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A Call Control and Multi-party usage framework for the Session Initiation Protocol (SIP) draft-ietf-sipping-cc-framework-05.txt

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Abstract

This document defines a framework and requirements for multi-party usage of SIP. To enable discussion of multi-party features and applications we define an abstract call model for describing the media relationships required by many of these. The model and actions described here are specifically chosen to be independent of the SIP signaling and/or mixing approach chosen to actually setup the media relationships. In addition to its dialog manipulation aspect, this framework includes requirements for communicating related information and events such as conference and session state, and session history. This framework also describes other goals which embody the spirit of SIP applications as used on the Internet.

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1. Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC-2119 [2].

2. Motivation and Background

The Session Initiation Protocol [1] (SIP) was defined for the initiation, maintenance, and termination of sessions or calls between one or more users. However, despite its origins as a large-scale multiparty conferencing protocol, SIP is used today primarily for point to point calls. This two-party configuration is the focus of the SIP specification and most of its extensions.

This document defines a framework and requirements for multi-party usage of SIP. Most multi-party operations manipulate SIP session dialogs (also known as call legs) or SIP conference media policy to cause participants in a conversation to perceive specific media relationships. In other protocols that deal with the concept of calls, this manipulation is known as call control. In addition to its dialog or policy manipulation aspect, "call control" also includes communicating information and events related to manipulating calls, including information and events dealing with session state and history, conference state, user state, and even message state.

Based on input from the SIP community, the authors compiled the following set of goals for SIP call control and multiparty applications:

- Define Primitives, Not Services. Allow for a handful of robust yet simple mechanisms which can be combined to deliver features and services. Throughout this document we refer to these simple mechanisms as "primitives". Primitives should be sufficiently robust that when they are combined they can be used to build lots of services. However, the goal is not to define a provably complete set of primitives. Note that while the IETF will NOT standardize behavior or services, it may define example services for informational purposes, as in service examples [6].
- Participant oriented. The primitives should be designed to provide services which are oriented around the experience of the participants. The authors observe that end users of features and services usually don't care how a media relationship is setup. Their ultimate experience is based only on the resulting media and other externally visible characteristics.
- Signaling Model independent: Support both a central control and a peer-to-peer feature invocation model (and combinations of the two). Baseline SIP already supports a centralized control model described in [3pcc], and the SIP community has expressed a great

deal of interest in peer-to-peer or distributed call control using primitives such as those defined in REFER [8], Replaces [9], and Join [10].

- Mixing Model independent: The bulk of interesting multiparty applications involve mixing or combining media from multiple participants. This mixing can be performed by one or more of the participants, or by a centralized mixing resource. The experience of the participants should not depend on the mixing model used. While most examples in this document refer to audio mixing, the framework applies to any media type. In this context a "mixer" refers to combining media in an appropriate, media-specific way. This is consistent with model described in the SIP conferencing framework.
- Invoker oriented. Only the user who invokes a feature or a service needs to know exactly which service is invoked or why. This is good because it allows new services to be created without requiring new primitives from all the participants; and it allows for much simpler feature authorization policies, for example, when participation spans organizational boundaries. As discussed in section 3.8, this also avoids exponential state explosion when combining features. The invoker only has to manage a user interface or API to prevent local feature interactions. All the other participants simply need to manage the feature interactions of a much smaller number of primitives.
- Primitives make full use of URIs. URIs are a very powerful mechanism for describing users and services. They represent a plentiful resource which can be extremely expressive and easily routed, translated, and manipulated--even across organizational boundaries. URIs can contain special parameters and informational headers which need only be relevant to the owner of the namespace (domain) of the URI. Just as a user who selects an http: URL need not understand the significance and organization of the web site it references, a user may encounter a SIP URL which translates into an email-style group alias, which plays a pre-recorded message, or runs some complex call-handling logic. Note that while this may seem paradoxical to the previous goal, both goals can be satisfied by the same model.
- Make use of SIP headers and SIP event packages to provide SIP entities with information about their environment. These should include information about the status / handling of dialogs on other user agents, information about the history of other contacts attempted prior to the current contact, the status of participants, the status of conferences, user presence information, and the status of messages.
- Encourage service decomposition, and design to make use of standard components using well-defined, simple interfaces. Sample components include a SIP mixer, recording service, announcement server, and voice dialog server. (This is not an exhaustive

list).

- o Include authentication, authorization, policy, logging, and accounting mechanisms to allow these primitives to be used safely among mutually untrusted participants. Some of these mechanisms may be used to assist in billing, but no specific billing system will be endorsed.
- o Permit graceful fallback to baseline SIP. Definitions for new SIP call control extensions/primitives MUST describe a graceful way to fallback to baseline SIP behavior. Support for one primitive MUST NOT imply support for another primitive.
- o There is no desire or goal to reinvent traditional models, such as the model used the [H.450] family of protocols, [JTAPI], or the [CSTA] call model, as these other models do not share the design goals presented in this document.

3. Key Concepts

3.1 "Conversation Space" Model

This document introduces the concept of an abstract "conversation space" (essentially as a set of participants who believe they are all communicating among one another). Each conversation space contains one or more participants.

Participants are SIP User Agents which send original media to or terminate and receive media from other members of the conversation space. Logically, every participant in the conversation space has access to all the media generated in that space (this is strictly true if all participants share a common media type). A SIP User Agent which does not contribute or consume any media is NOT a participant; nor is a user agent which merely forwards, transcodes, mixes, or selects media originating elsewhere in the conversation space. [Note that a conversation space consists of zero or more SIP calls or SIP conferences. A conversation space is similar to the definition of a "call" in some other call models.]

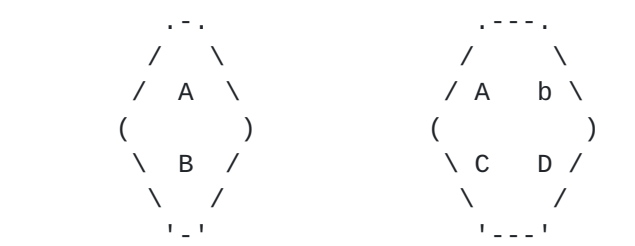
Participants may represent human users or non-human users (referred to as robots or automatons in this document). Some participants may be hidden within a conversation space. Some examples of hidden participants include: robots which generate tones, images, or announcements during a conference to announce users arriving and departing, a human call center supervisor monitoring a conversation between a trainee and a customer, and robots which record media for training or archival purposes.

Participants may also be active or passive. Active participants are expected to be intelligent enough to leave a conversation space when they no longer desire to participate. (An attentive human

participant is obviously active.) Some robotic participants (such as a voice messaging system, an instant messaging agent, or a voice dialog system) may be active participants if they can leave the conversation space when there is no human interaction. Other robots (for example our tone generating robot from the previous example) are passive participants. A human participant "on-hold" is passive.

An example diagram of a conversation space can be shown as a "bubble" or ovals, or as a "set" in curly or square brace notation. Each set, oval, or "bubble" represents a conversation space. Hidden participants are shown in lowercase letters.

{ A , B } [A , B]



3.2 Comparison with Related Definitions

In SIP, a call is "an informal term that refers to some communication between peers, generally set up for the purposes of a multimedia conversation." Obviously we cannot discuss normative behavior based on such an intentionally vague definition. The concept of a conversation space is needed because the SIP definition of call is not sufficiently precise for the purpose of describing the user experience of multiparty features.

Do any other definitions convey the correct meaning? SIP, and SDP [5] both define a conference as "a multimedia session identified by a common session description." A session is defined as "a set of multimedia senders and receivers and the data streams flowing from senders to receivers." Both of these definitions are heavily oriented toward multicast sessions with little differenciation among participants. As such, neither is particularly useful for our purposes. In fact, the definition of "call" in some call models is more similar to our definition of a conversation space.

Some examples of the relationship between conversation spaces, SIP call legs, and SIP sessions are listed below. In each example, a human user will perceive that there is a single call.

- A simple two-party call is a single conversation space, a single session, and a single call-leg.
- A locally mixed three-way call is two sessions and two call-legs. It is also a single conversation space.
- A simple dial-in audio conference is a single conversation space, but is represented by as many call-legs and sessions as there are human participants.
- A multicast conference is a single conversation space, a single session, and as many call-legs as participants.

3.3 Signaling Models

Obviously to make changes to a conversation space, you must be able to use SIP signaling to cause these changes. Specifically there must be a way to manipulate SIP dialogs (call legs) to move participants into and out of conversation spaces. Although this is not as obvious, there also must be a way to manipulate SIP dialogs to include non-participant user agents which are otherwise involved in a conversation space (ex: B2BUAs, 3pcc controllers, mixers, transcoders, translators, or relays).

Implementations may setup the media relationships described in the conversation space model using the approach described in 3pcc [7]. The 3pcc approach relies on only the following 3 primitive operations:

- Create a new call-leg (INVITE)
- Modify a call-leg (reINVITE)
- Destroy a call-leg (BYE)

The main advantage of the 3pcc approach is that it only requires very basic SIP support from end systems to support call control features. As such, third-party call control is a natural way to handle protocol conversion and mid-call features. It also has the advantage and disadvantage that new features can/must be implemented in one place only (the controller), and neither requires enhanced client functionality, nor takes advantage of it.

In addition, a peer-to-peer approach is discussed at length in this draft. The primary drawback of the peer-to-peer model is additional end system complexity. The benefits of the peer-to-peer model include:

- state remains at the edges
- call signaling need only go through participants involved (there are no additional points of failure)
- peers can take advantage of end-to-end message integrity or encryption

- setup time is shorter (fewer messages and round trips are required)

The peer-to-peer approach relies on additional "primitive" operations, some of which are identified here.

- Replace an existing dialog
- Join a new dialog with an existing dialog
- Support SIP conference policy control
- Locally perform media forking (multi-unicast)
- Ask another UA to send a request on your behalf

Many of the features, primitives, and actions described in this document also require some type of media mixing, combining, or selection as described in the next section.

3.4 Mixing Models

SIP permits a variety of mixing models, which are discussed here briefly. This topic is discussed more thoroughly in the SIP conferencing framework [15] and cc-conferencing [19]. SIP supports both tightly-coupled and loosely-coupled conferencing, although more sophisticated behavior is available in tightly-coupled conferences. In a tightly-coupled conference, a single SIP user agent (called the focus) has a direct dialog relationship with each participant (and may control non participant user agents as well). In a loosely-coupled conference there is no coordinated signaling relationships among the participants.

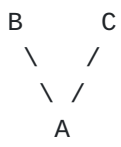
For brevity, only the two most popular conferencing models are significantly discussed in this document (local and centralized mixing). Applications of the conversation spaces model to loosely-coupled multicast and distributed full unicast mesh conferences are left as an exercise for the reader. Note that a distributed full mesh conference can be used for basic conferences, but does not easily allow for more complex conferencing actions like splitting, merging, and sidebars.

Call control features should be designed to allow a mixer (local or centralized) to decide when to reduce a conference back to a 2-party call, or drop all the participants (for example if only two automatons are communicating). The actual heuristics used to release calls are beyond the scope of this document, but may depend on properties in the conversation space, such as the number of active, passive, or hidden participants; and the send-only, receive-only, or send-and-receive orientation of various participants.

3.4.1 Tightly Coupled

3.4.1.1 (Single) End System Mixing

The first model we call "end system mixing". In this model, user A calls user B, and they have a conversation. At some point later, A decides to conference in user C. To do this, A calls C, using a completely separate SIP call. This call uses a different Call-ID, different tags, etc. There is no call set up directly between B and C. No SIP extension or external signaling is needed. A merely decides to locally join two call-legs.



A receives media streams from both B and C, and mixes them. A sends a stream containing A's and C's streams to B, and a stream containing A's and B's streams to C. Basically, user A handles both signaling and media mixing.

3.4.1.2 Centralized Mixing

In a centralized mixing model, all participants have a pairwise SIP and media relationship with the mixer. Common applications of centralized mixing include ad-hoc conferences and scheduled dial-in or dial-out conferences. [need diagram]

3.4.1.3 Centralized Signaling, Distributed Media

In this conferencing model, there is a centralized controller, as in the dial-in and dial-out cases. However, the centralized server handles signaling only. The media is still sent directly between participants, using either multicast or multi-unicast. Multi-unicast is when a user sends multiple packets (one for each recipient, addressed to that recipient). This is referred to as a "Decentralized Multipoint Conference" in [H.323].

3.4.2 Loosely Coupled

In these models, there is no point of central control of SIP signaling. As in the "Centralized Signaling, Distributed Media" case above, all endpoints send media to all other endpoints. Consequently every endpoint mixes their own media from all the other sources, and sends their own media to every other participant. [add diagrams]

3.4.2.1 Large-Scale Multicast Conferences

Large-scale multicast conferences were the original motivation for both the Session Description Protocol [SDP] and SIP. In a large-scale multicast conference, one or more multicast addresses are allocated to the conference. Each participant joins that multicast groups, and sends their media to those groups. Signaling is not sent to the multicast groups. The sole purpose of the signaling is to inform participants of which multicast groups to join. Large-scale multicast conferences are usually pre-arranged, with specific start and stop times. However, multicast conferences do not need to be pre-arranged, so long as a mechanism exists to dynamically obtain a multicast address.

3.4.2.2 Full Distributed Unicast Conferencing

In this conferencing model, each participant has both a pairwise media relationship and a pairwise SIP relationship with every other participant (a full mesh). This model requires a mechanism to maintain a consistent view of distributed state across the group. This is a classic hard problem in computer science. Also, this model does not scale well for large numbers of participants. because for <n> participants the number of media and SIP relationships is approximately n-squared. As a result, this model is not generally available in commercial implementations; to the contrary it is primarily the topic of research or experimental implementations. Note that this model assumes peer-to-peer signaling.

3.5 Conveying Information and Events

Participants should have access to information about the other participants in a conversation space, so that this information can be rendered to a human user or processed by an automaton. Although some of this information may be available from the Request-URI or To, From, Contact, or other SIP headers, another mechanism of reporting this information is necessary.

Many applications are driven by knowledge about the progress of calls and conferences. In general these types of events allow for the construction of distributed applications, where the application requires information on session dialog and conference state, but is not necessarily co-resident with an endpoint user agent or conference server. For example, a focus involved in a conversation space may wish to provide URLs for conference status, and/or conference/floor control.

The SIP Events [4] architecture defines general mechanisms for subscription to and notification of events within SIP networks. It

introduces the notion of a package which is a specific "instantiation" of the events mechanism for a well-defined set of events.

Event packages are needed to provide the status of a user's session dialogs, provide the status of conferences and its participants, provide user presence information, provide the status of registrations, and provide the status of user's messages. While this is not an exhaustive list, these are sufficient to enable the sample features described in this document.

The conference event package [12] allows users to subscribe to information about an entire tightly-coupled SIP conference. Notifications convey information about the pariticipants such as: the SIP URL identifying each user, their status in the space (active, declined, departed), URLs to invoke other features (such as sidebar conversations), links to other relevant information (such as floor control policies), and if floor control policies are in place, the user's floor control status. For conversation spaces created from cascaded conferences, converstation state can be gathered from relevant foci and merged into a cohesive set of state.

The session dialog package [11] provides information about all the dialogs the target user is maintaining, what conversations the user in participating in, and how these are correlated. Likewise the registration package [13] provides notifications when contacts have changed for a specific address-of-record. The combination of these allows a user agent to learn about all conversations occurring for the entire registered contact set for an address-of-record.

Note that user presence in SIP [14] has a close relationship with these later two event packages. It is fundamental to the presence model that the information used to obtain user presence is constructed from any number of different input sources. Examples of other such sources include calendaring information and uploads of presence documents. These two packages can be considered another mechanism that allows a presence agent to determine the presence state of the user. Specifically, a user presence server can act as a subscriber for the session dialog and registration packages to obtain additional information that can be used to construct a presence document.

The multi-party architecture may also need to provide a mechanism to get information about the status /handling of a dialog (for example, information about the history of other contacts attempted prior to the current contact). Finally, the architecture should provide ample opportunities to present informational URIs which relate to calls, conversations, or dialogs in some way. For example, consider the SIP

Call-Info header, or Contact headers returned in a 300-class response. Frequently additional information about a call or dialog can be fetched via non-SIP URIs. For example, consider a web page for package tracking when calling a delivery company, or a web page with related documentation when joining a dial-in conference. The use of URIs in the multiparty framework is discussed in more detail in Section 3.7.

Finally the interaction of SIP with stimulus-signaling-based applications, which allow a user agent to interact with an application without knowledge of the semantics of that application, is discussed in the SIP application interaction framework [16]. Stimulus signaling can occur to a user interface running locally with the client, or to a remote user interface, through media streams. Stimulus signaling encompasses a wide range of mechanisms, ranging from clicking on hyperlinks, to pressing buttons, to traditional Dual Tone Multi Frequency (DTMF) input. In all cases, stimulus signaling is supported through the use of markup languages, which play a key role in that framework.

3.6 Componentization and Decomposition

This framework proposes a decomposed component architecture with a very loose coupling of services and components. This means that a service (such as a conferencing server or an auto-attendant) need not be implemented as an actual server. Rather, these services can be built by combining a few basic components in straightforward or arbitrarily complex ways.

Since the components are easily deployed on separate boxes, by separate vendors, or even with separate providers, we achieve a separation of function that allows each piece to be developed in complete isolation. We can also reuse existing components for new applications. This allows rapid service creation, and the ability for services to be distributed across organizational domains anywhere in the Internet.

For many of these components it is also desirable to discover their capabilities, for example querying the ability of a mixer to host a 10 dialog conference, or to reserve resources for a specific time. These actions could be provided in the form of URLs, provided there is an a priori means of understanding their semantics. For example if there is a published dictionary of operations, a way to query the service for the available operations and the associated URLs, the URL can be the interface for providing these service operations. This concept is described in more detail in the context of dialog operations in section

3.6.1 Media Intermediaries

Media Intermediaries are not participants in any conversation space, although an entity which is also a media translator may also have a colocated participant component (for example a mixer which also announces the arrival of a new participant; the announcement portion is a participant, but the mixer itself is not). Media intermediaries should be as transparent as possible to the end users--offering a useful, fundamental service; without getting in the way of new features implemented by participants. Some common media intermediaries are desribed below.

3.6.2 Mixer

A SIP mixer is a component that combines media from all dialogs in the same conversation in a media specific way. For example, the default combining for an audio conference might be an N-1 configuration, while a text mixer might interleave text messages on a per-line basis. More details about how to manipulate the media policy used by mixers is being discussed in the XCON Working Group.

3.6.3 Transcoder

A transcoder translates media from one encoding or format to another (for example, GSM voice to G.711, MPEG2 to H.261, or text/html to text/plain), or from one media type to another (for example text to speech). A more thorough discussion of transcoding is described in SIP transcoding services invocation [17].

3.6.4 Media Relay

A media relay terminates media and simply forwards it to a new destination without changing the content in any way. Sometimes media relays are used to provide source IP address anonymity, to facilitate middlebox traversal, or to provide a trusted entity where media can be forcefully disconnected.

3.6.5 Queue Server

A queue server is a location where calls can be entered into one of several FIFO (first-in, first-out) queues. A queue server would subscribe to the presence of groups or individuals who are interested in its queues. When detecting that a user is available to service a queue, the server redirects or transfers the last call in the relevant queue to the available user. On a queue-by-queue basis, authorized users could also subscribe to the call state (dialog information) of calls within a queue. Authorized users could use this information to effectively pluck (take) a call out of the queue

(for example by sending an INVITE with a Replaces header to one of the user agents in the queue).

3.6.6 Parking Place

A parking place is a location where calls can be terminated temporarily and then retrieved later. While a call is "parked", it can receive media "on-hold" such as music, announcements, or advertisements. Such a service could be further decomposed such that announcements or music are handled by a separate component.

3.6.7 Announcements and Voice Dialogs

An announcement server is a server which can play digitized media (frequently audio), such as music or recorded speech. These servers are typically accessible via SIP, HTTP, or RTSP. An analogous service is a recording service which stores digitized media. A convention for specifying announcements in SIP URIs is described in [netann]. Likewise the same server could easily provide a service which records digitized media.

A "voice dialog" is a model of spoken interactive behavior between a human and an automaton which can include synthesized speech, digitized audio, recognition of spoken and DTMF key input, recording of spoken input, and interaction with call control. Voice dialogs frequently consist of forms or menus. Forms present information and gather input; menus offer choices of what to do next.

Spoken dialogs are a basic building block of applications which use voice. Consider for example that a voice mail system, the conference-id and passcode collection system for a conferencing system, and complicated voice portal applications all require a voice dialog component.

3.6.7.1 Text-to-Speech and Automatic Speech Recognition

Text-to-Speech (TTS) is a service which converts text into digitized audio. TTS is frequently integrated into other applications, but when separated as a component, it provides greater opportunity for broad reuse. Automatic Speech Recognition (ASR) is a service which attempts to decipher digitized speech based on a proposed grammar. Like TTS, ASR services can be embedded, or exposed so that many applications can take advantage of such services. A standardized (decomposed) interface to access standalone TTS and ASR services is currently being developed in the SPEECHSC Working Group.

3.6.7.2 VoiceXML

[VoiceXML] is a W3C recommendation that was designed to give authors control over the spoken dialog between users and applications. The application and user take turns speaking: the application prompts the user, and the user in turn responds. Its major goal is to bring the advantages of web-based development and content delivery to interactive voice response applications. We believe that VoiceXML represents the ideal partner for SIP in the development of distributed IVR servers. VoiceXML is an XML based scripting language for describing IVR services at an abstract level. VoiceXML supports DTMF recognition, speech recognition, text-to-speech, and playing out of recorded media files. The results of the data collected from the user are passed to a controlling entity through an HTTP POST operation. The controller can then return another script, or terminate the interaction with the IVR server.

A VoiceXML server also need not be implemented as a monolithic server. Below is a diagram of a VoiceXML browser which is split into media and non-media handling parts. The VoiceXML interpreter handles SIP dialog state and state within a VoiceXML document, and sends requests to the media component over another protocol.

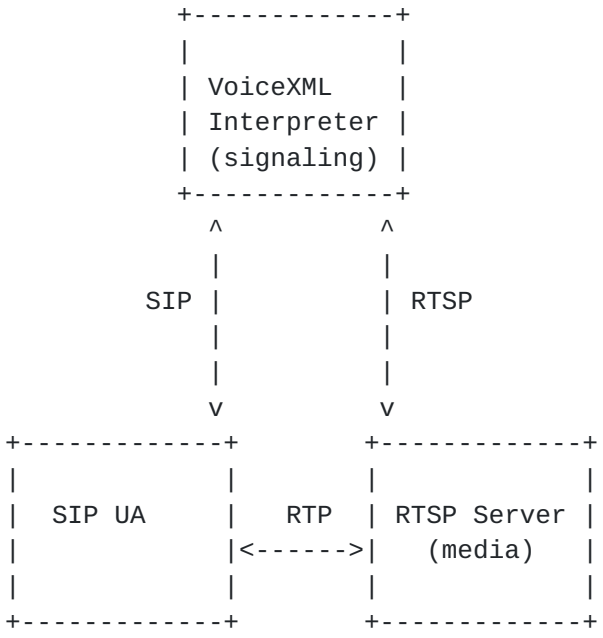


Figure : Decomposed VoiceXML Server

3.7 Use of URIs

All naming in SIP uses URIs. URIs in SIP are used in a plethora of contexts: the Request-URI; Contact, To, From, and *-Info headers; application/uri bodies; and embedded in email, web pages, instant messages, and ENUM records. The request-URI identifies the user or service that the call is destined for.

SIP URIs embedded in informational SIP headers, SIP bodies, and non-SIP content can also specify methods, special parameters, headers, and even bodies. For example:

```
sip:bob@babylon.biloxi.com;method=BYE?Call-ID=13413098
  &To=<sip:bob@biloxi.com>;tag=879738
  &From=<sip:alice@atlanta.com>;tag=023214

sip:bob@babylon.biloxi.com;method=REFER?
  Refer-To=<http://www.atlanta.com/~alice>
```

Throughout this draft we discuss call control primitive operations. One of the biggest problems is defining how these operations may be invoked. There are a number of ways to do this. One way is to define the primitives in the protocol itself such that SIP methods (for example REFER) or SIP headers (for example Replaces) indicate a specific call control action. Another way to invoke call control primitives is to define a specific Request-URI naming convention. Either these conventions must be shared between the client (the invoker) and the server, or published by or on behlf of the server. The former involves defining URL construction techniques (e.g. URL parameters and/or token conventions) as proposed in [netannc]. The latter technique usually involves discovering the URI via a SIP event package, a web page, a business card, or an Instant Message. Yet another means to acquire the URLs is to define a dictionary of primitives with well-defined semantics and provide a means to query the named primitives and corresponding URLs that may be invoked on the service or dialogs.

3.7.1 Naming Users in SIP

An address-of-record, or public SIP address, is a SIP (or SIPS) URI that points to a domain with a location server that can map the URI to set of Contact URIs where the user might be available. Typically the Contact URIs are populated via registration.

```
     Address of Record        Contacts

     sip:bob@biloxi.com   ->  sip:bob@babylon.biloxi.com:5060
                              sip:bbrown@mailbox.provider.net
```

sip:+1.408.555.6789@mobile.net

Callee Capabilities [20] defines a set of additional parameters to the Contact header that define the characteristics of the user agent at the specified URI. For example, there is a mobility parameter which indicates whether the UA is fixed or mobile. When a user agent registers, it places these parameters in the Contact headers to characterize the URIs it is registering. This allows a proxy for that domain to have information about the contact addresses for that user.

When a caller sends a request, it can optionally request Caller Preferences [21], by including the Accept-Contact and Reject-Contact headers which request certain handling by the proxy in the target domain. These headers contain preferences that describe the set of desired URIs to which the caller would like their request routed. The proxy in the target domain matches these preferences with the Contact characteristics originally registered by the target user. The target user can also choose to run arbitrarily complex "Find-me" feature logic on a proxy in the target domain.

There is a strong asymmetry in how preferences for callers and callees can be presented to the network. While a caller takes an active role by initiating the request, the callee takes a passive role in waiting for requests. This motivates the use of callee-supplied scripts and caller preferences included in the call request. This asymmetry is also reflected in the appropriate relationship between caller and callee preferences. A server for a callee should respect the wishes of the caller to avoid certain locations, while the preferences among locations has to be the callee's choice, as it determines where, for example, the phone rings and whether the callee incurs mobile telephone charges for incoming calls.

SIP User Agent implementations are encouraged to make intelligent decisions based on the type of participants (active/passive, hidden, human/robot) in a conversation space. This information is conveyed via the session dialog package or in a SIP header parameter communicated using an appropriate SIP header. For example, a music on hold service may take the sensible approach that if there are two or more unhidden participants, it should not provide hold music; or that it will not send hold music to robots.

Multiple participants in the same conversation space may represent the same human user. For example, the user may use one participant for video, chat, and whiteboard media on a PC and another for audio media on a SIP phone. In this case, the address-of-record is the same for both user agents, but the Contacts are different. In

addition, human users may add robot participants which act on their behalf (for example a call recording service, or a calendar reminder). Call Control features in SIP should continue to function as expected in such an environment.

3.7.2 Naming Services with SIP URIs

[Editor's Note: this section needs to be pared down considerably, and the examples replaced with example.{com|org|net} domain names.] A critical piece of defining a session level service that can be accessed by SIP is defining the naming of the resources within that service. This point cannot be overstated.

In the context of SIP control of application components, we take advantage of the fact that the standard SIP URI has a user part. Most services may be thought of as user automatons that participate in SIP sessions. It naturally follows that the user address, or the left-hand-side of the URI, should be utilized as a service indicator.

For example, media servers commonly offer multiple services at a single host address. Use of the user part as a service indicator enables service consumers to direct their requests without ambiguity. It has the added benefit of enabling media services to register their availability with SIP Registrars just as any "real" SIP user would. This maintains consistency and provides enhanced flexibility in the deployment of media services in the network.

There has been much discussion about the potential for confusion if media services URIs are not readily distinguishable from other types of SIP UA's. The use of a service namespace provides a mechanism to unambiguously identify standard interfaces while not constraining the development of private or experimental services.

In SIP, the request-URI identifies the user or service that the call is destined for. The great advantage of using URIs (specifically, the SIP request URI) as a service identifier comes because of the combination of two facts. First, unlike in the PSTN, where the namespace (dialable telephone numbers) are limited, URIs come from an infinite space. They are plentiful, and they are free. Secondly, the primary function of SIP is call routing through manipulations of the request URI. In the traditional SIP application, this URI represents people. However, the URI can also represent services, as we propose here. This means we can apply the routing services SIP provides to routing of calls to services. The result - the problem of service invocation and service location becomes a routing problem, for which SIP provides a scalable and flexible solution. Since there is such a vast namespace of services, we can explicitly name each service in a finely granular way. This allows the distribution of

services across the network.

Consider a conferencing service, where we have separated the names of ad-hoc conferences from scheduled conferences, we can program proxies to route calls for ad-hoc conferences to one set of servers, and calls for scheduled ones to another, possibly even in a different provider. In fact, since each conference itself is given a URI, we can distribute conferences across servers, and easily guarantee that calls for the same conference always get routed to the same server. This is in stark contrast to conferences in the telephone network, where the equivalent of the URI - the phone number - is scarce. An entire conferencing provider generally has one or two numbers. Conference IDs must be obtained through IVR interactions with the caller, or through a human attendant. This makes it difficult to distribute conferences across servers all over the network, since the PSTN routing only knows about the dialed number.

In the case of a dialog server, the voice dialog itself is the target for the call. As such, the request URI should contain the identifier for this spoken dialog. This is consistent with the Request-URI service invocation model of RFC 3087. This URL can be in one of two formats. In the first, the VoiceXML script is identified directly by an HTTP URL. In the second, the script is not specified. Rather, the dialog server uses its configuration to map the incoming request to a specific script.

Since the request URI could indicate a request for a variety of different services, of which a dialog server is only one type, this example request URI first begins with a service identifier, that indicates the basic service required. For VoiceXML scripts, this identification information is a URL-encoded version of the URL which references the script to execute, or if not present, the dialog server uses server-specific configuration to determine which script to execute.

Examples of URLs that invoke VoiceXML dialogs are: (line folding for clarity only)

```
sip:dialog.vxml.http%3a//dialogs.server.com/script32.vxml
 @vxmlservers.com

sip:dialog.vxml@vxmlservers.com
```

The first of these indicates that the dialog server (located at vxmlservers.com) should invoke a VoiceXML script fetched from http://dialogs.server.com/script32.vxml. Since the user part of the SIP URL cannot contain the : character, this must be escaped to %3a.

These types of conventions are not limited to application component servers. An ordinary SIP User Agent can have a special URIs as well, for example, one which is automatically answered by a speakerphone. Since URIs are so plentiful, using a separate URI for this service does not exhaust a valuable resource. The requested service is clear to the user agent receiving the request. This URI can also be included as part of another feature (for example, the Intercom feature described in Section 6.1.6). This feature can be specified with a SIP user parameter, since are part of the userpart of a SIP URI.

Likewise a Request URI can fully describe an announcement service through the use of the user part of the address and additional URI parameters. In our example, the user portion of the address, "annc", specifies the announcement service on the media server. The two URI parameters "play=" and "early=" specify the audio resource to play and whether early media is desired.

```
sip:annc@ms2.carrier.net;
 play=http://audio.carrier.net/allcircuitsbusy.au;early=yes

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

In practical applications, it is important that an invoker does not necessarily apply semantic rules to various URIs it did not create. Instead, it should allow any arbitrary string to be provisioned, and map the string to the desired behavior. The administrator of a service may choose to provision specific conventions or mnemonic strings, but the application should not require it. In any large installation, the system owner is likely to have pre-existing rules for mnemonic URIs, and any attempt by an application to define its own rules may create a conflict. Implementations should allow an arbitrary mix of URLs from these schemes, or any other scheme that renders valid SIP URIs to be provisioned, rather than enforce only one particular scheme.

For example, a voicemail application can be built using very different sets of URI conventions, as illustrated below:

URI Identity	Example Scheme 1 Example Scheme 2 Example Scheme 3
Deposit with standard greeting	sip:sub-rjs-deposit@vm.wcom.com sip:677283@vm.wcom.com sip:rjs@vm.wcom.com;mode=deposit
Deposit with on phone greeting	sip:sub-rjs-deposit-busy.vm.wcom.com sip:677372@vm.wcom.com sip:rjs@vm.wcom.com;mode=3991243
Deposit with special greeting	sip:sub-rjs-deposit-sg@vm.wcom.com sip:677384@vm.wcom.com sip:rjs@vm.wcom.com;mode=sg
Retrieve - SIP authentication	sip:sub-rjs-retrieve@vm.wcom.com sip:677405@vm.wcom.com sip:rjs@vm.wcom.com;mode=retrieve
Retrieve - prompt for PIN in-band	sip:sub-rjs-retrieve-inpin.vm.wcom.com sip:677415@vm.wcom.com sip:rjs@vm.wcom.com;mode=inpin

As we have shown, SIP URIs represent an ideal, flexbile mechanism for describing and naming service resources, be they queues, conferences, voice dialogs, announcements, voicemail treatments, or phone features.

3.8 Invoker Independence

With functional signaling, only the invoker of features in SIP need to know exactly which feature they are invoking. One of the primary benefits of this approach is that combinations of functional features work in SIP call control without requiring complex feature interaction matrices. For example, let us examine the combination of a "transfer" of a call which is "conferenced".

Alice calls Bob. Alice silently "conferences in" her robotic assistant Albert as a hidden party. Bob transfers Alice to Carol. If Bob asks Alice to Replace her leg with a new one to Carol then both Alice and Albert should be communicating with Carol (transparently).

Using the peer-to-peer model, this combination of features works fine if A is doing local mixing (Alice replaces Bob's call-leg with Carol's), or if A is using a central mixer (the mixer replaces Bob's

call leg with Carol's). A clever implementation using the 3pcc model can generate similar results.

New extensions to the SIP Call Control Framework should attempt to preserve this property.

3.9 Billing issues

Billing in the PSTN is typically based on who initiated a call. At the moment billing in a SIP network is neither consistent with itself, nor with the PSTN. (A billing model for SIP should allow for both PSTN-style billing, and non-PSTN billing.) The example below demonstrates one such inconsistency.

Alice places a call to Bob. Alice then blind transfers Bob to Carol through a PSTN gateway. In current usage of REFER, Bob may be billed for a call he did not initiate (his UA originated the outgoing call leg however). This is not necessarily a terrible thing, but it demonstrates a security concern (Bob must have appropriate local policy to prevent fraud). Also, Alice may wish to pay for Bob's session with Carol. There should be a way to signal this in SIP.

Likewise a Replacement call may maintain the same billing relationship as a Replaced call, so if Alice first calls Carol, then asks Bob to Replace this call, Alice may continue to receive a bill.

Further work in SIP billing should define a way to set or discover the direction of billing.

4. Catalog of call control actions and sample features

Call control actions can be categorized by the dialogs upon which they operate. The actions may involve a single or multiple dialogs. These dialogs can be early or established. Multiple dialogs may be related in a conversation space to form a conference or other interesting media topologies.

It should be noted that it is desirable to provide a means by which a party can discover the actions which may be performed on a dialog. The interested party may be independent or related to the dialogs. One means of accomplishing this is through the ability to define and obtain URLs for these actions as described in section .

Below are listed several call control "actions" which establish or modify dialogs and relate the participants in a conversation space. The names of the actions listed are for descriptive purposes only (they are not normative). This list of actions is not meant to be exhaustive.

In the examples, all actions are initiated by the user "Alice" represented by UA "A".

4.1 Early Dialog Actions

The following are a set of actions that may be performed on a single early dialog. These actions can be thought of as a set of remote control operations. For example an automaton might perform the operation on behalf of a user. Alternatively a user might use the remote control in the form of an application to perform the action on the early dialog of a UA which may be out of reach. All of these actions correspond to telling the UA how to respond to a request to establish an early dialog. These actions provide useful functionality for PDA, PC and server based applications which desire the ability to control a UA. A proposed mechanism for this type of functionality is described in Remote Call Control [23].

4.1.1 Remote Answer

A dialog is in some early dialog state such as 180 Ringing. It may be desirable to tell the UA to answer the dialog. That is tell it to send a 200 Ok response to establish the dialog.

4.1.2 Remote Forward or Put

It may be desirable to tell the UA to respond with a 3xx class response to forward an early dialog to another UA.

4.1.3 Remote Busy or Error Out

It may be desirable to instruct the UA to send an error response such as 486 Busy Here.

4.2 Single Dialog Actions

There is another useful set of actions which operate on a single established dialog. These operations are useful in building productivity applications for aiding users to control their phone. For example a CRM application which sets up calls for a user eliminating the need for the user to actually enter an address. These operations can also be thought of a remote control actions. A proposed mechanism for this type of functionality is described in Remote Call Control [23].

4.2.1 Remote Dial

This action instructs the UA to initiate a dialog. This action can be performed using the REFER method.

4.2.2 Remote On and Off Hold

This action instructs the UA to put an established dialog on hold. Though this operation can be conceptually be performed with the REFER method, there is no semantics defined as to what the referred party should do with the SDP. There is no way to distinguish between the desire to go on or off hold.

4.2.3 Remote Hangup

This action instructs the UA to terminate an early or established dialog. A REFER request with the following Refer-To URI performs this action. Note: this URL is not properly escaped.

```
sip:bob@babylon.biloxi.example.com;method=BYE?Call-ID=13413098
  &To=<sip:bob@biloxi.com>;tag=879738
  &From=<sip:alice@atlanta.example.com>;tag=023214
```

4.3 Multi-dialog actions

These actions apply to a set of related dialogs.

4.3.1 Transfer

The conversation space changes as follows:

```
         before            after
{ A , B }  -->   { C , B }
```

A replaces itself with C.

To make this happen using the peer-to-peer approach, "A" would send two SIP requests. A shorthand for those requests is shown below:

```
REFER B  Refer-To:C
BYE B
```

To make this happen instead using the 3pcc approach, the controller sends requests represented by the shorthand below:

```
INVITE C (w/SDP of B)
reINVITE B (w/SDP of C)
BYE A
```

Features enabled by this action: - blind transfer - transfer to a central mixer (some type of conference or forking) - transfer to park server (park) - transfer to music on hold or announcement server -

transfer to a "queue" - transfer to a service (such as Voice Dialogs service) - transition from local mixer to central mixer

This action is frequently referred to as "completing an attended transfer". It is described in more detail in cc-transfer [18].

4.3.2 Take

The conversation space changes as follows: { B , C } --> { B , A } A forcibly replaces C with itself. In most uses of this primitive, A is just "un-replacing" itself. Using the peer-to-peer approach, "A" sends: INVITE B Replaces: <call leg between B and C>

Using the 3pcc approach (all requests sent from controller) INVITE A (w/SDP of B) reINVITE B (w/SDP of A) BYE C

Features enabled by this action: - transferee completes an attended transfer - retrieve from central mixer (not recommended) - retrieve from music on hold or park - retrieve from queue - call center take - voice portal resuming ownership of a call it originated - answering-machine style screening (pickup) - pickup of a ringing call (i.e. early dialog)

Note: that pick up of a ringing call has perhaps some interesting additional requirements. First of all it is an early dialog as opposed to an established dialog. Secondly the party which is to pickup the call may only wish to do so only while it is an early dialog. That is in the race condition where the ringing UA accepts just before it receives signaling from the party wishing to take the call, the taking party wishes to yield or cancel the take. The goal is to avoid yanking an answered call from the called party.

This action is described in Replaces [9] and in cc-transfer [18].

4.3.3 Add

Note that the following 4 actions are described in cc-conferencing [19].

This is merely adding a participant to a SIP conference. The conversation space changes as follows: { A , B } --> { A, B, C } A adds C to the conversation. Using the peer-to-peer approach, adding a party using local mixing requires no signaling. To transition from a 2-party call or a locally mixed conference to centrally mixing A could send the following requests: REFER B Refer-To: conference-URI INVITE conference-URI BYE B To add a party to a conference: REFER C Refer-To: conference-URI or REFER conference-URI Refer-To: C Using the 3pcc approach to transition to centrally mixed, the controller

would send: INVITE mixer leg 1 (w/SDP of A) INVITE mixer leg 2 (w/SDP of B) INVITE C (late SDP) reINVITE A (w/SDP of mixer leg 1) reINVITE B (w/SDP of mixer leg 2) INVITE mixer leg3 (w/SDP of C) To add a party to a SIP conference: INVITE C (late SDP) INVITE conference-URI (w/SDP of C) Features enabled: - standard conference feature - call recording - answering-machine style screening (screening)

4.3.4 Local Join

The conversation space changes like this: { A, B} , {A, C} --> {A, B, C} or like this { A, B} , {C, D} --> {A, B, C, D} A takes two conversation spaces and joins them together into a single space. Using the peer-to-peer approach, A can mix locally, or REFER the participants of both conversation spaces to the same central mixer (as in 5.3) For the 3pcc approach, the call flows for inserting participants, and joining and splitting conversation spaces are tedious yet straightforward, so these are left as an exercise for the reader. Features enabled: - standard conference feature - leaving a sidebar to rejoin a larger conference

4.3.5 Insert

The conversation space changes like this: { B , C } --> {A, B, C } A inserts itself into a conversation space. A proposed mechanism for signaling this using the peer-to-peer approach is to send a new header in an INVITE with "joining" semantics. For example: INVITE B Join: <call id of B and C> If B accepted the INVITE, B would accept responsibility to setup the call legs and mixing necessary (for example: to mix locally or to transfer the participants to a central mixer) Features enabled: - barge-in - call center monitoring - call recording

4.3.6 Split

{ A, B, C, D } --> { A, B } , { C, D } If using a central conference with peer-to-peer REFER C Refer-To: conference-URI (new URI) REFER D Refer-To: conference-URI (new URI) BYE C BYE D Features enabled: - sidebar conversations during a larger conference

4.3.7 Near-fork

A participates in two conversation spaces simultaneously: { A, B } --> { B , A } & { A , C } A is a participant in two conversation spaces such that A sends the same media to both spaces, and renders media from both spaces, presumably by mixing or rendering the media from both. We can define that A is the "anchor" point for both forks, each of which is a separate conversation space. This action is purely local implementation (it requires no special signaling).

Local features such as switching calls between the background and foreground are possible using this media relationship.

4.3.8 Far fork

The conversation space diagram... { A, B } --> { A , B } & { B , C } A requests B to be the "anchor" of two conversation spaces. This is easily setup by creating a conference with two subconferences and setting the media policy appopriately such that B is a participant in both. Media forking can also be setup using 3pcc as described in Section 5.1 of RFC3264 [3] (an offer/answer model for SDP). The session descriptions for forking are quite complex. Controllers should verify that endpoints can handle forked-media, for example using prior configuration.

Features enabled:
- o barge-in
- o voice portal services
- o whisper
- o hotword detection
- o sending DTMF somewhere else

5. Security Considerations

Call Control primitives provide a powerful set of features that can be dangerous in the hands of an attacker. To complicate matters, call control primitives are likely to be automatically authorized without direct human oversight.

The class of attacks which are possible using these tools include the ability to eavesdrop on calls, disconnect calls, redirect calls, render irritating content (including ringing) at a user agent, cause an action that has billing consequences, subvert billing (theft-of-service), and obtain private information. Call control extensions must take extra care to describe how these attacks will be prevented.

We can also make some general observations about authorization and trust with respect to call control. The security model is dramatically dependent on the signaling model chosen (see section 3.2)

Let us first examine the security model used in the 3pcc approach. All signaling goes through the controller, which is a trusted entity. Traditional SIP authentication and hop-by-hop encrpytion and message integrity work fine in this environment, but end-to-end encrpytion and message integrity may not be possible.

When using the peer-to-peer approach, call control actions and

primitives can be legitimately initiated by a) an existing participant in the conversation space, b) a former participant in the conversation space, or c) an entity trusted by one of the participants. For example, a participant always initiates a transfer; a retrieve from Park (a take) is initiated on behalf of a former participant; and a barge-in (insert or far-fork) is initiated by a trusted entity (an operator for example).

Authenticating requests by an existing participant or a trusted entity can be done with baseline SIP mechanisms. In the case of features initiated by a former participant, these should be protected against replay attacks by using a unique name or identifier per invocation. The Replaces header exhibits this behavior as a by-product of its operation (once a Replaces operation is successful, the call-leg being Replaced no longer exists). For other requests, a "one-time" Request-URI may be provided to the feature invoker.

To authorize call control primitives that trigger special behavior (such as an INVITE with Replaces or Join semantics), the receiving user agent may have trouble finding appropriate credentials with which to challenge or authorize the request, as the sender may be completely unknown to the receiver, except through the introduction of a third party. These credentials need to be passed transitively in some way or fetched in an event body, for example.

6. IANA Considerations

This document required no action by IANA.

7. Appendix A: Example Features

Primitives are defined in terms of their ability to provide features. These example features should require an amply robust set of services to demonstrate a useful set of primitives. They are described here briefly. Note that the descriptions of these features are non-normative. Some of these features are used as examples in section 6 to demonstrate how some features may require certain media relationships. Note also that this document describes a mixture of both features originating in the world of telephones, and features which are clearly Internet oriented.

Example Feature Definitions:

Call Waiting - Alice is in a call, then receives another call. Alice can place the first call on hold, and talk with the other caller. She can typically switch back and forth between the callers.

Blind Transfer - Alice is in a conversation with Bob. Alice asks Bob

to contact Carol, but makes no attempt to contact Craol independently. In many implementations, Alice does not verify Bob's success or failure in contacting Carol.

Attended Transfer - The transferring party establishes a session with the transfer target before completing the transfer.

Consultative transfer - the transferring party establishes a session with the target and mixes both sessions together so that all three parties can participate, then disconnects leaving the transferee and transfer target with an active session.

Conference Call - Three or more active, visible participants in the same conversation space.

Call Park - A call participant parks a call (essentially puts the call on hold), and then retrieves it at a later time (typically from another location).

Call Pickup - A party picks up a call that was ringing at another location. One variation allows the caller to choose which location, another variation just picks up any call in that user's "pickup group".

Music on Hold - When Alice places a call with Bob on hold, it replaces its audio with streaming content such as music, announcements, or advertisements.

Call Monitoring - A call center supervisor joins an in-progress call for monitoring purposes.

Barge-in - Carol interrupts Alice who has a call in-progress call with Bob. In some variations, Alice forcibly joins a new conversation with Carol, in other variations, all three parties are placed in the same conversation (basically a 3-way conference).

Hotline - Alice picks up a phone and is immediately connected to the technical support hotline, for example.

Autoanswer - Calls to a certain address or location answer immediately via a speakerphone.

Intercom - Alice typically presses a button on a phone which immediately connects to another user or phone and casues that phone to play her voice over its speaker. Some variations immediately setup two-way communications, other variations require another button to be pressed to enable a two-way conversation.

Speakerphone paging - Alice calls the paging address and speaks. Her voice is played on the speaker of every idle phone in a preconfigured group of phones.

Speed dial - Alice dials an abbreviated number, or enters an alias, or presses a special speed dial button representing Bob. Her action is interpreted as if she specified the full address of Bob.

Call Return - Alice calls Bob. Bob misses the call or is disconnected before he is finished talking to Alice. Bob invokes Call return which calls Alice, even if Alice did not provide her real identity or location to Bob.

Inbound Call Screening - Alice doesn't want to receive calls from Matt. Inbound Screening prevents Matt from disturbing Alice. In some variations this works even if Matt hides his identity.

Outbound Call Screening - Alice is paged and unknowingly calls a PSTN pay-service telephone number in the Carribean, but local policy blocks her call, and possibly informs her why.

Call Forwarding - Before a call-leg is accepted it is redirected to another location, for example, because the originally intended recipient is busy, does not answer, is disconnected from the network, configured all requests to go soemwhere else.

Message Waiting - Bob calls Alice when she steps away from her phone, when she returns a visible or audible indicator conveys that someone has left her a voicemail message. The message waiting indication may also convey how many messages are waiting, from whom, what time, and other useful pieces of information.

Do Not Disturb - Alice selects the Do Not Disturb option. Calls to her either ring briefly or not at all and are forwarded elsewhere. Some variations allow specially authorized callers to override this feature and ring Alice anyway.

Distinctive ring - Incoming calls have different ring cadences or sample sounds depending on the From party, the To party, or other factors.

Automatic Callback: Alice calls Bob, but Bob is busy. Alice would like Bob to call her automatically when he is available. When Bob hangs up, alice's phone rings. When Alice answers, Bob's phone rings. Bob answers and they talk.

Find-Me - Alice sets up complicated rules for how she can be reached (possibly using [CPL], [presence] or other factors). When Bob calls

Alice, his call is eventually routed to a temporary Contact where Alice happens to be available.

Whispered call waiting - Alice is in a conversation with Bob. Carol calls Alice. Either Carol can "whisper" to Alice directly ("Can you get lunch in 15 minutes?"), or an automaton whispers to Alice informing her that Carol is trying to reach her.

Voice message screening - Bob calls Alice. Alice is screening her calls, so Bob hears Alice's voicemail greeting. Alice can hear Bob leave his message. If she decides to talk to Bob, she can take the call back from the voicemail system, otherwise she can let Bob leave a message. This emulates the behavior of a home telephone answering machine

Presence-Enabled Conferencing: Alice wants to set up a conference call with Bob and Cathy when they all happen to be available (rather than scheduling a predefined time). The server providing the application monitors their status, and calls all three when they are all "online", not idle, and not in another call.

IM Conference Alerts: A user receives an notification as an Instant Message whenever someone joins a conference they are also in.

Single Line Extension -- A group of phones are all treated as "extensions" of a single line. A call for one rings them all. As soon as one answers, the others stop ringing. If any extension is actively in a coversation, another extension can "pick up" and immediately join the conversation. This emulates the behavior of a home telephone line with multiple phones.

Click-to-dial - Alice looks in her company directory for Bob. When she finds Bob, she clicks on a URL to call him. Her phone rings (or possibly answers automatically), and when she answers, Bob's phone rings.

Pre-paid calling - Alice pays for a certain currency or unit amount of calling value. When she places a call, she provides her account number somehow. If her account runs out of calling value during a call her call is disconnected or redirected to a service where she can purchase more calling value.

Voice Portal - A service that allows users to access a portal site using spoken dialog interaction. For example, Alice needs to schedule a working dinner with her co-worker Carol. Alice uses a voice portal to check Carol's flight schedule, find a restauraunt near her hotel, make a reservation, get directions there, and page Carol with this information.

7.1 Implementation of these features

Example Features:	
Call Hold	[Offer/Answer] for SIP
Call Waiting	Local Implementation
Blind Transfer	[cc-transfer]
Attended Transfer	[cc-transfer]
Consultative transfer	[cc-transfer]
Conference Call	[conf-models]
Call Park	*[examples]
Call Pickup	*[examples]
Music on Hold	*[examples]
Call Monitoring	*Insert
Barge-in	*Insert or Far-Fork
Hotline	Local Implementation
Autoanswer	Local URI convention
Speed dial	Local Implementation
Intercom	*Speed dial + autoanswer
Speakerphone paging	*Speed dial + autoanswer
Call Return	Proxy feature
Inbound Call Screening	Proxy or Local implementation
Outbound Call Screening	Proxy feature
Call Forwarding	Proxy or Local implementation
Message Waiting	[msg-waiting]
Do Not Disturb	[presence]
Distinctive ring	*Proxy or Local implementation
Automatic Callback	2 person presence-based conference
Find-Me	Proxy service based on presence
Whispered call waiting	Local implementation
Voice message screening	*
Presence-based Conferencing	*call when presence = available
IM Conference Alerts	subscribe to conference status
Single Line Extension	*
Click-to-dial	*
Pre-paid calling	*
Voice Portal	*

7.1.1 Call Park

Call park requires the ability to: put a dialog some place, advertise it to users in a pickup group and to uniquely identify it in a means that can be communicated (including human voice). The dialog can be held locally on the UA parking the dialog or alternatively transferred to the park service for the pickup group. The parked dialog then needs to be labeled (e.g. orbit 12) in a way that can be communicated to the party that is to pick up the call. The UAs in the pick up group discovers the parked dialog(s) via the dialog

package from the park service. If the dialog is parked locally the park service merely aggregates the parked call states from the set of UAs in the pickup up group.

7.1.2 Call Pickup

There are two different features which are called call pickup. The first is the pickup of a parked dialog. The UA from which the dialog is to be picked up subscribes to the session dialog state of the park service or the UA which has locally parked the dialog. Dialogs which are parked should be labeled with an identifier. The labels are used by the UA to allow the user to indicate which dialog is to be picked up. The UA picking up the call invoked the URL in the call state which is labeled as replace-remote.

The other call pickup feature involves picking up an early dialog (typically ringing). This feature uses some of the same primitives as the pick up of a parked call. The call state of the UA ringing phone is advertised using the dialog package. The UA which is to pickup the early dialog subscribes either directly to the ringing UA or to a service aggregating the states for UAs in the pickup group. The call state identifies early dialogs. The UA uses the call state(s) to help the user choose which early dialog that is to be picked up. The UA then invokes the URL in the call state labeled as replace-remote.

7.1.3 Music on Hold

Music on hold can be implemented a number of ways. One way is to transfer the held call to a holding service. When the UA wishes to take the call off hold it basically performs a take on the call from the holding service. This involves subscribing to call state on the holding service and then invoking the URL in the call state labeled as replace-remote.

Alternatively music on hold can be performed as a local mixing operation. The UA holding the call can mix in the music from the music service via RTP (i.e. an additional dialog) or RTSP or other streaming media source. This approach is simpler (i.e. the held dialog does not move so there is less chance of loosing them) from a protocol perspective, however it does use more LAN bandwidth and resources on the UA.

7.1.4 Call Monitoring

Call monitoring is a Join operation. The monitoring UA sends a Join to the dialog it wants to listen to. It is able to discover the dialog via the dialog state on the monitored UA. The monitoring UA

sends SDP in the INVITE which indicates receive only media. As the UA is monitoring only it does not matter whether the UA indicates it wishes the send stream be mix or point to point.

7.1.5 Barge-in

Barge-in works the same as call monitoring except that it must indicate that the send media stream to be mixed so that all of the other parties can hear the stream from UA barging in.

7.1.6 Intercom

The UA initiates a dialog using INVITE in the ordinary way. The calling UA then signals the paged UA to answer the call. The calling UA may discover the URL to answer the call via the session dialog package of the called UA. The called UA accepts the INVITE with a 200 Ok and automatically enables the speakerphone.

Alternatively this can be a local decision for the UA to answer based upon called party identification.

7.1.7 Speakerphone paging

Speakerphone paging can be implemented using either multicast or through a simple multipoint mixer. In the multicast solution the paging UA sends a multicast INVITE with send only media in the SDP (see also RFC3264). The automatic answer and enabling of the speakerphone is a locally configured decision on the paged UAs. The paging UA sends RTP via the multicast address indicated in the SDP.

The multipoint solution is accomplished by sending an INVITE to the multipoint mixer. The mixer is configured to automatically answer the dialog. The paging UA then sends REFER requests for each of the UAs that are to become paging speakers (The UA is likely to send out a single REFER which is parallel forked by the proxy server). The UAs performing as paging speakers are configured to automatically answer based upon caller identification (e.g. To field, URI or Referred-To headers).

Finally as a third option, the user agent can send a mass-invitation request to a conference server, which would create a conference and send invitations to the conference to all user agents in the paging group.

7.1.8 Distinctive ring

The target UA either makes a local decision based on information in an incoming INVITE (To, From, Contact, Request-URI) or trusts an

Alert-Info header provded by the caller or inserted by a trusted proxy. In the latter case, the UA fetches the content described in the URI (typically via http) and renders it to the user.

7.1.9 Voice message screening

At first, this is the same as call monitoring. In this case the voicemail service is one of the UAs. The UA screening the message monitors the call on the voicemail service, and also subscribes to call-leg information. If the user screening their messages decides to answer, they perform a Take from the voicemail system (for example, send an INVITE with Replaces to the UA leaving the message)

7.1.10 Single Line Extension

Incoming calls ring all the extensions through basic parallel forking [bis]. Each extension subscribes to call-leg events from each other extension. While one user has an active call, any other UA extension can insert itself into that conversation (it already knows the call-leg information)in the same way as barge-in.

7.1.11 Click-to-dial

The application or server which hosts the click-to-dial application captures the URL to be dialed and can setup the call using 3pcc or can send a REFER request to the UA which is to dial the address. As users sometimes change their mind or wish to give up listing to a ringing or voicemail answered phone, this application illustrates the need to also have the ability to remotely hangup a call.

7.1.12 Pre-paid calling

For prepaid calling, the user's media always passes through a device which is trusted by the pre-paid provider. This may be the other endpoint (for example a PSTN gateway). In either case, an intermediary proxy or B2BUA can periodically verify the amount of time available on the pre-paid account, and use the session-timer extension to cause the trusted endpoint (gateway) or intermediary (media relay) to send a reINVITE before that time runs out. During the reINVITE, the SIP intermediary can reverify the account and insert another session-timer header.

Note that while most pre-paid systems on the PSTN use an IVR to collect the account number and destination, this isn't strictly necessary for a SIP-originated prepaid call. SIP requests and SIP URIs are sufficiently expressive to convey the final destination, the provider of the prepaid service, the location from which the user is calling, and the prepaid account they want to use. If a pre-paid IVR

is used, the mechanism described below (Voice Portals) can be combined as well.

7.1.13 Voice Portal

A voice portal is essentially a complex collection of voice dialogs used to access interesting content. One of the most desirable call control features of a Voice Portal is the ability to start a new outgoing call from within the context of the Portal (to make a restauraunt reservation, or return a voicemail message for example). Once the new call is over, the user should be able to return to the Portal by pressing a special key, using some DTMF sequence (ex: a very long pound or hash tone), or by speaking a hotword (ex: "Main Menu").

In order to accomplish this, the Voice Portal starts with the following media relationship:

{ User , Voice Portal }

The user then asks to make an outgoing call. The Voice Portal asks the User to perform a Far-Fork. In other words the Voice Portal wants the following media relationship:

{ Target , User } & { User , Voice Portal }

The Voice Portal is now just listening for a hotword or the appropriate DTMF. As soon as the user indicates they are done, the Voice Portal Takes the call from the old Target, and we are back to the original media relationship.

This feature can also be used by the account number and phone number collection menu in a pre-paid calling service. A user can press a DTMF sequence which presents them with the appropriate menu again.

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