------|
        |<-----------------------|                        |
        |     200 OK             |                        |
        |----------------------->|   200 OK               |
        |                        |----------------------->|
        |                        |                        |
```
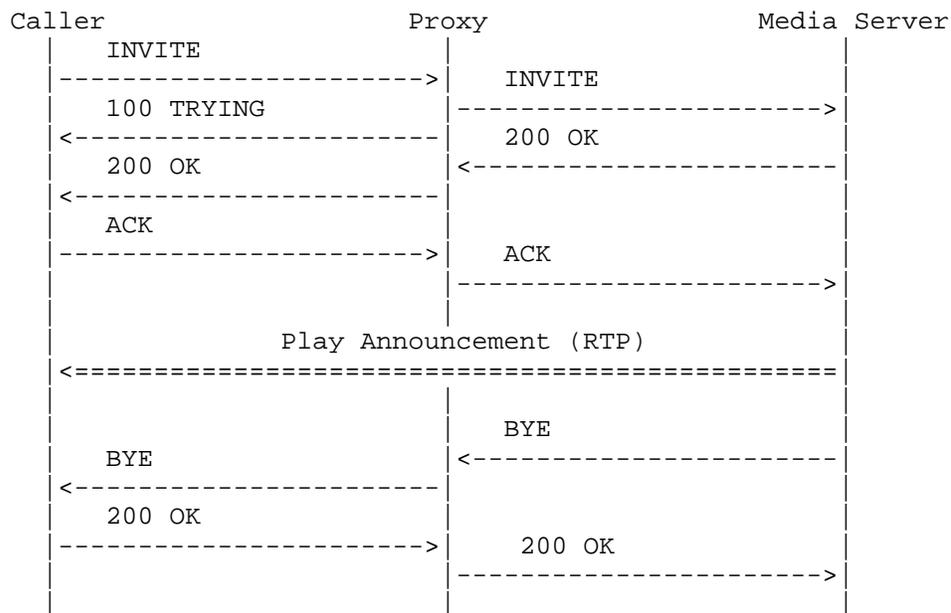
3.3. Formal Syntax

   The following syntax specification uses the augmented Backus-Naur
   Form (BNF) as described in RFC 4234 [7].

   ANNC-URL   = sip-ind annc-ind "@" hostport
                        annc-parameters uri-parameters

   sip-ind   = "sip:" / "sips:"
   annc-ind   = "annc"

   annc-parameters = ";" play-param [";" content-param]
                                    [";" delay-param]
                                    [";" duration-param]
                                    [";" repeat-param]
                                    [";" locale-param]
                                    [";" variable-params]
                                    [";" extension-params]

   play-param   = "play=" prompt-url

   content-param   = "content-type=" MIME-type

   delay-param   = "delay=" delay-value

```
delay-value      = 1*DIGIT

duration-param   = "duration=" duration-value

duration-value   = 1*DIGIT

repeat-param     = "repeat=" repeat-value

repeat-value     = 1*DIGIT / "forever"

locale-param     = "locale=" token
                        ; per RFC 3066, usually
                        ; ISO639-1_ISO3166-1
                        ; e.g., en, en_US, en_UK, etc.

variable-params = param-name "=" variable-value

param-name       = "param" DIGIT ; e.g., "param1"

variable-value   = 1*(ALPHA / DIGIT)

extension-params = extension-param [ ";" extension-params ]

extension-param  = token "=" token
```

"uri-parameters" is the SIP Request-URI parameter list as described
in RFC 3261 [10]. All parameters of the Request URI are part of the
URI matching algorithm.

The MIME-type is the MIME [1] [2] [3] [4] [5] content type for the
announcement, such as audio/basic, audio/G729, audio/mpeg,
video/mpeg, and so on.

A number of MIME registrations, which could be used here, have
parameters, for instance, video/DV. To accommodate this, and retain
compatibility with the SIP URI structure, the MIME-type parameter
separator (semicolon, %3b) and value separator (equal, %d3) MUST be
escaped. For example:

```
sip:annc@ms.example.net; \
    play=file://fs.example.net//clips/my-intro.dvi; \
    content-type=video/mpeg%3bencode%d3314M-25/625-50
```

The locale-value consists of a tag as specified in RFC 3066 [9].

The definition of hostport is as specified by RFC 3261 [10].

   The syntax of prompt-url consists of a URL scheme as specified by RFC
   3986 [8] or a special token indicating a provisioned announcement
   sequence. For example, the URL scheme MAY include any of the
   following.

   o http/https
   o ftp
   o file (referencing a local or NFS (RFC 3530 [16]) object)
   o nfs (RFC 2224 [14])

   If a provisioned announcement sequence is to be played, the value of
   prompt-url will have the following form:

   prompt-url   = "/provisioned/" announcement-id

   announcement-id = 1*(ALPHA / DIGIT)

   Note that the scheme "/provisioned/" was chosen because of a
   hesitation to register a "provisioned:" URI scheme.

   This document is strictly focused on the SIP interface for the
   announcement service and, as such, does not detail how announcement
   sequences are provisioned or defined.

   Note that the media type of the object the prompt-url refers to can
   be most anything, including audio file formats, text file formats, or
   URI lists. See the Prompt and Collect Service (Section 4) section
   for more on this topic.

4. Prompt and Collect Service

   This service is also known as a voice dialog. It establishes an
   aural dialog with the user.

   The dialog service follows the model of the announcement service.
   However, the service indicator is "dialog". The dialog service takes
   a parameter, voicexml=, indicating the URI of the VoiceXML script to
   execute.

   sip:dialog@mediaserver.example.net; \
       voicexml=http://vxmlserver.example.net/cgi-bin/script.vxml

   A Media Server MAY accept additional SIP request URI parameters and
   deliver them to the VoiceXML interpreter session as session
   variables.

   Although not good VoiceXML programming practice, VoiceXML scripts
   might contain sensitive information, such as a user's pass code in a

DTMF grammar. Thus, the media server MUST support the https scheme
for the voicexml parameter for secure fetching of scripts. Likewise,
dynamic grammars often do have user-identifying information. As
such, the VoiceXML browser implementation on the media server MUST
support https fetching of grammars and subsequent documents.

Returned information often is sensitive. For example, the
information could be financial information or instructions. Thus,
the media server MUST support https posting of results.

4.1. Formal Syntax for Prompt and Collect Service

The following syntax specification uses the augmented Backus-Naur
Form (BNF) as described in RFC 4234 [7].

```
DIALOG-URL         = sip-ind dialog-ind "@" hostport
                        dialog-parameters

sip-ind            = "sip:" / "sips:"
dialog-ind         = "dialog"

dialog-parameters  = ";" dialog-param [ vxml-parameters ]
                                      [ uri-parameters ]

dialog-param       = "voicexml=" vxml-url

vxml-parameters    = vxml-param [ vxml-parameters ]

vxml-param         = ";" vxml-keyword "=" vxml-value

vxml-keyword       = token

vxml-value         = token
```

The vxml-url is the URI of the VoiceXML script. If present, other
parameters get passed to the VoiceXML interpreter session with the
assigned vxml-keyword vxml-value pairs. Note that all vxml-keywords
MUST have values.

If there is a vxml-keyword without a corresponding vxml-value, the
media server MUST reject the request with a 400 BAD REQUEST response
code. In addition, the media server MUST state "Missing VXML Value"
in the reason phrase.

The media server presents the parameters as environment variables in
the connection object. Specifically, the parameter appears in the
connection.sip tree.

If the Media Server does not support the passing of keyword-value
pairs to the VoiceXML interpreter session, it MUST ignore the
parameters.

"uri-parameters" is the SIP Request-URI parameter list as described
in RFC 3261 [10]. All parameters in the parameter list, whether they
come from uri-parameters or from vxml-keyworks, are part of the URI
matching algorithm.

5. Conference Service

One identifies mixing sessions through their SIP request URIs. To
create a mixing session, one sends an INVITE to a request URI that
represents the session. If the URI does not already exist on the
media server and the requested resources are available, the media
server creates a new mixing session. If there is an existing URI for
the session, then the media server interprets it as a request for the
new session to join the existing session. The form of the SIP
request URI for conferencing is:

sip:conf=uniqueIdentifier@mediaserver.example.net

The left-hand side of the request URI is actually the username of the
request in the request URI and the To header. The host portion of
the URI identifies a particular media server. The "conf" user name
conveys to the media server that this is a request for the mixing
service. The uniqueIdentifier can be any value that is compliant
with the SIP URI specification. It is the responsibility of the
conference control application to ensure the identifier is unique
within the scope of any potential conflict.

In the terminology of the conferencing framework [22], this URI
convention tells the media server that the application server is
requesting it to act as a Focus. The conf-id value identifies the
particular focus instance.

As a focus in the conferencing framework, the media server MUST
support the ";isfocus" parameter in the Request URI. Note, however,
that the presence or absence of the ";isfocus" parameter has no
protocol impact at the media server.

It is worth noting that the conference URI shared between the
application and media servers provides enhanced security, as the SIP
control interface does not have to be exposed to participants. It
also allows the assignment of a specific media server to be delayed
as long as possible, thereby simplifying resource management.

One can add additional legs to the conference by INVITEing them to
the above-mentioned request URI. Per the matching rules of RFC 3261
[10], the conf-id parameter is part of the matching string.

Conversely, one can remove legs by issuing a BYE in the corresponding
dialog. The mixing session, and thus the conference-specific request
URI, remains active so long as there is at least one SIP dialog
associated with the given request URI.

If the Request-URI has "conf" as the user part, but does not have a
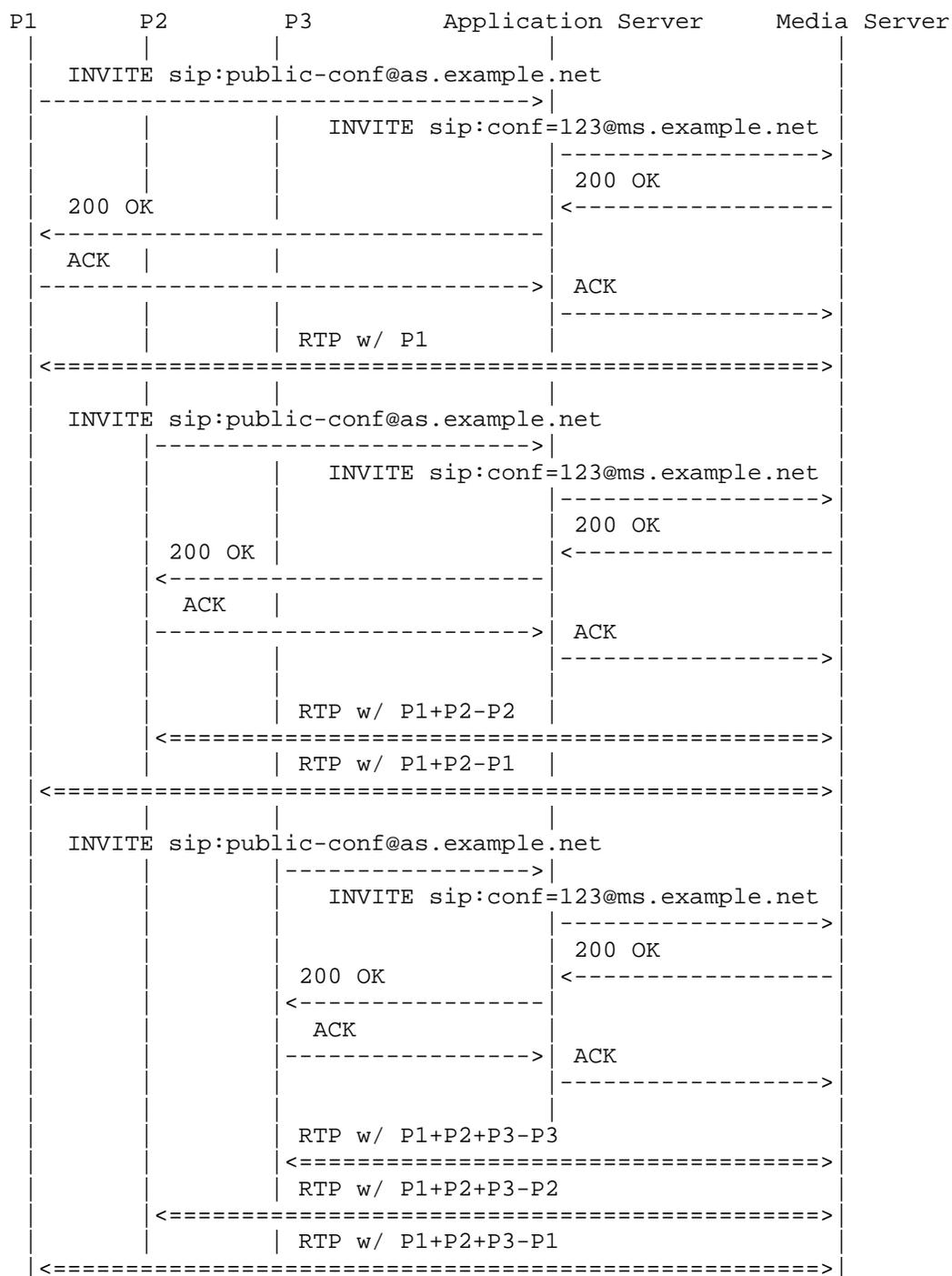conf-id parameter, the media server MUST respond with a 404 NOT
FOUND.

   NOTE: The media server could create a unique conference instance
   and return the conf-id string to the User Agent Clinet (UAC) if
   there is no conf-id present. However, such an operation may have
   other operational issues, such as permissions and billing. Thus
   an application server or proxy is a better place to do such an
   operation. Moreover, such action would make the media server into
   a Conference Factory in the terminology of conference-framework
   [22]. That is not the appropriate behavior for a media server.

Since some conference use cases, such as business conferencing, have
billing implications, the media server SHOULD authenticate the
application server or proxy. At a minimum, the media server MUST
implement sips:.

5.1. Protocol Diagram

This diagram shows the establishment of a three-way conference. This
section is informative. It is only one method of establishing a
conference. This example shows a simple back-to-back user agent.

The conference-framework [22] describes additional parameters and
behaviors of the Application Server. For example, the first INVITE
from P1 to the Application Server would include the ";isfocus"
parameter; the Application Server would act as a Conference Factory;
and so on. However, none of that protocol machinery has an impact on
the operation of the Application Server to Media Server interface,
which is the focus of this protocol document.

```
    P1        P2        P3        Application Server    Media Server
    |         |         |              |                    |
    |  INVITE sip:public-conf@as.example.net               |
    |-------------------------------->|                    |
    |         |         |   INVITE sip:conf=123@ms.example.net
    |         |         |              |----------------->|
    |         |         |              |  200 OK          |
    |  200 OK |         |              |<-----------------|
    |<--------------------------------|                    |
    |  ACK |  |         |              |                    |
    |-------------------------------->|  ACK               |
    |         |         |              |----------------->|
    |         |         |  RTP w/ P1   |                    |
    |<===================================================>|
    |         |         |              |                    |
    |  INVITE sip:public-conf@as.example.net               |
    |         |-------------------------->|                |
    |         |         |   INVITE sip:conf=123@ms.example.net
    |         |         |              |----------------->|
    |         |         |              |  200 OK          |
    |         |  200 OK |              |<-----------------|
    |         |<--------------------------|                |
    |         |   ACK   |              |                    |
    |         |-------------------------->|  ACK           |
    |         |         |              |----------------->|
    |         |         |              |                    |
    |         |         |  RTP w/ P1+P2-P2 |                |
    |         |<===================================================>|
    |         |         |  RTP w/ P1+P2-P1 |                |
    |<===================================================>|
    |         |         |              |                    |
    |  INVITE sip:public-conf@as.example.net               |
    |         |         |---------------->|                |
    |         |         |   INVITE sip:conf=123@ms.example.net
    |         |         |              |----------------->|
    |         |         |              |  200 OK          |
    |         |         |  200 OK      |<-----------------|
    |         |         |<-----------------|                |
    |         |         |   ACK        |                    |
    |         |         |---------------->|  ACK           |
    |         |         |              |----------------->|
    |         |         |              |                    |
    |         |         |  RTP w/ P1+P2+P3-P3               |
    |         |         |<===================================================>|
    |         |         |  RTP w/ P1+P2+P3-P2               |
    |         |<===================================================>|
    |         |         |  RTP w/ P1+P2+P3-P1               |
    |<===================================================>|
```

```
       |       |          |                    |                    |
       |       |          |                    |                    |
```

Using the terminology of conference-framework [22], the Application
Server is the Conference Factory, and the Media Server is the
Conference Focus.

Note that the above call flow does not show any 100 TRYING messages
that would typically flow from the Application Server to the UACs;
nor does it show the ACKs from the UACs to the Application Server or
from the Application Server to the Media Server.

Each leg can drop out either under the supervision of the UAC, by the
UAC sending a BYE, or under the supervision of the Application
Server, by the Application Server issuing a BYE. In either case, the
Application Server will either issue a BYE on behalf of the UAC or
issue it directly to the Media Server, corresponding to the
respective disconnect case.

It is left as a trivial exercise to the reader for how the
Application Server can mute legs, create side conferences, and so
forth.

Note that the Application Server is a server to the participants
(UACs). However, the Application Server is a client for mixing
services to the Media Server.

5.2. Formal Syntax

The following syntax specification uses the augmented Backus-Naur
Form (BNF) as described in RFC 4234 [7].

```
CONF-URL          = sip-ind conf-ind "=" instance-id "@" hostport
                      [ uri-parameters ]

sip-ind           = "sip:" / "sips:"

conf-ind          = "conf"

instance-id       = token
```

"uri-parameters" is the SIP Request-URI parameter list as described
in RFC 3261 [10]. All parameters in the parameter list are part of
the URI matching algorithm.

6. IANA Considerations

   The IANA has registered the following parameters in the SIP/SIPS URI
   Parameters registry, following the specification required policy of
   RFC 3969 [19]:

   Parameter Name   Predefined Values   Reference
   --------------   -----------------   ---------
   play   no   RFC 4240
   repeat   no   RFC 4240
   delay   no   RFC 4240
   duration   no   RFC 4240
   locale   no   RFC 4240
   param[n]   no   RFC 4240
   extension   no   RFC 4240

7. The User Part

   There has been considerable discussion about the wisdom of using
   fixed user parts in a request URI. The most common objection is that
   the user part should be opaque and a local matter. The other
   objection is that using a fixed user part removes those specified
   user addresses from the user address space.

   We address the latter issue first. The common example is the
   Postmaster address defined by RFC 2821 [15]. The objection is that
   by using the Postmaster token for something special, one removes that
   token for anyone. Thus, the Postmaster General of the United States,
   for example, cannot have the mail address Postmaster@usps.gov.
   However, one may debate whether this is a significant limitation.

   This document explicitly addresses this issue. The user names
   described in the text (namely annc, ivr, dialog, and conf) are
   available for whatever local use a given SIP user agent or proxy
   wishes for them. What this document does is give special meaning for
   these user names at media servers that implement this specification.
   If a media server chooses not to implement this specification,
   nothing breaks. If a user wishes to use one of the user names
   described in this document at their SIP user agent, nothing breaks
   and their user agent will work as expected.

   The key point is, one cannot confuse the namespace at a Media Server
   with the namespace for an organization. For example, let us take the
   case where a network offers services for "Ann Charles". She likes to
   use the name "annc", and thus she would like to use
   "sip:annc@example.net". We offer there is ABSOLUTELY NO NAME
   COLLISION WHATSOEVER. Why is this so? This is so because
   sip:annc@example.net will resolve to the specific user at a specific

device for Ann. As an example, example.net's SIP Proxy Server
resolves sip:annc@example.net to annc@anns-phone.example.net.
Conversely, one directs requests for the media service annc directly
to the Media Server, e.g., sip:annc@ms21.ap.example.net. Moreover,
by definition, requests for Ann Charles, or anything other than the
announcement service, will NEVER be directly sent to the Media
Server. If that were not true, no phone in the world could use the
user part "eburger", as eburger is a reserved user part in the
Brooktrout domain. Clearly, this is not the case.

If one wishes to make their media server accessible to the global
Internet, but retain one of the Media Server-specific user names in
the domain, a SIP Proxy can easily translate whatever opaque name one
chooses to the Media Server-specific user name. For example, if a
domain wishes to offer services for the above mentioned Ann Charles
at sip:annc@example.com, they can offer the announcement service at
sip:my-special-announcement-service@example.com. The former address,
sip:annc@example.com, would resolve to the actual device where annc
resides. The latter would resolve to the media server announcement
server address, sip:annc@mediaserver.example.com, as an example.
Note that this convention makes it easier to provision this service.
With a fixed mapping at the multifunction media server, there are
less provisioning data elements to get wrong.

Here is another way of looking at this issue. Unix reserves the
special user "root". Just about all Unix machines have a user root,
who has an address "root@a-specific-machine.example.com", where
"a-specific-machine" is the fully-qualified domain name (FQDN) of a
particular instance of a machine. There are very well-defined
semantics for the "root" user.

Even though most every Unix machine has a "root" user, often there is
no mapping for a "root" user in a domain, such as "root@example.com".
Conversely, there is no restriction on creating an MX record for
"root@example.com". That choice is fully up to the administrative
authority for the domain.

The "users" proposed by this document, "annc", "conf", and "dialog"
are all users at a Media Server, just as the "root", "bin", and
"nobody" users are "users" at a Unix host.

After much discussion, with input from the W3C URI work group, we
considered obfuscating the user name by prepending "__sip-" to the
user name. However, as explained above, this obfuscation is not
necessary. There is a fundamental difference between a user name at
a device and a user name at an MX record (SMTP) or Address-of-Record
(SIP). Again, there is no possibility that the name on the device
may "leak out" into the SIP routing network.

The most important thing to note about this convention is that the
left-hand side of the request URI is opaque to the network. The only
network elements that need to know about the convention are the Media
Server and client. Even proxies doing mapping resolution, as in the
example above for public announcement services, do not need to be
aware of the convention. The convention is purely a matter of
provisioning.

Some have proposed that such naming be a pure matter of local
convention. For example, the thesis of the informational RFC RFC
3087 [17] is that you can address services using a request URI.
However, some have taken the examples in the document to an extreme.
Namely, that the only way to address services is via arbitrary,
opaque, long user parts. Clearly, it is possible to provision the
service names, rather than fixed names. While this can work in a
closed network, where the Application Servers and Media Servers are
in the same administrative domain, this does not work across domains,
such as in the Internet. This is because the client of the media
service has to know the local name for each service / domain pair.
This is particularly onerous for situations where there is an ad hoc
relationship between the application and the media service. Without
a well-known relationship between service and service address, how
would the client locate the service?

One very important result of using the user part as the service
descriptor is that we can use all of the standard SIP machinery,
without modification. For example, Media Servers with different
capabilities can SIP Register their capabilities as users. For
example, a VoiceXML-only device will register the "dialog" user,
while a multi-purpose Media Server will register all of the users.
Note that this is why the URI to play is a parameter. Doing
otherwise would overburden a normal SIP proxy or redirect server.
Conversely, having the conference ID be part of the user part gives
an indication that requests get routed similarly (as opposed to
requiring a Globally Routable User Agent URI (GRUU), which would
restrict routing to the same device).

Likewise, this scheme lets us leverage the standard SIP proxy
behavior of using an intelligent redirect server or proxy server to
provide high-available services. For example, two Media Servers can
register with a SIP redirect server for the annc user. If one of the
Media Servers fails, the registration will expire and all requests
for the announcement service ("calls to the annc user") will get sent
to the surviving Media Server.

8. Security Considerations

   Exposing network services with well-known addresses may not be
   desirable. The Media Server SHOULD authenticate and authorize
   requesting endpoints per local policy.

   Some interactions in this document result in the transfer of
   confidential information. Moreover, many of the interactions require
   integrity protection. Thus, the Media Server MUST implement the
   sips: scheme. In addition, application developers are RECOMMENDED to
   use the security services offered by the Media Server to ensure the
   integrity and confidentiality of their user's data, as appropriate.

   Untrusted network elements could use the convention described here
   for providing information services. Many extant billing arrangements
   are for completed calls. Successful call completion occurs with a
   2xx result code. This can be an issue for the early media
   announcement service. This is one of the reasons why the early media
   announcement service is deprecated.

   Services such as repeating an announcement forever create the
   possibility for denial of service attacks. The media server SHOULD
   have local policies to deal with this, such as time-limiting how long
   "forever" is, analyzing where multiple requests come from,
   implementing white-lists for such a service, and so on.

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   of whom are listed in the Acknowledgements section. All I did was do
   the theory and write it up. That also means all of the mistakes are
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Note that at the time of this writing, there are 7 known independent
server implementations that are interoperable with 23 known client
implementations. Our apologies if we did not count your
implementation.

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