

[Contents](#)[Index](#)[Glossary](#)[Reference](#)

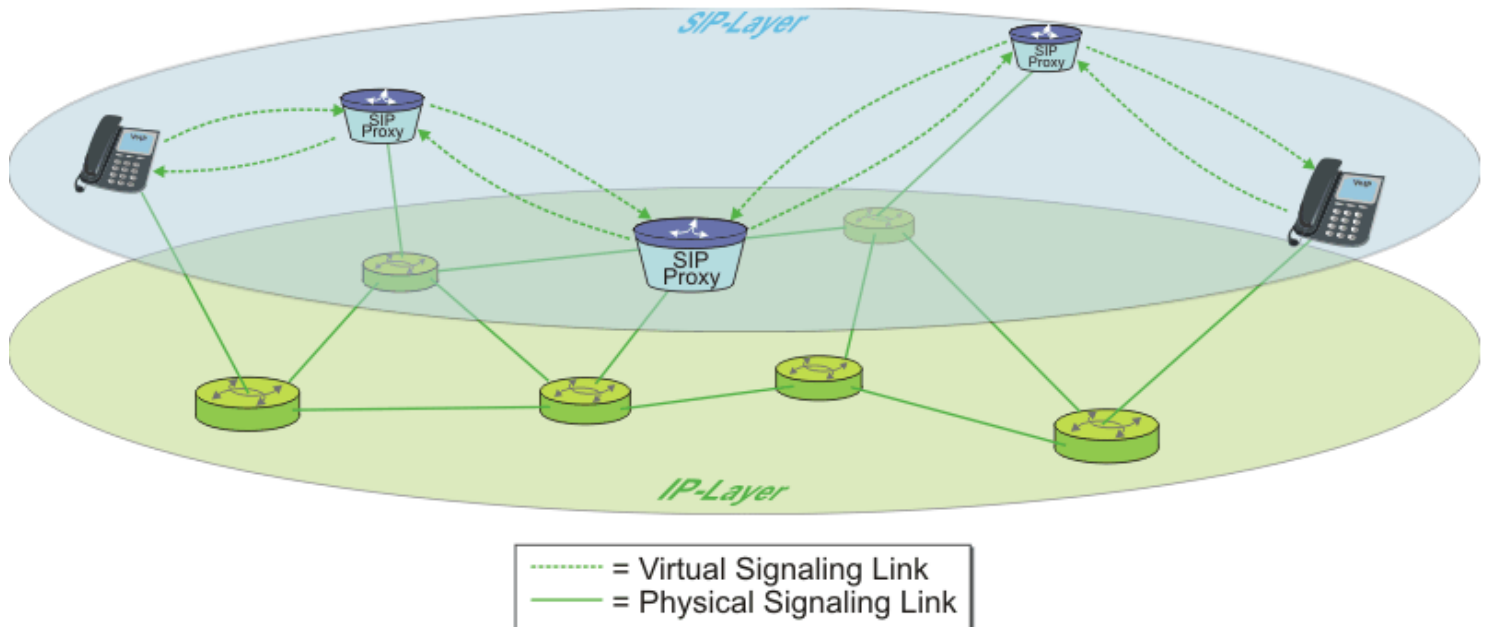
SIP, SDP and other NGN Protocols - Signaling & Protocol Analysis

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- ***Introducing the Playground of SIP / Reviewing SIP and SDP Basics***
- ***Detailed Consideration of Formal SIP-Protocol Aspects***
- ***Detailed Consideration of Formal SDP-Protocol Aspects***
- ***Advanced Use of SIP and SDP***
- ***SIP, SDP and DBP in 3GPP-Networks***

Philosophy of SIP-Operation

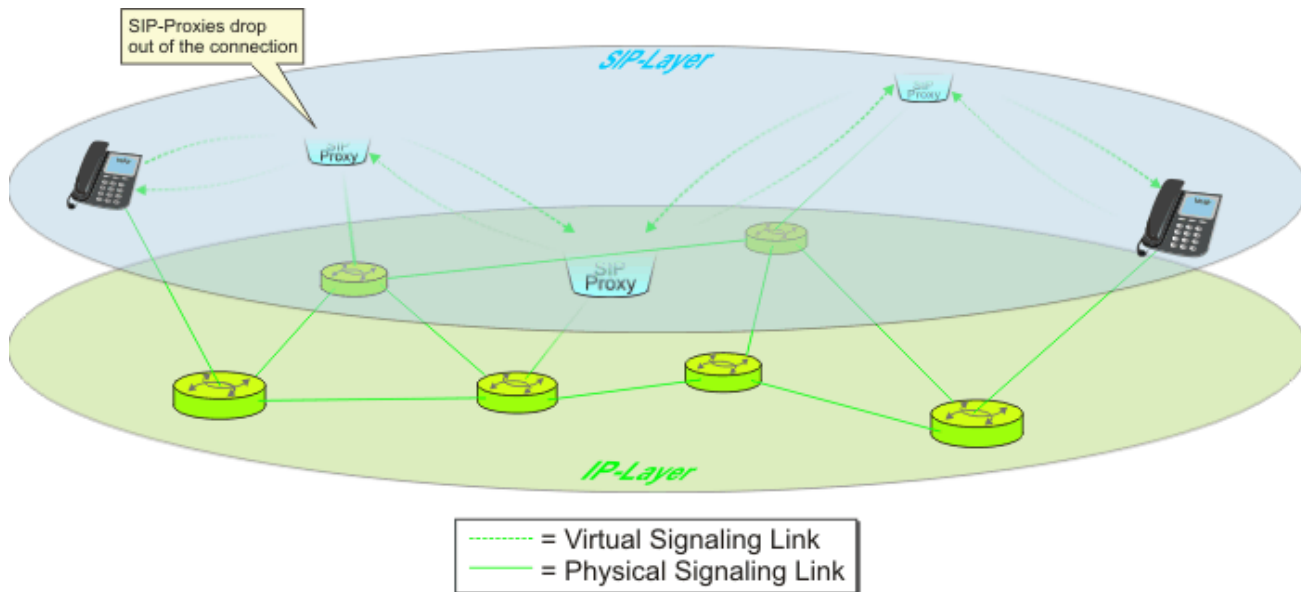
- Session Establishment Phase



Session Establishment Phase

- The figure illustrates it: During the session establishment phase, the two peers may require a number of SIP-proxy servers to route and handle session establishment requests. At this time, we don't want to be specific about the type of SIP-proxy.
- Note the two layers: The SIP-layer requires physical transport of the SIP-messages through the IP-layer and through the routers which are used there.

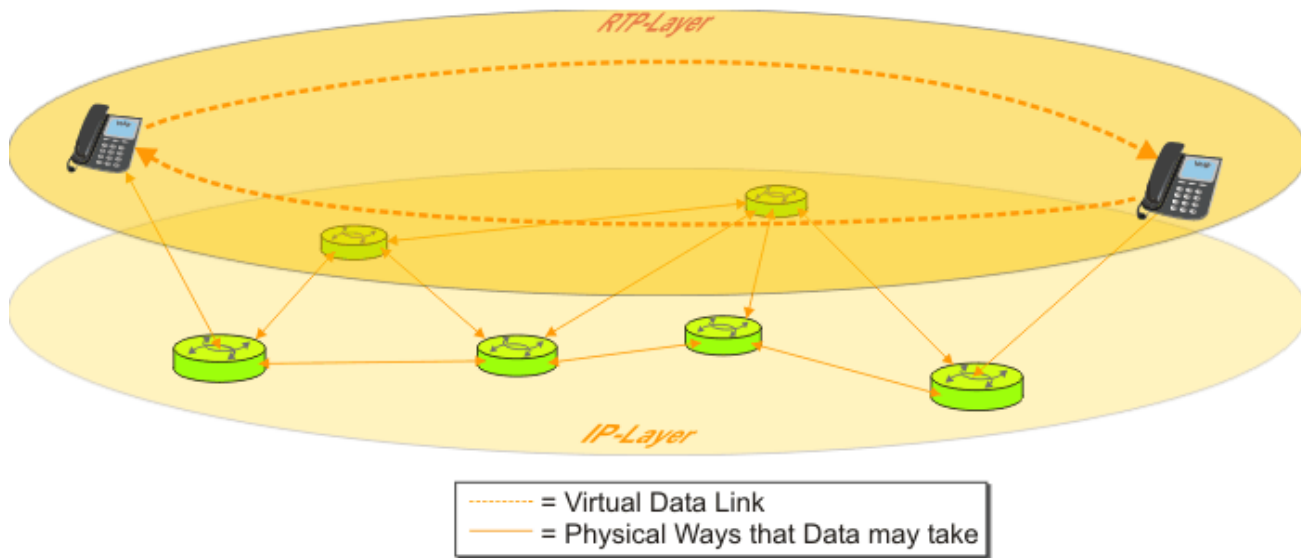
Session Completion Phase



- Note that the SIP-proxy servers drop out of the communication chain. This is the regular way of processing in SIP. That is, after the setup of the communication channel, there is no more involvement required of the proxies.
- This statement is, however, only true if standard conditions apply: No NAT, no media conversion needed, no IP-version Interworking between the two peers...)

Note that operators may also configure certain means to assure that the proxies remain in the signaling chain and are also receiving keep-alive messages periodically from one or both peers.

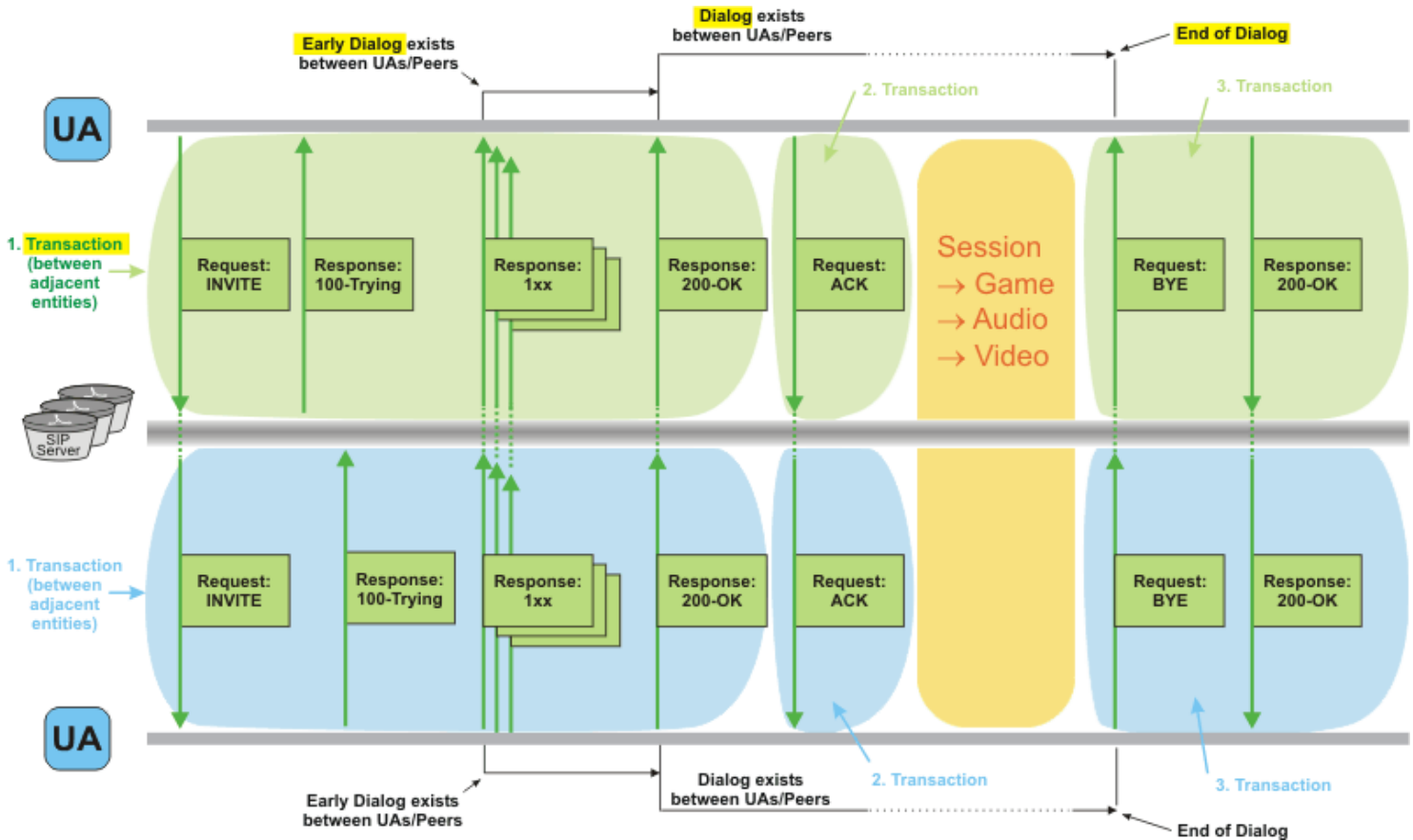
Session Active Phase



- While the session is active, the two peers exchange data, e.g. embedded into **RTP**- or **RTSP**-frames. These data frames are packed into **IP**-frames of which every single one can take a different route between the two peers.
- Note that there is no **SIP**-proxy in between.

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Important SIP-Terminology / Step 1: Two Users ...



We start with the limitation of only two users (no SIP-forking) to simplify things initially. In a second step we shall handle the case of multiple users).

Transaction)

Each SIP-transaction consists of a single request message which is sent by a UAC (User Agent Client) and the related final response message which is sent by the adjacent UAS (User Agent Server). If the request message is an INVITE, then there are zero or more provisional responses between the INVITE and the final response message. Note that the term “adjacent” in the previous sentence means that transactions ultimately exist between adjacent SIP-entities (e.g. UA and proxy) and not necessarily between peers (the two UA’s in the graphics) [RFC 3261 (p.24)].

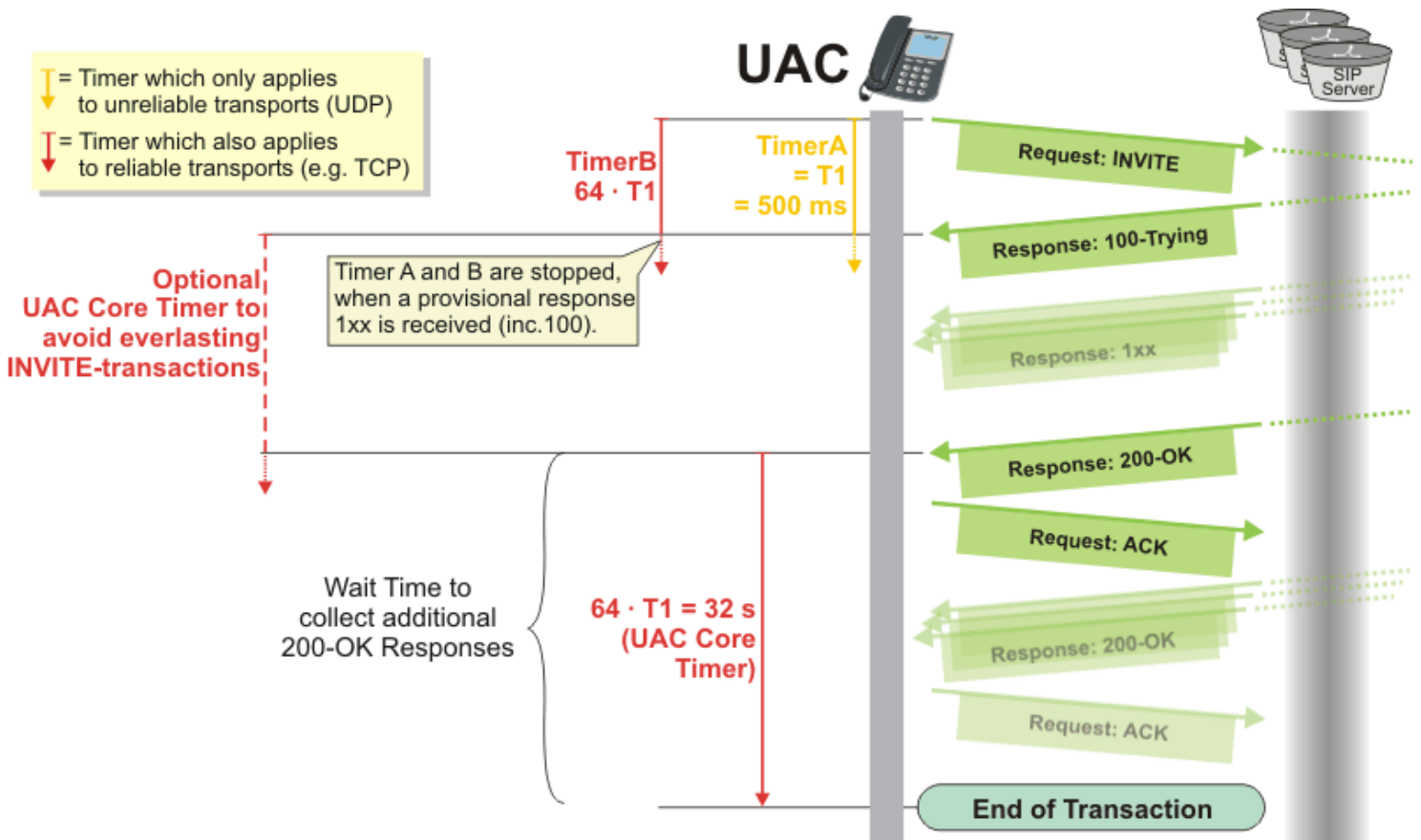
The exception to this rule is indicated in the figure: The successful dialog establishment through the initial INVITE-transaction shall be acknowledged by a Request: ACK-message which is considered to be a second transaction but which is not responded at all (although it is a SIP-Request). The Request: ACK really is a new transaction, considering the fact that a new branch-value is used. However, as we will see later, the transaction number (<--> CSeq) does not get incremented from the Request: INVITE that the Request: ACK relates to [RFC 3261 (p.24)].

Dialog / Call / Early Dialog (Definition)

- Dialog establishment is initiated when a UAC sends a Request: INVITE-message towards another peer, with this message possibly traversing one or more SIP-proxies.
- Dialog establishment can be triggered by Request: INVITE and (new with RFC 3515) also by Request: REFER.
- Dialogs exist between two UA’s / peers. There can be no dialogs between a UA and a SIP-proxy.
- A dialog has been established as soon as a UAS responds to a Request: INVITE with a non-failure final response message (<--> 200-OK). This rule means that the reception of a 2XX-response by a UAC establishes a dialog between these two users.
- An early dialog is there, if a UAS responds to a Request: INVITE with a provisional Response: “101 – 199” message (<--> which excludes “100” (Trying)) [RFC 3261 (12.1)]. The benefit of the definition of an “early dialog” is that the UAC may send further SIP-Requests (e.g. UPDATE) to the UAS already while the dialog is in its early state [RFC 3261 (13.2.2.1)].
- In SIP, a call consists of one or more dialogs [RFC 3261 (p.78)]. More than one dialog per call is only possible for multiparty calls.
- Dialogs are terminated by either party by sending a Request: BYE-message. Early dialogs can be terminated by the UAC by sending a Request: CANCEL-message

Each dialog is identified by the Call-ID-value which is initially allocated by the peer that sent the Request: INVITE-message and by the “To:”- and “From:”-tag values. We will get back to these identifiers in a few slides.

INVITE Transaction (UAC-Side - Response: 200-OK)



Overview

When the **UAC** issues the Request: INVITE-message, it will start timer A and timer B within its transaction handling sublayer.

Reception of Provisional Response

Upon reception of the first related provisional response message, timer A and timer B shall be stopped. Consequentially, any retransmissions of the Request: INVITE stop.

Also timer B shall be stopped when the first provisional response message is received. However, this may lead to everlasting transactions when the **UAS** does never send a final response message. To avoid this from happening, the initial Request: INVITE-message may be equipped with an "Expires:"-header field that indicates how long the INVITE will be valid. Implementation depending, the transaction user / **UAC**-core sublayer may operate another transaction timeout timer to be able to detect and delete dead INVITE-transactions.

Reception of Response: 200-OK

Upon reception of the first Response: 200-OK, the **UAC**-core (<--> one layer above the **UAC**) shall start a timer of $64 \times T1$ duration which allows the **UAC**-core to collect further incoming Response: 200-OK-messages. Each one needs to be responded to with a Request: **ACK**-message. [RFC 3261 (17.1.1), (p.82-83), Annex A]

10 - ??? Question Section ???

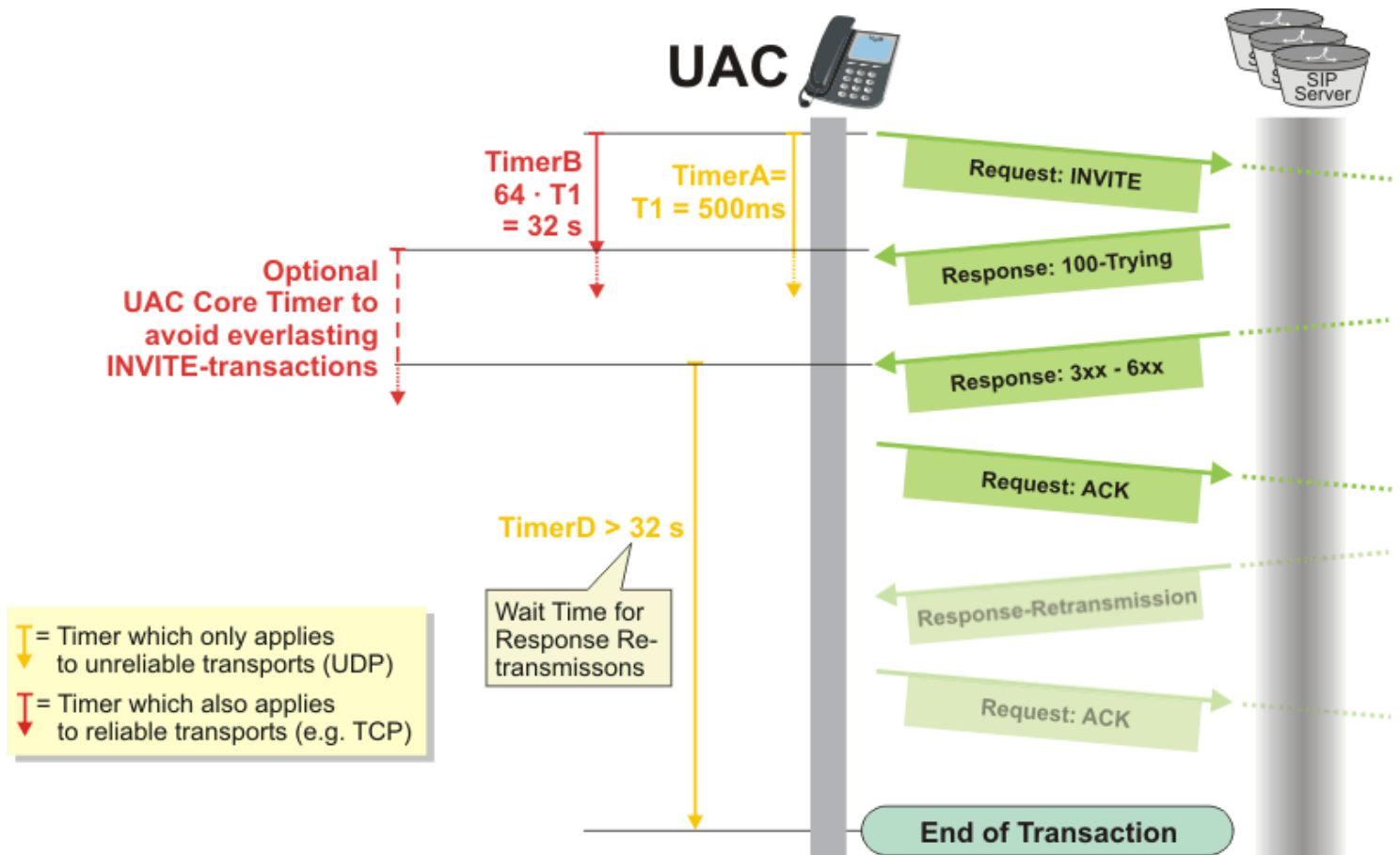
- When is a "To:"-tag included in the indicated Response: 100-Trying message?

Timer 1, Timer A and Timer B in case of 3GPP-Networks

Timer	Value to be used between P-CSCF and UE	Value to be used between SIP-nodes within the IMS
Timer 1 (T1)	2 s	0.5 s
Timer A	Initially T1	Initially T1
Timer B	128 s	32 s

[3GTS 24.229 (7.7)]

INVITE Transaction (UAC-Side - Response: 3XX – 6XX)



Overview

When the **UAC** issues the Request: INVITE-message, it will start timer A and timer B within its transaction handling sublayer.

Reception of Provisional Response

Upon reception of the first related provisional response message, timer A and timer B are stopped. Consequentially, any retransmissions of the Request: INVITE stop.

Also timer B shall be stopped when the first provisional response message is received. However, this may lead to everlasting transactions when the **UAS** does never send a final response message. To avoid this from happening, the initial Request: INVITE-message may be equipped with an "Expires:"-header field that indicates how long the INVITE will be valid. Implementation depending, the transaction management sublayer may operate another transaction timeout timer to be able to detect and delete dead INVITE-transactions.

Reception of Response: 3XX – 6XX

Upon reception of a failure response message (<--> Response: 3XX – 6XX), the **UAC** should start timer D within its transaction handling sublayer. Timer D is used in case of **UDP**-transport to allow the **UAC** to collect possibly incoming retransmissions of the failure response message.

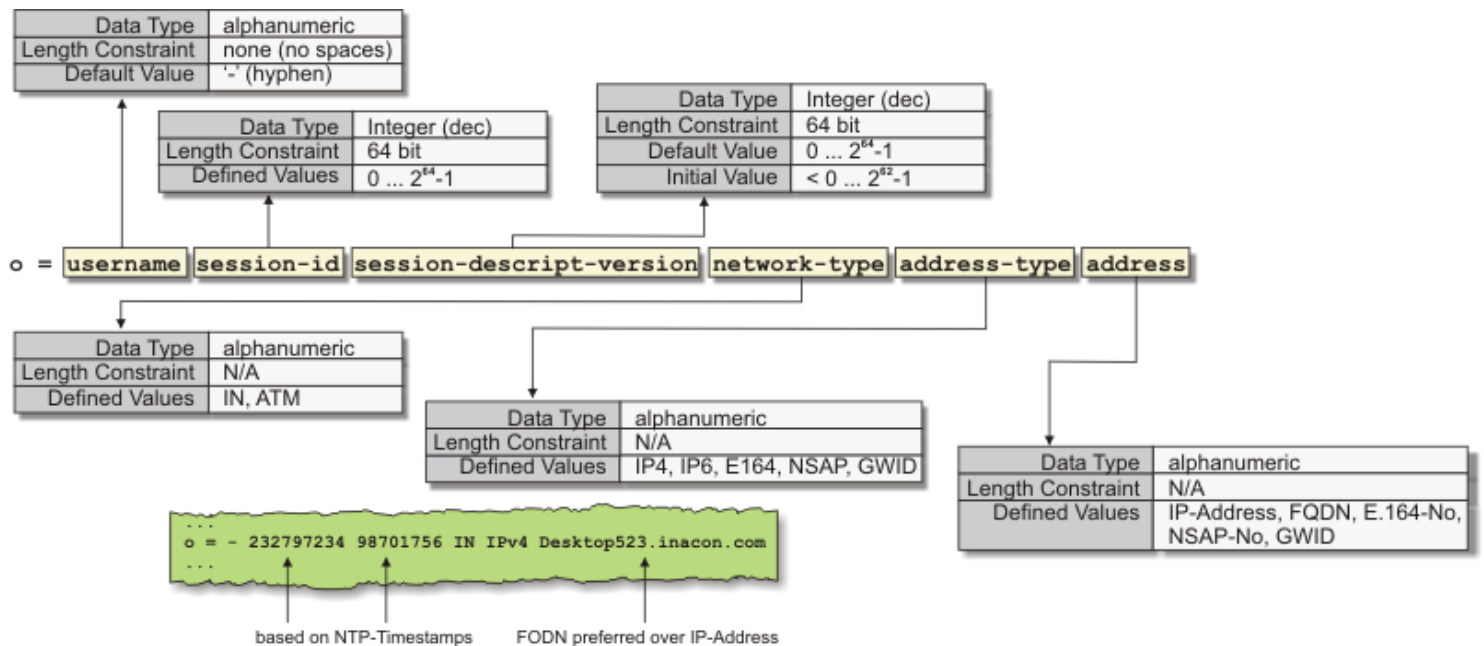
[RFC 3261 (17.1.1), Annex A]

11 - ??? Question Section ???

- Will the **UAC** on the left side be able to initiate another session / send another Request: INVITE while timer D is still running?
- Which entity selects the transport protocol (**UDP**, **TCP**) between two **SIP**-nodes?

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The "o=" -Line (Origin)



The o-line contains global session related information and information about the originator of the session. To be more specific:

Username

This parameter shall identify the user or the device that sends this session description. Still, usually there is no such information available and a simple hyphen ("-") is used instead [RFC 2327 (p.9)].

Session-ID

The session-ID shall be a 64 bit integer value, preferably generated according to the NTP-timestamp value of the sender's device. However, the use of NTP-timestamps is not mandated and therefore, any session-id which provides reasonable uniqueness is acceptable [RFC 2327 (p.9)].

Session-Description-Version

Similarly to the session-id, the session-description-version shall be a 64 bit integer value. However, its initial value shall be less than $2^{62}-1$ to avoid any overflows while a session is active. The reason is that either peer may during a session change session or media parameters and any new session description version shall be identified by the same session-id but an incremented session-description-version [RFC 2327 (p.9), RFC 3264 (p.5)]

Network-type

The only defined network type is "IN" for the internet [RFC 2327 (p.10)] and "ATM" for ATM-based networks [RFC3108].

Address-type

Defined address types are IPv4- and IPv6-addresses for network type "IN" and therefore the setting of this parameter is usually either "IP4" or "IP6". Note that RFC 2327 clearly recommends the use of FQDN's (e.g. device10.inacon.com) rather than plain IP-addresses and if plain IP-addresses are used then in should be globally unique (<--> not private) IP-addresses [RFC 2327 (p.10)]. In real life, the used address is rarely an FQDN and the IP-addresses are frequently private ones. As illustrated, ATM-networks use E164-numbers, NSAP (Network Service Access Point) and GWID (Gateway ID) instead [RFC 3108].

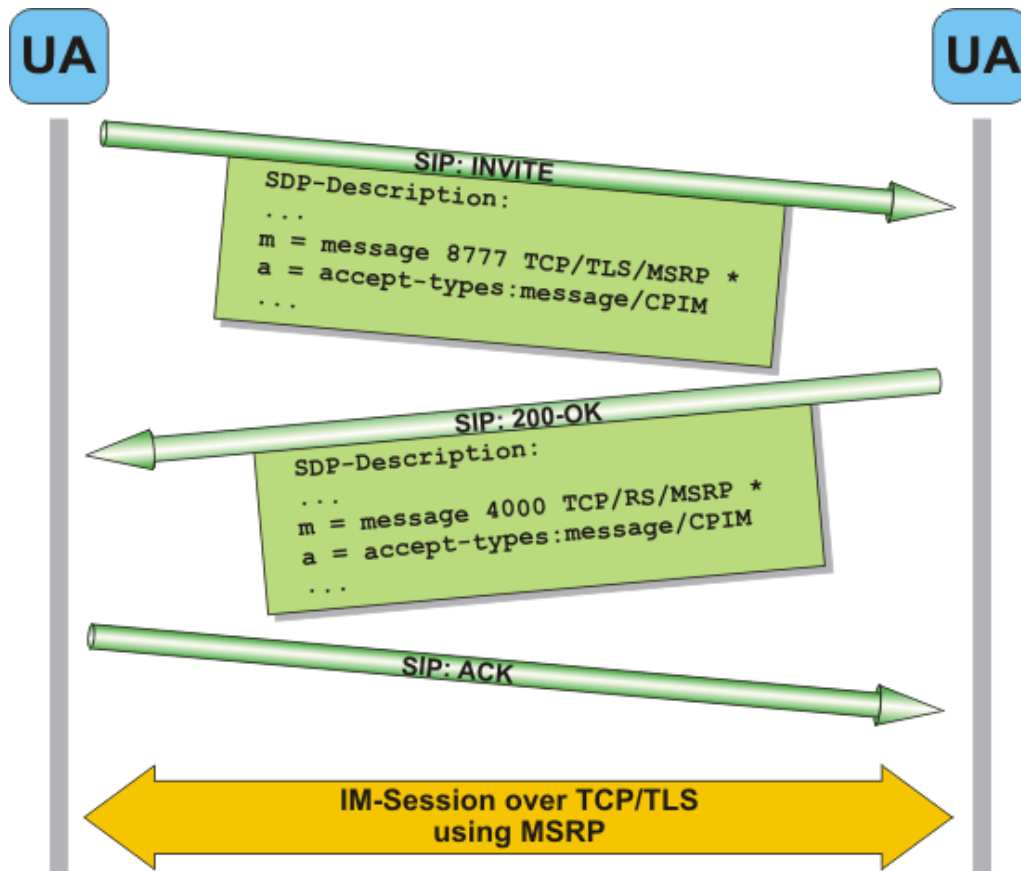
Address

This parameter includes the address of the originator of this session description. Note that this address is not necessarily identical to the connection address to which data shall be sent to (see next page).

The address will be an FQDN (e.g. device10.inacon.com), IPv4- or IPv6-address in case of IP-based networks or an NSAP, an E.164-telephone number or a GWID in case of ATM-based networks.

[draft-ietf-mmusic-sdp-new-26.txt (5.2), RFC 3108]

Media Type = message / Subtype = CPIM

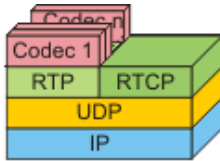


- The graphics illustrates the setup of an instant messaging session between two peers. Setup of a chatroom is different and described in draft-niemi-simple-chat-03.txt.
- Most important for our considerations is the new media type "message" and the use of the transport protocol "[TCP/TLS/MSRP](#)" which means, that [MSRP](#) (Message Session Relay Protocol) shall be used on top of a [TLS/TCP](#)-protocol stack.
- As can be seen from the following attribute line, the "acceptable [MIME](#)-types" (<--> accept-types) is set to "message/[CPIM](#)" which translates into the specific message format which has been defined in [RFC 3860](#) / [RFC 3862](#) for [CPIM](#) (Common Presence and Instant Messaging).
- The IM-session itself occurs independent from [SIP](#) and [SDP](#).

[<http://www.iana.org/assignments/media-types/>, draft-ietf-simple-message-sessions-14.txt]

(1) "m"-line / Details of the Transport Protocol Types

• RTP/AVP



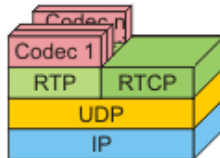
Applications:

Standard Voice and Video Calls, PoC, VoD, AoD

m-line (Example):

m = video 4500 RTP/AVP 31

• RTP/AVPF



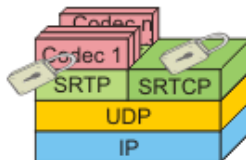
Applications:

Quality sensitive Voice and Video Calls or VoD, AoD sessions with the need for sophisticated RTCP-reporting.

m-line (Example):

m = audio 6888 RTP/AVPF 8 0 3

• RTP/SAVP



Applications:

Quality sensitive Voice and Video Calls or VoD, AoD sessions with the need for integrity protection and encryption.

m-line (Example):

m = video 45002 RTP/SAVP 34 31

RTP/AVP

The related protocol stack is most frequently used as it is well suited for real-time user applications like voice calls or video conferencing. Note that the protocol stack always contains two parts: one for [RTP](#) and the voice or video codec on top and another one for [RTCP](#).

[RTP/AVP](#) is also applicable in case of unidirectional VoD or AoD-sessions in which case a media stream only flows in one direction. The structure of the m-line is also indicated. The payload-type-list does contain one or more integer codec identifiers which are used within the payload type field of the related [RTP](#)-frames

[RFC 3551]

RTP/AVPF

The same applies which has been said about [RTP/AVP](#). The only difference between [RTP/AVP](#) and [RTP/AVPF](#) is the use of advanced "Feedback" reports on [RTCP](#)-level. These advanced feedback reports provide information about lost [RTP](#)-frames and their numbers or encoder specific feedback information about lost graphics frames (which is important for high quality video conferencing and VoD).

[draft-ietf-avt-rtcp-feedback-11.txt]

RTP/SAVP

The related protocol stack indicates the additional value of [RTP/SAVP](#) over [RTP/AVP](#): It uses [SRTP](#) and [SRTCP](#) to provide privacy in the user plane. In that respect, privacy relates to [RTP](#) / [RTCP](#)-data frame integrity protection, replay protection and encryption. The provision of the related keying material is out of the scope of [RTP/SAVP](#) and [SRTP](#).

[RFC 3711]

The “b”-Line (Bandwidth Information)

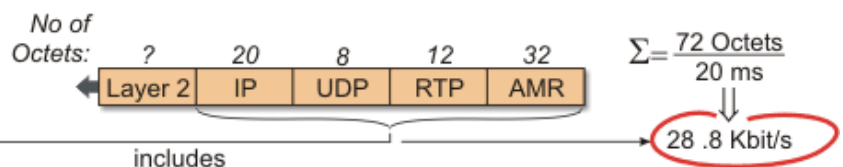
Data Type	alphanumeric
Length Constraint	N/A
Defined Values	AS, CT, TIAS, RR, RS

b = **modifier** : **value**

Data Type	Integer
Length Constraint	N/A
Unit	[Kbit/s] ↔ AS, CT [bit/s] ↔ RR, RS, TIAS

Example:

```
...
m = audio 5000 RTP/AVP 100
a = rtpmap:100 AMR/8000
b = AS:29
...
```



add 5% for RTCP (1.25% for Reports that are sent by this side and 3.75% for Reports that are received by this side)

The provision of the b-line is optional. The b-line may be present in the session description part and/or in the media description part. If it is present in both parts then the ones in the media description part only apply to that media stream.

CT (Conference Total)

This modifier is typically used within the session part of an SDP-description. CT provides a rough estimate of the bandwidth which is required for the entire session. The use of the CT-modifier is little helpful for resource reservation and QoS-related issues.

[RFC 2327 (p.14), draft-ietf-mmusic-sdp-new-26.txt (5.8)]

AS (Application Specific)

This modifier is typically used within the media description part of an SDP-description. AS provides information how much bandwidth is required for that media stream. In case of RTP-based data transfers (as defined through the transport protocol type) the related AS-bandwidth value takes into account the entire overhead caused by lower layer protocols down towards IP (but it does not consider any layer 1 or layer 2 overheads nor does it consider any compression algorithms [RFC 3550 (6.2 / p.24)]). The necessary RTCP-bandwidth for sending and receiving RTCP-reports is also not included in the AS-bandwidth value and is estimated to be an additional 5% of the AS-bandwidth value. Since 5% may be wrong, the RR- and RS-modifiers were defined.

The related AS-bandwidth value may be used by resource reservation protocols like RSVP in general and SM (Session Management) in case of 3GPP to understand how much bandwidth a media stream requires but the aforementioned constraints need to be taken into account.

This is illustrated in the example. Note that the bit rate of 28.8 kbit/s is rounded upwards to the next integer value of 29 kbit/s.

[RFC 2327 (p.14), RFC 3550 (6.2), draft-ietf-mmusic-sdp-new-26.txt (5.8), 3GTS 26.236 (Annex B), 3GTS 29.208 (7.2.2)]

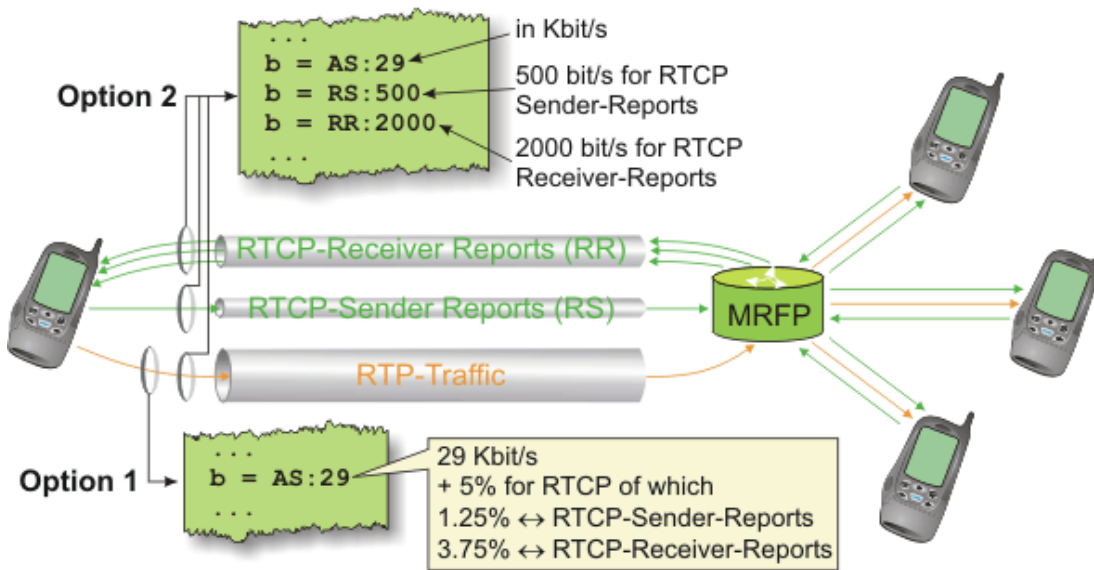
TIAS (Transport Independent Application Specific)

This modifier provides the same information as the AS-modifier but without consideration of any lower layer protocols. Therefore, the indicated value covers exclusively the bandwidth which is needed by a particular codec type. Neither RTP-headers nor bandwidth for RTCP-messages are considered. A bandwidth modifier TIAS is accompanied by an attribute maxprate (maximum packet rate) which provides the maximum number of packet per seconds for that media stream [RFC3890].

RR (Rtcp bandwidth for data Receiver) and RS (Rtcp bandwidth for data Sender)

Both are dealt with on the following pages [RFC3556].

Details of the Bandwidth Modifiers “RR” and RS”



- For each unidirectional RTP-session, the RTP-sender also generates SR's (Sender Reports) and the RTP-receiver generates RR's (Receiver Reports).

Note that if the session is bidirectional, then each peer generates only Sender Reports [RFC 3551]

- In case of multicast or conference sessions, a media stream sender will encounter situations in which the multiple media stream receivers each convey their own RTCP-receiver reports. Consequentially, there is an increased demand for RTCP-receiver bandwidth compared to the required RTCP-transmit bandwidth.
- This is a well-known and well-addressed issue which is already covered by the standard bandwidth modifier “AS”. Possible resource allocation mechanisms just add 5% to the AS-bandwidth for considering RTCP. Note that in this case, 1.25% are for RTCP-sender reports and 3.75% are for RTCP-receiver reports. We illustrate this possibility as option 1.
- However, this situation may not work well in all cases. There may for instance be very low bandwidth codecs on top in which case more bandwidth percentage than 5% is needed for RTCP or the opposite may be the case.
- This illustrates the requirements behind introducing the new bandwidth modifiers “RR” (Rtcp bandwidth for data Receiver) and “RS” (Rtcp bandwidth for data Sender).
- They allow for an explicit communication of the required RTCP-bandwidth values in each direction (note that the RS-bandwidth in such a case is actually added to the AS-bandwidth as it pertains to the same direction).
- In our graphics, option 2 illustrates the use of the RR- and RS-modifiers as amendments for the AS- or TIAS-modifiers. The AS-bandwidth is still 29 kbit/s but since specific RR- and RS-modifiers have been included, the 5%-rule no longer applies but the provided values of 500 bit/s and 2000 bit/s are applied.

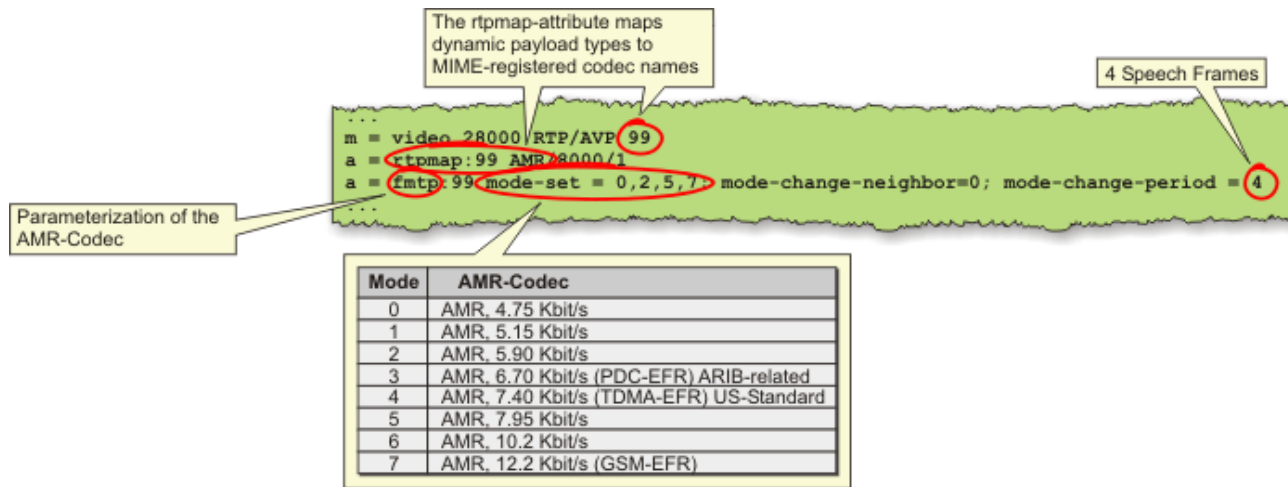
Note that the AS- and the CT-modifiers use kbit/s as unit while TIAS, RR and RS use bit/s.

[RFC 3556]

16 - ??? Question Section ???

- Taking these constraints about using RR and RS into consideration: How will a UA most likely indicate not to use RTCP at all in either or in both directions?

Example 2: AMR-Codec Definition and Parameterization

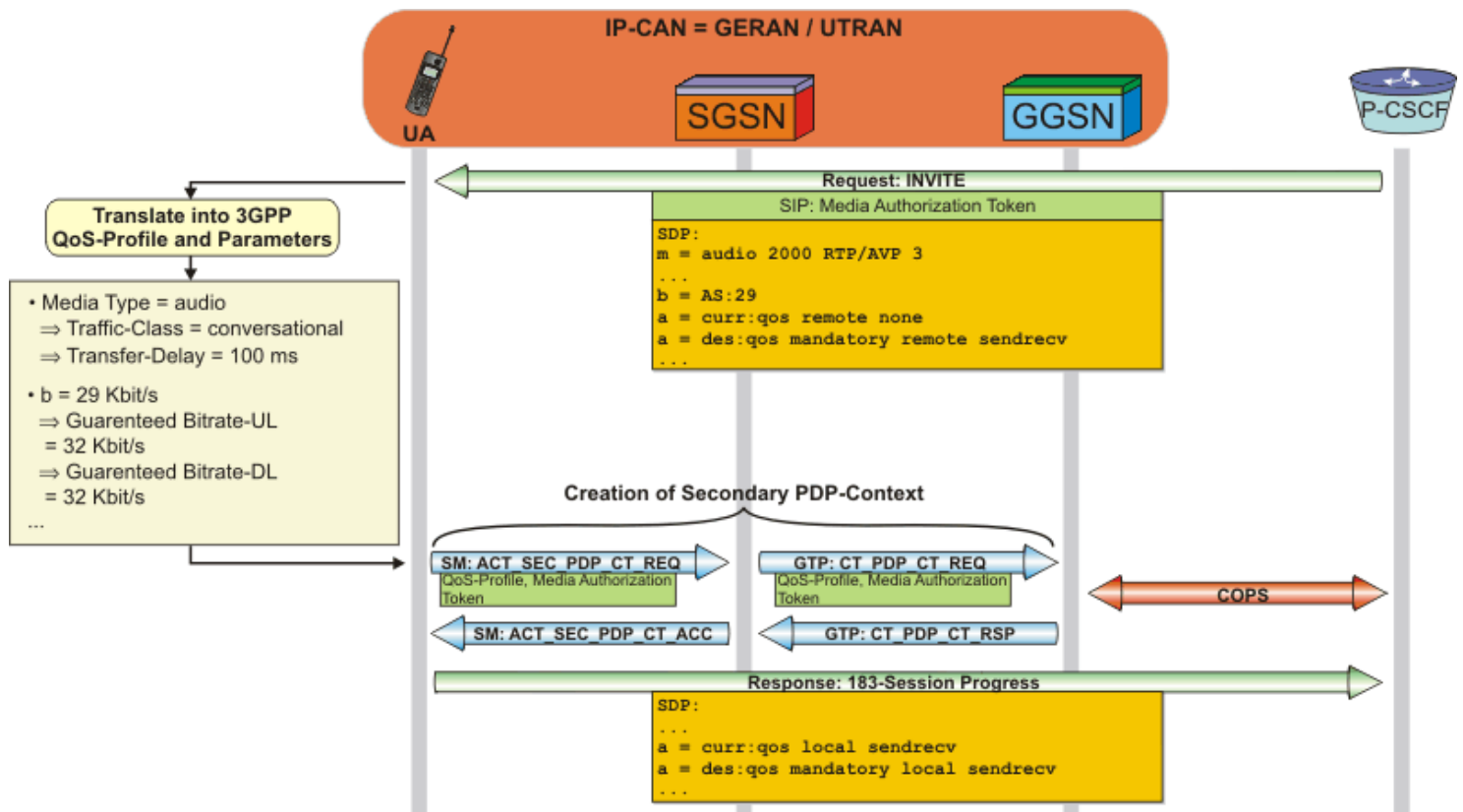


- The example illustrates how the rtpmap-attribute is used to map the dynamic payload number '99' to the MIME-registered codec type AMR. Of course, the rtpmap-parameter is not only limited to the AMR-codec.
- RFC 3267 (p.44) mandates the clock rate for AMR to be '8000' and we use it on one channel (<--> AMR/8000/1) which is the default and could have been omitted.
- In the next line, the fmtp-attribute defines all other AMR-specific attributes. In particular it limits the AMR-modes to 0, 2, 5 and 7. The table which AMR user data rates are reflected by which "mode-set" setting.
- The following parameter "mode-change-neighbor" indicates whether AMR-mode changes during the upcoming conversation are only allowed between adjacent modes (e.g. from 3 only to 2 or to 4) in which case "mode-change-neighbor = 1" or whether any mode change is allowed (e.g. from 5 to 1 or to 7) "mode-change-neighbor = 0".
- The final parameter "mode-change-period" indicates the minimum number of AMR-frames during which no mode change is allowed.

[draft-ietf-mmusic-sdp-new-26.txt (6), RFC 3267, 3GTS 26.101, 3GTS 26.235]

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Example 1: Resource Reservation if IP-CAN = GERAN/UTRAN



- The figure illustrates the case when the IP-CAN of the invited party is based on GERAN or UTRAN. In this case, the Request: INVITE will contain a media authorization token which is used by the UA / mobile station as "entry ticket" into real-time QoS. The media authorization token has previously been conveyed by the PEP (Policy Enforcement Point) to the PDF (policy decision function) upon request of the PDF. The PEP is usually part of the GGSN.
- Upon reception of the Request: INVITE, the UA needs to interpret the received SDP-parameters and translate them into IP-CAN-specific QoS-parameters which in this case means 3GPP-specific QoS-parameters.

Selection of 3GPP-specific QoS-Parameters

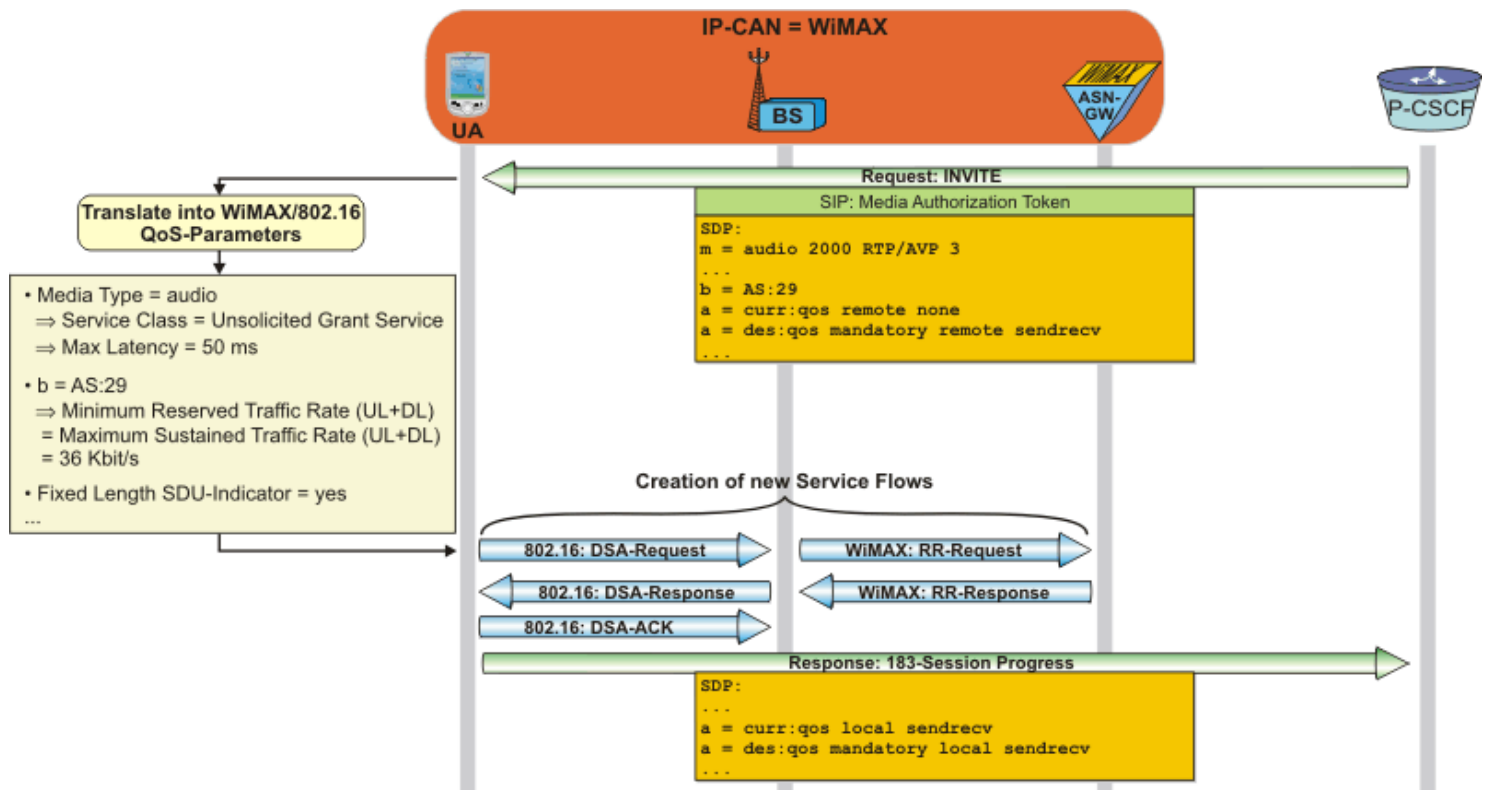
- Since the media type is audio or video rather than message, application or data the traffic class is selected with "Conversational Class". Alternatively, if the media stream would be marked as "sendonly" or "recvonly", the traffic class would have been selected with "Streaming Class".
- The transfer delay is requested with 100 ms.
- The SDP-parameter bandwidth "b=AS: 29" translates into "Guaranteed Bitrate" and "Maximum Bitrate" with app. 32 kbit/s (the additional bandwidth is there to considering RTCP-reporting).
- Of course there are many other QoS-parameters which are not illustrated here.

Activation of QoS-aware Media Tunnel

- Completely independent from SIP, the UA / mobile station then activates a secondary PDP-context through SM-signaling messages (Session Management). The UA / MS sends out an ACT_SEC_PDP_CT_REQ-message (Activate Secondary Packet Data Protocol Context Request) to the SGSN and upon successful PDP-context establishment it receives back an ACT_SEC_PDP_CT_RSP-message (Activate Secondary Packet Data Protocol Context Response).
- As illustrated, the UA / mobile station needs to include the media authorization token into the ACT_SEC_PDP_CT_REQ-message. After reception by the GGSN, the PEP (Policy Enforcement Point) which is physically part of the GGSN will check with the PDF (Policy Decision Function) which is part of the P-CSCF whether the requested resources have really been authorized and are necessary.
- For more details about PDP-context activation please refer to the INACON-book "GPRS – Signaling & Protocol Analysis (RAN & Mobile Station)".

[3GTS 23.060, 3GTS 24.008, 3GTS 29.208]

Example 2: Resource Reservation if IP-CAN = WIMAX



- This figure depicts another case in which the IP-CAN of the invited party is based on WIMAX / 802.16. In this case, the Request: INVITE may at some time in the future contain a media authorization token which can be used by the UA / subscriber station as “entry ticket” into real-time QoS.

Note that at least today, WIMAX has no means to convey a media authorization from the mobile station to the network.

- Upon reception of the Request: INVITE, the UA needs to interpret the received SDP-parameters and translate them into IP-CAN-specific QoS-parameters which in this case means WIMAX / 802.16-specific QoS-parameters.

Selection of WIMAX / 802.16-specific QoS-Parameters

- Since the media type is audio and the maximum latency is app. 50 ms, the service class can be selected with “Unsolicited Grant Service” which allows for the smallest latency.
- The SDP-parameter bandwidth “b=AS: 29” translates into “Minimum Reserved Traffic Rate” and “Maximum Sustained Traffic Rate” of 36 kbit/s. Note that the additional bandwidth is there to considering RTCP-reporting and because WIMAX QoS-parameters follow certain granularities.
- Of course there are many other QoS-parameters which are not illustrated here.

Activation of QoS-aware Media Tunnel

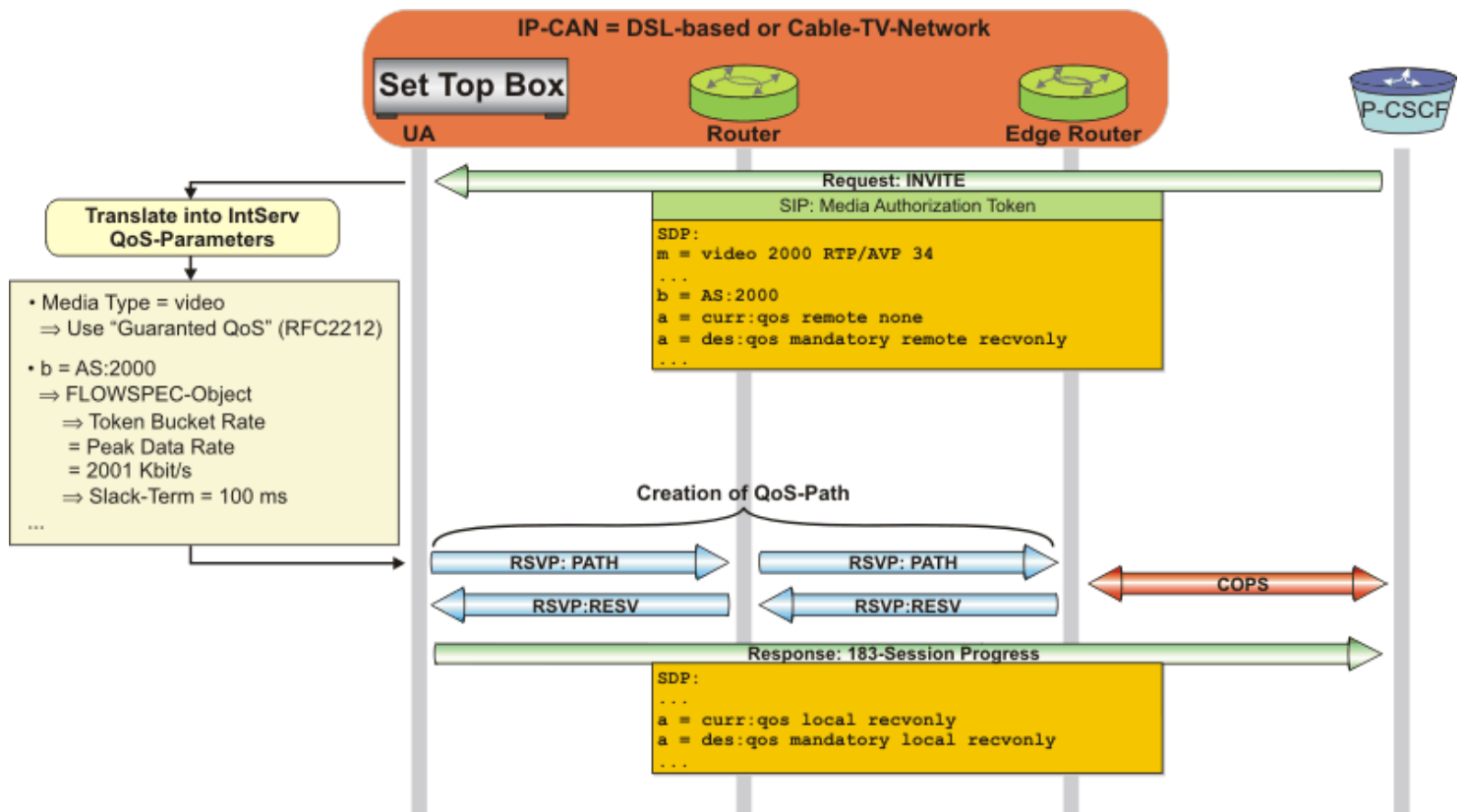
- Completely independent from SIP, the UA / mobile station establishes new “Service Flows” through 802.16-specific MAC-messages. The UA / MS sends out a DSA-REQ-message (Dynamic Service Addition) to the WIMAX base station which interprets and relays it towards a WIMAX ASN-Gateway (Access Service Network).
- It is the ASN-GW that finally approves the new service flows.

At least today, there are no means for the WIMAX mobile station to include the media authorization token. Neither does the WIMAX-forum support QoS-policing.

- For more details about DSA-procedure please refer to the INACON-book “WIMAX from A-Z”.

[IEEE 802.16-2005]

Example 3: Resource Reservation if IP-CAN = IntServ-aware



- The figure illustrates the case when the IP-CAN of the invited party is based on a DSL-access line or a Cable-TV network. Although several options exist in this case, we depict a case in which the UA (a set top box) uses IntServ to request a QoS-aware path within its access network to receive a movie from the backbone network (represented by the P-CSCF).
- As illustrated in case of IntServ-aware networks, the Request: INVITE will contain a media authorization token which is used by the UA / set top box as "entry ticket" into real-time QoS. The media authorization token has previously been conveyed by the PEP (Policy Enforcement Point) to the PDF (policy decision function) upon request of the PDF. The PEP is usually part of the edge router (see graphics). Unlike the IEEE and the WIMAX-forum, the IETF already integrated IntServ and Policy Control [RFC 2750, RFC 3521].
- Upon reception of the Request: INVITE, the UA / set top box needs to interpret the received SDP-parameters and translate them into IP-CAN-specific QoS-parameters which in this case means IntServ-specific QoS-parameters.

Selection of IntServ-specific QoS-Parameters

- In the IntServ-environment, the real-time QoS-requirement translates into the selection of "Guaranteed QoS" rather than "Controlled QoS" to cope with minimum delay requirements.
- This transfer delay is termed "Slack Term" in the IntServ-environment and it is selected with 100 ms (implementation specific).
- The SDP-parameter bandwidth "b=AS: 2000" translates into "Guaranteed Bitrate" and "Maximum Bitrate" with app. 2001 kbit/s (the additional 1 kbit/s bandwidth is there for RTCP-reporting).
- Of course there are many other QoS-parameters which are not illustrated here.

Activation of QoS-aware Media Tunnel

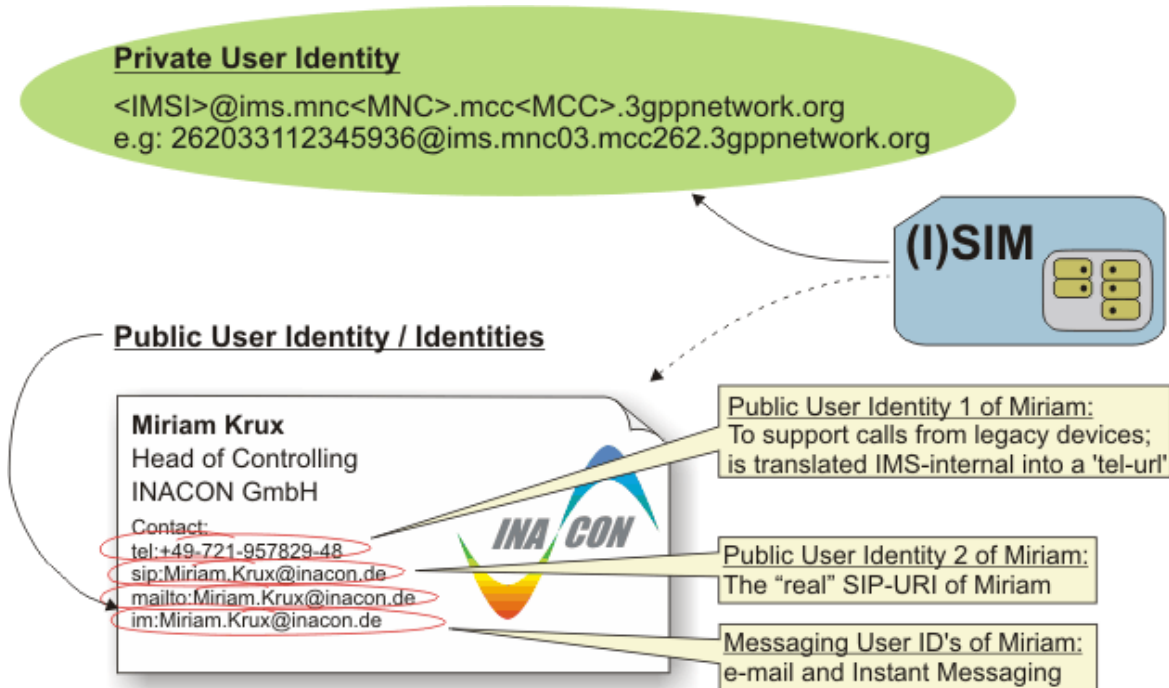
- Completely independent from SIP, the UA / mobile station establishes a unidirectional new "QoS-Path" through IntServ-specific RSVP-messages. The UA / MS sends out an RSVP: Path-message to its standard gateway which relays it internally towards other IntServ-aware routers. Alternatively, there could be a DiffServ-environment which uses DSCP's to indicate a certain QoS-requirement within each IP-frame
- For more details about IntServ and Policing please refer to the respective IETF-specifications RFC 2749 and RFC 3521.

[RFC 2749, RFC 2750, RFC 3084, RFC 3521, PacketCable PKT-SP-DQOS-I12-050812 "Dynamic Quality of Service Specification"]

- *Introducing the Playground of SIP / Reviewing SIP and SDP Basics*
- *Detailed Consideration of Formal SIP-Protocol Aspects*
- *Detailed Consideration of Formal SDP-Protocol Aspects*
- *Advanced Use of SIP and SDP*
- ***SIP, SDP and DBP in 3GPP-Networks***

IMS-related User Identities

- Private User Identity (IMPI) / Public User Identity (IMPU)



Overview / the ISIM

- Obviously, the [IMS](#) and [SIP](#) require new and different means for subscriber identification as legacy mobile telecommunication services. One example is the use of the [SIP-URI](#) in [SIP](#)-based networks for subscriber identification. Accordingly, [SIP-URI](#)'s need to be provided in [IMS](#)-based networks, too.

We are aware of the possibility to continue using legacy telephone numbers in the form of 'tel-url's but more advanced and easier to use are [SIP-URI](#)'s.

- Accordingly, [3GPP](#) has defined a new [SIM](#)-type, which is called the ISIM ([IMS](#)-Subscriber Identity Module). The availability of the ISIM is not mandatory for an [IMS](#)-subscriber but it is recommended. We will on the next slide line out the consequences of having or not having an ISIM in more detail.
- Most importantly, [3GPP](#) requires a subscriber to have two types of user identities, the private user identity and the public user identity.

[3GTS 23.003 (13.3, 13.4), 3GTS 23.228 (4.3.3),]

Private User Identity (IMPI)

The private user identity is stored in the [HSS](#) and on the ISIM (indirectly also on [SIM](#) and [USIM](#)). The private user identity unambiguously identifies a subscriber. The private user identity reuses the [IMSI](#) of a subscriber together with the [MCC](#) and [MNC](#) of the related mobile network operator to provide for a globally unique and routable identifier. More details follow on the next slide.

[3GTS 23.228 (4.3.3.1), RFC 2486 (Definition of NAI)]

Public User Identity (IMPU)

As the figure illustrates, a given subscriber may be using more than one public user identity. Public user identities are known to the subscriber and can be published on business cards. The public user identity can be compared with the [MS-ISDN](#) (Mobile Subscriber – Integrated Services Directory Number) from legacy [GSM](#)-networks. Public user identities take on the form of an [URI](#) and include messaging [URI](#)'s for IM and e-mail.

[3GTS 23.228 (4.3.3.2)]

22 - ??? Question Section ???

- How is it possible that Miriam has been allocated a [SIP-URI](#) that does not relate to the operator's host name (e.g. Vodafone.sip.co.uk) but to inacon.de?

Details of Private User Identities (IMPI)

- Private User Identities are based on the Network Access Identifier (NAI) as defined in RFC 2486

In 3GPP-networks, the private user identity is formatted as: <IMSI>@ims.mnc<MNC>.mcc<MCC>.3gppnetwork.org

- The private user identity is either stored within the memory of the ISIM or needs to be generated by the IMS-enabled UE/MS from the IMSI

Irrespective of whether there is an ISIM, the format of the private user identity will always be the same. On the network side, the HSS stores the private user identity (IMPI).

- Private User Identities (IMPI) can be compared with the IMSI in legacy mobile networks

Like the IMSI, the private user identity is never displayed to the subscriber. And like the IMSI, the IMPI cannot be altered by the subscriber.

Details of Public User Identities (IMPU)

- **Public User Identities are based on SIP-URI's or TEL-URL's**

As the figure on the previous slide illustrated, 'tel-url's are best suited to migrate your legacy phone number to the [IMS](#)-environment while [SIP-URI](#)'s represent human readable "telephone numbers".

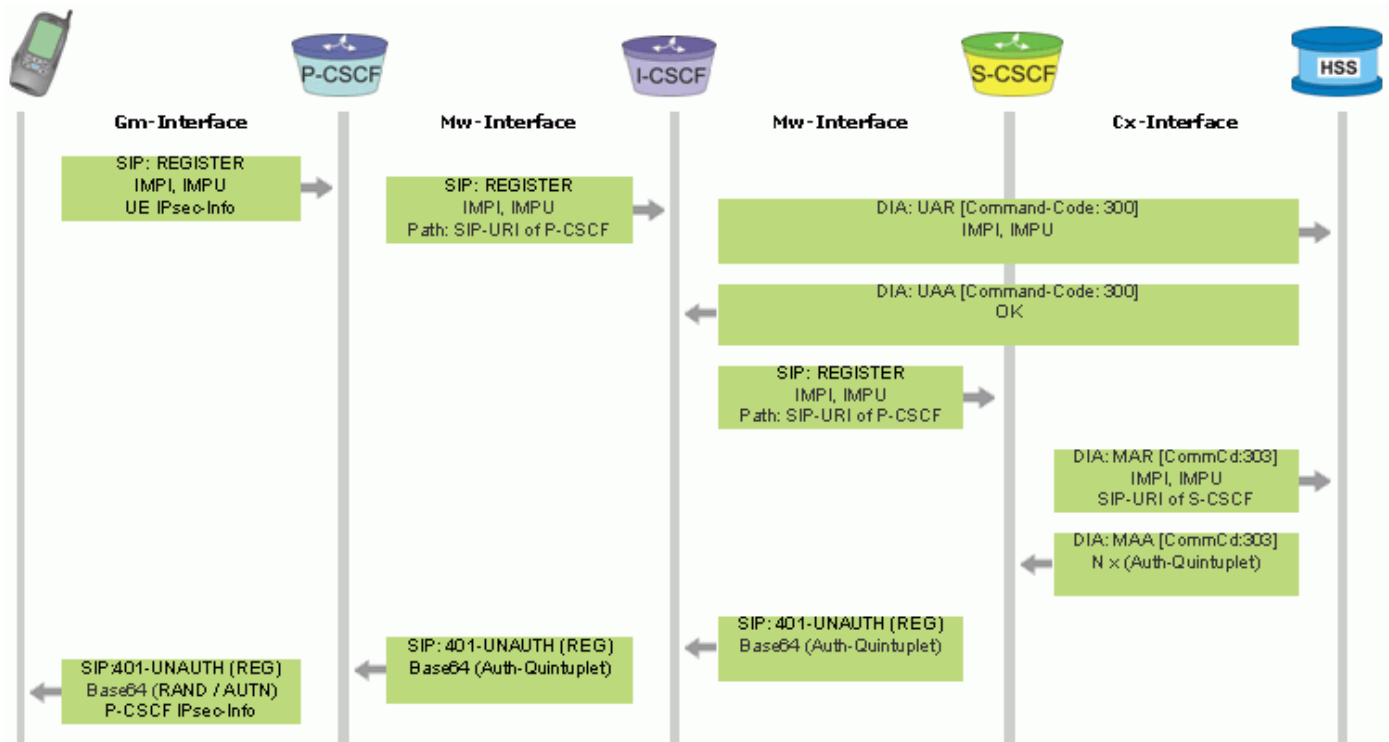
- **Public User Identities are particularly important for mobile terminating sessions**

Very simple: Everybody who wants to contact the [IMS](#)-enabled subscriber has to use a public user identity to specify the session destination.

- **If there is no ISIM, the MS/UE has to generate a temporary public user identity**

This public user identity is used during subscriber registration towards the [S-CSCF](#). More details will be provided on the next side.

(1) Registration to the IMS in 3GPP (Detailed Scenario)



Initial Conditions

The UE has been powered up and has just before activated its primary PDP-context and it has already discovered its P-CSCF.

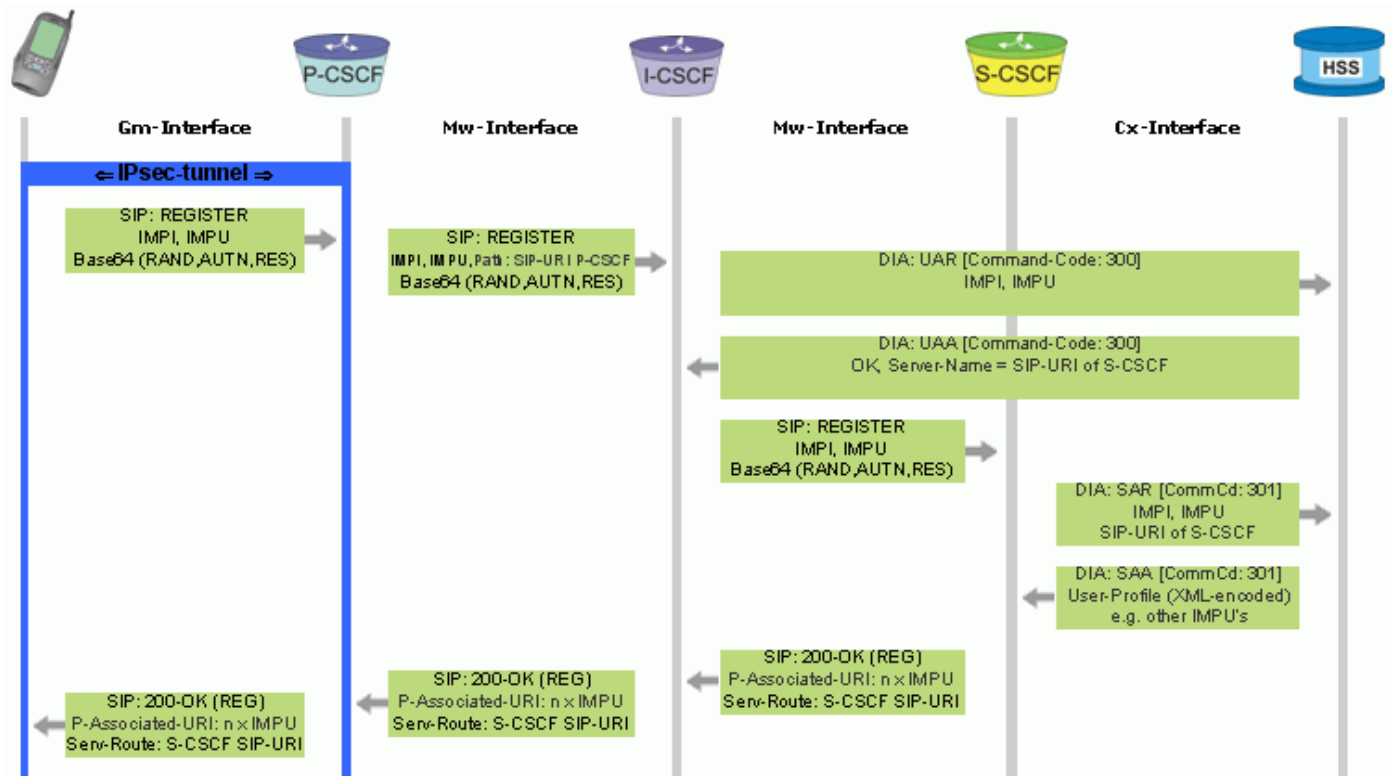
Applicability of this Procedure

The scenario is applicable to the initial registration of a UE with subscription of the UE and the P-CSCF to the registration event package.

Description

- The UE sends the Request: REGISTER-message to the P-CSCF which relays it towards an I-CSCF in the home-PLMN of the subscriber. This does obviously not preclude the P-CSCF and the I-CSCF to be in the same network.
- The I-CSCF uses the IMPI and IMPU of the subscriber for an initial authorization information retrieval towards the HSS. That is, the I-CSCF sends a DIAMETER: UAR-message (User Authorization Request) to the HSS to find out whether this subscriber may register in the first place.
- Let us assume that everything is OK and the HSS will reply with a DIAMETER: UAA-message (User Authorization Answer). Since this is the initial registration of the subscriber after power on, no "server-name" AVP will be included in this message. This "server-name" AVP is used to identify the S-CSCF of the subscriber.
- Since no S-CSCF is serving the subscriber yet, the I-CSCF will select an S-CSCF based on different considerations like load sharing, geographic issue or customer type (pre-paid / contract).
- In the next step, the I-CSCF will relay the Request: REGISTER-message to the selected S-CSCF.
- The S-CSCF uses the DIAMETER: MAR-message (Multimedia Authorization Request) to request authentication information for that subscriber ((identified through IMPI and IMPU) and to preset the HSS with its own S-CSCF Id (<--> SIP-URI of the S-CSCF).
- The HSS replies by sending back n authentication quintuplets (<--> RAND, SRES, CK, IK, AUTN) which will be used by the S-CSCF to authenticate the subscriber.
- The S-CSCF picks one entire quintuplet and relays it towards the I-CSCF. The relaying occurs through a Response: 401-Unauthorized message that terminates the initial SIP: Register-transaction.
- The I-CSCF relays this Response: 401-Unauthorized message towards the P-CSCF.
- The P-CSCF extracts all keys (IK, CK) and the authentication response (RES) from the authentication information and it adds its own IPsec-related information (SPI (32 bit) / new port number for SIP-signaling messages) to the Response: 401-Unauthorized message that it finally sends towards the UE.

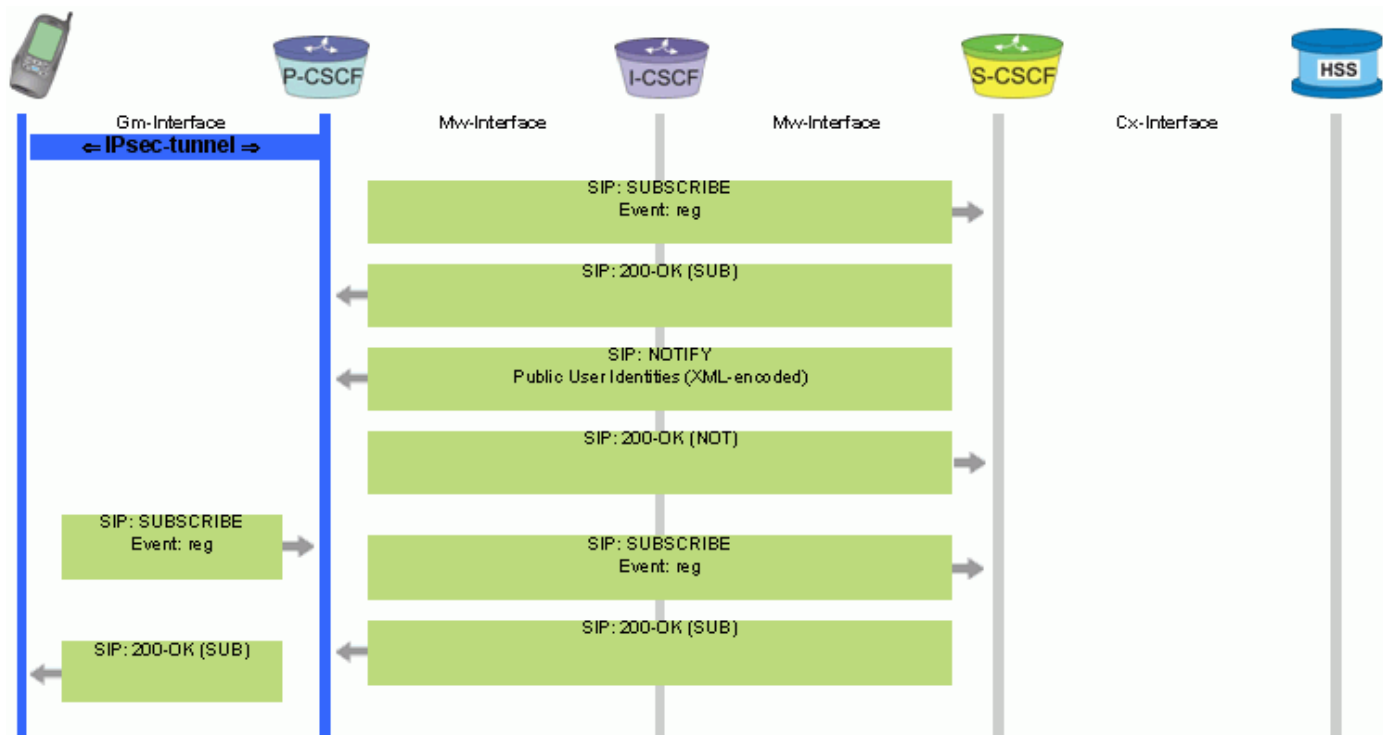
(2) Registration to the IMS in 3GPP (Detailed Scenario)



Additional Information

- The UE authenticates the network based on AUTN and calculates its own values for RES, CK and IK based on Ki and RAND. The calculated RES and the related RAND- and AUTN-values are embedded into a second Request: REGISTER-message which is then sent towards the P-CSCF.
- As the figure illustrates, this Request: REGISTER-message is integrity protected through IK and either HMAC-SHA-1-96 (RFC 2403) or HMAC-MD-5-96. The message is sent over the new IPsec-tunnel to the very port number that the P-CSCF previously conveyed to the UE.
- The P-CSCF will only accept initial Request: REGISTER-messages on the unprotected port number '5060'.
- The P-CSCF relays the Request: REGISTER-message towards an I-CSCF. Note that the P-CSCF will add a "Path:"-header field to make sure that the S-CSCF routes all future UE-terminating SIP-transactions through this P-CSCF.
- This I-CSCF has (again) to interrogate the HSS whether the subscriber is already registered to an S-CSCF and whether there are any registration restrictions. The interrogation is done as before through a DIAMETER: UAR-message (User Authorization Request) which contains the IMPI and IMPU of that subscriber.
- Different from the first UAR-message, the current answer message DIAMETER: UAA (User Authorization Answer) contains the SIP-URI of the S-CSCF that was previously selected by the I-CSCF to serve that customer (the HSS kept track since the MAR/MAA-messages while the I-CSCF keeps no registration states).
- Therefore, the I-CSCF is enabled to relay the Request: REGISTER-message towards that S-CSCF.
- The S-CSCF verifies that the RES and XRES-values match (<--> authentication successful) and then sends a DIAMETER: SAR-message (Server Assignment Request) to the HSS to finally register that S-CSCF as serving S-CSCF and to initiate the download of the XML-encoded user profile [3GTS 29.228 (Annex D and E)]. This user profile consists of the user's public user identities and other information. The HSS does so by sending a DIAMETER: SAA-message (Server Assignment Answer) to the S-CSCF.
- The S-CSCF generates the Response: 200-OK-message and inserts the IMPU's of the subscriber as received from the HSS. The S-CSCF also adds a "Service-Route:"-header field into the Response: 200-OK-message which includes its own SIP-URI (with port number) to be used from now on by the P-CSCF in the future for all SIP-transactions originated by the UE. That is, the "Service-Route:"-header field is used by the S-CSCF to indicate its own identity to the P-CSCF and to make sure that the P-CSCF will route all SIP-requests coming from that UE to that S-CSCF (if the P-CSCF is in another PLMN, then at least one I-CSCF will be between the P-CSCF and the S-CSCF during future transactions. If both are in the same PLMN then there will be no I-CSCF between them for all transactions except future registrations).
- The I-CSCF relays the Response: 200-OK-message to the P-CSCF which applies integrity protection upon it and relays it towards the UE.

(3) Registration to the IMS in 3GPP (Detailed Scenario)



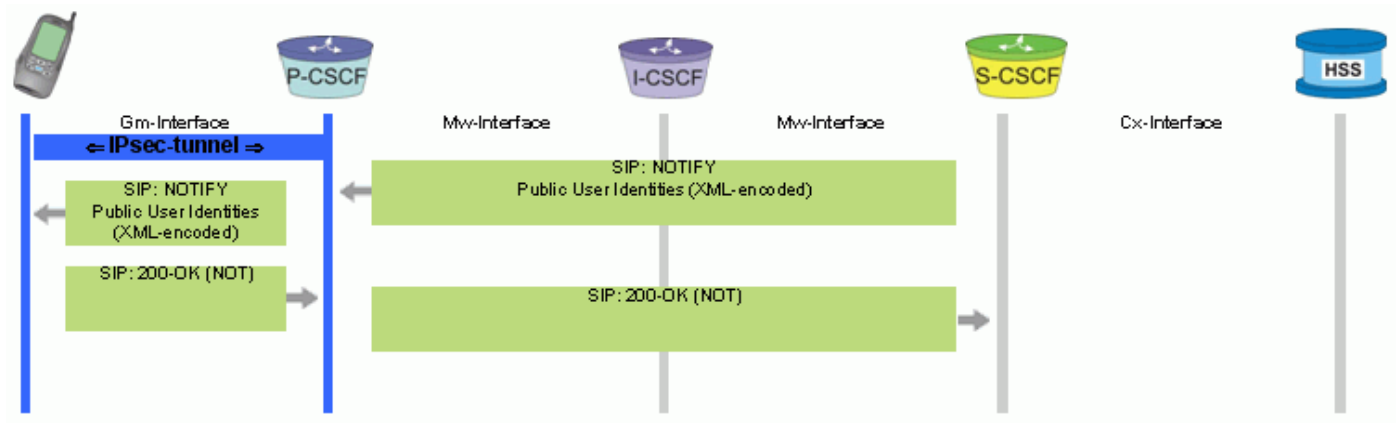
Additional Information

- The plain registration is over once that the **UE** receives the Response: 200-OK-message. Still, to allow for network initiated de-registrations and to keep the **UE** and the **P-CSCF** informed at all times whether or not the **UE** is registered (and with which public user identities), there is an immediate subscription of the **UE** and the **P-CSCF** to the registration event state package.

Note that this is an **IMS**-specific amendment to the genuine registration procedure as defined in [RFC 3261](#).

- For the registration event package, the **S-CSCF** takes on the role of the event server and the **P-CSCF** and the **UE** become the user agents that are notified once that a registration event state change occurs.
- As illustrated, the **P-CSCF** and the **UE** will both and independently issue a Request: SUBSCRIBE-message to the **S-CSCF** which identifies the "event: reg" as subscription event.
- The **S-CSCF** internally registers both subscriptions and sends a Request: NOTIFY-message to the **P-CSCF** to inform the **P-CSCF** about the current registration state of that subscriber.

(4) Registration to the IMS in 3GPP (Detailed Scenario)



Additional Information

- Finally, the S-CSCF sends a Request: NOTIFY-message to the UE to inform that UE about its current registration state.

In case of a network initiated de-registration, there will be another Request: NOTIFY-message telling the UE and the P-CSCF that the subscriber has been de-registered entirely or only one or more public user identities are de-registered.

Table of Contents

Introducing the Playground of SIP / Reviewing SIP and SDP Basics

What are the driving Forces behind the NGN-Hype?

- Easy Offering of Multimedia Services becomes possible
- Data and Voice Network Convergence
- Mobile and Fixed Network Convergence
- Service Convergence / Offering of Triple-Play Services

Next Generation Networks and their Components

- Typical Configuration and Interconnection of Next Generation Networks

- Network Type 1: Evolved ISP
 - Network Type 2: Former Telecom-Operator
 - Network Type 3: 3GPP Mobile Network Operator
 - Network Type 4: WIMAX Network Operator

- The IP Multimedia Subsystem (IMS)
- Overview and Internal Structure of the IMS
- The IMS in the 3GPP Environment

And where are SIP, SDP and all the other Protocols used?

- SIP Use within NGN
 - DIA (DIAMETER))
- H.248 / MEGACO
- RTP / RTSP / SRTP (Real-time Transport Protocol / Real-Time Streaming Protocol / Secure Real-time Transport Protocol)
- Interface specific View on Protocols (IMS-internal)
- Why SIP is used and not H.323 or other Alternatives?

Scope of SIP

- Session Establishment
- Clarification of the Term "Session"
- Session Modification
- Session Release
- Philosophy of SIP-Operation
 - Session Establishment Phase
- Comparison between SIP and HTTP

(1) Different Implementations of SIP

- Option 1: Amateur Use of SIP
- Option 2: Semi-Professional Use of SIP
- Option 3: Professional Use of SIP for VoIP (PSTN-Replacement)

Simple Example of a SIP-Scenario: VoIP Call Setup with SIP

- Overview
- Summary: Some SIP-Terminology
 - Message Types
 - SIP-Methods
 - Response Types
- Request: INVITE-Message
- Response: 100 (Trying)
- Response: 180 (Ringing)
- Response: 200 (OK)
- Request: ACK
- Example of one of the 4300 Speech Frames
- Request: BYE
- Response: 200 (OK)
- The Related Session Description Protocol (SDP) Contents

Detailed Consideration of Formal SIP-Protocol Aspects

SIP – Protocol Structure

Sublayers within SIP

- Transport Layer Control
- Transaction Handling (UAC and UAS)
- Transaction User / Core

SIP-Network Architecture

Overview

User Agents

- Dedicated VoIP-Phones
- Mobile Stations
- Softphones
- Set Top Boxes

SIP-Servers

- Stateless SIP-Proxy Server
- Stateful SIP-Proxy Server
- SBC (Session Border Controller), B2BUA (Back-to-Back User Agent)

Special SIP-Servers

- Application Layer Gateway (ALG)

Operation of Stateless SIP-Proxy Servers

- Advantages of Stateless SIP-Proxies
- Disadvantages of Stateless SIP-Proxies
- Other Assets of Stateless SIP-Proxies

Operation of Stateful SIP-Proxy Servers

Operation of Registrars

Operation of Redirect Servers

- Introduction
- Procedure Description

(1) Operation of Forking SIP-Proxy Servers (always stateful)

Operation of B2BUA and SBC

Operation of Event Servers

- Introduction
- Event Subscription
- Event Publishing and Notification
- The Presence Event

Soft Switches and their Controllers

Summary

Important SIP-Terminology / Step 1: Two Users ...

Transaction)

Dialog / Call / Early Dialog (Definition)

Session (Definition)

Dialog Identification (two Users / with or w/o Proxies)

Session Identification and Distinction

Transaction Identification (two UA's / no Proxies)

- The Branch Parameter
- Magic Cookie "z9hG4bK"
- Sequence Numbering (CSeq)

(1) Transaction Identification (two UA's / with Proxies)

- Transaction Identification through "branch" is done Hop-by-Hop
- Transaction Numbering through "CSeq:" applies end-to-end

Transaction-specific Messaging

- Option 1: Request = INVITE / Transaction = successful

Amendments in case of more than two Peers

Introducing Different Contact Addresses per User

Behavior of Forking Proxies

The Terms Call, Dialog, Session and Transaction in case of Forking

(1) Message and Parameter Details

Summary

SIP-Timers

Overview

INVITE Transaction (UAC-Side - Response: 200-OK)

- Overview
- Reception of Provisional Response
- Reception of Response: 200-OK
- Timer 1, Timer A and Timer B in case of 3GPP-Networks

INVITE Transaction (UAC-Side - Response: 3XX – 6XX)

- Overview

Reception of Provisional Response

Reception of Response: 3XX – 6XX

INVITE Transaction (UAC-Side - no Response received)

Overview

Expiry of Timer A

Expiry of Timer B

INVITE Transaction (UAS-Side sent 3XX – 6XX – wait for ACK)

Overview

Expiry of Timer G

Expiry of Timer H

Timer 1, Timer 2, Timer G and Timer H in case of 3GPP-Networks

INVITE Transaction (UAS-Side sent 200-OK – wait for ACK)

Overview

Expiry of UAS-Core Timer (T1)

Expiry of 2.UAS-Core Timer (64 x T1)

Timer 1 and Timer 2 in case of 3GPP-Networks

"None"-INVITE Transaction (UAC-Side - Response: 2XX – 6XX)

Overview

Timer E and Timer F

Timer K

Timer 1, Timer E, Timer F and Timer K in case of 3GPP-Networks

"None"-INVITE Transaction (UAC-Side - no Response)

Overview

Timer E

Expiry of Timer F

Timer 1, Timer 2, Timer E and Timer F in case of 3GPP-Networks

"None"-INVITE Transaction (UAS-Side - final Response sent)

Overview

Timer J

Timer 1 and Timer J in case of 3GPP-Networks

SIP-Message Format

General Information

Request Messages

Response Messages

SIP-Message Contents

The Request Line (Request Messages only)

(1) The Different Method-Types

Address Specification / Request-URI

The Status Line

The "From:" and the "To:" Header Fields

The "Call-ID:" and "Max-Forwards:" Header Fields

The "CSeq:" Header Field

The "Via:" Header Field

The "Contact:" Header Field

Detailed Consideration of Formal SDP-Protocol Aspects

The Session Description Protocol (SDP)

Structure of SDP-Parameters within a SIP-Message (Example)

Logfile Example: Session and Media Descriptors through SDP

Session Description Items

Overview

The "v=" -Line (Version)

The "s=" -Line (Session Name)

The "o=" -Line (Origin)

Username

Session-ID

Session-Description-Version

Network-type

Address-type

Address

The "c=" -Line (Connection Info)

Time Description Items

Overview

The "t=" -Line (Time)

Media Description Items

Overview

Presence of Attributes

The “m”-Line (Media Announcement)

Media Type (MIME)

Port-Number

Transport

Payload-Type-List

“m”-line Media Type Attribute (MIME) / some Examples

Media Type = application / Subtype = pdf

(1) “m”-line / Details of the Transport Protocol Types

RTP/AVP

RTP/AVPF

RTP/SAVP

TBCP (Talk Burst Control Protocol)

TCP (Transmission Control Protocol)

TCP/BFCP (TCP / Binary Floor Control Protocol)

TCP/MSRP (Message Session Relay Protocol)

TCP/RTP/AVPF

TCP/TLS/BFCP (Transport Layer Security / Binary Floor Control Protocol)

TCP/TLS/MSRP (Transport Layer Security / Message Session Relay Protocol)

UDP (User Datagram Protocol)

UDPTL (UDP Transport Layer)

Use Case Example: Floor Control during Conferencing (BFCP)

Configuration of the Participants and the Conference / Floor Control Server

The “b”-Line (Bandwidth Information)

CT (Conference Total)

AS (Application Specific)

TIAS (Transport Independent Application Specific)

RR (Rtcp bandwidth for data Receiver) and RS (Rtcp bandwidth for data Sender)

“a”-Lines (Attributes)

Example 1: Attribute “recvonly / sendonly”

The Offer / Answer Model

Session Identification Parameters at both Peers

Advanced Use of SIP and SDP

Reviewing the SIP-Scenario in Chapter 1: Critical Issues

Overview

How can the called party be found across networks?

How to tackle the loss of provisional response messages?

How is real-time QoS provided for the media streams?

Are there any means for secondary call treatment?

Question 1: How to find the called party across networks and how to route SIP-messages?

Answer:

SIP-clients need to register to “their” SIP-registrar to bind their current IP-address to their SIP-URI.

But how does a User Agent find “its” Registrar?

Option1: The “Preconfigured” Case

Option 2: User Agent performs DNS-query itself

Option 3: User Agent uses an intermediate SIP-proxy server to find the Registrar

DNS-Queries with NAPTR- and SRV-Records

NAPTR-Records

SRV-Records

Question 2: What happens if Provisional Responses get lost?

Provisional Response Messages are those with Status Code 100 – 199

Solution: Provide for the Option to acknowledge provisional Responses

Indicating Support or Requirement of acknowledged provisional Responses

Indicating Lacking Support for a Required Feature

Using PRACK to acknowledge Provisional Responses

Transaction Abort in Case of Lacking PRACK

Summary

Question 3: How to assure appropriate Resource Allocation in both Ways before alerting the Called Party?

Consequences of this Lack of Sophistication

Call Drop (No common Codec)

Answer: We define an additional Handshaking Procedure

Overview: Resource Management using SIP and SDP

Positive Outcome – Resource Reservation successful

Handling the Precondition Attributes “a = curr:” and “a = des:”

Overview

The new Option Tag “precondition”

The “m = ...” Line / Port Number and Payload Type

Interpretation of “local” and “remote” Direction-Tags

Interpretation of the “current-status” Attribute (“a = curr:”)

The “desired-status” and “confirm-status” Attributes

Preconditions fulfilled: the final Status

Example 1: Resource Reservation if IP-CAN = GERAN/UTRAN

Selection of 3GPP-specific QoS-Parameters

Activation of QoS-aware Media Tunnel

Example 2: Resource Reservation if IP-CAN = WIMAX

Selection of WIMAX / 802.16-specific QoS-Parameters

Activation of QoS-aware Media Tunnel

Example 3: Resource Reservation if IP-CAN = IntServ-aware

Selection of IntServ-specific QoS-Parameters

Activation of QoS-aware Media Tunnel

Summary

Question 4: Are there any Means for Secondary Call Treatment

Answer: Yes, there are means for

User Busy

Call Forwarding Unconditional

User not Responding

The “Do not Disturb” Feature

The “Find Me” Feature

(1) User Busy and “Do not Disturb” Feature – Detailed Message Sequence Chart

(1) “Call Forwarding Unconditional” / “User not Registered” – Detailed Message Sequence Chart

(1) User not Responding – Detailed Message Sequence Chart

(1) Find me / Follow me – Detailed Message Sequence Chart

Other important Questions

Is it possible to re-adjust an already established media stream setup?

How does a SIP-server detect that a call dropped? (Ungraceful Session Release)

Is there any means to compress SIP-messages since they are ASCII-encoded?

What do the abbreviations SIP-B, SIP-I and SIP-T stand for?

Media Stream Adjustments

Adding a Media Stream

Positive Response

Media Stream Modification

Removal of a Media Stream (Practical Exercise)

Ungraceful Session Release

Overview

Service Revocation

SIP-Devices may fail (Call Drop)

SIP is based on the approach to keep the call state only in the UA's (User Agents)

Options how to cope with the Problem

Option 1: Timer-based Session Release

Negative Outcome: Session Timer Expiry

(1) Summary

Introduction to SIP-I and SIP-T

SIP-Bridging

PSTN-Originating Session

SIP-Originating Session

SIP, SDP and DBP in 3GPP-Networks

Relationship between SIP, the IMS and 3GPP-Networks

Generic SIP-Servers vs. IMS-specific SIP-Servers

The Mobile's Way to SIP Registration and SIP-Sessions

Step 1: GPRS-Attachment

Step 2: Primary PDP-Context Activation Procedure / P-CSCF Discovery

Step 3: SIP-Registration (<--> REGISTER)

IMS-related User Identities

Overview / the ISIM

Private User Identity (IMPI)

Public User Identity (IMPU)

Details of Private User Identities (IMPI)

Details of Public User Identities (IMPU)

Use of Private and Public User Identities in REGISTER-Msgs.

Home Network Domain Name

Use of Private User Identity

Use of Public User Identity

Use of Temporary Public User Identity

Registration to the IMS in 3GPP-Networks (Overview)

Dependency between APN-Setting and P-CSCF-Selection

Subscriber registers to IMS while located in H-PLMN

Subscriber is Roaming

Authentication in 3GPP-based IMS

The Authentication Quintuplet of IMS-AKA

AK

AMF

AUTN

CK

IK

RAND

XRES

MAC

Authenticating the Network towards the MS/UE

The "base64"-Encoding Process

The IMS-AKA Authentication Process

Application of IPsec between MS/UE and P-CSCF

(1) Registration to the IMS in 3GPP (Detailed Scenario)

Initial Conditions

Applicability of this Procedure

Description

Mobile Originating Calls

Plain VoIP-Call

Bullet 1: Call Initiation

Bullet 2: QoS-Authorization

Bullet 3 and 4: I-CSCF in H-PLMN

Bullet 5: S-CSCF selects the Path

Bullet 6: Call reaches the called Party

Bullet 11: Response 183 (Session Progress) to Mobile Station

Bullet 12: Mobile Station establishes Secondary PDP-Context or modifies existing PDP-Context

Call towards the PSTN

Bullet 1: Call Initiation

Bullet 2: QoS-Authorization

Bullet 3: Tasks of the I-CSCF's

Bullet 4: S-CSCF in H-PLMN

Bullet 5: Task of BGCF

Bullet 6: Operation at the MGCF

Bullet 7: Response 183 (Session Progress) to Mobile Station

Bullet 8: Mobile Station establishes Secondary PDP-Context or modifies existing PDP-Context

Call from the PSTN

Bullet 1: Call Initiation

Bullet 2: Request Routing Information from the HSS

Bullet 3: Call is routed to the IMS

Bullet 4: H.248 Interaction with the Media Gateways

Bullet 5: Obtain detailed Routing Information from the S-CSCF

Bullet 6: Call is routed to P-CSCF

Bullet 7 and 8: Media Authorization / Relay Request: INVITE to MS/UE

Bullet 9: Mobile Station establishes Secondary PDP-Context

Solutions for the Practical Exercises

Responses to the Question Sections

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Glossary

Term	Explanation
16-QAM	16 symbols Quadrature Amplitude Modulation (<--> 3GTS 25.213)
2B1Q	Two Binary One Quaternary (<--> Line Coding used on the ISDN U-Interface)
3G ...	3rd Generation ...
3GPP	Third Generation Partnership Project (Collaboration between different standardization organizations (e.g. ARIB, ETSI) to define advanced mobile communications standards, responsible for UMTS)
3GPP2	Third Generation Partnership Project 2 (similar to 3GPP, but consisting of ANSI, TIA and EIA-41, responsible for cdma2000, EvDO and EVDV)
8-PSK	8 Symbol Phase Shift Keying
AA	Anonymous Access
AAA	Authentication, Authorization and Accounting
AAL-2	ATM Adaptation Layer 2 (for real-time services) (<--> ITU-T I.363.2)
AAL-5	ATM-Adaptation Layer 5 (non-real time) (<--> ITU-T I.363.5)
AAS	Adaptive Antenna Systems
A-Bit	Acknowledgement Request Bit (<--> used in LLC-protocol (Logical Link Control)
ABM	Asynchronous Balanced Mode
ABNF	Augmented Backus Naur Form (<--> RFC 2234)
ACC	Access Control Class (<--> 3GTS 22.011)
ACCH	Associated Control Channel (GSM / can be an SACCH or an FACCH)
ACK	Acknowledgement (<--> 3GTS 25.214)
ACS	Active Codec Set
ADM	Asynchronous Disconnected Mode
ADPCM	Adaptive Differential Pulse Code Modulation
AES	Advanced Encryption Standard / Cipher Key Lengths: 128 bit, 192 bit or 256 bit; AES is based on the Rijndael algorithm which is named after its two developers Joan Daemen and Vincent Rijmen
AESA	ATM End System Address
AG	Absolute Grant (<--> 3GTS 25.309)
AGCH	Access Grant Channel (GSM)
AH	Authentication Header (<--> RFC 2402)
AI	Acquisition Indicator
AICH	Acquisition Indicator Channel (UMTS Physical Channel)
AK	Authentication Key (<--> IEEE 802.16)
AK	Anonymity Key (<--> 3GTS 33.102)
AKA	Authentication and key agreement (<--> 3GTS 33.102)
ALCAP	Access Link Control Application Part (<--> ITU-T Q.2630.1 / Q.2630.2)
ALG	Application Layer Gaetway
AM	Acknowledged Mode operation (<--> e.g. in UMTS-RLC)
AM	Amplitude Modulation
AMC	Adaptive Modulation and Coding (<--> 3GTS 25.858)
AMD	Acknowledged Mode Data ((UMTS RLC PDU-type)
AMF	Authentication management field (<--> 3GTS 33.102)
AMI	Alternate Mark Inversion (<--> Line Coding)
AMPS	Advanced Mobile Phone System
AMR	Adaptive Multirate Encoding (<--> 3GTS 26.090)
ANSI	American National Standards Institute
AP	Access Point (<--> IEEE 802.11, 802.16)
AP	Access Preamble
AP-AICH	CPCH Access Preamble Acquisition Indicator Channel (<--> UMTS Physical Channel)
API	Access Preamble Acquisition Indicator
APN	Access Point Name (<--> Reference to a GGSN)
APP	A Posteriori Probability (<--> Turbo Decoding)
ARFCN	Absolute Radio Frequency Channel Number
ARIB	Association of Radio Industries and Businesses (Japanese)
ARP	Address Resolution Protocol (<--> RFC 826)
ARPU	Average Return Per User
ARQ	Automatic Repeat Request
AS	Application Server
AS	Access Stratum (<--> UMTS)
ASC	Access Service Class
ASCI	Advanced Speech Call Items (<--> GSM-R)
ASCII	American Standard Code for Information Interchange (<--> ANSI X3.4-1986)
ASIC	Application Specific Integrated Circuit

AS-ILCM	Application Server - Incoming Leg Control Model
ASN.1	Abstract Syntax Notation 1 (<--> ITU-T X.680 / X.681)
AS-OLCM	Application Server - Outgoing Leg Control Model
ATCA	Advanced Telecommunications Computing Architecture
AT-Command	Attention-Command
ATM	Asynchronous Transfer Mode (<--> ITU-T I.361)
AuC	Authentication Center
AUTN	Authentication Token (<--> 3GTS 33.102)
AV	Authentication Vector (<--> 3GTS 33.102)
B2BUA	Back-to-Back User Agent (<--> SIP term / RFC 3261, RFC 3725)
B8ZS	Bipolar with Eight-Zero Substitution (<--> Line Code used at the T1-Rate (1.544 Mbit/s))
BB	Base Band module
BC	Broadcast
BCC	Broadcast Call Control ((3GTS 44.069)
BCC	Base Station Color Code
BCCH	Broadcast Control Channel (UMTS Logical Cannel)
BCCH	Broadcast Control Channel (<--> GSM Logical Channel)
BCH	Broadcast Channel (UMTS Transport Channel)
BCTP	Bearer Control Tunneling Protocol ((ITU-T Q.1990)
BEC	Backward Error Correction
BEG	BEGin Message (<--> TCAP)
BER	Bit Error Rate
BFCP	Binary Floor Control Protocol ((draft-ietf-xcon-bfcp-05)
BFI	Bad Frame Indication
BG	Border Gateway
BGCF	Breakout Gateway Control Function
BIB	Backward Indicator Bit
BICC	Bearer Independent Call Control (<--> ITU-T Q.1902.1 – Q.1902.6)
BLER	Block Error Rate
BMC	Broadcast / Multicast Control (<--> 3GTS 25.324)
BM-IWF	Broadcast Multicast Interworking Function
BNF	Backus Naur Form (<--> RFC 2234)
BQA	Bluetooth Qualification Administer
BOB	Bluetooth Qualification Body
BQRB	Bluetooth Qualification Review Board
BQTF	Bluetooth Qualification Test Facility
BRAN	Broadband Radio Access Network
BS	Base Station (<--> IEEE 802.16)
BS_CV_MAX	Maximum Countdown Value to be used by the mobile station (<--> Countdown Procedure)
BS_EIRP	Base Station Effective Isotropic Radiated Power
BSC	Base Station Controller
BSIC	Base Station Identity Code
BSN	Block Sequence Number (<--> RLC) / Backward Sequence Number (<--> SS7)
BSS	Base Station Subsystem
BSSAP	Base Station Subsystem Application Part
BSSGP	Base Station System GPRS Protocol
BSSMAP	Base Station Subsystem Mobile Application Part ((3GTS 48.008)
BTAB	Bluetooth Technical Advisory Board
BTS	Base Transceiver Station
BVCI	BSSGP Virtual Connection Identifier
C/I	Carrier-to-Interference Ratio (<--> like SNR)
C/R-Bit	Command / Response Bit
C/T-Field	logical Channel / Transport channel identification Field
CAI	Channel Assignment Indicator
CAP	CAMEL Application Part (<--> CCS7)
CBC	Cipher Block Chaining (<--> DES-Operation Mode)
CBC	Cell Broadcast Center
CBCH	Cell Broadcast Channel (GSM)
CC	Call Control
CCC	CPCH Control Command
CCCH	Common Control Channel (UMTS Logical Channel)
CCCH	Common Control Channel (GSM Logical Channel)
CCH	Control Channel
CCITT	Comité Consultatif International Télégraphique et Téléphonique (International Telegraph and Telephone Consultative Committee)
CCM	Common Channel Management (Protocol Part on the GSM Abis-Interface / 3GTS 48.058)
CCM-Mode	Counter with CBC-MAC (<--> RFC 3610) Combined Authentication and Encryption with AES-Algorithm
CCN	Cell Change Notification (related to Network Assisted Cell Change / 3GTS 44.060)

CCPCH	Common Control Physical Channel (see also P-CCPCH and S-CCPCH)
CCS7	Common Channel Signaling System No. 7 (<--> ITU-T Q-series of specifications, in particular Q.700 – Q.703)
CCTrCH	Coded Composite Transport Channel (UMTS)
CCU	Channel Codec Unit
CD/CA-ICH	Collision Detection / Channel Assignment Indicator Channel (UMTS Physical Channel)
CDI	Collision Detection Indicator
CDMA	Code Division Multiple Access
CDR	Call Detail Record
CEPT	Conférence Européenne des Postes et Télécommunications
CESoP	Circuit Emulation Services over Packet
CFN	Connection Frame Number
CG	Charging Gateway
CGF	Charging Gateway Function
CGI	Cell Global Identification
CHAP	Challenge Handshake Authentication Protocol ((RFC 1334)
CIC	Circuit Identity Code (<--> ISUP)
CIC	Call Instance Code (<--> BICC)
CID	Channel Identity (<--> ATM)
CIDR	Classless Inter-Domain Routing (<--> RFC 1519)
CIF	Common Intermediate Format (352 x 240 pixels / (ITU-T H261 / H263)
CINR	Carrier to Interference and Noise Ratio
CIO	Cell Individual Offset (<--> 3GTS 25.331)
CK	Ciphering Key (<--> 3GTS 33.102)
CKSN	Ciphering Key Sequence Number
CMC	Codec Mode Command
CMI	Codec Mode Indication
CMR	Codec Mode Request
CMTS	Cable Modem Termination System
CN	Core Network
COFDM	Coded Orthogonal Frequency Division Multiplexing
CON	CONTinue Message (<--> TCAP)
COPS	Common Open Policy Service Protocol ((RFC 2748)
CPCH	Common Packet Channel (UMTS Transport Channel)(FDD only
CPCS	Common Part Convergence Sublayer
CPICH	Common Pilot Channel (UMTS Physical Channel / see also P-CPICH and S-CPICH)
CPIM	Common Presence and Instant Messaging ((RFC 3862)
CPS	Coding and Puncturing Scheme
CPU	Central Processing Unit
CQI	Channel Quality Indicator (<--> 3GTS 25.214)
CRNC	Controlling RNC
CS	Coding Scheme
C-SAP	Control Service Access Point
CSCF	Call Session Control Function (<--> SIP)
CSD	Circuit Switched Data
CSICH	CPCH Status Indicator Channel (UMTS Physical Channel)
CSMA-CA	Carrier-Sense Multiple Access – Collision Avoidance
CSPDN	Circuit Switched Public Data Network
CS-X	Coding Scheme (1 – 4)
CTCH	Common Traffic Channel (Logical) (PTM
CTFC	Calculated Transport Format Combination (<--> 3GTS 25.331)
CV	Countdown Value
CW	Code Word
cwnd	Congestion window
DARP	Downlink Advanced Receiver Performance (<--> 3GPP's adaptation of SAIC / 3GTS 45.015, 3GTS 24.008)
dBm	The unit dBm measures a power. The conversion of a power value from Watt [W] to dBm is done in the following way:X [dBm] = 10 x log10(X [W] / 0.001 [W])
DBP	Diameter Base Protocol ((RFC 3588)
DCCH	Dedicated Control Channel (UMTS Logical Channel)
DCH	Dedicated Channel (Transport)
DCM	Dedicated Channel Management (Protocol Part on the GSM Abis-Interface / 3GTS 48.058)
DCS	Digital Communication System
DDDS	Dynamic Delegation Discovery System ((RFC 3401 – RFC 3404)
DDI	Data Description Indicator (<--> 3GTS 25.309, 25.331)
DES	Data Encryption Standard
DHCP	Dynamic Host Configuration Protocol (<--> RFC 2131)
DIA	Diameter Protocol (<--> RFC 3588, RFC 3589)
Digit	4 bit
DL	Downlink

DLR	Destination Local Reference (<--> SCCP term)
DNS	Domain Name System
DOCSIS	Data Over Cable Service Interface Specification ((defined by CableLabs)
DPC	Destination Point Code
DPCCH	Dedicated Physical Control Channel (UMTS Physical Channel)
DPOCH	Dedicated Physical Channel (UMTS / Term to combine DPDCH and DPCCH)
DPDCH	Dedicated Physical Data Channel (UMTS Physical Channel)
DRNC	Drift Radio Network Controller
DRX	Discontinuous Reception
DS-CDMA	Direct Sequence Code Division Multiple Access
DSCH	Downlink Shared Channel (UMTS Transport Channel)
DSL	Digital Subscriber Line
DSLAM	Digital Subscriber Line Access Multiplexer
DSN	Digital Switching Network
DSS1	Digital Subscriber Signaling System No.1 (<--> also referred to as LAPD-signaling / ITU-T Q.931)
DTAP	Direct Transfer Application Part
DTCH	Dedicated Traffic Channel (UMTS Logical Channel)
DTM	Dual Transfer Mode (<--> 3GTS 43.055)
DTX	Discontinuous Transmission
DVB-H	Digital Video Broadcasting – Handheld
DVB-T	Digital Video Broadcasting – Terrestrial
E-AGCH	E-DCH Absolute Grant Channel (<--> 3GTS 25.211)
EAP	Extensible Authentication Protocol (<--> RFC 3748)
EAPOL	EAP encapsulation Over Lan or wlan ((IEEE 802.1X)
Ec/No	Received energy per chip / power density in the band
ECSD	Enhanced Circuit Switched Data (<--> HSCSD + EDGE)
E-DCH	Enhanced Uplink Dedicated Transport Channel (<--> 3GTS 25.211, 25.309)
EDGE	Enhanced Data Rates for Global Evolution
E-DPCCH	E-DCH Dedicated Physical Control Channel((3GTS 25.211)
E-DPDCH	E-DCH Dedicated Physical Data Channel((3GTS 25.211)
EDR	Enhanced Data Rate (<--> more speed with Bluetooth 2.0 (<--> 2.0 – 3.0 Mbit/s)
EFR	Enhanced Full Rate speech codec
EGPRS	Enhanced General Packet Radio Service
E-GSM	Extended GSM (GSM 900 in the Extended Band)
E-HICH	E-DCH HARQ Acknowledgement Indicator Channel (<--> 3GTS 25.211)
EIA	Electronic Industries Alliance (US-organization to support US industry)
EIR	Equipment Identity Register
EIRENE	European Integrated Railway Radio Enhanced Network (<--> GSM-R)
eMLPP	enhanced Multi-Level Precedence and Pre-emption (<--> 3GTS 23.067)
END	END Message (<--> TCAP)
ENUM	E.164-telephone number to URI (Uniform Resource Identifier) translation (<--> RFC 3761)
E-RGCH	E-DCH Relative Grant Channel (<--> 3GTS 25.211)
E-RNTI	E-DCH Radio Network Temporary Identifier (<--> 3GTS 25.401)
ESN	Electronic Serial Number (North American Market)
ESP	Encapsulating Security Payload (<--> RFC 2406)
E-TFC	E-DCH Transport Format Combination (<--> 3GTS 25.309)
Ethernet	Layer 2 Protocol for IP (<--> IEEE 802.3)
ETSI	European Telecommunications Standard Institute
EV-DO	Evolution Data Only or Evolution Data Optimized ((cdma2000)
EV-DV	Evolution Data/Voice (<--> cdma2000)
EVM	Error Vector Magnitude
FA	Foreign Agent (<--> Mobile IP / RFC 3344)
FACCH	Fast Associated Control Channel (GSM)
FACH	Forward Access Channel (UMTS Transport Channel)
FBI	Feedback Information (UMTS
FBI	Final Block Indicator
FCC	Federal Communications Commission
FCCH	Frequency Correction Channel (GSM)
FCS	Frame Check Sequence (CRC-Check)
FDD	Frequency Division Duplex
FDDI	Fiber Distributed Data Interconnect (optical Layer 2)
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FER	Frame Error Rate
FFH	Fast Frequency Hopping
FFT	Fast Fourier Transformation
FH-CDMA	Frequency Hopping Code Division Multiple Access
FIB	Forward Indicator Bit

FIPS	Federal Information Processing Standard
FISU	Fill In Signal Unit
FMC	Fixed Mobile Convergence
FN	Frame Number
FPB	First Partial Bitmap
FQDN	Fully Qualified Domain NameFully qualified domain names consist of a host and a domain name whereas the domain name needs to include a top-level domain (e.g. "de" or "org"). Examples: "www.inaon.de" and "PC10.inaon.com" are fully qualified domain names. ("www" and "PC10" represent the host, ("inaon" is the second-level domain,("de" and "com" are the top level domain.
FR	Fullrate or Frame Relay
FRMR	Frame Reject
FSN	Forward Sequence Number
FTP	File Transfer Protocol ((RFC 959)
GCC	Generic Call Control
GCF	General Certification Forum
GEA	GPRS Encryption Algorithm
GERAN	GSM EDGE Radio Access Network
GGSN	Gateway GPRS Support Node
GIF	Graphics Interchange Format
GK	Gatekeeper
GMM	GPRS Mobility Management
G-MSC	Gateway MSC
GMSC-S	Gateway MSC Server
GMSK	Gaussian Minimum Shift Keying
G-PDU	T-PDU + GTP-Header
GPRS	General Packet Radio Service
GPRS-CSI	GPRS CAMEL Subscription Information
GPRS-SSF	GPRS Service Switching Function (<--> CAMEL)
GPS	Global Positioning System
GRA	GERAN Registration Area
GSM	Global System for Mobile Communication
GSM-R	GSM for Railways
GSN	GPRS Support Node
GTP	GPRS Tunneling Protocol (<--> 3GTS 29.060)
GTP-C	GTP Control Plane
GTP-U	GTP User Plane
GTT	Global Text Telephony (<--> 3GTS 23.226)
GTTT	GPRS Transparent Transport Protocol (<--> 3GTS 44.018)
HA	Home Agent (<--> Mobile IP / RFC 3344)
HARQ	Hybrid ARQ (<--> 3GTS 25.212)
HCS	Hierarchical Cell Structure
HDB3	High Density Bipolar Three (<--> Line Coding used for E1 (PCM 30))
HDLC	High level Data Link Control
HFC-Network	Hybrid Fiber- / Coaxial-cable
HIPERLAN/2	High Performance Radio Local Area Network type 2
HLR	Home Location Register
HMAC	Keyed Hashing for Message Authentication (<--> RFC 2104)
H-PLMN	Home PLMN
HR	Halfrate
H-RNTI	HS-DSCH Radio Network Transaction Identifier (<--> 3GTS 25.331, 25.433)
HSCSD	High Speed Circuit Switched Data
HSDPA	High Speed Downlink Packet Access (<--> 3GTS 25.301, 25.308, 25.401, 3GTR 25.848)
HS-DPCCH	High Speed Dedicated Physical Control Channel (<--> 3GTS 25.211)
HS-DSCH	High Speed Downlink Shared Transport Channel (<--> 3GTS 25.211, 25.212, 25.308)
HS-PDSCH	High Speed Physical Downlink Shared Channel (<--> 3GTS 25.211)
HSS	Home Subscriber Server (<--> 3GTS 23.002). HSS replaces the HLR with 3GPP Rel. 5
HS-SCCH	High Speed Shared Control Channel (<--> 3GTS 25.211, 25.214)
HSUPA	High Speed Uplink Packet Access (<--> 3GTS 25.301, 25.309, 25.401, 3GTR 25.896)
HTTP	HyperText Transfer Protocol (<--> RFC 2616)
HUMAN	High-speed Unlicensed Metropolitan Area Network
I+S	Information + Supervisory
IAM	Initial Address Message (ISUP (ISDN User Part)
IANA	Internet Assigned Numbers Authority
ICANN	Internet Corporation for Assigned Names and Numbers
ICH	Indicator Channel (UMTS Physical Channel / see also PICH, AICH, CD/CA-ICH)
ICM	Initial Codec Mode
ICMP	Internet Control Message Protocol (<--> RFC 792)
ICS	Implementation Conformance Statement
I-CSCF	Interrogating Call Session Control Function (<--> SIP)
IE	Information Element

IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force (www.ietf.org)
IFFT	Inverse Fast Fourier Transformation
IHOSS	Internet Hosted Octet Stream Service
IK	Integrity Key (<--> 3GTS 33.102)
IKE	Internet Key Exchange (<--> RFC 2409)
IKMP	Internet Key Management Protocol
iLBC	Internet Low Bitrate Codec (<--> RFC 3951 / RFC 3952)
ILCM	Incoming Leg Control Model
IMEI	International Mobile Equipment Identity
IMPI	IP Multimedia Private Identity; the private user identity of an IMS-subscriber, formatted as an NAI (<--> 3GTS 33.203)
IMPU	IP Multimedia Public Identity; the public user identity of an IMS-subscriber, formatted as SIP-URI or TEL-URL (<--> 3GTS 33.203)
IMS	Internet Protocol Multimedia Core Network Subsystem (<--> Rel. 5 onwards)
IMS-AG	IMS-Access Gateway
IMSI	International Mobile Subscriber Identity
IMS-SSF	IP Multimedia Subsystem – Service Switching Function
IMT-2000	International Mobile Telecommunications for the year 2000
INAP	Intelligent Network Application Part (<--> CCS7)
IOV-I / IOV-UI	Input Offset Variable for I+S and UI-Frames (<--> for ciphering in GPRS)
IP	Internet Protocol (<--> RFC 791)
IPBCP	IP Bearer Control Protocol (<--> ITU-T Q.1970)
IP-CAN	Internet Protocol - Connectivity Access Network (e.g. DSL, TV-Cable, WIMAX, UMTS)
IPCP	Internet Protocol Control Protocol (<--> RFC 1332)
IPsec	Internet Protocol / secure (<--> RFC 2401)
IPv4	Internet Protocol (version 4)
IPv6	Internet Protocol (version 6)
IR	Incremental Redundancy (<--> ARQ II)
ISAKMP	Internet Security Association and Key Management Protocol (<--> RFC 2408)
ISC	IP multimedia subsystem Service Control-Interface
ISCP	Interference Signal Code Power (<--> 3GTS 25.215 / 3GTS 25.102)
ISDN	Integrated Services Digital Network
I-SIM	IMS capable SIM
ISO	International Standardization Organization
ISP	Internet Service Provider
ISPC	International Signaling Point Code (<--> ITU-T Q.708)
ISUP	ISDN User Part (<--> ITU-T Q.761 – Q.765)
ITU-T	International Telecommunication Union – Telecommunication Sector
Iub-FP	Iub-Frame Protocol (<--> 3GTS 25.427 / 25.435)
Iu-FP	Iu-Frame Protocol (<--> 3GTS 25.415)
Iur-FP	Iur-Frame Protocol (<--> 3GTS 25.424, 3GTS 25.425, 25.426, 25.435)
JPEG	Joint Picture Expert Group
kbps	kilo-bits per second
KEK	Key Encryption Key (<--> IEEE 802.16)
L1	Layer 1 (physical layer)
L2	Layer 2 (data link layer)
L2TP	Layer 2 Tunneling Protocol (<--> RFC 2661)
L3	Layer 3 (network layer)
LA	Location Area
LAC	Location Area Code
LAI	Location Area Identification (LAI = MCC + MNC + LAC)
LAPB	Link Access Procedure Balanced
LAPD	Link Access Protocol for the ISDN D-Channel
LBS	Location Based Service
LCP	Link Control Protocol (<--> PPP)
LCS	LoCation Service
LI	Length Indicator
LLC	Logical Link Control-Protocol
LMDS	Local Multipoint Distribution Services
LOS	Line Of Sight
LPD	Link Protocol Discriminator
LSB	Least Significant Bit
LSSU	Link Status Signal Unit
LTE	Long Term Evolution (of UMTS)
M3UA	MTP-3 User Adaptation Layer (<--> RFC 3332 / 3GPP 29.202 (Annex A))
MAC	Medium Access Control (UMTS (3GTS 25.321)
MAC	Medium Access Control ((E)GPRS (3GTS 04.60 / 3GTS 44.060)
MAC	Message Authentication Code (<--> 3GTS 33.102)

MAC-e	MAC-E-DCH (<--> 3GTS 25.321)
MAC-es	MAC-E-DCH SRNC (<--> 3GTS 25.321)
MAC-hs	MAC-High Speed (<--> 3GTS 25.321)
MAN	Metropolitan Area Network
MAP	Mobile Application Part
MASF	Minimum Available Spreading Factor
Max [X, Y]	The value shall be the maximum of X or Y, which ever is bigger
MBMS	Multimedia Broadcast / Multicast Service ((3GTS 23.246, 3GTS 43.846)
MBWA	Mobile Broadband Wireless Access (<--> IEEE 802.20 Specification of physical and medium access control layers of an air interface for interoperable mobile broadband wireless access systems, operating in licensed bands below 3.5 GHz, optimized for IP-data transport, with peak data rates per user in excess of 1 Mbps. It supports various vehicular mobility classes up to 250 Km/h in a MAN environment and targets spectral efficiencies, sustained user data rates and numbers of active users that are all significantly higher than achieved by existing mobile systems)
MBZ	Must Be Zero
MCC	Mobile Country Code
Mcps	Mega Chip Per Second
MCS-X	Modulation and Coding Scheme (1 – 9) and for HSDPA / HSUPA
MCU	Multipoint Control Unit (<--> H.323 equipment)
MD-X	Message Digest Algorithm (MD-2, 4, 5 are defined) (MD-5 (RFC 1321)
ME	Mobile Equipment (ME + SIM = MS)
MEGACO	Media Gateway Control Protocol (<--> ITU-T H.248 incl. Annex F – H and IETF RFC 3015)
MExE	Mobile Station Application Execution Environment
MGC	Media Gateway Controller
MGCF	Media Gateway Control Function
MGCP	Media Gateway Control Protocol (<--> RFC 2705)
MGW	Media Gateway
MIDI	Musical Instrument Digital Interface
MIH	Media Independent Handover (<--> IEEE 802.21)
MIME	Multipurpose Internet Mail Extensions
MIMO	Multiple In, Multiple Out
MIN	Mobile Identity Number (North American Market)
Min [X, Y]	The value shall be the minimum of X or Y, which ever is smaller
MINA	Mobile Internet Network Architecture
MLP	MAC Logical Channel Priority
MLPP	Multi-Level Precedence and Pre-emption (<--> ITU-T Q.85 / Clause 3)
MM	Mobility Management
MMCC	Multimedia Call Control
MMD	IP Multimedia Domain (<--> name of the IMS in 3GPP2)
MMDS	Multipoint Microwave Distribution System or Multi-channel Multi-point Distribution System
MMS	Multimedia Messaging Service (<--> 3GTS 22.140, 3GTS 23.140]
MNC	Mobile Network Code
MNRG	Mobile Not Reachable for GPRS flag
MOC	Mobile Originating Call
MPCC	Multiparty Call Control
MPEG	Motion Picture Expert Group
MPLS	Multi Protocol Label Switching
MRC	Maximum Ratio Combining
MRFC	Multimedia Resource Function Controller
MRFP	Multimedia Resource Function Processor
MRU	Maximum Receive Unit (<--> PPP)
MRW	Move Receiving Window
MS	Mobile Station
MS	Mobile Subscriber Station (<--> IEEE 802.16e)
MSB	Most Significant Bit
MSC	Mobile Services Switching Center
MSC-S	MSC-Server
MS-ISDN	Mobile Subscriber – International Service Directory Number
MSRP	Message Session Relay Protocol (<--> draft-ietf-simple-message-sessions-XX)
MSS	Maximum Segment Size (<--> TCP)
MSU	Message Signal Unit
MT	Mobile Terminal or Mobile Terminating
MTC	Mobile Terminating Call
MTP	Message Transfer Part (<--> ITU-T Q.701 – Q.709)
MTP-3b	Message Transfer Part level 3 / broadband (<--> ITU-T Q.2210)
MTU	Maximum Transmit Unit (<--> IP)
NACC	Network Assisted Cell Change (<--> 3GTS 44.060)
NACK	Negative Acknowledgement (<--> 3GTS 25.308, 25.309))
NAI	Network Access Identifier (<--> RFC 2486)
NAPT	Network Address Port Translation (<--> RFC 3022)
NAPTR	Naming Authority Pointer (<--> RFC 2915)
NAS	Non-Access-Stratum (<--> UMTS)

NASS	Network Attachment SubSystem (<--> part of the TISPAN NGN-architecture)
NAT	Network Address Translation (<--> RFC 1631)
NBAP	NodeB Application Part (<--> 3GTS 25.433)
NBNS	NetBios Name Service
NC	Neighbor Cell
NCC	Network Color Code
NCP	Network Control Protocol (<--> PPP)
NGN	Next Generation Networks
NI	Network Indicator
NIC	Network Interface Card
NLOS	Non Line Of Sight
NMT	Nordic Mobile Telephone (analog cellular standard, mainly used in Scandinavia)
NPB	Next Partial Bitmap
N-PDU	Network-Protocol Data Unit (<--> IP-Packet, X.25-Frame)
NS	Network Service
NSAP	Network Service Access Point
NSAPI	Network Service Access Point Identifier
N-SAW	N-Channel Stop and Wait (<--> 3GTS 25.309, 3GTR 25.848)
NSE	Network Service Entity
NSPC	National Signaling Point Code
NSS	Network Switching Subsystem
NS-VC	Network Service – Virtual Connection
NS-VCG	Network Service – Virtual Connection Group
NS-VL	Network Service – Virtual Link
NT	Network Termination
O&M	Operation and Maintenance
Octet	8 bit
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OLCM	Outgoing Leg Control Model
OMA	Open Mobile Alliance ((http://www.openmobilealliance.org/)
OMAC	One-Key CBC-MAC (<--> NIST standard: SP 800-38B and http://csrc.nist.gov/CryptoToolkit/modes/proposedmodes/)
OMC	Operation and Maintenance Center
OoBTC	Out of Band Transcoder Control (<--> 3GTS 23.153)
OPC	Originating Point Code
OPWA	One Pass With Advertising (<--> Term in RSVP)
OSA	Open Service Access
OSA-SCS	Open Service Access – Service Capability Server
OSCP	Online Certificate Status Protocol (<--> RFC 2560)
OSI	Open System Interconnection
OSP	Octet Stream Protocol
OTDOA	Observed Time Difference Of Arrival
OVSF	Orthogonal Variable Spreading Factor
P/F-Bit	Polling/Final - Bit
PA	Presence Agent (<--> RFC 3856)
PABX	Private Automatic Branch Exchange
PACCH	Packet Associated Control Channel ((E)GPRS)
PACS	Personal Access Communication System
PAD	Packet Assembly Disassembly
PAGCH	Packet Access Grant Channel ((E)GPRS)
PAP	Password Authentication Protocol (<--> RFC 1334)
PBCCH	Packet Broadcast Control Channel ((E)GPRS)
PCCCH	Packet Common Control Channel ((E)GPRS)
PCCH	Paging Control Channel (UMTS Logical Channel)
P-CCPCH	Primary Common Control Physical Channel (UMTS / used as bearer for the BCH TrCH)
PCH	Paging Channel (UMTS / Transport Channel)
PCH	Paging Channel (GSM / Logical Channel)
PCI	Peripheral Component Interconnect (computer bus standard to interconnect peripherals to the CPU)
PCM	Pulse Code Modulation
PCN	Personal Communication Network
PCPCH	Physical Common Packet Channel (UMTS Physical Channel)
P-CPICH	Primary Common Pilot Channel (UMTS Physical Channel)
PCS	Personal Communication System
P-CSCF	Proxy Call Session Control Function (<--> SIP)
PCU	Packet Control Unit
PD	Protocol Discriminator
PDC	Personal Digital Communication (ARIB-Standard)
PDCH	Packet Data Channel ((E)GPRS)

PDCP	Packet Data Convergence Protocol (<--> 3GTS 25.323)
PDF	Policy Decision Function (<--> Part of the IP Multimedia Subsystem)
PDG	Packet Data Gateway
PDG	Packet Data Gateway
PDH	Plesiochronous Digital Hierarchy
PDN	Packet Data Network
PDP	Packet Data Protocol
PDS	Packet Data Subsystem (<--> 3GPP2)
PDSCH	Physical Downlink Shared Channel (UMTS Physical Channel)
PDSN	Packet Data Support Node (<--> the SGSN in 3GPP2)
PDTCCH	Packet Data Traffic Channel ((E)GPRS)
PDU	Protocol Data Unit or Packet Data Unit
PEAP	Protected Extensible Authentication Protocol
PEP	Policy Enforcement Point (<--> 3GTS 23.209)
PER	Packed Encoding Rules (<--> ITU-T X.691)
PES	PSTN/ISDN Emulation Subsystem (<--> part of the TISPAN NGN-architecture)
PFC	Packet Flow Context
PFI	Packet Flow Identifier
PHS	Payload Header Suppression (<--> IEEE 802.16)
PHY	Physical Layer
PICH	Page Indicator Channel (UMTS Physical Channel)
PICMG	PCI (<--> Peripheral Component Interconnect) Industrial Computer Manufacturers Group (http://www.picmg.org/)
PIDF	Presence Information Data Format ((RFC 3863)
PKCS	Public Key Cryptography Standard
PLC	Power Line Communications
PLMN	Public Land Mobile Network
PMM	Packet Mobility Management
PN	Pseudo Noise
PNCH	Packet Notification Channel ((E)GPRS)
PoC	Push to talk over Cellular (<--> 3GTR 29.979 and various OMA-specifications)
PoE	Power over Ethernet
POP	Post Office Protocol (<--> RFC 1939)
POTS	Plain Old Telephone Service
PPCH	Packet Paging Channel ((E)GPRS)
PPP	Point-to-Point Protocol (<--> RFC 1661)
PRA	PCPCH Resource Availability
PRACH	Physical Random Access Channel (UMTS
PRACH	Packet Random Access Channel ((E)GPRS)
PRD	Bluetooth Qualification Program Reference Document
PRI	Primary rate access ISDN-user interface for PABX's (23 or 30 B-channels plus one D-Channel)
PS	Physical Slot (<--> IEEE 802.16)
PS	Puncturing Scheme
PSC	Primary Synchronization Code or Primary Scrambling Code (both used in UMTS)
P-SCH	Primary Synchronization Channel (physical)
PSD	Power Spectral Density (<--> 3GTS 25.215 / 3GTS 25.102)
PSK	Phase Shift Keying
PSPDN	Packet Switched Public Data Network
PSTN	Public Switched Telephone Network
PT	Protocol Type (<--> GTP or GTP')
PTCCH	Packet Timing Advance Control Channel ((E)GPRS)
PTCCH/D	Packet Timing Advance Control Channel / Downlink Direction ((E)GPRS)
PTCCH/U	Packet Timing Advance Control Channel / Uplink Direction ((E)GPRS)
PTM	Point to Multipoint
P-TMSI	Packet TMSI
PTP	Point to Point
PUA	Presence User Agent (<--> RFC 3856)
PVC	Permanent Virtual Circuit
QCIF	Quarter Common Intermediate Format (176 x 144 pixels (ITU-T H261 / H263)
QE	Quality Estimate
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying (<--> 3GTS 25.213)
QSIG	Q-interface signaling protocol
RA	Routing Area
RAB	Radio Access Bearer
RAC	Routing Area Code
RACH	Random Access Channel (UMTS Transport Channel)
RACH	Random Access Channel (GSM)
RACS	Resource and Admission Control Subsystem (<--> part of the TISPAN NGN-architecture)

RADIUS	Remote Authentication Dial In User Service (<--> RFC 2865)
RAI	Routing Area Identification
RANAP	Radio Access Network Application Part (<--> 3GTS 25.413)
RAND	Random Number
RAT	Radio Access Technology (e.g. GERAN, UTRAN, ...)
RATSCCH	Robust AMR Traffic Synchronized Control CHannel
RB	Receive Block Bitmap (<--> EGPRS)
RB	Radio Bearer
RBB	Receive Block Bitmap (<--> GPRS)
REJ	Reject
RF	Radio Frequency
RFC	Request for Comments (<--> Internet Standards)
RFID	Radio Frequency Identification
RG	Relative Grant (<--> 3GTS 25.309)
R-GSM	Railways-GSM
RL	Radio Link
RLC	Radio Link Control (UMTS (3GTS 25.322)
RLC	Radio Link Control ((E)GPRS / 3GTS 04.60 / 3GTS 44.060)
RLM	Radio Link Management (Protocol Part on the GSM Abis-Interface / 3GTS 48.058)
RLP	Radio Link Protocol (<--> 3GTS 24.022)
RLS	Radio Link Set (<--> 3GTS 25.309, 25.433)
RNC	Radio Network Controller
RNL	Radio Network Layer
RNR	Receive Not Ready
RNS	Radio Network Subsystem
RNSAP	Radio Network Subsystem Application Part (<--> 3GTS 25.423)
RNSN	Radio Network Serving Node
RNTI	Radio Network Temporary Identifier
RPID	Rich Presence Information Data
RPLMN	Registered PLMN
RPR	Resilient Packet Ring (<--> IEEE 802.17)
RR	Radio Resource Management
RR	Receive Ready (LAPD/LLC/RLP-Frame Type)
RRBP	Relative Reserved Block Period
RRC	Radio Resource Control (<--> 3GTS 25.331)
RRC-Filter	Root Raised Cosine Filter
RSA	Ron Rivest, Adi Shamir and Leonard Adleman-algorithm (Public Key Encryption / PKCS #1)
RSADP	RSA-Decryption Primitive (<--> RFC 3447 (5.1.2) or PKCS #1 (5.1.2); PKCS = Public Key Cryptography Standard]
RSAEP	RSA-Encryption Primitive (<--> RFC 3447 (5.1.1) or PKCS #1 (5.1.1); PKCS = Public Key Cryptography Standard]
RSAES-OAEP	RSA Encryption Scheme - Optimal Asymmetric Encryption Padding (<--> PKCS #1 / RFC 3447)
RSC	Recursive Systematic Convolutional Coder((Turbo Coding, 25.212)
RSCP	Received Signal Code Power (<--> 3GTS 25.215)
RSN	Retransmission Sequence Number (<--> 3GTS 25.309)
RSSI	Received Signal Strength Indicator
RSVP	Resource Reservation Protocol (<--> RFC 2205)
RTG	Receive transmit Transition Gap ((IEEE 802.16 (3.45)) the time between an uplink subframe and the subsequent downlink subframe in a TDD-system
RTO	Retransmission Time Out
RTP	Real-time Transport Protocol (<--> RFC 3550, RFC 3551)
RTP/AVP	Real-time Transport Protocol / Audio Video Profile ((RFC 3551) (used in SDP-descriptions)
RTP/AVPF	Real-time Transport Protocol / extended Audio Video Profile for rtcp Feedback (used in SDP-descriptions)[draft-ietf-avt-rtcp-feedback-11.txt]
RTP/SAVP	Real-time Transport Protocol / Secure Audio Video Profile ((RFC 3711) (used in SDP-descriptions)
RTSP	Real Time Streaming Protocol (<--> RFC 2326)
RTT	RoundTrip Time (<--> RFC 793)
RTWP	Received Total Wideband Power
RV	Redundancy and Constellation Version (<--> 3GTS 25.212)
RX	Receive
SA	Service Area
SAAL-NNI	Signaling ATM Adaptation Layer – Network Node Interface
SAB	Service Area Broadcast
SABM(E)	Set Asynchronous Balanced Mode (Extended for Modulo 128 operation) (LAPD/LLC/RLP-Frame Type)
SABP	Service Area Broadcast Protocol (<--> 3GTS 25.419)
SACCH	Slow Associated Control Channel (GSM)
SACCH/MD	SACCH Multislot Downlink (related control channel of TCH/FD / GSM)
SAI	Service Area Identifier
SAIC	Single Antenna Interference Cancellation
SANC	Signaling Area Network Code (<--> ITU-T Q.708)
SAP	Service Access Point

SAPI	Service Access Point Identifier
SAR	Segmentation And Reassembly (ATM-sublayer)
SBC	Session Border Controller (<--> SIP term, usually a B2BUA with NAT-function and media gateway)
SC	Serving Cell
SCCP	Signaling Connection Control Part (<--> ITU-T Q.711 – Q.714)
S-CCPCH	Secondary Common Control Physical Channel (used as bearer for the FACH and PCH TrCH's / UMTS Physical Channel)
SCH	Synchronization Channel (UMTS Physical Channel / see also P-SCH and S-SCH)
SCH	Synchronization Channel (GSM)
S-CPICH	Secondary Common Pilot Channel (UMTS Physical Channel)
SCR	Source Controlled Rate
S-CSCF	Serving Call Session Control Function (<--> SIP)
SCTP	Stream Control Transmission Protocol (<--> RFC 2960)
SDCCH	Stand Alone Dedicated Control Channel
SDH	Synchronous Digital Hierarchy
SDMA	Space Division Multiple Access
SDP	Session Description Protocol (<--> RFC 2327, RFC 3266, RFC 3264)
SDU	Service Data Unit (<--> the payload of a PDU)
SF	Spreading Factor
SFH	Slow Frequency Hopping
SFN	System Frame Number
SG	Security Gateway (IPsec / (RFC 2401)
SGSN	Serving GPRS Support Node
SGW	Signaling Gateway (SS7 (IP)
SHA	Secure Hash Algorithm
SHCCH	Shared Channel Control Channel (UMTS Logical Channel / (TDD only)
SI	Service Indicator
SIB	System Information Block
SID	Silence Insertion Descriptor
SID	Size InDex (<--> 3GPP 25.321)
SIF	Signaling Information Field
SIG	Special Interest Group (<--> e.g. Bluetooth)
SIM	Subscriber Identity Module
SIO	Service Information Octet
SIP	Session Initiation Protocol (<--> RFC 3261)
SIP-AS	SIP-Application Server
SIP-B	SIP for Businesses (<--> abbreviation for a set of PABX-specific SIP-extensions)
SIP-I	SIP with encapsulated ISUP (<--> ITU-T Q.1912.5)
SIP-T	SIP for Telephones (<--> RFC 3372, RFC 3398)
SIR	Signal to Interference Ratio
SLC	Signaling Link Code
SLF	Subscriber Locator Function
SLR	Source Local Reference
SLS	Signaling Link Selection
SLTA	Signaling Link Test Acknowledge
SLTM	Signaling Link Test Message
SM	Session Management (<--> 3GTS 23.060, 3GTS 24.008)
SME	Small and Medium size Enterprises (Type of Business)
SMS	Short Message Service (<--> 3GTS 24.011, 3GTS 23.040)
SM-SC	Short Message Service Center
SMSCB	Short Message Services Cell Broadcast
SMS-G-MSC	SMS Gateway MSC (for Short Messages destined to Mobile Station)
SMS-IW-MSC	SMS Interworking MSC (for Short Messages coming from Mobile Station)
SMTP	Simple Mail Transfer Protocol (<--> RFC 2821)
SN	Sequence Number
SND	Sequence Number Downlink (<--> GTP)
SNDCP	Subnetwork Dependent Convergence Protocol
SNM	Signaling Network Management Protocol (<--> ITU-T Q.704 (3))
SNN	SNDCP N-PDU Number Flag
SN-PDU	Segmented N-PDU (SN-PDU is the payload of SNDCP)
SNR	Signal to Noise Ratio
SNTM	Signaling Network Test & Maintenance ((ITU-T Q.707)
SNU	Sequence Number Uplink (<--> GTP)
SOAP	Simple Object Access Protocol (<--> http://www.w3.org/TR/2000/NOTE-SOAP-20000508)
SOHO	Small Office Home Office (Type of Business)
SPC	Signaling Point Code
SPI	Security Parameter Index (<--> RFC 2401)
SQCIF	Semi Quarter Common Intermediate Format (128 x 96 pixels (ITU-T H261 / H263)

SQN	Sequence number (used in UMTS-security architecture / 3GTS 33.102)
SRB	Signaling Radio Bearer
SRES	Signed Response
SRNC	Serving Radio Network Controller
SRNS	Serving Radio Network Subsystem
SRTTP	Secure RTP (<--> RFC 3711)
SRTT	Smoothed RoundTrip Time (<--> RFC 793)
SRV	Service Location (<--> RFC 2782)
SS	Subscriber Station (<--> IEEE 802.16)
SSC	Secondary Synchronization Code
SSCF	Service Specific Co-ordination Function
SSCF/NNI	Service Specific Coordination Function – Network Node Interface Protocol (<--> ITU-T Q.2140)
SSCF/UNI	Service Specific Coordination Function – User Network Interface Protocol (<--> ITU-T Q.2130)
S-SCH	Secondary Synchronization Channel (physical)
SSCOP	Service Specific Connection Oriented Protocol (<--> ITU-T Q.2110)
SSCOPMCE	Service Specific Connection Oriented Protocol in a Multi-link or Connectionless Environment (<--> ITUT Q.2111)
SSCS	Service Specific Convergence Sublayer
SSDT	Site Selection Diversity Transmission
SSID	Service Set Identifier (<--> IEEE 802.11)
SSN	Start Sequence Number (<--> related to ARQ-Bitmap in GPRS / EGPRS)
SSN	Send Sequence Number (<--> GSM MM and CC-Protocols)
SSRTG	Subscriber Station Receive to transmit Transition Gap (<--> IEEE 802.16 (3.53)) Time that the SS needs to switch from receive to transmit.
SSSAR	Service Specific Segmentation And Reassembly (<--> ITU-T I.366.1)
ssthresh	Slow start threshold (<--> RFC 2001)
SSTTG	Subscriber Station Transmit to receive Transition Gap (<--> IEEE 802.16 (3.54)) Time that the SS needs to switch from transmit to receive.
STC	Signaling Transport Converter on MTP-3 and MTP-3b (<--> ITU-T Q.2150.1) / Signaling Transport Converter on SSCOP and SSCOPMCE (<--> ITU-T Q.2150.2)
STTD	Space Time block coding based Transmission Diversity
STUN	Simple Traversal of UDP through Network Address Translators (<--> RFC 3489)
SUERM	Signal Unit Error Rate Monitor (<--> ITU-T Q.703 (10))
SUFI	Super Field (RLC-Protocol)
SVC	Switched Virtual Circuit
SWAP	Shared Wireless Access Protocol (<--> Home RF)
T.38	Fax Specification
TA	Terminal Adapter (<--> ISDN)
TA	Timing Advance
TACS	Total Access Communication System
TAF	Terminal Adopter Function (<--> 3GTS 27.001)
TAI	Timing Advance Index
TB	Transport Block
TBCP	Talk Burst Control Protocol
TBF	Temporary Block Flow
TBS	Transport Block Set
TCAP	Transaction Capabilities Application Part (<--> Q.771 – Q.773)
TCH	Traffic Channel
TCH/FD	Traffic Channel / Fullrate Downlink
TCH-AFS	Traffic CHannel Adaptive Full rate Speech
TCH-AHS	Traffic Channel Adaptive Half rate Speech
TCP	Transmission Control Protocol
TCP/BFCP	Transmission Control Protocol / Binary Floor Control Protocol ((draft-ietf-xcon-bfcp-05.txt)
TCP/RTP/AVP	Real-time Transport Protocol / Audio Video Profile over TCP (used in SDP-descriptions)[draft-ietf-avt-rtp-framing-contrans-06.txt]
TCP/TLS/BFCP	Transmission Control Protocol / Transport Layer Security / Binary Floor Control Protocol ((draft-ietf-xcon-bfcp-05.txt)
TCTF	Target Channel Type Field
TCTV	Transport Channel Traffic Volume
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TE	Terminal Equipment
TEID	Tunnel Endpoint Identifier (<--> GTP / 3GTS 29.060)
TEK	Traffic Encryption Key (<--> IEEE 802.16)
TF	Transport Format
TFC	Transport Format Combination
TFCl	Transport Format Combination Identifier
TFCS	Transport Format Combination Set
TFI	Transport Format Indication (<--> UMTS)
TFI	Temporary Flow Identity (<--> (E)GPRS)
TFO	Tandem Free Operation (<--> 3GTS 22.053)
TFRC	Transport Format and Resource Combination (<--> 3GTS 25.308)
TFRI	Transport Format and Resource Indicator (<=> 3GTS 25.308, 25.321)
TFS	Transport Format Set

TFTP	Trivial File Transfer Protocol (<--> RFC 1350)
TGD	Transmission Gap start Distance (<--> 3GTS 25.215)
TGL	Transmission Gap Length (<--> 3GTS 25.215)
TGPRC	Transmission Gap Pattern Repetition Count (<--> 3GTS 25.215)
TGSN	Transmission Gap Starting Slot Number (<--> 3GTS 25.215)
TH-CDMA	Time Hopping Code Division Multiple Access
THIG	Topology Hiding Inter Network Gateway
TI	Transaction Identifier
TIA	Telecommunications Industry Association
TID	Tunnel Identifier
TIPHON	Telecommunications and Internet Protocol Harmonization Over Networks (<--> ETSI Project)
TISPAN	Telecoms & Internet converged Services & Protocols for Advanced Networks (<--> ETSI Working Group to define IMS for fixed broadband access networks)
TLLI	Temporary Logical Link Identifier
TLS	Transport Layer Security (<--> RFC 2246 / RFC 3546 / formerly known as SSL or Secure Socket Layer)
TLV	Tag / Length / Value Notation
TM	Transparent Mode operation (<--> UMTS-RLC)
TM	Transmission Modules
TMD	Transparent Mode Data ((UMTS RLC PDU-type)
TMN	Telecommunication Management Network
TMSI	Temporary Mobile Subscriber Identity
TNL	Transport Network Layer (<--> 3GTS 25.401)
ToIP	Text over IP
TPC	Transmit Power Command
T-PDU	Payload of a G-PDU which can be user data, i.e. possibly segmented IP-frames, or GTP signaling information (<--> GTP)
TQI	Temporary Queuing Identifier
TRAU	Transcoder and Rate Adaption Unit
TrCH	Transport Channel (UMTS)
TrFO	Transcoder Free Operation
TrGw	Transition Gateway (IPv4 (IPv6)
TRX	Transmitter / Receiver
TS	Timeslot
TSC	Training Sequence Code
TSN	Transmission Sequence Number (<--> 3GTS 25.321)
TSTD	Time Switched Transmit Diversity
TTA	Telecommunications Technology Association (South Korean standards organization)
TTG	Tunnel Termination Gateway
TTG	Transmit receive Transition Gap ((IEEE 802.16 (3.63)) the time between a downlink subframe and the subsequent uplink subframe in a TDD-system
TTI	Transmission Time Interval
TTL	Time To Live (<--> IP-Header / RFC 791)
TX	Transmit
UA	User Agent (<--> SIP-Term / RFC 3261)
UA	Unnumbered Acknowledgement (LAPD/LLC/RLP-Frame Type)
UAC	User Agent Client (<--> SIP-Term / RFC 3261)
UARFCN	UMTS Absolute Radio Frequency Channel Number
UART	Universal Asynchronous Receiver and Transmitter
UAS	User Agent Server (<--> SIP-Term / RFC 3261)
UDP	User Datagram Protocol (<--> RFC 768)
UDPTL	UDP Transport Layer (used in SDP-description for T.38 fax-applications)
UE	User Equipment
UEA	UMTS Encryption Algorithm (<--> 3GTS 33.102)
UI	Unnumbered Information (<--> LAPD) / Unconfirmed Information (<--> LLC) / Frame Type
UIA	UMTS Integrity Algorithm (<--> 3GTS 33.102)
UICC	Universal Integrated Circuit Card (<--> 3GTS 22.101 / Bearer card of SIM / USIM)
UL	Uplink
UM	Unacknowledged Mode operation (<--> UMTS-RLC)
UMA	Unlicensed Mobile Access
UMA	Unlicensed Mobile Access (<--> 3GTS 43.318)
UMAN	Unlicensed Mobile Access Network
UMD	Unacknowledged Mode Data ((UMTS RLC PDU-type)
UMS	User Mobility Server (<--> HSS = HLR + UMS)
UMTS	Universal Mobile Telecommunication System
UNC	UMA Network Controller
UNC-SGW	UMA Network Controller Security Gateway
URA	UTRAN Registration Area
URI	Uniform Resource Identifier
URL	Uniform Resource Locator (<--> RFC 1738)
USAT	USIM Application Toolkit
USB	Universal Serial Bus

USCH	Uplink Shared Channel (UMTS Transport Channel (TDD only
USF	Uplink State Flag
USIM	Universal Subscriber Identity Module ((3GTS 31.102)
UTF-8	Unicode Transformation Format-X (Is an X-bit) lossless encoding of Unicode characters)
UTRA	UMTS Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
UII	User to User Information
UUS	User-User-Signaling (<--> 3GTS 23.087)
UWB	Ultra-Wide Band (<--> IEEE 802.15.3)
UWC	Universal Wireless Convergence (Merge IS-136 with GSM)
VAD	Voice Activity Detector
VBS	Voice Broadcast Service (<--> GSM-R)
VC	Virtual Circuit
VCI	Virtual Circuit Identifier (<--> ATM)
VGCS	Voice Group Call Service (<--> GSM-R)
VHE	Virtual Home Environment (<--> 3GTS 22.121, 3GTS 23.127)
VLR	Visitor Location Register
VPI	Virtual Path Identifier (<--> ATM)
V-PLMN	Visited PLMN
VPN	Virtual Private Network
WAG	WLAN Access Gateway
WAP	Wireless Application Protocol
WCDMA	Wide-band Code Division Multiple Access
WIMAX	Worldwide Interoperability for Microwave Access ((IEEE 802.16)
WINS	Windows Internet Name Service
W-LAN	Wireless Local Area Network (<--> IEEE 802.11)
WMAN	Wireless Metropolitan Area Network
WSN	Window Size Number
XID	Exchange Identification (LAPD/LLC-Frame Type)
XOR	Exclusive-Or Logical Combination
XRES	Expected Response (<--> 3GTS 33.102)

Index

1

100rel	b
100-Trying (<--> SIP-Response Code)	b
183-Session Progress (<--> SIP-Response Code)	b

3

302-Moved Temporarily (<--> SIP-Response Code)	b b
3pcc	b b

4

401-Unauthorized (<--> SIP-Response Code)	b
420-Bad Extension (<--> SIP-Response Code)	b
422- Session Interval Too Small (<--> SIP-Response Code)	b
480-Temporarily Unavailable (<--> SIP-Response Code)	b
481-Call Leg / Transaction Does Not Exist (<--> SIP-Response Code)	b
487-Request Terminated / Transaction Does Not Exist (<--> SIP-Response Code)	b
488-Not Acceptable Here (<--> SIP-Response Code)	b

5

504-Server Timeout (<--> SIP-Response Code)	b
580-Precondition Failure (<--> SIP-Response Code)	b

6

606-Not Acceptable (<--> SIP-Response Code)	b b
---	-------------------------------------

A

Address (<--> SDP o-line)	b
Address-type (<--> SDP o-line)	b
ALG (Application Layer Gateway)	b
a-line (<--> SDP)	b
AMR (<--> SDP-definition)	b
Application Layer Gateway	b
Application Server	b
AS (<--> SDP b-line / modifier)	b
AS (Application Server)	b
attribute lines (<--> SDP)	b

B

B2BUA	b
back-to-back user agent	b
BFCP	b
b-line (<--> SDP)	b b
branch parameter	b

C

















































































call	b
call (<--> definition in SIP)	b
call drop (no common codec)	b
call forwarding unconditional	b
Call-ID	b
capacity of SIP-server	b
c-line (<--> SDP)	b
codec type	b
common presence and instant messaging	b
confirmation (conf) (<--> SDP-attribute)	b
connection (<--> SDP-attribute)	b
Contact (<--> header field)	b
Core (<--> SIP-sublayer)	b
CPIM	b
CSeq	b
CT (<--> SDP b-line / modifier)	b
current (curr) (<--> SDP-attribute)	b




















































































D




















































































desired (des) (<--> SDP-attribute)	b
DIA	b
dialog	b b
dialog (<--> definition in SIP)	b
DIAMETER	b
do not disturb	b

E

early dialog	b
--------------	-------------------

event server	
event: reg	
expires-time (<--> 3GPP)	 
F	
find me	
fmp (<--> SDP-attribute)	
follow me	
forking	    
FQDN	
From-tag	
fully qualified domain name	
H	
H.248	
H.323	
header field	
P-Associated-URI	 
Path	
Service-Route	
home network domain name	
I	
iLBC	
IMPI	
IMPU	
IMS	
IP Multimedia Subsystem	
ISIM	
K	
keep-alive mechanism	
L	
local (<--> SDP-attribute)	
location based services	
M	
magic cookie ((z9hG4bK)	
media description (<--> SDP)	 
media gateway	
media gateway controller	 
media stream adjustment	
media type (MIME) (<--> SDP m-line)	
media types (MIME) / examples	
MEGACO	
MGC	 
MGW	
Min-SE	
m-line (<--> SDP)	
mode-change-neighbor (<--> SDP-attribute / AMR-specific)	
mode-change-period (<--> SDP-attribute / AMR-specific)	
mode-set (<--> SDP-attribute / AMR-specific)	
modifying a media stream	
N	
NAPTR	
network architecture (<--> SIP)	
Network-type (<--> SDP o-line)	
Next Generation Networks	 
NGN	 
notification (<--> event)	
NTP	
O	
offer / answer model	
o-line (<--> SDP)	
P	
Packed Encoding Rules	
P-Associated-URI (<--> header field)	 
Path (<--> header field)	
payload-type-list (<--> SDP m-line)	
PDF	  
PER	
policy decision function	  
port number (<--> RTCP)	
port number (<--> SIP)	 
port-number (<--> SDP m-line)	
PRACK	
precondition (<--> SIP-option tag)	
preconditions (<--> SDP)	

presence (<--> event)	
private user identity	
protocol stack (<--> SIP)	
provisional response messages	
provisional responses	
proxy (<--> stateful)	 
proxy (<--> stateless)	 
proxy server discovery (SIP)	 
PSTN-replacement	
public user identity	
publishing (<--> event)	
Q	
q-parameter	
R	
RAck	
Real-Time Streaming Protocol	
Real-time Transport Protocol	
recvonly (<--> SDP-attribute)	
redirect server	 
refresher (<--> timer based session release)	
Registrar	
registration event package	
remote (<--> SDP-attribute)	
Response Types (<--> SIP)	
Response: 100-Trying	
Response: 183-Session Progress	
Response: 302-Moved Temporarily	 
Response: 401-Unauthorized	
Response: 420-Bad Extension	
Response: 422- Session Interval Too Small	
Response: 481-Call Leg / Transaction Does Not Exist	
Response: 486-User Busy	
Response: 487-Request Terminated	
Response: 488-Not Acceptable Here	
Response: 504-Server Timeout	
Response: 580-Precondition Failure	
Response: 606-Not Acceptable	 
RR (<--> SDP b-line / modifier)	 
RS (<--> SDP b-line / modifier)	 
RSeq	 
RTCP-port number)	
RTP	
RTP/AVP (<--> SDP m-line / transport)	
RTP/AVPF (<--> SDP m-line / transport)	
RTP/SAVP (<--> SDP m-line / transport)	
rtptime (<--> SDP-attribute)	
RTSP	
S	
SBC	 
SBC initiated session release	
S-CSCF selection	
SDP-media description	
SDP-session description	
SDP-time description	
Secure Real-time Transport Protocol	
selection (of S-CSCF)	
sendonly (<--> SDP-attribute)	
Service-Route (<--> header field)	
session	
session (<--> identification)	
session border controller	
session description (<--> SDP)	
session establishment with SIP (<--> simplified example)	
session release (<--> SBC-initiated)	
session release (<--> timer based)	
session release (<--> ungraceful)	 
session timer	 
Session-Description-Version (<--> SDP o-line)	
session-description-version (<--> SDP-description)	
Session-ID (<--> SDP o-line)	
SigComp	
SIP ((Response types)	
SIP (<--> protocol stack)	
SIP (<--> scope of)	

SIP trapezoid	
SIP Use within NGN	
SIP-B	
SIP-bridging	
SIP-I	
SIPPING	
SIP-proxy server discovery	
SIP-server (<--> capacity of)	
SIP-T	 
SIP-transport protocol	
SKYPE	
s-line (<--> SDP)	
soft switch	 
SRTP	
SRV	
start time (<--> SDP t-line)	
stateful proxy	 
stateless proxy	 
stop time (<--> SDP t-line)	
sublayers of SIP	
subscription (<--> event)	
T	
T1	
tag (<--> From)	
tag (<--> To)	
TBCP (<--> SDP m-line / transport)	
TCP (<--> SDP m-line / transport)	
TCP/BFCP (<--> SDP m-line / transport)	
TCP/MSRP (<--> SDP m-line / transport)	
TCP/RTP/AVPF (<--> SDP m-line / transport)	
TCP/TLS/BFCP (<--> SDP m-line / transport)	
TCP/TLS/MSRP (<--> SDP m-line / transport)	
TIAS (<--> SDP b-line / modifier)	
time description (<--> SDP)	
timer 1	
timer A	
timer B	
timer C	 
timer D	
timer E	 
timer F	 
timer G	
timer H	
timer J	
timer K	
t-line (<--> SDP)	
To-tag	
Transaction (<--> definition in SIP)	
transaction (<--> identification)	  
transaction / CANCEL	
transaction / INVITE / successful	
transaction / INVITE / unsuccessful	
transaction / REGISTER	
transaction handling (<--> SIP-sublayer)	
transaction numbering	
transaction user (<--> SIP-sublayer)	
transport (<--> SDP m-line)	
transport layer control (<--> SIP-sublayer)	
transport protocol (<--> SIP)	
trapezoid (SIP)	
Triple-Play Services	
TU	
U	
UA	
UAC	
UAS	
UDP (<--> SDP m-line / transport)	
UDPTL (<--> SDP m-line / transport)	
UMA	
ungraceful session release	 
UPDATE	 
User Agent	
user busy	
user not registered	

user not responding

Username (<--> SDP o-line)

V

via

v-line (<--> SDP)

Z

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