IP Communications Expert

SIP INTEROPERABILITY CERTIFICATE

CERTIFICATION STATEMENT

ILEXIA , independent tests laboratory, certifies the interoperability, using SIP protocol, of :

- ✓ **Patton 5490 e-SBC** and Microsoft Skype for Business 2015 with compatible IP Phones
- ✓ Public SIP Trunking Service

CERTIFICATION AUTHORITY

ILEXIA, independent tests laboratory specialized in IP Telephony interoperability certification

CERTIFICATION DATE

December 2015

CERTIFICATION SCOPE

This certificate is valid in both the functional scope and the firmware/software release mentioned in this document

VENDOR CERTIFIED

Patton in **Bouygues Telecom** service provider's environment

PRODUCT CERTIFIED

Patton 5490 (Trinity OS 3.8.1)

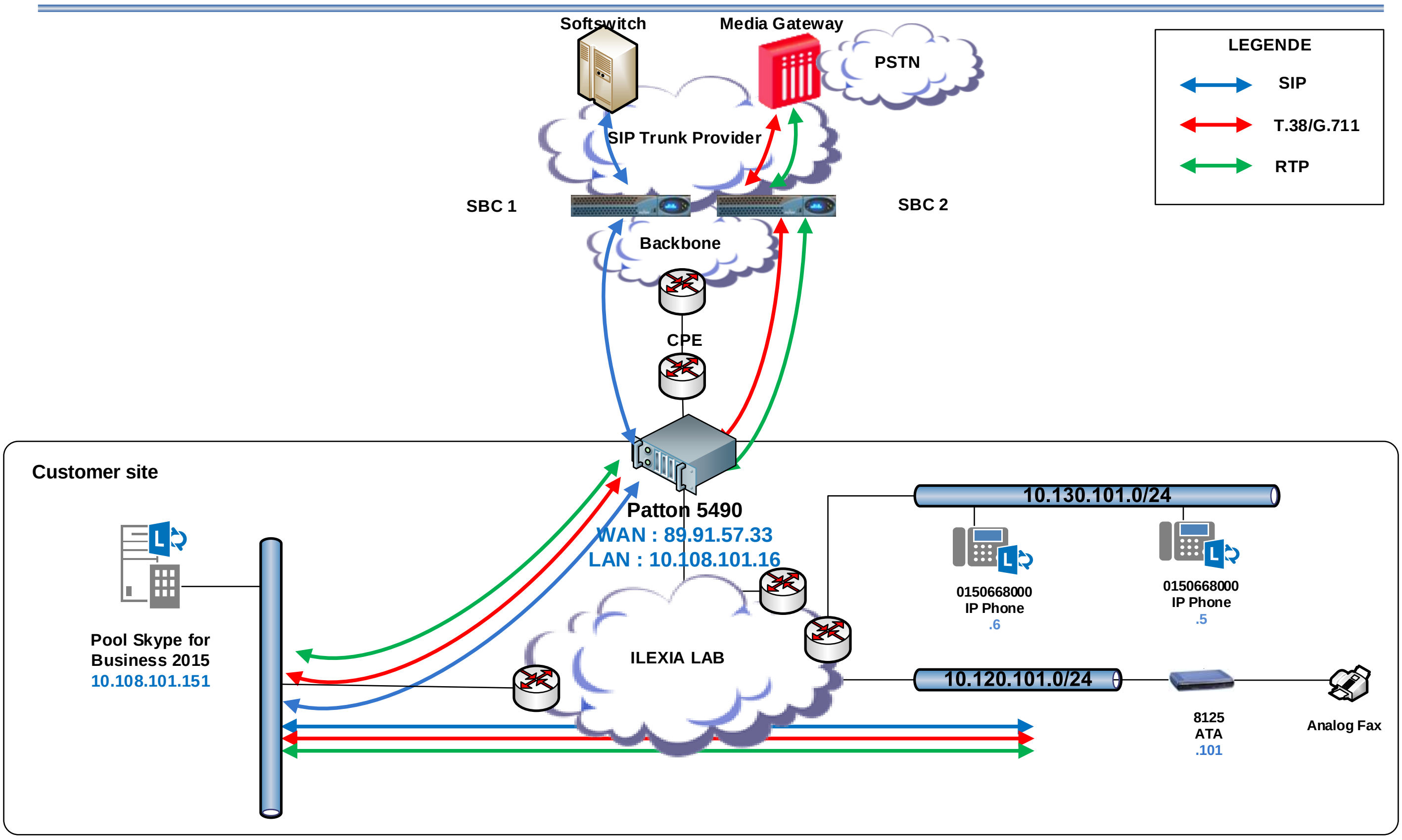
FUNCTIONAL SCOPE

FEATURES CERTIFIED	
Registration/Peering	**National/GSM calls**
Basic call with Skype compatible IP Phone	**Codecs G.729/G.711**
Caller id display	**DTMFs handling (RFC2833/Inband)**
Caller id restriction	**3 way conferencing**
CLIP/CLIR	**Call Hold/Retrieve without MOH**
Transfer (blind and supervised) G.711 only	**Fax (T.38 / G.711 Passthrough)**

LAB COMPONENTS AND SOFTWARE LEVEL

Manufacturer	System	Release	Protocol tested
Patton	**Patton 5490**	Trinity 3.8.1	**SIP**
Microsoft	**Skype for Business 2015**	6.0.9319.0	**SIP**
CIRPACK	**SBC**	**4.7**	**SIP**
CIRPACK	**Softswitch**	**4.7**	**SIP**

CERTIFICATION TEST ENVIRONMENT



IOT Patton 5490 – Skype for Business 2015 – SIP Trunk

02/12/2015

DETAILED SIP INTEROPERABILITY TESTS RESULTS TABLE

N°	TEST TITLE	DETAILS	RESULT	RESTRICTIONS or COMMENTS
	Basic call setup	**Off-net calls Outgoing/Incoming**		
1	Basic outgoing call to PSTN/national user	Proper calling number display G.729 20 ms as negotiated codec	PASS	-
2	Basic outgoing call to PSTN/national user	Proper calling number display G.711 20 ms as negotiated codec	PASS	-
3	Basic outgoing call to GSM	Proper calling number display G.729 20 ms as negotiated codec	PASS	-
4	Basic outgoing call to GSM	Proper calling number display G.711 20 ms as negotiated codec	PASS	-
5	Basic incoming call from PSTN/national user	Proper calling number display G.729 20 ms as negotiated codec	PASS	-
6	Basic incoming call from PSTN/national user	Proper calling number display G.711 20 ms as negotiated codec	PASS	-
7	Basic outgoing call to PSTN	CLIP/CLIR	PASS	-
8	Basic incoming call to PSTN	CLIP/CLIR	PASS	-
9	Basic with hold/retrieve		PASS	Without Music on Hold
10	Basic with hold/retrieve (MOH)		KO	Music on Hold played but not forwarded to provider
	Call transfer	**Off-net/On-net and Off-net/Off-net**		
11	Blind transfer initiated by Audiocode IP Phone G.711 20 ms as negociated codec	Off-net / Off-net	PASS	-
12	Blind transfer initiated by Aastra IP Phone G.711 20 ms as negociated codec	Off-net / Off-net	PASS	-
13	Monitored transfer initiated by Audiocode IP-Phone G.711 20 ms as negociated codec	Off-net / Off-net	PASS	-
14	Monitored transfer initiated by Aastra IP-Phone G.711 20 ms as negociated codec	Off-net / Off-net	PASS	-
15	Monitored transfer initiated by Audiocode IP-Phone G.729 20 ms as negociated codec	Off-net / Off-net	PASS	-
16	Monitored transfer initiated by Aastra IP-Phone G.729 20 ms as negociated codec	Off-net / Off-net	KO	DTMF RFC2833 Payload not sent on Call Transfer 200OK(SDP)
	Conferencing	**3 way conferencing**		
17	Conference initiated by Aastra IP-Phone	Off-net / Off-net	PASS	-

N°	TEST TITLE	DETAILS	RESULT	RESTRICTIONS or COMMENTS
18	Conference initiated by Audiocodes IP-Phone	Off-net / Off-net	PASS	-
	DTMFs handling			
19	RFC 2833/INBAND - Outgoing call initiated with RFC2833 codec transcoded in G.711/INBAND	-	PASS	-
20	RFC 2833/INBAND - Incoming call initiated with G.711/INBAND codec transcoded in RFC2833	-	PASS	-
	Fax services			
21	Outgoing Fax	Fax (T.38)	PASS	-
22	Incoming Fax	Fax (T.38)	PASS	-
23	Outgoing Fax	Fax (G.711 Passthrough)	PASS	-
24	Incoming Fax	Fax (G.711 Passthrough)	PASS	-
25	Outgoing Fax	Fax (T.38 Fallback G.711)	PASS	-
26	Incoming Fax	Fax (T.38 Fallback G.711)	PASS	-

TABLE LEGEND	
DTMF	Digital Tone Multi Frequency
PASS	Test passed successfully
NOK	Test passed successfully with restriction/clarification
KO	Tests failed
N/A	Not Applicable or not testable because not available or not supported on the system
SIP	Session Initiation Protocol

Since 2002, ILEXIA is an independent laboratory specialized in testing and validating IP Telephony systems in numerous environments (enterprises, operators). Our goal is to validate the compatibility of enterprise and service provider IP telephony products.

The tests & validations that we undertake rely heavily on our LAB equipped with up to date telecommunication systems used in both enterprise and Service Provider market.
Our engineers have been trained and certified on the different IP Telephony products available, and have acquired a complete and detailed understanding of the protocols used for IP Telephony such as : SIP, SIPS, SDP, SIP/CSTA, MGCP, H.248, MEGACO, RTP, SRTP, IMS, WebRTC.
In this way, we provide different testing sessions for various customers (Manufacturers, Operators, Distributors, as well as for Businesses and Local governing entities).

The advantages of ILEXIA Interoperability certification

- ✓ **Field Expertise on implementation of business trunking offer**
- ✓ **An "all inclusive" offer (equipment, services and resources)**
- ✓ **The constant update of our IP telephony solutions (including IPBX, application server, gateways, IP Phones, ….) and a guarantee of performance with the latest software releases**
- ✓ **Adaptability to different customers situations (WAN, MAN and LAN multi-sites architecture, IPBX specific software version, interaction with external applications, ...)**
- ✓ **Very competitive prices including the provisioning of equipment to be tested**
- ✓ **A recognized know-how**
- ✓ **A rapid response to requests for certification**
- ✓ **Consulting support to product development from design phase to production**

Equipments available in ILEXIA LAB for interoperability certification
Alcatel OmniPCX Enterprise & Office
Avaya / Nortel Communication Server 1000
Asterisk
Oracle Communications (Acme Packet) SBC
Audiocodes Voice Gateway, SBC, SBA
Centile Istra SIP Server
Cisco Unified Communication Manager / CUPS / Unity
Cisco Unified Communication Manager Express / UC500
Comverse IMS Softswitch
Ingate SIParator SBC
Innovaphone PBX
Microsoft Skype for Business
Mitel MiVoice (Aastra) 5000 / UCP / TWP
Mitel MiVoice Business Controller (MCD)
OneAccess Gateway
Patton SmartNode SBC and Voice Gateway
Shoretel ShoreGear PBX
SIP Express Router
Sonus Voice Gateway, E-SBC, SBA
Tiptel PBX
Unified (Siemens) OpenScape Voice / Entreprise

Testing equipment in ILEXIA LAB for interoperability certification
Spirent Abacus 5000
Tektronix Spectra 2

Functionality usually certified
Registration process
Basic call (incoming and outgoing)
Call Hold/Retrieve
Speed Call List
Missed, received and placed call list
Multi-line
Multiple lines appearance
Music on Hold
Management DTMF (access to voice servers)
Name Display
Call Transfer with and without consultation
Forwarding (all types)
Voicemail (storage, consultation, notification)
Message Waiting indication
Conference
Fax handling (T38, T30)
Distinctive ringing (internal and external call)
Call admission control
Audio codec selection (G.729, G.711 and G.722)
Video codec selection (H.261 and H.263)
Hot Desking
Call park
Camp On Busy
Click to Call
Management of potential situations of failure (failure to VPN access, network and service platform)

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