ATIS-1000040

PROTOCOL SUITE PROFILE FOR IP NETWORK TO NETWORK INTERCONNECTION RELEASE 1.0

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ATIS-1000040, *Protocol Suite Profile for IP Network to Network Interconnection Release 1.0*

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ATIS Standard on

PROTOCOL SUITE PROFILE FOR IP NETWORK TO NETWORK INTERCONNECTION RELEASE 1.0

Alliance for Telecommunications Industry Solutions

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Abstract

This document identifies a set of protocols and specifies their profile so that signaling, media, and network related parameters can be uniformly and consistently utilized across the Interconnection interface. The objective is to support a service seamlessly across an IP Network to Network Interconnection as identified by the test scenarios defined in ATIS-1000041.

FOREWORD

The Alliance for Telecommunication Industry Solutions (ATIS) serves the public through improved understanding between providers, customers, and manufacturers. The Packet Technologies and Systems Committee (PTSC) develops and recommends standards and technical reports related to services, architectures, and signaling, in addition to related subjects under consideration in other North American and international standards bodies. PTSC coordinates and develops standards and technical reports relevant to telecommunications networks in the U.S., reviews and prepares contributions on such matters for submission to U.S. ITU-T and U.S. ITU-R Study Groups or other standards organizations, and reviews for acceptability or per contra the positions of other countries in related standards development and takes or recommends appropriate actions.

The mandatory requirements are designated by the word *shall* and recommendations by the word *should*. Where both a mandatory requirement and a recommendation are specified for the same criterion, the recommendation represents a goal currently identifiable as having distinct compatibility or performance advantages. The word *may* denotes a optional capability that could augment the standard. The standard is fully functional without the incorporation of this optional capability.

Shall indicates a strict requirement to be exercised and validated. *May* indicates an optional feature that may be supported, but is not required and will not be validated.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, PTSC, 1200 G Street NW, Suite 500, Washington, DC 20005.

The Next Generation Carrier Interconnection (NG-CI) Task Force was responsible for the development of this document.

At the time of consensus on this document, PTSC, which was responsible for its development, had the following leadership:

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Protocol Suite Profile for IP Network to Network Interconnection Release 1.0

1 SCOPE, PURPOSE, & APPLICATION

The ATIS Next Generation Carrier Interconnection (NG-CI) Task Force (TF) TRs specify a standards-based interface for network-to-network interconnection for next generation network services.

This document identifies a set of protocols and specifies their profile so that signaling, media, and network related parameters can be uniformly and consistently utilized across the Interconnection interface. The objective is to support a service seamlessly across an IP Network to Network Interconnection as identified by the test scenarios defined in ATIS-1000041.

The purpose of this document is to specify the protocol profiles which will be used in order to support the interoperability of a service between providers.

This document may be used to develop detailed system conditions to be checked for standards conformance and service interoperability.

It is a component of a family of documents that taken as a whole provides an IP network to network interconnection (NNI) guideline that supports next generation service interoperability.

This Release 1.0 addresses Basic VoIP service, Basic VoIP service with DTMF support, and Basic voice service with T.38 Fax support for native IP endpoints.

Post Release 1.0 will address support for RPH as well as services and capabilities such as domestic mobile-to-mobile voice service interconnect over IP and international mobile and nomadic endpoints. Other services like wireline-TDM, managed peer-to-peer, mobile-to-mobile peering, IPTV, multimedia services, gaming, fixed-mobile service convergence, etc., may also be considered.

The ATIS NG-CI TRs cover four aspects of interconnection and this specific TR has three other companion TRs that together address VoIP services Interconnection over IP-based links/networks.

The companion TRs include the following:

- The "Test Suite" TR (ATIS-1000041) provides the tests used for IP NNI interconnection testing in order to support service interoperability.
- The "Technical Parameters" TR (ATIS-1000038) provides a description of the Provider data for collection and eventual exchange for service planning in order to support service interoperability.
- The "Test Configuration" TR (ATIS-1000039) provides the network interface configuration in order to support service interoperability.

Establishing and specifying the fixed protocol and configuration profiles, and the variable or selectable parameters that can then be applied uniformly and consistently for interconnects, will ensure a reliable level of conformance to a standard that supports the establishment of successful interoperability.

1.1 Assumptions

1. Due to the variety of security configurations and credentials possible between providers, the use of IPSEC or TLS to support signaling or media streams will be subject to agreement between those providers and will not be defined within this document.
2. The level of information presented here is intended to be sufficient to support interoperability testing events; however, additional work will be required to develop actual test scripts based on the test scenarios, configurations, and protocol suites presented
3. It is understood that test SIP device endpoint E.164 addresses will need to be exchanged prior to testing. SIP URIs converted from TEL URI format will be used to convey the E164 addresses. (See ATIS-1000009.2006 for example URIs.)
4. IPv4 is assumed unless otherwise stated.
5. The term Provider is used to generically represent all types of parties.

2 References

The following standards contain provisions which, through reference in this text, constitute provisions of this ATIS Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this ATIS Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

2.1 ATIS[1] (normative)

ATIS-1000009.2006, *IP Network-to-Network Interface (NNI) Standard for VoIP*.

ATIS-1000026.2008, *Session/Border Control Functions and Requirements*.

2.2 IETF[2] (normative)

RFC 4244, *An Extension to the Session Initiation Protocol (SIP) for Request History Information*.

2.3 ATIS[3] (informative)

PTSC-SAC-2010-033R1, *TR on History Info in Carrier Network*.

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[1] These documents are available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G Street N.W., Suite 500, Washington, DC 20005. < https://www.atis.org/docstore/default.aspx >

[2] This document is available from the Internet Engineering Task Force (IETF). < http://www.ietf.org >

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3 ACRONYMS & ABBREVIATIONS

ANSI	American National Standards Institute
ATIS	Alliance for Telecommunications Industry Solutions
DTMF	Dual-Tone Multi-Frequency
ENUM	Electronic NUMbering
HTTP	Hypertext Transfer Protocol
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPSec	Internet Protocol Security
IPv4	Internet Protocol Version 4
IPv6	Internet Protocol Version 6
ITU	International Telecommunications Union
MSF	Multi-Service Forum
NG-CI TF	Next Generation Carrier Interconnection Task Force
NNI	Network to Network Interface
OAMP	Operations, Administration, Maintenance and Provisioning
PTSC	Packet Technologies and Systems Committee
QoS	Quality of Service
RFC	Request for Comments
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SIP	Session Initiated Protocol
SP	Service Provider
TCP	Transmission Control Protocol
TLS	Transport Layer Security
UDP	User Datagram Protocol
URI	Universal (or Uniform) Resource Identifier (or Indicator)
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network

4 APPLICATION/SERVICE LAYER PROTOCOL PROFILE

4.1 Basic Voice Service

Each reference to standards document below is followed either by a list of exceptions to the standards or by additional conditions not explicitly stated. The specifications are used for an NNI and all requirements are written for that purpose. The term "across the NNI" may be used explicitly to emphasize the scope of the requirements. The term "may be ignored" is used to limit the scope of coverage of the protocol profile without imposing unnecessary changes to the systems under test which may also be configured for a different purpose. The term is used to modify the optional requirements that are known to be out of scope of Release 1.0.

It is important to note that while the standards and references cited may be normative for the interface, for the purposes of this document the requirements explicitly highlighted or excluded are considered normative only for defining and testing the IP Network to Network Guideline as developed by the ATIS PTSC NG-CI Task Force. The "requirements" as stated below shall be noted in that context and are to be considered as test criteria in support of ATIS-1000041.

4.1.1 Signaling

4.1.1.1 SIP Profile for SIP based VoIP Service Interconnection

ATIS-1000009.2006 includes a detailed SIP profile based on RFC 3261.

The following is the minimal SIP protocol profile for NGCI Phase 1:

- Section 11.1.1 of ATIS-1000009.2006 and IETF RFC 3261:
    - Specifications in clause 11.1.1 from ATIS-1000009.2006 shall be supported except as noted in the following.
        1. The SIP method **Registration** will not be tested across the NNI. This is optional in NNI but out of scope for Release 1.0.
        2. The SIP media session negotiation shall support the use of Session Description Protocol (SDP) via message body type "**application/sdp**".
        3. The SIP method **Invite** shall include a media session offer.
        4. The SIP method **re-Invite** shall include a media session offer.
        5. The first SIP reliable non-failure responses, 200 (OK), 180 (Ringing), and 183 (Session Progress) sent reliably, to the **INVITE** or **re-INVITE** shall include a media connection answer.
        6. In response to a SIP method **Cancel**, the SIP 200 (OK) shall be sent for the received Cancel request and SIP 487 (Request Terminated) shall be sent to the corresponding invite transaction to cancel the Invite request. [see RFC 3261]
        7. The SIP method **Options** shall be sent only in the context of a SIP Invite dialog. [RFC:11, use of may for dialog]
        8. In response to a SIP method **Options**, the SIP 200 (OK) shall include at least SIP headers Supported, Accept, and Allow. [see RFC 3261]
        9. The SIP **Invite** dialog shall not be considered canceled as if a SIP **Cancel** request is received when the underlying TCP transport protocol is lost. [new]
        10. The transport of SIP messages over UDP and Fragmented UDP shall both be supported. [new]
        11. The SIP header **Alert-Info** if present may be forwarded across the NNI but its effect on the UA is subject to local policy.

            NOTE: This is optional in NNI spec.
        12. The SIP header **Content-Language** shall support the value "en".

            NOTE: This is optional in NNI spec.

13. The SIP header **Content-Disposition** shall use the value "session."

    NOTE: The NNI spec allows other values, however this is the only value that will be tested.

14. The SIP header **Content-Language** shall support the value "en".

    NOTE: This is optional in NNI spec.

15. The SIP header **Authentication-Info** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
16. The SIP header **Authorization** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
17. The SIP header **Expire** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
18. The SIP header **Error-Info** shall be tested. Note: This is optional in the NNI spec.
19. The SIP header **In-Reply-To** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
20. The SIP header **Min-Expires** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
21. The SIP header **MIME-Version** shall be tested only for "1.0". Note that other values are supported in the NNI spec.
22. The SIP header **Max-Forward** of any message shall have a value equal to or greater than 35 when it is delivered across the NNI. [RFC call for 70: 35 is allocated to each side of the NNI.]
23. The SIP header **Organization** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
24. The SIP header **Reply-To** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
25. The SIP header **Retry-After** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
26. The SIP header **Supported** shall be tested with standards-track RFCs' option tags unless otherwise stated in this document.
27. The SIP header **Require** and **Proxy-Require** shall be tested with standards-track RFCs' option tags unless otherwise stated in this document.
28. The SIP header **Priority** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
29. The SIP header **Server** shall be tested by the last proxy server before crossing the NNI and shall provide the software and firmware versions information. This is optional in the NNI spec. [w. new]
30. The SIP header **Subject** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
31. The SIP header **User-Agent** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.

32. The SIP **WWW-Authenticate** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
33. The SIP Usage of **HTTP Authenticated** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.
34. The SIP **Warning** will be supported. This is optional in the NNI spec but out of scope for Release 1.0.
35. The SIP **S/MIME** may be ignored. This is optional in the NNI spec but out of scope for Release 1.0.

- IETF RFC 2046, *Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types*, November 1996.
    1. Testing of this RFC is out of scope for Release 1.0.
- IETF RFC 3204, *MIME media types for ISUP and QSIG Objects*, December 2001.
    1. Testing of this RFC is out of scope for Release 1.0.
- IETF RFC 3264, *An Offer/Answer Model with the Session Description Protocol (SDP)*, June 2002.
    1. The RFC 3264 shall be complied with in handling SIP **Offer-Answer** media coordination procedures.
    2. The Media Port and the existing media session using a previously agreed on parameters shall continue if the offered codec in a re-Invite is not supported. [based on RFC 3264]
- IETF RFC 3323, *A Privacy Mechanism for the Session Initiation Protocol (SIP)*, November 2002.
    1. Testing of this RFC is out of scope for Release 1.0.
- IETF RFC 3325, *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks*, November 2002.
    1. Testing of this RFC is out of scope for Release 1.0.
- IETF RFC 3326, *The Reason Header Field for the Session Initiation Protocol (SIP)*, December 2002.
    1. The SIP header **Reason Header** field may be ignored. This is out of scope for Release 1.0.
- IETF RFC 3420, *Internet Media Type message/sipfrag*, November 2002.
    1. **s/MIME** is out of scope for Release 1.0.
- IETF RFC 3515, *The Session Initiation Protocol (SIP) Refer Method*, April 2003.
    1. The SIP **Refer** Method is out of scope for Release 1.0. The support of RFC 3515 is contingent upon whether Call Center Transfer Services are supported.
- IETF RFC 3824, *Using E.164 numbers with the Session Initiation Protocol (SIP)*, June, 2004.
    1. Testing of this RFC is out of scope for Release 1.0.
- ATIS-1000009.2006, clause 10/IETF RFC 3966, The *tel URI for Telephone Calls*, December 2004.
    1. SIP **URI format** shall be tested in the form of :

       "sip:+1NPANXXXXXX@host;user=phone"

       Other formats are out of scope for Release 1.0.

- IETF RFC 3911, *The Session Initiation Protocol (SIP) "Join" Header*, September 2004.
    1. Testing of this RFC is out of scope for Release 1.0. The support of RFC 3911 is contingent upon whether Call Center Transfer Services are supported.
- IETF RFC 3892, *The SIP Referred-By Mechanism*, September 2004.
    1. Testing of this RFC is out of scope for Release 1.0. The support of RFC 3892 is contingent upon whether Call Center Transfer Services are supported.
- IETF RFC 3891, *The Session Initiation Protocol (SIP) "Replaces" Header*, September 2004.
    1. Testing of this RFC is out of scope for Release 1.0. The support of RFC 3891 is contingent upon whether Call Center Transfer Services are supported.
- IETF RFC 4412, *Communications Resource Priority for the Session Initiation Protocol (SIP)*, February 2006.
    1. Testing of this RFC will be supported; however, there are no specific requirements to be highlighted or excluded.
- IETF RFC 4028, *Session Timers in SIP*, February 2005.
    1. The SIP Timer refresh mechanism is out of scope for Release 1.0.
- IETF RFC 3960, *Early Media and Ringback Tone Generation in the Session Initiation Protocol*, December 2004.
    1. After a SIP method **Invite** with offer has been sent or received across the NNI, the Media Ports on both sides of the NNI shall be open accordingly.
    2. The SIP **180 Ringing** with no SDP shall be accepted (and a local ring back tone is assumed to be generated accordingly).
    3. When an SIP **180 Ringing** with SDP is sent or received, the Media Ports on both sides of the NNI shall be open for the specified media packets.
    4. When an SIP **183 Session Progress** is received, the Media Port may only admit the media packets associated with the first RTP stream until a subsequent response is received.
    5. The RFC 2543 call hold mechanism using the hold SDP (c=0.0.0.0) may be ignored. Testing of this RFC is out of scope for Release 1.0.
- IETF RFC 4694, *Number Portability Parameters for the "tel" URI*.
    1. Testing of this RFC is out of scope for Release 1.0.
- IETF RFC 3311, *The Session Initiation Protocol (SIP) UPDATE method*.
    1. This is out of scope for Release 1.0.
- Draft-ietf-levy-diversion-11 (expired August 2010)
    1. This is out of scope for Release 1.0.
- IETF RFC 4244, *An Extension to SIP for Request History Information*.
    1. This is out of scope for Release 1.0.

4.1.1.2 SDP

- IETF RFC 4566, *SDP: Session Description Protocol*.
    1. SDP unicast session shall be supported.
    2. SDP address via IPv4 format shall be supported.
    3. SDP for multi-cast sessions is out of scope for Rel. 1.0.
    4. SDP address format via FQDN is out of scope for Rel. 1.0.

4.1.2 Media

4.1.2.1 Voice Codec Profile

- Section 9 of ATIS-1000009.2006:
    - The ITU-T G.711 codec shall be tested during SIP media session negotiation via SDP.
    - The ITU-T G.729 codec shall be tested during SIP media session negotiation via SDP.
    - The ITU-T T.38 fax over UDPTL over UDP shall be tested during SIP media session negotiation via SDP.

4.1.2.2 RTP Profile for SIP based VoIP Service Interconnection (RTP/ RTCP)

References:

1. IETF RFC 3550 (2003), *RTP: A Transport Protocol for Real-Time Applications*.
2. IETF RFC 3551 (2003), *RTP Profile for Audio and Video Conferences with Minimal Control*.
3. IETF RFC 3389 (2002), *RTP Payload for Comfort Noise*.

4.2 DTMF over RTP

4.2.1. Signaling

Reference:

- IETF RFC 4733, *RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals*.
    1. SDP shall support the use of dynamic payload type to specify telephone-event for events 0-11 covering 0-9, #, and *.

4.2.2. Media

Reference:

IETF RFC 4733, *RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals*.

1. The DTMF Named Events for 0-9, * and # in RFC 4733 shall be tested.
2. The dynamic payload type of RFC 4733 shall be tested.
3. The same dynamic payload type first offered shall be used in both directions.

4. The SDP answer shall indicate the list of supported named events.

4.3 Basic Fax Service with T.38 Support

4.3.1. Signaling

Same as those in Basic Voice Service with the following:

- In the SIP Offer and Answer messages, SDP shall support AVP image/t38.
- In the SIP Offer and Answer messages, SDP may support AVP audio/t38.

4.3.2 Media

4.3.2.1 ITU-T T.38

T.38 is an ITU-T standards recommendation for protocols used for the transmission of Group 3 fax over an IP network. This section defines a T.38 protocol profile to be used in conjunction with the T.38 Fax Test Suites as defined in ATIS-1000041. A summary of the key requirements and exceptions is listed below in section 4.3.3.1.1. Annex A, *T.38 Protocol Normative Reference*, specifies variances to the T.38 recommendations and is assumed as a base. Fax Service references are included in section 4.3.3.1.2.

4.3.3.1.1 T.38 NNI Protocol Profile

To support the exchange of fax documents over a network to network interface, the following requirements are defined for this document.
T.38 Protocol Service Profile at the network to network interface (NNI):

1. SIP v.2 (RFC 3265) shall be supported to set up or tear down T.38 Fax connections.
2. Transport of T.38 fax data via UDP using the UDP transport layer (UDPTL) shall be supported.
3. Transport of T.38 fax data via RTP over UDP or TCP may be supported.
4. Transport of T.38 fax data over TCP may be supported.
5. The media gateway Autonomous Transition Method described in T.38 Annex E may be supported.
6. The negotiation for T.38 Error Correction Mode (ECM) for UDPTL connections shall be supported so that an Invite (or re-Invite) with ECM parameters will not be rejected by default.
7. The use of the parity based forward error correction described in T.38 Annex C for UDPTL connections may be supported.
8. A G.711 u-Law Fall-Back and Pass-Through negotiation shall be supported if a T.38 fax connection negotiation failed.
9. When a T.38 or G.711 Pass-Through fax connection is successfully negotiated, the existing media (voice) connection shall be dropped.
10. Upon completion of a T.38 or G.711Pass-Through fax transmission, a new re-Invite negotiation for re-Invite to restore the voice connection shall be supported.

11. If the originating endpoint rejects the re-Invite for a T.38 connection for reasons such as 415 “Unsupported Media Type” response or 606/488 “Not Acceptable Response”, the existing media (voice) connection shall be kept alive.
12. The nominal maximum bit rate for fax transmission at 14.4 Kbits/s shall be supported. Operation at a higher bit rate is not precluded subjecting to the capabilities of the endpoints.

4.3.3.1.2 Basic Fax Service References

ITU-T:
ITU-T Recommendation T.38 (2007), *Procedures for real-time Group 3 facsimile communication over IP networks*.
ITU-T Recommendation T.30 (1996), *Procedures for document facsimile transmission in the general switched telephone network*.

IETF:
IETF RFC 2833 (2000), *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*.
IETF RFC 3261 (2002), *SIP: Session Initiation Protocol*.
IETF RFC 3362 (2002), *Real-time Facsimile (T.38) - image/t38 MIME Sub-type Registration*.
IETF RFC 3264 (2002), *An Offer/Answer Model with the Session Description Protocol (SDP)*.
IETF RFC 4566 (2006), *SDP: Session Description Protocol*.
IETF RFC 4612 (2006), *Real-Time Facsimile (T.38) - audio/t38 MIME Sub-type Registration*.

1. This RFC is out of scope for Rel. 1.0.

5 Transport, Network, Tunnel/Link and Transport Layers

This clause provides a generic list of transport/network protocols to support the VoIP service in Release 1.0.

5.1 Transport

5.1.1 UDP

IETF RFC 768, *User Datagram Protocol*.

5.1.2. TCP

IETF RFC 769, *Transmission Control Protocol*.

5.1.3 TLS

IETF RFC 5246, *The Transport Layer Security (TLS) Protocol*, Version 1.2.

5. 2 IP Network

5.2.1 IP Routing and Control

Static Route is assumed for Release 1.0.

5.2.2 IP Forwarding

IETF RFC 791, *Internet Protocol*.

ATIS-1000007.2006, *Generic Signaling and Control Plane Security Requirements for Evolving Networks* [defines the options for IPSEC].

5.3 Tunneling or Link Layer

5.3.1 Ethernet

IEEE 802.3, *LAN/MAN CSMA/CD (Ethernet) Access Method*.

5.4 Physical Layer

5.4.1 Ethernet

IEEE 802.3, *LAN/MAN CSMA/CD (Ethernet) Access Method*.

Annex A

A T.38 Protocol Normative Reference

The following normative reference to ITU-T Recommendation T.38 (2007), *Procedures for real-time Group 3 facsimile communication over IP networks*, specifies the sections that are informational, will be supported, or are in scope for the ANS IP Network-to-Network Interface (NNI) Standard for VoIP. All sections that are not explicitly mentioned are deemed to be out of scope.

Sections 1 thru 5 are informational. Section 6.1, Internet Protocol – TCP or UDP, only UDP will be supported. The sections that follow are deemed to be in scope:

6.2 Gateway facsimile data transfer functions
7 IFT protocol definition and procedures
9.1 IFT over UDP transport using UDPTL protocol: IFT/UDPTL/UDP
10.1 V.8 negotiation
10.3 Facsimile mode.
10.4 Compatibility with equipment conforming to prior versions of this Recommendation
Annex A – ASN.1 notation
Annex D – SIP/SDP call establishment procedures
Appendix V.3.3 (V.3.3 Incorrect use of the colon (":") in several T.38 attributes in Annex D)
Appendix V.3.4 (Case sensitivity of udptl and T38MaxBitRate in SIP and H.248.1)