ATIS-1000041

TEST SUITES FOR IP NETWORK TO NETWORK INTERCONNECTION RELEASE 1.0

TECHNICAL REPORT

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ATIS-1000041

ATIS Standard on

TEST SUITES FOR IP NETWORK TO NETWORK INTERCONNECTION RELEASE 1.0

Alliance for Telecommunications Industry Solutions

Approved August 2010

Abstract

This document specifies a set of call test scenarios involving SIP and other signaling messages which for various situations may be required to provide an expected reaction to an event or a sequence of events appropriate to the previously signaled message. This "expected reaction" is based upon the protocol profile established in the messages that flow across the NNI.

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:

M. Dolly, PTSC Chair (AT&T)
V. Shaikh, PTSC Vice-Chair (Telcordia)
G. Yum, PTSC NG-CI Chair (Verizon)
M. Hammer, PTSC NG-CI Vice-Chair (Cisco)
M. Desterdick, Technical Editor (Verizon)

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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 specifies a set of call test scenarios involving SIP and other signaling messages which for various situations may be required to provide an expected reaction to an event or a sequence of events appropriate to the previously signaled message. This "expected reaction" is based upon the protocol profile established in the messages that flow across the NNI.

The purpose of these tests are to confirm appropriate implementation of SIP messages across the NNI involved in common call scenarios and verify their conformance with the respective protocol profiles.

The purpose of these tests are to serve as the basis for developing detailed test cases and scripts to validate conformance to the NNI profile and service interoperability including the protocols used therein.

This document may be used to develop test cases to test service interoperability between providers before service deployment.

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 VoIP 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 "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.
- The "Protocol Suites" TR (ATIS-1000040) provides the service specific protocol requirements to support service interoperability. While the other documents in this Release 1.0 set make reference to the specifications covered, the "Protocol Suites" TR identifies the specific details of protocol elements to be tested.

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 NORMATIVE 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.

ATIS-1000007.2006, *Generic Signaling and Control Plane Security Requirements for Evolving Networks*.[1]

ATIS-1000008.2006, *ANSI Extensions to Q.1980.1 - The Narrowband Signaling Syntax (NSS) – Syntax Definitions*.[1]

ATIS-1000009.2006, *IP Network -To-Network Interface Standard for VoIP*. [1]

ATIS-1000010.2006, *Support of Emergency Telecommunications Service (ETS) in IP Networks*.[1]

ATIS-1000011, *ETS Packet Priority for IP NNI Interfaces - Use of Existing DiffServ Per Hop Behaviors*.[1]

ATIS-1000013.2007, *Lawfully Authorized Electronic Surveillance (LAES) for Internet Access and Services*. [1]

ATIS-1000013.a.2009, *Supplement to ATIS-1000013.2007, LAES for Internet Access and Services*. [1]

ATIS-1000014, VoIP *Network-to-Network Interface Testing Framework*. [1]

[1] This document is 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 >

ATIS-1000017.2008, *Network Interface Protocol and the Session Initiation Protocol (SIP) with ANSI Extensions to the Narrowband Signaling Syntax (NSS).*[1]

ATIS-1000020, *ETS Packet Priority for IP NNI Interfaces - Requirements for a Separate Expedited Forwarding Mechanism.*[1]

ATIS-1000026.2008, *Session/ Border Control Functions and Requirements.*[1]

ATIS-1000028.2008, *IP Device (SIP for UA) to Network Interface Standard.*[1]

ATIS-1000029.2008, *NGN Security Requirements.*[1]

ATIS-1000030.2008, *NGN Authenticating Requirements.*[1]

ATIS-1000035.2009, *NGN Identity Management (IdM) Framework.*[1]

ATIS-1000036, *NGN Operator Services Architectures and Interfaces.*[1]

ATIS-1000104.1991 (R2008), *Exchange-Interexchange Carrier Interfaces - Individual Channel Signaling Protocols.*[1]

ATIS-1000018, *NGN Architecture.*[1]

ATIS-1000019.2007, *Network to Network (NNI) Standard for Signaling & Control Security for Evolving VoIP/Multimedia Networks.*[1]

ATIS-1000607.a.2006, *Supplement to T1.607-2000 (R2004).*[1]

ATIS-1000619.a.1994 (R2007), *Integrated Services Digital Network (ISDN) – Multi-Level Precedence and Preemption (MLPP) Service Capability (MLPP service domain and cause value changes).*[1]

ATIS-1000625.a.1998 (R2008), *Supplement to ATIS-1000625.1993 (R2008).*[1]

ATIS-1000627.1993 (R2009), *Broadband ISDN - ATM Layer Functionality and Specification.*[1]

ATIS-1000678.2006, *Lawfully Authorized Electronic Surveillance (LAES) for Voice over Packet Technologies in Wireline Telecommunications Networks, Version 2.*[1]

ATIS-1000678.a.2007, *Supplement to ATIS-1000678.2006.*[1]

ATIS-1000679.2004 (R2010), *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control or ISDN User Part.*[1]

T1.TR.76-2001, *Operation of the BICC Protocol with IP Bearer control Protocol (WD).*[1]

T1.TR.82-2003, *SIP Network Operators Implementers Guide for Pre-Paid Calling Cards, with DTMF Detection at the PSTN-IP Gateway.*[1]

3 ACRONYMS & ABBREVIATIONS

ANSI	American National Standards Institute
ATIS	Alliance for Telecommunications Industry Solutions
DTMF	Dual-Tone Multi-Frequency
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
NG-CI TF	Next Generation Carrier Interconnection Task Force
NNI	Network to Network Interface
PTSC	Packet Technologies and Systems Committee
QoS	Quality of Service
RPH	Resource Priority Header
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 TEST SUITES

4.1 Network Layer Tests

4.1.1 PING

Description: PING (Packet Internet Groper) is a TCP/IP utility that tests for network connectivity, which in this case will test whether each service provider's SIP signaling ports are reachable by using their IP addresses. The Internet Control Message Protocol (ICMP) Echo Request and Echo Reply message is used to confirm there are no problems with the Network Access layer, cabling, or routers, but not Application/Session and Transport Protocol (UDP or TCP) problems.

Purpose: To ensure IP-layer connectivity between operators' border elements corresponding to their signaling ports. (PING to Media ports out of scope for this test.)

> NOTE**:** With respect to future NG-CI phases, and in particular with respect to network deployment, care should be taken if PING is used outside of a managed network. Use of PING in a broader context opens up the potential for DoS attacks and is dependent on intermediate routers/switches being configured to forward the ICMP echo reply back. In general, this cannot be assumed to be the case. In the longer term, L5/L7 mechanisms for verifying connectivity will be required.

Conditions to be checked:

1. Verify reachability using the SIP Signaling Port IP address with the ICMP Echo utility.
2. Ensure replies received from all ICMP Echo requests.

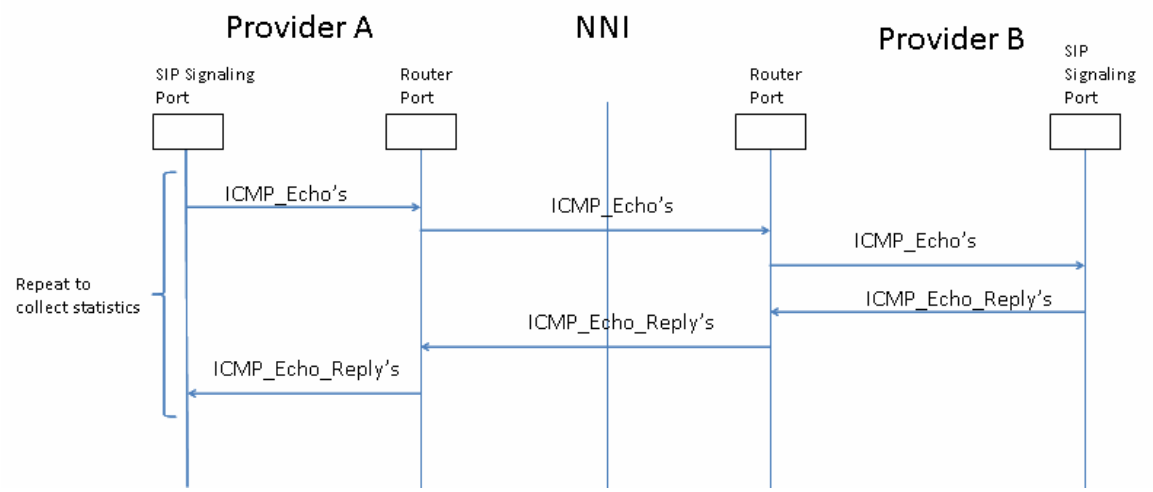

Figure 1: Ping Test

Service Parameters: The IP address of the SIP Signaling Port that provides connectivity to the signaling endpoints in each service provider's network.

References:

1. IETF RFC 792, *Internet Control Message Protocol*, 1981-09.

4.2 Application Layer Tests

4.2.1 Basic VoIP service

The test suites for basic voice service included in this section are Codec Type and Configuration Tests in 4.2.1.1 and Basic Call Flow tests in 4.2.1.2.

4.2.1.1 Codec Type and Configuration Tests

Description: The negotiation of the codec and packetization period (pp) will use the Session Description Protocol (SDP) offer, and shall at least include the codecs mutually selected by the service providers, listed in the offer in the order of preference. The opportunity to avoid transcoding events and codec mismatch in the speech path should be reflected in the codec policy.

Purpose: Check that codecs and packetization period (pp) offered and their order in the SDP is consistent with agreed upon codec list and order, that the codec invoked was correct, and that the codec selected is actually used and properly performs during the call. It is equally important to ensure that if the two service providers for any reason cannot select any common codec for a call, the call will be dropped and the event recorded.

Conditions to be checked:

1. The corresponding order of codec preference is reflected in the offer message when originating a call.
2. The highest preferred codec in the offer that is supported is the codec identified in the answer message.
3. An answer to an offer shows only one codec.

4. Mid-session codec changes are made only with a re-INVITE.

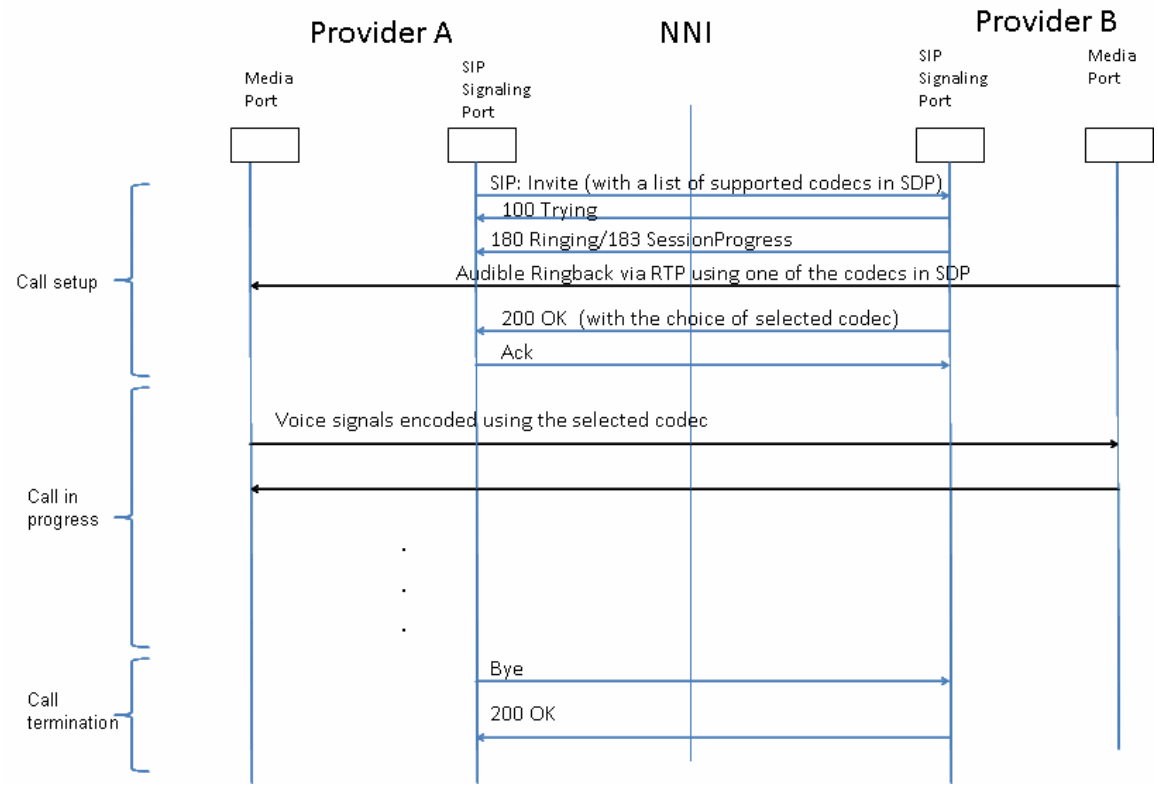

Figure 2: Codec Negotiation Call Flow

Service Parameters: The Codec type, packetization period, and order of preference.

Sample Technical Parameters:

Codecs Accepted and Packetization Period 1. Codec 1 2. Codec 2 3. Codec 3	 G.711 mu 20 ms G.729A 20 ms G.729 20 ms	 G.711 mu 20 ms G.729 20 ms G.729A 20 ms
Codec Order Preference 1. First Preference 2. Second Preference 3. Third Preference	 G.711 mu 20 ms G.729A 20 ms G.729 20 ms	 G.711 mu 20 ms G.729 20 ms G.729A 20 ms

References:

1. IETF RFC 3264, *An Offer/Answer Model with the Session Description Protocol (SDP)*, 2002-06.
2. IETF RFC 4566, *SDP: Session Description Protocol*, 2006-07.

NOTE: RFC 4566 obsoletes RFC 2327 & RFC 3266.

4.2.1.2 Basic Call Flow Tests

For each of these application layer tests, it is assumed that the Network Layer Tests and Codec Type and Configuration Tests have previously been successfully performed and completed.

The basic call flow tests included in this subsection are as follows:

- Provider A/B originating normal call release - Calling party clears after answer.
- Provider A/B originating normal call release - Calling party abandon.
- Provider A/B originating normal call release - Called party clears after answer.
- Provider A/B originating normal call setup to ring No Answer/Timeout.
- Provider A/B originating normal call setup to Busy Line/Calling party release.
- Provider A/B originating verify proper handling for no route to destination.

4.2.1.2.1 Provider A/B originating normal call release - Calling party clears after answer

Description: After the called party answers the call, the calling party says "Hello, testing. I'm going to hang up and you should hear dial tone." The called party replies, "Okay, thank you. Good bye." The calling party then hangs up.

Purpose: To verify that the calling party's equipment clears the call and that the called party hears dial tone after the calling party hangs up.

Conditions to be checked:

1. Provider B uses one of the codecs on the list to encode the voice signals.
2. Provider A uses the same codec Provider B uses to encode the voice signals. Provider A and Provider B continue to use the same codec throughout the call.
3. Provider B stops sending media traffic after it sends Bye message.
4. Provider A stops sending media traffic after it sends Bye-200 OK message.
5. Provider A and Provider B will respond to any future requests for the call 481 messages.

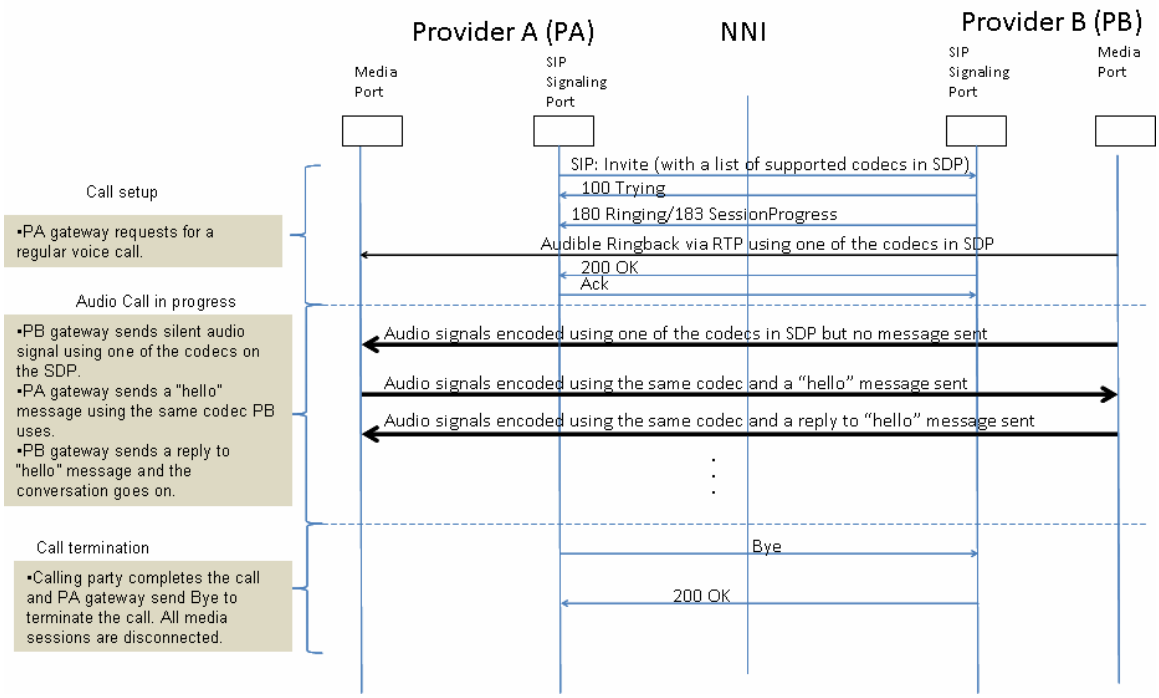

Figure 3: Normal Call Release - Calling party clears after answer

Service Parameters: Same as for Codec Type and Configuration Tests.

References:

1. IETF RFC 3555, *MIME Type Registration of RTP Payload Formats*, July 2003.
2. IETF RFC 4566, *SDP: Session Description Protocol*, July 2006.

4.2.1.2.2 Provider A/B originating normal call release - Calling party abandon

Description: The calling party initiates a call, waits to hear audible ring back tone, and then hangs up before the call is answered. The called party takes no action.

Purpose: To verify that Provider B acknowledges the Cancel request, responds with a 487 Request Terminated message and stops the ringing at the called party.

Conditions to be checked:

1. Provider B receives a cancel request prior to the timeout period.
2. Any media traffic between Provider A and Provider B is stopped.

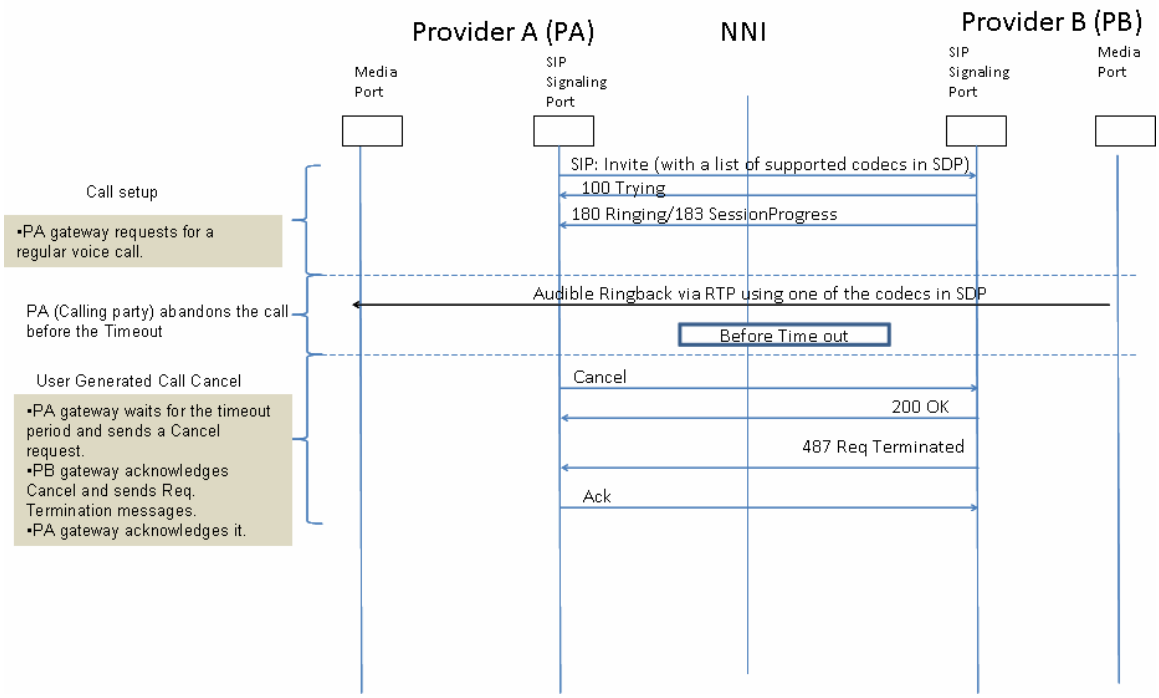

Figure 4: Calling party abandons before the call is answered

Service Parameters: Same as for Codec Type and Configuration Tests.

References:

1. IETF RFC 3555, *MIME Type Registration of RTP Payload Formats*, July 2003.
2. IETF RFC 4566, *SDP: Session Description Protocol*, July 2006.

4.2.1.2.3 Provider A/B originating normal call release - Called party clears after answer

Description: After the called party answers the call, the called party says "Hello, testing. I'm going to hang up and you should hear dial tone." The calling party replies, "Okay, thank you. Good bye." The called party then hangs up.

Purpose: To verify that the called party clears the call and that the calling party hears dial tone after called party hangs up.

CONDITIONS TO BE CHECKED:

1. Provider B uses one of the codecs on the list to encode the voice signals.
2. Provider A uses the same codes Provider B uses to encode the voice signals.
3. Provider A and Provider B continue to use the same codec throughout the call.
4. Provider A stops sending media traffic after it sends Bye message.
5. Provider B stops sending media traffic after it sends Bye-200 OK message.
6. Provider A and Provider B will respond to any future requests for the call 481 messages.

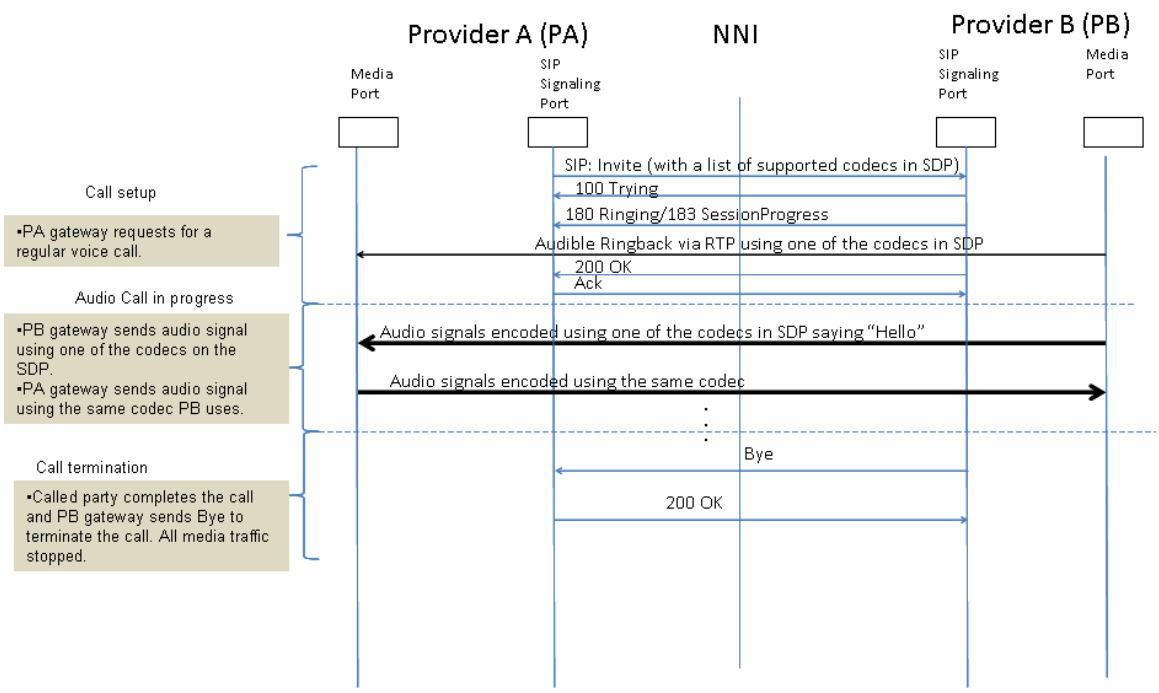

Figure 5: Called party hangs up after the call is answered

Service Parameters: Same as for Codec Type and Configuration Tests.

References:

1. IETF RFC 3555, *MIME Type Registration of RTP Payload Formats*, July 2003.
2. IETF RFC 4566, *SDP: Session Description Protocol*, July 2006.

4.2.1.2.4 Provider A/B originating normal call setup to ring No Answer / Timeout

Description: The calling party initiates the call and listens to audible ringback tone for a period longer than the agreed upon timeout period. The called party takes no action and the call times out to a No Answer condition.

Purpose: To verify that the calling party hears ringing tone and then dial tone after timeout per timer setting.

Conditions to be checked:

1. Provider B receives a Cancel message from Provider A between the period of the Timeout and Timeout + inter-carrier timeout tolerance.
2. Provider A receives a 487 Request message for the call.
3. CA and CB media traffic, if any, from each direction is stopped.

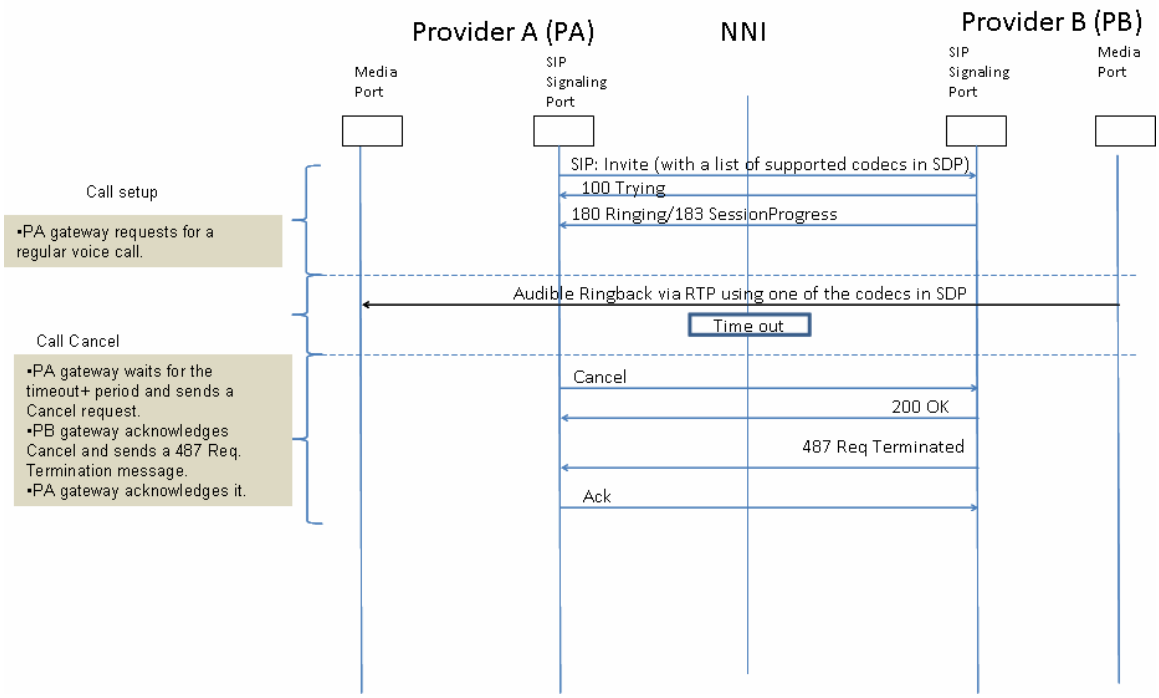

Figure 6: No Answer Timeout

Service Parameters: Same as for Codec Type and Configuration Tests, Ring/No Answer Service Timeout period.

References:

1. IETF RFC 3555, *MIME Type Registration of RTP Payload Formats,* July 2003.
2. IETF RFC 4566, *SDP: Session Description Protocol*, July 2006.

4.2.1.2.5 Provider A/B originating normal call setup to Busy Line / Calling party release

Description: The called party goes off hook and creates a busy condition. The calling party initiates a call to the now busy called party and hears a busy signal.

Purpose: To verify proper call handling in that the calling party hears a busy signal.

Conditions to be checked:

1. Provider A receives a 486 Busy Here message.
2. Provider A and Provider B do not send any media traffic to each other.

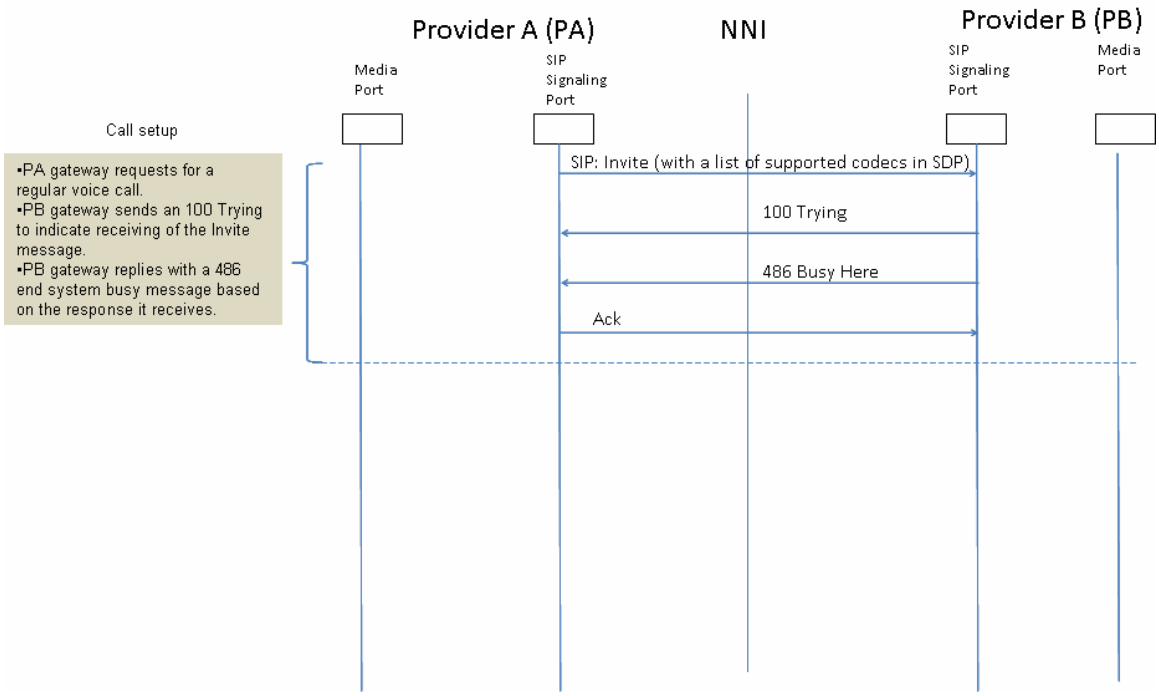

Figure 7: Call to Busy Line

Service Parameters: Same as for Codec Type and Configuration Tests.

References:

1. IETF RFC 3555, *MIME Type Registration of RTP Payload Formats*, July 2003.
2. IETF RFC 4566, *SDP: Session Description Protocol*, July 2006.

4.2.1.2.6 Provider A/B originating verify proper handling for no route to destination

Description: The calling party initiates a call to known code in Provider B's network with no route.

Purpose: Verify proper call handling for a destination that is not reachable in the foreign network.

Conditions to be checked:

1. Provider B receives an Invite with a called number in Provider B's routing table.
2. Provider A receives a 404 Not Found message.
3. Provider A and Provider B do not send any media traffic to each other.

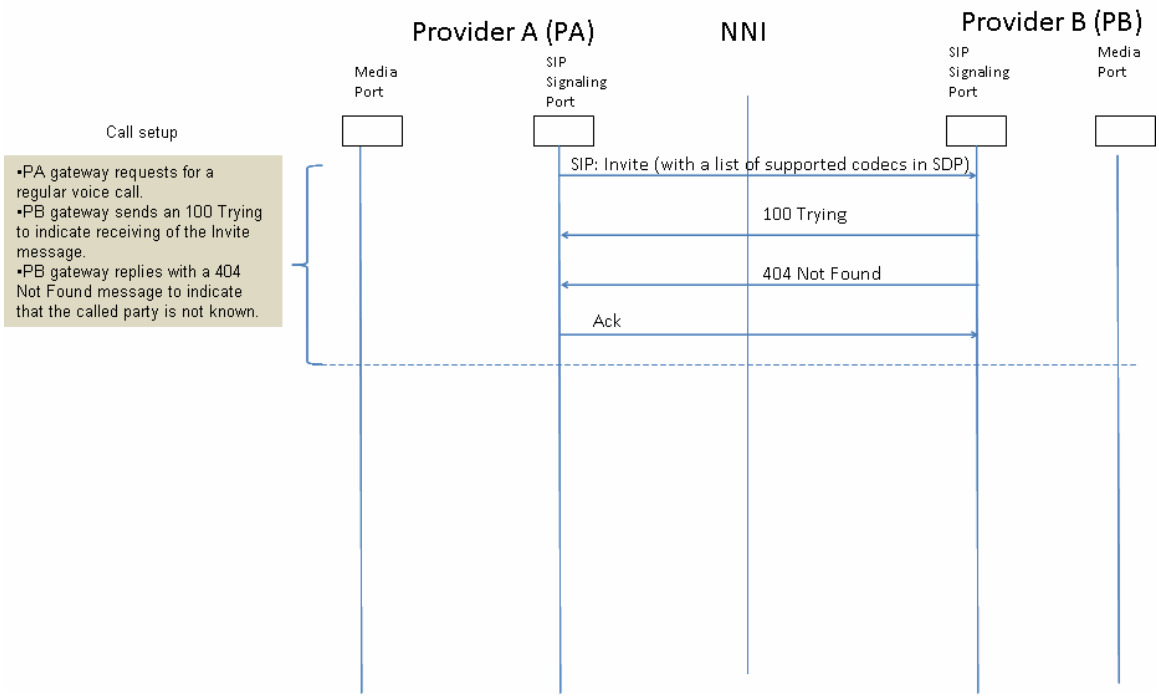

Figure 8: No Route to Destination

Service Parameters: Same as Codec Test, list of test numbers.

References:

1. IETF RFC 3555, *MIME Type Registration of RTP Payload Formats*, July 2003.
2. IETF RFC 4566, *SDP: Session Description Protocol*, July 2006.

4.2.3 DTMF Over RTP

For each of these application layer tests, it is assumed that the Network Layer Tests and Codec Type and Configuration Tests have previously been successfully performed and completed.

The following test suites for basic voice service with DTMF support are included in this section:

- Verify digits (0-9 and * and #) received post-answer from the caller.
- Verify digits (0-9 and * and #) received post-answer from the callee.
- Verify digits (0-9 and * and #) to a network based call prompter post-answer from the callee.
- Verify digits (0-9 and * and #) to a network based call prompter post-answer from the caller.

The description, purpose, service parameters, and references are the same for each of these tests.

Description: Tone representations may be used by Application Servers, Telephone Switches, or Gateways. Gateways can detect DTMF, have digital signal processers (DSP) used to process audio packets, and transmit the frequency components of the voice-band signal via RTP. Dual-Tone Multi Frequency (DTMF) tones in the RTP streams are susceptible to problems where compression is used in low bit rate codecs optimized for voice patterns, which can distort DTMF tones -- e.g., G.723.1. "Separate RTP payload formats for telephony tone signals are desirable since low-rate voice codecs cannot be guaranteed to reproduce these tone signal accurately enough for automatic recognition

[RFC4733].” It is possible to change to a higher bandwidth codec, or use the telephone-event payload per IETF RFC 4733, where DTMF events are identified and negotiated using DSP where the receiver produces the tone signal.

Purpose: To verify that DTMF tone generated by the end user device on each side of the NNI is transported over RTP streams as DTMF Names Events.

Service Parameters: Support for DTMF over RTP.

References:

1. IETF RFC 4733, *RTP Payload for DTMF Digits, Telephony Tones & Telephony Signals*, 12/06.

NOTE – RFC 4733 obsoletes RFC 2833.

4.2.3.1 Verify digits (0-9 and * and #) received post-answer from the caller

Description: Regular call session in progress; User enters DTMF; the VoIP phone or media gateway generates RTP packets per 4733.

Conditions to be checked:

1. Verify Provider B receives all the DTMF digits 0-9, *, and # with accurate representation of frequency, duration, and volume.
2. Verify that DTMF signals are transported via RTP across the NNI.

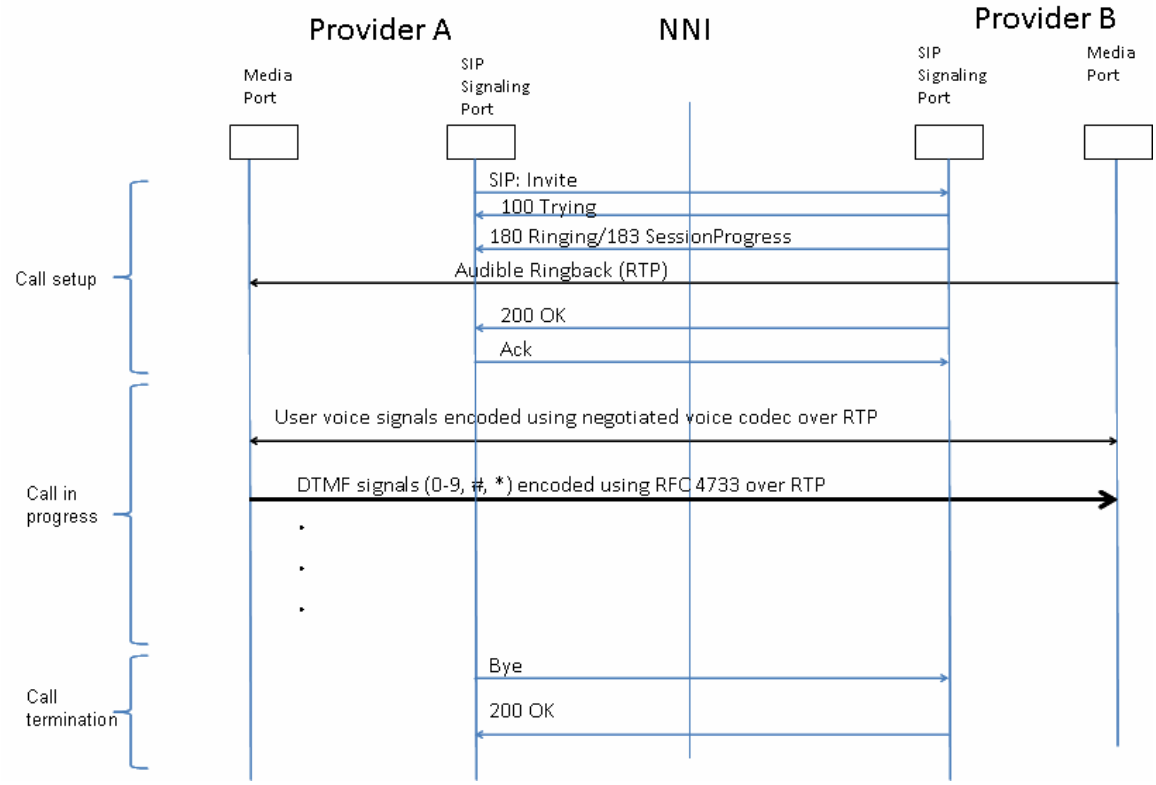

Figure 9: DTMF Test 1 (received post answer, from the caller)

4.2.3.2 Verify digits (0-9 and * and #) received post-answer from the callee

Description: Regular call session in progress; User enters DTMF; the VoIP phone or media gateway generates RTP packets per 4733.

Conditions to be checked:

1. Verify Provider A receives all the DTMF digits 0-9, *, and # with accurate representation of frequency, duration, and volume.
2. Verify that DTMF signals are transported via RTP across the NNI.

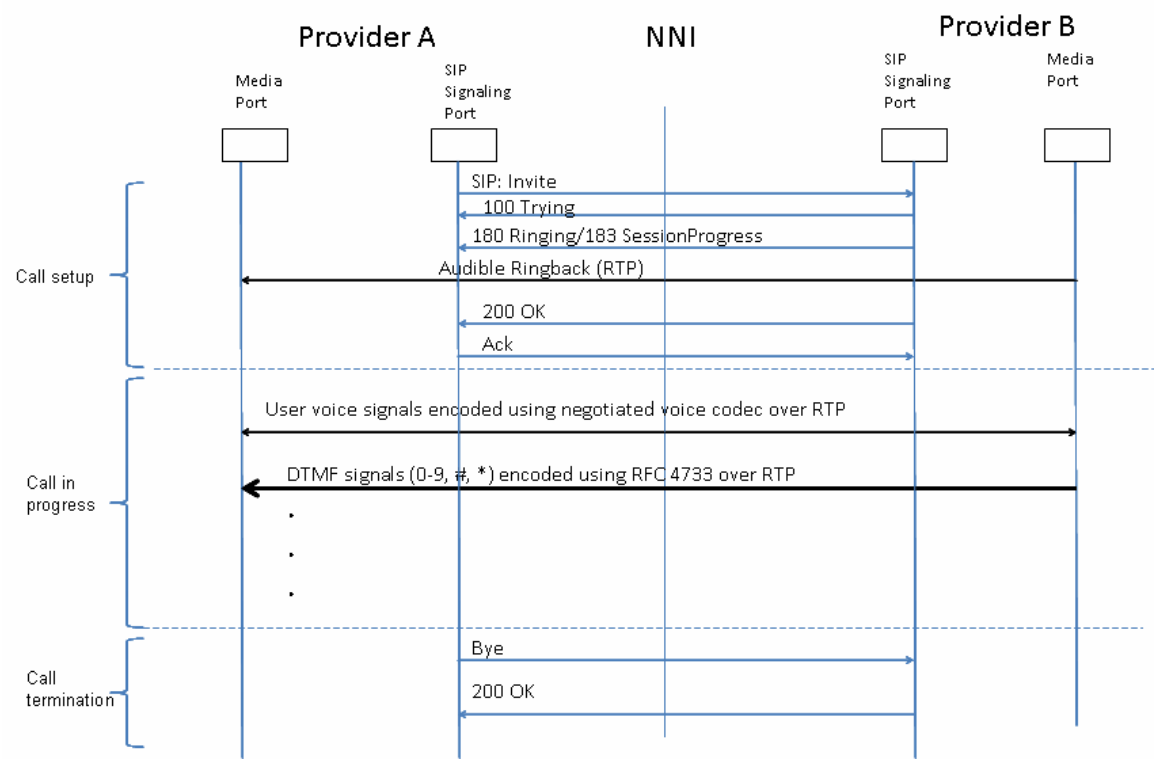

Figure 10: DTMF Test 2 (received post-answer, from the callee)

4.2.3.3 Verify digits (0-9 and * and #) to a network based call prompter post-answer from the callee

Description: Regular call session in progress; Prompter requests a DTMF indication response; User enters DTMF; the VoIP phone or media gateway generates RTP packets per 4733.

Conditions to be checked:

1. Verify Provider A receives all the DTMF digits 0-9, *, and # with accurate representation of frequency, duration, and volume.
2. Verify that DTMF signals are transported via RTP across the NNI.

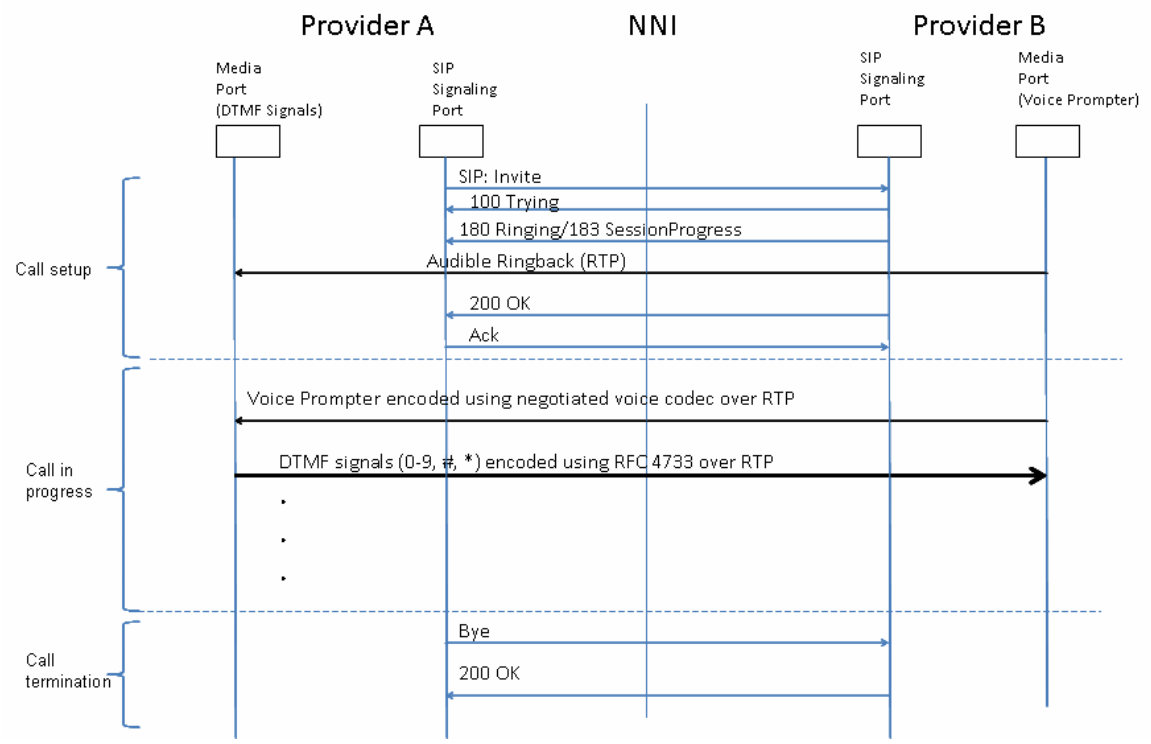

Figure 11: DTMF Test 3 (network based call prompter, post answer from the callee)

4.2.3.4 Verify digits (0-9 and * and #) to a network based call prompter post-answer from the caller

Description: Regular call session in progress; Prompter requests a DTMF indication response; User enters DTMF; the VoIP phone or media gateway generates RTP packets per RFC 4733.

Conditions to be checked:

1. Verify Provider B receives all the DTMF digits 0-9, *, and # with accurate representation of frequency, duration, and volume.
2. Verify that DTMF signals are transported via RTP across the NNI.

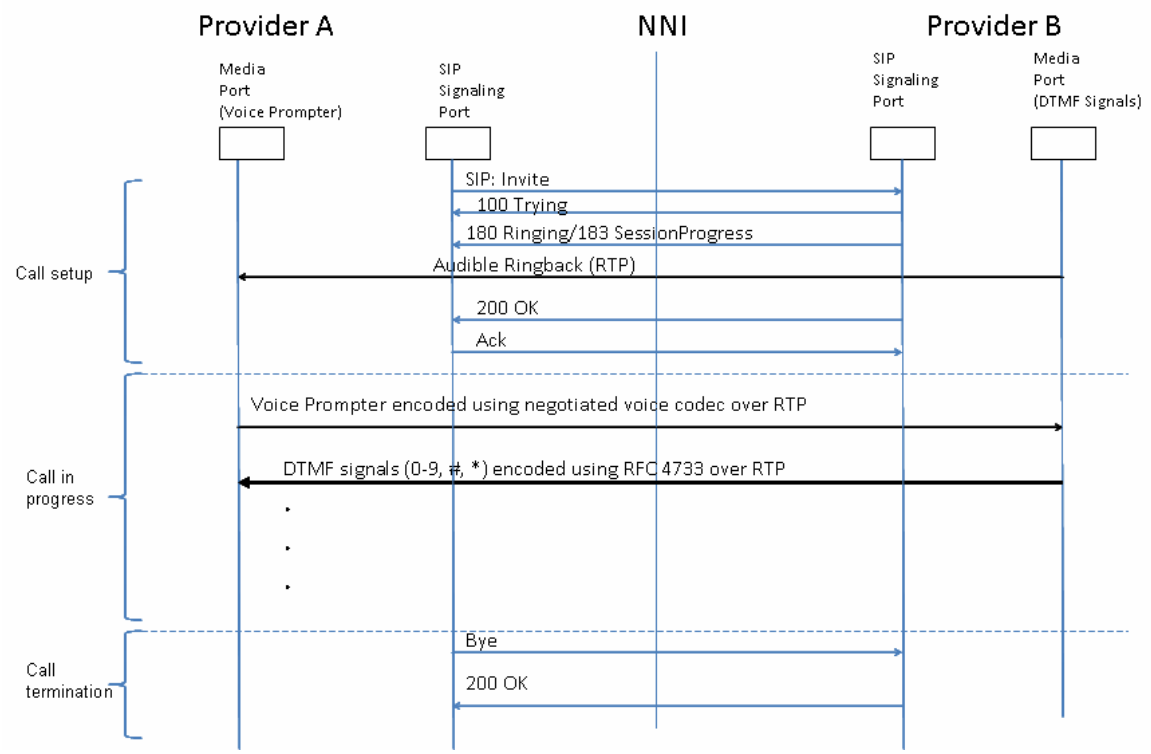

Figure 12: DTMF Test 4 (network based call prompter, post answer from the caller)

4.2.4 Fax service with T.38 support

For each of these application layer tests, it is assumed that the Network Layer Tests and Codec Type and Configuration Tests have previously been successfully performed and completed.

The following test suites for Fax service with T.38 support are included in this section:

- G.711 Pass-Thru Mode without Re-Negotiation
- G.711 Fall Back (originating media gateway does not support T.38)
- G.711 Fall Back (from non-G.711 session)
- T.38 Fax Mode

The description, purpose, service parameters, and references are the same for each of these tests.

Description: Fax calls can be supported at the NNI via circuit emulation using G.711 pass-thru or T.38 packet transmission mode. In G.711 pass-thru mode, during a fax call, an audio channel is established between the originating and terminating T.30 fax devices. The fax-tones originated from a T.30 fax device are encoded using in G.711 format and the G.711 traffic is sent across the NNI to the receiving T.30 fax device terminal. In T.38 mode, a T.38 capable fax device terminal is used to encode the fax tones into T.38 format and send the T.38 traffic across the NNI to the receiving fax device terminal, which in turn decodes the T.38 traffic format to fax tones.

Signaling and fax tone exchange faults may occur during set-up (fax discrimination, transition, capabilities negotiation). Faults during the call-phase (transmission) may affect image transmission quality due to packet loss, delay, and jitter. Gateways and customer equipment during set-up impact fax set-up success. Gateway-to-gateway tests under varying configurations for signaling and media is

warranted. However, it is not feasible to test CPE terminal and gateway incompatibility and deficiencies at the NNI.

Purpose: To verify the use of T.38 protocol to transport fax and the use of G.711 flow through as a fallback method.

Configuration Note for Fax Endpoint:

1. The use of Echo Cancellers by the gateways when switching from an audio to a fax session can impact the set-up phase and fax image transmission and it should be turned off.
2. The use of Voice Activity Detection (VAD) features needs to be deactivated by the gateways during the fax session.

Notes for T.38 Fax Mode:

The use of TCP and UDP to transport T.38 packets between T.38 capable gateways, servers, and devices are specified in the ITU-T T.38 recommendation. SIP is used to set up and negotiate the use of T.38 for fax transmission across the NNI. A SIP re-INVITE will close the audio channel and the gateway will switch to fax-to-relay (T.38) upon detection of a fax event, or for unsuccessful fax relay negotiation, fallback to pass-through with up-speed to G.711. An integrated access device (IAD) fax could setup a call offering only a T.38 fax relay.

1. To minimize signaling failures, the correct SIP ReINVITE message offering the generic SDP offer for G.711 codec is needed to switch from audio to G.711 up-speed for fax tone transmission.
2. Voice Activity Detection (VAD) needs to be deactivated by the MGC for successful set-up.
3. The T.38 fax "no signal detected" failure may be caused when no fax tone (V.21 preamble?) is detected (which are optional for T.38, but required for T.30 fax) but not always supported by all fax machines. The use of called-tone prior to the T.38 ReINVITE to ensure tone detection as required by the T.30 protocol may reduce signaling failure and should be tested.

Service Parameters: N/A.

References:

1. ITU-T Recommendation T.30, *Fax PSTN*.
2. ITU-T Recommendation T.38, *Fax*
3. IETF RFC 4040, *RTP Payload Format for a 64 kbit/s Transparent Call*, 2005-04.
4. IETF RFC 3362, *Real-time Facsimile (T.38) – image/t.38 MIME Sub-type Registration*, 2002-08.

4.2.4.1 G.711 Pass-Thru Mode without Re-Negotiation

Description: Regular call session in progress using G.711 codec; Provider A or Provider B initiates fax transmission using existing G.711 circuit emulation mode to transmit fax signal.

Conditions to be checked:

1. The fax document is successfully transmitted over G.711 RTP and decoded accurately on the receiving side.
2. The same G.711 media streams connected to transmit voice signals are used to transit fax tones on both directions.

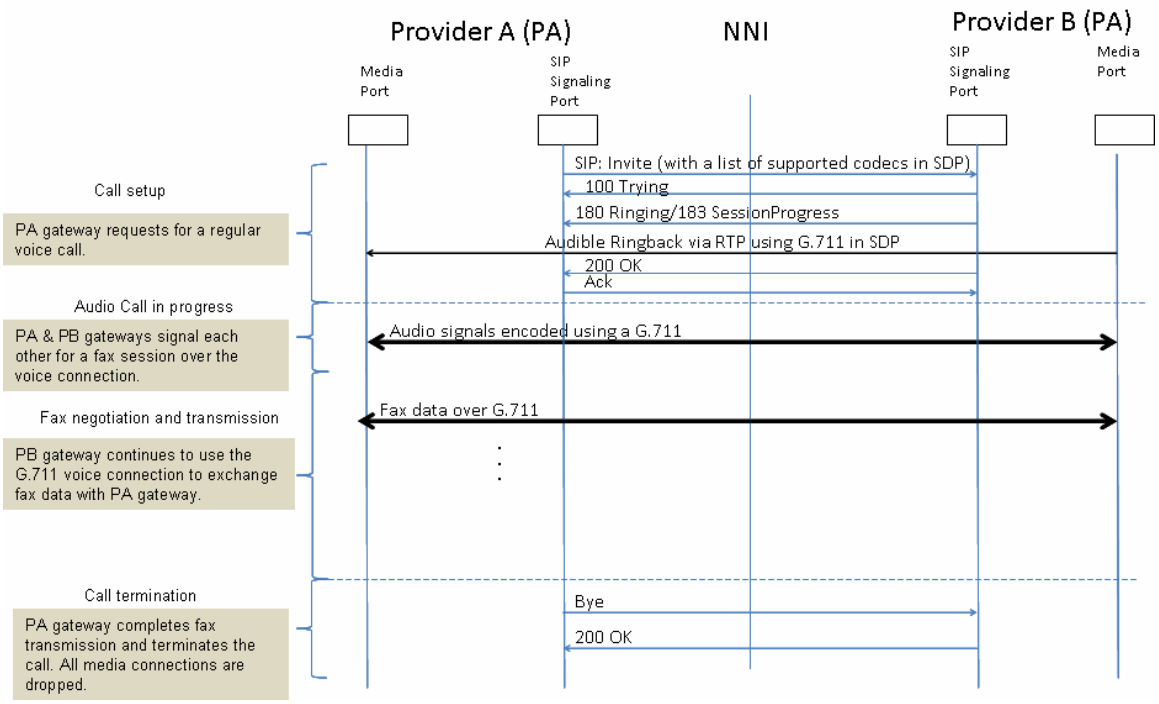

Figure 13: G.711 Pass-Thru Mode without Re-Negotiation

4.2.4.2 G.711 Fall Back (originating media gateway does not support T.38)

Description: Regular call session in progress using non-G.711 codec; Provider B initiates fax transmission offering a T.38 connection; Provider A does not support T.38; Provider B re-invites using G.711 circuit emulation mode to transmit fax signal.

Conditions to be checked:

1. Verify that a Re-INVITE with T.38 is first offered and rejected before a re-Invite with G.711 offer is sent.
2. The fax document is successfully transmitted over G.711 RTP and decoded accurately on the receiving side.

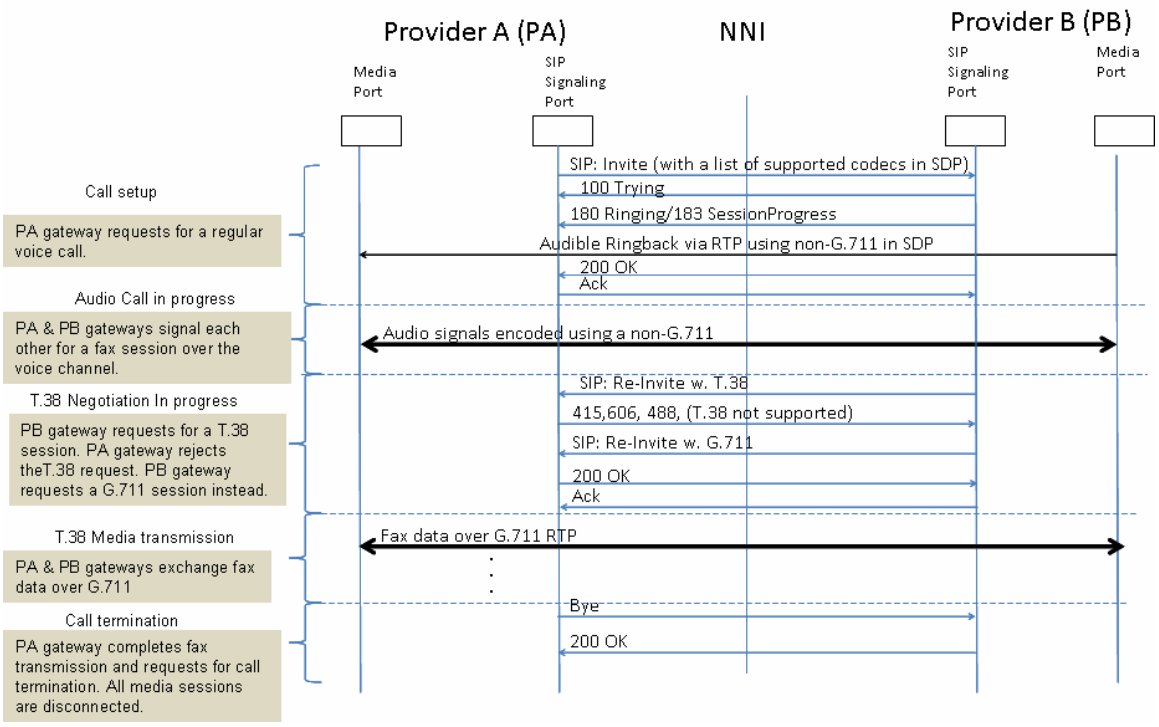

Figure 14: G.711 Fall Back (originating media gateway does not support T.38)

4.2.4.3 G.711 Fall Back (from non-G.711 session)

Description: Regular call session in progress using non-G.711 codec; Provider B does not support T.38 and initiates fax transmission offering a G.711 connection; Provider A accepts the offer and the transmission is completed using G.711.

Conditions to be checked:

1. Verify that a Re-INVITE with G.711 is offered and accepted before a G.711 RTP connections is established.
2. The fax document is successfully transmitted over G.711 RTP and decoded accurately on the receiving side.

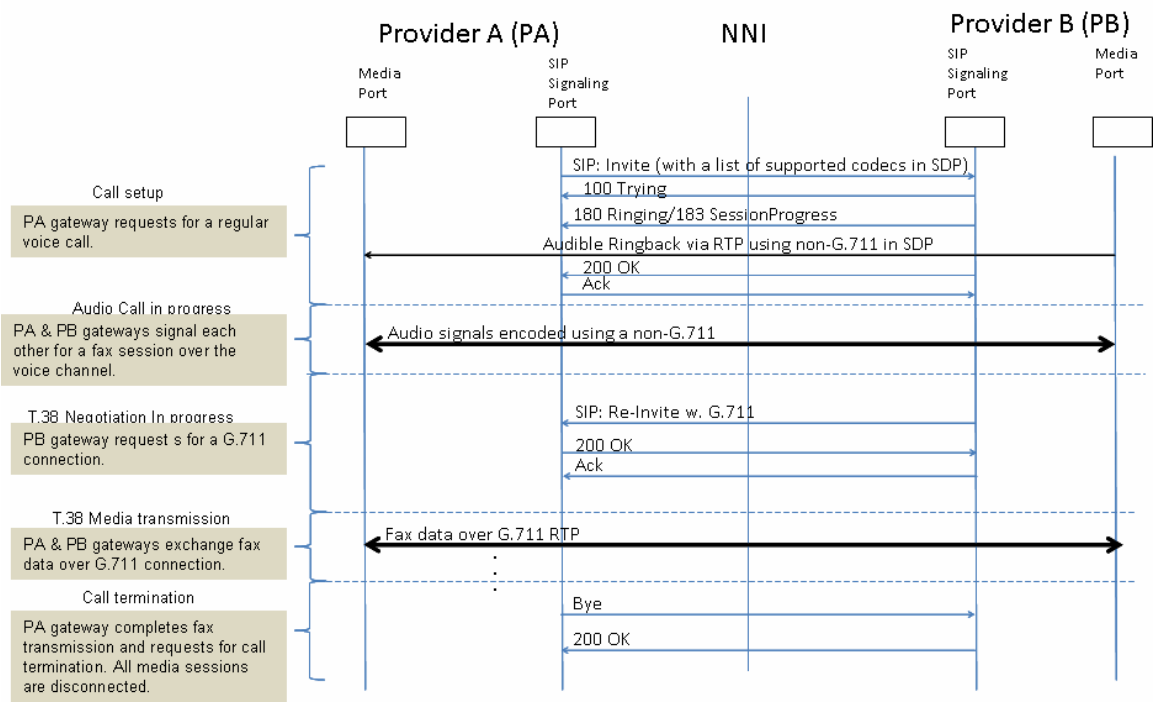

Figure 15: G.711 Fall Back (from non-G.711 session)

4.2.4.4 T.38 Fax Mode

Description: Regular call session in progress using any negotiated voice codec; Provider B initiates fax transmission offering a T.38 connection; Provider A supports T.38, a T.38 session is established and the transmission is completed using T.38.

Conditions to be checked:

1. A SIP re-Invite with a T.38 offer is sent and accepted before the T.38 connection is established.
2. Verify that while the T.38 connection is up the previous connection it disconnected.

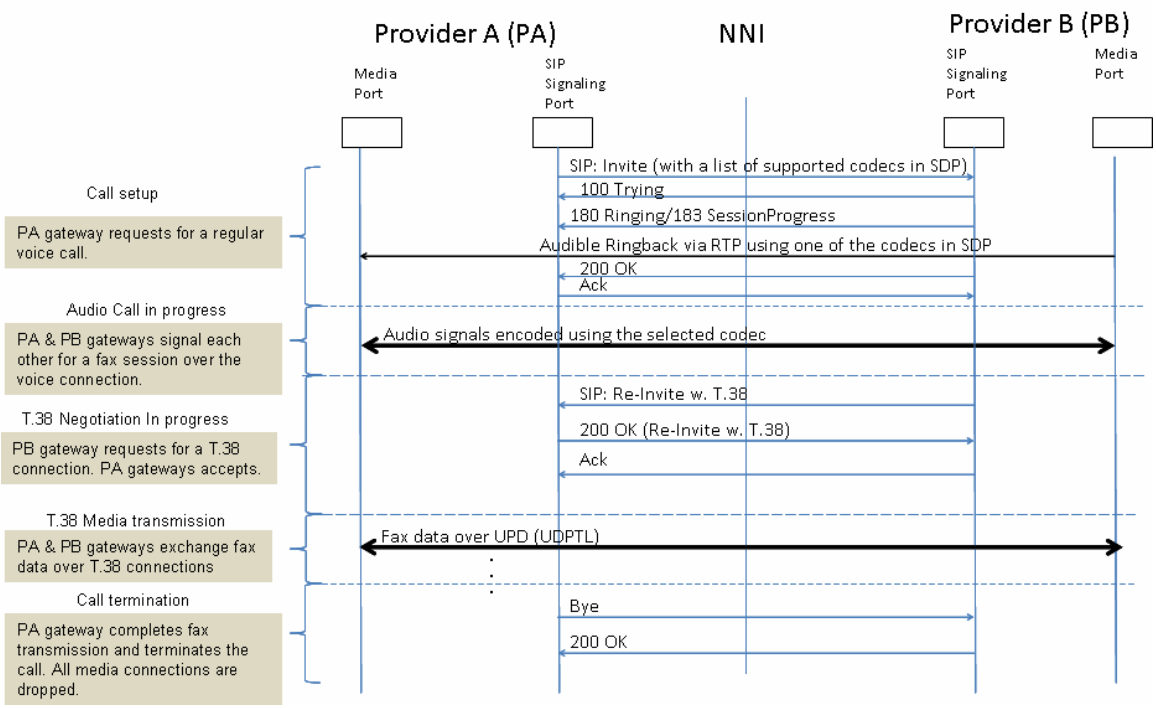

Figure 16: T.38 Fax Mode