Architectures and Supporting Protocols for VOIP/3G -- A

IETF and 3GPP at work
NGN and IMS (why)
3G Network Elements
Numbering and Naming (ENUM)
Session Description Protocol (SDP)
Media Gateway Control (Megaco/MGCP)
Agenda

- IETF
- Networking framework – 3G, wireline
- Why control what users can do?
  - Justification for 3G IMS architecture
- 3G terminal
IETF

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

3.1.2014

- 128 active WGs
- here are some of them
IETF Areas and Area Directors

2.1.2014

Comnet PhD

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Jari Arkko, Ericsson (Finland)

Applications Area (app)
Barry Leiba, Huawei Technologies
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Internet Area (int)
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Real-time Applications and Infrastructure Area (rai)
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Routing Area (rtg)
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Security Area (sec)
Stephen Farrell, Trinity College Dublin
Sean Turner, IECA, Inc.

Transport Area (tsv)
Spencer Dawkins, Huawei
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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.
3GPP Recent/Upcoming Releases

IMS - all based on WCDMA
- HSPA - Both CS and PS
- LTE - Only packet service

Rel 5  6/2003
Rel 6  12/2004
Rel 7  9/2007
Rel 8  Dec 2008
Rel 9  Dec 2009
Rel 10 2011
Rel 11

LTE-A 100M-1G
Final 12/2009
3G is composed of many Subsystems

- UTRAN
- Circuit Switched Domain
  - Packet Switched Domain → Enhanced Packet Core
  - IMS IP Multimedia Subsystem
- Other IP Connectivity Access Network
  - LTE
- UE

Gb
A

IP-CAN = IP Connectivity Access Network
3G IP Multimedia core network Subsystem (3G IMS)

- AS – Application Server
- CAP - Camel Application Part
- IM-SSF – IP Multimedia Service Switching Function
- ISC – IP Multimedia Service Control
- S-CSCF – Serving Call Session Control Function
- HSS – Home Subscriber Server
- HSS – Home Subscriber Server
- Cx
- ISC
- Sh
- ISC
- MAP
- Si
- Mr
- Mr
- MAP
- MRFC
- CAP
- MRFC
- Camel Service Environment
- OSA service capability server (SCS)
- OSA application server
- OSA API
- SCIM
- AS
- SCIM
- AS
- S-CSCF
- S-CSCF
- S-CSCF
- S-CSCF
- S-CSCF
Alternative to IMS?

• With a 3G device a user can access the open Internet and use any services that are available on the Internet: www, e-mail, conferencing, Skype, other VOIP etc.
  – QoS is the Best Effort QoS of regular Internet
  – Charging can be either volume based or flat rate.
  – Flat rate leads to fast growth of traffic and can finally lead to overuse of the cellular capacity and poor QoS

• Take the CS domain signaling and call control, map TDM trunks to IP ”connections” → retains the existing CS – domain services control and architecture, replace TDM transport by IP (this was called UMA – universal mobile access → now GAN = Generic Access to A/Gb interfaces in rel 6)
Motivation for IMS

- IMS = Integration of cellular and Internet worlds. Why, when a user already can take an Internet connection from a cellular device and use all Internet Services?
  - Controlled QoS for Interactive voice and video (shortest path for Media while service control at home network!)
  - Proper Charging for QoS and Freedom of charging based on any business model for the services
  - Integration of services on a single packet platform: access to all aspects of sessions from any service.
  - Ease of interworking with Internet Services(?)

Q: Is this enough?
Q: Operators have been asking: Why should they switch from circuit based voice services to IMS based voice services in 3G? But the move is now taking up speed. In LTE, Voice over LTE is based on IMS.
**IMS Objectives (official)**

Support for the following:

1. establishing IP Multimedia Sessions
2. negotiation of QoS
3. interworking with the Internet and the CSN
4. roaming
5. strong control by the operator with respect to the services delivered to the end user
6. rapid service creation without requiring standardization
7. access independence (starting from release 6)

*Carrier Grade service* is a service provided by an operator or a set of operators such that the operator(s) can take full responsibility for the quality of the service. For this, the environment should be predictable and no 3rd party should be able to interfere with the provisioning.
What is missing from the previous list?

• Mobile Operators are not yet willing to guarantee *malware free operation* of smart phones/Mobile devices that are open to user downloaded software

• Operators do not guarantee *privacy of communications*

• Yet if Mobile Device= open platform for any application written for it
  
  – Provides new types of opportunities for fraud compared to PC environment
  
  – E.g. Android recently had 1.5 M applications, +2000/day!!!
Next Generation Network (NGN) is the ETSI effort to harmonize packet telephony

The network architecture is layered in a much more strict sense than in case of CSN

- **Services**
  - IP Applications
  - Virtual Home Environment
  - Open Service Architecture

- **Control**
  - call control
  - session management
  - mobility management

- **Switching**
  - Transcoding at the edge
  - Switching
  - Routing

In practice this means that ETSI has decided to adopt the IMS framework as a basis for services over all kinds of networks wireline or wireless.

This is much more modular than what we have in CS networks!
Post IP Core and Services Architecture – a vision for 10y?

Open source content (www)

Copyrighted Content

Internet Services
Peer to Peer Services
Voice, audio, Video

Managed Services
- charging, DRM
- telephony?
- virtual networks

Legacy networks

Identity, Name services

IP-address, HIP, DHT
URI, SIM-card, etc.

Routing
Mobility management
Overlay technology

802.1 802.X

radio  copper  Fiber
IMS Technology and Architecture

- 3G context
- 3G network element roles
- ENUM
- Megaco
- SDP
Comparison of IN and IMS Architectures

IN
- Purpose: centralized implementation of low penetration services possibly independent of network vendor → software updates only to few nodes when new services are introduced
- Protocols: INAP is used for accessing service logic, ISUP etc are signaling protocols, so 2 protocols are involved → protocol inter-working occurs, if a new signaling protocol is introduced also INAP would need to be updated.
- Services implementation environment is SCF+SDF+SCE available from a few vendors only
- Services triggering takes place in SSF that may reside even in a visited network

IMS
- Purpose: === identical!
- Protocols: Application Server processes SIP directly, only 1 protocol → all signaling info available for services implementation. This is possible because user plane is completely separate from signaling plane.
- AS: implementation can be whatever – not regulated in any way
- Services triggering always takes place in the home network S-CSCF. This means that services implementation is 100% home network responsibility
3G Application Triggering

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

The result is the same as in IN: for low penetration services, only one or a few servers need to be upgraded instead of upgrading all CSCF network elements.

iFC – Initial Filter Criteria
sFC – Subsequent Filter Criteria
SPT – Service Point Trigger
sFC is considered historical (obsolete)
The role of HSS

Mobility Management
User security info. generation
User security support
Service Provisioning support
Call / Session establishment support
GUP Data Repository

Identification handling
Service authorization support
Access authorization
Application Services Support
CAMEL Services Support

Source: 23002-700.doc
Release 7
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.
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

source: www.3gpp.org/specs/archive/23002-580 (Rel 5)
UE has a tunnel to visited IMS

PDP – Packet data protocol (IPv4, IPv6 or X.25 …)

Virtual presence of UE in visited network IM subsystem (UE’s IP-address is here)
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
- GGSN assigns an IP address for the terminal

In this case UE is easier to reach for incoming traffic.

For mobile office applications Intranet connectivity at this level is not popular. Instead IP VPNs are used.
The UMTS terminal functional model

Browser  
Streaming  
Point-to-Point data  
Messaging

FTP  LDAP  DNS  HTTP  SLP  SIP  IMAP  SMTP  X.509  Radius  H.323 ...

QoS extension  
QoS Management

DiffServ  RSVP  TCP  UDP

Socket API  DHCP  RTP/RTCP  WAP

IP  
Packet Classifier  PPP

UMTS
IMS Interworking with the PSTN

• IMS Interworking with PSTN is also possible. Role of BGCF is to locate the best Media Gateway Control F == at which point the technology boundary will be crossed.
Supporting protocols for IP telephony – wired and wireless

- ENUM – addressing and naming
- Gateway control – Megaco = H.248
- 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 telephone 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

"." the root

.arpa

.com

.net

... second-level node

.fi

.e164.arpa

in-addr

second-level node

second-level node

second-level node

".e164.arpa" – server is the root of the ENUM hierarchy. Countries have started reserving names under it and establishing ENUM services/country. Enum pilot in Finland in 2004-2006.

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!".

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

Tier 1 maps a number of a number block to ENUM op, Tier 2 gives the NATPR records. (this is the planned deployment model in 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.

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….,
From Oct 2006 is run by Ficora

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.comnet.tkk.fi!".

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

Tier 3

Corporate numbering schemas…

In Finland ENUM pilot until oct-2006, hence commercial service: Tier 1 and Tier 2 present!
ENUM use and future

- Since DNS is used by everybody, ENUM is a likely survivor, policy routing etc additions may emerge.
- Due to Number Portability, the Provision of ENUM service and the 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.
- “User” ENUM does not support secret telephone numbers
ENUM Varieties

One potential source of confusion, when talking about ENUM, is the variety of ENUM implementations in place today. Quite often, people speaking of ENUM are really referring to only one of the following:

• **Public ENUM**: The original vision of ENUM as a global, public directory-like database, with subscriber opt-in capabilities and delegation at the country code level in the e164.arpa domain. This is also referred to as *user ENUM*.

• **Private ENUM**: A carrier, VoIP operator or ISP may use ENUM techniques within its own networks, in the same way DNS is used internally to networks.

• **Carrier ENUM**: Groups of carriers or communication service providers agree to share subscriber information via ENUM in private peering relationships. The carriers themselves control subscriber information, not the individuals. Carrier ENUM is also referred to as *infrastructure ENUM*, and is being the subject of new IETF recommendations to support VoIP peering.....

http://en.wikipedia.org/wiki/Telephone_number_mapping
Use of ENUM in 3G IMS

• If the callee is identified by tel URL (tel: +358-59-345-897), the originating S-CSCF tries to map this to a SIP URI using a NAPTR query to ENUM
  – successful if the target is a VOIP number
  – if call is made from IMS to GSM, we first try to find the destination in an IP network. This may take a while because the query escalates up in the DNS hierarchy.

• If no mapping is found, it is assumed that the target is a PSTN or any other CSN number and the call signaling is routed to a BGCF (Breakout Gateway Control Function) that is specialised at routing based on telephone numbers.

• The assumption is that only VOIP numbers are found in ENUM and that ENUM does not store telephone numbers of Circuit Networks.
Call from PSTN to a SIP phone

1. Caller dials 4512471

2. Query
   1.7.4.2.1.5.4.9.8.5.3.e164.arpa

3. Response
   sip:raimo.kantola@sip.comnet.hut.fi

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

5. INVITE

6. INVITE

SIP phone
192.151.79.187
or
+358-9-451 2471

How about NP across the technology boundary?
ENUM issues and problems

- Long chain of DNS servers results in low reliability
- Result is always the same for a number irrespective of from where the call is originating in a domain → Non-optimal routing and no policy routing.
- Number Portability across technology boundary would require changes in PSTN (link between IN and ENUM) → Carrier ENUM stores all telephone numbers and their allocation to operators → we avoid changing PSTN or IN
- Using ENUM for calls from PSTN is difficult because of overlap sending: non-complete numbers are not described in ENUM records (leads to many queries with result: Not Found).
- Management of numbering data. DNS mgt tools are not optimal.
- Security (DNSSec?)
IP Telephony Research in the Networking Laboratory till 2010

• Technology evaluation
  – Delay measurements breakdown (1997…)
  – SIP call waiting

• Numbering and Routing Information Interoperability with ISDN
  – TRIP (Telephony Routing over IP) and ENUM protocols
  – CTRIP (Circuit TRIP) protocol proposed
  – Database (mySQL) solution to Number Portability (Antti Paju)
  – Nicklas Beijar’s Lic thesis (Spring 2004) on alternative solutions for NP

• Mobile Peer-to-Peer from 2005
More recent Research at Comnet inspired by VOIP and NATs

• Customer Edge Switching =
  – Cooperative Firewall and Realm Gateway (enhanced NAT)
  – Allows hosts in private address space X to initiate communication with hosts in private address space Y. Hosts in Y do not need to poll in order to maintain NAT binding
  – Introduces communications IDs – all packets are tunnelled thru the core from edge to edge
  – For ease of deployment – Realm GW → legacy hosts can initiate communication with servers that are behind the new type edge device
  – Falls back to NAT operation if server in legacy Internet
  – No compulsory changes to hosts

• This is still a research concept – not standardised yet (RGW does not need ”mandatory” standards).
Megaco - Media Gateway Control protocol controls Media Gateways and Media Processing

- First VOIP/PSTN gateways were monolithic – all functions from signaling to media processing in the same box
- MGCP was promoted by Cablelabs = US CATV R&D body as the CATV Telephony standard, later ITU-T created 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
Gateway decomposed

Call Control

SCN - SIG (CCS)

MGC

IP - SIG
= SIP
= H.323
= ISUP/IP

Megaco

MG

SCN-SIG - CAS
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.
- Original CATV model: MG resides in set-top box
SDP: Session Description Protocol

- Used to describe sessions (to link the session with media tools)
  - end to end negotiation of session parameters before establishing consent for the session – both parties must agree to the session
  - Describes conference/session addresses and ports + other parameters needed by RTP, RTSP and other media tools
  - Is really not a protocol, rather a description language
- SDP was initially designed for Mbone. Mbone was/is a multicast overlay network on the Internet
- 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 a multicast group
• Mbone – part of Internet that supported 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.)
• Has its direct use as a service in corporate networks and as a service enabler in public networks.
Multicast vs point to point

• All Multicast protocols have to deal with the fact that there are many receivers for a message sent by the sender
  - it follows that achieving reliable delivery can not easily be based on acknowledgements
  - ack flooding would follow and the sender would be overwhelmed by them
  - There are specialized solutions for reliable multicast…

• In p-to-p, reliable delivery is easy to achieve by requiring acknowledgements either to each message or within a window of N (= for each N messages)
SDP can describe

- Session name and purpose
- Time(s) the session is active
  - start, stop time, repetition (relevant for conferences)
- 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= (originator 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 (cont’d)

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

Media description
- m= (media name and transport address) – IP address + port = 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)

NB: SDP itself allows using names instead of addresses of endpoints in ”o=” but not in ”c=”!!
\text{→ problem with NAT traversal!!!}

Some SDP documents:
- RFC 4566 SDP: Session Description Protocol
- RFC 2974 Session Announcement Protocol
- RFC 3264 An Offer/Answer Model with SDP
- RFC 4317 Session Description Protocol (SDP) Offer/Answer Examples
- RFC 4583 Session Description Protocol (SDP) Format for Binary Floor Control Protocol (BFCP) Streams
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<thead>
<tr>
<th>RFC</th>
<th>Title</th>
<th>Status</th>
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<tbody>
<tr>
<td>2326</td>
<td>Real Time Streaming Protocol (RTSP) (Proposed Standard)</td>
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<tr>
<td>4566</td>
<td>SDP: Session Description Protocol (Proposed Standard) updated by RFC3266</td>
<td></td>
</tr>
<tr>
<td>2974</td>
<td>Session Announcement Protocol (Experimental)</td>
<td></td>
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<tr>
<td>3108</td>
<td>Conventions for the use of SDP for ATM Bearer Connections (PS)</td>
<td></td>
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<tr>
<td>3259</td>
<td>A Message Bus for Local Coordination (Informational)</td>
<td></td>
</tr>
<tr>
<td>3264</td>
<td>An Offer/Answer Model with SDP (Proposed Standard) Updated by RFC6157</td>
<td></td>
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<tr>
<td>3524</td>
<td>Mapping of Media Streams to Resource Reservation Flows (Proposed Standard)</td>
<td></td>
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<tr>
<td>3605</td>
<td>Real Time Control Protocol (RTCP) attribute in SDP (PS)</td>
<td></td>
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<tr>
<td>3890</td>
<td>A Transport Independent Bandwidth Modifier for the SDP (PS)</td>
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<tr>
<td>4145</td>
<td>TCP-Based Media Transport in SDP (PS) Updated by RFC4572</td>
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<tr>
<td>4317</td>
<td>SDP Offer/Answer Examples (Informational)</td>
<td></td>
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<tr>
<td>4435</td>
<td>A Framework for the Usage of Internet Media Guides (IMGs) (Informational)</td>
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<tr>
<td>4473</td>
<td>Requirements for Internet Media Guides (IMGs) (Informational)</td>
<td></td>
</tr>
<tr>
<td>4566</td>
<td>SDP: Session Description Protocol (Proposed Standard)</td>
<td></td>
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<tr>
<td>4567</td>
<td>Key Management Extensions for SDP and Real Time Streaming Protocol (RTSP) (PS)</td>
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</tr>
<tr>
<td>4568</td>
<td>Session Description Protocol (SDP) Security Descriptions for Media Streams (PS)</td>
<td></td>
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<tr>
<td>4570</td>
<td>Session Description Protocol (SDP) Source Filters (PS)</td>
<td></td>
</tr>
<tr>
<td>4572</td>
<td>Connection-Oriented Media Transport over TLS Protocol in SDP (PS)</td>
<td></td>
</tr>
<tr>
<td>4574</td>
<td>The Session Description Protocol (SDP) Label Attribute (PS)</td>
<td></td>
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<tr>
<td>4583</td>
<td>SDP Format for Binary Floor Control Protocol (BFCP) Streams (PS)</td>
<td></td>
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music RFCs 04.01.2014 (2)

RFC 4796 The Session Description Protocol (SDP) Content Attribute (PS)
RFC 5027 Security Preconditions for SDP Media Streams (PS)
RFC 5245 Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translators (NAT) Traversal for Offer/Answer Protocols (PS) Updated by RFC6336
RFC 5432 Quality of Service (QoS) Mechanism Selection in SDP (PS)
RFC 5547 An SDP Offer/Answer Mechanism to Enable File Transfer (PS)
RFC 5576 Source-Specific Media Attributes in SDP (PS)
RFC 5583 Signaling Media Decoding Dependency in SDP (PS)
RFC 5888 SDP Grouping Framework (PS)
RFC 5898 Connectivity Preconditions for SDP Media Streams (PS)
RFC 5939 SDP Capability Negotiation (PS) Updated by RFC6871
RFC 5956 Forward Error Correction Grouping Semantics in SDP (PS)
RFC 6236 Negotiation of Generic Image Attributes in SDP (PS)
RFC 6336 IANA Registry for Interactive Connectivity Establishment (ICE) Options (PS)
RFC 6544 TCP Candidates with Interactive Connectivity Establishment (ICE) (PS)
RFC 6849 An Extension to SDP and Real-time Transport Protocol (RTP) for Media Loopback (PS)
RFC 6871 SDP Media Capabilities Negotiation (PS)
RFC 7006 Miscellaneous Capabilities Negotiation in SDP (PS)
Mmusic active drafts (4.1.2014)

Delayed Duplication Attribute in SDP (submitt for pub)
Duplication Grouping Semantics in SDP (submitt for pub)
Using Interactive Connectivity Establishment (ICE) with SDP offer/answer and SIP
Latching: Hosted NAT Traversal (HNT) for Media in Real-Time Communication
Cross Session Stream Identification in SDP
Real Time Streaming Protocol 2.0 (RTSP) (subm to pub)
SDP: Session Description Protocol WG Document Jun 2014
ICE: A Protocol for Network Address Translator (NAT) Traversal for Offer/Answer Protocols
A Network Address Translator (NAT) Traversal mechanism for media controlled by RTSP
The Evaluation of Different NAT Traversal Techniques for Media Controlled by RTSP
Stream Control Transmission Protocol (SCTP)-Based Media Transport in SDP
Multiplexing Negotiation Using SDP Port Numbers
SDP Extension For Setting Audio and Video Media Streams Over Circuit-Switched Bearers In PSTN (s-p)
Offer/Answer Considerations for G723 Annex A and G729 Annex B (s-p)
A Framework for SDP Attributes when Multiplexing
SDP 'trafficclass' Attribute
Trickle ICE: Incremental Provisioning of Candidates for ICE Protocol
UDP Transport Layer (UDPTL) over Datagram Transport Layer Security (DTLS)
Summary

• In 3GPP, the drive is towards providing PSTN like control over services and over what a user can do in the IP environment. This view believes in the importance of assuring QoS for user sessions. Conversely, if there is enough capacity (and low delay) for everyone, BE is all that is needed.

• Assuring low delay and proper trust handling → IMS

• IP telephony requires many supporting protocols.

• IETF development model is one protocol for one problem.

• Client-Server model is used whenever possible.

• The need for trust establishment in the mobile environment is higher than in the fixed environment: mobile based FW does not make sense → FW must be network based

• Through access to the Internet, the open Internet model lives on.