Standardization Trends in ITU-T NGN UNI and NNI Signaling

Takumi Ohba[†] *and Koji Tanida*[††]

Abstract

The International Telecommunication Union, Telecommunication Standardization Sector (ITU-T) released the Recommendations for Next Generation Network (NGN) signaling for the Network Network Interface (NNI) and the User Network Interface (UNI) in March 2007 and February 2008, respectively. Consequently, the specifications for basic communications of session-oriented services are now ready. This paper overviews the NGN signaling for UNI and NNI in ITU-T.

1. Scope of NGN signaling

The first set of ITU-T (International Telecommunication Union, Telecommunication Standardization Sector) Recommendations for the Next Generation Network (NGN), called NGN Release 1, aims to provide interactive multimedia communication services (session-oriented services) based on audio and video between two parties [1], [2]. As part of NGN Release 1, the Recommendations for NGN signaling for the Network to Network Interface (NNI) and the User to Network Interface (UNI) were specified in March 2007 and February 2008, respectively. The NNI is the interface to other NGN operators and the UNI is the interface to terminals. They are defined in the NGN architecture of Y.2012 (**Fig. 1**) [3].

Those signaling Recommendations specify control signals that handle the setup and release of sessions. They also specify audio and video used during the communication, such as recommended codecs to be used, because the information, e.g., types of audio and video for session-oriented communications, is negotiated over control signals at the time of session setup.

2. Specifications of NGN signaling

The control signals of session-oriented services

over the NGN use the session initiation protocol (SIP), which is specified by the Internet Engineering Task Force (IETF), the Internet-related standards developing organization. SIP defines a communication that lasts for a definite period of time over IP (Internet protocol) networks as a *session* and provides various control functions including session setup and release as well as registration of terminals with the network.

When we say "SIP" in IETF, its specifications in IETF cover a wide range of functionalities, as is obvious from the fact that more than one hundred documents have been specified since the first publication of RFC3261 [4] in 2002, which specifies very basic procedures for SIP. However, it is not necessary to implement all of these specifications to provide session-oriented services over the NGN. Moreover, in the SIP specifications in IETF, there is a lot of room for choosing options to cover general usages. Therefore, it is necessary to sort out specifications, i.e., to perform profiling, in order to improve interoperability or make implementations easy.

Consequently, the profile for applying the SIP specifications to the NGN NNI is specified in Q.3401, and that for the UNI is specified in Q.3402. Specifically, these profiles include distinctions between mandatory (M) and optional (O) support for the document (Request for Comments (RFC)) and for messages and parameters and include clarifications and choices of options within the RFCs. The mandatory and optional distinctions at the document (RFC) level as part of the profile of the NNI in Q.3401 are listed

---

† NTT Service Integration Laboratories
††NTT Network Service Systems Laboratories
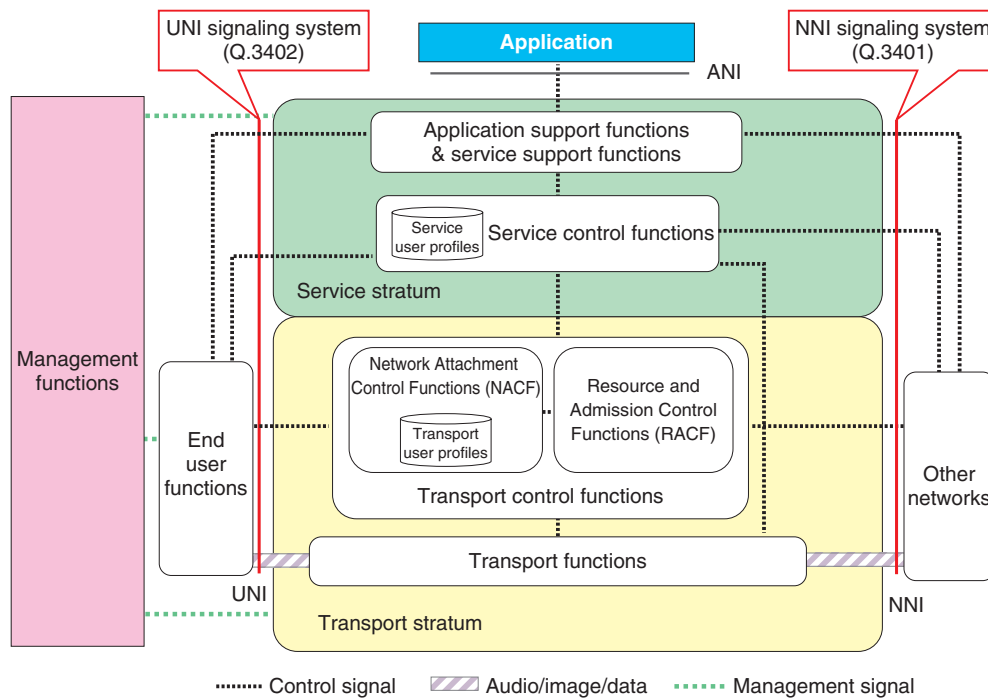 Musashino-shi, 180-8585 Japan

Fig. 1. Reference points of the NGN signaling.

in **Table 1**.

The contents of Q.3401 and Q.3402 resemble each other because SIP can be applied to both the UNI and NNI, but there are differences between them. One example is the network-asserted calling party ID (identification), which is certainly forwarded between networks, but is not configurable at the originating UNI and must be forwarded to the user at the destination UNI if the calling party ID presentation is configured.

Additionally, Q.3401 and Q.3402 define Real-time Transport Protocol (RTP) profiles, which are used for audio and video between users, and codecs recommended to be used. However, Q.3401 and Q.3402 do not specify implementations of terminals themselves, but only the signaling over the reference points—the NNI and UNI. Therefore, the codecs that are actually used in communications are determined by end-to-end negotiation at the time of session setup. A list of codecs recommended to be configured in the session setup signals for this negotiation is specified (**Table 2**).

Furthermore, as the reference point between the NGN and applications in the NGN architecture in Y.2012, the Application Network Interface (ANI) is defined as well as the NNI and UNI, as shown in Fig.

1. Y.2001 states that open APIs, e.g., Parlay-X, should be provided over the ANI. Currently, application support functions and service support functions are under study to enable the ANI specifications to be completed.

3.   Relationship with 3GPP and ETSI TISPAN

The 3rd Generation Partnership Project (3GPP) and ETSI TISPAN (European Telecommunication Standards Institute, Telecoms & Internet converged Services & Protocols for Advanced Network)) also specify the SIP profiles in the detailed internal structure of the IP Multimedia Subsystem (IMS) and the NGN [5], [6]. ITU-T focuses on the reference points between the NGN and its externals from the interoperability viewpoints while referencing 3GPP and TISPAN specifications. More precisely, the specifications of ITU-T are closely related to interconnections between NGNs or between NGN and users, and those of 3GPP/ETSI TISPAN to implementations of function blocks constructing the NGN. Therefore, the two specifications have compatibility with and complement one another, without any mutual duplication or contradiction.

In the standardization activities at ITU-T, once a

Table 1.   RFC documents supported by NNI.

Number	Title	M/O
RFC2046	Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types	O
RFC2327	SDP: Session Description Protocol	M
RFC2976	The SIP INFO Method	O
RFC3087	Control of Service Context using SIP Request-URI	O
RFC3204	MIME media types for ISUP and QSIG Objects	O
RFC3261	SIP: Session Initiation Protocol	M
RFC3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)	M
RFC3264	An Offer/Answer Model with the Session Description Protocol (SDP)	M
RFC3265	Session Initiation Protocol (SIP)-Specific Event Notification	O
RFC3311	The Session Initiation Protocol (SIP) UPDATE Method	M
RFC3312	Integration of Resource Management and Session Initiation Protocol (SIP)	O
RFC3323	A Privacy Mechanism for the Session Initiation Protocol (SIP)	M
RFC3324	Short Term Requirements for Network Asserted Identity	O
RFC3325	Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks	M
RFC3326	The Reason Header Field for the Session Initiation Protocol (SIP)	M
RFC3398	Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping	O
RFC3420	Internet Media Type message/sipfrag	O
RFC3428	Session Initiation Protocol (SIP) Extension for Instant Messaging	O
RFC3455	Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)	O
RFC3515	The Session Initiation Protocol (SIP) Refer Method	O
RFC3824	Using E.164 numbers with the Session Initiation Protocol (SIP)	O
RFC3840	Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)	O
RFC3841	Caller Preferences for the Session Initiation Protocol (SIP)	O
RFC3891	The Session Initiation Protocol (SIP) Replaces Header	O
RFC3892	The Session Initiation Protocol (SIP) Referred-By Mechanism	O
RFC3893	Session Initiation Protocol (SIP) Authenticated Identity Body (AIB) Format	O
RFC3911	The Session Initiation Protocol (SIP) Join Header	O
RFC3959	The Early Session Disposition Type for the Session Initiation Protocol (SIP)	O
RFC3960	Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)	O
RFC3966	The tel URI for Telephone Numbers	M
RFC4028	Session Timers in the Session Initiation Protocol (SIP)	M
RFC4032	Update to the Session Initiation Protocol (SIP) Preconditions Framework	O
RFC4235	An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)	O
RFC4244	An Extension to the Session Initiation Protocol (SIP) for Request History Information	O
RFC4412	Communications Resource Priority for the Session Initiation Protocol (SIP)	O
RFC4458	Session Initiation Protocol (SIP) URIs for Applications such as Voicemail and Interactive Voice Response (IVR)	O
RFC4483	A Mechanism for Content Indirection in Session Initiation Protocol (SIP) Messages	O
RFC4566	SDP: Session Description Protocol	M
RFC4694	Number Portability Parameters for the tel URI	O

M: support of RFC is mandatory.
O: support of RFC is optional.

Recommendation has been issued, it is not modified frequently because completeness is fully achieved before publication. However, at 3GPP/ETSI TISPAN, documents are published when they achieve certain levels of completeness; thereafter, amendments are issued repeatedly as necessary. Policies for revising documents are considerably different in these standardization bodies. Q.3401 (NNI) and Q.3402 (UNI)

Table 2.   Codecs in the codec list.

Default	G.711 A/mu law: Pulse Code Modulation (PCM) of Voice Frequencies
Recommended	AMR: Adaptive Multi-Rate EVRC: Enhanced Variable Rate Codec G.729: Coding of Speech at 8 kbit/s using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP) G.729A: Reduced complexity 8-kbit/s CS-ACELP speech codec G.722.1: 7-kHz Audio-coding at 24 and 32 kbit/s for Hands Free Operation in Systems with Low Frame Loss G.726: 40-, 32-, 24-,16-kbit/s Adaptive Differential Pulse code Modulation (ADPCM) AMR WB: Wideband Coding of Speech at around 16 kbit/s Using Adaptive Multi-rate Wideband (AMR-WB) VMR-WB: Source-Controlled Variable-Rate Multimode Wideband Speech Codec G.722: 7-kHz Audio Coding within 64 kbit/s G.729.1: G.729 based Embedded Variable bit-rate coder: An 8–32-kbit/s scalable wideband coder bitstream interoperable with G.729 T.140: Protocol for multimedia application text conversation

were specified with sufficient consistency with the 3GPP and ETSI/TISPAN specifications at the time of their publication by ITU-T.

4.   Relationship with domestic standardization activities

The Telecommunications Technology Committee in Japan (TTC) issued two technical reports, TR-9025 and TR-9024 [7], based on existing standards for operators' SIP networks, but not for NGN, ahead of ITU-T. These corresponded to the NNI and UNI respectively, with limited functions.

The NNI was already standardized as JT-Q.3401 at TTC following publication at ITU-T in 2007 and the UNI was scheduled to be done in 2008. These TTC standards, which are downstream from ITU-T Recommendations, adhere directly to the contents in the previous technical reports issued, and they enhance the functionalities so that they conform to the NGN architecture perfectly.

5.   Future activities

The NGN signaling in ITU-T first specifies the NNI and UNI for basic session-oriented services. NTT Laboratories has proactively contributed to ITU-T for the international standardization and downstreaming of ITU-T Recommendations to TTC. These specifi-

cations are very important for providing various services of NTT's NGN and for interconnecting to other carriers' NGNs and user terminals. All the various NGN services provided by NTT fully conform to the ITU-T Recommendations regarding NGN signaling, e.g., Q.3401, and the corresponding TTC standards.

ITU-T is studying more advanced services than the basic session-oriented services, for example, customized ringback tones and ringing tones. Based on its pioneering experiences in Japan's broadband environment, NTT Laboratories will continue to contribute to NGN international standardization.

References

[1]   N. Morita, H. Imanaka, O. Kamatani, T. Ohba, and K. Tanida, "Overview and Status of NGN Standardization Activities at ITU-T," NTT Technical Review, Vol. 5, No. 11, 2007.
https://www.ntt-review.jp/archive/ntttechnical.php?contents=ntr2007 11gls.html

[2]   N. Morita and H. Imanaka, "ITU-T NGN Release 1 Ready," NTT Technical Review, Vol. 6, No. 11, 2008.
https://www.ntt-review.jp/archive/ntttechnical.php?contents=ntr2008 11gls.html

[3]   http://www.itu.int/rec/T-REC/e

[4]   http://www.ietf.org/rfc.html

[5]   A. Kurokawa and I. Higashi, "Standardization Trends of the Next Generation Network in ETSI TISPAN," NTT Technical Review, Vol. 4, No. 6, pp. 53–57, 2006.
https://www.ntt-review.jp/archive/ntttechnical.php?contents= ntr200606053.pdf

[6]   http://www.3gpp.org/Specification-Numbering

[7]   http://www.ttc.or.jp/cgi/document-db/index.html (in Japanese).

Takumi Ohba
Senior Research Engineer, NGN System Architecture Project, NTT Service Integration Laboratories.
He received the M.S. degree in physics from the University of Tokyo, Tokyo, in 1993. He joined NTT in 1993 and has been studying ATM and IP signaling. Since July 2005, he has been promoting globalization of the next-generation network in SIP standardization. He is a member of the Institute of Electronics, Information and Communication Engineers of Japan.

Koji Tanida
Fifth Promotion Project, NTT Network Service Systems Laboratories.
He received the M.E. degree in electronics from Tohoku University, Miyagi, in 2004. He joined NTT in 2004. He has been engaged in developing a call control server for the NGN and studying IP signaling. Since January 2006, he has been promoting globalization of the NGN in SIP standardization.