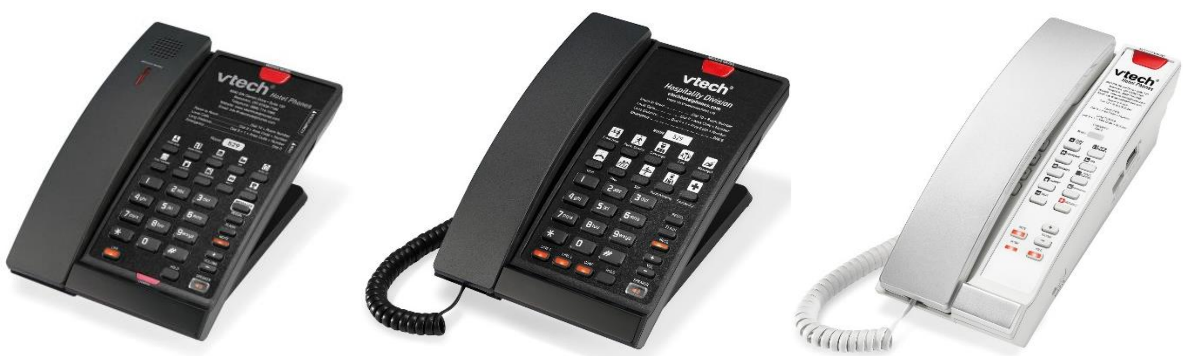
NEC

Enterprise Solutions

Standard SIP Terminal
Generic Interoperability Test Plan



Phone model	Serial nr
VTech CTM-S2411 Matte Black	14AEDB1E6A36
VTech S2220 Black	14AEDB1941A4
VTech S2211 Silver	00122A484E45

Manufacturer : VTech	**Status :** Accepted	**Version:** 1.1
Models: S2411, S2220, S2211 **Firmware version :** SIP_58.3.80.10 / SIP_45.3.77.00	**Tested by:** André Blokland **Department:** NEC TS Technical Support	
Telephone system: SV9300	**Initial Test Date:** 5 October 2017	

Test History

Version	Date	Test Engineer	Description
0.1	11 September 2017	André Blokland	Document creation
0.2	12 September 2017	André Blokland	Added test results
1.0	14 September 2017	André Blokland	Added conclusion
1.1	5 October 2017	André Blokland	Hotline support added

Section 1: Introduction

NEC telephone systems provide a SIP extension interface that can be used to connect both NEC and third party manufacturer SIP telephones ("devices"). Interoperability Testing (IOT) is carried out to determine if the SIP telephone is compatible with the basic functionality of the telephone system. If these tests are satisfactorily completed, the device can then be "Certified" by NEC. This Interoperability Test Plan (ITP) defines a set of test procedures that can be used to verify the interoperability between the NEC telephone system and a SIP terminal. The test cases described in this document cover industry-accepted requirements including but not exclusive to:

- RFC3261 "SIP: Session Initiation Protocol"
- RFC4566 "SDP: Session Description Protocol"
- RFC3264 "An Offer/Answer Model with the Session Description Protocol (SDP)"

The intention of this document is to briefly describe the test results of two types of VTech SIP phones running on the SV9300.

1.1 Scope

The following hardware and software is part of the configuration to test the VTech phones:

- SV9300 PBX.
- DT700, DT300 and standard SIP phones.
- VTech SIP phone CTM-S2411 Matte Black.
- VTech SIP phone S2220.
- VTech SIP phone S2211.

1.2 Not tested features

The following features are not tested during the certification of VTech on SV9300:

- NAT, STUN, Net service, firewall features, VLAN.
- VAD: Voice Activity Detection. VAD can reduce RTP data to not send silence packets.
- AGC: Automatic Gain Control. AGC can keep constant voice volume.
- AEC: Automatic Echo Cancellation. AEC can improve voice quality.
- SRTP: Secure Real-time Transport Protocol
- Packet Loss Compensation, Adaptive Jitter Buffer.
- Support two SIP servers working at the same time.
- Provide a backup SIP server.
- Black List: list of numbers not to be accepted. Restricted List: list of numbers not to be dialed.
- 802.1x authentication.
- Link Layer Discovery Protocol. (LLDP, IEEE 802.1AB)
- Point-to-Point Protocol over Ethernet. (PPPoE)
- Syslog server logging.
- Dialing rules.
- Call Park.

1.3 Test Environment
Tests are carried out within NEC labs using a LAN environment. Tests are done in a single IP domain with one switch to which all equipment is connected. Only a few Wireshark traces per test category will be created.

1.4 References
More information about the products in the configuration can be read in:
- SV9300 manuals for the hardware, commands, system and programming.
- VTech SIP phone configuration guide.

1.5 Glossary of Terms

CLIP	Calling Line Identification Presentation
DDI	Direct Dial-In
DHCP	Dynamic Host Configuration Protocol
DND	Do Not Disturb
DNR	Directory number
DNS	Dynamic Name Server
DTMF	Dual Tone Multi-Frequency
FTP	File Transfer Protocol
IPG	IP Gateway
ISG	In-System Gateway
IVR	Interactive Voice Response
MAC	Media Access Control
NTP	Network Time Protocol
ODN	Originally Dialled Number
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
SNTP	Simple Network Time protocol
TCP	Transmission Control Protocol

1.6 Test Equipment and Software
The following hardware and software is part of the configuration to test the VTech phones:
- SV9300 firmware release V5.3.0.
  SV9300 programming can be read in appendix A.
- DT800, DT400 and standard SIP phones.

Vtech phone Item	CTM-S2411	S2220	S2211
MAC address	14:ae:db:1e:6a:36	14:ae:db:19:41:a4	00:12:2a:48:4e:45
Hardware version	8422011_RF_LARGE	8422011	8350005
Feature mask	0x00000E81	0x0000000A	0x00000909
Boot version	VTechBoot 1.05.00	VTechBoot 1.05.00	VTechBoot 1.06.0B
Firmware version	SIP_58.3.80.10	SIP_58.3.80.10	SIP_45.3.77.00
Release date	July 7 2015 17:38:59	July 7 2015 - 17:38:59	Dec 13 2013 - 14:54:19
Audio profile version	S2410 S2420 0006	S2100 S2210 S2220 0007	S2211 S2221 0009
IP Address	192.168.121.171	192.168.121.172	192.168.121.178
Telephone number	2031	2032	2033

SECTION 2: Registration

2.1 Registration without Authentication
The object of these tests is to determine if the VTech phones are able to register to the SV9300 and maintain that registration.

2.1.1 Successful Registration (Test result OK)
VTech registers to the SV9300.
REGISTER request message sent from the VTech to the SV9300 and replied with a 200 OK.

2.1.2 Registration Refresh (Test result OK)
Re-registration requests from the VTech phones to the SV9300 occurs before the registration expires.
The registration of the VTech with the telephone is maintained.
Periodic REGISTER request messages sent from the VTech to the SV9300 and replied with a 200 OK response.

2.1.3 Registration Failure (Test result OK)
When using a wrong extension number, the registration fails and a 404 Not found response from the SV9300 is sent. Calls from the VTech are disabled.

2.1.4 De-Registration (Test result not OK)
Test Procedure:
- Step 1: Verify that the VTech phones are registered with the telephone system and working.
- Step 2: Disable registration on the VTech phones. Calls from the VTech phones should be disabled.
- Step 3: Verify that registration is successful after re-enabling the registration.

VTech sends a REGISTER request to the telephone system including an Expires field value of 0 to inform the SV9300 the registration should be removed and replied with a 200 OK response. Obviously VTech phones do not support this feature since no message is sent to SV9300.

2.2 Registration with Authentication
Authentication can be enabled on SV9300 in order to increase security so SIP terminals require a valid password and handshake to connect. These tests have this authentication mode set to Enabled therefore the SIP Extension should always send its password during registration.

2.2.1 Successful Registration (Test result OK)
VTech phone register to the SV9300. REGISTER request message sent from the VTech phones to the SV9300 and a “407 Proxy Authentication Required” returned from the SV9300 including authentication handshake details.
VTech phones then re-send the REGISTER request to the SV9300 including authentication details and replied with a 200 OK response.

2.2.2 Registration Refresh (Test result OK)
Re-registration requests from the VTech to the SV9300 occurs before the registration expires. The registration of the VTech with the telephone is maintained.

2.2.3 Registration Failure (Test result OK)
The registration fails and a “404 Not Found” response from the SV9300 is sent. Calls from the VTech are disabled.
Registration is successful after correcting the information.

2.2.4 De-Registration (Test result not OK)
Barring any major network failures, the phones do not de-register once registered. Obviously, if they are rebooted, they will de-register (expiry time = 0) until the boot process has completed. However, considering standard operation

with no major events occurring in the network, the phones will not unregister after registered unless the authentication details are removed.

2.2.5 Registration Failure 480 (Test result OK)
Attempt to register to SV9300 with an already allocated phonenumber fails with 480 Temporarily unavailable. Calls are disabled.

Section 3 Outgoing Calls

This section ensures that the SIP terminal can make calls successfully to other devices on the SV9300, and also externally via the SV9300s lines to the PSTN.
The (seond generation) VTech phones only support En-Block dialing, no overlap dialnig.

3.1 Call to PSTN / ISDN

3.1.1 Call Setup to PSTN/ISDN (Test result OK)
Calls from VTech phones to the PSTN are made. PSTN telephone answers call.
Calling Party hears RBT. INVITE from SIP Extension, ‘100 Trying’ from PBX, ‘183 Session progress’ from PBX, ‘200 OK’ from SIP Extension. Called Party hears Ringing. ACK received from the PBX.
CLIP is displayed at Called Party. Two-way voice path when Called Party answers.
Do this test for external calls via networks based on ISDN, CCIS and SIP.

3.1.2 Call Continuance (Test result OK)
Make a call to the PSTN (SIP). PSTN telephone answers call. Call left in progress for 120 mins
Call is maintained. Periodically REGISTER, ACK and Subscribe messages are output.

3.1.3 Disconnect by Calling Party (Test result OK)
After 2-1-1, the Calling Party hangs up.
Call is cleared by VTech phone, all VoIP resources are freed. BYE sent to the PBX and an ‘200 OK’ returned.
Called Party hears the indication ‘The other person has hung up’.

3.1.4 Disconnect by Called Party (Test result OK)
After 2-1-1, the Called Party hangs up. PSTN Call is cleared.
Calling Party hears the indication ‘The other person has hung up’.
After about 2 seconds the BYE message is sent from the PBX and is given a ‘200 OK’ response from the SIP extension. The Network message is replaced with the SIP extensions method of ending the call. This may be a NU tone or just freeing up the interface. All VoIP resources are freed.

3.1.5 Call Cancel (Test result OK)
Make a call to the PSTN from a VTech phone. Calling Party clears call before it is answered.
Call is cleared. CANCEL request is sent to PBX and ‘200 OK’ in return.
SV9300 also sends ‘487 Request terminated’, ACK from SIP extension,
CANCEL request sent to the SV9300 to again and 200 OK in return.
Do this test for external calls via networks based on ISDN, CCIS and SIP.

Section 3.2: Call internal to phones

3.2.1 Call Setup to telephone (Test result OK)
Make a call to a telephone on the same system. Telephone answers call.
Calling Party hears RBT. Called Party hears Ringing. INVITE from SIP Extension, ‘100 trying’ from PBX, ‘180 Ringing’ from the PBX. CLIP is displayed at Called Party.

Two-way voice path when Called Party answers. ‘200 OK’, ACK.
During the call, only call duration is displayed on the VTech phones, no name or telephone number.
Do this test for different type of destination phones: standard SIP, DT800, DT400, UT880 and DECT.

3.2.2 Call Continuance (Test result OK)
Make a call to a telephone on the same SV9300. Telephone answers call.
Call left in progress for 120 minutes. Call should be maintained.
Periodically REGISTER and ACK messages are output.

3.3.3 Disconnect by Calling Party (Test result OK)
After 2-2-1, the Calling Party hangs up
Call is cleared. BYE message sent from SIP extension and 200 OK from the PBX
All VoIP resources are freed. Called Party hears Busy Tone.

3.3.4 Disconnect by Called Party (Test result OK)
After 2-2-1, the Called Party hangs up. Called party call is cleared.
Calling Party SIP extension hears Busy tone played by the PBX.
After about 2 seconds the BYE message is sent from the PBX and is given a ‘200 OK’ response from the SIP extension. The Busy tone is replaced with the SIP extensions method of ending the call.
All VoIP resources are freed.

3.3.5 Call Cancel (Test result OK)
Make a call to a telephone on the same system. Calling Party clears call before it is answered.
Call is cleared. CANCEL request is sent to PBX and ‘200 OK’ in return.
SV9300 also sends ‘487 Request terminated’, ACK from SIP extension,
CANCEL request sent to the SV9300 to again and 200 OK in return.
All VoIP resources are freed. Called Party stops ringing

Section 4 Inconing Calls

This section ensures that the VTech telephones can successfully receive calls from other extensions on the SV9300, and via the PSTN/ISDN (DDI). Incoming PSTN calls have CLI enabled.

4.1 Incoming call from PSTN/ISDN

4.1.1 Call Setup from PSTN/ISDN (Test result OK)
Incoming call to VTech telephone from PSTN. Call answered.
Calling Party hears RBT. INVITE sent to SIP Extension. ‘100 Trying’, ‘180 Ringing’ sent from SIP Extension.
Called Party hears Ringing. CLIP is displayed at Called Party.
Two-way voice path when (Called) SIP Party answers. SIP extension sends 200 OK and the PBX returns an ACK back.
Note that VTech phones only show the CLIP when ringing; during the call you see the call duration)

4.1.2 Call Continuance (Test result OK)
Incoming call to VTech telephone from PSTN. Call answered. Call left in progress for 120 mins.
Call is maintained.Periodically REGISTER and ACK messages may be output. Also additional INVITE with SD followed by 200 OK may be output.

4.1.3 Disconnect by Calling Party (Test result OK)
After 3-1-1, the Calling Party (PSTN) hangs up. SIP Extension hears Network Call termination Tone (or indication ‘The other person has hung up’). After about 20 seconds the BYE message is sent from the PBX and should be given a ‘200 OK’ response from the SIP extension. The Network message is replaced with the SIP extensions method of ending the call. This may be a NU tone or just freeing up the interface. All VoIP resources are freed.

4.1.4 Disconnect by Called Party (Test result OK)
After 3-1-1, the Called Party hangs up. BYE message sent from SIP extension and '200 OK' from the PBX.
All VoIP resources are freed. Calling Party (PSTN) hears Network Call termination Tone (or indication 'The other person has hung up').

4.1.5. Call Cancel (Test result OK)
Incoming call to VTech telephone from PSTN. Calling Party clears call before it is answered.
PSTN incoming call. INVITE to SIP extension, '100 Trying' and '180 ringing' from SIP extension.
PSTN clears down. CANCEL sent by the PBX, '200 OK' returned by SIP extension.
'487 Request cancelled' sent from SIP extension with 200 OK returned from the PBX.
All VoIP resources are freed. Called SIP Extension stops ringing

4.2 Incoming call from Telephone

4.2.1 Call Setup from telephone (Test result OK)
Incoming call to VTech telephone from Telephone. Call answered.
Calling Party hears RBT. Called Party hears Ringing.
PBX sends INVITE and the SIP Extension returns with '100 Trying', '180 Ringing'.
CLIP is displayed at Called Party.
SIP Extension answers the call. 200 OK' sent to the PBX which returns ACK.
Two-way voice path when Called Party answers.

4.2.2 Call Continuance (Test result OK)
Incoming call to VTech telephone from Telephone. Call answered. Call left in progress for 120 minutes.
Call should maintained. Periodically REGISTER and ACK messages are output.

4.2.3 Disconnect by Calling Party (Test result OK)
After 3-2-1, the Calling Party hangs up. Telephone ends the call. SV9300 plays busy tone to the SIP extension.
After 16 seconds a BYE message is sent to the SIP extension and it replies with '200 OK'.
The Busy tone is replaced with the SIP extensions method of ending the call. This may be a NU tone or just freeing up the interface. All VoIP resources are freed

4.2.4 Disconnect by Called Party (Test result OK)
After 3-2-1, the Called Party hangs up. SIP extension ends the call. A BYE message is sent to the PBX which responds with '200 OK'. Telephone receives busy tone immediately. All VoIP resources are freed

4.2.5 Call Cancel (Test result OK)
Incoming call to VTech telephone from Telephone. Calling Party clears call before it is answered.
CANCEL sent from the PBX to SIP extension which replies with '200 OK'.
SIP extension sends '487 Request cancelled' which the PBX responds with ACK.
Call should be cleared, all VoIP resources are freed. SIP Extension stops ringing.

Section 5: Codec & Media Tests
The purpose of these tests is to confirm which supported Codecs on the SV9300 are also supported by the VTech.

5.1 Supported Codecs and payload types
According to the configuration guide, VTech phones supports G.711 (u-law and A-law) and G.722 Codecs.
Since SV9300 does not support G.722 for standard SIP phones, during the test the focus is only on G.711.

5.2 G711 a-law Codec Tests (Test result OK)
Based on the possibilities of VTech phones and SV9300, the following features are tested:

Nr	VTech setting	SV9300 Setting	VTech calls DT800/400	DT800/400 calls VTech	VTech calls VTech	VTech calls Peer2P DT800/400	VTech calls Peer2P VTech
1	G711a (payload cannot be adjusted)	G711 20ms	OK	OK	OK	SV9300 does not support	OK
2		G711 30ms	OK	OK	OK	SV9300 does not support	OK
3		G711 40ms	OK	OK	OK	SV9300 does not support	OK

5.2 G711 µ-law Codec Tests (Test result OK)
Based on the possibilities of VTech phones and SV9300, the following features are tested:

Nr	VTech setting	SV9300 Setting	VTech calls DT800/400	DT800/400 calls VTech	VTech calls VTech	VTech calls Peer2P DT800/400	VTech calls Peer2P VTech
1	G711µ (payload cannot be adjusted)	G711 20ms	OK	OK	OK	SV9300 does not support	OK
2		G711 30ms	OK	OK	OK	SV9300 does not support	OK
3		G711 40ms	OK	OK	OK	SV9300 does not support	OK

5.3 G729 Codec Tests (Test result not OK)
VTech phones do not support G729.

5.4 CODECs Disabling and reordering
Some SIP phones will have the ability to use CODECs that are not supported by the PBX. In this case, an alternative supported CODEC should be used automatically. Tests are to prove PBX CODECs are chosen.

5.4.1 Disabled CODEC outgoing call (Test result OK)
Enable only the CODECs on the VTech phones which are not supported by SV9300. Make outgoing calls from the VTech phones to a telephone or PSTN Trunk. Call should fail with VTech phones being presented wih Busy Tone. INVITE message will be sent to the SV9300. The SV9300 responds with '488 Not Acceptable' followed by the ACK from the VTech phone letting the PBX know it understands the '488'.

5.4.2 Disabled CODEC incoming call (Test result OK)
Enable only the CODECs on the VTech phones which are not supported by SV9300. Make incoming call from a Telephone or PSTN Trunk into the VTech phone. Call rings at VTech. VTech sends INVITE to the SV9300 and the SV9300 returns '100 Trying' and 180 ringing. When answered the calling party will be presented with Busy tone and the call will not connect. The VTech phone will stop ringing and will provided with a Busy Tone. '200 OK' with SDP will be sent from the VTech phone and a BYE will be returned from the SV9300. '200 OK' will be returned from the VTech phone.

5.4.3 CODEC Order outgoing call (Test result OK)
Enable all unsupported CODECs on the VTech phones apart from the last in the list which should be a supported SV9300 CODEC. Make outgoing call from VTech phone to a Telephone or PSTN Trunk.

INVITE from the VTech phone with SDP, ‘100 Trying’ from the SV9300 followed by ‘183 Session progress’. Answer the called party. ‘200 OK’ with SDP will be sent from the SV9300 then a return of ACK Should then come back from the VTech phone. The alternative CODEC should be used.

5.4.4 CODEC order incoming call (Test result OK)
Enable all unsupported CODECs on the VTech phones apart from the last in the list which should be a supported SV9300 CODEC. Make incoming call from a Telephone or PSTN Trunk into the VTech phone.
INVITE from the SV9300 with SDP, ‘100 Trying’ from the VTech phone followed by ‘183 Session progress’.
Answer the called SIP party. ‘200 OK’ with SDP will be sent then a return of ACK Should then come back from the SV9300. The alternative CODEC should be used instead. Make a note of actual CODEC used (use Wireshark capture)

Section 6 Functional (Feature) Tests

In case SIP phones connected to SV9300, DTMF signal sending and receiving is possible based on In-band DTMF or based on Out-band DTMF (RFC2833). All commands related to DTMF settings in SV9300 can be found in the SV9300 System manual.

In case of trunk calls, the following DTMF relay method (assigned by cmBA52) can be used:
- Out-band DTMF (RFC2833).
- In-band DTMF (Voice pass through).

The table below shows the possible combinations of DTMF relay methods in SV9300:

Setting at opposite office	**RFC 2833 method**		**In-band method**	
Setting at own office	Receiving at own office	Receiving at opposite office	Receiving at own office	Receiving at opposite office
RFC 2833 method	RFC 2833	RFC 2833	In-band	In-band
In-band method	In-band	In-band	In-band	In-band

DTMF (RFC2833) that can be sent or received are “0-9”, * and “#”.
The values “A”, “B”, “C” and “D” are not supported.

6.1 Out-band DTMF (RFC2833) for internal calls (Test result OK)
Switch on DTMF relay (RFC2833) on the VTech phones.
Make some internal calls and check for example with the UM8000 voicemail menu whether DTMF works fine.

6.2 In-band DTMF for internal calls (Test result OK)
Switch off DTMF relay (RFC2833) on the VTech phones.
Make some internal calls and check for example with the UM8000 voicemail menu whether DTMF works fine.

6.3 Out-band DTMF (RFC2833) for trunk calls (Test result OK)
Switch on DTMF relay (RFC2833) on the VTech phones. Make some internal calls and check for example with the UM8000 voicemail menu whether DTMF works fine.

6.4 In-band DTMF for trunk calls (Test result OK)
Switch off DTMF relay (RFC2833) on the VTech phones. Make some internal calls and check for example with the UM8000 voicemail menu whether DTMF works fine.

6.5 Consult Transfer by flash hook or recall (Test result OK)
Make a call from the VTech phone to a phone, answer that call.

Use Line 2 of this VTech phone to make an enquiry call to another phone, and answer it.
Transfer the call by hanging up the VTech phone. Call is transferred and established.

6.6 Blind Transfer by flash hook or recall (Test result OK)
Make a call from the VTech phone to a phone, answer that call.
Use Line 2 of this VTech phone to make an enquiry call to another phone, and do not answer it.
Transfer the call by hanging up the VTech phone. Call is transferred and is established when anwering it.

Section 9: Outgoing Calls with Peer to Peer enabled

This section ensures that the VTech telephones can make calls successfully to other extensions on the SV9300, and via the PSTN with Peer to Peer enabled.

9.1 Call to PSTN / ISDN Peer to Peer enabled (Not supported by SV9300)
With regard to standard SIP phones, the peer to peer SIP feature is only supported for internal calls between SIP phones. More details about this feature can be found in the SV9300 System manual, section "Conditions for Peer-to-Peer Connection with Standard SIP Terminal".

9.2 Call internal to Telephone Peer to Peer enabled

9.2.1 Call Setup (Telephone) Peer to Peer enabled (Test result OK)
Make a call to a telephone on the same SV9300. Telephone answers call.
Calling Party hears RBT. Called Party hears Ringing. INVITE from SIP Extension, '100 trying' from PBX, '180 Ringing' from the PBX. CLIP is displayed at Called Party. Two-way voice path when Called Party answers. '200 OK', ACK.

9.2.2 Call Continuance (Telephone) Peer to Peer enabled (Test result OK)
Make a call to a telephone on the same SV9300. Telephone answers call. Call left in progress for 30 minutes.
Call should be maintained.Periodically REGISTER and ACK messages may be output. Also additional INVITE with SD followed by '200 OK' may be output.

9.2.3 Disconnect by Calling Party (Phone) Peer to Peer enabled (Test result OK)
After 9.2.1, the Calling Party hangs up. Call should be cleared. BYE message sent from SIP extension and 200 OK from PBX. All VOIP resources are freed. Called Party hears Busy Tone.

9.2.4 Disconnect by Called Party (Phone) Peer to Peer enabled (Test result OK)
After 9.2.1, the Called Party hangs up. Called party call should be cleared.
Calling Party SIP extension hears Busy tone played by the PBX.
BYE message is sent from the PBX and should be given a '200 OK' response from the SIP extension.
The Busy tone is replaced with the SIP extensions method of ending the call. This may be a NU tone or just freeing up the interface. All VOIP resources are freed.

9.2.5 Call Cancel (Phone) Peer to Peer enabled (Test result OK)
Make a call to a Phone on the same system. Calling Party clears call before it is answered.
Call should be cleared. CANCEL request set to PBX and '200 OK' in return. SV9300 also sends '487 Request terminated', ACK from SIP extension, CANCEL request sent to PBX to again and 200 OK in return.
All VOIP resources are freed. Called Party stops ringing

Section 10: Specific hotel functions

Check in and check out the VTech phones. During check in, use some specific hotel features like wake up and message waiting. Verify whether the correct messages are sent from/to PMS. During check in, the phone must be able to make external calls, whereas during check out this possibility is blocked.

The following **wake** up functions are tested:

- Set wake up call on VTech guest phone. (Ok)
- Set wake up call via VTech administrative phone. (Ok)
- Check wake up call on VTech guest phone. (Ok)
- Check wake up call via VTech administrative phone. (Ok)
- Cancel wake up call on VTech guest phone. (Ok)
- Cancel wake up call via VTech administrative phone. (Ok)
- Select different tone sources, e.g. speech synthesis, hold Tone Source on CPU blade and Internal Tone Generator. Redo the tests above. (Ok)
- Enter a few maid status codes (room status) and check whether the correct settings / actions are activated. (Ok)
- Activate and de-activate Message Waiting for the VTech phones and check the lamp status. (Ok)

Section 11: Local functions on VTech phones

The following local functions of VTech phones are tested with successful result:

- **Redial**: enter Speaker key and afterwards the redail key. Last number is successfully dialed. (Ok)
- **Speeddial**: program speeddial keys and enter them to make immediately calls.
- **Hotline**: program hotline key, go of hook to make immediately a call.
  Hotline is referred to by VTech as Emergency Offhook Dialing. You can find it in the web GUI on the Other Phone Settings page. This feature works when you lift the handset and (on phones that support handsfree) by pressing the SPEAKER key. (Ok)
- **Auto Answer**: Program auto answer for certain phonenumbers and call the VTech phone. This call is immediately answered. (Not Ok)
- **Hold**: Answer a call on a VTech phone and click on the Hold key to put the call on hold. Hold tone is played at the opposite part. Click on Line 1 to unhold the call and check if two parties can communicate. (Ok)
- **Mute**: Press the Mute key during a call. Then the oppossite party will not hear your conversation or background noise. You will still hear the opposite party. Press mute again to unmute the call and continue the conversation. (Ok)
- **Three-party Conference**: Use the Line 1 and Line 2 to set up two different calls and enter the Conference key to start a conference call for the three actual parties. Enter the Conference key again to leave the conference. (Not Ok)

The following VTech features can be used by applying the <u>webbrowser</u>:

- **Do Not Disturb**. Use the webbrowser to activate DND on the VTech phone. Then the display indicates: Do Not Disturb. When making a call to this phone, it does not ring. The originator just hears ringing all the time. (Limited support)
- **Call Forwarding**. Activating Call Forwarding on VTech phones (via webbrowser!) is not supported on SV9300.
  As an alternative on VTech phones you can enter PBX codes like *21 to successfully forward calls to another destination. You can also configure “s*219876” under a speed dial key to activate Call Forwarding to phone 9876. Configure “s#21” to cancel it. Finally you can also use the Line 1 and Line 2 functions to transfer calls. (Limited support)
- **Timing setting**: SNTP is not supported by VTech phones. (Not Ok)

Section 12: Resilience Tests

12.1 Power out

Switch SIP Peer to Peer in SV9300 off.

12.1.1 Unexpected phone termination (Idle) (Test result OK)

Remove the power for the VTech phone, wait 20 seconds and re-connected the power for this phone.

Re-REGISTER takes place and afterwards the phone can used without problems.

12.1.2 Unexpected phone termination (In conversation) (Test result OK)
Establish a call from the VTech phone to another phone.
Remove the power for the VTech phone, wait 20 seconds and re-connected the power for this phone.
Re-REGISTER takes place and afterwards the phone can used without problems.

Section 12.2: Network issues

12.2.1 Disconnect the Ethernet cable (During idle) (Test result OK)
Ensure the VTech phone is registered and able to make calls. Disconnect Ethernet cable and wait for five minutes. Re-connected the cable to this phone. Re-REGISTER takes place and afterwards the phone can used without problems.

12.2.2 Disconnect the Ethernet cable (During conversation) (Test result OK)
Ensure the VTech phone is registered and able to make calls. Set up a call and keep the conversation ongoing. Disconnect Ethernet cable and wait for five minutes. Re-connected the cable to this phone.
Re-REGISTER takes place and afterwards the phone can used without problems.

12.2.3 Disconnect the SV9300 cable (During idle) (Test result OK)
Ensure the VTech phone is registered and able to make calls.
Disconnect the switch cable connected to the VoIP port of the SV9300 and wait for five minutes.
Re-connected the VoIP cable to SV9300.
Re-REGISTER by VTech takes place and afterwards this phone can used without problems.

12.2.4 Disconnect the SV9300 cable (During conversation) (Test result OK)
Ensure the VTech phone is registered and able to make calls.
Disconnect the switch cable connected to the VoIP port of the SV9300 and wait for five minutes.
Re-connected the VoIP cable to SV9300.
Re-REGISTER by VTech takes place and afterwards this phone can used without problems.

Section 13: Conclusion

This section presents a summary of the test results and just focusses on the problems and limitations of the VTech phones.

13.1 Failures
During the test several limitations are encountered (next section) but no failures.

13.2 Limitations
During the certification the following limitations in Vtech SIP phones are encountered:
- De-registration is not supported.
- Only the G.711 A-law CODEC and the G.711 µ-law CODEC are supported.
- When enabling the SIP peer-to-peer feature in SV9300, this can only be used for internal calls.
- Several local features on the VTech phones can not supported or only limited supported.

13.3 Remarks

13.4 Compatibility Summary

	Compatible/Incompatible
Registration - No Authentication	

Successful Registration	Ok
Registration Refresh	Ok
Registration Failure	Ok
De-registration	Not Ok
Registration with Authentication	
Successful Registration	Ok
Registration Refresh	Ok
Registration Failure	Ok
De-registration	Not Ok
Registration Failure 480	Ok
Call to PSTN/ISDN	
Call Setup	Ok
Call Continuance	Ok
Disconnect by Calling Party	Ok
Disconnect by Called Party	Ok
Call Cancel	Ok
Call internal to Telephone	
Call Setup	Ok
Call Continuance	Ok
Disconnect by Calling Party	Ok
Disconnect by Called Party	Ok
Call Cancel	Ok
Incoming call from PSTN/ISDN	
Call Setup	Ok
Call Continuance	Ok
Disconnect by Calling Party	Ok
Disconnect by Called Party	Ok
Call Cancel	Ok
Incoming call from Telephone	
Call Setup	Ok
Call Continuance	Ok
Disconnect by Calling Party	Ok
Disconnect by Called Party	Ok
Call Cancel	Ok
CODECs Disabling and Reordering	
G.711a-law CODEC	Ok
G.711µ-law CODEC	Ok
G.729	Not supported by VTech
G.723	Not supported by SV9300
G.722	Not supported by SV9300
G.726	Not supported by SV9300
Disabled CODEC outgoing call	Ok
Disabled CODEC incoming call	Ok
CODEC order outgoing call	Ok
CODEC order incoming call	Ok
Functional (Feature) tests	
Out-band DTMF (RFC2833) for internal calls	Ok
In-band DTMF (RFC2833) for internal calls	Ok
Out-band DTMF (RFC2833) for trunk calls	Ok
In-band DTMF (RFC2833) for trunk calls	Ok

Consult Transfer (Flash hoOk/Not Ok/recall)	Ok
Blind Transfer (Flash hoOk/Not Ok/recall)	Ok
Outgoing calls to PSTN/ISDN Peer to Peer enabled	
Call Setup	Not supported by SV9300
Call Continuance	Not supported by SV9300
Disconnect by Calling Party	Not supported by SV9300
Disconnect by Called Party	Not supported by SV9300
Call Cancel	Not supported by SV9300
Call internal to Telephone Peer to Peer enabled	
Call Setup	Ok
Call Continuance	Ok
Disconnect by Calling Party	Ok
Disconnect by Called Party	Ok
Call Cancel	Ok
Hotel functions	
Regular hotel functions	Ok
Local functions on phone	
Redail, Speeddial	Ok
Hotline, Auto answer	Ok
Hold, Mute	Ok
Three party conference	Not Ok
Do not disturb	Limited support
Call forwarding	Limited support
Time setting via SNTP	Not Ok
Power out	
Unexpected SIP extension termination (Idle)	Ok
Unexpected SIP extension termination (In conversation)	Ok
Network issues	
Disconnect the Ethernet cable (During idle)	Ok
Disconnect the Ethernet cable (During conversation)	Ok
Disconnect the Ethernet cable (During idle)	Ok
Disconnect the Ethernet cable (During conversation)	Ok

Appendix A: SV9300 configuration

System Check Report generated on the SV9300 which is used for the certification:

PCPro Version	5.2.0.150
Connection Account	RS-232C(9600bps)
Read Date/Time(PC)	9/14/2017 11:37:56 AM
MJ Alarms	Off

Unit No.	Version information
01	SC-4600 V5.3.0.0

Unit No.	Status	Alarm	JACK Insertion Status
01	Operating normally	No alarm	Not inserted

Unit No.	VoIPDB Kind	Firmware Version	Operation Status	LAN Cable Connection Status

		Blade	CPU Blade	Difference		
01	GPZ-64IPLD	A1-0010.01	A1-0010.01	OK	Normal operation	Connected

Unit No.	Port	IP Address	Subnet Mask	Default Gateway	VLAN	VLAN ID	Priority of VLAN ID
01	Maintenance Port	192.168.120.100	255.255.255.0				
	VoIP Port	192.168.121.104	255.255.255.0	192.168.121.1			
	RTP Port	192.168.121.105					

Hardware Key Code	36LYRAHK0100000000B4Y01372DQ

No.	License Data Name	Software Key Code	Authentication Status
001	System Version License V4	NPDG-JV6M-OQRH-CYNB-3OLU	A
002	Capacity/Function/Terminal License	NPDa-Jn6M-OQRF-CKNH-JOLU	A

No.	Feature Name	Used	Valid	Remains
001	PORT Capacity	79	2300	2221
002	ISDN Terminals	0	256	256
010	VoIP Channels	64	1280	1216
011	IP Trunk	4	346	342
012	IP Ports	2	640	638
016	Remote Unit	0	16	16
017	SIP TRK Channels	16	692	676
018	SoftPhone	0	144	144
019	SoftPhone ACD	0	16	16
020	PS	0	64	64
023	Mobility Access	0	1578	1578
024	STD SIP Phone	3	1040	1037
025	Embedded 32P CNF	0	56	56
043	UMS PORT	0	272	272
044	UMS FAX PORT	0	4	4
045	UMS TTS PORT	0	6	6
046	UMS CLIENT	0	2005	2005
047	UMS LANGUAGE	0	25	25
048	UMS HOSPI LANG	0	15	15
049	UMS TTS LANG	0	10	10
060	PVA PORT	0	256	256
061	RGA PORT	0	512	512
062	RGA Enhance I	0	2	2
065	RGA Language	0	120	120

Unit 01

#1			
01	DLCA(16)	(COT)	04
02	LC(8) *NV		05
03	QSIG	(BRI)	06

Unit No.	Expansion Chassis	Operation Status	Alarm
01	#3 (SLOT 13-18)	Disconnected	-
	#2 (SLOT 07-12)	Disconnected	-
	#1 (SLOT 01-06)	Normal operation	No alarm

Unit No.	Slot No.	Mounted Blade	Firmware Version			System Data		Ports		Operation Status	Link Connection	Serial No.	Lot No.
			Line/Trunk Blade	CPU Blade	Difference	Setup condition	Detail	Blade	Assigned				
01	01	DLCA(16)	2.8	2.8	OK	OK	10:DLC	16	8	Normal operation	-	A13020B4800502	432CHA
01	02	LC(8) *NV	3.2	3.2	OK	OK	20:LC	8	8	Normal operation	-	2B0185BJ300349	J12AAA
01	03	QSIG	3.1	3.1	OK	OK	45:QSIG	32	31	Normal operation	Line 1 alarm	A13037B4800337	432CDC
01	04	(COT)	-	2.A	Unknown	Only System Data	30:COT	-	8	-	-	-	-
01	05	-	-	-	-	-	-	-	-	-	-	-	-
01	06	(BRI)	-	3.3	Unknown	Only System Data	40:BRT	-	4	-	-	-	-

Unit No.	Slot No.	CKT No.	Port	Data	Mounted Blade	Terminal Type	Line Keys
01	01	01	010101	F1021	DLCA(16)	44 : DT330	32
01	01	02	010102	F1022	DLCA(16)		
01	01	03	010103	F1023	DLCA(16)		
01	01	04	010104	F1024	DLCA(16)		
01	01	05	010105	F1025	DLCA(16)	48 : DT430 Chinese	24
01	01	06	010106	F1026	DLCA(16)		
01	01	07	010107	F1027	DLCA(16)		
01	01	08	010108	F1028	DLCA(16)		
01	01	09	010109		DLCA(16)		
01	01	10	010110		DLCA(16)		
01	01	11	010111		DLCA(16)		

01	01	12	010112		DLCA(16)		
01	01	13	010113		DLCA(16)		
01	01	14	010114		DLCA(16)		
01	01	15	010115		DLCA(16)		
01	01	16	010116		DLCA(16)		
01	02	01	010201	1001	LC(8) *NV		
01	02	02	010202	1002	LC(8) *NV		
01	02	03	010203	1003	LC(8) *NV		
01	02	04	010204	1004	LC(8) *NV		
01	02	05	010205	1005	LC(8) *NV		
01	02	06	010206	1006	LC(8) *NV		
01	02	07	010207	1007	LC(8) *NV		
01	02	08	010208	1008	LC(8) *NV		
01	03	01	010301	D101	QSIG		
01	03	02	010302	D102	QSIG		
01	03	03	010303	D103	QSIG		
01	03	04	010304	D104	QSIG		
01	03	05	010305	D105	QSIG		
01	03	06	010306	D106	QSIG		
01	03	07	010307	D107	QSIG		
01	03	08	010308	D108	QSIG		
01	03	09	010309	D109	QSIG		
01	03	10	010310	D110	QSIG		
01	03	11	010311	D111	QSIG		
01	03	12	010312	D112	QSIG		
01	03	13	010313	D113	QSIG		
01	03	14	010314	D114	QSIG		
01	03	15	010315	D115	QSIG		
01	03	16	010316	D116	QSIG		
01	03	17	010317	D117	QSIG		
01	03	18	010318	D118	QSIG		
01	03	19	010319	D119	QSIG		
01	03	20	010320	D120	QSIG		
01	03	21	010321	D121	QSIG		
01	03	22	010322	D122	QSIG		
01	03	23	010323	D123	QSIG		
01	03	24	010324	D124	QSIG		
01	03	25	010325	D125	QSIG		
01	03	26	010326	D126	QSIG		
01	03	27	010327	D127	QSIG		
01	03	28	010328	D128	QSIG		
01	03	29	010329	D129	QSIG		
01	03	30	010330	D130	QSIG		
01	03	31	010331	D131	QSIG		

01	03	32	010332		QSIG		
01	04	01	010401	D061	(COT)		
01	04	02	010402	D062	(COT)		
01	04	03	010403	D063	(COT)		
01	04	04	010404	D064	(COT)		
01	04	05	010405	D065	(COT)		
01	04	06	010406	D066	(COT)		
01	04	07	010407	D067	(COT)		
01	04	08	010408	D068	(COT)		
01	06	01	010601	D140	(BRI)		
01	06	02	010602	D141	(BRI)		
01	06	03	010603	D142	(BRI)		
01	06	04	010604	D143	(BRI)		
01	06	05	010605		(BRI)		
01	06	06	010606		(BRI)		
01	06	07	010607		(BRI)		
01	06	08	010608		(BRI)		

Virtual Port	Data	IP Address	Status	Terminal Type	Line Keys
0020	F1120				
0021	F1121	192.168.121.170	A	18 : DT730CG	12
0022	F1122	192.168.121.173	A	21 : DT830/DT830DG	32
0023	F1123				
0024	F1124				
0025	F1125				
0026	F1126				
0027	F1127				
0028	F1128				
0029	F1129				
0100	EEC000				
0101	EEC001				
0102	EEC002				
0103	EEC003				
0104	EEC004				
0105	EEC005				
0106	EEC006				
0107	EEC007				
0108	EEC008				
0109	EEC009				

Peer to peer CCIS trunk ports

Virtual Port	Trunk No.
000	D050

001	D051
002	D052
003	D053
004	D054

SIP trunk ports

Unit No.	Virtual Port	Trunk No.
01	000	D000
01	001	D001
01	004	D004
01	010	D010
01	011	D011
01	012	D012
01	013	D013
01	014	D014
01	015	D015
01	016	D016
01	017	D017
01	042	D042
01	043	D043
01	044	D044
01	045	D045
01	046	D046
01	047	D047
01	048	D048
01	049	D049

Wireless PCS/PHS terminal ports/Standard SIP ports

Virtual Port	Station No.	IP Address	Status	Terminal Type
000	1030			
001	1031	192.168.121.171	A	80 : Standard SIP Station
002	1032	192.168.121.172	A	80 : Standard SIP Station
003	1033	192.168.121.178	A	80 : Standard SIP Station
004	1034			
005	1035			
006	1036			
007	1037			
008	1038			
009	1039			

Numbering Plan Group 0

Access Code	Data	Description
0	A128	LCR Group 2
10	804	4 digits Station
11	804	4 digits Station
12	A129	LCR Group 3
2	A129	LCR Group 3
502	A129	LCR Group 3
52	A129	LCR Group 3
61	A021	Call Pickup-Direct
62	A020	Call Pickup-Group
63	A129	LCR Group 3
7	A129	LCR Group 3
80	A126	LCR Group 0
90	802	2 digits Station
91	802	2 digits Station
*21	A010	Call Forwarding-All Calls Set
*23	A241	Call Forwarding-Logout/Call Forwarding-PS Out of Cell (Zone)/Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out Set
*24	A024	Wake Up Call/Timed Reminder Set
*25	A025	Wake Up Call/Timed Reminder Cancel
*26	A026	Wake Up Call/Timed Reminder Check
*27	A027	Wake Up Call Set from Predetermined Station (Single Wake Up time operation)
*28	A028	Wake Up Call Set from Predetermined Station (Multiple Wake Up time operation)
*50	A100	Voice Response System Access Record
*51	A101	Voice Response System Access Replay
*52	A102	Voice Response System Access Delete
*53	A022	Do Not Disturb Set From station
*61	A040	MW Lamp Control Set
*63	A263	Peer-to-Peer Connection for Station-to-Station Call with Standard SIP station Set
*64	A264	Peer-to-Peer Connection for Station-to-Station Call with Standard SIP station Cancel
**	A239	IP Station Logout
#21	A011	Call Forwarding-All Calls Cancel
#23	A242	Call Forwarding-Logout/Call Forwarding-PS Out of Cell (Zone)/Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out Cancel
#25	A015	Call Forwarding-Busy Line Cancel
#26	A017	Call Forwarding-No Answer Cancel
#27	A013	Call Forwarding-No Answer/Busy Line Cancel
#37	A003	
#53	A023	Do Not Disturb/Return Message Schedule Cancel/Return Message Schedule Voice
#61	A041	MW Lamp Control Reset