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

NGN NNI signalling profile (protocol set 1)

ITU-T Recommendation Q.3401



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ITU-T Recommendation Q.3401

NGN NNI signalling profile (protocol set 1)

Summary

ITU-T Recommendation Q.3401 defines the IP network-to-network interface (NNI) signalling profile for use between NGN carriers of protocol set 1 for voiceband services.

Source

ITU-T Recommendation Q.3401 was approved on 9 March 2007 by ITU-T Study Group 11 (2005-2008) under the ITU-T Recommendation A.8 procedure.

Keywords

NGN, profile, SDP, SIP, signalling.

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ITU-T Recommendation Q.3401

NGN NNI signalling profile (protocol set 1)

1 Scope

This Recommendation contains a service-level profile, i.e., SIP/SDP interface description, between two network operators (NNI signalling profile), where the two different network operators may support different SIP/SDP profiles (i.e., they differ in terms of the SIP extensions, SIP information elements and SDP lines which are supported). Transport-level profile, e.g., RTP, is described as necessary as the media are described in the service-level signalling.

NOTE 1 – Specification of the interworking between the SIP/SDP profile contained in this Recommendation and the SIP/SDP profiles in each operator's network (i.e., the functions contained inside the interworking entity/entities at the network boundary) is out of scope of this Recommendation.

For the protocol set 1 of the NGN NNI signalling profile, this Recommendation covers the voiceband services such as VoIP (audio, text and so on), DTMF and T.38 fax.

NOTE 2 – Support for a mobility management interface is not supported in this version of this interface. However, nomadicity of users between different networks is supported.

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.

2.1 ITU-T references

- [ITU-T G.711] ITU-T Recommendation G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.
- [ITU-T G.722] ITU-T Recommendation G.722 (1988), *7 kHz audio-coding within 64 kbit/s*.
- [ITU-T G.722.1] ITU-T Recommendation G.722.1 (2005), *Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss*.
- [ITU-T G.722.2] ITU-T Recommendation G.722.2 (2003), *Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)*.
- [ITU-T G.726] ITU-T Recommendation G.726 (1990), *40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)*.
- [ITU-T G.729] ITU-T Recommendation G.729 (1996), *Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)*.
- [ITU-T G.729A] ITU-T Recommendation G.729 Annex A (1996), *Reduced complexity 8 kbit/s CS-ACELP speech codec*.
- [ITU-T G.729.1] ITU-T Recommendation G.729.1 (2006), *G.729-based embedded variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729*.
- [ITU-T Q.761] ITU-T Recommendation Q.761 (1999), *Signalling System No. 7 – ISDN User Part functional description*.

- [ITU-T Q.762] ITU-T Recommendation Q.762 (1999), *Signalling System No. 7 – ISDN User Part general functions of messages and signals.*
- [ITU-T Q.763] ITU-T Recommendation Q.763 (1999), *Signalling System No. 7 – ISDN User Part formats and codes.*
- [ITU-T Q.764] ITU-T Recommendation Q.764 (1999), *Signalling System No. 7 – ISDN User Part signalling procedures.*
- [ITU-T Q.1912.5] ITU-T Recommendation Q.1912.5 (2004), *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part.*
- [ITU-T T.38] ITU-T Recommendation T.38 (2005), *Procedures for real-time Group 3 facsimile communication over IP networks.*
- [ITU-T T.140] ITU-T Recommendation T.140 (1998), *Protocol for multimedia application text conversation.*
- [ITU-T Y.2012] ITU-T Recommendation Y.2012 (2006), *Functional requirements and architecture of the NGN release 1.*

2.2 IETF references

2.2.1 Service-level signalling specifications

- [RFC 2046] IETF RFC 2046 (1996), *Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types.*
- [RFC 2327] IETF RFC 2327 (1998), *SDP: Session Description Protocol.*
- [RFC 2976] IETF RFC 2976 (2000), *The SIP INFO Method.*
- [RFC 3087] IETF RFC 3087 (2001), *Control of Service Context using SIP Request-URI.*
- [RFC 3204] IETF RFC 3204 (2001), *MIME media types for ISUP and QSIG Objects.*
- [RFC 3261] IETF RFC 3261 (2002), *SIP: Session Initiation Protocol.*
- [RFC 3262] IETF RFC 3262 (2002), *Reliability of Provisional Responses in the Session Initiation Protocol (SIP).*
- [RFC 3264] IETF RFC 3264 (2002), *An Offer/Answer Model with the Session Description Protocol (SDP).*
- [RFC 3265] IETF RFC 3265 (2002), *Session Initiation Protocol (SIP)-Specific Event Notification.*
- [RFC 3311] IETF RFC 3311 (2002), *The Session Initiation Protocol (SIP) UPDATE Method.*
- [RFC 3312] IETF RFC 3312 (2002), *Integration of Resource Management and Session Initiation Protocol (SIP).*
- [RFC 3323] IETF RFC 3323 (2002), *A Privacy Mechanism for the Session Initiation Protocol (SIP).*
- [RFC 3324] IETF RFC 3324 (2002), *Short Term Requirements for Network Asserted Identity.*
- [RFC 3325] IETF RFC 3325 (2002), *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks.*
- [RFC 3326] IETF RFC 3326 (2002), *The Reason Header Field for the Session Initiation Protocol (SIP).*

- [RFC 3398] IETF RFC 3398 (2002), *Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping*.
- [RFC 3420] IETF RFC 3420 (2002), *Internet Media Type message/sipfrag*.
- [RFC 3428] IETF RFC 3428 (2002), *Session Initiation Protocol (SIP) Extension for Instant Messaging*.
- [RFC 3455] IETF RFC 3455 (2003), *Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)*.
- [RFC 3515] IETF RFC 3515 (2003), *The Session Initiation Protocol (SIP) Refer Method*.
- [RFC 3824] IETF RFC 3824 (2004), *Using E.164 numbers with the Session Initiation Protocol (SIP)*.
- [RFC 3840] IETF RFC 3840 (2004), *Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)*.
- [RFC 3841] IETF RFC 3841 (2004), *Caller Preferences for the Session Initiation Protocol (SIP)*.
- [RFC 3891] IETF RFC 3891 (2004), *The Session Initiation Protocol (SIP) Replaces Header*.
- [RFC 3892] IETF RFC 3892 (2004), *The Session Initiation Protocol (SIP) Referred-By Mechanism*.
- [RFC 3893] IETF RFC 3893 (2004), *Session Initiation Protocol (SIP) Authenticated Identity Body (AIB) Format*.
- [RFC 3911] IETF RFC 3911 (2004), *The Session Initiation Protocol (SIP) Join Header*.
- [RFC 3959] IETF RFC 3959 (2004), *The Early Session Disposition Type for the Session Initiation Protocol (SIP)*.
- [RFC 3960] IETF RFC 3960 (2004), *Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)*.
- [RFC 3966] IETF RFC 3966 (2004), *The tel URI for Telephone Numbers*.
- [RFC 4028] IETF RFC 4028 (2005), *Session Timers in the Session Initiation Protocol (SIP)*.
- [RFC 4032] IETF RFC 4032 (2005), *Update to the Session Initiation Protocol (SIP) Preconditions Framework*.
- [RFC 4235] IETF RFC 4235 (2005), *An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)*.
- [RFC 4244] IETF RFC 4244 (2005), *An Extension to the Session Initiation Protocol (SIP) for Request History Information*.
- [RFC 4412] IETF RFC 4412 (2006), *Communications Resource Priority for the Session Initiation Protocol (SIP)*.
- [RFC 4458] IETF RFC 4458 (2006), *Session Initiation Protocol (SIP) URIs for Applications such as Voicemail and Interactive Voice Response (IVR)*.
- [RFC 4483] IETF RFC 4483 (2006), *A Mechanism for Content Indirection in Session Initiation Protocol (SIP) Messages*.
- [RFC 4566] IETF RFC 4566 (2006), *SDP: Session Description Protocol*.
- [RFC 4694] IETF RFC 4694 (2006), *Number Portability Parameters for the tel URI*.

2.2.2 Transport-level specifications

- [RFC 2833] IETF RFC 2833 (2000), *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*.
- [RFC 3267] IETF RFC 3267 (2002), *Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs*.
- [RFC 3389] IETF RFC 3389 (2002), *Real-time Transport Protocol (RTP) Payload for Comfort Noise*.
- [RFC 3550] IETF RFC 3550 (2003), *RTP: A Transport Protocol for Real-Time Applications*.
- [RFC 3551] IETF RFC 3551 (2003), *RTP Profile for Audio and Video Conferences with Minimal Control*.
- [RFC 4103] IETF RFC 4103 (2005), *RTP Payload for Text Conversation*.

2.3 ETSI references

- [EN 301 703] ETSI EN 301 703 V7.0.2 (1999), *Digital cellular telecommunications system (Phase 2+) (GSM); Adaptive Multi-Rate (AMR); Speech processing functions; General description (GSM 06.71 version 7.0.2 Release 1998)*.

2.4 Other references

- [TIA-127-A] TIA-127-A (2004), *Enhanced Variable Rate Codec Speech Service Option 3 for Wideband Spread Spectrum Digital Systems*.
- [TIA-1016] TIA-1016-A (2006), *Source-Controlled Variable-Rate Multimode Wideband Speech Codec (VMR-WB), Service Options 62 and 63 for Spread Spectrum Systems*.

3 Definitions

For SIP- and SDP-specific terminology, reference shall be made to [RFC 3261], [RFC 3264] and [RFC 2327]. 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 codec list: A codec list defines the codecs that should be announced in SIP/SDP messages exchanged over the NNI based on a bilateral agreement established between the NGNs.

NOTE – The purpose of the codec list is to promote interoperability, limit the number of transcodings on network connections and possibly improve network resource management.

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

3.3 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 [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 behaviors are defined in [RFC 3261].) A B2BUA reformulates a message before sending it as a new request.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

ACM	Address Complete Message
AMR	Adaptive Multirate (codec)
ANM	Answer Message
B2BUA	Back-to-Back User Agent
BGC-FE	Breakout Gateway Control Functional Entity
CSC-FE	Call Session Control Functional Entity
EVRC	Enhanced Variable Rate Codec
FQDN	Fully Qualified Domain Name
IAM	Initial Address Message
IBC-FE	Interconnection Border gateway Control Functional Entity
IBG-FE	Interconnection Border Gateway Functional Entity
ISUP	ISDN User Part
MG	Media Gateway
MGC-FE	Media Gateway Control Functional Entity
MIME	Multipurpose Internet Mail Extensions
NGN-TE	NGN Terminal Equipment
NNI	Network-to-Network Interface
PSTN	Public Switched Telephone Network
PT	Payload Type
REL	Release
RFC	Request For Comments
RLC	Release Complete
RTCP	RTP Control Protocol
RTP	Real-time Transport Protocol
SCF	Service Control Functions
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SG-FE	Signalling Gateway Functional Entity
SIP	Session Initiation Protocol
SIP-I	SIP with encapsulated ISUP
SIPS	Session Initiation Protocol Secure
TCP	Transmission Control Protocol
TMG-FE	Trunk Media Gateway Functional Entity
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

5 Reference model

Figure 5-1 illustrates the interface covered by this Recommendation in the NGN architecture defined in [ITU-T Y.2012].

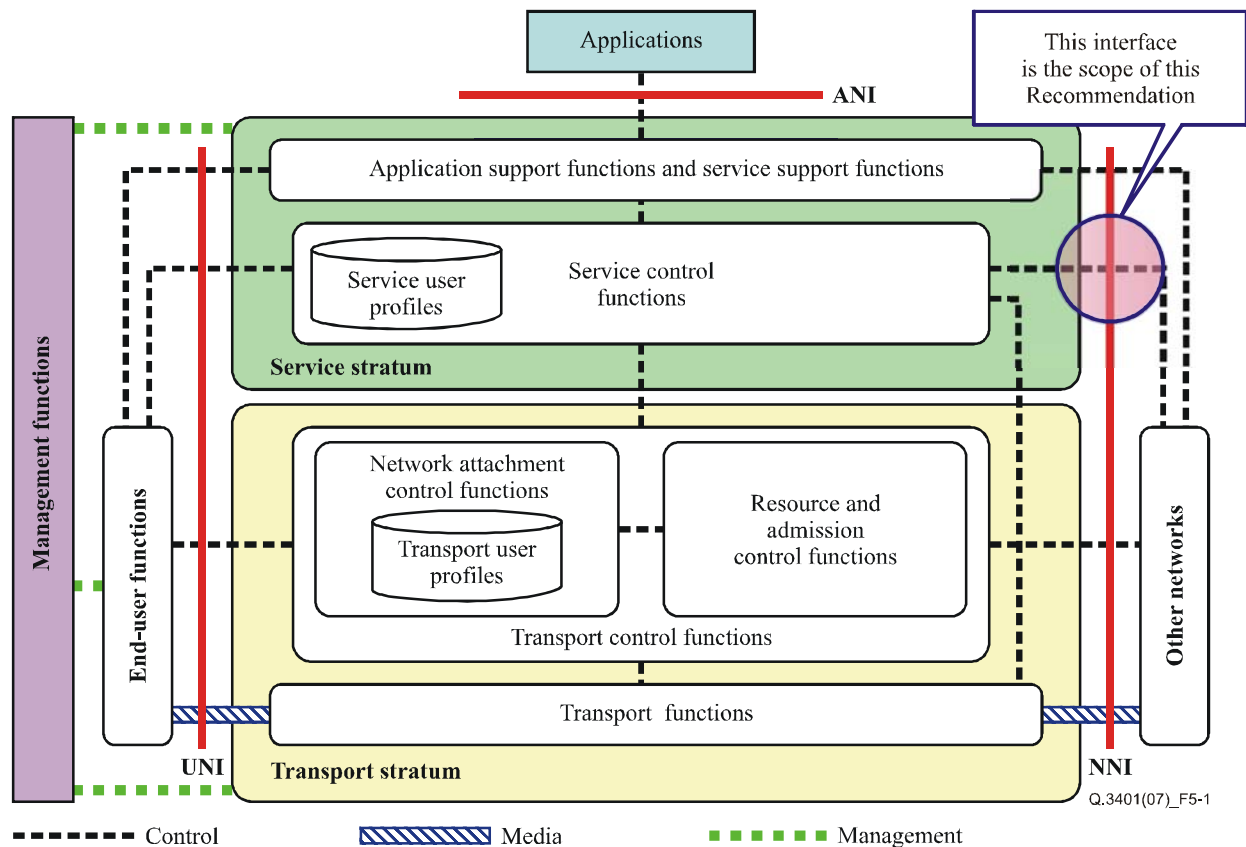


Figure 5-1 – The interface covered by this Recommendation within the NGN architecture

Figure 5-2 illustrates the interconnection reference model for the NNI.

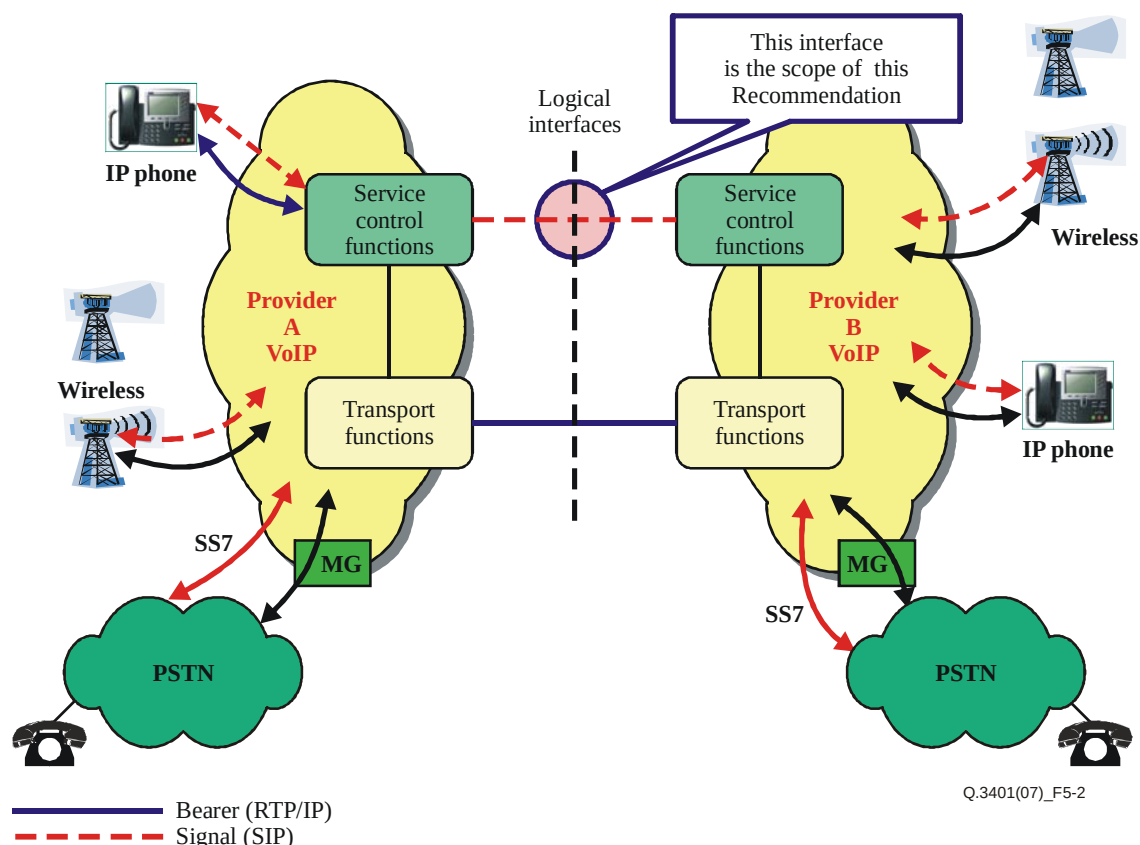


Figure 5-2 – VoIP interconnection reference model

6 Assumptions

This Recommendation is based on the following set of assumptions:

- 1) It is recognized that data-oriented call control (e.g., http to a web portal on an application server), in addition to SIP-oriented call control, may occur across the NNI. For example, a network-based service may allow a user to initiate calls or control the disposition of incoming calls through a web browser interface using http. It is assumed that such data-oriented call control involves no differentiated treatment from other data traffic, and so is not described further in this Recommendation. Differentiated treatment may be desirable in some cases.
- 2) The logical interfaces associated with the service control functions in the service stratum and the transport functions in the transport stratum enable interconnection between two service provider networks in a peering environment.
- 3) Each provider may use a set of service control functions and transport functions to connect to multiple peer networks.
- 4) SIP back-to-back user agent (B2BUA) functions may be used in support of network interconnection.
- 5) Only the network-to-network interface (NNI) is supported between two peering VoIP carriers. There may be an IP transit network between two peering VoIP service providers.
- 6) SIP is used for service-level signalling. The SIP messages may contain MIME encapsulated ISUP in order to facilitate interoperability with the PSTN. ISUP is specified in [ITU-T Q.761]-[ITU-T Q.764].

- 7) RTP is used for voice transport.
- 8) IPv4 is supported at the NNI. Support for IPv6 is optional based on bilateral agreement.
- 9) The IPv6 operator is responsible for IPv4/IPv6 interworking.

7 Media availability in a SIP session

The following apply to any media session established across the NNI using SIP:

- a) The terminating-side network of the NNI shall pass any media packets in the direction toward the originating party as soon as they are available. A primary reason is to allow the caller to hear inband call progress tones if PSTN interworking is encountered on a voice call.
- b) The originating-side network of the NNI:
 - shall pass media packets from the originating party in the direction toward the terminating party upon and after receiving a final SDP answer within a SIP 2xx response to the INVITE for normal dialog;
 - may pass media packets from the originating party in the direction toward the terminating party as early as the first SDP answer has occurred, which is in a SIP 1xx response to the INVITE when early dialog has been set up. A network, as a policy, may choose not to send media packets from the originating party until the final SDP offer/answer has been made in order to avoid theft-of-service in cases where usage-sensitive billing is employed.
- c) As per [RFC 3261], once a SIP dialog has ended, the flow of media packets shall be halted.
- d) The absence of media packets across the NNI over any time interval in either direction shall not be taken by either network as a sufficient reason to clear the SIP session. When the status of media flows is active according to the SDP negotiation, the absence of packets across the NNI for a given duration may constitute a reason to clear the SIP session.

NOTE – The absence of packets across the NNI for a given duration can be a reason to clear the SIP session only when it is sure that it is because of the failure.

8 Codec

8.1 Codec list

It is the responsibility of entities at the rim of the NGN (e.g., NGN-TE) and network equipment originating and terminating the NGN IP media flows to negotiate and select a common codec for each "end-to-end" media session. Therefore, the NGN shall allow end-to-end negotiation within the agreed codec list between NGN entities (terminals and network elements) and may allow it outside the list based on its network policy.

NOTE 1 – In case a common codec cannot be negotiated, this Recommendation does not provide procedures for the NNI.

NOTE 2 – In the interest of promoting interoperability, limiting the number of transcodings on network connections and possibly improving network resource management, it is desirable that the NGNs establish bilateral agreements about the codec list. SIP/SDP messages exchanged over the NNI indicate a request to use one or more of the codecs in this codec list.

The way of handling messages with codecs that are not in the codec list or with no codec in the list depends on the network policy, i.e., some networks may allow the use of codecs that are not in the codec list, while others may reject such messages.

Agreement on a codec list does not put any direct requirement on the codecs that have to be implemented in the network for transcoding purposes, nor does it mean that terminals shall support