VIPR
Internet-Draft
Intended status: Standards Track
Expires: September 13, 2012

M. Petit-Huguenin
Unaffiliated
J. Rosenberg
jdrosen.net
C. Jennings
Cisco
March 12, 2012

The Public Switched Telephone Network (PSTN) Validation Protocol (PVP) draft-petithuguenin-vipr-pvp-04

Abstract

One of the main challenges in inter-domain federation of Session Initiation Protocol (SIP) calls is that many domains continue to utilize phone numbers, and not email-style SIP URI. Consequently, a mechanism is needed that enables secure mappings from phone numbers to domains. The main technical challenge in doing this securely is to verify that the domain in question truly is the "owner" of the phone number. This specification defines the PSTN Validation Protocol (PVP), which can be used by a domain to verify this ownership by means of a forward routability check in the PSTN.

Status of this Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF). Note that other groups may also distribute working documents as Internet-Drafts. The list of current Internet-Drafts is at http://datatracker.ietf.org/drafts/current/.

Internet-Drafts are draft documents valid for a maximum of six months and may be updated, replaced, or obsoleted by other documents at any time. It is inappropriate to use Internet-Drafts as reference material or to cite them other than as "work in progress."

This Internet-Draft will expire on September 13, 2012.

Copyright Notice

Table of Contents

1. Introduction

The validation protocol is the key security mechanism in ViPR. It is used to couple together PSTN calls with IP destinations based on shared knowledge of a PSTN call. This document relies heavily on the concepts and terminology defined in [VIPR-OVERVIEW] and will not make sense if you have not read that document first.

The protocol assumes that two enterprises, the originating one (enterprise O) initiates a call on the PSTN to an E.164 number ECALLED that terminates on the terminating enterprise (enterprise T). Each enterprise has a ViPR server, acting as a P2P node. The node in enterprise O is PO, and the node in enterprise T is PT. This PSTN call completes successfully, and knowledge of this call is known to PO and PT. Later on, PO will query the P2P network with number ECALLED. It comes back with a Node-ID PCAND for a node. At this time, PO can't know for sure that PCAND is in fact PT. All it knows is that some node, PCAND, wrote an entry into the DHT claiming that it was the owner of number ECALLED. The objective of the protocol is for PO to determine that node PCAND can legitimately claim ownership of number ECALLED, by demonstrating knowledge of the previous PSTN call. It demonstrates that knowledge by demonstrating it knows the start time, stop timer, and possibly caller ID for the PSTN call made previously.

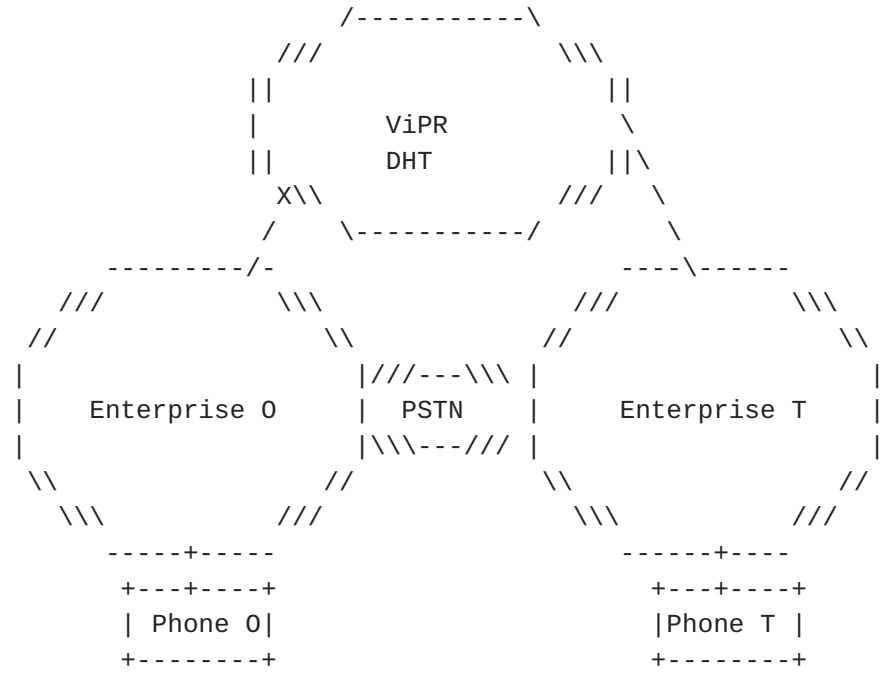


Figure 102: Validation Model

If node PCAND can demonstrate such knowledge, then enterprise O can assume that node PCAND had in fact received the call, which could only have happened if it had knowledge of the call to number ECALLED, which could only have happened if PCAND is in enterprise T, and thus it is PT. This is because PSTN routing is assumed to be "secure", in that, if someone calls some number through the PSTN, it will in fact reach a terminating line (whether it be analog, PRI, or other) which is the rightful "owner" of that number. If enterprise T was not the owner of the number, if would not have received the call, would not know its start/stop/caller ID, not be able to provide that information to PT, and not be able to satisfy the knowledge proof. This basic approach is shown in Figure 102.

A first question commonly asked is, why not just do regular authentication? What if we give each node a certificate, and then have the nodes authenticate each other? The answer is that a certificate certifies that a particular node belongs to a domain - for example, that node PT is part of example.com. A certificate does not assert that, not only is PT example.com, but example.com owns the following phone numbers. Therefore simple certificate authentication does not provide any guarantee over ownership of phone numbers.

In principle, it might be possible to ask certificate authorities, such as Verisign, to assert just that. However, traditionally, certificate authorities have been extremely hesitant to certify much at all. The reason is, the certifier needs to be able to assure that the information is correct. How can a certifier like Verisign verify that, in fact, a particular enterprise owns phone numbers? It could make a few test calls, perhaps, to check if they look right. However, these test calls are disruptive to users that own the numbers (since their phones will ring!). If the test calls are done for a subset of the numbers, it is not secure. If the certifier simply required, as part of the business agreement, that the enterprises provided correct information, the certifier might avoid legal liability, but the legitimacy of the service will be compromised and customers will stop using it. Furthermore, it has proven incredibly hard to do this kind of certification worldwide with a single certificate authority.

ViPR has, as a goal, to work anywhere in the world and do guarantee correct call routing with five nines of reliability. Consequently, traditional certificates and authentication do not work. It turns out to be quite hard to design a secure version of this validation protocol. To demonstrate this, we will walk through some initial attempts at it, and show how they fail.

2. The Wrong Way

The first attempt one might make is the following. PO takes the caller ID for the call, ECALLING and called number ECALLED for the call, and sends them to candidate node PCAND. These two identifiers - the called number E and the caller ID, form a unique handle that can be used to identify the call in question. Node PCAND looks at all of the ViPR Call Records (VCRs) of the calls over the last 48 hours, and takes those with the given called party number and calling party number. If there is more than one match, the most recent one is used. We now have a unique call.

Now, node PCAND demonstrates knowledge of this call by handing back the start and stop times for this call in a message back to PO. This approach is shown in Figure 103.

Figure 103: Incorrect Validation Protocol: Take 1

Unfortunately, this method has a major problem, shown in Figure 104.

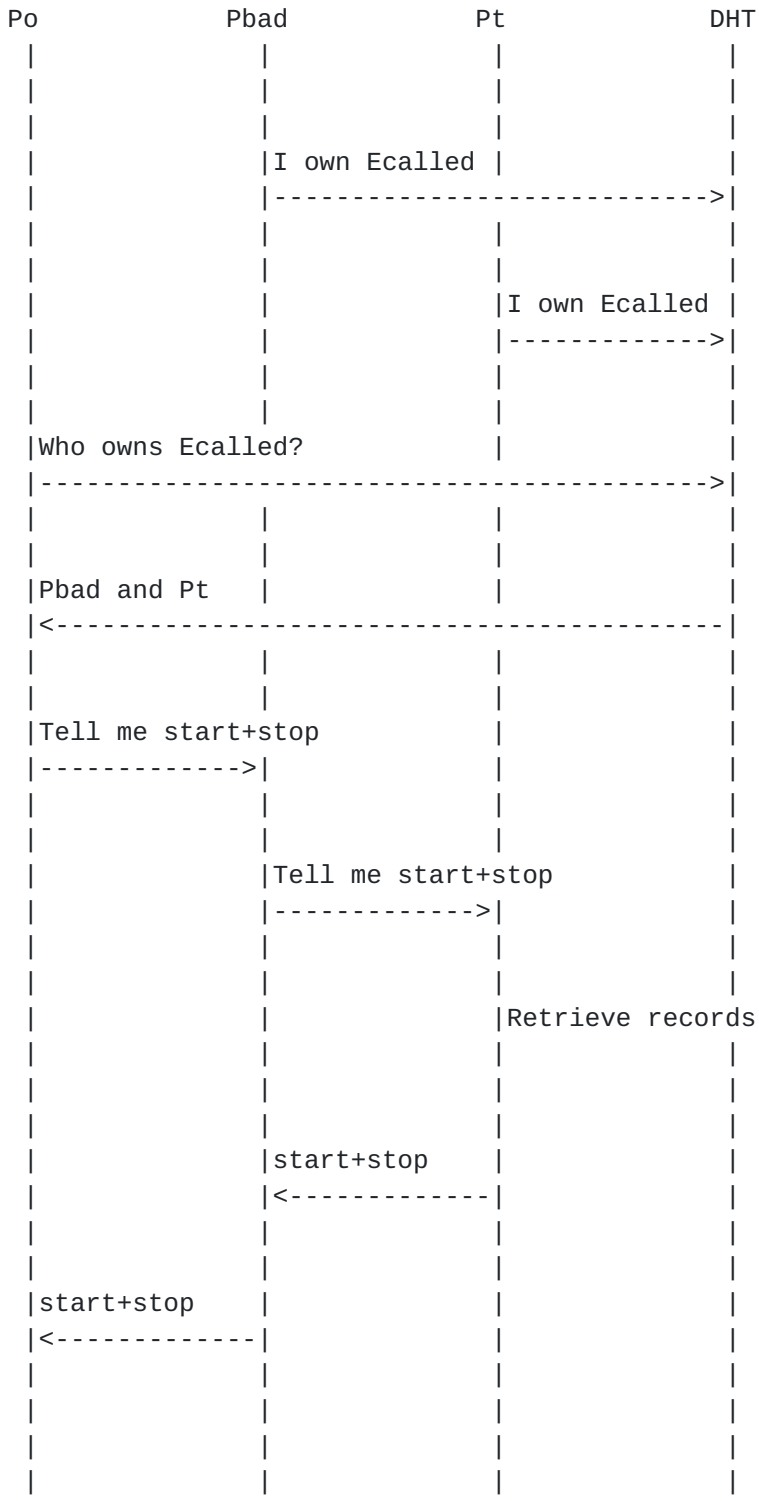


Figure 104: Attack for Incorrect Validation Protocol

Consider an attacker BadGuy PBAD. PBAD joins the P2P network, and advertises a number prefix they do NOT own, but which is owned by

enterprise T and node PT. Now, when PO queries the DHT with number ECALLED, it comes back with two results - the one from PBAD and the one from node PT. Details of querying the DHT are provided in [VIPR-RELOAD-USAGE]. It begins validation procedures with both. PBAD will now be asked to show the start and stop times for the call, given ECALLED and ECALLING. It doesn't know that information. However, node PT does. So now, PBAD, acting as if it where the originating party, begins the validation protocol with node PT. It passes the calling and called numbers sent by PO. PT finds a match and returns the call start and stop times to PBAD. PBAD, in turn, relays them back to PO. They are correct, and as a consequence, PO has just validated PBAD!

Typically, the first response to this is, "Well the problem is, you let two separate people write the same number into the DHT. Why don't you make sure on the right one is allowed to write it in?". That is not possible, since there is no mechanism by which an arbitrary node in the DHT can determine who is the rightful owner of this number. "OK", the reader responds, "So instead, why don't you define a rule that says, if there are two entries in the DHT for a particular number, consider this an attack and don't try to validate the number". That would prevent the attack above. However, it introduces a Denial of service attack. An attacker can pick a target number, write it into the DHT, and prevent successful validation from happening towards that number. They can't misroute calls, but they can stop ViPR from working for targeted numbers. That is not acceptable. ViPR has to be immune from attacks like this; it should not be possible, through simple means such as configuration, for an attacker to cause a targeted number to never be validated.

One might be tempted to add a signature over the call start and stop times, but it does not help. BadGuy can just resign them and relay them on.

In essence, this simple approach is like a login protocol where the client sends the password in the clear. Such mechanisms have serious security problems.

Realizing the similarities between the validation protocol and a login protocol, a next attempt would be to use a much more secure login mechanism - digest authentication. To do this, domain O takes the called number E and the caller ID, and send them to node P. Node P treats these as a "username" of sorts - an index to find a single matching call. The start time and stop times of the call become the "password". Enterprise O also sends a big random number - a nonce - to node P. Node P then takes the random number, takes the password, hashes them together, and sends back the hash. All of this is done over a TLS connection between enterprise O and node P. Digest over

TLS is very secure, so surely this must be secure too, right? Wrong!

It is not. Indeed it is susceptible to EXACTLY the same attack described previously. This is shown in Figure 105.

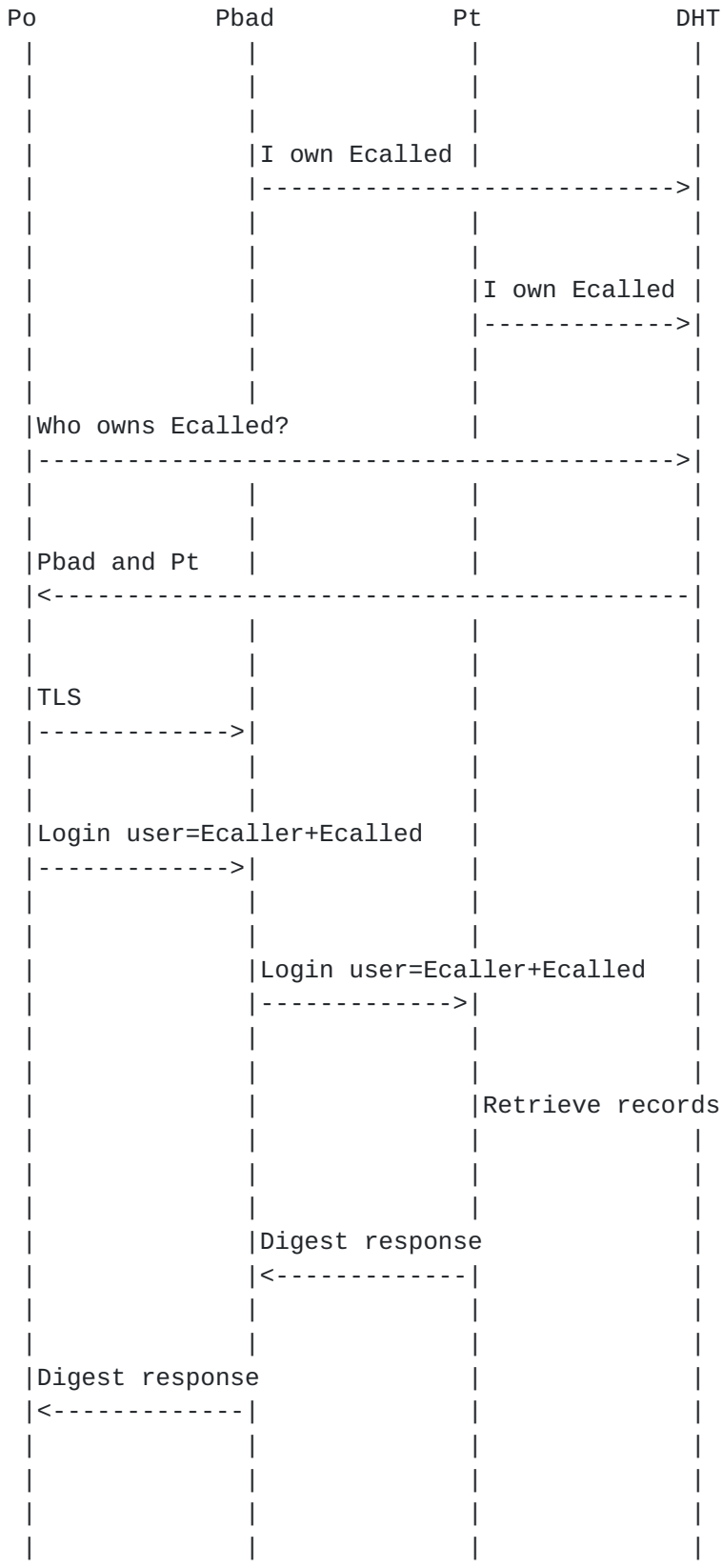

Po
Pbad
Pt
DHT
I own Ecalled
I own Ecalled
Who owns Ecalled?
Pbad and Pt
TLS
Login user=Ecaller+Ecalled
Login user=Ecaller+Ecalled
Retrieve records
Digest response
Digest response

Figure 105: Trying Digest for Validation

In a similar attack, PBAD could pick a random called number it is interested in, query the P2P network for it, find node PT. Then, provide node PT the number ECALLED to attack, and ECALLING, assuming it can guess a likely caller ID. It then takes the received digest response, and goes through every possible start/stop time over the last 24 hours, running them through the hash function. When the hash produces a match, the PBAD has just found a full VCR for node PT. It can then write into the DHT using number E as a key, pointing to itself, and satisfy validation requests against it, without even needing to ask node P again. Our first attempt is susceptible to this attack too.

The problem here is that the call start and stop times have "low entropy" - they are not very random and are easily guessable, just like a poorly chosen password.

What we really want to do here is have a "login" protocol that creates a secure connection between a client and a server, where we use the called number and caller ID as a "username" to identify a PSTN call, and then use the start and stop times as a "password". But our login protocol has to have some key features:

1. Someone posing as a server, but which does not have the username and password, cannot determine the username and password easily as a consequence of an authentication operation started by a valid client, aside from successfully guessing in the one attempt it is given on each connection attempt.
2. Someone posing as a client, but which does not have the username and password, cannot determine the username and password as a consequence of an authentication operation started against a valid server, aside from guessing in the one attempt it is given on each TLS connection attempt.
3. An active MITM, who is explicitly on the path of the exchanges and has visibility and the ability to modify messages, cannot obtain the shared secret, nor can it observe or modify information passed between the client and real server.
4. It is impossible for a passive observer to view the exchange and obtain the shared secret or any of the material that is exchanged.
5. It is impossible for a rogue client or rogue server to participate in a login with a legitimate peer, and then take the messages exchanged, and run an offline dictionary attack to work through every possible combination of start and stop times. Fortunately, these properties are provided by a class of password authentication protocols called Encrypted Key Exchange or EKE protocols.

3. EKE Protocols

EKE protocols were proposed in 1992 by Steve Bellovin. Since their proposal, numerous variations have been defined. One of them, the Secure Remote Password protocol, was standardized by the IETF in RFC 2945 [RFC2945]. A TLS mode of SRP was later defined in RFC 5054 [RFC5054]. It is the latter protocol which is actually used by ViPR. A high level overview of EKE protocols is shown in Figure 106. Alice and Bob share a shared secret P. Alice generates a public/private keypair. She then takes her public key, and encrypts it using her password as a symmetric encryption key. She sends this encrypted key to Bob. Bob, who shares the password, uses it as a symmetric key and decrypts the message, obtaining Alice's new public key. Bob then constructs a big random number R, which is to be used as a session key. Bob then encrypts R with the public key he just got from Alice, and sends that to Alice. Now, Alice, using her public key, decrypts the message and obtains the session key R.

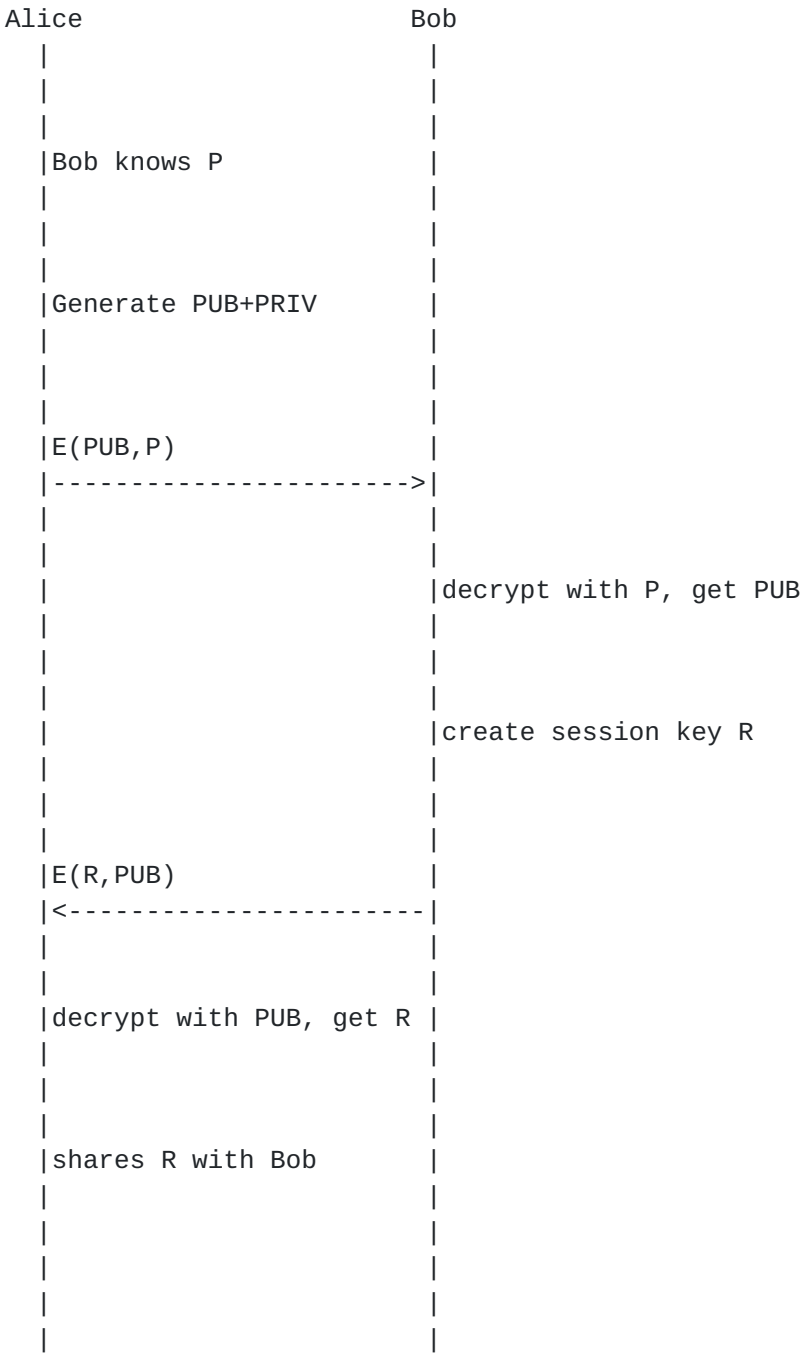
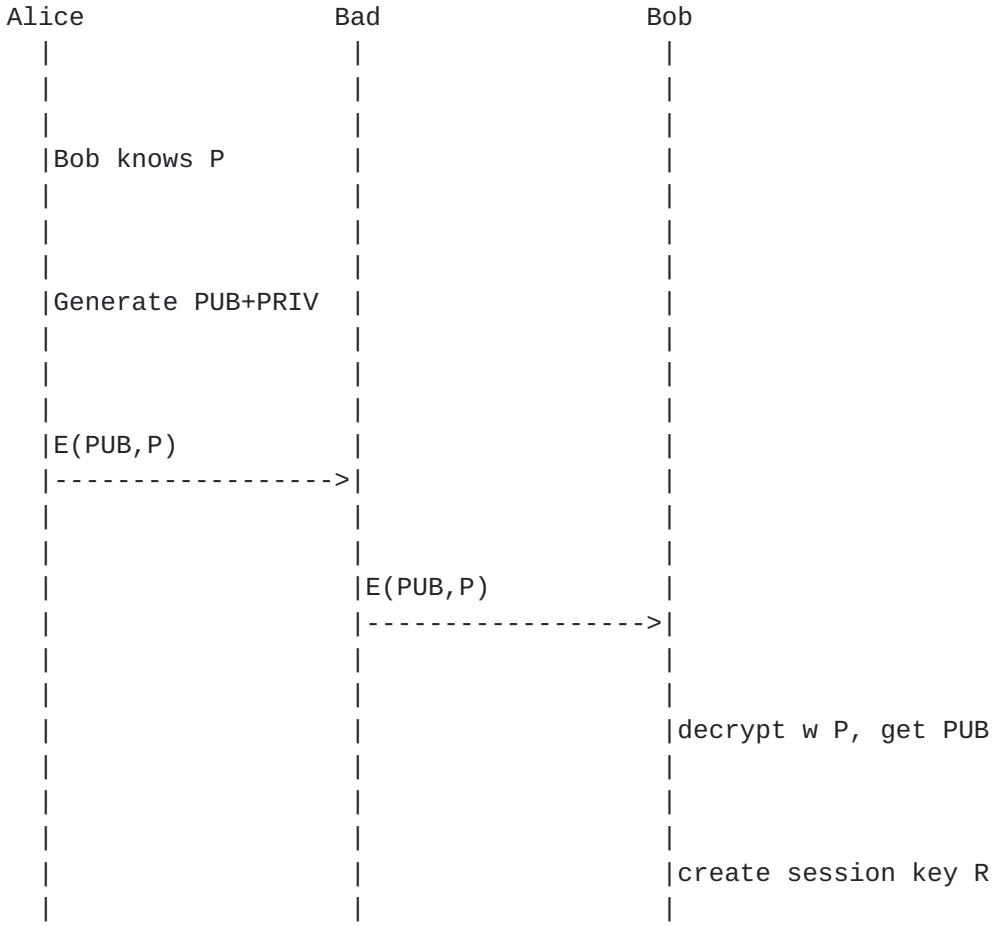


Figure 106: High Level EKE Model

At this point Alice and Bob share a session key R which can be used for authentication (by having Alice and Bob prove to each other that they have the same value for R) or for encrypting data back and forth. How does this help? Consider our man-in-the-middle attack again, in Figure 107. Once again, Alice shares a password with legitimate user Bob. However, she begins the "login" process with

BadGuy. She passes E(PUB,P) to BadGuy. BadGuy doesn't know P, so he can't decrypt the message. More importantly, he can't run through each possible password P and decrypt the message. If he did, he wouldn't be able to tell if he got it right, since PUB appears random; the decryption process would produce a random string of bits whether it was successful or not. So for now, BadGuy can only pass it on. BadGuy now intercepts E(R,PUB). Now, BadGuy can try the following. He can run through each P, decrypt E(PUB,R), obtain PUB. However, since we are using asymmetric encryption (i.e., public key encryption), even with PUB he cannot DECRYPT E(R,PUB)! BadGuy does not have the private key, which he needs to decrypt. Given a public key, he cannot guess the private key either. That is how public/ private keying systems work. That is the secret here to making this work. So, once again, BadGuy has no choice but to pass the message on. Now, Alice and Bob share R but it is unknown to BadGuy. Bob now takes his Node-ID, encrypts it with R, and sends to Alice. Once again, BadGuy doesn't have R and can't get it, so he has no choice but to pass it on. Alice decrypts this Node-ID with R, and now knows that she is actually talking to Bob - since she has Bob's Node-ID. Other data can be substituted for the Node-ID, and indeed this is what happens in the actual validation protocol.



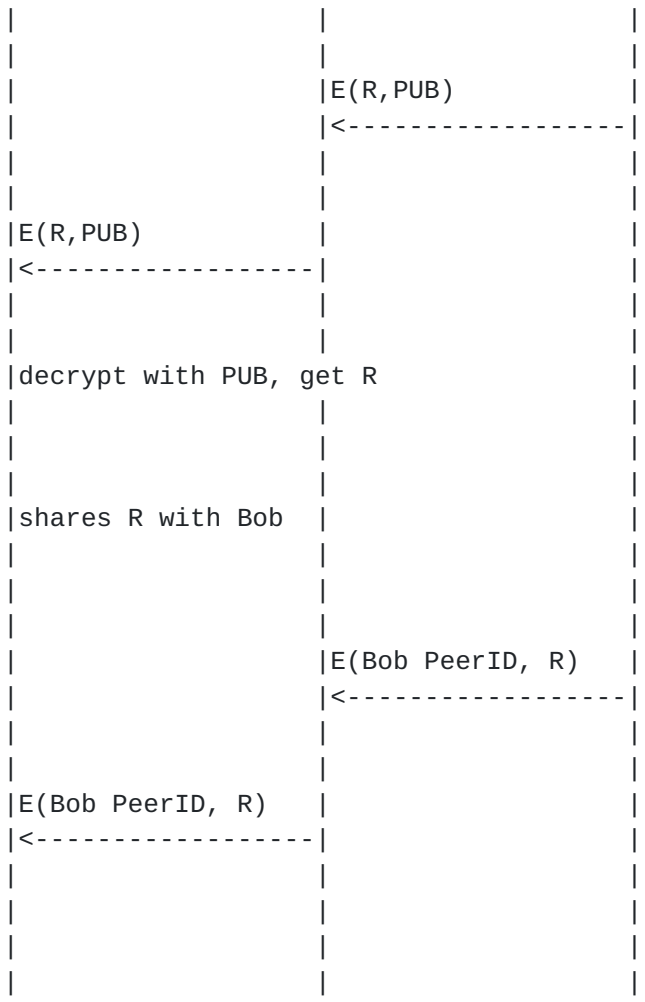


Figure 107: Attacking EKE Protocols

However, the main point of this exercise is to demonstrate that EKE protocols have the desired properties.

4. Terminology

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].

5. Protocol Overview

The validation protocol begins with the following assumptions:

1. Node PO wishes to validate with node PCAND, and has its Node-ID (which it obtained via the DHT) and VServiceID (which it also obtained via the DHT Fetch).
2. Node PO and PCAND have a series of call records over the last 48 hours, uploaded by their call agents. Each call record contains an E.164 calling and called party number, and a start and stop time in NTP time. On the terminating side, each call record is also associated with a VServiceID.
3. Node PO is seeking to validate a call to called number ECALLED with caller ID ECALLING.

The validation protocol operates by having the originating node make a series of attempts to connect to, and "login" to the terminating node. Each "login" attempt consists of establishment of a TCP connection, and then execution of TLS-SRP procedures over that connection. TLS-SRP[RFC5054] relies on a shared secret - in the form of a username and password - in order to secure the connection. In ViPR, the username and password are constructed by using information from a target VCR along with the VServiceID learned from the DHT. The "username", instead of identifying a user, identifies a (hopefully) unique VCR shared between the originating and terminating nodes. The "password" is constructed from the VCR such that it knowledge of the information is unique to knowledge of the VCR itself.

Unfortunately, it is difficult to construct usernames and passwords that always uniquely identify a VCR. To deal with this, the validation protocol requires the originator to construct a series of usernames and passwords against a series of different nodes and their corresponding IP addresses and ports, and then run through them until a connection is securely established.

6. Username and Password Algorithms

ViPR provides two different algorithms for mapping from a particular VCR to a username and password:

1. Method A: This method makes use of the called party and calling party number to form the username, and the start and stop times of the call to form the password.
2. Method B: This method makes use of the called party number, along with a point in time in the middle of the call, as the username, and then the start and stop times to form the password.

The originating node will first try validations with method A, and if those all fail, then try with method B. The method itself, along with necessary information on how to use the method, is encoded into the

username itself. The format of the username is (using ABNF [RFC4234]):

```
username = method-a / method-b / future-method
future-method = method ":" method-data
method = private-method / experimental-method / standard-method
private-method = "p-" 1*(alphanum / "-" / ".")
experimental-method = "x-" 1*(alphanum / "-" / ".")
standard-method = 1*(alphanum / "-" / ".")
            ; Cannot start with "p-" or "x-"
method-data = 1*(alphanum / method-unreserved)

method-a = "a:" vserviceid orig-number terminating-number
           rounding-time
method-b = "b:" vserviceid terminating-number timekey rounding-time

vserviceid = "vs=" 1*32HEXDIG ";"
orig-number = "op=$2a$" 1*2DIGIT "$" 1*(alphanum / "/" / ".") ";"
terminating-number = "tp=+" 1*15DIGIT ";"
timekey = "tk=" 1*10DIGIT "." 1*10DIGIT ";"
rounding-time = "r=" 1*6DIGIT ";" ; Cannot be equal to 0
```

This format starts with the method, followed by a colon, followed by a sequence of characters that are specific to the method. Both methods a and b rely on conveyance of information attributes that make up the username. Each attribute follows a specific format.

Because having the originating number in clear in the username would create some privacy issues (see [I-D.procter-vipr-privacy-concerns]), it is first hashed using a bcrypt hash [1].

Examples include (the first example is split in 3 lines for formatting purpose):

```
a:vs=7f5a8630b6365bf2;
  op=$2a$10$uhNBlMT5O063n5/YMlg3Y.xTmZuHMeAFbSxhwSacX87aujFxPvoxG;
  tp=+14085553084;r=1000;
b:vs=7f5a8630b6365bf2;tp=+14085553084;tk=172636364.133622;r=1000;
```

Both methods use a rounding factor R. This is used to round the start and stop times in the password to a specific nearest multiple of R (which is in milliseconds). This rounding is done because the passwords need to be bit exact and we need to compensate for different measured values.

If we will fallback to method B (which works more often), why have both? There are two answers:

1. The caller ID mechanism (method A) will work, and the non-caller ID (method B) won't work, for numbers like 8xx.
2. Method A has much higher entropy (see analysis in Section 10.1). Validating with it provides greater confidence in the validity of the number. In this phase, nothing is done with this "confidence". However, in later phases, it is anticipated that low-confidence numbers will require multiple validations for different calls to occur before they are trusted. To allow for this feature to be added later, validation with both methods must be present in the initial release.

The sections below detail precisely how these are constructed.

7. Originating Node Procedures

Most of the work for validation is on the side of the originator. It establishes connections and performs a series of validation checks.

7.1. Establishing a Connection

The first step in the process is to establish a TCP connection to PCAND. To do that, PO sends a RELOAD AppAttach message targeted towards PCAND, using the Application-ID defined in [VIPR-RELOAD-USAGE]. Once connected, TLS-SRP is run over the connection.

7.2. Constructing a Username and Password

When a terminating node receives a username in a format it doesn't understand, it fails the validation. This allows for graceful upgrade to new mechanisms in the future.

7.2.1. Method A

The PO examines the VCR it is using for validation. It extracts the called party numbers which is E.164 based. It also extracts the calling number, which is hashed with bcrypt hash and extract the salt and the number of rounds from it. This VCR will have been uploaded at a previous point in time. PO then examines the VCRs posted in the time since this one was uploaded, and looks for any more recent VCRs with the same called party numbers and a calling party number which, when hashed with the salt and the number of rounds, is equals to the has extracted, regardless of VService. If it finds one or more, it takes the most recent one (as measured by its end time). If it finds no more recent, it uses the VCR which triggered the validation in the first place.

Why do this? This deals with the following case. User A calls user B, causing a VCR to be uploaded. The originating node sets a timer, which fires 12 hours later. However, within that 12 hour period, A called B again. If node A provides the caller ID and called party numbers as the "key" to select a VCR, it will match multiple records over the past day. We need to pick one, so the most recent is always used. This requires the originating node to know and use the most recent VCR. Furthermore, we must choose the most recent VCR regardless of its VService, because the originating Upload VCRs are sent using an arbitrary VService. Thus, the more recent call may have been done using a different VService than the one which triggered the validation. Since the actual Vservices are not common between originating and terminating sides, we must choose the VCR on the originating side regardless of VService. The username is constructed by using the syntax for method A described above. The calling party number is set to "op", and the called party number is set to "tp", and "r" is the value of Tr as an integral number of milliseconds. The VServiceID learned from the dictionary entry is used as the value of "vs".

This username will select the identical VCR at the terminating node, under the following conditions:

1. PT is aware of all calls made to the called party number. This property is true so long as each incoming number is handled by a single call agent within a domain, and furthermore, the VCR for calls to that number is always posted to a ViPR server which advertises that number into the DHT. These properties are readily met by ViPR for typical single user numbers. For 8xx numbers, which are translated within the PSTN and may route to a multiplicity of non-8xx numbers, it is more difficult. ViPR will only work with 8xx numbers if all calls to those numbers get sent to agents which share the same ViPR server.
2. PO is aware of all calls made to the called party number with the given caller ID. This property can be hard to meet. If the caller ID for a call is set to the number of the calling phone, and all VCRs made from that phone are posted to the same ViPR server, that server will know about all calls made by the domain with the given DID in the caller ID. However, in domains that set the PSTN caller ID to the attendant line number, it is possible that there would be two separate agents, each utilizing different ViPR servers. A user in each agent calls the same number, and the same PSTN caller ID is used. However, one ViPR server knows about one of the calls, and a different ViPR server knows about the other call. However, PT knows about both. In that case, validation from one of the ViPR servers will fail, and from the other, succeed.

3. There were no calls on the PSTN to the called party which spoofed the caller ID to match the caller ID used by the valid enterprise. In that case, PT will have a VCR for a call with a matching calling/called party number, but this VCR is unknown to PO since the call was not actually made by the originating enterprise. This attack is described in more detail in XXXX.

Next, the password is selected. The password is basically the start and stop times for the call. However, the SRP protocol requires a bit exact agreement on the password. Unfortunately, the calling and called parties will not have the same values for the start and stop times, for several reasons:

1. The call start time at the originating and terminating ends will differ by the propagation delay of the call acceptance message through the PSTN. This can be several hundreds of milliseconds.
2. The clocks at the originating and terminating ends may not be synchronized, which can also introduce different values for the start and stop times.
3. The call termination time at the originating and terminating ends will also differ by the propagation time; this propagation time may in fact be different for the call acceptance and termination.

It is also important to note that agreement on a call acceptance and termination time assumes an explicit signaling message is sent for these two events. In the case of analog FXO ports, there is no signaling at all, and consequently, these points in time cannot be measured. It is possible to agree upon other call characteristics when analog lines are in use, but they have much worse accuracy and consequently much, much lower entropy. For this reason, this specification of ViPR only works in telephony systems with explicit messaging for call acceptance and termination, which includes PRI, SS7, BRI, analog trunks with answer and disconnect supervision, and CAS trunks.

To deal with these inaccuracies in timing, the start and stop times need to be rounded. Let Tr be the rounding interval, so that each time is rounded to the value of N*Tr for integral N, such that N*Tr is less than the start or stop time, and (N+1)*Tr is greater than it. In other words, "round down". If Tr=1 second, this would round down to the nearest second.

Unfortunately, rounding doesn't fully help. Lets say that the difference between the start times on the originating and terminating nodes is delta. We can still have different values for the start time if one side rounds to one value, and the other side to a different value. If delta=100ms and Tr=1s, consider a start time of 10.08 seconds on one side, and 9.98 on the other side. One side will

round to 10 seconds and the other to nine seconds. The probability of this happening is approximately delta/Tr. We could just make Tr really large to compensate, but this reduces the entropy of the system (see below).

To deal with this, the originating node will actually compute FOUR different passwords. For the start time and stop time both, the originating node will round down as follows. Let T be the time in question. Let N be the value such that N*Tr <= T < (N+1)*Tr. In other words, N*Tr is the nearest round-down value, and (N+1)*Tr is the nearest round up. Let T1 and T2 be the two rounded values of T. We have:

```
if (T >= ((2N+1)/2) * Tr) then:
                          T1 = N*Tr
                          T2 = (N+1)*Tr
if (T <  ((2N+1)/2) * Tr) then:
                          T1 = N*Tr
                          T2 = (N-1)*Tr
```

In other words, if T is in the top half of the rounding interval, we try the rounded values above and below. If T is in the bottom half, we try the rounded values below, and below again. Pictorially:

[[TBD]]

Figure 108: Rounding mechanism for validation protocol

These are tried in the following sequence:

1. Try Tstart-1 and Tend-1.
2. Try Tstart-2 and Tend-1.
3. Try Tstart-1 and Tend-2.
4. Try Tstart-2 and Tend-2.

For example, if the originating side has a start time of 10.08 and a stop time of 30.87, the four start and stop times with Tr=1s are:

Start	Stop
10	30
9	30
10	31
9	31

Each of these times is represented in 64 bit NTP time (Tr can be

configured to less than 1s in which case there will be non-zero values in the least significant 32 bits). Each password is then computed by taking the 64 bit start time, followed by the 64 bit end time, resulting in a 128 bit word. This word is base64 encoded to produce an ASCII string representation of 21 characters. To perform the caller ID based validation, the SRP-TLS procedure is done four times, once with each of the four username/password combinations (of course the username is identical in all four cases). As long as delta is less than Tr/2, one of this is guaranteed to work.

7.2.2. Method B

Unfortunately, in many cases caller ID cannot be used as an identifier for the VCR. This is because:

1. CallerID is frequently suppressed in the PSTN, and not delivered. This is especially true in international cases.
2. CallerID is sometimes munged by the PSTN, and delivered, but with a different value than was sent by the originator. This happens in certain arbitrage interexchange carriers.

Consequently, if no caller ID was delivered at all, the terminating side will not have a matching record. In that case, it informs the calling side that it should abort and revert to method B. If munged, it will also abort for the same reason.

If the caller ID attempt aborts, PO now tries a different approach. In this approach, the "username" is the combination of the called party number and a point during the call, selected at random. The password is equal to the start and stop times of the call. This method uses the method-tag "B" in the username.

Unlike method A, with method B, the VCR which triggered the validation is used, regardless of whether there were other, more recent, calls to the same calledparty number! This is because, in method B (unlike method A), the time itself is used as a key to select a VCR. Furthermore, using a more recent VCR does not interact properly with multi-tenancy. The called number and point during that call will select an identical VCR on the terminating side if the following conditions are met:

1. For the called party number, there was not more than one call in progress made to that number at the same time. This is generally true for numbers for a single user; typically there is only one active call at a time. Of course, it is possible a user receives a call, and then gets another. It then puts the first on hold while the second call is taken. In these cases, it is possible that the "username" will select a different VCR on PT, in which

case the validation fails. More troubling are numbers representing call centers, conference bridges, 8xx numbers, and attendant numbers, all of which frequently have multiple calls in progress to them at the same time. As a consequence, for these types of called numbers, validation is typically only going to work if caller ID is delivered. Fortunately, 8xx numbers are only national in the first place, so it is likely that this will work.

2. PO is aware of all calls made from within its enterprise to ECALLED. This can fail if there are multiple ViPR servers serving different agents, and a call is made from one agent, sent to one ViPR server, and a call to that same number is made on a different agent, send to a different ViPR server. As in the caller ID case, this will still be OK in many cases - the validation from one ViPR server succeeds, the other fails.
3. PT is aware of all calls made to ECALLED. The same caveats as described above for the caller ID mechanism apply. PO takes the VCR, and chooses a time Tkey which is uniformly distributed between Tstart+Tr and Tstop-Tr. The usage of the Tr here is to make sure that Tkey is squarely inside of the call start and stop for PT as well. Note that, because Tkey is not a password, it is sent in the clear and does not need to be rounded.

The username encodes the called party number, Tkey, the DHT, and the VServiceID learned from the DHT query. The password is computed identically to method A.

7.3. Requesting Validation

Once the SRP-TLS connection is up, data is exchanged. This is done through a single VAP transaction initiated by PO. This transaction is only VAP in the sense that it utilizes the basic syntax (the header and TLV attribute structure), and its request/response model. Other than that, it is effectively a different protocol - the validation protocol.

PO sends a VAP request with method ValExchange (0x00d). It contains one attribute, Domain. The originating ViPR server obtains this domain by looking at the VService of the VCR that was eventually used for the validation. Note that, in cases where the VCR which triggered the validation, is different than the one actually used for validation (because a more recent VCR to the same number was found), it is important to use the VService associated with the VCR which was actually used for validation, and NOT the VService associated with the VCR which triggered the attempt. Multi-tenancy does not work properly without this. The domain from the VService is placed into the message. This is basically the domain name of the originating enterprise. It is included since it is needed by PT to compute the

ticket.

PO will then receive a response. If it never receives a response within a timeout, it considers the validation to have failed, and continues to the next choice. If it receives any kind of error response, including a rejection due to a blacklisted domain, it considers validation to have failed, and continues to the next choice. If it is a success response, it will contain one attribute - ServiceContent, which contains a ValInfo XML object. ValInfo is an XML object which contains the SIP URIs and an optional ticket. The ViPR server must parse the ValInfo XML object and perform verification on it to avoid attacks. The following checks are done:

1. Extract the <number> element. This will contain a single number. That value is compared with the E.164 called party number which was just validated. If they do not match, this is a potential attack, and the XML is discarded and the ViPR server acts as if validation failed. However it does not generate an alarm.
2. Remove any extensions to the XML which are not supported by the ViPR server (no extensions defined, so in this version, any elements except for the <ticket>, <number>, <route>s and their embedded <SIPURI> are removed.
3. If there is no <route> element and no <ticket> element in the XML object, it means that the PVP server can not provide these elements yet because not enough entropy was accumulated. The PVP client MUST stop trying new methods.
4. Verify that the <route> element contains a single element, <SIPURI>.
5. Verify that the SIP URI is not larger than 614 characters, contains a domain name that is a valid set of domain name characters, contains a user part that is a valid set of characters, if it contains maddr, that the maddr is a valid domain or IP and less than 255 characters, and if there is a port, it is within 0-65535. This is for security purposes; to make sure a malicious ViPR server on the terminating side cannot send invalid URI and attack the call agent.
6. Verify that each SIP URI contains the same domain name. Once the checks and fixes are done, the patched XML is passed to subscribers in a Notify as described in [VIPR-VAP].

8. Terminating Node Procedures

8.1. Waiting for SRP-TLS

PT will wait for an AppAttach request on the Application-ID defined in [VIPR-RELOAD-USAGE] and the connection is established, it begins waiting for SRP-TLS. The TLS messaging will provide PT with a

username.

It parses the username and determines the method. If the value of the method is not "a" or "b", this is a new method not supported by the node. The SRP-TLS procedures should be failed. If the method is "a", it is the caller ID mechanism. The called number, calling number, VService, and rounding time are extracted. PT then searches through its VCRs over the last 48 hours for one with a matching called number and caller ID and VService whose VServiceID matches the one from the username:

1. If none are found, PT proceeds with the SRP-TLS exchange, but using a fake username and password. This will cause the validation to eventually fail.
2. If one is found, it is used.
3. If more than one is found, the one with the most recent end time is used.

The start and stop times from the selected VCR are taken. Using the value of Tr from the username, both times are rounded down to the nearest multiple of Tr. Note that, this rounding is different than the one used on the originating side. The values are ALWAYS rounded down. So if the stop time is 10.99 and Tr is one second, the rounded down value of 10 is used. The start and stop times are then represented as 64 bit NTP times (after rounding), concatenated, and base64ed to produce a 21 character password. This is the password used with SRP-TLS.

Note that, the originating node will try up to four different password combinations. One of these should work, the others will cause SRP-TLS to fail due to differing shared secrets. However, it is the job of the originator to perform these four; to the terminating node, they are four separate attempts. Processing of SRP-TLS login attempts is stateless on the terminating side. This means that each attempt is treated independently by PT. It performs identical processing on each SRP-TLS attempt - examine the username, find a matching VCR, extract password, and fail the attempt or continue to success. The originating side has the main burden of sequencing through the various mechanisms.

If the method is "b", PT uses the extracted called party ID and a time in the middle of the call. It searches through all VCR records whose called number matches and whose VServiceID matches, and of those, takes the ones where Tkey is between Tstart and Tstop. Of those, if more than one match, the one with the most recent Tstop is used. Tstart and Tstop for that VCR are extracted, and converted to a password just as is done for the PO. The resulting SRP-TLS procedure will then either succeed or fail. Note that, if a domain

has multiple Vservices that contain the same number, there will be multiple VCRs for calls to that number, and there will be multiple validation attempts, one for each of the Vservices.

Note that there could be multiple successful validations coming from different domains for one specific VCR, so VCRs should not be removed before the end of the 48 hours period. This can happen when a calling domain uses a PSTN provider that is itself VIPR enabled.

8.2. Receiving Validation Requests

PT listens for incoming VAP/validation requests once the TLS connection is up. It rejects anything but a ValExchange method with a 400 response. This allows for future extensibility of the validation protocol. If the request was ValExchange, it extracts the domain name. This will be something like "example.com". PT knows the VCR against which validation succeeded. That VCR is associated with a VService. The ViPR server checks the domain in the ValExchange request against the black/white list associated with that VService. If no VService is currently active, the ValExchange is rejected with a 403. If there is one active, and if the domain appears on the black list, or does not appear in the white list, the ViPR server rejects the ValExchange request with a 403 error response, indicating that this domain is not allowed to call.

If the domain was in the whitelist or not in the blacklist, or there was no whitelist/blacklist, PT constructs a successful response to the ValExchange request. It contains one attribute: ServiceContent. It has a ValInfo XML object, which contains a number, and optionaly a ticket and a series of routes.

PT then check if there is enough entropy accumulated for this domain and this number by checking an internal database. If entropy has already been stored by a previous validation, it is added to the entropy provided by this new validation. If the new total, or the entropy provided by this validation, is higher than the threshold configured in PT, then a route and ticket will be sent as described in the following paragraphs. If the new total is lower than the threshold, then it is stored in the internal database and the ServiceContent returned contains no ticket and no route.

The number is always the E.164 number which was just validated, including the plus sign. Note that this will also appear in the ticket. The route element is the sequence of route elements for each instance associated with the vservice.

Details of the ticket are provided in [VIPR-SIP-ANTISPAM] but the ticket attribute is constructed as follows:

1. A ticket unique ID TLV is created, containing a randomly chosen 128 bit value as the ticket ID. That is the first TLV in the ticket.
2. A salt TLV is created, containing a random 32 bit value. This is the second TLV in the ticket.
3. The validity has the start time set using the current time as the start time, and the current time + the ticket lifetime as the end time. The ticket lifetime is a per-DHT configurable parameter. The terminating ViPR server will have performed the validation using a particular VService; the DHT for that VService is used to find the right value for this parameter.
4. Number: This is the terminating number, in E.164 format, which was just validated.
5. Granting node: this is set to one of the Node-IDs associated with this ViPR server. Any will do.
6. Granting domain: This value is taken from the domain part of the SIP URI associated with the VService in which the validated VCR was found.
7. Granted-To domain: This is formed using the Domain sent in the ValExchange request.
8. Epoch: This is the current epoch associated with the password.
9. Integrity: Using the current password, this is computed from the rest of the Ticket.

The resulting sequence of TLVs is base64 encoded and that is placed into the ticket element in the ServiceContent attribute in the ValExchange response.

9. Syntax Details

This section enumerates the methods and attributes used by PVP.

The methods and their corresponding method values, are:

Method	Value
ValExchange	0x00d

Figure 1: PVP Methods

The attribute names and corresponding types are:

Attribute Name	Type
Domain	0x3001

Figure 2: PVP Attributes

10. Security Considerations

[[This section is mostly missing and needs to be done.]]

10.1. Entropy

[[The entropy obtained in the information from the PSTN calls significantly impacts the security of this protocol. This section needs to provide an analysis of how much entropy actually exists in this information.]]

[[Defines the worst case of conference calls and resulting entropy]]

10.2. Forward Routing Assumptions

[[Discuss forward routing security in PSTN and explain how this protocol is reliant on that.]]

11. IANA Considerations

11.1. PVP Method Registry

IANA shall establish and maintain a "PVP Method" Registry, listing the following information:

- Method tag: A string uniquely defining a method. This string cannot use a "p-" or "x-" prefix.
- Defining publication: The RFC used to define the PVP method, as defined in the registration process below.

11.2. Registration Process

Private PVP methods starts with "p-" (e.g. "p-fingerprint"). Private methods MUST be only used inside a specific domain and MUST NOT be registered.

Experimental PVP methods starts with "x-" followed by a domain name owned by the entity describing this method, with the domain name reversed (e.g. "x-com.example.fingerprint"). Experimental methods can be used between domains, MAY BE registered, but implementers use them at their own risk.

All other PVP methods MUST be registered based on the "specification required" option defined in [RFC2434], with the further stipulation that the "specification" is an RFC of any category.

The defining RFC MUST clearly identify and describe, for each tag

being registered:

- Selectors: A list of name-values used to find the matching call record on the terminating side.
- Selectors usage: A text describing how the selectors are processed on the terminating side, especially what to do when multiple call records match the selectors.
- Parameters: A list of name-values that are used to guide the PVP processing.
- Secret: The secret data that is used to verify that the VCRs on both side match.
- Secret usage: A text describing how the secret data is generated, especially when one call record can generate multiple different secrets.
- Priority: The priority assigned to this method which an unsigned integer between -32768 and +32767. The priority for a new method MUST be the median of the priority of the method that will be selected immediately before and the priority of the method that will be selected immediately after this method, using the minimum and maximum values if there no lower or higher methods. Multiple methods can share the same priority to mean that there is no specific order to try them.
- Replacement: The name of an experimental method that this new method optionally replaces. An implementation supporting both the experimental and standard method and processing a VipRRegistration record containing both MUST NOT try the experimental method. The text MUST also indicates the date after which implementation will stop using the experimental method.
- The entropy accumulated when this method succeeds.
- Interoperability considerations.
- Security considerations.
- Privacy considerations.

11.3. Initial PVP Method Registry

IANA shall populate the initial PVP Method Registry with the following methods:

Method	RFC
a	RFCXXXX
b	RFCXXXX

[[NOTE TO RFC EDITOR: Please change XXXX to the number assigned to this specification.]]

12. Acknowledgements

Thanks to Patrice Bruno for his comments, suggestions and questions that helped to improve this document.

This document was improved by the input from the VIPR Design Team:

Mary Barnes
Daryl Malas
Hakim Mehmood
Muthu Arul Mozhi Perumal
Jon Peterson
Marc Petit-Huguenin
Michael Procter
John Ward
Dean Willis

This document was written with the xml2rfc tool described in [RFC2629].

13. References

13.1. Normative References

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.

[RFC2434] Narten, T. and H. Alvestrand, "Guidelines for Writing an IANA Considerations Section in RFCs", BCP 26, RFC 2434, October 1998.

[RFC4234] Crocker, D., Ed. and P. Overell, "Augmented BNF for Syntax Specifications: ABNF", RFC 4234, October 2005.

[RFC5054] Taylor, D., Wu, T., Mavrogiannopoulos, N., and T. Perrin, "Using the Secure Remote Password (SRP) Protocol for TLS Authentication", RFC 5054, November 2007.

[VIPR-OVERVIEW] Barnes, M., Jennings, C., Rosenberg, J., and M. Petit-Huguenin, "Verification Involving PSTN Reachability: Requirements and Architecture Overview", draft-jennings-vipr-overview-02 (work in progress), September 2011.

[VIPR-VAP] Jennings, C., Rosenberg, J., and M. Petit-Huguenin,

"Verification Involving PSTN Reachability: The ViPR Access Protocol (VAP)", draft-jennings-vipr-vap-02 (work in progress), March 2012.

[VIPR-SIP-ANTISPAM]
Petit-Huguenin, M., Rosenberg, J., and C. Jennings, "Session Initiation Protocol (SIP) Extensions for Blocking VoIP Spam Using PSTN Validation", draft-petithuguenin-vipr-sip-antispam-03 (work in progress), January 2012.

[VIPR-RELOAD-USAGE]
Petit-Huguenin, M., Rosenberg, J., and C. Jennings, "A Usage of Resource Location and Discovery (RELOAD) for Public Switched Telephone Network (PSTN) Verification", draft-petithuguenin-vipr-reload-usage-04 (work in progress), March 2012.

13.2. Informative References

[RFC2629] Rose, M., "Writing I-Ds and RFCs using XML", RFC 2629, June 1999.

[RFC2945] Wu, T., "The SRP Authentication and Key Exchange System", RFC 2945, September 2000.

[I-D.procter-vipr-privacy-concerns]
Procter, M., "Privacy Concerns relating to the PSTN Validation Protocol (PVP)", draft-procter-vipr-privacy-concerns-00 (work in progress), October 2011.

URIs

[1] <http://www.openbsd.org/papers/bcrypt-paper.ps>

Appendix A. Release notes

This section must be removed before publication as an RFC.

A.1. Modifications between vipr-04 and vipr-03

- Add entropy processing.
- Private methods do not need the reversed domain.
- Updated the BNF to permit experimental and private methods.

- Updated the BNF to permit multiple characters, digits and "-" "." symbols in method names.
- Added section in RFC defining a new method to migration mechanism between experimental and standard methods.
- Added privacy section in RFC defining a new method.

A.2. Modifications between vipr-03 and vipr-02

- Added IANA registration sections.

A.3. Modifications between vipr-02 and vipr-01

- The Calling number is method "a" is now hashed with the bcrypt hash.

A.4. Modifications between vipr-01 and vipr-00

- Added text explaining that VCRs should not be removed before the end of the 48 hours delay.
- Inserted Terminology section.
- Fixed the timekey ABNF.
- Specified that rounding-time cannot be equal to 0.

A.5. Modifications between vipr-00 and dispatch-03

- Moved to new Working Group.

A.6. Modifications between dispatch-03 and dispatch-02

- Nits.
- Shorter I-Ds references.
- Removed sentence saying that Tkey is converted to base64.
- Added ValExchange method and Domain attribute definitions.
- Fixed the last sentence of 7.2 - the ticket goes into the ticket element in the ServiceContent attribute.
- Expanded first usage of VCR initialism.
- Replaced any insteance of peerID by Node-ID.
- Rewrote the ABNF.

Authors' Addresses

Marc Petit-Huguenin
Unaffiliated

Email: petithug@acm.org

Jonathan Rosenberg
jdrosen.net
Monmouth, NJ
US

Email: jdrosen@jdrosen.net
URI: http://www.jdrosen.net

Cullen Jennings
Cisco
170 West Tasman Drive
San Jose, CA 95134
USA

Phone: +1 408 421-9990
Email: fluffy@cisco.com