Internet Engineering Task Force
INTERNET DRAFT
draft-gross-sipaq-01.txt
April, 2001
Expires: October 2001
SIP Working Group

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QoS and AAA Usage with SIP Based IP Communications

Status of this Memo

Abstract

This document specifies the architecture, protocols and messages for SIP based IP communications between Internet domains in support of QoS and AAA. Detailed message flows and message contents are discussed and specified. AAA requirements including inter-domain and user authorization in mobile and non-mobile environments are addressed. A solution is proposed where session setup and teardown are linked with QoS and AAA signaling using AAA policy servers and clearinghouses.

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1. Introduction

Commercial grade IP communications may require voice and other media transport of equal or higher quality than present 3.1 kHz circuit switched voice. Conversely, lower media quality may be acceptable in cases where other service advantages exist, similar to wireless audio/video streaming.

We focus here on those circumstances where voice and other media quality are important and require end-to-end network resources for QoS with larger bandwidth and/or lower delay. Dynamic setup of end-to-end network resources for QoS requires highly scalable mechanisms to authenticate, authorize and account (AAA) for network resources across the Internet. This must be able to happen between a very large number of independent and various access domains that may have no business or trust relationship with each other, or may not even know of each other's existence before initiating end-to-end communications.

The term IP communications is used here for a large class of new communications enabled by IP and the Internet such as presence, instant text, voice or multimedia conferences, the integration of messaging and real time communications, the integration of communications in productivity software, with transactions, games, entertainment and other. Not all IP communication services can be extended to other networks, such as PSTN, ISDN, H.323, BICC or the so-called "softswitch" networks that use MEGACO or similar master-slave signaling protocols.

AAA services are important because QoS and IP to PLMN and PSTN termination through gateways are services that are billable. Unauthorized users should be prevented from using these services and service providers should be able to appropriately bill authorized users.

This document specifies the architecture, protocols and messages for SIP based IP communications between Internet domains in support of QoS and AAA [1]. It builds on information presented in [2] and [3] and is inline with the AAA framework and architecture of [4] and [5]. This document discusses AAA requirements including inter-domain and user authorization in mobile and non-mobile environments. One AAA solution is proposed. The developments in this document assume 2 party call control only. The term originator implies a user who initiates and participates in the IP communication session.

2. Terminology

This section defines terms used throughout this document. Some of

these terms may have other meanings outside the context of this work. Their outside meanings are not guaranteed to coincide with the definitions given here.

APS: AAA and Policy Server. The APS acts as a policy decision point, PDP, for network usage related policies, including IP telephony and QoS requests such as bandwidth reservation. The APS acts on service requests, such as SIP INVITEs for calls requiring QoS and possibly outsources requests to other PDPs, such as a DIR, ADB, or CH. The APS renders a decision and may then apply policy to network elements.

CH: Clearinghouse. The CH authorizes inter-domain calls with QoS and collects usage reports for settlement between service providers.

DIR: Directory. The DIR is the source for end user names and their services. A DIR has the list of all individual users and some attributes that the APS may use as criteria for policy decisions on an individual basis.

ADB: Accounts Database. The ADB is part of the "back office" of the service provider. The ADB contains a list of all corporate accounts and the respective service level specifications that apply at the edges of their networks. It may have classes of policies or customized policies for various corporate accounts.

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 [6].

3. Overview

The components involved in AAA and QoS enabled IP communication sessions are shown in Figure 1. SP represents a SIP Proxy. The clearinghouse (CH) exists outside the access domains and is responsible for providing inter-domain AAA services. In some cases a customer APS may exist outside the access domain. One such case may be where the customer has his or her own corporate domain with a corresponding corporate APS.

Routers internal to the domain are labeled "R" and the edge router is labeled "ER". The media agent, "MA", is the application that sends/receives telephony/multimedia session data. The ER may be responsible for aggregating RSVP flows into appropriate Diffserv classes.

All developments discussed in this document apply using SIP for setting up IP communications. Other signaling means such as H.323, MEGACO or related protocols, may have a different structure in the dependency between signaling, QoS setup and AAA and are therefore not the object of this draft.

The SIP User Agent, SIP UA, acts as client and server on behalf of the user. If an IP phone originated the session, the SIP UA resides on the IP phone. If the signal entering the access domain came directly from a gateway, the SIP UA may reside in the access domain,

as shown in the figure. In any case, the specific details of this configuration are not relevant to the development in this document.

The large box in Figure 1 is labeled as an ISP. In general, it may be an enterprise domain. Additionally, an enterprise domain may outsource SIP services to an ISP. In that case, both domains may be shown in the figure. For simplicity, Figure 1 is shown with a single domain.

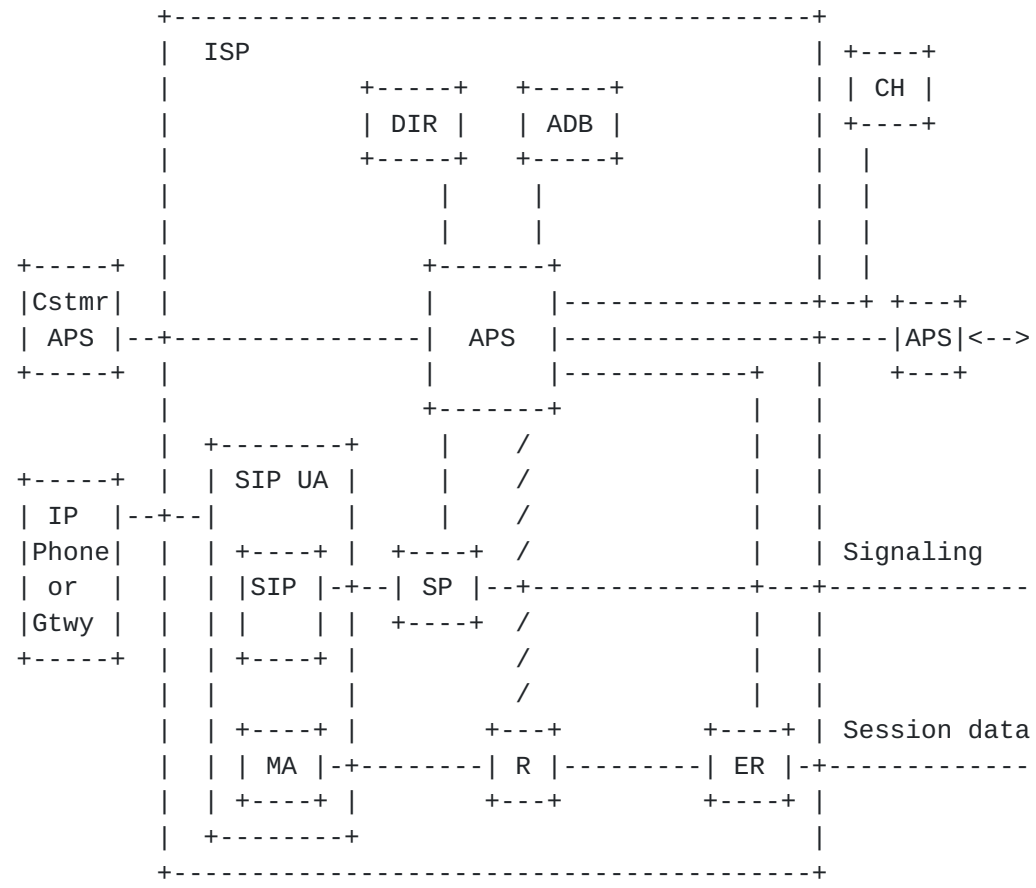


Figure 1: Components to Support Inter-domain AAA QoS Services

The SIP proxy acts as a policy enforcement point, PEP, for IP communication sessions that use SIP. An extension to COPS has been defined that outlines the syntax and semantics of COPS messages and COPS objects for use with SIP [7]. The APS is the corresponding policy decision point, PDP, for SIP sessions. The APS may also interact with the DIR, ADB, customer APS, an APS from other domains and/or one or more clearinghouses for information on which to base a service usage decision. If QoS is requested and granted, the APS may also interact with applicable network device(s) to aid in QoS setup.

Clearinghouses are used to provide scalable, inter-domain AAA

services. One or more clearinghouses may be involved in an IP communications session. Network devices along the session data path use one or more QoS mechanisms to provide a user specified level of service. RSVP [8] may be one such QoS mechanism in access networks and Diffserv [9] may be a QoS mechanism in the transit network.

Internal routers may or may not participate in bandwidth reservation. If it can be assumed that internal routers are not bottlenecks and will always have sufficient resources to handle requests without reservations, message exchanges with the APS may be eliminated. In this case, the slanted line connecting the APS with R in Figure 1 may be ignored.

4. Authorization and Authentication

Proper authorization and authentication may be accomplished in a variety of ways. In pursuit of the most suitable technique, various factors must be considered. These factors include:

1) minimizing call setup time
2) minimizing the time and occurrences that authorization tokens/messages are placed on the wire
3) minimizing implementation complexity
4) optimization of business relationship models (use of clearinghouses as opposed to many bi-lateral agreements).

With these factors in mind, this document discusses some of the requirements of authorization and authentication and suggests one technique as a solution.

Requirements of authorization and authentication include:

1) user authorization
2) inter-domain authorization

In this document, user authorization is taken for granted when the session originator is in his/her home domain. This scenario is discussed in the first sub-section below. When the session originator roams into a foreign domain, or another session participant roams into a foreign domain, user authorization is necessary. Foreign domains are referred to as visited domains in this document. This scenario is considered in the second sub-section below.

While roaming, user registration is assumed to occur before an IP communication session is initiated [10]. During the user registration process, relevant information is exchanged between the roaming user and the visited domain. This information includes the roaming users home domain. The registration process may use the AAA infrastructure described in this document. Further details concerning user registration are out of the scope of this document.

The solution suggested within this document to handle AAA requirements in QoS enabled IP communication sessions includes the

use of one or more authorization tokens and clearinghouses. This solution makes every attempt to address the factors listed previously to provide the most suitable solution.

The methods described in this document use the IP communication session setup protocol, such as SIP, to transport authorization tokens. An alternative approach is to pass authorization tokens between the AAA/Policy servers directly. The second approach may have operational advantages but would require substantially more development before an implementation could be realized. Development along these lines has begun and is discussed in [4] and [5].

Although accounting is not discussed in this section, it is an implied necessity. IP communication sessions must trigger the submission of usage indications or reports after session completion. This facilitates correct accounting.

4.1. Inter-domain

Assume Domain 1, D1, and Domain 2, D2, register with Clearinghouse x (CHx). The necessary security infrastructure is put in place between CHx and D1 and CHx and D2. The security infrastructure must support authentication and authorization services. In some situations it may also be important to support a non-repudiation service to prevent the false denial of IP communication services/resources.

Such an infrastructure can be supplied in the form of a public key infrastructure, PKI, based system or a symmetric key based system such as Kerberos. This infrastructure must be operational before the services of CHx are available. If a PKI based system is used D1 may register with CHx, and in the process exchange public key certificates with CHx. CHx may then use the certificate from D1 to authenticate that any message received from D1 actually came from D1. Analogously, D1 may use the certificate from CHx to authenticate that any message received from CHx actually came from CHx. This can be facilitated by the built in security functionality of OSP [11].

Authentication of the SIP proxy and all end users within a domain is the responsibility of the domain administrator. Maintenance of intra-domain security is required because CHx only authenticates D1 as a whole through the APS. Intra-domain authentication of users requires another level of authentication not discussed in this section. This level of authentication is an especially important consideration for roaming users.

When an end host caller from D1, EH1, calls an end host callee in D2, EH2, the APS in D1, APS1, requests an authorization token from CHx to verify that an appropriate business relationship exists with D2. This development is illustrated in Figure 2. The figure only illustrates messaging that includes the authorization token.

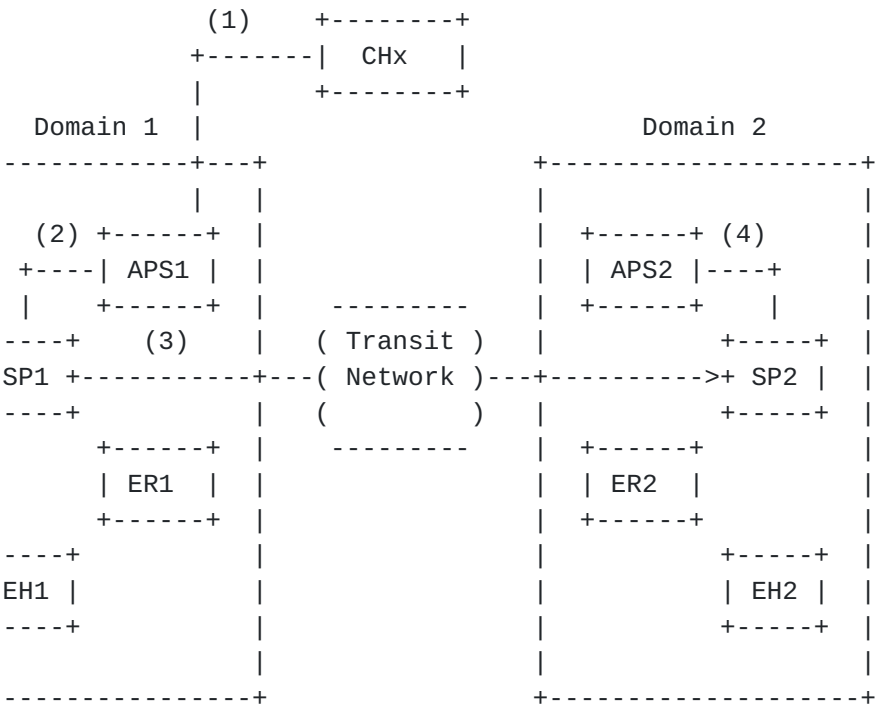
If authorization is granted, CHx replies with an authorization token

to APS1. This token carries sufficient authentication information so that D2 (and possibly CHx) can later verify that it was created by CHx (or possibly another trusted CH) and not by a fraudulent source. This is in addition to the digital signatures contained in the

messages exchanged between CHx and APS1 used to authenticate individual messages.

The APS1 may now authorize the call to the SIP proxy of D1, SP1. APS1 may store the session source, destination, and media information to aid in QoS setup. SP1 embeds the authorization token in the SIP INVITE messages and forwards it to the SIP proxy of D2, SP2.

Upon reception of the INVITE message, SP2 contacts APS2, passing it the authorization token, to request local and inter-domain authorization to complete session setup. Because the authorization token contains self-authentication information, APS2 can authenticate that CHx has actually authorized the incoming call. This removes the requirement for APS2 to contact CHx.



1) APS1 requests an auth token from CH, triggered by session setup request from EH1
2) APS1 returns auth token to SP1 (SP1 embeds token in session signaling message)
3) SP1 signals SP2 with embedded token
4) SP2 sends auth token to APS2 for authorization and authentication
5) auth token no longer used, may be destroyed

Figure 2: Authorization Token Life Cycle

After EH1 and EH2 have been notified that the IP communication session has been authorized, they may now proceed to reserve bandwidth using a mechanism such as RSVP (mapped to Diffserv at the domain egress routers). Bandwidth management services determine if the requested bandwidth is available over the specified path.

The originating and terminating domains have now completed proper inter-domain authorization. If authorization was granted call setup can complete and data may be exchanged over the reserved path. In some models, pre-session establishment of QoS may be a requirement. In such cases, completion of session setup may only transpire if QoS has been established in the required direction(s) [2], [12]. In other models, establishment of QoS and the IP communication session may be decoupled, where each proceeds independently [2].

4.2. Application Level and Pre-call Mobility

Mobility of an end user demands additional infrastructure to authenticate and authorize the user while roaming. Three or more domains may be involved in the AAA and call setup process. The visited domain may be defined as the domain that the user has roamed into. The home domain may be defined as the domain where the user maintains a user profile that contains, among other things, a list of services for that user. The called domain is the domain in which the called party is present. Note that in the general case, this may not be the called party's home domain. That is, the called party may also be roaming. An additional level of authorization and authentication would be necessary in this case. For simplicity, this document addresses the case where the called party is reached at his or her home domain.

In general, authorization and authentication between domains is necessary because billable QoS services may be requested along a path between them. All service providers included in this path who transmit the session data and provide the requested QoS services will expect accounting and/or payment.

However, there is a distinction between the end domains where per call accounting may be required and the transit networks that need not be burdened with the knowledge or understanding of each individual qos flow. For scalability, the transit networks will group all qos flows in some Diffserv class of traffic. The discussion that follows applies only to end domains where calls originate and terminate.

Accounting agreements can be encompassed in a business relationship with a clearinghouse. An authorization token can be used to verify the business relationship between the visited domain and the clearinghouse and the called domain and the clearinghouse.

A business relationship may also include roaming allowances and services. This implies that a user who roams into a foreign domain may proceed to use the IP communication services that he/she pays

for and normally receives from his/her home domain. An authorization token can be used between the visited and home domains to verify such a business relationship with a clearinghouse. Accounting, and ultimately charging, is facilitated with the use of authorization tokens to verify business agreements.

In most cases, SIP signaling sessions which involve mobile SIP terminals will be directed through the home domain of the mobile user in order to maintain home control. Home control is important for the execution of signaled services because the user then sees the same services irrespective of the network capabilities of the visited domain he/she has roamed into. These services are contained in a user profile at the user's home domain.

The authorization tokens obtained from one or more clearinghouses may be embedded in the SIP signaling during session establishment. The order in which tokens are obtained is not clear at this time. Also, the domain(s) that contact the clearinghouse(s) for authorization token requests is not clear at this time. Two possible scenarios are illustrated in Figures 3 and 4. We note that in each, options exist concerning clearinghouse contacts and messaging order.

In Figure 3, session signaling (SIP INVITE message) arrives at the SIP proxy of the visited domain (1). The SP, acting as a SIP PEP, makes a request to the APS, acting as a SIP PDP (2). The APS of the visited domain contacts one or more clearinghouses seeking appropriate business relationships with the home and called domains (3, 4). If all three domains are registered with the same clearinghouse, only that clearinghouse need be contacted. The clearinghouses return one or more authorization tokens to the APS, which passes them back to the SP. The SP embeds the tokens within the SIP INVITE message and forwards it to the home domain of the caller (5).

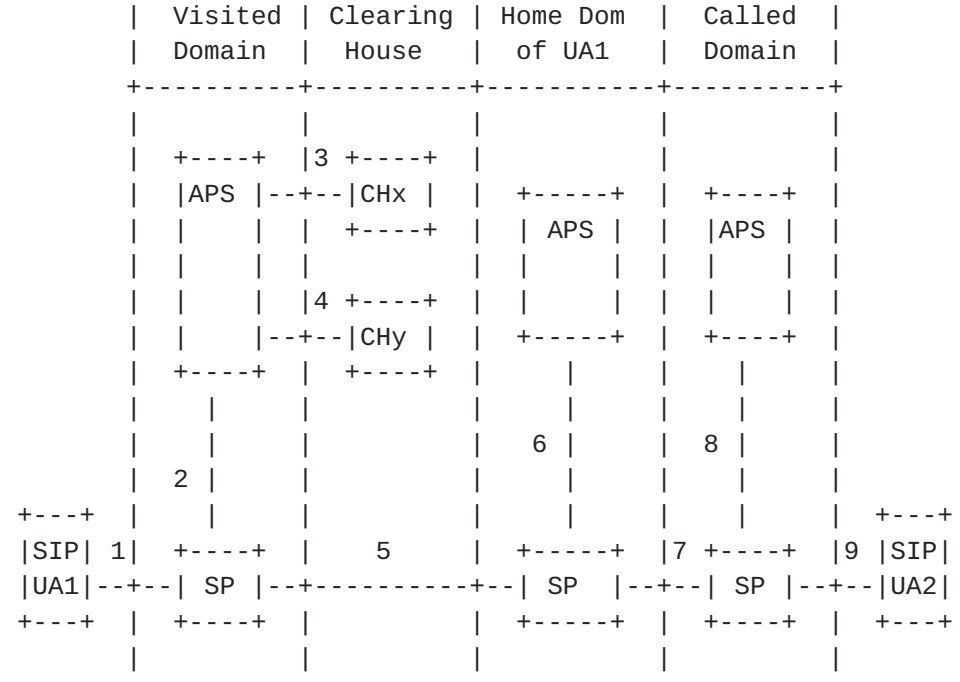


Figure 3: Scenario 1 - Session Setup with AAA Support

The SP in the home domain contacts the APS using a PEP/PDP relationship (6). The SP or APS checks the user profile to verify

that the requested services are allowed. The APS verifies the applicable authorization token and may make additional policy-based admission decisions. The decision is passed back to the SP. The INVITE message is then forwarded to the SP of the called domain (7). The SP contacts the APS using a PEP/PDP relationship (8). The APS verifies the applicable authorization token and may make additional policy-based admission decisions. The decision is passed back to the SP. The SP finally forwards the INVITE message to the called user, SIP UA2 (9).

In Figure 4, the messaging is different only in one way. The APS of the visited domain contacts the APS of the home domain before any authorization tokens are obtained (3). The APS of the home domain checks the user's profile to authorize the use of services. If the home domain authorizes the session, the APS in the visited domain receives a success reply and proceeds to obtain necessary authorization tokens as described for Figure 3.

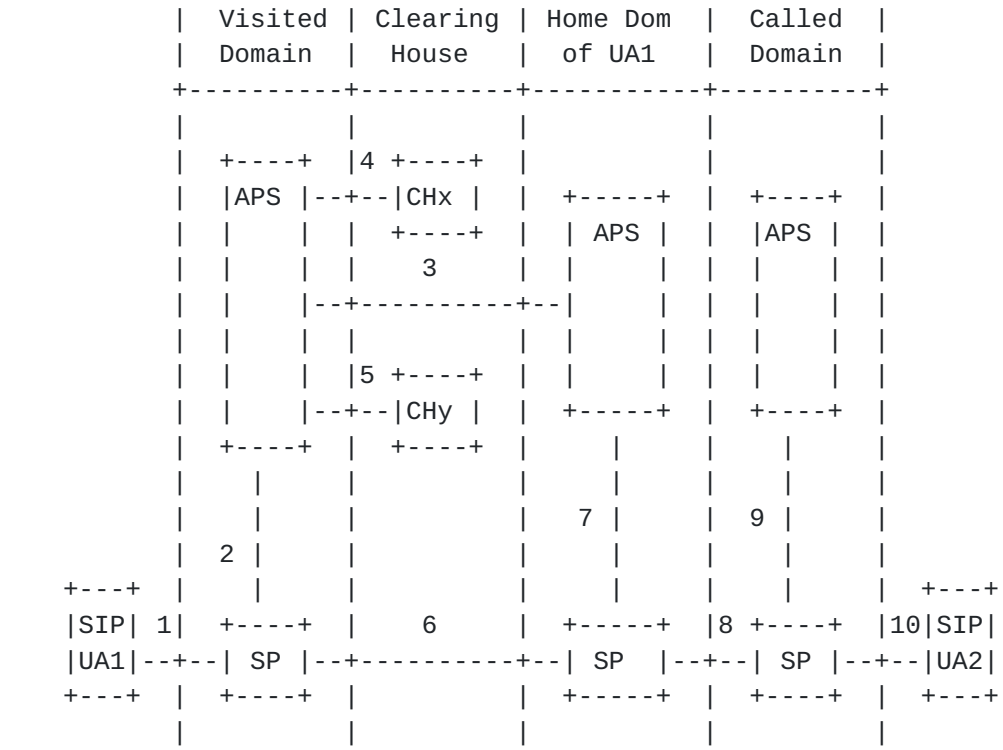


Figure 4: Scenario 2 - Session Setup with AAA Support

The advantage of the scenario in Figure 4 is that the user's profile is checked before any authorization tokens are obtained. In this way, valuable airtime in the wireless case would not be wasted transporting authorization tokens if the user's profile did not allow the session to proceed. A disadvantage is that call setup time may be lengthened somewhat due to an extra inter-domain message

exchange. Another disadvantage may be that a messaging protocol between APSs would have to be defined. COPS is one protocol that may be used for this interaction [13].

The authorization tokens ensure that authentic, billable parties exist and that account, billing and charge settlement may be achieved in a previously agreed upon manner. To achieve scalability, business relationships with a clearinghouse can be used to remove the necessity for bi-lateral business agreements.

5. Message Contents

This section defines the data content of messages passed between the SIP proxies, APS, and clearinghouse to provide QoS and AAA linkage with SIP for inter-domain IP communication sessions. The messaging discussed does not consider mobility. It considers an originating domain and a terminating domain, each containing a SIP UA, a SIP proxy, and an APS, similar to the illustration in Figure 2. If inter-domain authorization and gateway location services are not required, interaction with a CH and inclusion of an authorization token in all exchanges listed in this section can be ignored.

A companion document defines a COPS extension for SIP that details the messaging and message contents between the SP and APS [7]. The current document merely discusses this messaging in general terms. Please refer to the referenced document for full details.

5.1. SIP Proxy to APS (Originating Domain)

When a SIP proxy receives an INVITE, it sends source, destination, and possibly media information to the APS. The source and destination information may be used by the APS for intra-domain policy-based admission authorization. Some or all of the information must be sent to the CH for inter-domain authorization.

The SIP call identifier (callID) can be used by the APS for intra-domain accounting. The callID must be sent to the CH in the authorization request message. The CH may use this identifier when generating an authorization token.

If the INVITE contains a SDP attribute specifying bandwidth reservation [12], the APS may need to know the amount of bandwidth requested. The bandwidth is implied by the media description [14], [15]. Bandwidth information is necessary if the APS is charged with the responsibility of managing bandwidth allocation based on user and application information. For example, the APS may have a policy that allows user X to reserve 300 Kbps for multimedia sessions with users A through E and only 50 Kbps multimedia sessions with other users.

5.2. APS to CH (from Originating Domain)

The information required by the CH for accurate AAA support may be

CH dependent. It is expected that all CHs require knowledge of the source and destination domains for basic AAA purposes. The call identifier may be useful for authorization token generation. Bandwidth reservation amount may be required by a CH dependent on

the specific business agreement with the domains. This quantity may be used as a basis for granting or denying authorization. If bandwidth reservation amount is required, it must be put into a form agreed upon by the parties involved, such as bits per second.

5.3. CH to APS (to Originating Domain)

The clearinghouse has the responsibility of making an authorization decision concerning IP communication between the specified source and destination domains. If gateway services are also requested, the decision may also be based upon available gateways and subsequent QoS or cost characteristics corresponding with that gateway. For example, the business relationship of the end domains with the clearinghouse may exclude IP communication sessions that incur costs above a specific limit. Costs may result from long distance charges over the PSTN or from bandwidth reservation over the IP network. The location of a gateway will likely affect one or both of these costs.

The CH must return to the APS an authorization decision. If the request is granted, the CH must also return to the APS an authorization token. The authorization token is expected to contain authentication information so that both end point domains recognize it as coming from a trusted CH, as described in a previous section. A unique transaction identifier may be returned by the CH and subsequently used to correlate original authorization requests with usage indication reports generated after the session has completed.

5.4. APS to SIP Proxy (Originating Domain)

The APS finally responds to the SIP proxy INVITE message with a status, indicating if the user is allowed to send the INVITE further along the SIP signaling chain. If the authorization is granted, the APS passes the authorization token it received from the CH to the SIP proxy. The SIP proxy includes this token inside the authorization token header field of the INVITE message [16].

5.5. SIP Proxy to SIP Proxy (Originating to Terminating Domain)

The SIP proxy forwards the INVITE message containing the authorization token from the originating domain to the terminating domain in the manner defined by SIP.

5.6. SIP Proxy to APS (Terminating Domain)

The APS in the terminating domain requires the same information as the APS in the originating domain on which to base an intra-domain authorization decision. Additionally, if the APS receives an authorization token, it performs an authentication check on it using the security infrastructure previously established with the CH. This

action verifies that the authorization token is authentic and that the requested service has been authorized by a trusted source.

6. Protocol Support

The exchange of information between the APS and the SIP proxy may be handled using COPS. As mentioned previously, a new COPS extension for SIP is being developed concurrently with this work [7]. Exchanges between the APS and the CH can be handled using OSP [11]. OSP defines all messaging and message data field specifications that are outlined in this document. No additional protocol work is required for this exchange. Also, OSP has been embraced and is in use by a number of large industry participants.

7. Messaging Overview

This section details the message exchange and message contents to initiate and terminate a SIP-based IP multimedia session. Mobility is not considered in this section. The caller and callee are both present and active at their respective home domains.

It is assumed that the media will be transmitted and received from both ends and that bandwidth reservation requests are made in both directions using the SDP attributes discussed in [12]. In the first subsection the inter-domain authorization occurs at the originating domain. In the second subsection the inter-domain authorization occurs at the terminating domain.

Descriptions of SIP messages, including requests and responses, are not discussed in this document. They are discussed in detail in [1]. Descriptions of RSVP messages are not discussed in this document but are discussed in detail in [8]. It is assumed that edge routers are RSVP enabled PEPs. Furthermore, it is assumed that the edge routers interact with the APS as PDPs of their respective domains for policy-based management decisions concerning RSVP. Message exchanges of this nature are assumed implicit in this section and are not included in the messaging overview. Details concerning this message exchange are in [17].

As discussed in [2], two QoS models may be defined for IP communication sessions. The QoS assured model is one in which the session and QoS signaling are coupled. In this case the session does not proceed until/unless QoS has been successfully established. The QoS enabled model decouples session and QoS signaling. In this case, the session setup may complete and session data may be exchanged even if QoS has not yet been, or never gets, established. The messaging discussed in this document only considers the QoS enabled model. The development in this section may be extended to work with the QoS enabled model with the inclusion of messages defined in [12] and [18].

Since QoS assured is assumed in the session flow examples, the RSVP

and SIP messaging are decoupled and may occur asynchronously. The SIP, COPS and OSP messages outlined in the call flow examples may therefore not necessarily be in chronological order with the RSVP messages.

In the following call flows, the APS of the respective domain may be contacted for a policy based decision whenever new session data arrives. The SIP protocol defines whether session data may be transmitted in INVITE, ACK and/or response messages. For simplicity, the call flows in this document assume that session data is transmitted in INVITE and 200 OK responses.

All components involved in message passing are illustrated in Figure 1. Those in the originating domain are subscripted with "o" and components in the terminating domain are subscripted with "t". The protocol used for each exchange is indicated at each step. Each step also contains notes concerning the contents or special attributes of the message.

7.1. Originating Domain Contacts Clearinghouse

The following discussion refers to Figures 5 and 6. The step numbers of Figure 5 are indicated as message labels in Figure 6. In step 1, the caller picks up the phone, requests a bandwidth reserved connection and dials a number. This causes the UAo to include appropriate SDP attributes in the SIP INVITE message indicating that QoS is being requested in both directions. The necessary information is passed from the SPo to APSo in step 2. APSo checks if local policy allows the call. This may include interaction with an external policy database, a DIR, an ADB, and/or another APS.

If local policy authorizes the call, APSo sends the CH an authorization request for inter-domain authorization in step 3. The CH considers the source and destination information to determine if appropriate business relationships exist with the originating and terminating domains. Bandwidth and/or media information may also be used on which to base a decision. If authorization is granted, the CH generates an authorization token using the call identifier as well as other data. The CH sends the token to the APSo in step 4 along with a status value indicating if authorization has been granted or not. The APSo passes this information on to the SPo in step 5.

The SPo forwards the INVITE with the embedded authorization token down the SIP signaling chain in step 6 to SPt in the terminating domain. The SPt sends the necessary information to APSt in step 7, which includes the authorization token. The APSt requests local authorization in a similar fashion as that described for the originating domain. The APSt then uses the authorization token and other message data to authenticate the token and determine if inter-domain authorization has been granted. In step 8 the APSt passes a status to the SPt indicating if the session has been authorized. If the call and all requested services are granted, the SPt forwards

the INVITE on to the UAt in step 9.

The UAt may then send the RSVP Path message to UAo in step 10 since it knows the source and destination addresses and ports and the traffic specifications. Note that in some cases, the SDP sent from

UAt to UAo may contain multiple media formats, each having different bandwidth requirements. In this case, the UAo must make a final decision considering the choice(s) of media formats. If the traffic specifications change, additional RSVP messages may be sent to update the network devices along the session data path.

In step 11 the UAt sends back a 180 (ringing) SIP response to UAo via SPo and SPt. In some cases, the UAt may send a 183 (session progress) SIP response followed by a 180 response. The 183 response would follow the same path as the 180 response. For simplicity, only the 180 response is shown in the figures.

```
1)   UAo   -->   SPo    (SIP)  INVITE; (SDP "a=qos:")
2)   SPo   -->   APSo   (COPS) src, dest, media, callID
3)   APSo  -->   CH     (OSP)  src, dest, callID
4)   CH    -->   APSo   (OSP)  status, auth token
5)   APSo  -->   SPo    (COPS) status, auth token
6)   SPo   -->   SPt    (SIP)  INVITE, (SDP "a=qos:"), auth token
7)   SPt   -->   APSt   (COPS) src, dest, media, callID, auth
                                  token
8)   APSt  -->   SPt    (COPS) status
9)   SPt   -->   UAt    (SIP)  INVITE
10)  UAt   -->   R/ER   (RSVP) Path
11)  UAt   -->   UAo    (SIP)  180
12)  UAo   -->   R/ER   (RSVP) Path
13)  UAt   -->   R/ER   (RSVP) Resv
14)  UAo   -->   R/ER   (RSVP) Resv
15)  UAt   -->   SPo    (SIP)  200 OK via SPt (SDP (a=qos:))
16)  SPo   -->   APSo   (COPS) src, dest, media
17)  APSo  -->   SPo    (COPS) status
18)  SPo   -->   UAo    (SIP)  200 OK (SDP (a=qos:))
19)  UAo   -->   UAt    (SIP)  ACK

20)  UAo   <->   UAt    (RTP)  session data exchanged

21)  UAo   -->   SPo    (SIP)  BYE
22)  SPo   -->   APSo   (COPS) RPT, DRQ
23)  SPo   -->   SPt    (SIP)  BYE
24)  SPt   -->   APSt   (COPS) RPT, DRQ
25)  SPt   -->   UAt    (SIP)  BYE
26)  UAo   -->   ERo    (RSVP) PathTear and ResvTear
27)  UAt   -->   ERt    (RSVP) PathTear and ResvTear
28)  APSo  -->   CH     (OSP)  usage report - session duration,
                                  bandwidth consumed, etc.
29)  APSt  -->   CH     (OSP)  usage report - session duration,
                                  bandwidth consumed, etc.
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Figure 5: Message Flow Details

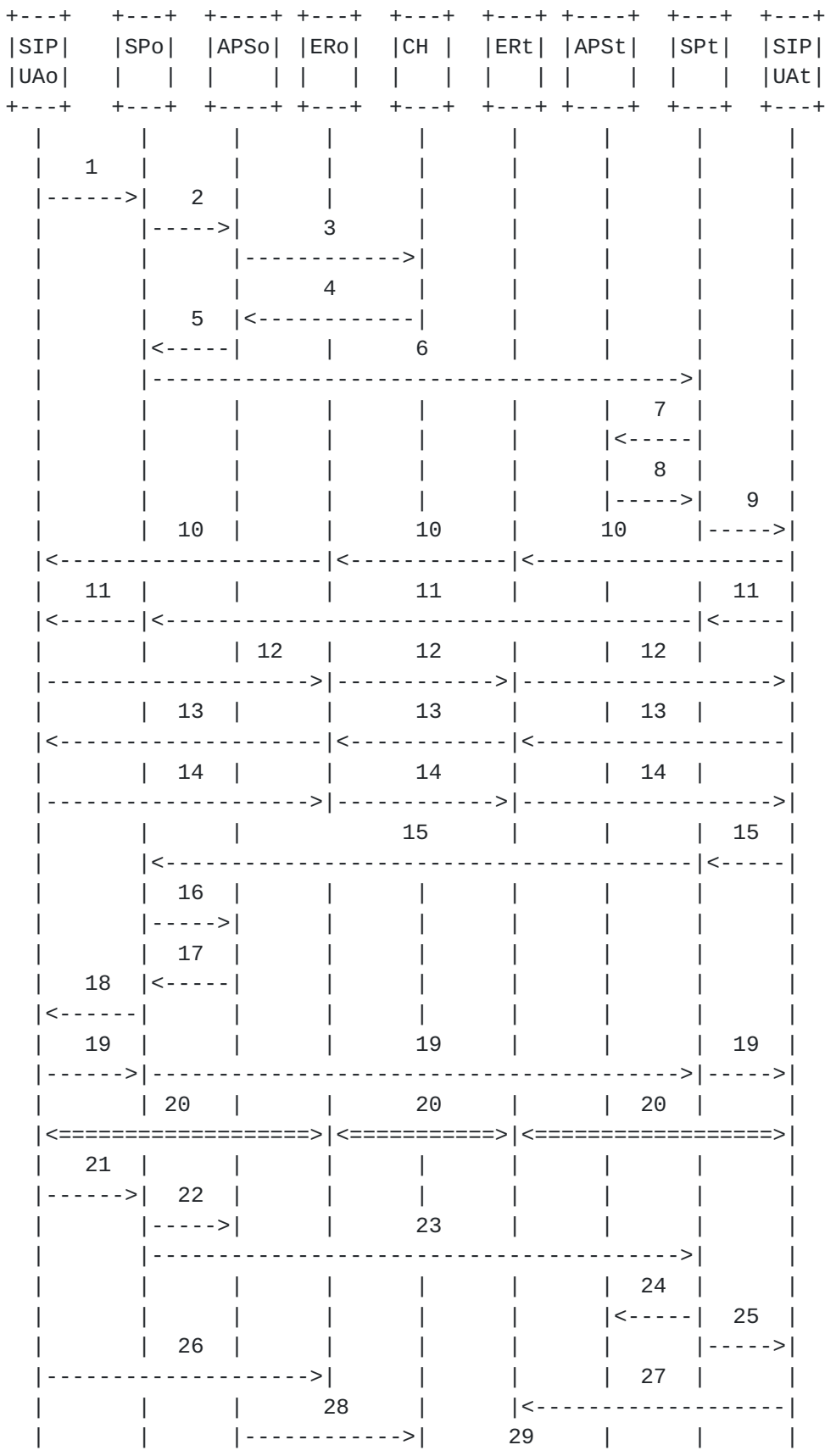
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Figure 6: Message Flow

UAo may now send a RSVP Path message to UAt in step 12 since it knows the source and destination addresses and ports and the traffic specifications. When RSVP Path messages arrive at UAo and UAt, they may send a RSVP Resv message as shown in steps 13 and 14.

In step 15, UAt sends a final 200 (OK) response to SPo via SPt. If the source, destination or media information are different from what was specified previously, SPo sends a message to APSo in step 16. APSo checks local policy and returns a status to SPo in step 17 indicating if the session is still allowed. If so, SPo sends the 200 response to UAo in step 18.

The UAo confirms reception of the 200 response and establishment of the session by sending a SIP ACK message to SPo in step 19. The session data is now exchanged via RTP, or some similar protocol, in step 20. The session ends for UAo when, for example, a SIP BYE message is sent to SPo in step 21. SPo sends a COPS report and deletes the COPS request state in APSo in step 22. The BYE message is simultaneously forwarded to SPt in step 23. SPt sends a COPS report and deletes the COPS request state in APSt in step 24. The BYE message is sent to UAt in step 25.

PathTear and ResvTear messages are sent from UAo and UAt upon call completion (from UAt when it receives the BYE message) in steps 26 and 27. This triggers a COPS DRQ from ERo and ERt to APSo and APSt, respectively. These messages are not shown but implicitly implied. APSo and APSt may then perform necessary accounting/bookkeeping chores. Alternately, ResvErr messages may be sent by an APS under cancellation circumstances by APSo or APSt.

The APSo and APSt send usage reports to the CH in steps 28 and 29 to indicate session duration, QoS services consumed such as bandwidth, and possibly other session details necessary for inter-domain accounting and settlement.

Other circumstances may cause the reservation in one or both directions to fail. In this case, ResvErr messages may be injected into the session data path. Upon reception of a PathErr or ResvErr message, an ERo/t would remove the corresponding state and notify APSo/t.

7.2. Terminating Domain Contacts Clearinghouse

The caller may have a reason to allow the terminating domain to contact the CH for inter-domain authorization. For example, the caller may require that the callee make an unusually large bandwidth reservation for a critical IP communication exchange. The caller does not know the limits of services the callee is entitled to. In

this case the caller may forego inter-domain authorization during call setup. Instead, the APSt, on behalf of the callee, receives the INVITE without a token, negotiates for an allowable level of service, then contacts the CH for an authorization request. The resulting authorization token is then placed inside the 200 response

sent back to the caller. Such a scenario may only make sense if the CH requires bandwidth information (or other QOS service information such as latency or jitter) to make an authorization decision.

The benefits of this are debatable. One advantage is that if successful negotiation is not possible, the call may be cancelled without contacting the CH, saving a message exchange. The message exchange scenario is similar to that of Figure 5 with minor changes.

8. Accounting, Charging and Billing

We make a clear distinction between accounting, charging and billing and provide a summary, possibly incomplete description.

Accounting: The process of logging the usage of network resources, such as QoS or PSTN gateway services. The accounting data may or may not be used for usage charging, depending on the service model. This draft addresses accounting only and does not address charging and billing.

Charging: Allocating the cost to various parties according to some business policies. Cost allocation may depend on such parameters as class of service, geographic locations, time of day, promotions, incentives for resellers and others. Examples of different business policies and classes of service are usage based charging and flat rate subscription rates.

Billing: Informing various users of the charges and expected payments.

9. Security Considerations

This document addresses some security issues concerning authentication of inter-domain business relationships. The protocols discussed in this work, SIP, COPS and OSP contain their own security mechanisms. Any security issues not addressed by SIP, COPS or OSP have not been considered in this work and are left as open issues.

10. Acknowledgments

The authors would like to thank Russ Fenger, Arun Raghunath, Changwen Liu, Mark Grosen, Jeff Mark, Kalon Kelley and Dave Durham for insightful discussions and valuable contributions.

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