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Registration for Multiple Phone Numbers in the Session Initiation  
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Abstract

This document defines a mechanism by which a Session Initiation Protocol (SIP) server acting as a traditional Private Branch Exchange (SIP-PBX) can register with a SIP Service Provider (SSP) to receive phone calls for SIP User Agents (UAs). In order to function properly, this mechanism requires that each of the Addresses of Record (AORs) registered in bulk map to a unique set of contacts. This requirement is satisfied by AORs representing phone numbers regardless of the domain, since phone numbers are fully qualified and globally unique. This document therefore focuses on this use case.

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## 1. Introduction

The Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. One of SIP's primary functions is providing rendezvous between users. By design, this rendezvous has been provided through a combination of the server look-up procedures defined in RFC 3263 [4], and the registrar procedures described in RFC 3261 [3].

The intention of the original protocol design was that any user's AOR (Address of Record) would be handled by the authority indicated by the hostport portion of the AOR. The users would register individual reachability information with this authority, which would then route incoming requests accordingly.

In actual deployments, some SIP servers have been deployed in architectures that, for various reasons, have requirements to provide dynamic routing information for large blocks of AORs, where all of the AORs in the block were to be handled by the same server. For purposes of efficiency, many of these deployments do not wish to maintain separate registrations for each of the AORs in the block. This leads to the desire for an alternate mechanism for providing dynamic routing information for blocks of AORs.

Although the use of SIP REGISTER request messages to update reachability information for multiple users simultaneously is somewhat beyond the original semantics defined for REGISTER requests by RFC 3261 [3], this approach has seen significant deployment in certain environments. In particular, deployments in which small to medium SIP-PBX servers are addressed using E.164 numbers have used this mechanism to avoid the need to maintain DNS entries or static IP addresses for the SIP-PBX servers.

In recognition of the momentum that REGISTER-based approaches have seen in deployments, this document defines a REGISTER-based approach. Since E.164-addressed UAs are very common today in SIP-PBX environments, and since SIP URIs in which the user portion is an E.164 number are always globally unique regardless of the domain, this document focuses on registration of SIP URIs in which the user portion is an E.164 number.

## 2. Constraints

Within the problem space that has been established for this work, several constraints shape our solution. These are defined in the MARTINI requirements document [22], and analyzed in Appendix A. In

terms of impact to the solution at hand, the following two constraints have the most profound effect: (1) The SIP-PBX cannot be assumed to be assigned a static IP address; and (2) No DNS entry can be relied upon to consistently resolve to the IP address of the SIP-PBX.

### 3. Terminology and 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].

Further, the term "SSP" is meant as an acronym for a "SIP Service Provider," while the term "SIP-PBX" is used to indicate a SIP Private Branch Exchange.

Indented portions of the document, such as this one, form non-normative, explanatory sections of the document.

Although SIP is a text-based protocol, some of the examples in this document cannot be unambiguously rendered without additional markup due to the constraints placed on the formatting of RFCs. This document uses the <allOneLine/> markup convention established in RFC 4475 [17] to avoid ambiguity and meet the RFC layout requirements. For the sake of completeness, the text defining this markup from Section 2.1 of RFC 4475 [17] is reproduced in its entirety below:

Several of these examples contain unfolded lines longer than 72 characters. These are captured between <allOneLine/> tags. The single unfolded line is reconstructed by directly concatenating all lines appearing between the tags (discarding any line feeds or carriage returns). There will be no whitespace at the end of lines. Any whitespace appearing at a fold-point will appear at the beginning of a line.

The following represent the same string of bits:

Header-name: first value, reallylongsecondvalue, third value

```
<allOneLine>
Header-name: first value,
  reallylongsecondvalue
, third value
</allOneLine>
```

```
<allOneLine>
Header-name: first value,
  reallylong
second
value,
  third value
</allOneLine>
```

Note that this is NOT SIP header-line folding, where different strings of bits have equivalent meaning.

#### 4. Mechanism Overview

The overall mechanism is achieved using a REGISTER request with a specially-formatted Contact URI. This document also defines an option tag that can be used to ensure a registrar and any intermediaries understand the mechanism described herein.

The Contact URI itself is tagged with a URI parameter to indicate that it actually represents multiple phone-number-associated contacts.

We also define some lightweight extensions to the Globally Routable UA URIs (GRUU) mechanism defined by RFC 5627 [20] to allow the use of public and temporary GRUUs assigned by the SSP.

Aside from these extensions, the REGISTER request itself is processed by a registrar in the same way as normal registrations: by updating its location service with additional AOR-to-Contact bindings.

Note that the list of AORs associated with a SIP-PBX is a matter of local provisioning at the SSP and at the SIP-PBX. The mechanism defined in this document does not provide any means to detect or recover from provisioning mismatches (although the registration event package can be used as a standardized means for auditing such AORs; see Section 7.2.1).

#### 5. Registering for Multiple Phone Numbers

##### 5.1. SIP-PBX Behavior

To register for multiple AORs, the SIP-PBX sends a REGISTER request to the SSP. This REGISTER request varies from a typical REGISTER request in two important ways. First, it MUST contain an option tag of "gin" in both a "Require" header field and a "Proxy-Require" header field. (The option tag "gin" is an acronym for "generate

implicit numbers".) Second, in at least one "Contact" header field, it MUST include a Contact URI that contains the URI parameter "bnc" (which stands for "bulk number contact"), and no user portion (hence no "@" symbol). A URI with a "bnc" parameter MUST NOT contain a user portion. Except for the SIP URI "user" parameter, this URI MAY contain any other parameters that the SIP-PBX desires. These parameters will be echoed back by the SSP in any requests bound for the SIP-PBX.

Because of the constraints discussed in Section 2, the host portion of the Contact URI will generally contain an IP address, although nothing in this mechanism enforces or relies upon that fact. If the SIP-PBX operator chooses to maintain DNS entries that resolve to the IP address of his SIP-PBX via RFC 3263 resolution procedures, then this mechanism works just fine with domain names in the Contact header field.

The "bnc" URI parameter indicates that special interpretation of the Contact URI is necessary: instead of indicating the insertion of a single Contact URI into the location service, it indicates that multiple URIs (one for each associated AOR) should be inserted.

Any SIP-PBX implementing the registration mechanism defined in this document MUST also support the Path mechanism defined by RFC 3327 [10], and MUST include a 'path' option-tag in the Supported header field of the REGISTER request (which is a stronger requirement than imposed by the Path mechanism itself). This behavior is necessary because proxies between the SIP-PBX and the Registrar may need to insert Path header field values in the REGISTER request for this document's mechanism to function properly, and per RFC 3327 [10], they can only do so if the User Agent Client (UAC) inserted the option-tag in the Supported header field. In accordance with the procedures defined in RFC 3327 [10], the SIP-PBX is allowed to ignore the Path header fields returned in the REGISTER response.

## 5.2. Registrar Behavior

The registrar, upon receipt of a REGISTER request containing at least one Contact header field with a "bnc" parameter will use the value in the "To" header field to identify the SIP-PBX for which registration is being requested. It then authenticates the SIP-PBX (using, e.g., SIP Digest authentication, mutual TLS [18], or some other authentication mechanism). After the SIP-PBX is authenticated, the registrar updates its location service with a unique AOR-to-Contact mapping for each of the AORs associated with the SIP-PBX. Semantically, each of these mappings will be treated as a unique row in the location service. The actual implementation may, of course, perform internal optimizations to reduce the amount of memory used to

store such information.

For each of these unique rows, the AOR will be in the format that the SSP expects to receive from external parties (e.g. "sip:+12145550102@ssp.example.com"), and the corresponding Contact will be formed by adding to the REGISTER request's Contact URI a user portion containing the fully-qualified, E.164-formatted number (including the preceding "+" symbol) and removing the "bnc" parameter. Aside from the initial "+" symbol, this E.164-formatted number MUST consist exclusively of digits from 0 through 9, and explicitly MUST NOT contain any visual separator symbols (e.g., "-", ".", "(", or ")"). For example, if the "Contact" header field contains the URI <sip:198.51.100.3:5060;bnc>, then the Contact value associated with the aforementioned AOR will be <sip:+12145550102@198.51.100.3:5060>.

Although the SSP treats this registration as a number of discrete rows for the purpose of re-targeting incoming requests, the renewal, expiration, and removal of these rows is bound to the registered contact. In particular, this means that REGISTER requests that attempt to de-register a single AOR that has been implicitly registered MUST NOT remove that AOR from the bulk registration. In this circumstance, the registrar simply acts as if the UA attempted to un-register a contact that wasn't actually registered (e.g., return the list of presently registered contacts in a success response). A further implication of this property is that an individual extension that is implicitly registered may also be explicitly registered using a normal, non-bulk registration (subject to SSP policy). If such a registration exists, it is refreshed independently of the bulk registration, and is not removed when the bulk registration is removed.

A registrar that receives a REGISTER request containing a Contact URI with both a "bnc" parameter and a user portion MUST NOT send a 200-class (success) response. If no other error is applicable, the registrar can use a 400 (Bad Request) response to indicate this error condition.

Note that the preceding paragraph is talking about the user portion of a URI:

```
sip:+12145550100@example.com
^^^^^^^^^^^^^^
```

A Registrar compliant with this document MUST support the Path mechanism defined in RFC 3327 [10]. The rationale for support of this mechanism is given in section Section 5.1.

Aside from the "bnc" parameter, all URI parameters present on the "Contact" URI in the REGISTER request MUST be copied to the Contact value stored in the location service.

If the SSP servers perform processing based on User Agent Capabilities (as defined in RFC 3840 [13]), they will treat any feature tags present on a Contact header field with a "bnc" parameter in its URI as applicable to all of the resulting AOR-to-Contact mappings. Similarly, any option tags present on the REGISTER request that indicate special handling for any subsequent requests are also applicable to all of the AOR-to-Contact mappings.

### 5.3. SIP URI "user" Parameter Handling

This document does not modify the behavior specified in RFC 3261 [3] for inclusion of the "user" parameter on request URIs. However, to avoid any ambiguity in handling at the SIP-PBX, the following normative behavior is imposed on its interactions with the SSP.

When a SIP-PBX registers with an SSP using a contact containing a "bnc" parameter, that contact MUST NOT include a "user" parameter. A registrar that receives a REGISTER request containing a Contact URI with both a "bnc" parameter and a "user" parameter MUST NOT send a 200-class (success) response. If no other error is applicable, the registrar can use a 400 (Bad Request) response to indicate this error condition.

Note that the preceding paragraph is talking about the "user" parameter of a URI:

```
sip:+12145550100@example.com;user=phone
      ^^^^^^^^^^^
```

When a SIP-PBX receives a request from an SSP, and the Request-URI contains a user portion corresponding to an AOR registered using a contact containing a "bnc" parameter, then the SIP-PBX MUST NOT reject the request (or otherwise cause the request to fail) due to the absence, presence, or value of a "user" parameter on the Request-URI.

## 6. SSP Processing of Inbound Requests

In general, after processing the AOR-to-Contact mapping described in the preceding section, the SSP Proxy/Registrar (or equivalent entity) performs traditional Proxy/Registrar behavior, based on the mapping. For any inbound SIP requests whose AOR indicates an E.164 number assigned to one of the SSP's customers, this will generally involve



setting the target set to the registered contacts associated with that AOR, and performing request forwarding as described in section 16.6 of RFC 3261 [3]. An SSP using the mechanism defined in this document MUST perform such processing for inbound INVITE requests and SUBSCRIBE requests to the "reg" event package (see Section 7.2.2), and SHOULD perform such processing for all other method types, including unrecognized SIP methods.

## 7. Interaction with Other Mechanisms

The following sections describe the means by which this mechanism interacts with relevant REGISTER-related extensions currently defined by the IETF.

### 7.1. Globally Routable User-Agent URIs (GRUU)

To enable advanced services to work with UAs behind a SIP-PBX, it is important that the GRUU mechanism defined by RFC 5627 [20] work correctly with the mechanism defined by this document -- that is, that User Agents served by the SIP-PBX can acquire and use GRUUs for their own use.

#### 7.1.1. Public GRUUs

Support of public GRUUs is OPTIONAL in SSPs and SIP-PBXes.

When a SIP-PBX registers a Bulk Number Contact (a Contact with a "bnc" parameter), and also invokes GRUU procedures for that Contact during registration, then the SSP will assign a public GRUU to the SIP-PBX in the normal fashion. Because the URI being registered contains a "bnc" parameter, the GRUU will also contain a "bnc" parameter. In particular, this means that the GRUU will not contain a user portion.

When a UA registers a contact with the SIP-PBX using GRUU procedures, the SIP-PBX provides to the UA a public GRUU formed by adding an "sg" parameter to the GRUU parameter it received from the SSP. This "sg" parameter contains a disambiguation token that the SIP-PBX can use to route inbound requests to the proper UA.

So, for example, when the SIP-PBX registers with the following contact header field:

```
Contact: <sip:198.51.100.3;bnc>;  
+sip.instance="urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6>"
```

Then the SSP may choose to respond with a Contact header field that

looks like this:

```
<allOneLine>
Contact: <sip:198.51.100.3;bnc>;
pub-gruu="sip:ssp.example.com;bnc;gr=urn:
uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6";
+sip.instance="<urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6>"
;expires=7200
</allOneLine>
```

When its own UAs register using GRUU procedures, the SIP-PBX can then add whatever device identifier it feels appropriate in an "sg" parameter, and present this value to its own UAs. For example, assume the UA associated with the AOR "+12145550102" sent the following Contact header field in its REGISTER request:

```
Contact: <sip:line-1@10.20.1.17>;
+sip.instance="<urn:uuid:d0e2f290-104b-11df-8a39-0800200c9a66>"
```

The SIP-PBX will add an "sg" parameter to the pub-gruu it received from the SSP with a token that uniquely identifies the device (possibly the URN itself; possibly some other identifier); insert a user portion containing the fully-qualified E.164 number associated with the UA; and return the result to the UA as its public GRUU. The resulting Contact header field sent from the SIP-PBX to the registering UA would look something like this:

```
<allOneLine>
Contact: <sip:line-1@10.20.1.17>;
pub-gruu="sip:+12145550102@ssp.example.com;gr=urn:
uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6;sg=00:05:03:5e:70:a6";
+sip.instance="<urn:uuid:d0e2f290-104b-11df-8a39-0800200c9a66>"
;expires=3600
</allOneLine>
```

When an incoming request arrives at the SSP for a GRUU corresponding to a bulk number contact ("bnc"), the SSP performs slightly different processing for the GRUU than it would for a URI without a "bnc" parameter. When the GRUU is re-targeted to the registered bulk number contact, the SSP MUST copy the "sg" parameter from the GRUU to the new target. The SIP-PBX can then use this "sg" parameter to determine which user agent the request should be routed to. For example, the first line of an INVITE request that has been re-targeted to the SIP-PBX for the UA shown above would look like this:

```
INVITE sip:+12145550102@198.51.100.3;sg=00:05:03:5e:70:a6 SIP/2.0
```

### 7.1.2. Temporary GRUUs

In order to provide support for privacy, the SSP SHOULD implement the temporary GRUU mechanism described in this section. Reasons for not doing so would include systems with an alternative privacy mechanism which maintains the integrity of public GRUUs (i.e., if public GRUUs are anonymized then the anonymizer function would need to be capable of providing as the anonymized URI a globally routable URI that routes back only to the target identified by the original public GRUU).

Temporary GRUUs are used to provide anonymity for the party creating and sharing the GRUU. Being able to correlate two temporary GRUUs as having originated from behind the same SIP-PBX violates this principle of anonymity. Consequently, rather than relying upon a single, invariant identifier for the SIP-PBX in its UA's temporary GRUUs, we define a mechanism whereby the SSP provides the SIP-PBX with sufficient information for the SIP-PBX to mint unique temporary GRUUs. These GRUUs have the property that the SSP can correlate them to the proper SIP-PBX, but no other party can do so. To achieve this goal, we use a slight modification of the procedure described in appendix A.2 of RFC 5627 [20].

The SIP-PBX needs to be able to construct a temp-gruu in a way that the SSP can decode. In order to ensure that the SSP can decode GRUUs, we need to standardize the algorithm for creation of temp-gruus at the SIP-PBX. This allows the SSP to reverse the algorithm to identify the registration entry that corresponds to the GRUU.

It is equally important that no party other than the SSP is capable of decoding a temporary GRUU, including other SIP-PBXes serviced by the SSP. To achieve this property, an SSP that supports temporary GRUUs MUST create and store an asymmetric key pair, {K\_e1, K\_e2}. K\_e1 is kept secret by the SSP, while K\_e2 is shared with the SIP-PBXes via provisioning.

All base64 encoding discussed in the following sections MUST use the character set and encoding defined in Section 4 of RFC 4648 [8], except that any trailing "=" characters are discarded on encoding, and added as necessary to decode.

The following sections make use of the term "HMAC-SHA256-80" to describe a particular HMAC algorithm. In this document, HMAC-SHA256-80 is defined to mean the application of the SHA-256 [24] secure hashing algorithm, and truncating the results to 80 bits by discarding the trailing (least significant) bits.

#### 7.1.2.1. Generation of temp-gruu-cookie by the SSP

An SSP that supports temporary GRUUs MUST include a "temp-gruu-cookie" parameter on all Contact header fields containing a "bnc" parameter in a 200-class REGISTER response. This "temp-gruu-cookie" MUST have the following properties:

1. It can be used by the SSP to uniquely identify the registration to which it corresponds.
2. It is encoded using base64. This allows the SIP-PBX to decode it into as compact a form as possible for use in its calculations.
3. It is of a fixed length. This allows for extraction of it once the SIP-PBX has concatenated a distinguisher onto it.
4. The temp-gruu-cookie MUST NOT be forgeable by any party. In other words, the SSP needs to be able to examine the cookie and validate that it was generated by the SSP.
5. The temp-gruu-cookie MUST be invariant during the course of a registration, including any refreshes to that registration. This property is important, as it allows the SIP-PBX to examine the temp-gruu-cookie to determine whether the temp-gruus it has issued to its UAs are still valid.

The above properties can be met using the following algorithm, which is non-normative. Implementors may chose to implement any algorithm of their choosing for generation of the temp-gruu-cookie, as long as it fulfills the five properties listed above.

The registrar maintains a counter, I. This counter is 48 bits long, and initialized to zero. This counter is persistently stored, using a back-end database or similar technique. When the registrar creates the first temporary GRUU for a particular SIP-PBX and instance ID (as defined by [20]), the registrar notes the current value of the counter, I<sub>i</sub>, and increments the counter in the database. The registrar then maps I<sub>i</sub> to the Contact and instance ID using the database, a persistent hash-map or similar technology. If the registration expires such that there are no longer any contacts with that particular instance ID bound to the GRUU, the registrar removes the mapping. Similarly, if the temporary GRUUs are invalidated due to a change in Call-ID, the registrar removes the current mapping from I<sub>i</sub> to the AOR and instance ID, notes the current value of the counter I<sub>j</sub>, and stores a mapping from I<sub>j</sub> to the contact containing a "bnc" parameter and instance ID. Based on these rules, the hash-map will contain a single mapping for each contact containing a "bnc" parameter and instance ID for which there is a currently valid registration.

The registrar maintains a symmetric key SK\_a, which is regenerated every time the counter rolls over or is reset. When the counter rolls over or is reset, the registrar remembers the old value of SK\_a for a while. To generate a temp-gruu-cookie, the registrar computes:

```
SA = HMAC(SK_a, I_i)
temp-gruu-cookie = base64enc(I_i || SA)
```

where || denotes concatenation. "HMAC" represents any suitably strong HMAC algorithm; see RFC 2104 [1] for a discussion of HMAC algorithms. One suitable HMAC algorithm for this purpose is HMAC-SHA256-80.

#### 7.1.2.2. Generation of temp-gruu by the SIP-PBX

According to RFC5627 [20] section 3.2, every registration refresh generates a new temp-gruu that is valid for as long as the contact remains registered. This property is both critical for the privacy properties of temp-gruu and is expected by UAs that implement the temp-gruu procedures. Nothing in this document should be construed as changing this fundamental temp-gruu property in any way. SIP-PBXes that implement temporary GRUUs MUST generate a new temp-gruu according to the procedures in this section for every registration or registration refresh from GRUU-supporting UAs attached to the SIP-PBX.

Similarly, if the registration that a SIP-PBX has with its SSP expires or is terminated, then the temp-gruu cookie it maintains with the SSP will change. This change will invalidate all the temp-gruus the SIP-PBX has issued to its UAs. If the SIP-PBX tracks this information (e.g., to include <temp-gruu> elements in registration event bodies, as described in RFC 5628 [9]), it can determine that previously issued temp-gruus are invalid by observing a change in the temp-gruu-cookie provided to it by the SSP.

A SIP-PBX that issues temporary GRUUs to its UAs MUST maintain an HMAC key, PK\_a. This value is used to validate that incoming GRUUs were generated by the SIP-PBX.

To generate a new temporary GRUU for use by its own UAs, the SIP-PBX MUST generate a random distinguisher value D. The length of this value is up to implementors, but MUST be long enough to prevent collisions among all the temporary GRUUs issued by the SIP-PBX. A size of 80 bits or longer is RECOMMENDED. See RFC 4086 [16] for further considerations on the generation of random numbers in a

security context. After generating the distinguisher D, the SIP-PBX then MUST calculate:

```
M    = base64dec(SSP-cookie) || D
E    = RSA-Encrypt(K_e2, M)
PA   = HMAC(PK_a, E)
```

```
Temp-Gruu-userpart = "tgruu." || base64(E) || "." || base64(PA)
```

where || denotes concatenation. "HMAC" represents any suitably strong HMAC algorithm; see RFC 2104 [1] for a discussion of HMAC algorithms. One suitable HMAC algorithm for this purpose is HMAC-SHA256-80.

Finally, the SIP-PBX adds a "gr" parameter to the temporary GRUU that can be used to uniquely identify the UA registration record to which the GRUU corresponds. The means of generation of the "gr" parameter are left to the implementor, as long as they satisfy the properties of a GRUU as described in RFC 5627 [20].

One valid approach for generation of the "gr" parameter is calculation of "E" and "A" as described in Appendix A.2 of RFC 5627 [20], and forming the "gr" parameter as:

```
gr = base64enc(E) || base64enc(A)
```

Using this procedure may result in a temporary GRUU returned to the registering UA by the SIP-PBX that looks similar to this:

```
<allOneLine>
Contact: <sip;line-1@10.20.1.17>
;temp-gruu="sip:tgruu.MQyaRiLEd78RtaWkcP7N8Q.5qVbsasdo2pkKw@
ssp.example.com;gr=YZGSCjKD42ccx008pA7HwAM4XNDIlMSL0H1A"
;+sip.instance="urn:uuid:d0e2f290-104b-11df-8a39-0800200c9a66">
;expires=3600
</allOneLine>
```

#### 7.1.2.3. Decoding of temp-gruu by the SSP

When the SSP proxy receives a request in which the user part begins with "tgruu.", it extracts the remaining portion, and splits it at the "." character into E' and PA'. It discards PA'. It then computes E by performing a base64 decode of E'. Next, it computes:

$M = \text{RSA-Decrypt}(K_{e1}, E)$

The SSP proxy extracts the fixed-length temp-gruu-cookie information from the beginning of this M, and discards the remainder (which will be the distinguisher added by the SIP-PBX). It then validates this temp-gruu-cookie. If valid, it uses it to locate the corresponding SIP-PBX registration record, and routes the message appropriately.

If the non-normative, exemplary algorithm described in Section 7.1.2.1 is used to generate the temp-gruu-cookie, then this identification is performed by splitting the temp-gruu-cookie information into its 48-bit counter I and 80-bit HMAC. It validates that the HMAC matches the counter I, and then uses counter I to locate the SIP-PBX registration record in its map. If the counter has rolled over or reset, this computation is performed with the current and previous SK\_a.

#### 7.1.2.4. Decoding of temp-gruu by the SIP-PBX

When the SIP-PBX receives a request in which the user part begins with "tgruu.", it extracts the remaining portion, and splits it at the "." character into E' and PA'. It then computes E and PA by performing a base64 decode of E' and PA' respectively. Next, it computes:

$PAC = \text{HMAC}(PK_a, E)$

where HMAC is the HMAC algorithm used for the steps in Section 7.1.2.2. If this computed value for PAC does not match the value of PA extracted from the GRUU, then the GRUU is rejected as invalid.

The SIP-PBX then uses the value of the "gr" parameter to locate the UA registration to which the GRUU corresponds, and routes the message accordingly.

## 7.2. Registration Event Package

Neither the SSP nor the SIP-PBX is required to support the Registration event package defined by RFC 3680 [12]. However, if they do support the Registration event package, they MUST conform to the behavior described in this section and its subsections.

As this mechanism inherently deals with REGISTER transaction behavior, it is imperative to consider its impact on the Registration

Event Package defined by RFC 3680 [12]. In practice, there will be two main use cases for subscribing to registration data: learning about the overall registration state for the SIP-PBX, and learning about the registration state for a single SIP-PBX AOR.

#### 7.2.1. SIP-PBX Aggregate Registration State

If the SIP-PBX (or another interested and authorized party) wishes to monitor or audit the registration state for all of the AORs currently registered to that SIP-PBX, it can subscribe to the SIP registration event package at the SIP-PBX's main URI -- that is, the URI used in the "To" header field of the REGISTER request.

The NOTIFY messages for such a subscription will contain a body that contains one record for each AOR associated with the SIP-PBX. The AORs will be in the format expected to be received by the SSP (e.g., "sip:+12145550105@ssp.example.com"), and the Contacts will correspond to the mapped Contact created by the registration (e.g., "sip:+12145550105@98.51.100.3").

In particular, the "bnc" parameter is forbidden from appearing in the body of a reg-event NOTIFY request unless the subscriber has indicated knowledge of the semantics of the "bnc" parameter. The means for indicating this support are out of scope of this document.

Because the SSP does not necessarily know which GRUUs have been issued by the SIP-PBX to its associated UAs, these records will not generally contain <temp-gruu> or <pub-gruu> elements defined in RFC 5628 [9]. This information can be learned, if necessary, by subscribing to the individual AOR registration state, as described in Section 7.2.2.

#### 7.2.2. Individual AOR Registration State

As described in Section 6, the SSP will generally retarget all requests addressed to an AOR owned by a SIP-PBX to that SIP-PBX according to the mapping established at registration time. Although policy at the SSP may override this generally expected behavior, proper behavior of the registration event package requires that all "reg" event SUBSCRIBE requests are processed by the SIP-PBX. As a consequence, the requirements on an SSP for processing registration event package SUBSCRIBE requests are not left to policy.

If the SSP receives a SUBSCRIBE request for the registration event package with a Request-URI that indicates an AOR registered via the "Bulk Number Contact" mechanism defined in this document, then the SSP MUST proxy that SUBSCRIBE to the SIP-PBX in the same way that it would proxy an INVITE bound for that AOR, unless the SSP has and can



maintain a copy of complete, accurate, and up-to-date information from the SIP-PBX (e.g., through an active back-end subscription).

If the Request-URI in a SUBSCRIBE request for the registration event package indicates a contact that is registered by more than one SIP-PBX, then the SSP proxy will fork the SUBSCRIBE request to all the applicable SIP-PBXes. Similarly, if the Request-URI corresponds to a contact that is both implicitly registered by a SIP-PBX and explicitly registered directly with the SSP proxy, then the SSP proxy will semantically fork the SUBSCRIBE request to the applicable SIP-PBX or SIP-PBXes and to the registrar function (which will respond with registration data corresponding to the explicit registrations at the SSP). The forking in both of these cases can be avoided if the SSP has and can maintain a copy of up-to-date information from the PBXes.

Section 4.9 of RFC 3680 [12] indicates that "a subscriber MUST NOT create multiple dialogs as a result of a single [registration event] subscription request." Consequently, subscribers who are not aware of the extension described by this document will accept only one dialog in response to such requests. In the case described in the preceding paragraph, this behavior will result in such client receiving accurate but incomplete information about the registration state of an AOR. As an explicit change to the normative behavior of RFC 3680, this document stipulates that subscribers to the registration event package MAY create multiple dialogs as the result of a single subscription request. This will allow subscribers to create a complete view of an AOR's registration state.

Defining the behavior as described above is important, since the reg-event subscriber is interested in finding out about the comprehensive list of devices associated with the AOR. Only the SIP-PBX will have authoritative access to this information. For example, if the user has registered multiple UAs with differing capabilities, the SSP will not know about the devices or their capabilities. By contrast, the SIP-PBX will.

If the SIP-PBX is not registered with the SSP when a registration event subscription for a contact that would be implicitly registered if the SIP-PBX were registered, then the SSP SHOULD accept the subscription and indicate that the user is not currently registered. Once the associated SIP-PBX is registered, the SSP SHOULD use the subscription migration mechanism defined in RFC 3265 [5] to migrate the subscription to the SIP-PBX.

When a SIP-PBX receives a registration event subscription addressed to an AOR that has been registered using the bulk registration mechanism described in this document, then each resulting

registration information document SHOULD contain an 'aor' attribute in its <registration/> element that corresponds to the AOR at the SSP.

For example, consider a SIP-PBX that has registered with an SSP that has a domain of "ssp.example.com". The SIP-PBX used a contact of "sip:198.51.100.3:5060;bnc". After such registration is complete, a registration event subscription arriving at the SSP with a Request-URI of "sip:+12145550102@ssp.example.com" will be re-targeted to the SIP-PBX, with a Request-URI of "sip:+12145550102@198.51.100.3:5060". The resulting registration document created by the SIP-PBX would contain a <registration/> element with an "aor" attribute of "sip:+12145550102@ssp.example.com".

This behavior ensures that subscribers external to the system (and unaware of GIN procedures) will be able to find the relevant information in the registration document (since they will be looking for the publicly-visible AOR, not the address used for sending information from the SSP to the SIP-PBX).

A SIP-PBX that supports both GRUU procedures and the registration event packages SHOULD implement the extension defined in RFC 5628 [9].

### 7.3. Client-Initiated (Outbound) Connections

RFC 5626 [19] defines a mechanism that allows UAs to establish long-lived TCP connections or UDP associations with a proxy in a way that allows bidirectional traffic between the proxy and the UA. This behavior is particularly important in the presence of NATs, and whenever TLS [18] security is required. Neither the SSP nor the SIP-PBX is required to support client-initiated connections.

The outbound mechanism generally works with the solution defined in this document without any modifications. Implementors should note that the instance ID used between the SIP-PBX and the SSP's registrar identifies the SIP-PBX itself, and not any of the UAs registered with the SIP-PBX. As a consequence, any attempts to use caller preferences (defined in RFC 3841[14]) to target a specific instance are likely to fail. This shouldn't be an issue, as the preferred mechanism for targeting specific instances of a user agent is GRUU (see Section 7.1).

### 7.4. Non-Adjacent Contact Registration (Path) and Service Route Discovery

RFC 3327 [10] defines a means by which a registrar and its associated

proxy can be informed of a route that is to be used between the proxy and the registered user agent. The scope of the route created by a "Path" header field is contact-specific; if an AOR has multiple contacts associated with it, the routes associated with each contact may be different from each other. Support for non-adjacent contact registration is required in all SSPs and SIP-PBXes implementing the multiple-AOR-registration protocol described in this document.

At registration time, any proxies between the user agent and the registrar may add themselves to the Path. By doing so, they request that any requests destined to the user agent as a result of the associated registration include them as part of the Route towards the User Agent. Although the Path mechanism does deliver the final Path value to the registering UA, UAs typically ignore the value of the Path.

To provide similar functionality in the opposite direction -- that is, to establish a route for requests sent by a registering UA -- RFC 3608 [11] defines a means by which a UA can be informed of a route that is to be used by the UA to route all outbound requests associated with the AOR used in the registration. This information is scoped to the AOR within the UA, and is not specific to the Contact (or Contacts) in the REGISTER request. Support of service route discovery is OPTIONAL in SSPs and SIP-PBXes.

The registrar unilaterally generates the values of the service route using whatever local policy it wishes to apply. Although it is common to use the Path and/or Route information in the request in composing the Service-Route, registrar behavior is not constrained in any way that requires it to do so.

In considering the interaction between these mechanisms and the registration of multiple AORs in a single request, implementors of proxies, registrars, and intermediaries must keep in mind the following issues, which stem from the fact that GIN effectively registers multiple AORs and multiple Contacts.

First, all location service records that result from expanding a single Contact containing a "bnc" parameter will necessarily share a single path. Proxies will be unable to make policy decisions on a contact-by-contact basis regarding whether to include themselves in the path. Second, and similarly, all AORs on the SIP-PBX that are registered with a common REGISTER request will be forced to share a common Service-Route.

One interesting technique that Path and Service-Route enable is the inclusion of a token or cookie in the user portion of the Service-Route or Path entries. This token or cookie may convey information

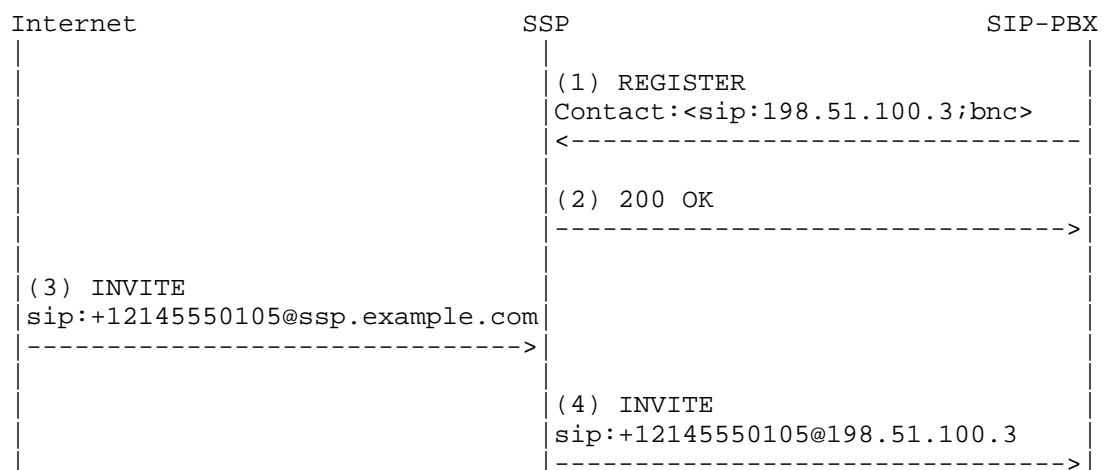
to proxies about the identity, capabilities, and/or policies associated with the user. Since this information will be shared among several AORs and several Contacts when multiple AOR registration is employed, care should be taken to ensure that doing so is acceptable for all AORs and all Contacts registered in a single REGISTER request.

## 8. Examples

Note that the following examples elide any steps related to authentication. This is done for the sake of clarity. Actual deployments will need to provide a level of authentication appropriate to their system.

### 8.1. Usage Scenario: Basic Registration

This example shows the message flows for a basic bulk REGISTER transaction, followed by an INVITE addressed to one of the registered UAs. Example messages are shown after the sequence diagram.



- (1) The SIP-PBX registers with the SSP for a range of AORs.

```
REGISTER sip:ssp.example.com SIP/2.0
Via: SIP/2.0/UDP 198.51.100.3:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
To: <sip:pbx@ssp.example.com>
From: <sip:pbx@ssp.example.com>;tag=a23589
Call-ID: 843817637684230@998sdasdh09
CSeq: 1826 REGISTER
Proxy-Require: gin
Require: gin
Supported: path
Contact: <sip:198.51.100.3:5060;bnc>
Expires: 7200
Content-Length: 0
```

- (3) The SSP receives a request for an AOR assigned to the SIP-PBX.

```
INVITE sip:+12145550105@ssp.example.com SIP/2.0
Via: SIP/2.0/UDP foo.example;branch=z9hG4bKa0bc7a0131f0ad
Max-Forwards: 69
To: <sip:2145550105@some-other-place.example.net>
From: <sip:gsmith@example.org>;tag=456248
Call-ID: f7aecbfc374d557baf72d6352e1fbcd4
CSeq: 24762 INVITE
Contact: <sip:line-1@192.0.2.178:2081>
Content-Type: application/sdp
Content-Length: ...

<sdp body here>
```

- (4) The SSP retargets the incoming request according to the information received from the SIP-PBX at registration time.

```

INVITE sip:+12145550105@198.51.100.3 SIP/2.0
Via: SIP/2.0/UDP ssp.example.com;branch=z9hG4bKa45cd5c52a6dd50
Via: SIP/2.0/UDP foo.example.com;branch=z9hG4bKa0bc7a0131f0ad
Max-Forwards: 68
To: <sip:2145550105@some-other-place.example.net>
From: <sip:gsmith@example.org>;tag=456248
Call-ID: f7aecbfc374d557baf72d6352elfbcd4
CSeq: 24762 INVITE
Contact: <sip:line-1@192.0.2.178:2081>
Content-Type: application/sdp
Content-Length: ...

<sdp body here>

```

## 8.2. Usage Scenario: Using Path to Control Request URI

This example shows a bulk REGISTER transaction with the SSP making use of the "Path" header field extension [10]. This allows the SSP to designate a domain on the incoming Request URI that does not necessarily resolve to the SIP-PBX when the SSP applies RFC 3263 procedures to it.



- (1) The SIP-PBX registers with the SSP for a range of AORs. It includes the form of the URI it expects to receive in the Request-URI in its "Contact" header field, and includes information that routes to the SIP-PBX in the "Path" header field.

```
REGISTER sip:ssp.example.com SIP/2.0
Via: SIP/2.0/UDP 198.51.100.3:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
To: <sip:pbx@ssp.example.com>
From: <sip:pbx@ssp.example.com>;tag=a23589
Call-ID: 326983936836068@998sdasdh09
CSeq: 1826 REGISTER
Proxy-Require: gin
Require: gin
Supported: path
Path: <sip:pbx@198.51.100.3:5060;lr>
Contact: <sip:pbx.example;bnc>
Expires: 7200
Content-Length: 0
```

- (3) The SSP receives a request for an AOR assigned to the SIP-PBX.

```
INVITE sip:+12145550105@ssp.example.com SIP/2.0
Via: SIP/2.0/UDP foo.example;branch=z9hG4bKa0bc7a0131f0ad
Max-Forwards: 69
To: <sip:2145550105@some-other-place.example.net>
From: <sip:gsmith@example.org>;tag=456248
Call-ID: 7ca24b9679ffe9aff87036a105e30d9b
CSeq: 24762 INVITE
Contact: <sip:line-1@192.0.2.178:2081>
Content-Type: application/sdp
Content-Length: ...
```

<sdp body here>

- (4) The SSP retargets the incoming request according to the information received from the SIP-PBX at registration time. Per the normal processing associated with "Path," it will insert the "Path" value indicated by the SIP-PBX at registration time in a "Route" header field, and set the request URI to the registered Contact.

```
INVITE sip:+12145550105@pbx.example SIP/2.0
Via: SIP/2.0/UDP ssp.example.com;branch=z9hG4bKa45cd5c52a6dd50
Via: SIP/2.0/UDP foo.example.com;branch=z9hG4bKa0bc7a0131f0ad
Route: <sip:pbx@198.51.100.3:5060;lr>
Max-Forwards: 68
To: <sip:2145550105@some-other-place.example.net>
From: <sip:gsmith@example.org>;tag=456248
Call-ID: 7ca24b9679ffe9aff87036a105e30d9b
CSeq: 24762 INVITE
Contact: <sip:line-1@192.0.2.178:2081>
Content-Type: application/sdp
Content-Length: ...
```

<sdp body here>

## 9. IANA Considerations

This document registers a new SIP option tag to indicate support for the mechanism it defines, two new SIP URI parameters, and a "Contact" header field parameter. The process governing registration of these protocol elements is outlined in RFC5727 [21].

### 9.1. New SIP Option Tag

This section defines a new SIP option tag per the guidelines in Section 27.1 of RFC 3261 [3].

Name: gin

Description: This option tag is used to identify the extension that provides Registration for Multiple Phone Numbers in SIP. When present in a Require or Proxy-Require header field of a REGISTER request, it indicates that support for this extension is required of registrars and proxies, respectively, that are a party to the registration transaction.

Reference: RFCXXXX (this document)

### 9.2. New SIP URI Parameters

This specification defines two new SIP URI parameters, as per the registry created by RFC 3969 [7].



#### 9.2.1. 'bnc' SIP URI parameter

Parameter Name: bnc  
Predefined Values: No (no values are allowed)  
Reference: RFCXXXX (this document)

#### 9.2.2. 'sg' SIP URI parameter

Parameter Name: sg  
Predefined Values: No  
Reference: RFCXXXX (this document)

#### 9.3. New SIP Header Field Parameter

This section defines a new SIP header field parameter per the registry created by RFC3968 [6].

Header field: Contact  
Parameter name: temp-gruu-cookie  
Predefined values: none  
Reference: RFCXXXX (this document)

### 10. Security Considerations

The change proposed for the mechanism described in this document takes the unprecedented step of extending the previously-defined REGISTER method to apply to more than one AOR. In general, this kind of change has the potential to cause problems at intermediaries -- such as proxies -- that are party to the REGISTER transaction. In particular, such intermediaries may attempt to apply policy to the user indicated in the "To" header field (i.e. the SIP-PBX's identity), without any knowledge of the multiple AORs that are being implicitly registered.

The mechanism defined by this document solves this issue by adding an option tag to a "Proxy-Require" header field in such REGISTER requests. Proxies that are unaware of this mechanism will not process the requests, preventing them from mis-applying policy. Proxies that process requests with this option tag are clearly aware of the nature of the REGISTER request, and can make reasonable policy decisions.

As noted in Section 7.4, intermediaries need to take care if they use a policy token in the Path and Service-Route mechanisms, as doing so will cause them to apply the same policy to all users serviced by the same SIP-PBX. This may frequently be the correct behavior, but circumstances can arise in which differentiation of user policy is

required.

Section 7.4 also notes that techniques that use a token or cookie in the Path and/or Service-Route values, and that this value will be shared among all AORs associated with a single registration. Because this information will be visible to User Agents under certain conditions, proxy designers using this mechanism in conjunction with the techniques describe in this document need to take care that doing so does not leak sensitive information.

One of the key properties of the outbound client connection mechanism discussed in Section 7.3 is assurances that a single connection is associated with a single user, and cannot be hijacked by other users. With the mechanism defined in this document, such connections necessarily become shared between users. However, the only entity in a position to hijack calls as a consequence is the SIP-PBX itself. Because the SIP-PBX acts as a registrar for all the potentially affected users, it already has the ability to redirect any such communications as it sees fit. In other words, the SIP-PBX must be trusted to handle calls in an appropriate fashion, and the use of the outbound connection mechanism introduces no additional vulnerabilities.

The ability to learn the identity and registration state of every user on the PBX (as described in Section 7.2.1) is invaluable for diagnostic and administrative purposes. For example, this allows the SIP-PBX to determine whether all the its extensions are properly registered with the SSP. However, this information can also be highly sensitive, as many organizations may not wish to make their entire list of phone numbers available to external entities. Consequently, SSP servers are advised to use explicit (i.e. white-list) and configurable policies regarding who can access this information, with very conservative defaults (e.g., an empty access list or an access list consisting only of the PBX itself).

The procedure for generation of temporary GRUUs requires the use of an HMAC to detect any tampering that external entities may attempt to perform on the contents of a temporary GRUU. The mention of HMAC-SHA256-80 in Section 7.1.2 is intended solely as an example of a suitable HMAC algorithm. Since all HMACs used in this document are generated and consumed by the same entity, the choice of an actual HMAC algorithm is entirely up to an implementation, provided that the cryptographic properties are sufficient to prevent third parties from spoofing GRUU-related information.

The procedure for generation of temporary GRUUs also requires the use of RSA keys. The selection of the proper key length for such keys requires careful analysis, taking into consideration the current and

foreseeable speed of processing for the period of time during which GRUUs must remain anonymous, as well as emerging cryptographic analysis methods. The most recent guidance from RSA Laboratories [25] suggests a key length of 2048 bits for data that needs protection through the year 2030, and a length of 3072 bits thereafter.

Similarly, implementors are warned to take precautionary measures to prevent unauthorized disclosure of the private key used in GRUU generation. Any such disclosure would result in the ability to correlate temporary GRUUs to each other, and potentially to their associated PBXes.

Further, the use of RSA decryption when processing GRUUs received from arbitrary parties can be exploited by DoS attackers to amplify the impact of an attack: because of the presence of a cryptographic operation in the processing of such messages, the CPU load may be marginally higher when the attacker uses (valid or invalid) temporary GRUUs in the messages employed by such an attack. Normal DoS mitigation techniques, such as rate-limiting processing of received messages, should help to reduce the impact of this issue as well.

Finally, good security practices should be followed regarding the duration an RSA key is used. For implementors, this means that systems MUST include an easy way to update the public key provided to the SIP-PBX. To avoid immediately invalidating all currently issued temporary GRUUs, the SSP servers SHOULD keep the retired RSA key around for a grace period before discarding it. If decryption fails based on the new RSA key, then the SSP server can attempt to use the retired key instead. By contrast, the SIP-PBXes MUST discard the retired public key immediately, and exclusively use the new public key.

## 11. Acknowledgements

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#### Appendix A. Requirements Analysis

The document "Requirements for multiple address of record (AOR) reachability information in the Session Initiation Protocol (SIP)" [22] contains a list of requirements and desired properties for a mechanism to register multiple AORs with a single SIP transaction. This section evaluates those requirements against the mechanism described in this document.

REQ1 - The mechanism MUST allow a SIP-PBX to enter into a trunking arrangement with an SSP whereby the two parties have agreed on a set of telephone numbers deemed to have been assigned to the SIP-PBX.

The requirement is satisfied.

REQ2 - The mechanism MUST allow a set of assigned telephone numbers to comprise E.164 numbers, which can be in contiguous ranges, discrete, or in any combination of the two.

The requirement is satisfied; the DIDs associated with a registration is established by bilateral agreement between the SSP and the SIP-PBX, and is not part of the mechanism described in this document.

REQ3 - The mechanism MUST allow a SIP-PBX to register reachability information with its SSP, in order to enable the SSP to route to the SIP-PBX inbound requests targeted at assigned telephone numbers.

The requirement is satisfied.

REQ4 - The mechanism MUST allow UAs attached to a SIP-PBX to register with the SIP-PBX for AORs based on assigned telephone numbers, in order to receive requests targeted at those telephone numbers, without needing to involve the SSP in the registration process.

The requirement is satisfied; in the presumed architecture, SIP-PBX UAs register with the SIP-PBX, and require no interaction with the SSP.

REQ5 - The mechanism MUST allow a SIP-PBX to handle requests originating at its own UAs and targeted at its assigned telephone

numbers, without routing those requests to the SSP.

The requirement is satisfied; SIP-PBXes may recognize their own DID and their own GRUUs, and perform on-SIP-PBX routing without sending the requests to the SSP.

REQ6 - The mechanism MUST allow a SIP-PBX to receive requests to its assigned telephone numbers originating outside the SIP-PBX and arriving via the SSP, so that the SIP-PBX can route those requests onwards to its UAs, as it would for internal requests to those telephone numbers.

The requirement is satisfied

REQ7 - The mechanism MUST provide a means whereby a SIP-PBX knows which of its assigned telephone numbers an inbound request from its SSP is targeted at.

The requirement is satisfied. For ordinary calls and calls using Public GRUUs, the DID is indicated in the user portion of the Request-URI. For calls using Temp GRUUs constructed with the mechanism described in Section 7.1.2, the "gr" parameter provides a correlation token the SIP-PBX can use to identify which UA the call should be routed to.

REQ8 - The mechanism MUST provide a means of avoiding problems due to one side using the mechanism and the other side not.

The requirement is satisfied through the 'gin' option tag and the 'bnc' Contact parameter.

REQ9 - The mechanism MUST observe SIP backwards compatibility principles.

The requirement is satisfied through the 'gin' option tag.

REQ10 - The mechanism MUST work in the presence of a sequence of intermediate SIP entities on the SIP-PBX-to-SSP interface (i.e., between the SIP-PBX and the SSP's domain proxy), where those intermediate SIP entities indicated during registration a need to be on the path of inbound requests to the SIP-PBX.

The requirement is satisfied through the use of the Path mechanism defined in RFC 3327 [10]

REQ11 - The mechanism MUST work when a SIP-PBX obtains its IP address dynamically.

The requirement is satisfied by allowing the SIP-PBX to use an IP address in the Bulk Number Contact URI contained in a REGISTER Contact header field.

REQ12 - The mechanism MUST work without requiring the SIP-PBX to have a domain name or the ability to publish its domain name in the DNS.

The requirement is satisfied by allowing the SIP-PBX to use an IP address in the Bulk Number Contact URI contained in a REGISTER Contact header field.

REQ13 - For a given SIP-PBX and its SSP, there MUST be no impact on other domains, which are expected to be able to use normal RFC 3263 procedures to route requests, including requests needing to be routed via the SSP in order to reach the SIP-PBX.

The requirement is satisfied by allowing the domain name in the Request URI used by external entities to resolve to the SSP's servers via normal RFC 3263 resolution procedures.

REQ14 - The mechanism MUST be able to operate over a transport that provides end-to-end integrity protection and confidentiality between the SIP-PBX and the SSP, e.g., using TLS as specified in [3].

The requirement is satisfied; nothing in the proposed mechanism prevent the use of TLS between the SSP and the SIP-PBX.

REQ15 - The mechanism MUST support authentication of the SIP-PBX by the SSP and vice versa, e.g., using SIP digest authentication plus TLS server authentication as specified in [3].

The requirement is satisfied; SIP-PBXes may employ either SIP digest authentication or mutually-authenticated TLS for authentication purposes.

REQ16 - The mechanism MUST allow the SIP-PBX to provide its UAs with public or temporary Globally Routable UA URIs (GRUUs) [20].

The requirement is satisfied via the mechanisms detailed in Section 7.1.

REQ17 - The mechanism MUST work over any existing transport specified for SIP, including UDP.

The requirement is satisfied to the extent that UDP can be used for REGISTER requests in general. The application of certain extensions and/or network topologies may exceed UDP MTU sizes, but such issues arise both with and without the mechanism described in



this document. This document does not exacerbate such issues.

REQ18 - Documentation MUST give guidance or warnings about how authorization policies may be affected by the mechanism, to address the problems described in Section 3.3 (of RFC5947).

These issues are addressed at length in Section 10, as well as summarized in Section 7.4.

REQ19 - The mechanism MUST be extensible to allow a set of assigned telephone numbers to comprise local numbers as specified in RFC3966 [15], which can be in contiguous ranges, discrete, or in any combination of the two.

Assignment of telephone numbers to a registration is performed by the SSP's registrar, which is not precluded from assigning local numbers in any combination it desires.

REQ20 - The mechanism MUST be extensible to allow a set of arbitrarily assigned SIP URI's as specified in RFC3261 [3], as opposed to just telephone numbers, without requiring a complete change of mechanism as compared to that used for telephone numbers.

The mechanism is extensible in such a fashion, as demonstrated by the document "GIN with Literal AoRs for SIP in SSPs (GLASS)" [23].

DES1 - The mechanism SHOULD allow an SSP to exploit its mechanisms for providing SIP service to ordinary subscribers in order to provide a SIP trunking service to SIP-PBXes.

The desired property is satisfied; the routing mechanism described in this document is identical to the routing performed for singly-registered AORs.

DES2 - The mechanism SHOULD scale to SIP-PBX's of several thousand assigned telephone numbers.

The desired property is satisfied; nothing in this document precludes DID pools of arbitrary size.

DES3 - The mechanism SHOULD scale to support several thousand SIP-PBX's on a single SSP.

The desired property is satisfied; nothing in this document precludes an arbitrary number of SIP-PBXes from attaching to a single SSP.

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SIP Verification with Event-package  
for Resolution of Managed Open-ended Username Target Handles  
(VERMOUTH)  
draft-kaplan-martini-vermouth-01

Status of this Memo

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## Abstract

The Martini Working Group is defining a mechanism for SIP IP-PBX type devices to REGISTER and obtain SIP service for E.164-based Address of Records, using the GIN mechanism defined in [draft-gin]. Two other drafts, [draft-olive] and [draft-glass], propose the same for non-E.164-based AoRs. This document defines a means by which the IP-PBX can verify the resolution entries in the SSP for open-ended or full AoRs of any GIN-based mechanism, using a new Event-Package named "vermouth".

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## 1. Introduction

In many deployed SIP Service Provider (SSP) architectures, it is common to use REGISTER requests to provide the reachability information for IP-PBXs, instead of DNS-based resolution and routing. An IETF-defined mechanism for doing so is defined in [draft-gin]. Another draft, [draft-olive], uses the [draft-gin] GIN mechanism for Local-Number AoRs as well; and a new draft [draft-glass] does the same for literal alpha-numeric/email-style AoRs.

In all cases, the IP-PBX or another SIP entity may wish to learn about all of the AoRs which were implicitly Registered by [draft-gin] or [draft-olive], or to learn about changes in their provisioned AoRs through asynchronous notifications. Even in non-Registration scenarios, where requests for specific AoRs in a SSP may instead be statically routed to an IP-PBX, it may be useful for the IP-PBX to learn what those AoRs are in order to detect mismatches or changes.

In theory, the [draft-gin] mechanism is simply a short-hand single REGISTER transaction for a bulk set of AoRs in lieu of multiple, separate REGISTER transactions for each AoR. In practice, however, the E.164 user numbers may be an "open" numbering plan/range, such that the SSP only really knows about a certain number of digits and the rest are only known to the IP-PBX. Likewise, when [draft-olive] is used, the Local-Number may be only partially known to the SSP.

Therefore, it is not possible for the SSP to actually provide state information for each possible unique AoR instance. Instead, it needs to provide an indication for the registration state of the prefix or digit portion it does know about.

This document proposes to provide such information using a new Event-Package.

## 2. Definitions

For brevity's sake, this document uses the word "request" instead of "out-of-dialog request", but in all case means out-of-dialog request.

AoR: address-of-record, as defined by RFC 3261: a URI by which the user is canonically known (e.g., on their business cards, in the From header field of their requests, in the To header field of REGISTER requests, etc.).

Bulk-AoR: a SIP or SIPS address-of-record with a "range" URI user parameter which expands the user string based on a heuristic.

Local-Number: an AoR which follows the form of local-number in [RFC3966], but may be encoded in a SIP or TEL URI. The local-number contains a 'phone-context' parameter identifying the scope of its number.

Email-style URI: a SIP AoR which does not identify a global E.164 number or Local-Number.

Implicit Registration: implicitly providing the reachability information for something other than the AoR explicitly indicated in the Register transaction.

Reachability Information: a set of URI's identifying the host and path of Proxies to reach that host; like any URI, these URI's may identify the specific connection transport, IP Address, and port information, or they may only identify FQDN's.

SSP: SIP Service Provider, as defined by [RFC5486].

### 3. The Solution - an Overview

The general concept is a SIP device, such as an IP-PBX, Subscribes to a new "vermouth" Event-Package by issuing a SUBSCRIBE request targeted at the SIP URI AoR it explicitly registered using GIN, or some other mutually-agreed-upon SIP-URI if GIN was not used.

If the Subscription is successful, the returned NOTIFY contains a userinfo XML document that lists all of the usernames of the AoR's domain that the SSP will route to the IP-PBX. The XML document does not contain the Contact/Path routing reachability information, since that information is already in the reg-event package information for the explicitly registered AoR of the IP-PBX, and may also be more sensitive in nature.

To handle the open-numbering-plan problem, an XML "range" attribute is used, which is similar to a regular expression pattern but with a very limited, specified syntax. The limited syntax is used to avoid ambiguities and reduce confusion - rationale for this is provided in Appendix A.

Furthermore, this document specifies that the To-URI used for the [draft-gin] REGISTER request, be usable as the target for the SUBSCRIBE request, both for the new 'vermouth' Event-Package, and for Subscribing to the [RFC3680] registration event-package for that explicitly registered AoR.

### 4. Event Package Definition

This section fills in the details needed to specify an event package as defined in Section 4.4 of [RFC3265].

#### 4.1. Event Package Name

The SIP Events specification requires package definitions to specify the name of their package or template-package.

The name of this package is "vermouth". As specified in [RFC3265], this value appears in the Event header present in SUBSCRIBE and NOTIFY requests.

#### 4.2. Event Package Parameters

The SIP Events specification requires package and template-package definitions to specify any package specific parameters of the Event header that are used by it.

No package specific Event header parameters are defined for this event package.

#### 4.3. SUBSCRIBE Bodies

The SIP Events specification requires package or template-package definitions to define the usage, if any, of bodies in SUBSCRIBE requests.

A SUBSCRIBE for registration events MAY contain a body. This body would serve the purpose of filtering the subscription. The definition of such a body is outside the scope of this specification.

A SUBSCRIBE for the registration package MAY be sent without a body. This implies that the default registration filtering policy has been requested. The default policy is:

- o Notifications are generated every time there is any change in the state of any of the registered contacts for the resource being subscribed to. Those notifications only contain information on the contacts whose state has changed.

- o Notifications triggered from a SUBSCRIBE contain full state (the list of all contacts bound to the address-of-record).

Of course, the server can apply any policy it likes to the subscription.

#### 4.4. Subscription Duration

The SIP Events specification requires package definitions to define a default value for subscription durations, and to discuss reasonable choices for durations when they are explicitly specified.

The Event Package defined herein is not tied to registration state, nor to any value that has natural expiry times. Therefore, the suggested subscription duration is 86400 seconds (1 day).



Of course, clients MAY include an Expires header in the SUBSCRIBE request asking for a different duration.

#### 4.5. NOTIFY Bodies

The SIP Events specification requires package definitions to describe the allowed set of body types in NOTIFY requests, and to specify the default value to be used when there is no Accept header in the SUBSCRIBE request.

The body of a notification of a change in provisioned usernames contains a user information document. This document describes some or all of the username expansions associated with the particular address-of-record subscribed to. All subscribers and notifiers MUST support the "application/userinfo+xml" format described in Section 5. The subscribe request MAY contain an Accept header field. If no such header field is present, it has a default value of "application/userinfo+xml". If the header field is present, it MUST include "application/userinfo+xml", and MAY include any other types capable of representing registration information.

Of course, the notifications generated by the server MUST be in one of the formats specified in the Accept header field in the SUBSCRIBE request.

#### 4.6. Notifier Processing of SUBSCRIBE Requests

The SIP Events framework specifies that packages should define any package-specific processing of SUBSCRIBE requests at a notifier, specifically with regards to authentication and authorization.

Provisioned usernames can be sensitive information. Therefore, all subscriptions to it SHOULD be authenticated and authorized before approval. Authentication MAY be performed using any of the techniques available through SIP, including digest, S/MIME, TLS or other transport specific mechanisms [1]. Authorization policy is at the discretion of the administrator, as always. However, a few recommendations can be made.

It is RECOMMENDED that an IP-PBX be allowed to subscribe to its own provisioned usernames. Such subscriptions are useful for detecting errors and changes.

#### 4.7. Notifier Generation of NOTIFY Requests

The SIP Event framework requests that packages specify the conditions under which notifications are sent for that package, and how such notifications are constructed.

Instead of delivering the full list every time a notification is sent, it is RECOMMENDED that notifications only list the username entries that have changed state (i.e., been added or removed).

Notifications triggered as a result of a fetch operation (a SUBSCRIBE with Expires of 0) or a new Subscription SHOULD result in the full list of all usernames to be present in the NOTIFY.

#### 4.8. Subscriber Processing of NOTIFY Requests

The SIP Events framework expects packages to specify how a subscriber processes NOTIFY requests in any package specific ways, and in particular, how it uses the NOTIFY requests to construct a coherent view of the state of the subscribed resource.

Typically, the NOTIFY will only contain information for usernames whose state has changed. To construct a coherent view of the total state of all usernames, the subscriber will need to combine NOTIFYs received over time. The details of this process depend on the document format used to convey registration state. Section 5 outlines the process for the application/userinfo+xml format.

#### 4.9. Handling of Forked Requests

The SIP Events framework mandates that packages indicate whether or not forked SUBSCRIBE requests can install multiple subscriptions.

Provisioned usernames are normally stored in some repository (whether it be co-located with a proxy/registrar or in a separate database). As such, there is usually a single place where the username information for a particular address-of-record is resident. This implies that a subscription for this information is readily handled by a single element with access to this repository. There is, therefore, no compelling need for a subscription to username information to fork. As a result, a subscriber MUST NOT create multiple dialogs as a result of a single subscription request. The required processing to guarantee that only a Section 4.4.9 of the SIP single dialog is established is described in Events framework [RFC3265].

#### 4.10. Rate of Notifications

The SIP Events framework mandates that packages define a maximum rate of notifications for their package.

For reasons of congestion control, it is important that the rate of notifications not become excessive. As a result, it is RECOMMENDED

that the server not generate notifications for a single subscriber at a rate faster than once every 5 seconds.

#### 4.11. State Agents

The SIP Events framework asks packages to consider the role of state agents in their design.

State agents have no role in the handling of this package.

### 5. Username Information

#### 5.1. Structure of Username Information

Username information is an XML document [4] that MUST be well-formed and SHOULD be valid. Username information documents MUST be based on XML 1.0 and MUST be encoded using UTF-8. This specification makes use of XML namespaces for identifying registration information documents and document fragments. The namespace URI for elements defined by this specification is a URN [5], using the namespace identifier ietf defined by [6] and extended by [7]. This URN is:

urn:ietf:params:xml:ns:userinfo

A username information document begins with the root element tag "userinfo". It consists of any number of "userlist" sub-elements, each of which contains the provisioning state for a particular list of usernames, associated with the address-of-record subscribed to. The username information for a particular address-of-record MUST be contained within a single "userlist" element; it cannot be spread across multiple "userlist" elements within a document. Other elements from different namespaces MAY be present for the purposes of extensibility; elements or attributes from unknown namespaces MUST be ignored.

There are two attributes associated with the "userinfo" element, both of which MUST be present:

version: This attribute allows the recipient of username information documents to properly order them. Versions start at 0, and increment by one for each new document sent to a subscriber. Versions are scoped within a subscription. Versions MUST be representable using a 32 bit integer.

state: This attribute indicates whether the document contains the full list of provisioned usernames, or whether it contains only information on those registrations which have changed since the previous document (partial).

Note that the document format explicitly allows for conveying information on multiple addresses-of-record. This enables subscriptions to groups of usernames, where such a group is identified by some kind of URI. For example, a domain might define sip:allusers@example.com as a subscribe-able resource that generates notifications when the provisioning state of any address-of-record in the domain changes.

The "userlist" element has a list of any number of "user" sub-elements, each of which contains information on a single username entry, which may itself be a range-patterned name. Other elements from different namespaces MAY be present for the purposes of extensibility; elements or attributes from unknown namespaces MUST be ignored.

There are three attributes associated with the "userlist" element, all of which MUST be present:

aor: The aor attribute contains a URI which is the address-of-record this list is associated with.

id: The id attribute identifies this list. It MUST be unique amongst all other id attributes present in other userlist elements conveyed to the subscriber within the scope of their subscription. Furthermore, the id attribute for a "userlist" element for a particular address-of-record MUST be the same across all notifications sent within the subscription.

state: The state attribute indicates the state of the username list. The valid values are "active" and "removed".

The "user" element contains the username. There are several attributes associated with the "contact" element which MUST be present:

id: The id attribute identifies this user name. It MUST be unique amongst all other id attributes present in other user elements conveyed to the subscriber within the scope of their subscription.

state: The state attribute indicates the state of the user name. The valid values are "active" and "removed".

type: The type attribute identifies the user name type. Valid values are "e614", "private", and "alpha".

range: the range attribute is defined in the next section.

context: the context attribute is only meaningful when the type attribute is "private", and in such a case the context identifies the context of the private name space.

## 5.2. The "range" Attribute

The range attribute's value defines the expansion of the username, using a syntax similar to regular expressions. The range pattern applies after the last character of the user element's value.

range-value = exp-char-set exp-char-count

exp-char-set = digit-char-set / any-char-set

digit-char-set = "[" dsc-begin "-" dsc-end "]"

dsc-begin = DIGIT

dsc-end = DIGIT

any-char-set = "."

exp-char-count = "{" exp-min "," exp-max "}"

exp-min = DIGIT

exp-max = DIGIT

The "digit-char-set" defines a range of digit characters, for example 0-9 or 3-5, inclusive. The "dsc-begin" digit value must be less than or equal to the "dsc-end" digit value.

The "any-char-set" defines any single character allowed in the 'user' token field of [RFC3261].

The "exp-char-count" defines a minimum and maximum number of times a character within the exp-char-set may be repeated, inclusive. The "exp-min" digit value must be less than or equal to the "exp-max" digit value.

## 6. Examples

Detailed scenario examples will be provided once the WG decides which way to go with this mechanism.

The following is an example username information document:

```
<?xml version="1.0"?>

<userinfo xmlns="urn:ietf:params:xml:ns:userinfo"
  xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
  version="23" state="full">

  <userlist aor="sip:ip-pbx1@ssp.example.com" state="active">

    <user id="76" state="active"
      type="e164">+12345678901</user>

    <user id="77" state="removed" type="alpha">bob</user>

    <user id="78" state="active" type="e614"
      range="[0-9]{4}">+1781555</user>

    <user id="79" state="active" type="private"
      range="[0-9]{4,10}"
      context="pbx.ssp.example.com"></user>

  </userlist>

</userinfo>
```

## 7. IANA Considerations

This document makes no request of IANA yet, but will if it goes forward.

## 8. Security Considerations

This section is still TBD.

## 9. Normative References

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## Appendix A - Rationale for Constraining the Expansion Pattern

This document's mechanism defines a limited set of patterns which may be used in the "<expansion>" portion of the Bulk-AoR. This is in contrast to the "Wildcarded AoR" mechanism used in some deployments, which use any regular expressions (regex) for the

pattern. One of the reasons this document restricts the regex syntax is to maintain [RFC3261] compliance, which does not allow common regex characters such as '^', '[', ']', '{', and '}' to appear in SIP URIs.

The other reason this document does not use any arbitrary regex is that one of the goals of this document is to be useful for an IP-PBX to determine provisioning mismatches. An arbitrary regex is typically useful for verifying a given input string matches the pattern, and not for actually determining the complete set of strings the regex pattern implies. In other words, a regex is useful for authenticating a given number matches the pattern, but not for determining what all of the provisioned numbers are.

For example, a regex syntax model for "sip:1234![5-9][0-9]\*!@example.com" is useful for checking if "sip:123456@example.com" is a matching number, but is extremely difficult for an IP-PBX to verify that the SSP does not include numbers the PBX does not have provisioned. The IP-PBX could check each of its locally provisioned numbers against the regex pattern, but has no clean way to determine if the set allowed by the regex is not *\*greater\** than its locally provisioned set.

Furthermore, numerous regex patterns can be used to mean the exact same set. For example "sip:1234!(5|6|7|8|9)[0-9]\*!@example.com", "sip:1234![5-9][0-9]{0,}!@example.com", "sip:1234![5-9][[:digits:]]\*!@example.com", and "sip:123!4[5-9][0-9]\*!@example.com" all represent the same set of user strings as the first regex example.

Therefore, to avoid such issues, this document uses a very narrow set of possible "patterns", which can be used for both matching and provisioning verification.



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SIP GIN MARTINI with  
Open-plan Local-number Identifier Values for Enterprises (OLIVE)  
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Status of this Memo

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## Abstract

The Martini Working Group has defined a mechanism for SIP IP-PBX type devices to REGISTER and obtain SIP service for E.164-based Address of Records, using the GIN mechanism [draft-gin]. This document defines a means for open-plan Local-Numbers to be used with Martini-based IP-PBXs.

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## 1. Introduction

In many deployed SIP Service Provider (SSP) architectures, it is common to use REGISTER requests to provide the reachability information for IP-PBXs, instead of DNS-based resolution and routing. An IETF-defined mechanism for doing so is being worked on in the Martini Working Group, with the GIN mechanism [draft-gin].

The current GIN mechanism only supports E.164-based AoRs, however in actual deployments private-extension or "local" numbers are used for hosted and carrier-provided intra-Enterprise calling services. These forms of AoRs are not supported by the current GIN mechanism. This document defines a means by which they can be supported, in a manner consistent with [RFC3261] and [draft-gin].

## 2. Definitions

For brevity's sake, this document uses the word "request" instead of "out-of-dialog request", but in all case means out-of-dialog request.

AoR: address-of-record, as defined by RFC 3261: a URI by which the user is canonically known (e.g., on their business cards, in the From header field of their requests, in the To header field of REGISTER requests, etc.).

Open-plan: an open-plan is a dialing-plan which is not constrained to be a specific set of numbers all known to the SSP; some specific numbers may be known by the SSP, and/or a beginning set of digits are known to the SSP and used to route calls to different branches

Local-Number: an AoR which follows the form of local-number in [RFC3966], but may be encoded in a SIP or TEL URI. The local-number contains a 'phone-context' parameter identifying the scope of its number.

Implicit Registration: implicitly providing the reachability information for something other than the AoR explicitly indicated in the Register transaction.

Reachability Information: a set of URI's identifying the host and path of Proxies to reach that host; like any URI, these URI's may identify the specific connection transport, IP Address, and port information, or they may only identify FQDN's.

SSP: SIP Service Provider, as defined by [RFC5486].

## 3. Background

### 3.1. The GIN Mechanism

The GIN mechanism, defined in [draft-gin], allows a SIP UA such as an IP-PBX to Register a set of E.164 AoRs in "bulk". Instead of creating a separate REGISTER transaction for every E.164 AoR, the IP-PBX sends one REGISTER request with a 'bnc' Contact URI parameter which indicates the Contact URI needs to be expanded in the Registrar's location service database. The expansion is such that each E.164 user number becomes the user portion of the registered Contact URI, one for each implicitly registered E.164 number-based AoR.

SIP Request routing to the Registered E.164 AoR then follows normal [RFC3261] procedures, replacing the Request URI with the expanded

registered Contact URI, and adding any Path information as a Route set, etc.

### 3.2. Local-Numbers

The Local-Number syntax for TEL URIs is defined in [RFC3966], such that a local-number is a set of digits, possibly with an extension or isdn-subaddress parameter, and scoped to the domain name or global-number-digits in its phone-context. In theory, a SIP UA can target its request to a Local-Number using the [RFC3966] syntax in a TEL URI or SIP URI, and have the request delivered to the UA or IP-PBX identified by the user digits for the given phone-context, which may subsequently route the request to the specific extension or isdn-subaddress.

In practice it's not that simple. Most branch-office IP-PBXs do not use the Local-Number syntax for their targets, and do not recognize such syntax if they receive it in the Request URI. Often the SSP adds the phone-context to received requests from the IP-PBX, and removes it when sending to the IP-PBX. The reasons for this include: (a) the IP-PBX is wholly within the context and thus has no knowledge of, nor concern that there could be, other contexts; (b) the IP-PBX may not actually know all the numbers in the private number plan, only its local ones; and (c) historically it has not been necessary for them to add such explicit indicators for things to work, and thus the status-quo is difficult to change.

Furthermore, Local-Numbers are difficult because they are doubly-scoped: once at the URI level by the domain name, and internally by the phone-context URI user parameter. The authoritative system for the Local-Number user portion (the system(s) which knows what they are and how to process them) is not necessarily identified by the URI's domain name, but rather may be identified by the phone-context's value. In other words, the SSP may not know about all possible Local-Number numbers, and even a given IP-PBX may not know them all for its Enterprise; the knowledge may be distributed. This presents difficulties for certain GIN functions such as reg-event, and is why this document refers to GIN support for Local-Numbers as being for "open-plan" scenarios.

## 4. The Solution - an Overview

The general concept proposed in this document is to logically apply GIN for the complete set of Local-Number "AoRs" of the Registered-to domain, as if they were individually Registered. The GIN-based REGISTER request would cause the Registrar to logically populate the set of AoR-to-Contact bindings, as it did before.

The Contact URI user portion would also be "expanded", using the same user portion as that of the implicitly registered AoRs: namely a Local-Number format. The Local-Number username is "normalized" in the same manner as [draft-gin].

Note that the list of Local-Number AoRs associated with a PBX is a matter of local provisioning at the SSP and at the PBX, as it was in [draft-gin]. The mechanism defined in this document does not provide any means to detect or recover from provisioning mismatches (although the registration event package can be used as a standardized means for auditing such AoRs).

No new option-tag is required, because this document's mechanism does not require any changes in GIN [draft-gin] registration nor in subsequent [RFC3261] routing behavior in the IP-PBX, nor in any proxies along the path. The routing follows the [RFC3261] Registered AoR-contact resolution model, which is a basic function of SIP.

The only SIP devices affected by this document's mechanism is the SSP's Registrar, which needs to update the appropriate AoR entries, and any proxy/ies of the SSP which perform route resolution by looking up the contents of the (logical) location-service database. Since such proxies may not even be in the path of the REGISTER request, an option-tag will not help. And since the Registrar and Proxies in question are all under control of the same administrative entity (the SSP), it is reasonable to expect them all to support this document's mechanism, if any do.

## 5. Registering for Local-Number AoRs

This document's mechanism relies on the GIN [draft-gin] Registration mechanism. The IP-PBX Registers into the SSP, using a REGISTER request with the "gin" option-tag in the Require and Proxy-Require header fields, and a Contact URI containing the "bnc" URI parameter and no user portion. After the PBX is authenticated, the registrar updates its location service so that each of the Local-Number AoRs associated with the PBX creates a unique AOR to Contact mapping.

In practice, however, the SSP domain may not have specific knowledge of any or all user names within a given phone-context's scope. In fact, the Local-Number TEL URI parameters (which are URI user parameters in SIP URIs) may only have meaning to the ultimate target of the request, or some entity which is authoritative for the phone-context's user names. Those parameters cannot be removed by the SSP if it does not actually process the user portions of the Local-Number. (i.e., if it does not have the dial-plan, etc.)

With regard to this document's mechanism, what this means is that such an SSP cannot physically instantiate an AoR in a database for every possible Local-Number and cannot physically instantiate an expanded Contact URI for every possible Local-Number user name with every possible user parameter. That does not inhibit the mechanism from working or being usable, however, because the location-service database model is purely an abstract concept. What's important is that the route-resolving Proxy be able to lookup and replace an AoR it is authoritative for, to a Registered Contact URI, such that the resultant Request URI matches what the IP-PBX expects to receive.

It is "safe" to do this because the explicitly Registered Contact URI of the [draft-gin] REGISTER request had no user portion, and thus no possible URI user parameters. As defined in [draft-gin], the Contact URI parameters of the REGISTER are saved and reused, but not URI user parameters.

There are multiple ways of describing the logical AoR instantiation and Contact URI expansion rules. They could be described as covering every possible ABNF expansion, such that every possible user and parameter logically exists in the location-service database (but obviously not physically exists). Or it could be described as only the phone-context value itself being an "AoR" entry and Contact URI expansion, with a policy to allow any and all user names and parameters to be copied instead of replaced by the Contact URI.

This remains an open issue for discussion, as discussed in section 8.

Regardless, for an implicitly Registered SIP AoR with a URI user portion matching the syntax outlined for "local-number" TEL URIs in [RFC3966]: the Contact is expanded following the other AoR models, EXCEPT that all URI user parameters are also included. For example, if the logically provisioned "AoR" from the previous examples were: "sip:12345;ext=678;phone-context=+1212555@ssp.example.com", it would logically get an automatically generated Contact value of <sip:12345;ext=678;phone-context=+1212555@198.51.100.3:5060;foo=bar> and if the AoR were "sip:12345;ext=678;phone-context=ssp.example.com@ssp.example.com", the resultant Contact value would be <sip:12345;ext=678;phone-context=ssp.example.com@198.51.100.3:5060;foo=bar>.

Note that in practice it is not uncommon to receive a SIP URI which does not strictly comply with the formatting rules of [RFC3966], but is processed as if it were, based on local policies. That is legal, of course, but from a logical perspective the SIP URI is actually retargeted or transformed into the syntactically valid form following [RFC3966], and that form MUST be the one used for routing, Contact URI expansions, etc. Likewise, if the URI were a TEL URI,

it MUST be logically transformed into a SIP URI of the SSP's domain as defined in section 19.1.6 of [RFC3261], with an appropriate phone-context, before executing the rules.

As in [draft-gin], aside from the "bnc" parameter, all URI parameters present in the Contact URI in the REGISTER message MUST be copied to the Contact value stored in the location service.

Note that the location service database, and any entry model described here, is purely an abstract concept used by [RFC3261], [draft-gin], and this document; an actual implementation may do whatever it likes internally, so long as the external behavior follows the model. For example, if an SSP does not maintain any specific knowledge of the Local Number dial-plan, but simply performs prefix or default routing for an Enterprise's private extensions, the SSP could just route based on the E.164 phone-context field value without having a separate physical "AoR" database entry for each local number of that context.

## 6. SSP Processing of Inbound Non-E.164 Requests

The SSP Proxy/Registrar (or equivalent entity) performs traditional Proxy/Registrar behavior, based on the logical mapping described in Section 5 and [draft-gin].

## 7. Interaction with Other Mechanisms

The following sections describe the means by which this mechanism interacts with relevant REGISTER-related extensions currently defined by the IETF.

Currently, the descriptions are somewhat informal, and omit some details for the sake of brevity. If the MARTINI working group expresses interest in furthering the mechanism described by this document, they will be fleshed out with more detail and formality.

### 7.1. Globally Routable User-Agent URIs (GRUU)

The GRUU mechanism for this document's mechanism works exactly the same way as defined in [draft-gin]. The GIN GRUU mechanism has no dependency on the AoR being an E.164.

### 7.2. Registration Event Package

The Registration Event Packet behavior for this document's mechanism works exactly the same way as defined in [draft-gin]. The [draft-gin] reg-event model has no dependency on the AoR being an E.164.

There is, however, an issue for Local-Numbers, if the SSP does not actually know the full list of Local-Number user names in the given phone-context scope. In such a case, it is TBD for how to handle this.

This remains an open issue for discussion, as discussed in section 8.

### 7.3. Non-Adjacent Contact Registration (Path) and Service Route Discovery

The Path and Service-Route behavior and considerations for this document's mechanism are exactly the same as defined in [draft-gin]. The [draft-gin] Path and Service-Route model has no dependency on the AoR being an E.164.

## 8. Open Issues

This document has several open issues, which were noted previously. They center around the handling of Local-Numbers. Local-Numbers are difficult because they are doubly-scoped: once at the URI level by the domain name, and internally by the phone-context URI user parameter. The authoritative system for the Local-Number user portion (the system(s) which knows what they are and how to process them) is not necessarily identified by the URI's domain name, but rather may be identified by the phone-context's value.

If the phone-context identifies the SSP domain, all's well - but that's rarely the case. More likely is that it identifies an E.164 number, or a sub-domain of the SSP, or another domain entirely. This causes issues with certain functions such as the reg-event package, which has been identified as an open issue.

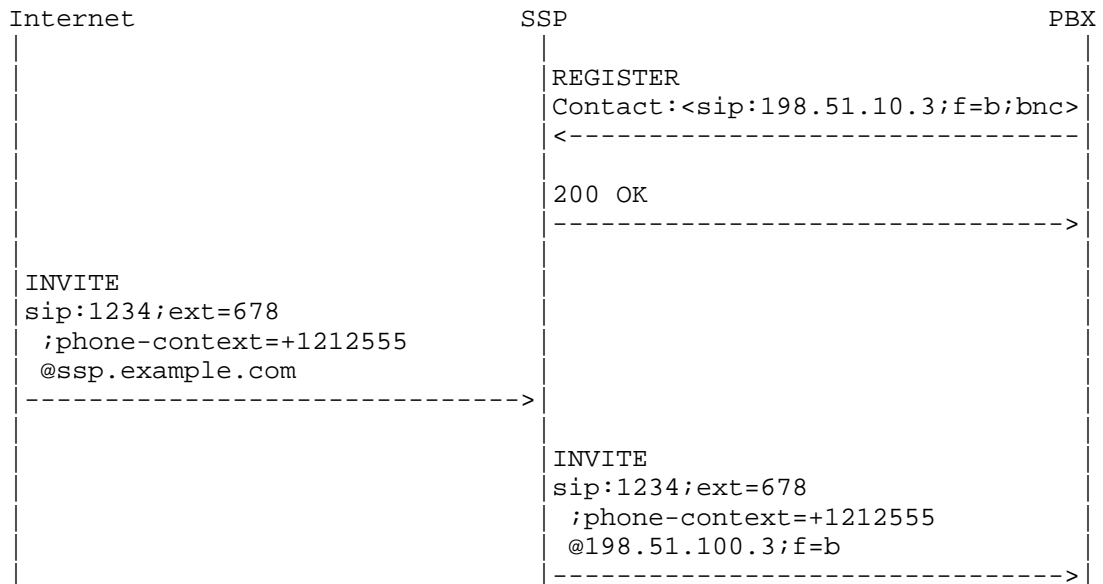
## 9. Examples

These will be fleshed out more in later versions of the draft, with explanations of the processing performed at each step. For the time being, they just show the basic syntax described above.



## 9.1. Usage Scenario: Basic Registration case

This example shows a basic bulk REGISTER transaction, followed by an INVITE addressed to one of the registered terminals, for a Local-Number AoR.



```

REGISTER sip:ssp.example.com SIP/2.0
Via: SIP/2.0/UDP 198.51.100.3:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
To: <sip:pbx123@ssp.example.com>
From: <sip:pbx123@ssp.example.com>;tag=a23589
Call-ID: 843817637684230998sdasdh09
CSeq: 1826 REGISTER
Proxy-Require: gin
Require: gin
Supported: path
Contact: <sip:198.51.10.3:5060;f=b;bnc>
Expires: 7200
Content-Length: 0
  
```

```
INVITE sip:1234;ext=678;phone-context=+1212555
      @ssp.example.com SIP/2.0
Via: SIP/2.0/UDP foo.example;branch=z9hG4bKa0bc7a0131f0ad
Max-Forwards: 69
To: <sip:2145550105@some-other-place.example.net>
From: <sip:alice@rabbithole.example.org>;tag=456248
Call-ID: f7aecbfc374d557baf72d6352elfbcd4
CSeq: 24762 INVITE
Contact: <sip:line-1@192.0.2.178:2081>
Content-Type: application/sdp
Content-Length: ...
```

<sdp body here>

```
INVITE sip:1234;ext=678;phone-context=+1212555
      @198.51.100.3;f=b SIP/2.0
Via: SIP/2.0/UDP foo.example;branch=z9hG4bKa0bc7a0131f0ad
Via: SIP/2.0/UDP ssp.example.com;branch=z9hG4bKa45cd5c52a6dd50
Max-Forwards: 68
To: <sip:2145550105@some-other-place.example.net>
From: <sip:alice@rabbithole.example.org>;tag=456248
Call-ID: 7ca24b9679ffe9aff87036a105e30d9b
CSeq: 24762 INVITE
Contact: <sip:line-1@192.0.2.178:2081>
Content-Type: application/sdp
Content-Length: ...
```

<sdp body here>

## 10. IANA Considerations

This document makes no request of IANA.

## 11. Security Considerations

This section is still TBD, but it should follow/have the same issues as [draft-gin].

## 12. Acknowledgements

Thanks to Adam Roach for providing text copied from [draft-gin].

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## Appendix A - Why Local-Numbers may need processing by SSPs

There is some debate about how a non-E.164 AoR could even be received by the SSP for processing to begin with. This section describes how such could be the case.

It should be noted that this document only deals with SIP AoRs of the same URI domain name as that of the REGISTER's To URI - namely the SSP's domain.

A SIP Request targeted to a Local-Number could require processing by the SSP because:

- The SSP provides IP-Centrex type services for some of the AoRs of an Enterprise, for example for small branches, while providing SIP-Trunk service to the main IP-PBX(s). Requests from the IP-Centrex UAs will thus be targeted to Local-numbers as they are received by the SSP Proxy on their way to the IP-PBX.
- The SSP provides inbound extension dialing, for example by offering private calling-card services, such that a E.164 number call is terminated by an Application Server of the SSP which authenticates the caller belongs to an Enterprise and then allows private extension dialing, as a UAC, thereby originating a new SIP session Request using a Local-Number target.
- The SSP provides inter-branch private dialing, by routing on some number of leading digits of a Local-Number.

There are other possibilities as well, of course, but this section is only intended to provide some basic rational for why it is possible for a local-number AoR to be used and appear in the SSP.