

Day One





Workshop at NGN Lab Instrumentation; Protocols: SIP



Workshop at NGN LabProtocols: H.248



 Workshop at NGN Lab interoperability aspects. SIP-ISUP SIP –I (ITU-T Rec. Q.1912.5 Profile C)



Workshop at NGN Lab Voice Quality (PESQ ITU-T Rec. P.862)



SIP protocol test scenario

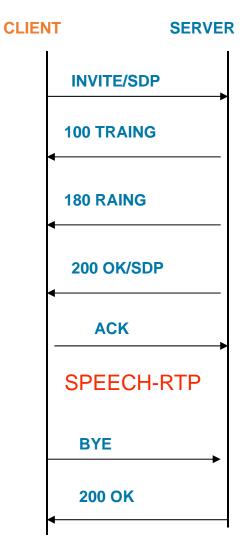


Tests will be run in two groups

- NGN network tests
- > Tests with test equipment

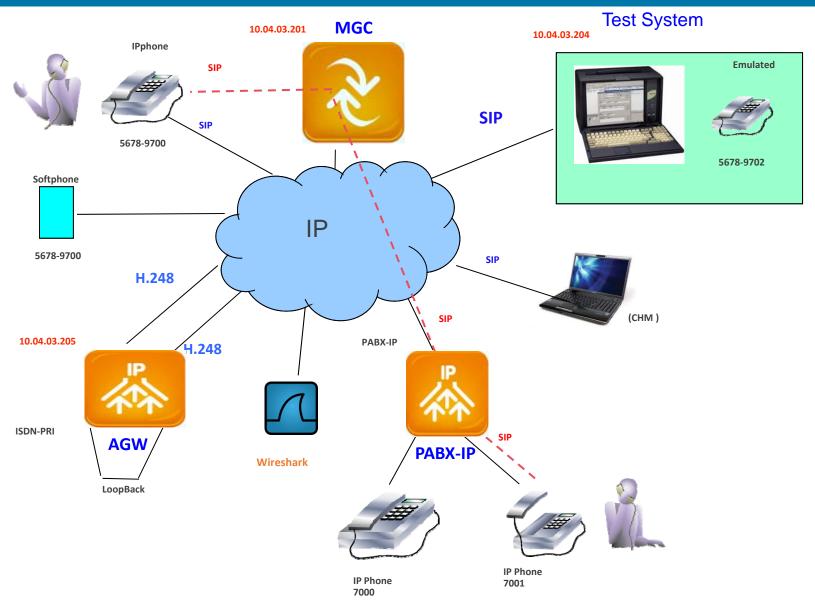
NGN network tests

- NGN network call SIP protocol Success Case
- Subscriber A originates call
- Subscriber B originates call
- NGN network call SIP protocol Failure Case
- Release: busy
- Release: congested



Topology SIP Protocol





SIP protocol call using the network Success Case – A originates



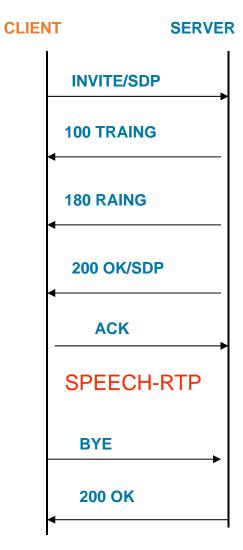
TEST-1	SUCCESS CASE Subscriber A originates
Scope	Verify how system behaves when subscriber is free
Procedure	 Subscriber A originates call (soft phone) SIP Protocol, using CODEC G.711 From soft phone to IP phone Monitored via wireshark
Objective	•

Verify how system behaves when interfunctioning

 Verify success of call completion, communication, and termination, involving NGN elements and correct codec negotiation

Expected results:

 Success completing call, talking and hanging up when subscriber A originates the call



SIP protocol call using the network Success Case – B originates

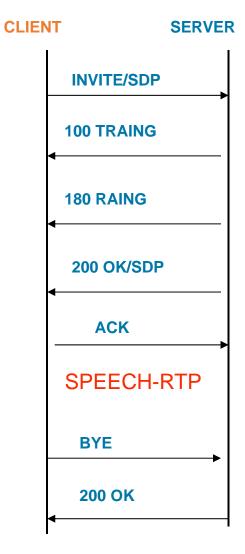


TEST-2	SUCCESS CASE Subscriber B originates
Scope	Verify how system behaves when subscriber is free
Procedure	 Subscriber B originates call SIP Protocol, using CODEC G.711 From soft phone to IP phone Monitored via wireshark
Objective	'

- Verify how system behaves when interfunctioning
- Verify success of call completion, communication, and termination, involving NGN elements and correct codec negotiation

Expected results:

Success completing call, talking and hanging up when subscriber B originates the call



SIP protocol call using the network with Failure Case – B busy



TEST-3	FAILURE CASE B busy
Scope	Verify how system behaves when subscriber number changed
Procedure	Originate Call to subscriber B, whose number must be busy SIP protocol, using CODEC G.711 From test equipment to IP phone Monitored via wireshark

Objective

- Verify how system behaves when interfunctioning
- Verify if caller receives a busy tone and message 486 Busy Here
- Verify correct mapping of referred causes (ITU reference Q.850) in the respective SIP messages

(Reason:Q.850;cause=17;text="user busy")

Expected results on the caller's end:

- Busy tone
- Message 486 Busy Here

INVITE/SDP 100 TRAING 486 Busy Here (Reason:Q.850;cause=17; text="User busy")

SIP protocol call using the network with Failure Case – B congested



TEST-4	FAILURE CASE B congested
Scope	Verify how system behaves when receiving subscriber number is non-existent
Procedure	 Originate Call to subscriber B, whose number must be congested SIP protocol, using CODEC G.711 From test equipment to IP phone Monitored via wireshark

Scope

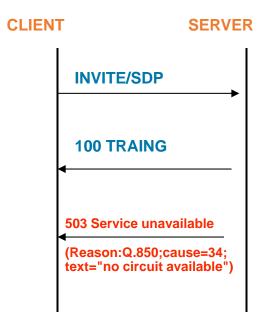
- Verify how system behaves when interfunctioning
- Verify if caller receives an empty number tone and message 503 Service Unavailable
- Verify correct mapping of referred causes (ITU reference Q.850) in the respective SIP messages

(Reason:Q.850;cause=34;text="no circuit available")

Expected results on the caller's end:

Empty number tone

(Reason:Q.850;cause=34;text="no circuit available")





Workshop at NGN Lab Instrumentation; Protocols: SIP; Tests with test equipment.

SIP protocol test scenario

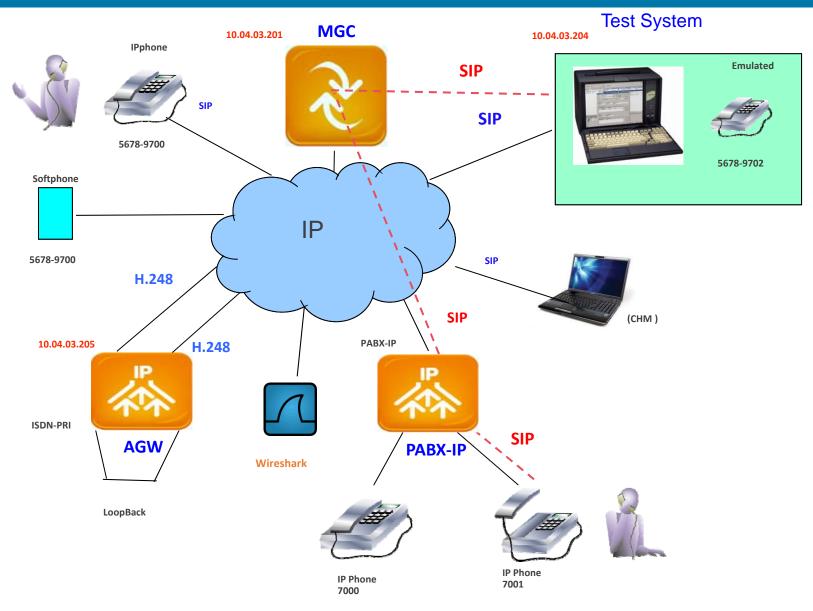


Tests with test equipment

- NGN network call SIP protocol Success Case
- Subscriber A originates call
- Subscriber B originates call
- NGN network call SIP protocol Failure Case
- Release: number changed
- □ Conformance Test ETSI TS 102 027-2 V4.1.1 (2006-07) Methods for Testing and Specification (MTS); Conformance Test Specification for SIP (IETF RFC 3261);
- SIP_CC_OE_CE_V_032
- SIP_CC_OE_CR_V_010
- SIP_CC_OE_CE_TI_003
- SIP_MG_TE_V_013

Topology SIP Protocol





SIP protocol call, Failure Case – B number changed



TEST-1	FAILURE CASE B number changed
Scope	Verify how system behaves when subscriber fails – 22 number changed
Procedure	Originate Call to subscriber B (number changed) SIP protocol, using CODEC G.711 From test equipment to IP phone Monitored via wireshark

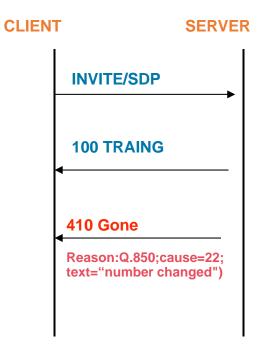
Objective

- Verify how system behaves when interfunctioning
- Verify if caller receives a busy tone and message 410 Gone
- Verify correct mapping of referred causes (ITU reference Q.850) in the respective SIP messages

(Reason:Q.850;cause=22;text="number changed")

Expected results on the caller's end:

- Busy tone
- Message 410 Gone



Conformance Test: SIP_CC_OE_CR_V_010 - CANCEL



■ Conformance Test:

TPId: SIP_CC_OE_CR_V_010

Status: Mandatory

Ref: RFC 3261

Purpose: Ensure that the IUT having received a Trying (100 Trying) response to its INVITE request, to give up the call, sends a CANCEL request.

Conformance Test: SIP_CC_OE_CE_V_032 (NotFound (404 Not Found))



Conformance Test:

TPId: SIP_CC_OE_CE_V_032

Status: Mandatory

Ref: RFC 3261

Purpose: Ensure that the IUT when an INVITE client transaction is in the Calling state, on receipt of a Not Found (404 Not Found) response sends an ACK request with the same Call-ID, From headers and Request-URI as in the original INVITE request and the same Tag in the To header as in this response.

Conformance Test: SIP_CC_OE_CE_TI_003 - Timer



Conformance Test:

TPId: SIP_CC_OE_CE_TI_003

Status: Mandatory

Ref: RFC 3261

Purpose: If an unreliable transport (UDP) is used, ensure that the IUT, when an INVITE client transaction is in the Calling state having already repeated its INVITE wait for a timer A set with a value of 2*T1 before sending it again.

Conformance Test: SIP_MG_TE_V_013 – short names



Conformance Test:

TPId: SIP_MG_TE_V_013

Status: Mandatory

Ref: RFC 3261

Purpose: Ensure that the IUT on receipt of an INVITE request including headers set with short names, sends a Success (200 OK) response preceded optionally by informational (1XX) response.

Day One





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Workshop at NGN LabProtocols: H.248



 Workshop at NGN Lab interoperability aspects. SIP-ISUP SIP –I (ITU-T Rec. Q.1912.5 Profile C)



Workshop at NGN Lab Voice Quality (PESQ ITU-T Rec. P.862)



Workshop at NGN LabProtocols: H.248.

Signaling protocol H.248 scenario

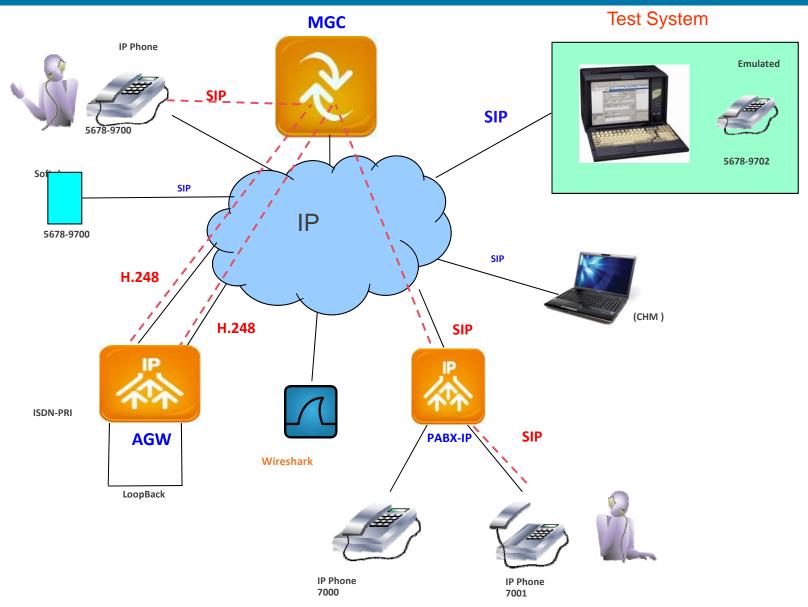


Four kinds of tests will be run:

- MGW alignment
- Service Change
- NGN network call H.248 protocol Success Case
- Subscriber A originates call
- Subscriber B originates call
- H.248 tests with RTP EVENTE
- NGN network call H.248 protocol Failure Case
- Release: busy
- Release: non-existent

Topology H.248 NGN network, VOICE call





H.248 protocol signaling call Alignment



TEST-1	ALIGNMENT	
Scope	Verify system behavior when starting up	
Procedure	 Remove board and insert it again to verify how system behaves when starting up Monitored via wireshark 	
Objective		
Verify start up behavior of a new alignment between MGW and MGC, using the Service Change command		
Expected results: • Alignment successful		

H.248 protocol signaling call Success Case – A originates



3	Subscriber A originates
Scope Ve	erify how system behaves when subscriber is free
Procedure •	Originate Call from IP phone subscriber H.248 protocol, using CODEC G.711 From IP phone to soft phone Monitored via wireshark

Objective

- · Verify how system behaves when interfunctioning
- Verify success of call completion, communication, and termination, involving NGN elements and correct codec negotiation.

Expected results:

 Success completing call, talking and hanging up when subscriber A originates the call

H.248 protocol signaling call Success Case – B originates



TEST-3	SUCCESS CASE Subscriber B originates
Scope	Verify how system behaves when subscriber is free
Procedure	 Originate Call from IP phone subscriber H.248 protocol, using CODEC G.711 From IP phone to soft phone Monitored via wireshark

Objective

- · Verify how system behaves when interfunctioning
- Verify success of call completion, communication, and termination, involving NGN elements and correct codec negotiation.

Expected results:

Success completing call, talking and hanging up when subscriber B originates the call

H.248 protocol signaling call Success Case – DTMF Digits



TEST-4	SUCCESS CASE DTMF Digits
Scope	Verify how system behaves when subscriber is free
Procedure	 Originate Call from IP phone subscriber H.248 protocol, using CODEC G.711 From IP phone to IP phone Digits from 0 to 9 shall be input Monitored via wireshark
Objective	

- Verify how system behaves when interfunctioning
- Verify success of call completion, communication, and termination, involving NGN elements and correct codec negotiation
- Verify digits in RTP events

Expected results:

• Success completing call, talking and hanging up when subscriber A originates the

Verify digits in RTP events

H.248 protocol call Failure Case – Subscriber B busy



TEST-5	FAILURE CASE Receiver must be busy
Scope	Verify how system behaves when receiving subscriber number is busy.
Procedure	 Originate Call to subscriber B, whose number must be busy H.248 protocol, using CODEC G.711 From test equipment to IP phone Monitored via wireshark

Objective

- Verify how system behaves when interfunctioning
- Verify if caller receives a busy tone and the message Subtract,
- Verify correct mapping of referred causes (ITU reference Q.931) in the respective ISDN messages, cause 17.

(Reason:Q.931;cause=17;text="user busy").

Expected results on the caller's end:

- Busy tone
- Message CAUSE# 17 User busy

H.248 protocol call using the network Failure Case – B congested



TEST-6	Failure Case B congested
Scope	Verify how system behaves when receiving subscriber number is non-existent
Procedure	 Originate Call to subscriber B, whose number must be congested H.248 protocol, using CODEC G.711 From test equipment to IP phone Monitored via wireshark

Objective

- · Verify how system behaves when interfunctioning
- Verify if caller receives an empty number tone and message 486 Busy Here,
- Verify correct mapping of referred causes (ITU reference Q.931) in the respective release messages

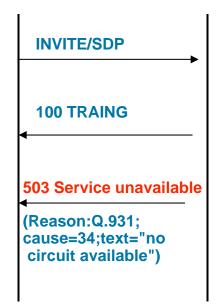
(Reason:Q.931;cause=34;text="no circuit available")

Expected results on the caller's end:

Empty number tone

(Reason:Q.931;cause=34;text="no circuit available")

CLIENT SERVER



Day Two





Workshop at NGN Lab Instrumentation; Protocols: SIP.



Workshop at NGN LabProtocols: H.248



 Workshop at NGN Lab interoperability aspects. SIP-ISUP SIP –I (ITU-T Rec. Q.1912.5 Profile C)



Workshop at NGN Lab Voice Quality (PESQ ITU-T Rec. P.862)



Workshop at NGN Lab interoperability aspects. SIP-ISUP, SIP-I (ITU-T Rec. Q.1912.5 Profile C).

SIP-I protocol test scenario



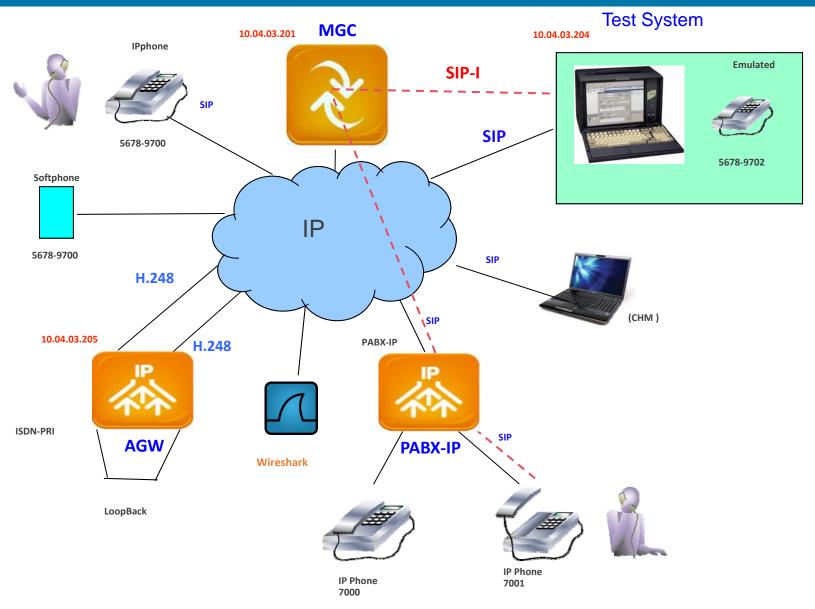
Tests with test equipment

NGN network call SIP-I protocol

- CODEC G.711A success case
- Subscriber A originates call
- Subscriber B originates call
- CODEC G.729A success case
- Subscriber A originates call
- Subscriber B originates call
- ☐ Failure Case
- Release: busy
- Release: congested

Topology SIP-I Protocol





Signaling call SIP-I protocol – Success Case Subscriber A originates CODEC-G.711



TEST-1	SUCCESS CASE From A, CODEC-G.711	
Scope	Verify how system behaves when subscriber is free	
Procedure	 Subscriber A originates a call Protocol SIP-I, using CODEC G.711 From test equipment to IP phone Monitored via wireshark 	
Objective		
 Verify how system behaves when interfunctioning Verify success of call completion and termination, involving NGN elements 		
Expected results: Success completing call, correct codec negotiation and termination.		

CLIENT SERVER INVITE-SDP/IAM 100 TRAING 180 RAING/ ACM 200 OK-SDP/ ANM **ACK** BYE/RELEASE 200 OK/ RLC

Signaling call SIP-I protocol – Success Case Subscriber B originates CODEC-G.711



SUCCESS CASE From A, CODEC-G.711				
Verify how system behaves when subscriber is free				
 Subscriber B originates a call Protocol SIP-I, using CODEC G.711 From test equipment to IP phone Monitored via wireshark 				
 Verify how system behaves when interfunctioning Verify success of call completion and termination, involving NGN elements 				

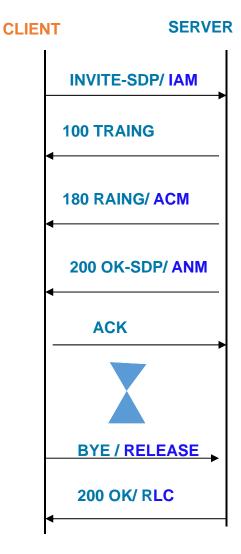
• Success completing call, correct codec negotiation and termination.

CLIENT SERVER INVITE-SDP/IAM 100 TRAING 180 RAING/ ACM 200 OK-SDP/ ANM **ACK** BYE / RELEASE 200 OK/ RLC

Signaling call SIP-I protocol – Success Case Subscriber A originates CODEC-G.729



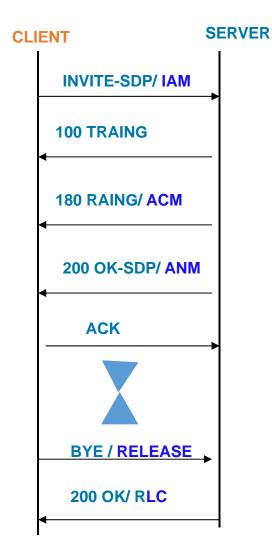
TEST-3	SUCCESS CASE From A, CODEC-G.729			
Scope	Verify how system behaves when subscriber is free			
Procedure	 Subscriber A originates a call Protocol SIP-I, using CODEC G.729 From test equipment to IP phone Monitored via wireshark 			
Verify how system behaves when interfunctioning Verify success of call completion and termination, involving NGN elements				
 Expected results: Success completing call, correct codec negotiation and termination. 				



Signaling call SIP-I protocol – Success Case Subscriber B originates CODEC-G.729



TEST-4	SUCCESS CASE From B, CODEC-G.729			
Scope	Verify how system behaves when subscriber is free			
Procedure	 Subscriber B originates a call Protocol SIP-I, using CODEC G.729 From test equipment to IP phone Monitored via wireshark 			
Objective				
 Verify how system behaves when interfunctioning Verify success of call completion and termination, involving NGN elements 				
Expected results: • Success completing call, correct codec negotiation and termination.				



Signaling call SIP-I protocol Failure Case – B Busy



TEST-5	FAILURE CASE B Busy
Scope	Verify how system behaves when receiving subscriber number is busy.
Procedure	If making an IP phone call, test equipment must receive it Protocol SIP-I, using CODEC G.711 Monitored via wireshark

Objective

- · Verify how system behaves when interfunctioning
- Verify message 486 Busy Here
- Verify correct mapping of referred causes (ITU reference Q.850) in the respective SIP messages

(Reason: Q.850;cause=17;text="user busy").

Expected results:

- Busy tone
- Message 486 Busy Here / REL #17

INVITE-SDP/ IAM 100 TRAING 486 Busy Here / REL #17

SIP-I protocol call Failure Case – B congested



TEST-6	FAILURE CASE B congested			
Scope	Verify how system behaves when receiving subscriber number is non-existent			
Procedure:	 Originate call to subscriber B, whose number must be congested SIP protocol, using CODEC G.711 From test equipment to IP phone Monitored via wireshark 			

Objective

- Verify how system behaves when interfunctioning
- Verify if caller receives an empty number tone and message 486 Busy Here
- Verify correct mapping of referred causes (ITU reference Q.850) in the respective SIP messages

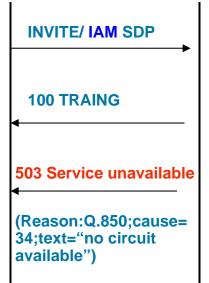
(Reason:Q.850;cause=34;text="no circuit available")

Expected results on the caller's end:

Empty number tone

(Reason:Q.850;cause=34;text="no circuit available")

CLIENT SERVER



Day Two





Workshop at NGN Lab Instrumentation; Protocols: SIP.



Workshop at NGN LabProtocols: H.248



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Workshop at NGN Lab Voice Quality (PESQ ITU-T Rec. P.862)



Workshop at NGN Lab Voice Quality (PESQ ITU-T Rec. P.862)

Voice Quality Measurement Scenario – MOS



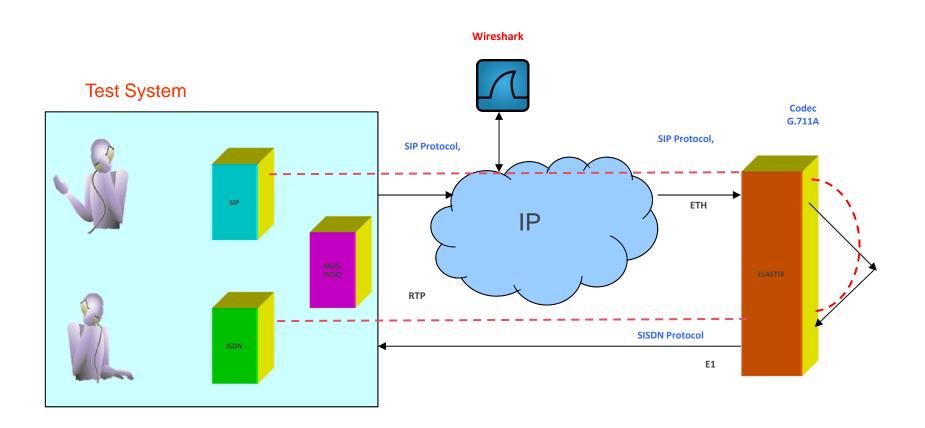
Two test cases will be run:

Case 1: VOICE calls (female and male sentences, SIP-ISDN protocol, using codec **G.711**

Case 1: VOICE calls (female and male sentences, SIP-ISDN protocol, using codec **G.729**

SIP-ISDN calls test topology, with voice quality measurement – MOS



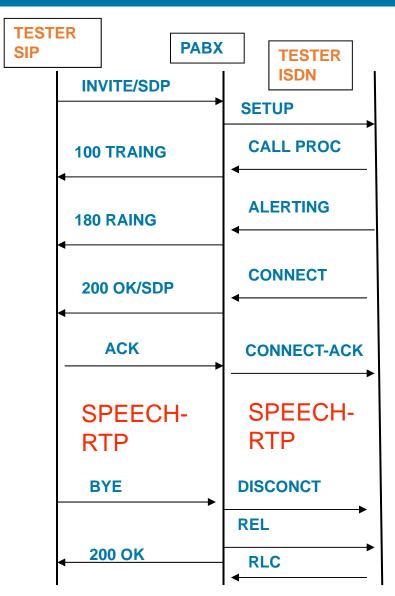


Perceptual Evaluation of SpeechQuality (PESQ) in standard P.862

MOS Tests with CODEC G.711



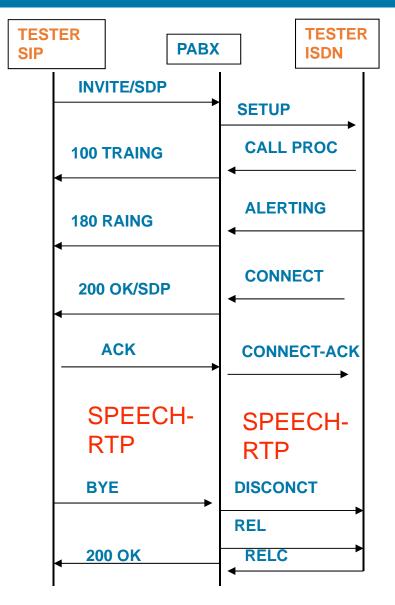
TEST-1	MOS tests with CODEC G.711			
Scope	Verify call completion and mos Mos measurements for codec G.711			
Procedure	 Simultaneous SIP calls shall be made, originating from test equipment, with male and female sentences, to be received by the IP-PABX The PABX-IP shall forward the calls to the ISDN route, to be answered by the test equipment CODEC G.711 From SIP test equipment, passing through the IP-PABX, to be received by the ISDN test equipment. Monitored via wireshark 			
Objective				
 Verify how system behaves when interfunctioning Verify call completion Verify results of MOS FROM MOS score Verify packet loss jitter delay 				
Expected Results: Complete calls (without loss) Mos SCORE within satisfactory standards				



MOS Tests with CODEC G.729



TEST-2	MOS G.729	tests	with	CODEC	
Scope	Verify call c codec G.729		d mos Mos r	measurements for	
Procedure	Simultaneous SIP calls shall be made, originating from test equipment, with male and female sentences, to be received by the IP-PABX The PABX-IP shall forward the calls to the ISDN route, to be answered by the test equipment CODEC G.729 From SIP test equipment, passing through the IP-PABX, to be received by the ISDN test equipment Monitored via wireshark				
Objective					
 Verify how system behaves when interfunctioning Verify call completion Verify results of MOS FROM MOS score Verify packet loss jitter delay 					
Expected Results: Complete calls (without loss) Mos SCORE within satisfactory standards					





Thanks!

<<NomeNomeNome>>

<<Área (UA)>>

Converged Networks Corporate Department – DRC

Phone: (+55) 19 3705-XXXX

xxxxxxx@cpqd.com.br

www.cpqd.com.br