Architectures and Supporting Protocols for VOIP/3G

IETF at work

NGN and 3G Network Elements

Numbering and Naming (ENUM, TRIP)

Session Description Protocol (SDP)

NAT traversal

Diameter

Media Gateway Control (Megaco/MGCP)

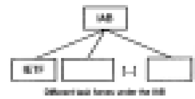
Common Open Policy Service (COPS)

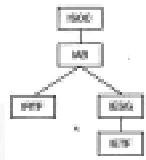
Agenda

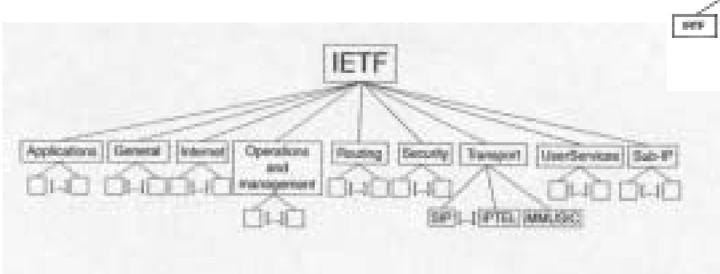
- IETF
- Networking framework – 3G, wireline
- 3G terminal
- ENUM – naming and addressing

IETF

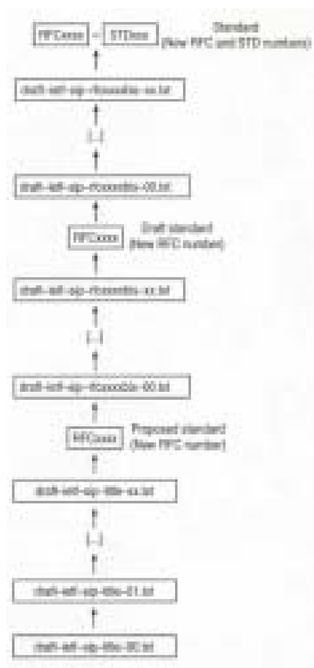
- IETF toolkit
  - bottom-up approach *("one problem – one protocol")*
  - Protocols should be simple, reusable, scalable, robust







IETF specifications

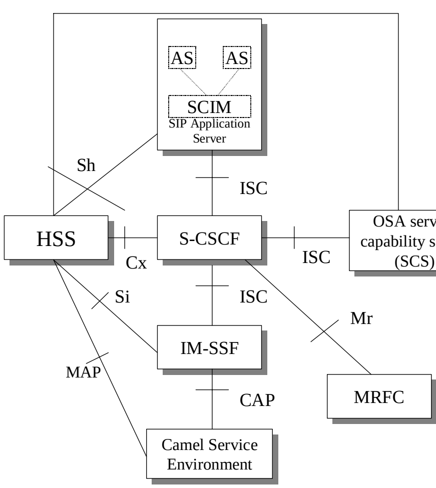


•Every standard follows the route Proposed standard-> Draft Standard-> Standard

ETSI, etc have delegated the 3G standardisation work to 3GPP

- 3GPP – is the 3G Partnership Project
- this gives a key role to vendors
- site: www.3gpp.org has all their documents!
- The idea is that ETSI etc will rubberstamp 3G documents as standards.

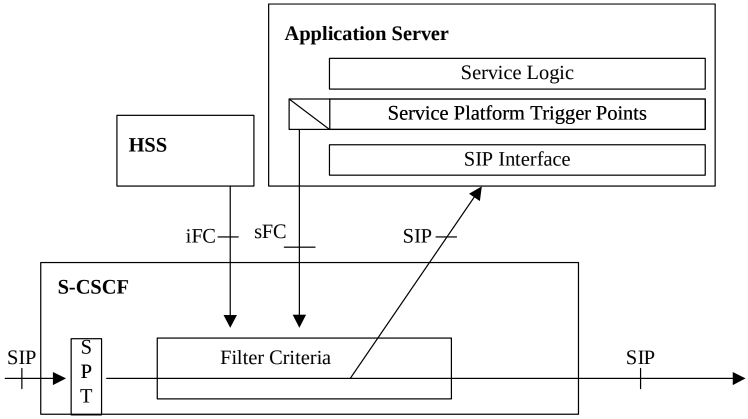
3G IP Multimedia core network Subsystems (3G IMS)



AS – Application Server
CAP - Camel Application Part
IM-SSF – IP Multimedia Service Switching Function
ISC – IP Multimedia Service Control

MAP - Mobile Application Part
MRFC - Media Resource Function Controller
OSA – Open Service Access
SCIM – Service Capability Interaction Manager
SCS – Service Capability Server

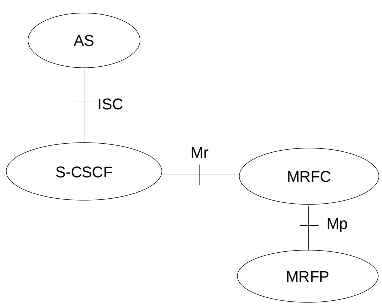
3G Application Triggering



iFC – Initial Filter Criteria
sFC – Subsequent Filter Criteria
SPT – Service Point Trigger

Service processing can be delegated to Application Servers with a fine grained control

Media processing in 3G



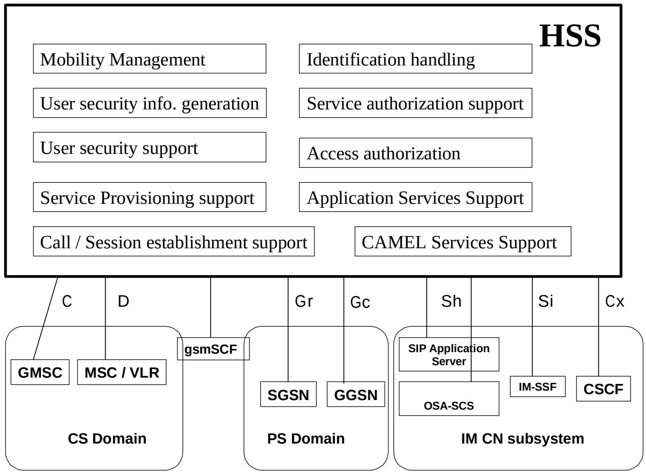
MRFC - Media Resource Function Controller
MRFP – Media Resource Function Processor

MRFC likely to have a general purpose processor,
MRFP has many DSPs – digital signal processors.

All this takes place in the IP domain.
Examples:
- transcoding Wideband AMR/ Narrowband AMR codec
- Multiparty conference media processing

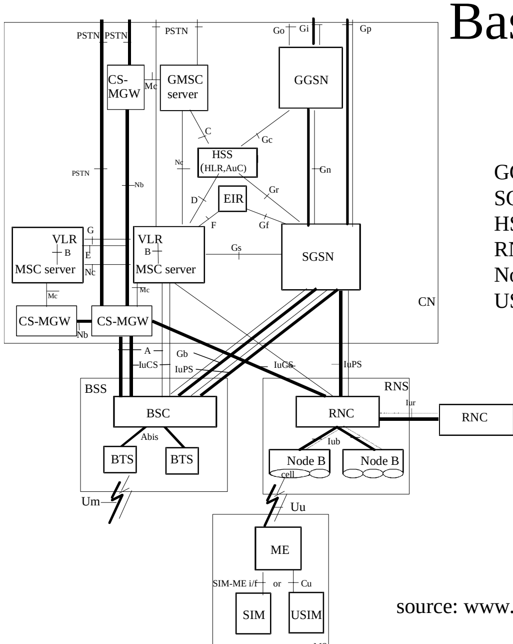
In practice it is convenient to implement MRFP in the same device as the Media Gateway between CS/PS domains

The role of HSS



source: www.3gpp.org/specs/archive/23002-580

Basic Configuration of a PLMN



GGSN – Gateway GPRS Support Node
SGSN – Serving GPRS Support Node
HSS – Home Subscriber Server
RNC – Radio Network Controller
Node B = 3G base station
USIM – UMTS Subscriber Identity Module

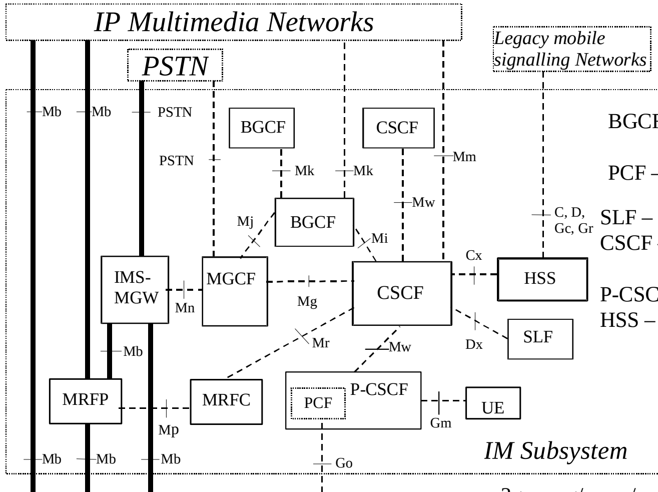
On CS side breakdown of MSC to Media Gateway and MSC server.

3G and GSM/GPRS are based on the same packet core elements.

source: www.3gpp.org/specs/archive/23002-580

The IP Multimedia Subsystem

sits on top of the Packet Core

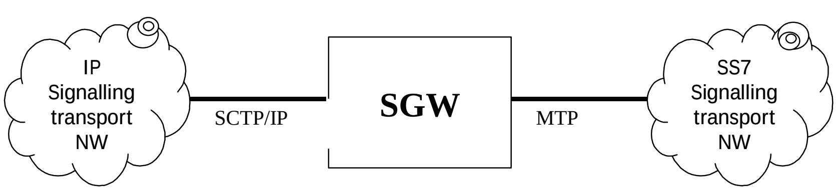


BGCF – Breakout Gateway Control Function
PCF – Policy Control Function

SLF – Service Location Function
CSCF – Call Session Control Function
P-CSCF – Proxy CSCF
HSS – Home Subscriber Server

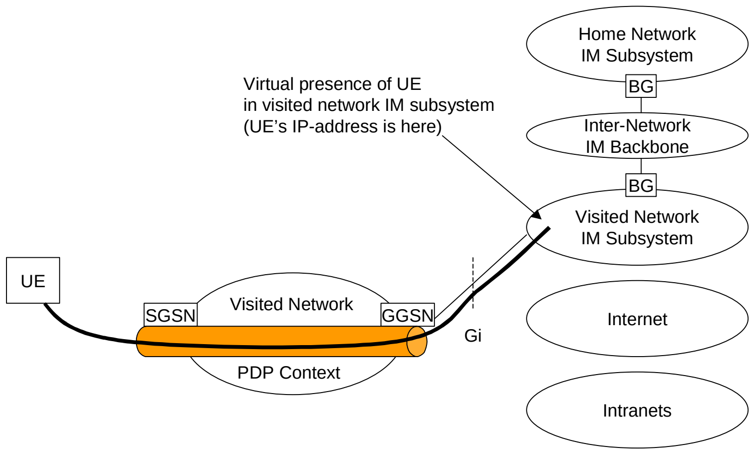
source: www.3gpp.org/specs/archive/23002-580

Signaling Gateway maps SS7 MTP to SCTP/IP transport

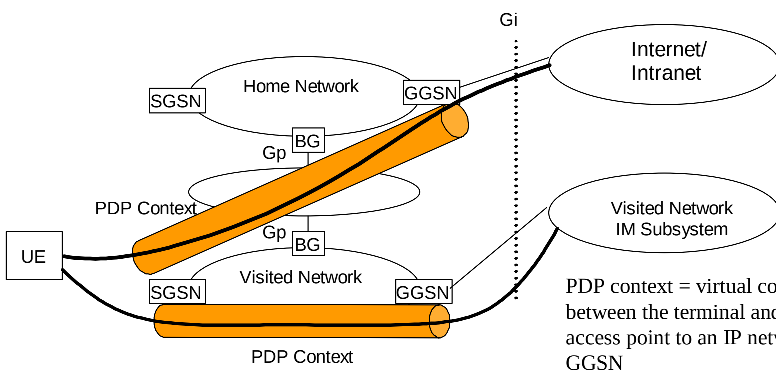


This allows to transfer signaling and service processing responsibility to IP based environment.

UE has a tunnel to visited IMS



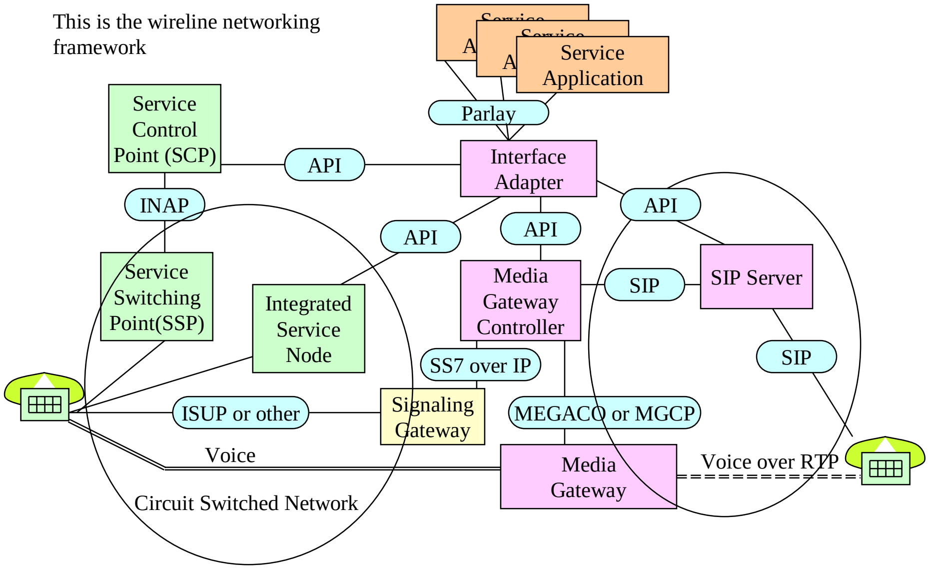
3G UE can use several services at the same time



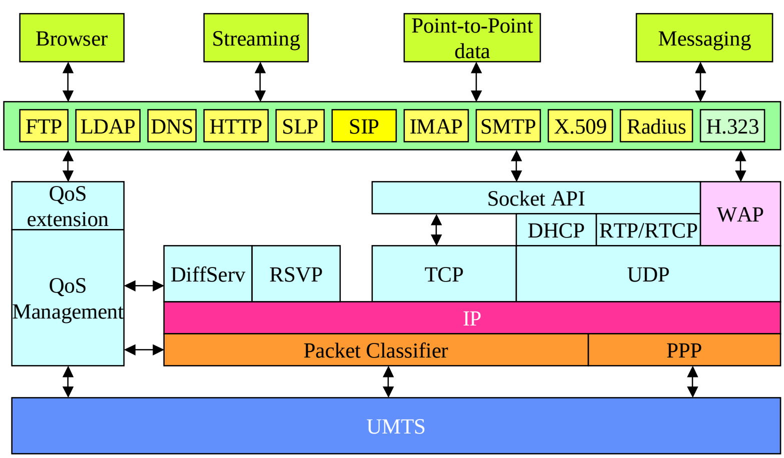
PDP context = virtual connection between the terminal and an access point to an IP network thru GGSN

ETSI SoftSwitch Architecture for NGN

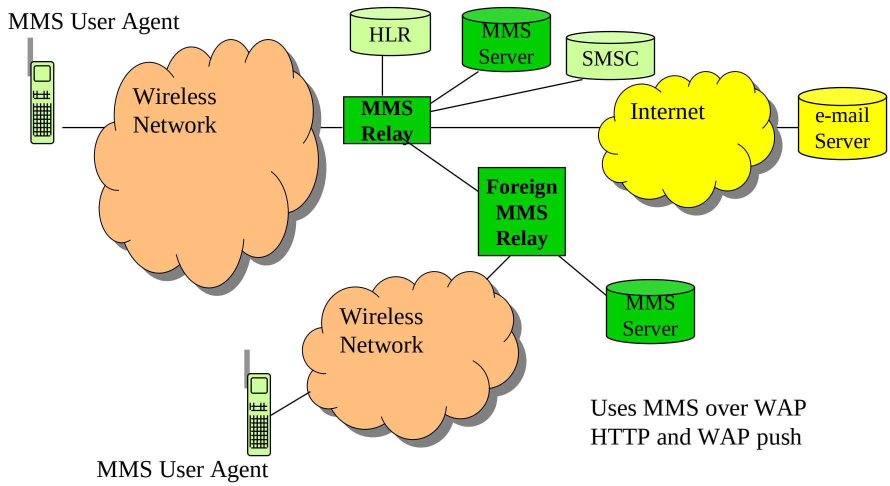
This is the wireline networking framework



The UMTS terminal functional model



The GPRS and 3G networks implement the Multimedia Messaging Service



Uses MMS over WAP
HTTP and WAP push

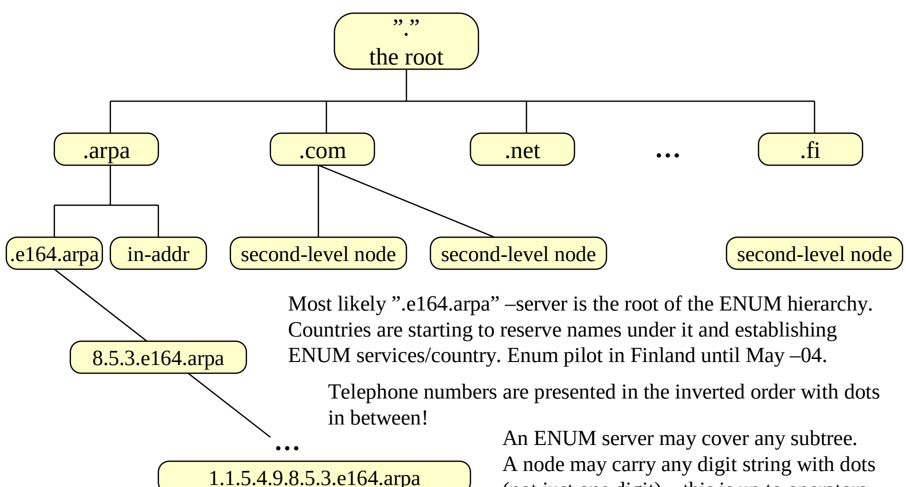
Supporting protocols for IP telephony – wired and wireless

- ENUM – addressing and naming
- Gateway location – TRIP
- Gateway control - Megaco
- Policy Control – COPS
- Session description – SDP
- AAA - Diameter

Naming and Addressing in NGN and 3G IMS vs. Telephone numbering

- A **Name identifies** a domain, a user or a service. An **address points to** a user or to an interface or to an inlet/outlet in a network.
- Internet heavily relies on the Domain Name System (DNS) to translate names to addresses. The specs of using DNS for Telephony names and addresses is called ENUM – tElephone-NUmber-Mapping.
- ENUM was originally meant for mapping IP telehone numbers (e.g. 3G IMS phonenumbers) to logical names (and IP addresses).
- With Naming and Addressing, at the same time we need to solve the problem of Gateway (CSN/IP) location and Number Portability across the technology boundary.

ENUM uses DNS to store telephone numbers



Most likely ”.e164.arpa” –server is the root of the ENUM hierarchy. Countries are starting to reserve names under it and establishing ENUM services/country. Enum pilot in Finland until May –04.

Telephone numbers are presented in the inverted order with dots in between!

An ENUM server may cover any subtree.
A node may carry any digit string with dots (not just one digit) – this is up to operators.

ENUM introduces NAPTR records

RFC 2915 - The Naming Authority Pointer (NAPTR) DNS Resource Record (Sep 2000)

NAPTR – Naming Authority PoinTeR = Record in DNS containing an URI.

E.g. IN NAPTR 10 10 ”u” ”sip+E2U” ”!^.*$!sip:raimo.kantola@sip.elisa.com!”.

Internet

Record type

Order and Pref

expect uri

SIP, E.164 to URI

The URI itself!
Can be modified by Regular expression

NAPTR format is: Domain TTL Class Type Order Preference Flags Service Regexp Replacement

Domain=first well known key e.g. <something>.uri.arpa
TTL=Time-To-Live – validity time of the record (time to cache)
Class=IN=Internet
Type=NAPTR=35
Order=low nrs are processed before high, once target found, stop (excepting flags)
Pref=if same order value, all with diff pref can be processed, take lowest first.
Flags=“S”-next lookup for SRV record, “A”-next lookup for A, AAAA or A6 record, “U” – the reminder has an URI+this is the last record, P –protocol specific processing
Service=protocol-name + resolver, resolver is used to resolve the result of regexp
Regexp=replacement-rule for whatever querier is holding.
Replacement=a fully qualified domain name to query next for NAPTR, SRV or address records (“S”, “A”)

Example from RFC 2915

In order to convert the phone number to a domain name for the first iteration all characters other than digits are removed from the telephone number, the entire number is inverted, periods are put between each digit and the string ".e164.arpa" is put on the left-hand side. For example, the E.164 phone number "+1-770-555-1212" converted to a domain-name it would be "2.1.2.1.5.5.5.0.7.7.1.e164.arpa."

For this example telephone number we might get back the following NAPTR records:

$ORIGIN 2.1.2.1.5.5.5.0.7.7.1.e164.arpa.
IN NAPTR 100 10 "u" "sip+E2U" "!^.*$!sip:information@tele2.se!" .
IN NAPTR 102 10 "u" "mailto+E2U" "!^.*$!mailto:information@tele2.se!" .

This application uses the same 'u' flag as the URI Resolution application. This flag states that the Rule is terminal and that the output is a URI which contains the information needed to contact that telephone service. ENUM uses the Service field by defining the 'E2U' service. The example above states that the available protocols used to access that telephone's service are either the Session Initiation Protocol or SMTP mail.

A possible ENUM hierarchy

This follows the ”US model” suggested by Tuomo Rostela for Finland.

Tier 0

$ORIGIN e164.arpa.
1 IN NS att_enum.com.
6.4 IN NS sweden_enum.se.
8.5.3 IN NS ficora_enum.fi.

358 is delegated to ficora_enum

Tier 1

ficora_enum.fi
8.5.3.e164.arpa

$ORIGIN 4.9.8.5.3.e164.arpa.
5 IN NS enum.elisa.fi.
6 IN NS enum.elisa.fi.

Elisa is chosen as the ENUM operator for HUT numbers 09-45….,

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Tier 2

enum.elisa.fi

$ORIGIN 1.7.4.2.1.5.4.9.8.5.3.e164.arpa.
IN NAPTR 10 10 ”u” ”sip+E2U” ”!^.*$sip:raimo.kantola@sip.netlab.hut.fi!”.

My office phone number is mapped to a (non-existing at the moment) SIP server operated by the NETLAB

Tier 3

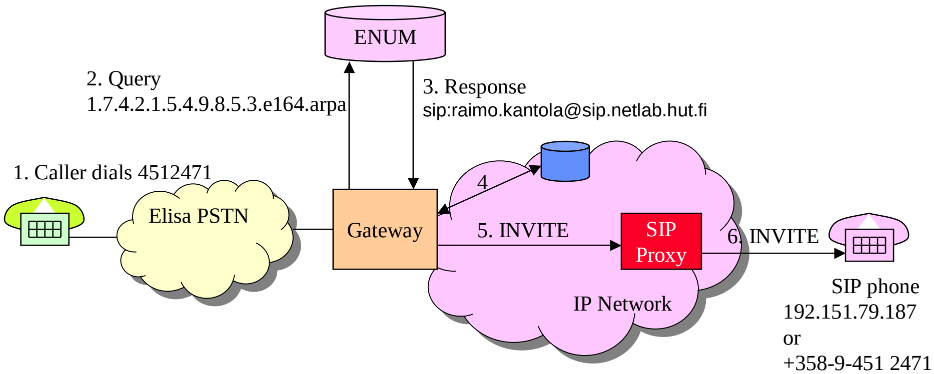
Corporate numbering schemas…

In Finnish ENUM pilot until 31.5.2004 only Tier 1 and Tier 2 present!

ENUM use and future

- Since DNS is used by everybody, ENUM is a likely surviver, policy routing etc additions may emerge
- Due to Number Portability Provision of ENUM service and provision of VOIP service to end-customers are two independent services.
- User may need to select the Numbering service provider separately from the VOIP service provider.

Call from PSTN to a SIP phone



4 - sip:raimo.kantola@sip.netlab.hut.fi is translated to an IP address of the SIP proxy serving the number by another DNS query that returns an address record.

ENUM issues and problems

- Long chain of DNS servers results low reliability
- Secret telephone numbers seem to require two ENUM systems: the ”Operator ENUM” with no direct access by users and ”user ENUM”.
- Result is always the same for a number irrespective of from where the call is originating in a domain →Non-optimal routing.
- Number Portability accross technology boundary would require changes in PSTN (link between IN and ENUM)
- Using ENUM for calls from PSTN is difficult because of overlap sending: non-complete numbers are not described in ENUM records.
- Management of numbering data.
- Security (DNSSec under development…?)
- Nicklas Beijar of Netlab suggests solutions to some of the above problems in his Lic thesis 2004.
- ENUM pilot in Finland until 31.5.2004, after that commercial operation?

IP Telephony Research in the Networking Laboratory

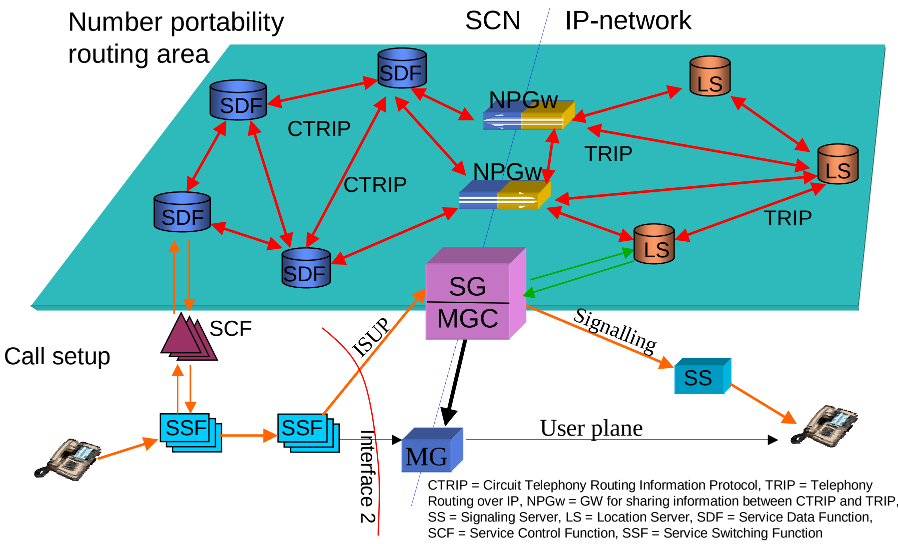
- Technology evaluation
  - Delay measurements breakdown (1997…)
  - SIP call waiting
- Numbering and Routing Information Interoperability with ISDN
  - TRIP and ENUM protocols
  - CTRIP protocol proposed
  - Database (mySQL) solution to Number Portability (Antti Paju)
  - Nicklas Beijar’s Lic thesis (Spring 2004) on alternative solutions for NP

Agenda (10.3.2004)

- TRIP, CTRIP – gateway location
- Megaco
- SDP – session descriptions
- COPS – policy based networking
- STUN and TURN – NAT traversal

The solution is CTRIP + Numbering gateway

IPANA->IMELIO->INTERO



CTRIP = Circuit Telephony Routing Information Protocol, TRIP = Telephony Routing over IP, NPGw = GW for sharing information between CTRIP and TRIP, SS = Signaling Server, LS = Location Server, SDF = Service Data Function, SCF = Service Control Function, SSF = Service Switching Function

TRIP (Telephony Routing over IP)

Framework in RFC 2871
Protocol defined in RFC 3219 (Jan 2002)

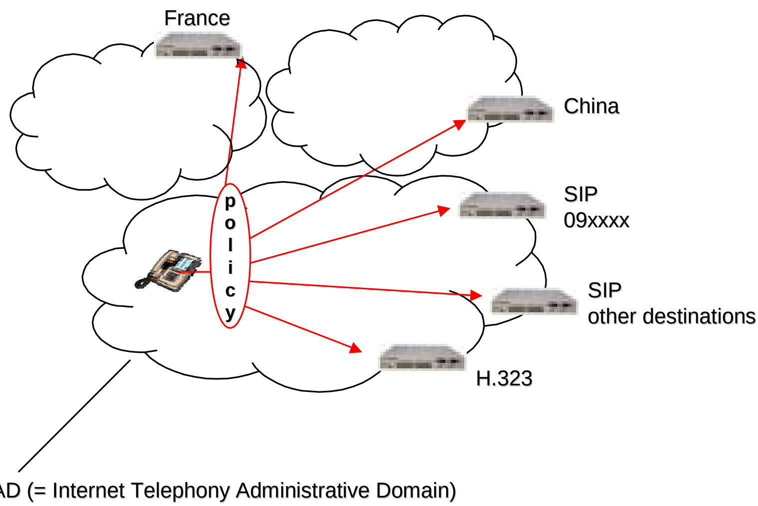
Purpose to advertise
- Reachability of telephony destinations (in ISDN)
- The attributes of the destinations
- The attributes of the path towards the destinations

Advertisements sent between location servers (LS)
⇒ Forms routes to gateways (passing through signaling servers)

Solves the gateway location problem for calls from the IP network to the ISDN.

N.Beijar 8.4.2002

TRIP motivation



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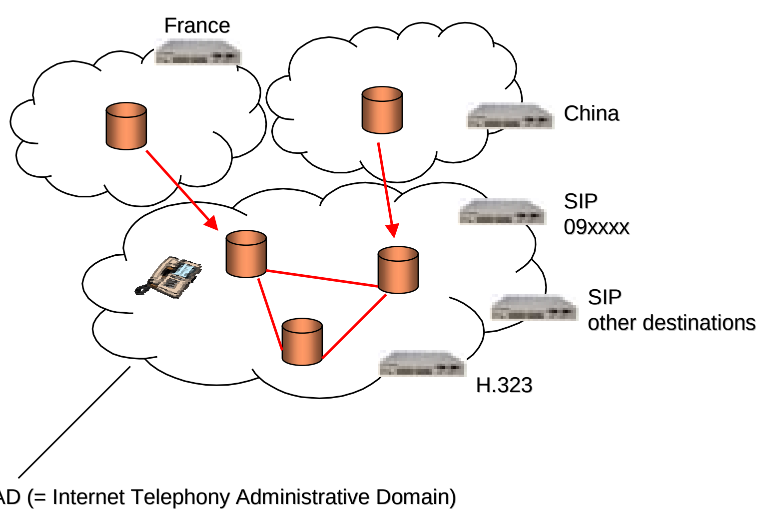
TRIP principle



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TRIP

Interdomain distribution between ITADs
- Based on BGP-4 (Border Gateway Protocol)
- Gateway selection driven by policies

Intradomain synchronization within the ITAD
- Based on OSPF, SCSP, IS-IS

Information transported as attributes of the UPDATE message
- Attributes can be added -> Expandable
- Flags control how unrecognized attributes are handled

Independent of signaling protocol

N.Beijar 8.4.2002

Policies

Gateway selection criteria
- Location
- Business relationships (charging arrangement)
- Policies
- Features
  - Signaling protocol
  - Codec
  - Service
- Capacity

The primary criteria for selecting a gateway is that the gateway can and is willing to route the call to the ISDN destination. For that the gateway needs to know the destination address.
Policies make the selection more accurate.

N.Beijar 8.4.2002

TRIP attributes

Name	Description
Withdrawn routes	List of telephone numbers that are no longer available.
Reachable routes	List of reachable telephone numbers.
Next hop server	The next signaling server on the path towards the destination.
Advertisement path	The path that the route advertisement has traveled.
Routed path	The path that the signaling messages will travel.
Atomic aggregate	Indicates that the signaling may traverse ITADs not listed in the routed path attribute.
Local preference	The intra-domain preference of the location server.
Multi exit disc	The inter-domain preference of the route if several links are used.
Communities	For grouping destinations in groups with similar properties.
ITAD topology	For advertising the ITAD topology to other servers in the same ITAD.
Authentication	Authentication of selected attributes.

N.Beijar 8.4.2002

TRIP for Gateways

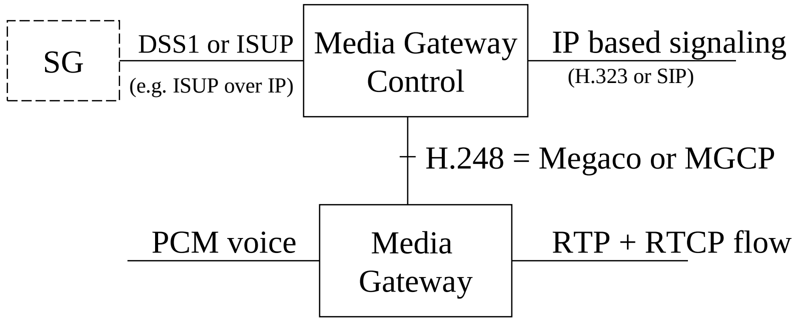
- Draft: draft-rs-trip-gw-03.txt
- Exports routing information from gateways to location servers
- New attributes
  - Circuit capacity
  - DSP capacity
- Due to the dynamic nature, only used for the first hop
- Lightweight
  - Send-only mode
  - No databases
- Compatibible with TRIP

N.Beijar 8.4.2002

Megaco - Media Gateway Control protocol controls Media Gateways and Media Processing

- MGCP was promoted by Cablelabs = US CATV R&D body as the CATV Telephony standard
- ITU-T has its own variant called Megaco=H.248
- Megaco, MGCP are master-slave protocols by which media gateways can be configured e.g to services - in case of residential media gateway, MGCP becomes a subscriber signalling system

Gateway decomposition

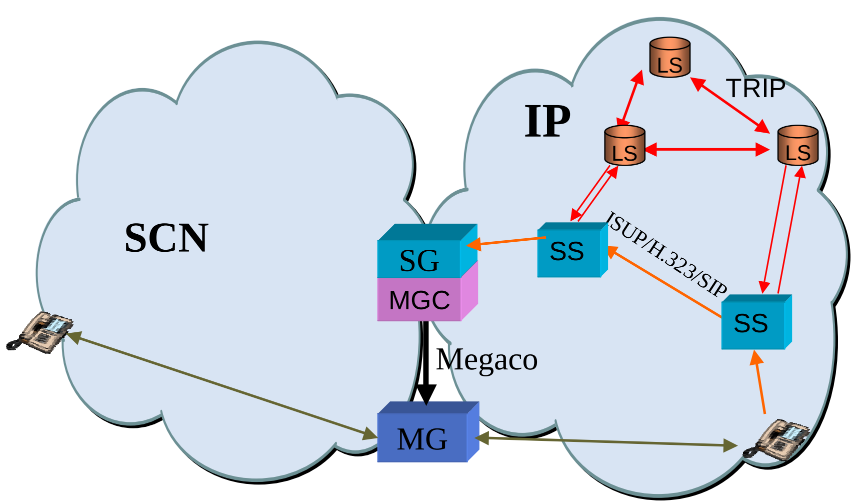


MG - Trunk gateway, residential gateway etc.
Many MGs can be controlled by one MGC, MGCs can be a mated pair --> higher availability performance.

Megaco functions

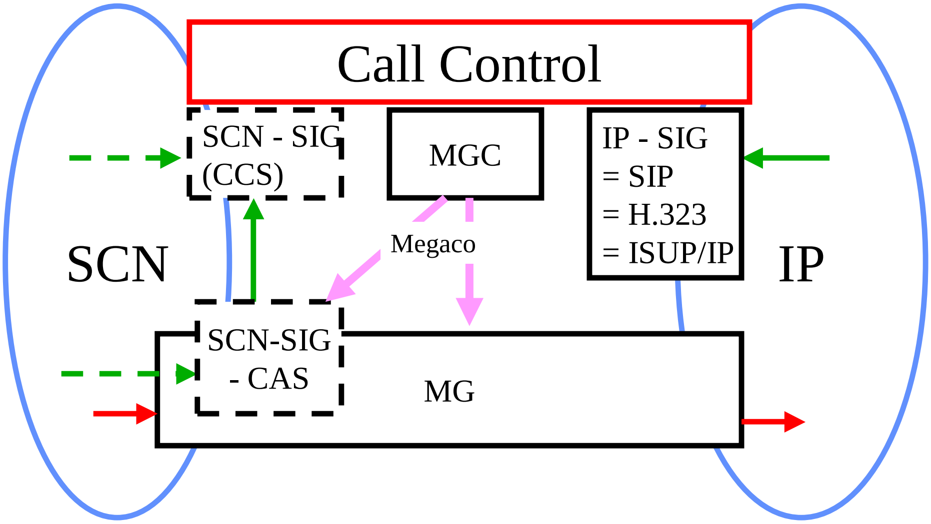
- Establishment of connections between terminations
  - PCM –timeslots for voice
  - ephemeral packet stream terminations: IP-address + source + dest UDP-port number
- Release of connections
- separation of signaling from voice band in case of CAS and analogue subsc signaling

Current Architecture



TRIP = Telephony Routing over IP, SG - Signalling Gateway, MGC - Media Gateway Controller
MG - Media Gateway, SS = Signaling Server, LS = Location Server

Gateway decomposed



Megaco for Residential Gateways

- Residential MG processes analogue subscriber signaling – inband, can not be separated from media plane
- Controller gives a dialling pattern for MG to look for. When detected, report to MGC. MGC gives a new pattern to look for. Etc.
- Real time processing of signals is delegated to the residential gateway, while MGC retains overall control over what is happening and what is the interpretation of the patterns.

QoS – Integrated Serv. and DiffServ help resolving the QoS issue in VOIP and 3G IMS

- Integrated Services
  - Different treatment to different flows
  - State info stored in network, routers examine packets!!!(not good)
  - Reservation merging
  - RSVP protocol – for reservation of resources

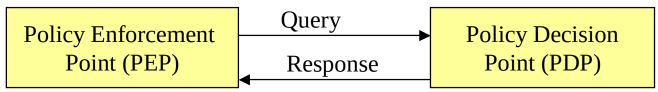
- DiffServ
  - Defines a small nrof traffic classes with different priority levels
  - Packets tagged with level tags at the beginning(ingress)
  - Routers just examine tags
  - Better scaling
  - Requires policy management: e.g. which packets to assign to which class.

SIP Sessions require policy control

- Parties can release the "call session" but since they have obtained each others IP-addresses, they can continue sending media streams to each other!!
- How to push INVITE to B-party, if B-party does not have a permanent IP address which is most often the case!

Integration of Proxy with Firewall and NAT

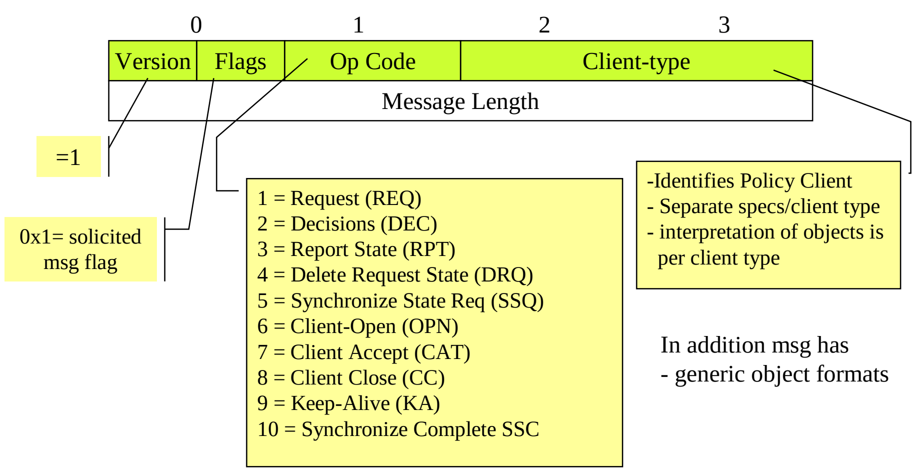
Common Open Policy Service Protocol (COPS) can be used to exchange policy info



- Examples of PEPs are Network Address Translators (NAT), Firewalls, RSVP Routers, GGSN in 3G
- PEP sends requests, updates, deletes to PDP
- PDP returns decisions to PEP (can also overwrite its decision at any time)
- Uses TCP for transport, Extensible for different PEPs
- PEP and PDP share state
- In case of PDP failure, PEP can make local policy decisions

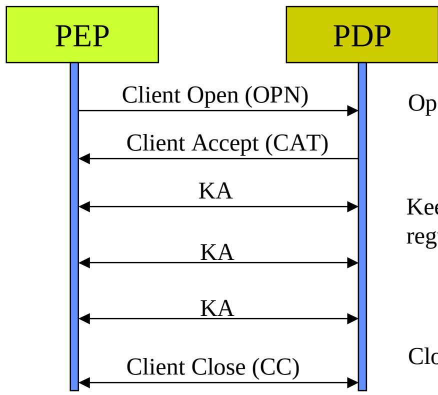
COPS Common Header

RFC 2748 of Jan 2000



In addition msg has
- generic object formats

COPS maintains a TCP session

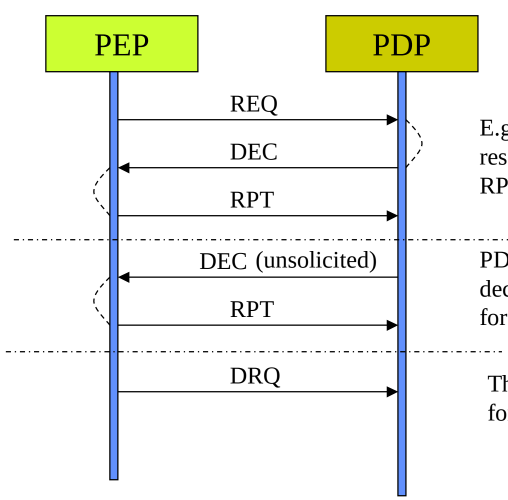


Opening a session establises a context

Keep-alive messages must be sent on regular intervals

Closing the session removes all state

PDP makes policy decisions on request or at any time

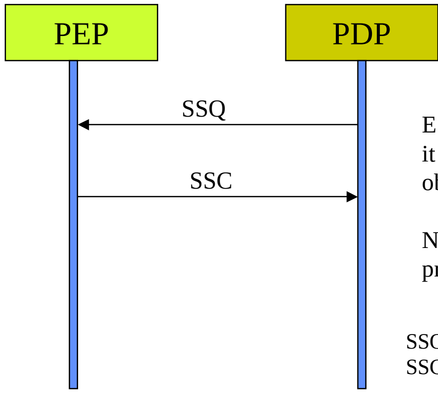


E.g. PEP may need to allocate some resourse – PDP makes the decision
RPT – reports the state change at PEP

PDP may at any time change its previous decision: e.g. default policy is overridden for a time. PEP must abide always!

There may be a need to remove state for a object: PDP needs to know.

PDP may need to synchronize its state with PEP



E.g. PDP has failed and after recovery it needs to restore the state of policy objects from the network (i.e. from PEPs)

NB: PEP does not change its state in this procedure!

SSQ – Synchronize State reQuest
SSC – SynchroniSe Complete

Use examples for COPS

- Wireline VOIP: COPS can be used to control a NAT+Firewall (PEP) from a Proxy Server (PDP).
  - Default policy is: all TCP/IP ports for media streams are closed (deny policy)
  - Per SIP session Proxy sends a DEC message to “open the gate” for bi-directional media flow.
  - When BYE is received, gate is again closed
- 3G IMS: to authorize resources for PDP contexts of media flows.

SDP: Session Description Protocol

- SDP was initially designed for Mbone. Mbone was/is a multicast overlay network on the Internet
- Used to describe sessions (to link the session with media tools)
- Describes conference/session addresses and ports + other parameters needed by RTP, RTSP and other media tools
- SDP is carried by SIP, SAP: Session Announcement Protocol etc.

Multicast

- Several parties involved
  - IPv4 Multicast from 224.0.0.0 – 239.255.255.255
- Saves bandwidth cmp to *n* times p2p connection
- Entity that is sending does not have to know all the participants
- Multicast Routing protocols
  - Dense Mode (shortest-path tree per sender)
  - Sparse Mode (shared tree used by all sources)
- IGMP (Internet Group Management Protocol)
  - For hosts that want to become part of multicast group
- Mbone – part of Internet that supports multicast
- RTP – transport of real-time data such as voice or video
  - Sequence number, timestamps
- RTCP – controls RTP transport (every RTP session has a parallel RTCP session.)

SDP can describe

- Session name and purpose
- Time(s) the session is active
  - start, stop time, repetition
- The media comprising the session
  - video, audio, etc
  - transport protocol: RTP, UDP, IP, H.320 etc
- Parameters to receive media: addresses, ports, formats etc.
  - H.261 video, MPEG video, PCMU law audio, AMR audio
- Approximate bandwidth needed for the session
- Contact info for person responsible

SDP info is <type>=<value> in strict order

<type> is a single, case sensitive character.
<value> is a text string or a nrof fields delimited by a single white space char.
SDP has one session level description and optionally *n* media descriptions.

Session description

* = optional

v= (protocol version)
o= (owner/creator and session identifier).
s= (session name)
i=* (session information)
u=* (URI of description)
e=* (email address)
p=* (phone number)
c=* (connection information - not required if included in all media)
b=* (bandwidth information)

One or more time descriptions (see below)
z=* (time zone adjustments)
k=* (encryption key)
a=* (zero or more session attribute lines)
Zero or more media descriptions (see below)

SDP items continued

Time description
- t= (time the session is active)
- r=* (zero or more repeat times)

Media description
- m= (media name and transport address)
- i=* (media title)
- c=* (connection information - optional if included at session-level)
- b=* (bandwidth information)
- k=* (encryption key)
- a=* (zero or more media attribute lines)

3G document refer to a newer SDP- draft from may 2002.

Some SDP documents:

RFC 2327: SDP Session Description Protocol (dated 1998), now Proposed Std
RFC 3407: SDP Simple Capability Declaration
RFC 3264 - An Offer/Answer Model with Session Description Protocol (SDP)
RFC 3266 - Support for IPv6 in Session Description Protocol (SDP)
RFC 3556 SDP Bandwidth modifiers for RTCP

NAT Traversal

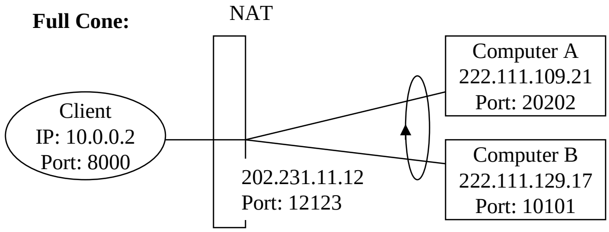
RFC 3489 Title: STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
Author(s): J. Rosenberg, J. Weinberger, C. Huitema, R. Mahy
Status: Standards Track Date: March 2003
See also: http://corp.deltathree.com/technology/nattraversalinsip.pdf
Traversal Using Relay NAT (TURN) draft-rosenberg-midcom-turn-03

- For the purpose of IPv4 address saving, many users sit behind Network Address Translators.
- NATs are of 4 types: Full Cone, Restricted Cone, Port Restricted Cone and Symmetric.
- NAT address/port mappings will be dropped after some time of not seeing packets thru the mapping

Internet is an A-subscriber’s Network! B-subscribers are not connected!

NAT Types 1, 2, 3

Full Cone:



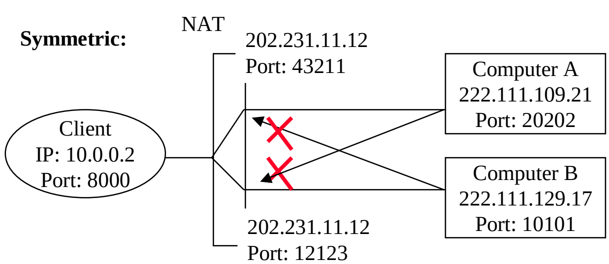
1. Client sends a msg to A
2. A responds to NAT address and port
3. Also B can send msgs to client

Restricted Cone: NAT will block messages from B until Client has sent a msg to B, After that both A and B will see the same mapping in NAT

Port Restricted Cone: NAT will block packets from all ports but the one to which Client has previously sent packets.

NAT type: Symmetric

Symmetric:



NAT provides a different mapping for different destinations. Messages from Computer B to Cient will be blocked thru the mapping established for Computer A.

STUN does not allow incoming TCP connections to traverse thru NATs,
STUN does not allow incoming UDP packet thru Symmetric NATs.

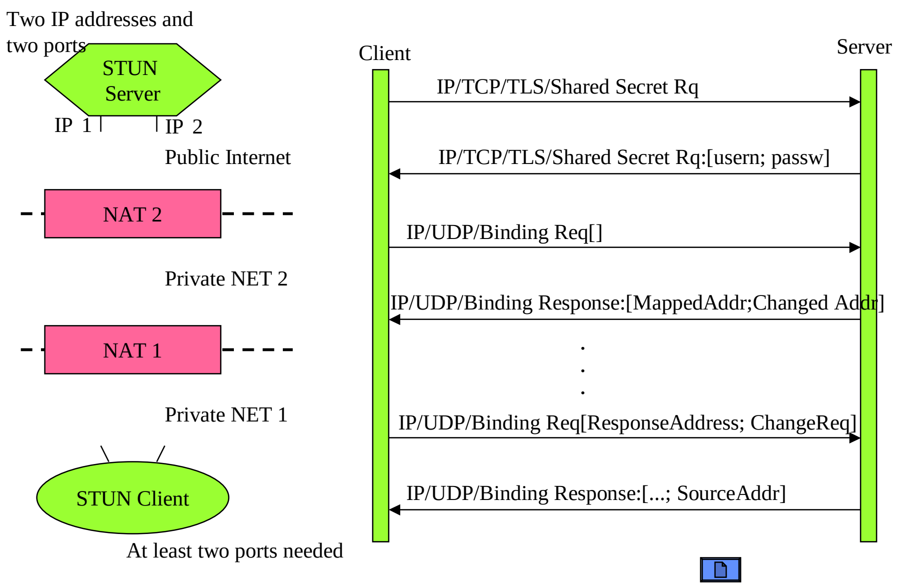
Symmetric NATs are common in large Enterprises.

STUN does not allow communication between two parties behind the same NAT using public Internet addresses.

Alternative approaches of NAT traversal

- Application Gateway: Application functions are embedded in the NAT. These functions rewrite parameters in Application protocol fields, e.g. in SIP messages.
- MIDCOM (RFC 3303) – a protocol is used to control the NAT by an Application proxy server. Requires changes to existing NATs. Requires a control relationship between the NAT and the proxy.
- STUN - allows entities behind a NAT to first discover the presence of a NAT and the type of NAT, and then to learn the addresses bindings allocated by the NAT. STUN requires no changes to NATs, and works with an arbitrary number of NATs in tandem between the application entity and the public Internet.

STUN model assumes nested NATs



Types of NAT are discovered by sending responses from different source address and port

Flags	Source Address	Source Port	CHANGED-ADDRESS
none	Da	Dp	Ca:Cp
Change IP	Ca	Dp	Ca:Cp
Change port	Da	Cp	Ca:Cp
Change IP and Change port	Ca	Cp	Ca:Cp

Table 1: Impact of Flags on Packet Source and CHANGED-ADDRESS in Binding Response

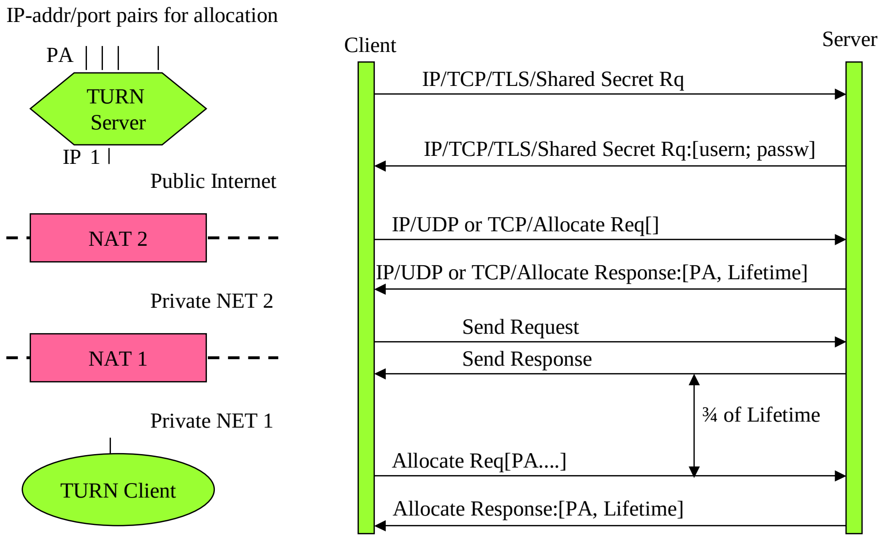
The full procedure of discovering the type of NAT and Firewall is in the RFC

STUN plays with the identity of the user: opens a door for inpersonation. Therefore, security, excluding man-in-the-middle attacks is crucial!
When a SIP application fills in SDP fields and some SIP fields, NAT traversal needs to be taken into account!

Traversal Using Relay NAT(TURN) helps with Symmetric NATs

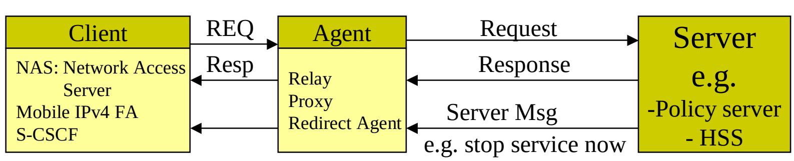
- **TURN allows for an element behind a NAT or firewall to receive incoming data over TCP or UDP connections from a single Peer.**
- **TURN does not allow for users to run servers on well known ports if they are behind a NAT**
- **Based on draft: draft-rosenberg-midcom-turn-03.**
- **Technically TURN is an extension to STUN (protocol formats and attributes), TURN can be co-implemented with STUN. TURN-server+STUN-server and TURN-cliet + STUN-client**
- **a TURN server allocates a Public Internet IP-address/port pair (PA) to the Client. Relays messages sent to PA to the Client wrapped in TURN headers.**

TURN model is similar to STUN



Diameter is the emerging AAA protocol for the Internet and 3G

- Applications include:
  - Network Access Servers for dial-up with PPP/SLIP,
  - Mobile IPv4 Foreign Agents,
  - roaming 3G and Internet users.
- Provides Authentication of users, Authorization and Accounting of use
- Carried over TCP or SCTP



Diameter documents

Diameter Base Protocol
draft-ietf-aaa-diameter-16.txt

Transport Profile (AAATRANS)
- transport issues
- server failover

Applications

Mobile IPv4 (DIAMMIP)
-FA

NASREQ
- PPP/SLIP access
  to Internet

Diameter Multimedia Application (3GPP)
- defined by 3GPP for 3G IMS
- Client: S-CSCF or I-CSCF
- Server HSS
- Ridirect: SLF

Diameter features include

- Delivery of attribute value pairs: AVPs
- Capability negotiation
- Error Notification
- Extensibility
- Sessions and Accounting

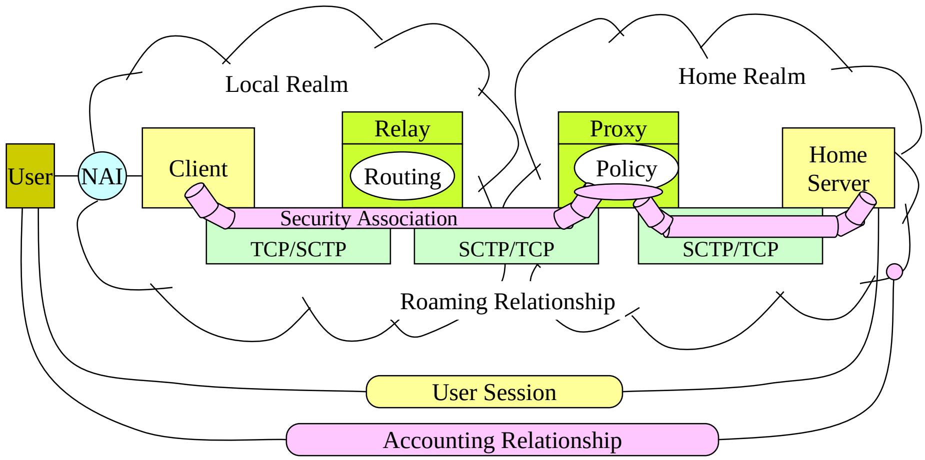
User Authentication

Service specific authentication info -> grant service or not

Resource usage information
- accounting and capacity planning is supported

Relay, proxy and redirect of requests thru a server hierarchy

Diameter operation model



NAI – Network Access Identifier = user's-identity + realm

Diameter terms and definitions

Accounting
The act of collecting information on resource usage for the purpose of capacity planning, auditing, billing or cost allocation.

Authentication
The act of verifying the identity of an entity (subject).

Authorization
The act of determining whether a requesting entity (subject) will be allowed access to a resource (object).

AVP
The Diameter protocol consists of a header followed by one or more Attribute-Value-Pairs (AVPs).
AVP = header encapsulating protocol-specific data (e.g. routing information) + AAA information.

Broker
A broker is a business term commonly used in AAA infrastructures. A broker is either a relay, proxy or redirect agent, and MAY be operated by roaming consortiums. Depending on the business model, a broker may either choose to deploy relay agents or proxy agents.

Diameter Agent = Diameter node that provides either relay, proxy, redirect or translation services.

Diameter Node = a host process that implements the Diameter protocol, and acts either as a Client, Agent or Server.

More Diameter terms

Diameter Security Exchange = a process through which two Diameter nodes establish end-to-end security.

Diameter Server = one that handles AAA requests for a particular realm. By its very nature, a Diameter Server MUST support Diameter applications in addition to the base protocol.

End-to-End Security
TLS and IPsec provide hop-by-hop security, or security across a transport connection. When relays or proxy are involved, this hop-by-hop security does not protect the entire Diameter user session. End-to-end security is security between two Diameter nodes, possibly communicating through Diameter Agents. This security protects the entire Diameter communications path from the originating Diameter node to the terminating Diameter node.

Home Realm = the administrative domain with which the user maintains an account relationship.

Interim accounting
An interim accounting message provides a snapshot of usage during a user's session. It is typically implemented in order to provide for partial accounting of a user's session in the case of a device reboot or other network problem prevents the reception of a session summary message or session record.

Local Realm
A local realm is the administrative domain providing services to a user. An administrative domain MAY act as a local realm for certain users, while being a home realm for others.

Still more terms

Network Access Identifier or NAI [NAI] = a user's identity + realm.
The identity is used to identify the user during authentication and/or authorization, the realm is used for message routing purposes.

Proxy Agent or Proxy
- forward requests and responses,
- proxies make policy decisions relating to resource usage and provisioning. This is typically accomplished by tracking the state of NAS devices.
- proxies typically do not respond to client Requests prior to receiving a Response from the server,
- they may originate Reject messages in cases where policies are violated.
- proxies need to understand the semantics of the messages passing through them, and
- may not support all Diameter applications.

Real-time Accounting
Real-time accounting involves the processing of information on resource usage within a defined time window. Time constraints are typically imposed in order to limit financial risk.

Relay Agent or Relay
- Relays forward requests and responses based on routing-related AVPs and realm routing table entries.
- do not make policy decisions, they do not examine or alter non-routing AVPs.
- relays never originate messages, do not need to understand the semantics of messages or non-routing AVPs,
- are capable of handling any Diameter application or message type.
- do not keep state on NAS resource usage or sessions in progress.

The last terms

Redirect Agent
- refer clients to servers and allow them to communicate directly.
- do not sit in the forwarding path → they do not alter any AVPs transiting between client and server.
- do not originate messages and
- are capable of handling any message type, although they may be configured only to redirect messages of certain types, while acting as relay or proxy agents for other types.
- do not keep state with respect to sessions or NAS resources.

Roaming Relationships
Roaming relationships include relationships between companies and ISPs, relationships among peer ISPs within a roaming consortium, and relationships between an ISP and a roaming consortium.

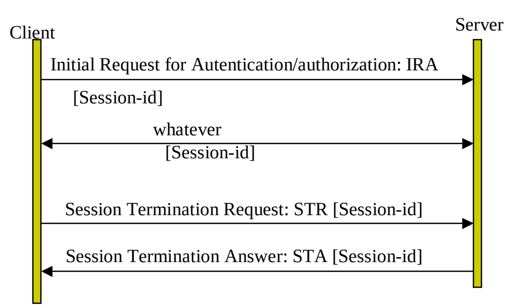
Security Association
A security association is an association between two endpoints in a Diameter session which allows the endpoints to communicate with integrity and confidentially, even in the presence of relays and/or proxies.

Session = a related progression of events devoted to a particular activity. Each application SHOULD provide guidelines as to when a session begins and ends. All Diameter packets with the same Session-Identifier are part of the same session.

Sub-session represents a distinct service (e.g. QoS or data characteristics) provided to a given session. These services may happen concurrently (e.g. simultaneous voice and data transfer during the same session) or serially. These changes in sessions are tracked with the Accounting-Sub-Session-Id.

Translation Agent performs protocol translation between Diameter and another AAA protocol, such as RADIUS.

Access is broken into sessions: Diameter authorizes sessions



A diameter node has a peer table

Host identity	Status	Stat/Dyn	Expiration time	TLS enabled	Additional Security info

origin host
-from capability
exchange:
CER/CEA

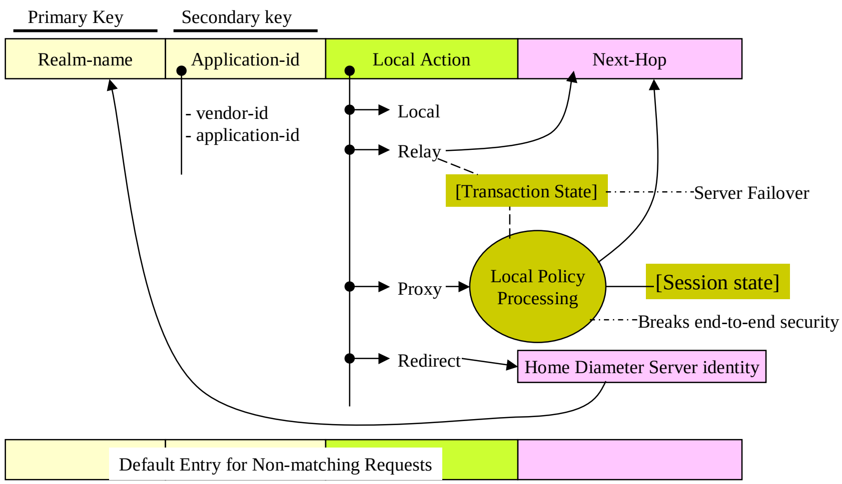
- Closed
- Wait-conn-ack
- wait-I-CEA
- wait-I-CEA/Elect
- wait-returns
- R-Open
- I- Open
- ….
- …
- Stop
- = state of the “dialog” with the peer

The peer table is referenced by Realm Routing Table.
The peer relationship may be dynamically established – will have an expiration time.

Diameter peer discovery helps scalability: order is as follows

- Search manually configured peer agent list
- Use SLPv2 (service location protocol)
- NAPTR query to DNS (”AAA+D2x where x=T|S, T=tcp, S=sctp) – gives the preferred SRV record, a new query gives the IP address
- query `_diameter._sctp´.realm and `_diameter._tcp´.realm, where realm is the destination realm

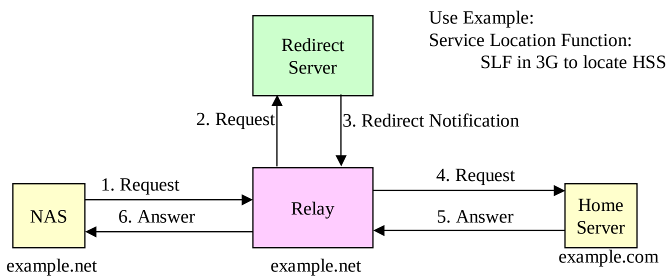
Realm Routing Table describes the actions of a Diameter Node



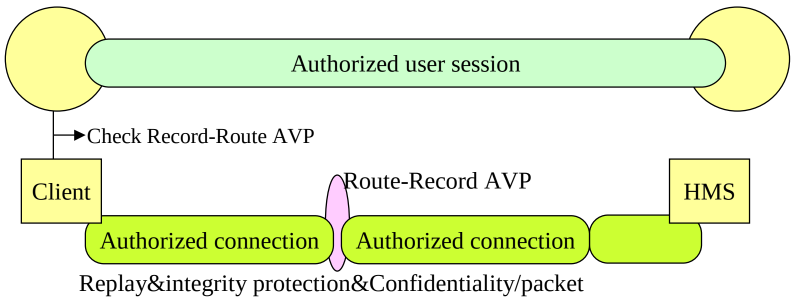
A node can act as proxy for some user connections and as a relay for others.
The Routing Table is configuration information.

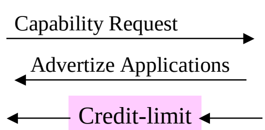
Redirect server helps to centralize Diameter request routing in a roaming consortium

Use Example:
Service Location Function:
SLF in 3G to locate HSS



A node must watch over its peers to achieve security





- Capability negotiation tells a node what to expect of a peer
- Authorization means taking a business risk, limited by Credit limit agreed by the peer realms.

Diameter header is designed for max flexibility

Version=1	Message Length
Command Flags	Command-Code
Application-ID	
Hop-by-Hop Identifier	
End-to-End Identifier	
AVPs	

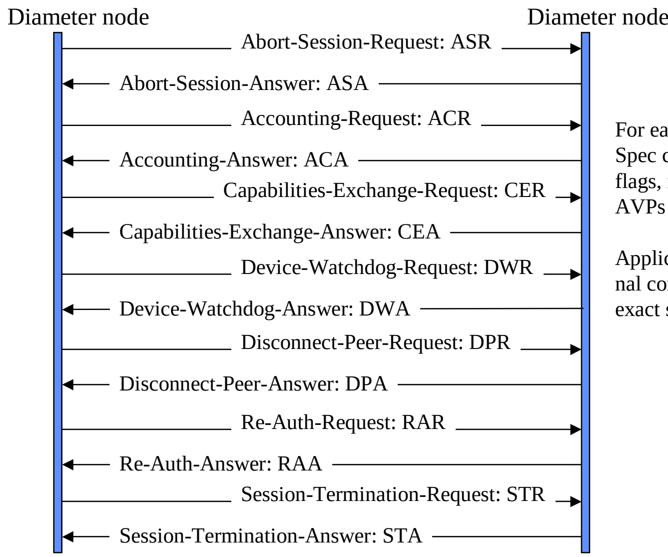
Application-ID: e.g. 3GPP application

Normally +1 increasing number on a connection
Same for Request and the corresponding Answer

R(equest) – if 0 = Answer
P(roxiable) – if 0 msg must be locally processed
E(rror) – only set in Answer msgs.
T(potentially re-transmitted message
- set after failover to help remove duplicate messages

Client sets to locally unique value (4 min) even over Reboots
Server copies from Request to Answer

Base Diameter protocol Requests and Answers



For each Command-code Spec contains exact possible flags, required and optional AVPs and their nr.

Applications introduce additio-nal command-codes and their exact syntax.

Base protocol AVPs

AVPs have a common header

AVP Code	
VMPrrrrr	AVP Length
Vendor-ID (opt)	
Data…	

V-vendor-id present
M-Mandatory AVP
P-encryption for e-2-e sec

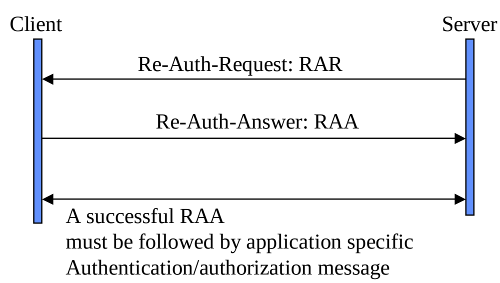
In AVPs e.g. the following items may appear:
- IPaddress
- Time
- UTF8String
- Diameter Identity = FQDN
  (fully qualified domain name)
- Diameter URI such as
  ”aaa://” FQDN [port] [transport] [protocol]
  aaa://host.example.com:1813;transport=sctp; protocol=radius
- IPFilterRule such as
  action dir proto from src to dst [options], where
  action =permit|deny
  dir=in|out (in = from the terminal)
  src/dst = <address/mask> [ports]

You can specify firewall rules in Diameter.

A diameter node operation is described as a set of state machines

- Peer state machine
- Authorization Session State Machines (4)
  - Server maintains session state: client FSM and server FSM
  - Server does not maintain session state: client FSM and server FSM
- Accounting Session State Machines
  - Client state machine
  - Server state machines: stateless and stateful
  - may be overridden by applications

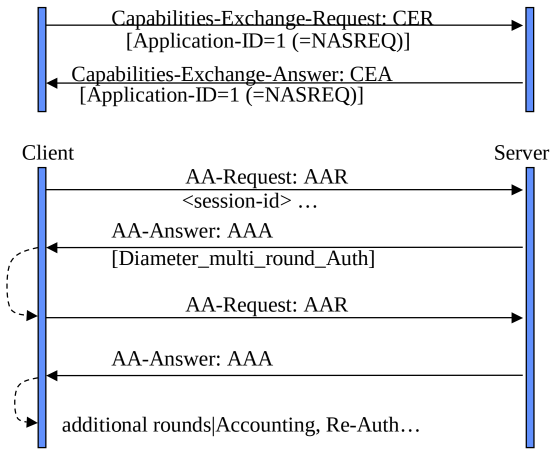
Server may require Re-authentication/authorization



Use example: enforcing a credit limit on a user during a long telephone call.

NASREQ defines an authentication and authorization application

draft-ietf-aaa-diameter-nasreq-10.txt of Nov 2002.



In Capabilities exchange peers agree to understand NASREQ commands.

NAS (PoP) initiates a new session.

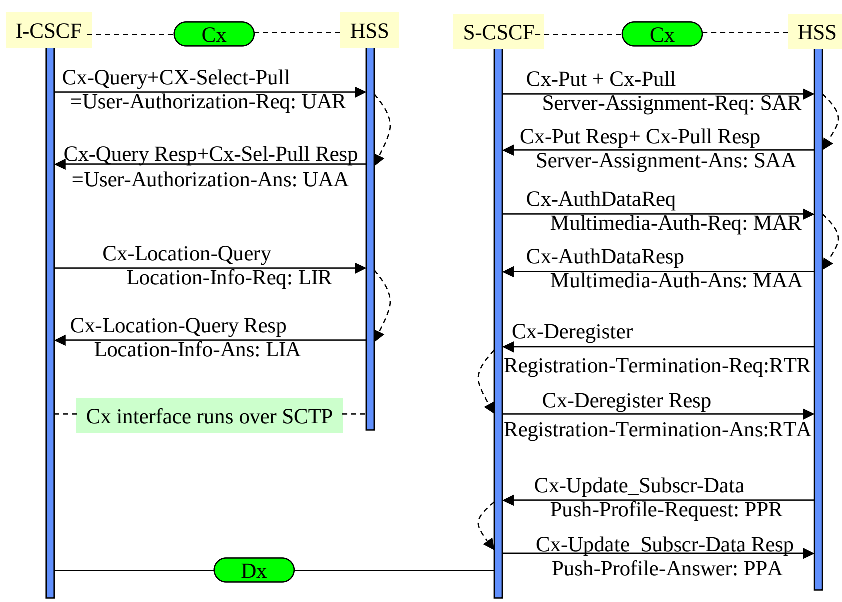
HMS **may** challenge the user.

User has to respond to challenge

Accounting is defined in BASE.

AAR and AAA have loads of AVPs!

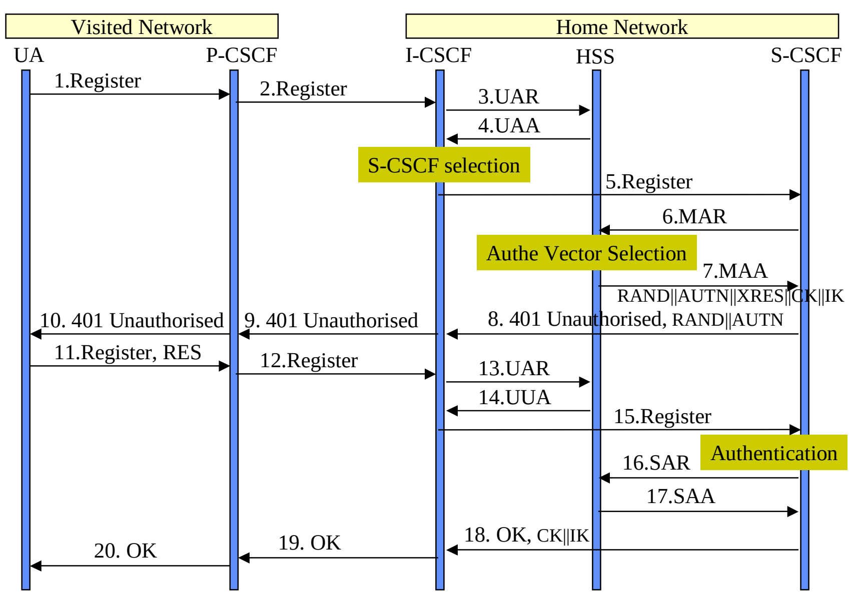
3GPP defines Diameter Multimedia Application



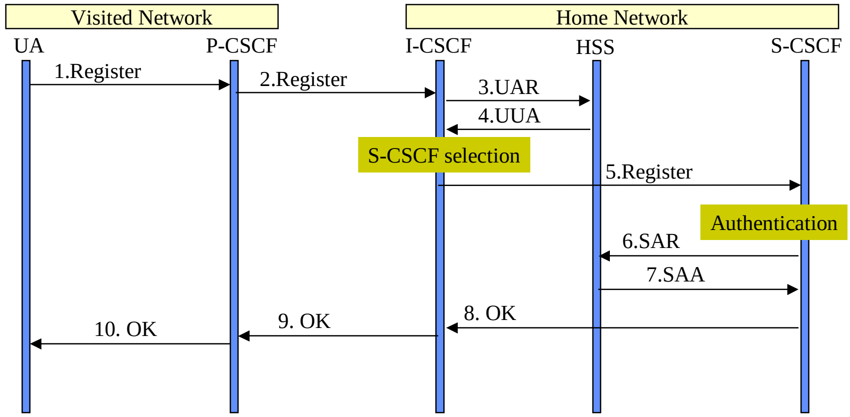
MM Application properties

- 3GPP has a Vendor-ID, 3GPP MM Application is defined as a vendor specific application.
- ”Cellular” Location management maps into MAP operations in SGSN+GGSN+ Registration/De-Registration in SIP terms maps to Authorization-Request/-Answer in Diameter + S-CSCF obtaining Subcr data = Diameter Profile-Push etc.
- User-Location-Query is used to obtain S-CSCF identity
- I-CSCF can use Diameter Redirect capability in SLF: Server-Location-Function to select S-CSCF/user-identity
  - I-CSCF is stateless, so SLF has to be used for every query
  - S-CSCF is stateful and will cash HSS address for the session.

Registration – user not registered

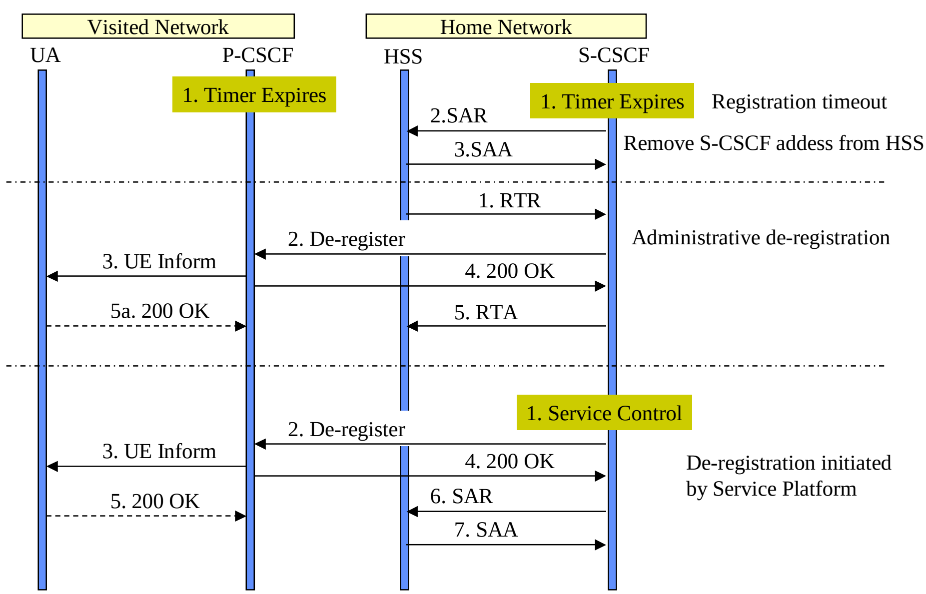


Registration – user currently registered

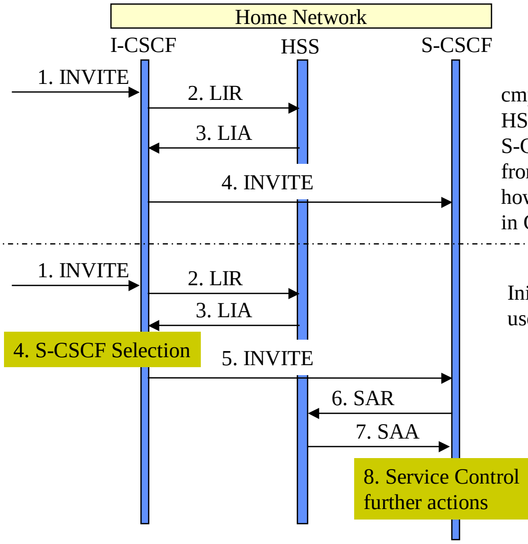


- Registration may need to be refreshed from time to time.
- Location changes may require re-registration.
- Mobile Initiated de-registration looks exactly the same!

Many ways/reasons to de-register



Mobile Terminated SIP Session Set-up is similar to MAP MT call



cmp: SendRoutingInformation of MAP
HSS knows the name (and address) of S-CSCF – no RoutingNumber is needed from ”VLR”. So there is a difference in how routing and addressing operates in GSM and in 3G IMS.

Initiation of a session to a non-registered user.

Further on,
For S-CSCF operation HSS issues Push-Profile-Request: PPR and S-CSCF answers by PPA.

Summary

- IP telephony requires many supporting protocols.
- Many IETF protocols overlap with GSM protocols (e.g. Diameter with MAP) in terms of functionality
- IETF development model is one protocol for one problem.
- Client-Server model is used whenever possible.
- The drive is towards providing PSTN like control over services and over what a user can do in the IP environment.
- Through access to the Internet, the open Internet model lives on.