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SERIES Q: SWITCHING AND SIGNALLING

Signalling requirements and protocols for the NGN –
Service and session control protocols

NGN UNI signalling profile (Protocol set 1)

Recommendation ITU-T Q.3402

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Recommendation ITU-T Q.3402

NGN UNI signalling profile (Protocol set 1)

Summary

Recommendation ITU-T Q.3402 defines the NGN user-to-network interface (UNI) signalling profile for use between users and networks of protocol set 1 for voice, video and data services.

Source

Recommendation ITU-T Q.3402 was approved on 29 February 2008 by ITU-T Study Group 11 (2005-2008) under Recommendation ITU-T A.8 procedure.

Keywords

NGN, profile, SDP, signalling, SIP.

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Recommendation ITU-T Q.3402

NGN UNI signalling profile (Protocol set 1)

1 Scope

This Recommendation specifies a service-level profile, i.e., SIP/SDP interface description between a user and a network, and a transport-level profile, e.g., RTP.

For protocol set 1 of the NGN UNI profile, this Recommendation covers multimedia (voice, video, and data) which include VoIP, multimedia telephony, DTMF, T.38 fax, and multimedia ring back and ringing tones and announcements.

This Recommendation covers all terminal types, e.g., SIP residential gateway, SIP phone, and SIP IP PBX. Therefore, the following UNI interfaces are specified:

- SIP residential gateway-to-service provider, where PSTN/ISDN terminals or IP phones can be connected to residential gateway.
- SIP phone-to-service provider, where the SIP phone can be either a soft phone or a hard phone implemented by IMS-based SIP specifications.
- SIP IP PBX-to-service provider, where the IP PBX can be either a proxy or B2BUA.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

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3 Definitions

For SIP- and SDP-specific terminology, reference shall be made to [IETF RFC 3261], [IETF RFC 3264], [IETF RFC 2327] and [IETF RFC 4566]. For NGN-specific terminology, reference shall be made to [ITU-T Y.2012]. Definitions for additional terminology used in this Recommendation are as follows:

3.1 end-user function (EUF): Includes end-user equipment, whether legacy or NGN, and also includes customer networks. End-user equipment may be either mobile or fixed. The end-user interfaces via which the EUF is connected to NGN are supported by both physical and functional (control) interfaces.

3.2 recommended-codec list: A recommended-codec list contains the codecs that should be shown by the network to the user in SIP/SDP messages exchanged over the UNI.

NOTE – The purpose of the recommended-codec list is just to show the codecs that a network recommends for use in the UNI, and the recommended-codec list does not recommend terminals to implement all the codecs shown in the list.

3.3 service control functions (SCF): The service control functions establish, monitor, support, and release multimedia sessions and manage the user's service interactions.

3.4 SIP back-to-back user agent (B2BUA): A back-to-back user agent (B2BUA) is a concatenation of a SIP user agent client (UAC) and user agent server (UAS).

NOTE – The IETF defines the B2BUA in [IETF RFC 3261] as "a logical entity that receives a request and processes it as a user agent server (UAS). In order to determine how the request should be answered, it acts as a user agent client (UAC) and generates requests. Unlike a proxy server, it maintains dialog state and shall participate in all requests sent on the dialogs it has established. Since it is a concatenation of a UAC and

UAS, no explicit definitions are needed for its behavior." (UAC and UAS behaviours are defined in [IETF RFC 3261].) A B2BUA reformulates a message before sending it as a new request.

4 Abbreviations

This Recommendation uses the following abbreviations:

3GPP	3rd Generation Partnership Project
AKA	Authentication and Key Agreement
AMR	Adaptive Multirate (codec)
AMR NB	AMR Narrowband
AMR WB	AMR Wideband
B2BUA	Back-to-Back User Agent
CSC-FE	Call Session Control Functional Entity
CSMA/CD	Carrier Sense Multiple Access with Collision Detection
DTMF	Dual-Tone Multifrequency
EUF	End-User Functions
EVRC	Enhanced Variable Rate Codec
FQDN	Fully Qualified Domain Name
GRUU	Globally Routable User Agent URIs
HTTP	HyperText Transfer Protocol
IBC-FE	Interconnection Border gateway Control Functional Entity
ICMP	Internet Control Message Protocol
IP PBX	IP Private Branch eXchange
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
IVR	Interactive Voice Response
KPML	Key Press Stimulus
MIME	Multi-purpose Internet Mail Extensions
MPEG	Moving Picture Experts Group
NAT	Network Address Translation
NGN	Next Generation Network
NGN-TE	NGN Terminal Equipment
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RGW	Residential Gateway
RTCP	RTP Control Protocol
RTCP XR	RTCP eXtended Reports
RTP	Real-time Transport Protocol

SCF	Service Control Functions
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIPS	Session Initiation Protocol Secure
SMV	Selectable Mode Vocoders
SRTP	Secure Real-time Transport Protocol
TCP	Transmission Control Protocol
TLS	Transport Layer Security
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
UNI	User-to-Network Interface
URI	Universal Resource Identifier
VMR-WB	Variable-Rate Multi-Mode Wideband
VoIP	Voice over IP

5 Reference model

Figure 1 illustrates the UNI interface covered by this Recommendation in the NGN architecture defined in [ITU-T Y.2012].

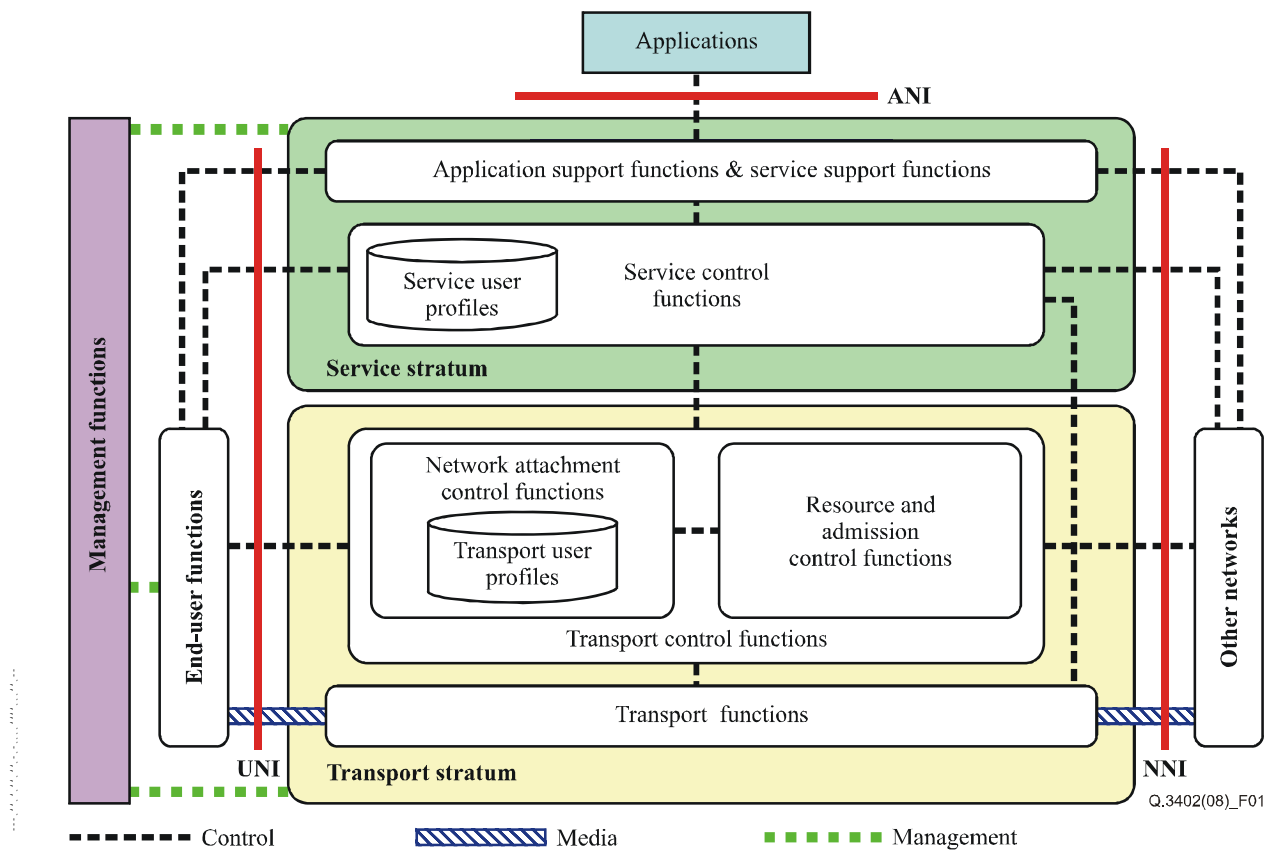


Figure 1 – UNI interface covered by this Recommendation within the NGN architecture

Figures 2, 3 and 4 illustrate the possible scenarios of terminal types within the EUF.

Figure 2 shows the scenario for PSTN/ISDN terminals and IP phones connected to a service provider via a SIP residential gateway.

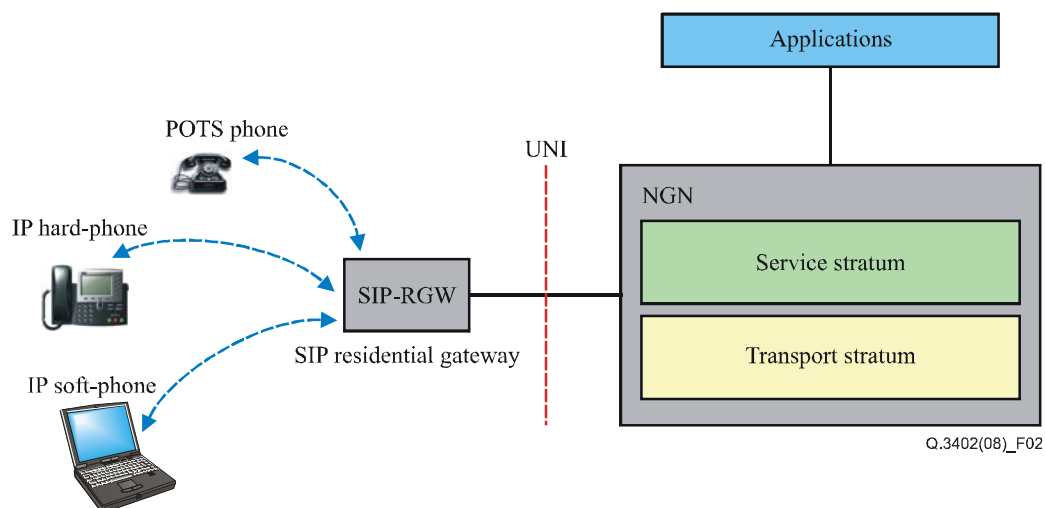


Figure 2 – Scenario for SIP residential gateway

Figure 3 shows the scenario for IMS-based SIP phones connected directly to a service provider.

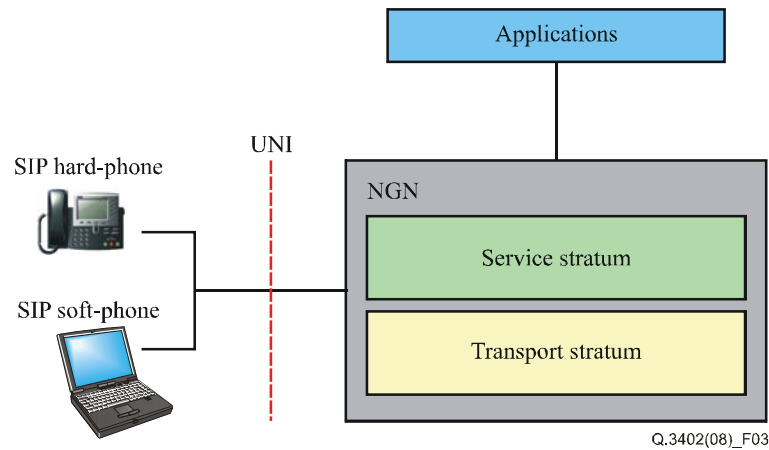


Figure 3 – Scenario for IMS-based SIP phone

Figure 4 shows the scenario for SIP phones connected to a service provider via a SIP IP PBX.

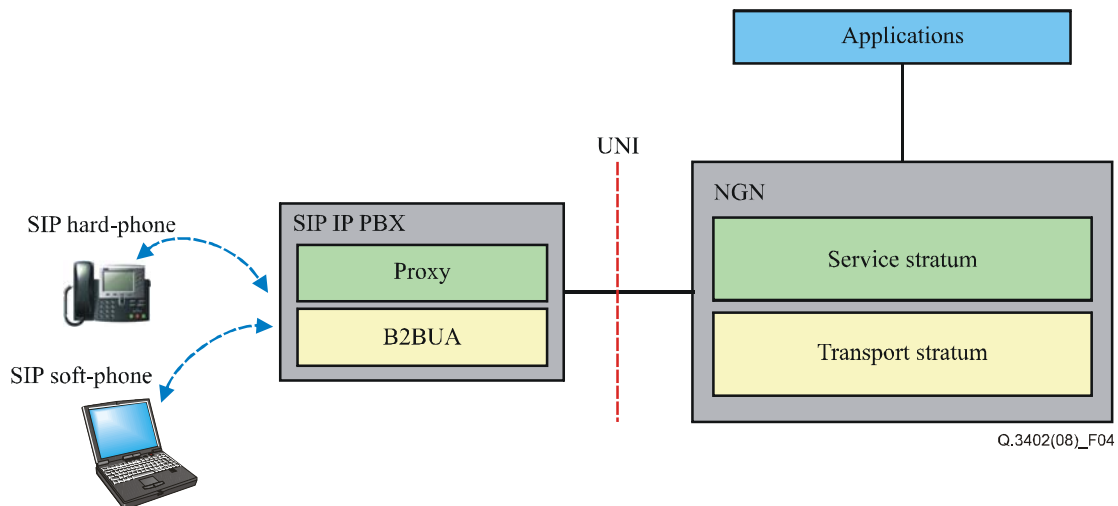


Figure 4 – Scenario for SIP IP PBX

6 Assumptions

This Recommendation is based on the following set of assumptions:

- 1) SIP/SDP is used for session control.
- 2) RTP or SRTP is used for voice and video transport; other transport protocols may be used for data applications.