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SIP MARTINI with Other Logical Identifier Values which are not E.164 (OLIVE)
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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. This document defines a means for other, non-E.164 AoRs 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, in [draft-gin].

The current Martini mechanism only supports E.164-based AoR's, however in actual deployments private-extension or "local" numbers are used for hosted and carrier-provided intra-Enterprise calling services, and domain-scoped "email-style" URIs may become more popular in the near future. Neither of these forms of AoRs are supported by the current Martini 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.).

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. Background on Current Martini Mechanism

The current Martini 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 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.

3.1. Why non-E.164 AoRs 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 AoR 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 AoR.

A SIP Request targeted to an Email-style AoR could require processing by an SSP because:

- The SSP provides Email-style AoRs for business customers - for example "sip:bob@ssp.example.net".
- The SSP has a sub-division or sub-entity which uses IP-PBX Martini trunks into the SSP main domain, for which the SSP provides AoRs of the main SSP domain - for example, a sales division IP-PBX in a sub-domain of sales.ssp.example.net, but for which the SSP provides an AoR of "sip:sales@ssp.example.net".
- The Email-style AoR is scoped to the Enterprise's domain, but the Request originated from within the SSP - for example by one of its subscriber SIP UAs. Since SIP UAs generally send their Requests through their SSP's proxies, the Request will be processed by them first. This type of AoR (i.e., one scoped outside of the SSP's domain) is NOT in scope for this document.

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 or email-style AoR to be used and appear in the SSP.

3.2. The issues with non-E.164 Identifiers

At the initial outset of Martini's work, the problem space to work on was narrowed to E.164 numbers only, for several reasons. This section attempts to identify the reasons in detail, and address them.

3.2.1 Globally unique AoR

One of the initial benefits cited for only supporting E.164 is that an E.164 user name is globally unique by itself, and thus changing the host portion of the Request-URI does not impact the identity of the intended target of the Request.

Fortunately, [draft-gin] does not in fact rely on that property of the AoR username. What [draft-gin] does is emulate what would have happened had the IP-PBX Registered each E.164 AoR separately. It is not in fact Registering an "E.164 number" - it is Registering typical [RFC3261] AoRs, which just happen to be formatted as E.164 numbers. For example, the IP-PBX is not implicitly Registering the identity of "+12125551212", it is instead implicitly Registering for the SIP AoR of "sip:+12125551212@ssp.example.net". The SSP may well associate and route requests for "tel:+12125551212" to that AoR's Registered contact, but in so doing it can be described as logically performing a transformation of the TEL URI to a SIP URI and looking up that SIP URI in its location service database.

Interestingly, the AoR that was implicitly Registered is globally unique, *regardless* of the fact that the user portion happens to be an E.164 number. It is globally unique, because the AoR is of the domain "ssp.example.net" - just as any [RFC3261] SIP AoR is globally unique due to its host domain portion.

3.2.2 Registered Contact Username

The other rational for needing a globally unique identifier is the Registered Contact URI. Because the Martini model performs a bulk Registration and uses a defined expansion rule for how to expand the Registered Contact into a unique Contact URI per AoR, the rule's output needs to be unambiguous and generate a unique Contact per AoR. For example, if the IP-PBX has multiple SSPs it Registers into, it can't just receive a request to "sip:bob@pbx.enterprise.com", because it won't know if that's to

"sip:bob@ssp1.example.net" or "sip:bob@ssp2.example.net", which may be different users.

This is a real problem, but what it requires/needs is a globally unique Contact URI, not a globally unique AoR user name. In fact, [draft-gin] suffers from this problem as well, in some ways. Although it is unambiguous what the target of the request may be (because the username is globally unique), it still may be important for the IP-PBX to know which implicitly Registered AoR the request was resolved from, before being sent to it. For example, the IP-PBX may want to know the request is for "sip:+12125551212@ssp.example.net.uk" vs. "sip:+12125551212@ssp.example.net.fr". The IP-PBX *would* have Registered separately for each domain, but if its contact is the same (and if it went through the same path, etc.), there would be no unique identifier for inbound Requests.

This problem could be solved numerous ways. For one, this is a similar issue to that solved by [RFC4244] or being worked on in [draft-4244bis]. Secondly, and more importantly, the IP-PBX could simply make the Contact unique by inserting a parameter in the Contact URI it Registers, or inserting a Path header field with unique information per Register target domain.

Regardless of the solution, it is solvable without impacting the user portion of the AoR being implicitly Registered. Therefore, there is no need to make the username portion globally unique, and thus once again there is no need for [draft-gin] to only be used for E.164 AoRs.

3.2.3 Avoiding syntax mismatches

Another issue with the Contact bulk-expansion rule and implicitly Registering Contact URIs is guaranteeing the Registrar uses a username the IP-PBX will accept/expect. In particular, there are numerous syntactic forms a phone number can take in the user portion of a SIP URI, for example with or without a leading "+", or with visual-separators or not. Since the IP-PBX is not actually Registering an explicit Contact URI user portion for each AoR, there needs to be an assurance that the format of the expanded-to user portion matches what it expects.

For Email-style URIs, this is actually quite straight-forward, because there is only one form the user name can take: if the AoR is "sip:bob@ssp.example.net", then a simple AoR-user-based expansion rule would suffice because "bob" can only have one form. (e.g., even case-sensitivity matters) Thus there is no syntax mismatch concern for generic string names.

Local-Numbers in TEL URIs, or SIP URIs formed from TEL URIs, are more difficult, however. Like E.164 (i.e., global-number) formats, they are not as well constrained. They may have visual-separators, for example, and if their phone-context parameter value is a global-number, it too has the same formatting "looseness" of E.164 number user names.

To make matters worse, Local-Numbers are frequently NOT within the scope of the SSP, or at least the SSP does not always have full knowledge of all Local-Numbers it can route requests for.

This document will address this Local-Number issue, but for the case of Local-Numbers scoped to the SSP's domain only. Local-Numbers scoped to a phone-context that is an E.164 or not the same domain name as the REGISTER transaction's explicit AoR are outside the scope of this document.

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

4. The Solution - an Overview

The general concept proposed in this document is to simply apply [draft-gin] for any set of AoRs of the Registered-to domain, for any valid set of user strings as defined in [RFC3261]. The [draft-gin] REGISTER request would cause the Registrar to populate the set of AoRs 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. In the case of a Local-Number, for the specific scopes described later, the username is "normalized" in the same manner as [draft-gin].

Note that the list of 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 Martini [draft-gin] Registration or subsequent [RFC3261] routing behavior in the IP-PBX or 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 AoRs

This document's mechanism relies on the Martini [draft-gin] Registration mechanism. The IP-PBX Registers into the SSP, using a REGISTER request with the [draft-gin] option-tag in the Require and Proxy-Require header fields, and a Contact URI containing the [draft-gin] "bnc" URI parameter and no user portion. After the PBX is authenticated, the registrar updates its location service so that each of the AoRs associated with the PBX creates a unique AOR to Contact mapping. 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:bob@ssp.example.com"), and the corresponding Contact will be formed adding a user portion to the REGISTER's Contact URI containing the AoR user name and removing the "bnc" parameter. For example, if the "Contact" header field contains the URI <sip:198.51.100.3:5060;foo=bar;bnc>, then the Contact value associated with the aforementioned AOR will be <sip:bob@198.51.100.3:5060;foo=bar>.

Local-Numbers have slightly different Contact URI expansion rules, as defined later.

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.

5.1. Registering Local Number AoRs

Local-Numbers present a complexity for AoR Registration, because their user portion is scoped to the phone-context's value. In practice the SSP domain identified by the entire AoR URI domain name may not have specific knowledge of any user name within a given phone-context's scope. In fact, the Local-Number TEL URI parameters (which are URI user parameters in SIP URIs), likely 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]: if the 'phone-context' URI user parameter value is the same SSP domain as that in the REGISTER transaction's To URI, then

Contact is then formed following the other AoR models, EXCEPT that all URI user parameters are also included, other than the phone-context parameter. For example, if the logically provisioned "AoR" from the previous examples were: "sip:12345;ext=678;phone-context=ssp.example.net@ssp.example.net", it would logically get an automatically generated Contact URI of <sip:12345;ext=678@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 would be logically transformed into a SIP URI of the SSP's domain before executing the rules.

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

The SSP Proxy/Registrar (or equivalent entity) performs traditional Proxy/Registrar behavior, based on the mapping described in Section 5 and [draft-gin]. For Local-Numbers in particular, the following section provides additional detail.

6.1. Processing of Local-Number Requests

As discussed in section 5.1, Local-Numbers present a special case which may need to be handled with more explicit rules than [RFC3261] or [draft-gin] currently prescribe. If the location-service database is described as having every possible expansion, then the Request would be processed in the same manner as 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 [draft-gin] GRUU mechanism

has no dependency on the global uniqueness of the AoR username, nor on being digits or 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 global uniqueness of the AoR username, nor on being digits or 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. One obvious way would be to define a syntactic format for the AoR and Contact URIs which only includes the phone-context value and makes it clear the user portion and any other parameters are unknown, such as "sip:!.*!;phone-context=ssp.example.com@ssp.example.com". Since a "!" is not a legal character in a Local-Number user name, but is legal in a SIP URI user name, it would be both ABNF compliant and unambiguous from Local-Numbers.

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 global uniqueness of the AoR username, nor on being digits or 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 identified by the URI's domain name, but rather 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 document tries to propose a mechanism to support the case when the phone-context identifies the SSP's domain, and which has a matching SIP AoR implicitly Registered by the IP-PBX. However, even in such a case, it does not imply the SSP has authoritative information for the user name - typically the IP-PBX does. This causes issues with certain functions such as the reg-event package.

In theory, this can probably be made to work as defined in this document, but there is an alternative: if we had a model for using the Registered reachability information of [draft-gin] for routing of requests which are not AoRs of the Registered-to domain (e.g., to an AoR of "sip:charlie@pbx.example.com"), then Local-Numbers could follow that routing model instead. This would follow a more natural [RFC3263] model, since effectively the phone-context indicates that the authoritative administrative entity is not in fact the SSP.

A draft detailing how [draft-gin] can be used in such a model is forthcoming.

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 1

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

```
Internet                        SSP                              PBX
|                                |                                 |
|                                |REGISTER                         |
|                                |Contact:<sip:198.51.100.3;bnc>   |
|                                |<--------------------------------|
|                                |                                 |
|                                |200 OK                           |
|                                |------------------------------->|
|                                |                                 |
|INVITE                          |                                 |
|sip:bob@ssp.example.com         |                                 |
|------------------------------->|                                 |
|                                |                                 |
|                                |INVITE                           |
|                                |sip:bob@198.51.100.3             |
|                                |------------------------------->|
```

```
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: bulknumbercontact
Require: bulknumbercontact
Contact: <sip:198.51.100.3:5060;user=phone;bnc>
Expires: 7200
Content-Length: 0
```

```
INVITE sip:bob@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: f7aecbfc374d557baf72d6352e1fbcd4
CSeq: 24762 INVITE
Contact: <sip:line-1@192.0.2.178:2081>
Content-Type: application/sdp
Content-Length: ...

<sdp body here>
```

```
INVITE sip:bob@198.51.100.3 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>
```

9.2. Usage Scenario: Basic Registration case 2

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

```
Internet                        SSP                             PBX
|                                |                                 |
|                                |REGISTER                         |
|                                |Contact:<sip:198.51.10.3;f=b;bnc>|
|                                |<--------------------------------|
|                                |                                 |
|                                |200 OK                           |
|                                |-------------------------------->|
|                                |                                 |
|INVITE                          |                                 |
|sip:1234;ext=678                |                                 |
| ; ssp.example.com              |                                 |
| @ssp.example.com               |                                 |
|------------------------------->|                                 |
|                                |                                 |
|                                |INVITE                           |
|                                |sip:1234;ext=678                 |
|                                | @198.51.100.3;f=b               |
|                                |-------------------------------->|
```

```
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: bulknumbercontact
Require: bulknumbercontact
Contact: <sip:198.51.10.3:5060;f=b;bnc>
Expires: 7200
Content-Length: 0
```

```
INVITE sip:1234;ext=678;phone-context=ssp.example.com
   @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: f7aecbfc374d557baf72d6352e1fbcd4
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@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 (unknowingly) providing text which I stole^M^M^M copied from [draft-gin].

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