

ECRIT	H. Schulzrinne	
Internet-Draft	Columbia University	
Intended status: Informational	H. Tschofenig	
Expires: September 10, 2010	Nokia Siemens Networks	
	M. Patel	
	InterDigital Communications	
	March 09, 2010	

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Public Safety Answering Point (PSAP) Callbacks draft-schulzrinne-ecrit-psap-callback-03.txt

Abstract

After an emergency call is completed (either prematurely terminated by the emergency caller or normally by the call-taker) it is possible that the call-taker feels the need for further communication or for a clarification. For example, the call may have been dropped by accident without the call-taker having sufficient information about the current situation of a wounded person. A call-taker may trigger a callback towards the emergency caller using the contact information provided with the initial emergency call. This callback could, under certain circumstances, then be treated like any other call and as a consequence, it may get blocked by authorization policies or may get forwarded to an answering machine.

The IETF emergency services architecture addresses callbacks in a limited fashion and thereby covers a couple of scenarios. This document discusses some shortcomings and raises the question whether additional solution techniques are needed.

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1. Introduction

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Summoning police, the fire department or an ambulance in emergencies is one of the fundamental and most-valued functions of the telephone. As telephone functionality moves from circuit-switched telephony to Internet telephony, its users rightfully expect that this core functionality will continue to work at least as well as it has for the

legacy technology. New devices and services are being made available that could be used to make a request for help, which are not traditional telephones, and users are increasingly expecting them to be used to place emergency calls.

Regulatory requirements demand that the emergency call itself provides enough information to allow the call-taker to initiate a call back to the emergency caller in case the call dropped or to interact with the emergency caller in case of further questions. Such a call, referred as PSAP callback subsequently in this document, may, however, be blocked or forwarded to an answering machine as SIP entities (SIP proxies as well as the SIP UA itself) cannot associate the potential importance of the call based on the SIP signaling.

Note that the authors are, however, not aware of regulatory requirements for providing preferential treatment of callbacks initiated by the call-taker at the PSAP towards the emergency caller.

Section 10 of [\[I-D.ietf-ecrit-framework\] \(Rosen, B., Schulzrinne, H., Polk, J., and A. Newton, "Framework for Emergency Calling using Internet Multimedia," July 2009.\)](#) discusses the identifiers required for callbacks, namely AOR URI and a globally routable URI in a Contact: header. Section 13 of [\[I-D.ietf-ecrit-framework\] \(Rosen, B., Schulzrinne, H., Polk, J., and A. Newton, "Framework for Emergency Calling using Internet Multimedia," July 2009.\)](#) provides the following guidance regarding callback handling:

A UA may be able to determine a PSAP call back by examining the domain of incoming calls after placing an emergency call and comparing that to the domain of the answering PSAP from the emergency call. Any call from the same domain and directed to the supplied Contact header or AoR after an emergency call should be accepted as a call-back from the PSAP if it occurs within a reasonable time after an emergency call was placed.

This approach mimics a stateful packet filtering firewall and is indeed helpful in a number of cases. It is also relatively simple to implement. Below, we discuss a few cases where this approach fails.

1.1. Routing Asymmetry

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In some deployment environments it is common to have incoming and outgoing SIP messaging to use different routes.

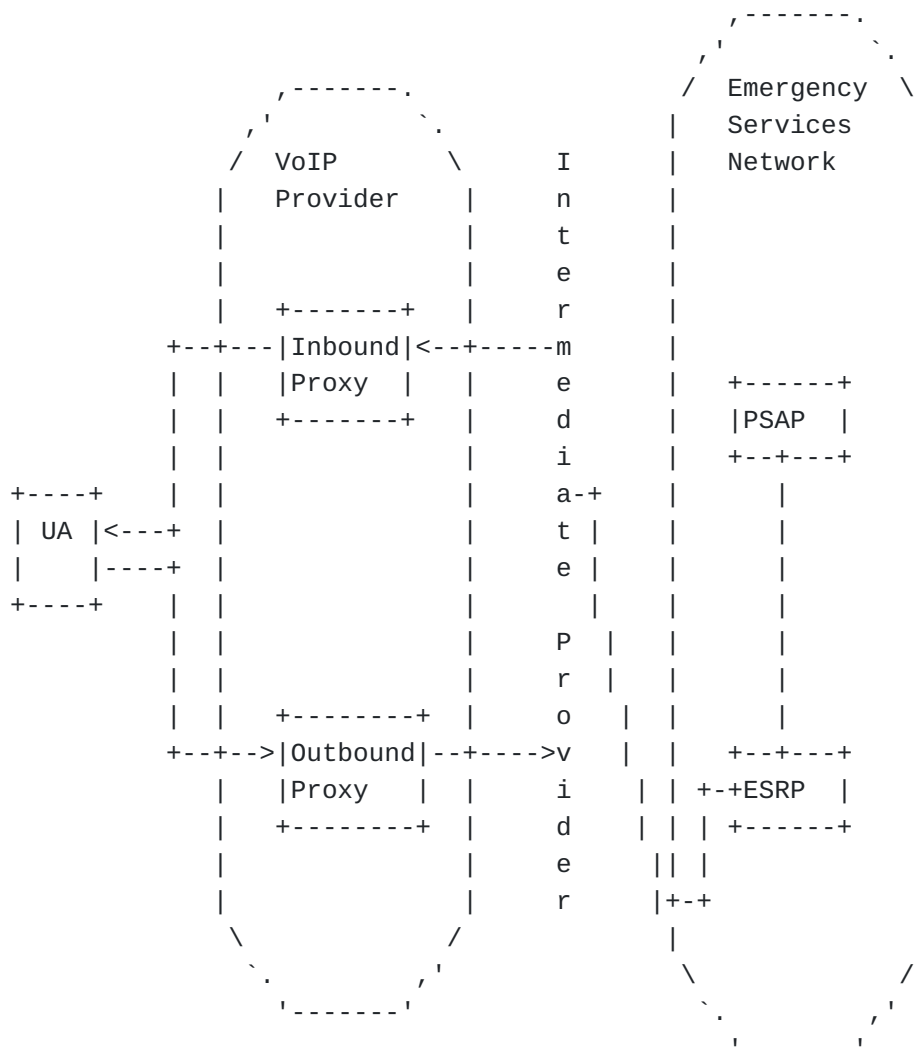


Figure 1: Example for Routing Asymmetry

1.2. Multi-Stage Resolution

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Consider the following emergency call routing scenario shown in [Figure 2 \(Example for Multi-Stage Resolution\)](#) where routing towards the PSAP occurs in several stages. An emergency call uses a SIP UA that does not run LoST on the end point. Hence, the call is marked with the 'urn:service:sos' Service URN [\[RFC5031\] \(Schulzrinne, H., "A Uniform Resource Name \(URN\) for Emergency and Other Well-Known Services," January 2008.\)](#). The user's VoIP provider receives the emergency call and determines where to route it. Local configuration or a LoST lookup might, in our example, reveal that emergency calls are routed via a

dedicated provider FooBar and targeted to a specific entity, referred as esrp1@foobar.com. FooBar does not handle emergency calls itself but performs another resolution step to let calls enter the emergency services network and in this case another resolution step takes place and esrp-a@esinet.org is determined as the recipient, pointing to an edge device at the IP-based emergency services network. Inside the emergency services there might be more sophisticated routing taking place somewhat depending on the existing structure of the emergency services infrastructure.

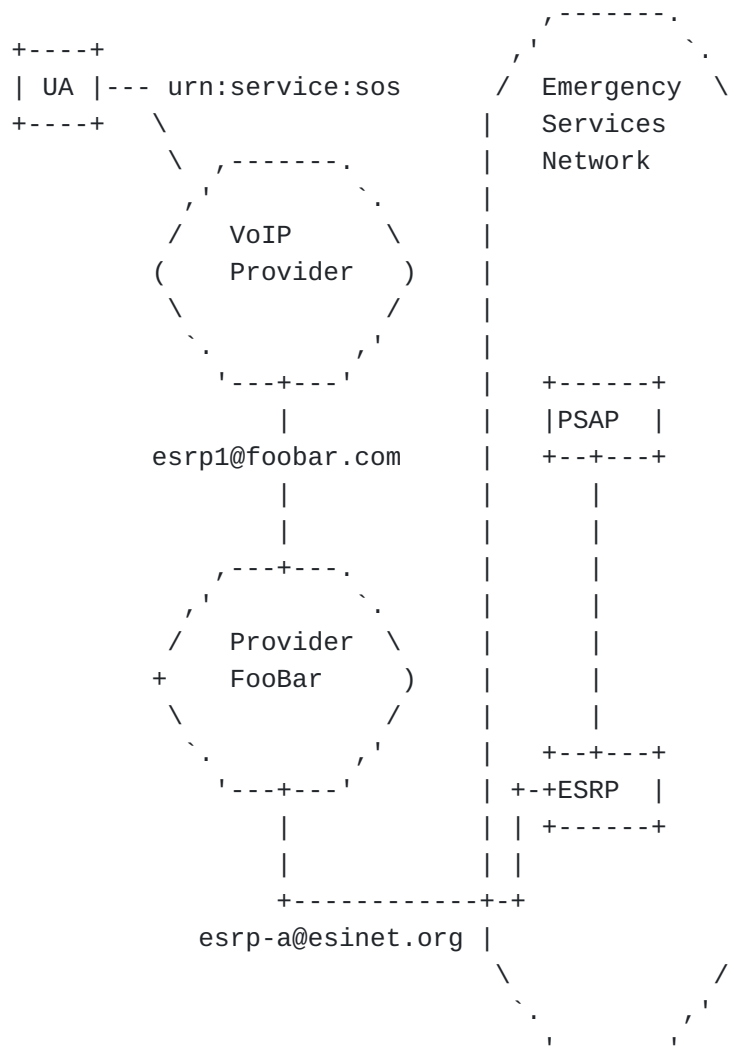


Figure 2: Example for Multi-Stage Resolution

1.3. Call Forwarding

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Imagine the following case where an emergency call enters an emergency network (state.org) via an ERSP but then gets forwarded to a different emergency services network (in our example to police-town.org, fire-town.org or medic-town.org). The same considerations apply when the police, fire and ambulance networks are part of the state.org sub-domains (e.g., police.state.org).

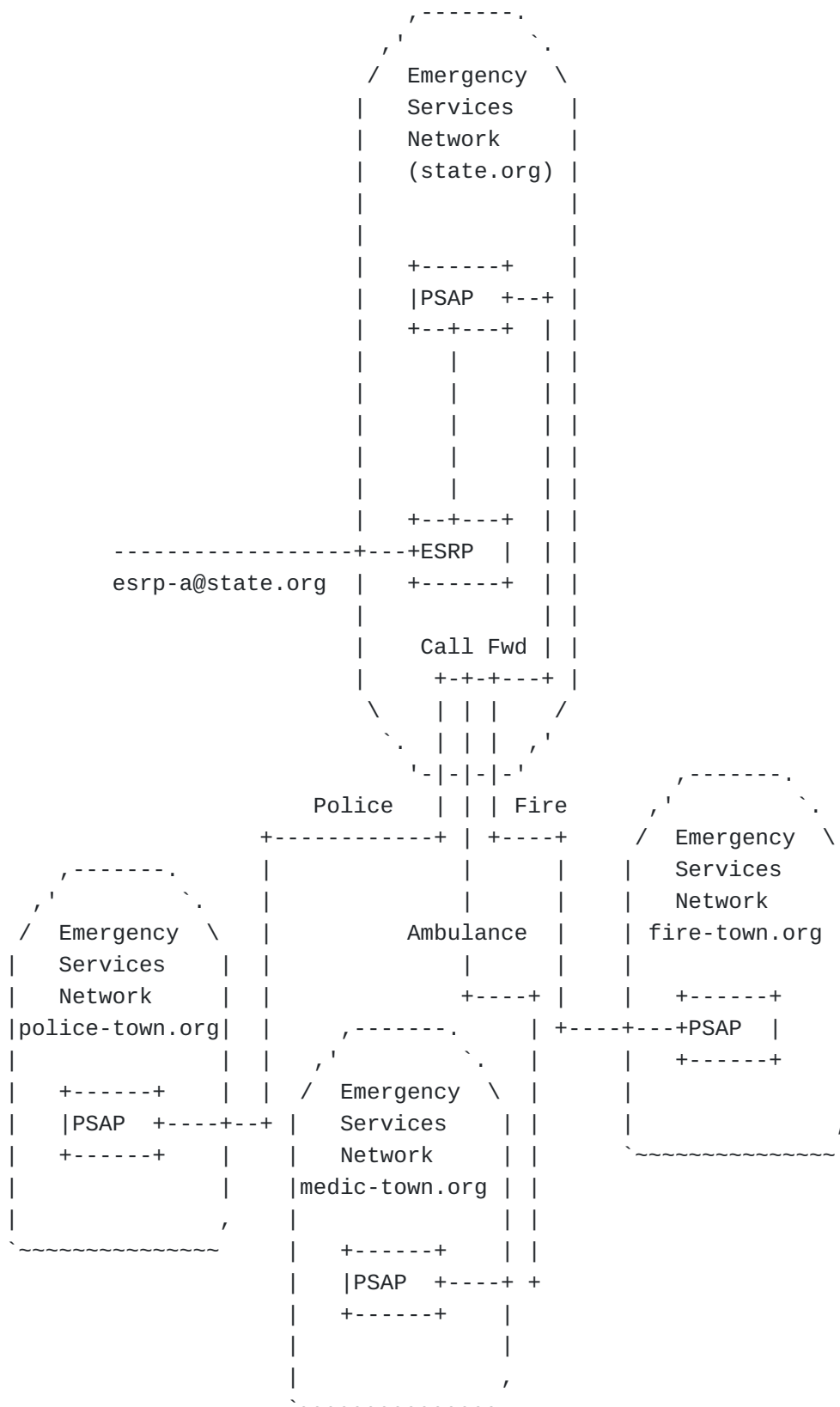


Figure 3: Example for Call Forwarding

1.4. PSTN Interworking

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In case an emergency call enters the PSTN, as shown in [Figure 4 \(Example for PSTN Interworking\)](#), there is no guarantee that the callback some time later does leave the same PSTN/VoIP gateway or that the same end point identifier is used in the forward as well as in the backward direction making it difficult to reliably detect PSAP callbacks.

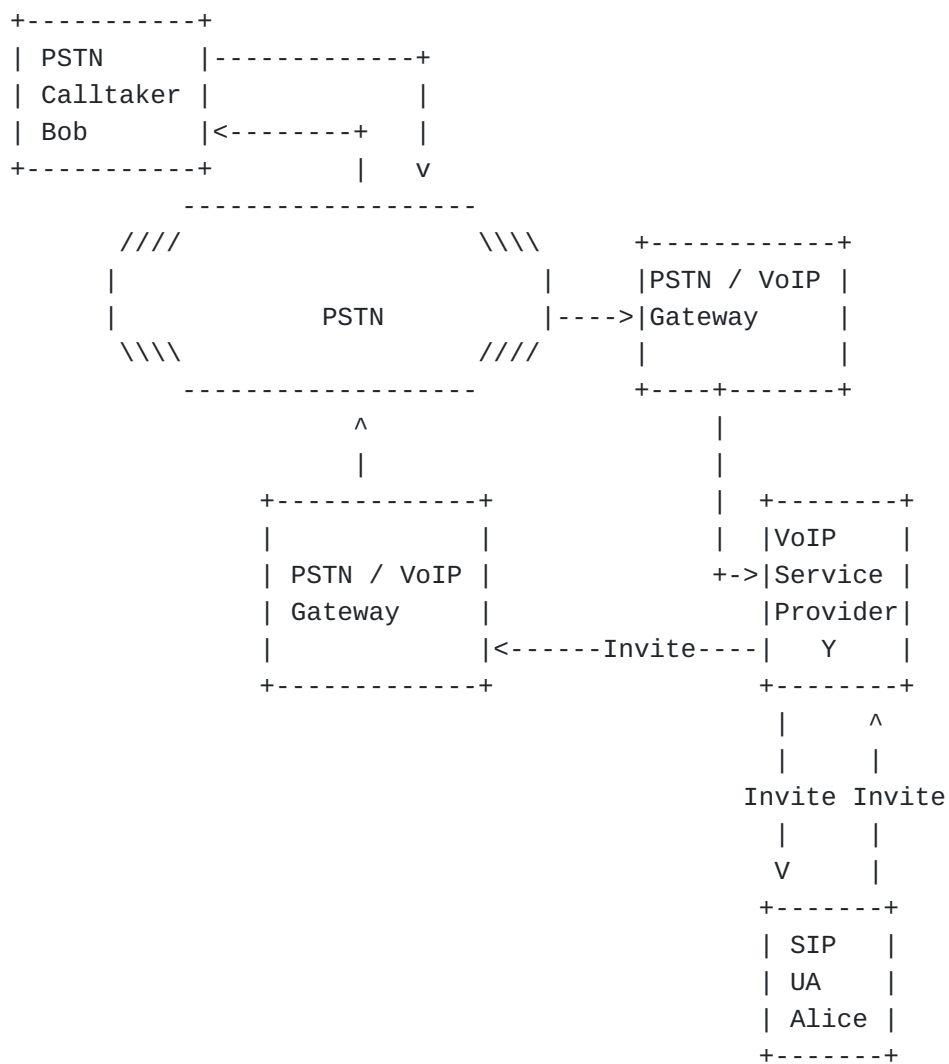


Figure 4: Example for PSTN Interworking

1.5. Network-based Service URN Resolution

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The mechanism described in [\[I-D.ietf-ecrit-framework\]](#) (Rosen, B., Schulzrinne, H., Polk, J., and A. Newton, "Framework for Emergency Calling using Internet Multimedia," July 2009.) assumes that all devices at the call signaling path store information about the domain of the communication recipient. This is necessary to match the stored domain name against the domain of the sender when an incoming call arrives.

However, the IETF emergency services architecture also considers those cases where the resolution from the Service URN to the PSAP URI happens somewhere in the network rather than immediately at the end point itself. In such a case, the end device is therefore not able to match the domain of the sender with any information from the outgoing emergency call.

[Figure 5 \(Example for Network-based Service URN Resolution\)](#) shows this message exchange graphically.

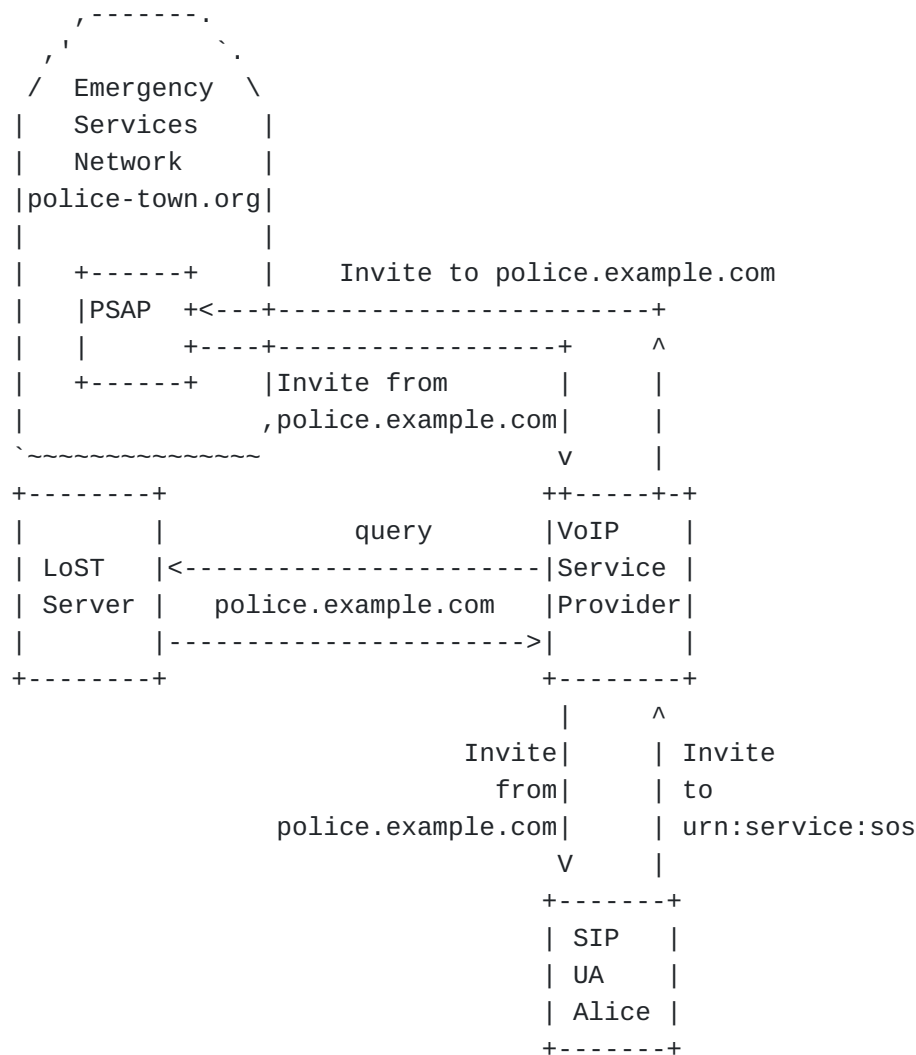


Figure 5: Example for Network-based Service URN Resolution

2. Terminology

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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 [\[RFC2119\] \(Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels," March 1997.\)](#).

Emergency services related terminology is borrowed from [\[RFC5012\] \(Schulzrinne, H. and R. Marshall, "Requirements for Emergency Context Resolution with Internet Technologies," January 2008.\)](#).

3. Design Approaches

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The starting point of the investigations is the currently provided functionality in Section 13 of [\[I-D.ietf-ecrit-framework\]](#) ([Rosen, B., Schulzrinne, H., Polk, J., and A. Newton, "Framework for Emergency Calling using Internet Multimedia," July 2009.](#)). It focuses on identifying a response to a previously made emergency call. As described in the introduction this approach is quite coarse grained since any call from the PSAP's domain is given preferential treatment. This approach is, however, likely going to be practical. Still there are a couple of limitations, as discussed in this document. To expand on the initially provided solution the following description starts with attempt to identify the caller as a PSAP. There are two approaches for accomplishing this functionality.

```
+-----+
| List of |+
| valid  ||
| PSAP ids ||
+-----+|
+-----+
*
* whitelist
*
V
Incoming   +-----+   Normal
SIP Msg    | SIP      |+   Treatment
----->| Entity  ||=====>
+ Identity |          ||(if not in whitelist)
+-----+|
+-----+
||
||
|| Preferential
|| Treatment
+++++=>
      (in whitelist)
```

Figure 6: Identity-based Authorization

In [Figure 6 \(Identity-based Authorization\)](#) an interaction is presented that allows a SIP entity to make a policy decision whether to bypass installed authorization policies and thereby providing preferential

treatment. To make this decision the sender's identity is compared with a whitelist of valid PSAPs. The identity assurances in SIP can come in different forms, such as SIP Identity [RFC4474] (Peterson, J. and C. Jennings, "Enhancements for Authenticated Identity Management in the Session Initiation Protocol (SIP)," August 2006.) or with P-Asserted-Identity [RFC3325] (Jennings, C., Peterson, J., and M. Watson, "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks," November 2002.). The former technique relies on a cryptographic assurance and the latter on a chain of trust.

The establishment of a whitelist with PSAP identities is operationally complex and does not easily scale world wide. When there is a local relationship between the VSP/ASP and the PSAP then populating the whitelist is far simpler.

An alternative approach to an identity based authorization model is outlined in Figure 7 (Trait-based Authorization). In fact, RFC 4484 [RFC4484] (Peterson, J., Polk, J., Sicker, D., and H. Tschofenig, "Trait-Based Authorization Requirements for the Session Initiation Protocol (SIP)," August 2006.) already illustrated the basic requirements for this technique.

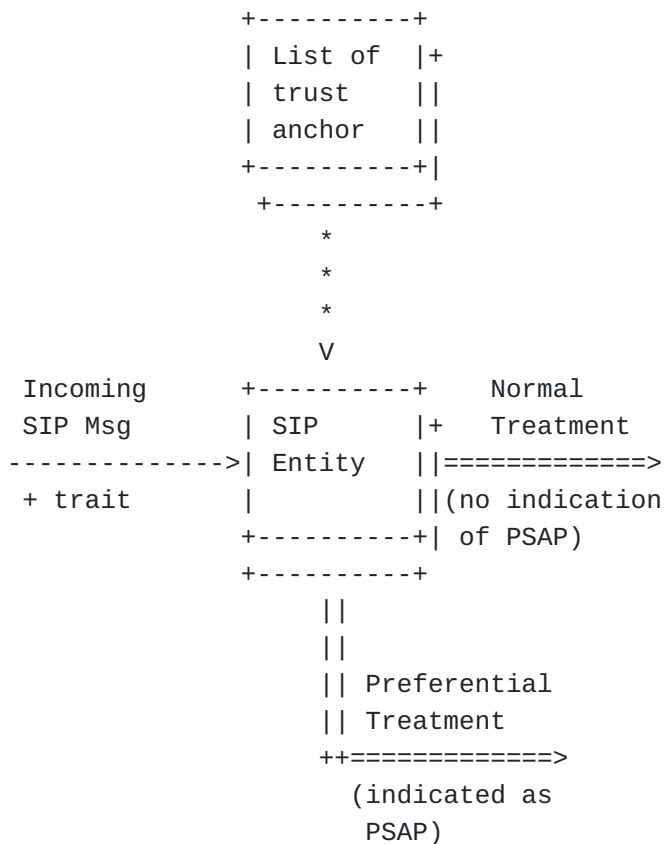


Figure 7: Trait-based Authorization

In a trait-based authorization scenario an incoming SIP message contains a form of trait, i.e. some form of assertion. The assertion contains an indication that the sending party has the role of a PSAP (or similar emergency services entity). The assertion is either cryptographically protected to enable end-to-end verification or an chain of trust security model has to be assumed. In [Figure 7 \(Trait-based Authorization\)](#) we assume an end-to-end security model where trust anchors are provisioned to ensure the ability for a SIP entity to verify the received assertion.

From a solution point of view various approaches are feasible, such as SIP SAML (see [\[I-D.ietf-sip-saml\]](#) (Tschofenig, H., Hodges, J., Peterson, J., Polk, J., and D. Sicker, "SIP SAML Profile and Binding," March 2010.)) or URI Parameters for indicating the Calling Party's Category and Originating Line Information (see [\[I-D.patel-dispatch-cpc-oli-parameter\]](#) (Patel, M., Jesske, R., and M. Dolly, "Uniform Resource Identifier (URI) Parameters for indicating the Calling Party's Category and Originating Line Information," November 2009.)).

Still, a drawback of the outlined approaches above is that it does not allow any mechanism to distinguish different types of calls initiated by PSAPs. Not every call from a PSAP is indeed a response to an emergency call.

This leads us to another mechanism on top of the previously presented ones, namely the indication is that the communication attempt is of emergency nature. As such, it is a slight modification of the one presented previously. In addition to the indication that the calling party is a PSAP there is an expression that the specific call is of emergency services nature. This indication cannot be verified by external parties, similarly to the emergency call marking for a citizen-to-authority emergency call using a Service URN, because it heavily depends on the intention of the call taker itself.

4. Topics for Investigation

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When you make an IP-based emergency call to an IP-based PSAP then the PSAP will get two pieces of identity information about the emergency caller:

- *Contact-URI: Information that uniquely identifies the device the call came from.

- *Address of Record: Long-term contact information

Should the callback functionality be tied to a previous emergency call setup and as such enabled only for a specific time? For example, preferential treatment for callbacks could be provided only within one hour after the initial emergency call was made.

Is it expected that the callback reaches primarily the device that initiated the emergency call? In some cases the device that was used to originally initiate the call does not respond anymore to a callback (e.g. imagine a fixed line phone that was used to report a fire in a house and is out of order soon afterwards). Since the initial emergency call provided a second contact mechanism (namely the address of record) it could be used by the call taker as well. Should this communication also experience the same type of override privilege as the initially transmitted callback to the emergency caller's device?

Should any restrictions be made regarding the media being used for callback? Is it acceptable to return an instant message when the caller started the conversation with audio?

5. Security Considerations

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This document provides discussions problems of PSAP callbacks and explores the design space.

An important aspect from a security point of view is the relationship between the emergency services network and the VSP (assuming that the emergency call travels via the VSP and not directly between the SIP UA and the PSAP). If there is some form of relationship between the emergency services operator and the VSP then the identification of a PSAP call back is less problematic than in the case where the two entities have not entered in some form of relationship that would allow the VSP to verify whether the marked callback message indeed came from a legitimate source.

The main attack surface can be seen in the usage of PSAP callback marking to bypass blacklists, ignore call forwarding procedures and similar features to interact with users and to get their attention. For example, using PSAP callback marking devices would be able to recognize these types of incoming messages leading to the device overriding user interface configurations, such as vibrate-only mode. As such, the requirement is to ensure that the mechanisms described in this document can not be used for malicious purposes, including SPIT.

It is important that PSAP callback marked SIP messages, which cannot be verified adequately, are treated like a call that does not have any marking attached instead of failing the call processing procedure.

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6. Acknowledgements

We would like to thank members from the ECRIT working group, in particular Brian Rosen, for their discussions around PSAP callbacks. The working group discussed the topic of callbacks at their virtual interim meeting in February 2010 and the following persons provided valuable input: John Elwell, Bernard Aboba, Cullen Jennings, Keith Drage, Marc Linsner, Roger Marshall, Dan Romascanu, Geoff Thompson, Milan Patel, Janet Gunn.

7. References

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7.1. Informative References

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[RFC2119]	Bradner, S. , " Key words for use in RFCs to Indicate Requirement Levels ," BCP 14, RFC 2119, March 1997 (TXT , HTML , XML).
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7.2. Informative References

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[I-D.ietf-ecrit-framework]	Rosen, B., Schulzrinne, H., Polk, J., and A. Newton, " Framework for Emergency Calling using Internet Multimedia ," draft-ietf-ecrit-framework-10 (work in progress), July 2009 (TXT).
[I-D.ietf-sip-saml]	Tschofenig, H., Hodges, J., Peterson, J., Polk, J., and D. Sicker, " SIP SAML Profile and Binding ," draft-ietf-sip-saml-07 (work in progress), March 2010 (TXT).
[I-D.patel-dispatch-cpc-oli-parameter]	Patel, M., Jesske, R., and M. Dolly, " Uniform Resource Identifier (URI) Parameters for indicating the Calling Party's Category and Originating Line Information ," draft-patel-dispatch-cpc-oli-parameter-02 (work in progress), November 2009 (TXT).
[I-D.patel-ecrit-sos-parameter]	Patel, M., " SOS Uniform Resource Identifier (URI) Parameter for Marking of Session Initiation Protocol (SIP) Requests related to Emergency Services ," draft-patel-ecrit-sos-parameter-08 (work in progress), February 2010 (TXT).
[RFC3325]	Jennings, C., Peterson, J., and M. Watson, " Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks ," RFC 3325, November 2002 (TXT).
[RFC4474]	Peterson, J. and C. Jennings, " Enhancements for Authenticated Identity Management in the Session Initiation Protocol (SIP) ," RFC 4474, August 2006 (TXT).
[RFC4484]	Peterson, J., Polk, J., Sicker, D., and H. Tschofenig, " Trait-Based Authorization Requirements for the Session Initiation Protocol (SIP) ," RFC 4484, August 2006 (TXT).
[RFC5012]	Schulzrinne, H. and R. Marshall, " Requirements for Emergency Context Resolution with Internet Technologies ," RFC 5012, January 2008 (TXT).
[RFC5031]	Schulzrinne, H., " A Uniform Resource Name (URN) for Emergency and Other Well-Known Services ," RFC 5031, January 2008 (TXT).

Authors' Addresses

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	Henning Schulzrinne
	Columbia University
	Department of Computer Science
	450 Computer Science Building

	New York, NY 10027
	US
Phone:	+1 212 939 7004
Email:	hgs+ecrit@cs.columbia.edu
URI:	http://www.cs.columbia.edu
	Hannes Tschofenig
	Nokia Siemens Networks
	Linnoitustie 6
	Espoo 02600
	Finland
Phone:	+358 (50) 4871445
Email:	Hannes.Tschofenig@gmx.net
URI:	http://www.tschofenig.priv.at
	Milan Patel
	InterDigital Communications
Email:	Milan.Patel@interdigital.com