



SIP Extension Compatibility Report

2N IP Verso
SIP Video Modular Doorphone
SV9100 / SL2100

NEC Enterprise Solutions has performed Interoperability Testing with the Platform(s) and the Device listed above on the date specified on the individual Compatibility Report.

Please always refer to the latest edition of a specific Compatibility Report on BusinessNet before considering connection.

If a Device is no longer mentioned in the Index and/or the Compatibility Report is not available then the Compatibility Report has been withdrawn and connection will no longer be supported by NEC Enterprise Solutions.

IMPORTANT – A specific Device may not necessarily be available in every territory of the EMEA Region. Verification should be obtained before proceeding.

Test Completion Date:	1 st March 2021
Manufacturer:	2N
Model:	IP Verso
Software Level:	2.31.0.40.5
Web Site	https://cie-group.com
NEC	
System Tested:	SV9100 / SL2100
Software Version:	10.60.55/ 3.00.02

Please refer to the following page(s) for further
Information and Configuration Notes.

Description of SIP Extension

2N® IP Verso is an elegant and reliable intercom equipped with lots of useful functions. Thanks to SIP support and compatibility with major brands of PBX manufacturers, it can benefit from using VoIP networks. 2N® IP Verso can be used as a door or special purpose intercom for office buildings, residential areas and other applications.

Advantages of Use

- Elegant design
- Weather resistant
- Various modes of installation (flush, surface, plasterboard)
- Sensitive microphone and loud speaker
- Both-way audio communication – acoustic echo cancellation
- Integrated colour HD camera with wide-angle lense and hidden night vision
- Selectable number of quick dial buttons with nametags and backlight
- Optional numeric keypad with backlight
- Option to have multiple modules of the same kind – for example, card reader for both entering and leaving the building
- Integrated switches of electric locks with wide setting options
- Optional integrated RFID card reader module
- PoE or 12 V DC power supply
- Configuration using web interface or dedicated PC application
- VoIP standard SIP 2.0 support
- 10 000 Phone Book positions
- 20 user time profiles
- Video codecs (H.263, H.263+, H.264, MPEG-4, MJPEG)
- Audio codecs (G.711, G.729, G.722, L16/16kHz)
- HTTP server for configuration
- SNTP client for time synchronisation
- RTSP server for audio and video streaming, ONVIF compatible
- SMTP client for email sending, Picture to Email feature
- TFTP/HTTP client for automated firmware and configuration upgrade and update

We have tested the 2N IP Verso SIP Door Entry using a Single Button.



Recommended Software Versions

SV9100:



SL2100



CCPU Version 10.60.55 / PC Pro Version 10.52.56

CCPU Version 3.00.02 / PC Pro Version 3.00.01

Licensing requirements

SV9100:

- 1 x System Port license required
- 1 x IP Advanced License

SL2100:

- 1 x SIP Extension License

System Configuration

The SV9100 CP20 CPU Card must have an IPLe VoIP Daughter Board installed.

The SL2100 has VoIP support as standard called “Embedded VoIP” with an Option to install the VoIP DB card for further support enhancements. Tests performed in this guide was using the Embedded VoIP option.

System Programming

The following items should be changed – all other items are considered irrelevant and as such left as default. Screenshots are for example purposes only and will have been taken from the PBX under test but will apply to the other PBXs listed on the cover of the Compatibility Report. Only differences in programming will be documented where necessary.

Advanced Edit	PRG	Item	Setting
Advanced Items + VoIP + General Settings + IP Addressing + CPU IPL IP Network Setup	10-12-02	Default Subnet Mask	Set according to customers network requirements
	10-12-01	IP Address	Must be in a different network range to IPLE IP Address (10-12-09) Must not be set as 0.0.0.0
	10-12-03	Default Gateway	Set according to customers network requirements
	10-12-09	IPLE/VOIPDB IP Address	Set according to customer's network requirements.
	10-12-10	IPLE/VOIPDB Subnet Mask	Set according to customers network requirements

SV9100

The screenshot shows the 'Easy edit' interface for SV9100 Test [SV9100 CP20 EMEA V10.5] - PCPro. The ribbon includes 'File', 'Home', 'View', 'Reports', 'Filter options', 'Tools', 'Grid style', 'Actions', and 'Ribbon search'. The 'Advanced view' tab is active, showing a tree view on the left with 'SIP Terminal DTMF Settings' selected. The main area displays a table of settings:

IP Address	1.2.3.4
Subnet Mask	255.0.0.0
Default Gateway	192.168.103.254
Default Gateway MAC Address	00-00-00-00-00-00
Time Zone	(GMT) Greenwich Mean Time, Dublin, Edinburgh, Lisb...
NIC Setting	Automatic detection
NAPT Router IP Address	0.0.0.0
ICMP Redirect	<input type="checkbox"/>
IPL IP Address	192.168.103.10
IPL Subnet Mask	255.255.255.0
DNS Primary Address	0.0.0.0
DNS Secondary Address	0.0.0.0
DNS Port	53
IPL NIC Port Setting	MDI
CPU MTU	1450
IPL MTU	1450

SL2100

The screenshot shows the 'Easy edit' interface for SL2100 Test. The ribbon includes 'File', 'Home', 'View', 'Reports', 'Filter options', 'Tools', 'Grid style', 'Actions', and 'Ribbon search'. The 'Advanced view' tab is active, showing a tree view on the left with 'Queue Messages' selected. The main area displays a table of settings:

IP Address	1.2.3.4
Subnet Mask	255.0.0.0
Default Gateway	192.168.103.254
NIC Setting	Automatic detection
NAT Router	Not used
NAPT Router IP Address	0.0.0.0
ICMP Redirect	<input type="checkbox"/>
VOIP IP Address	192.168.103.10
VOIP Subnet Mask	255.255.255.0
VOIP NIC Setting	Automatic detection
DNS Primary Address	0.0.0.0
DNS Secondary Address	0.0.0.0
DNS Port	53

Advanced Edit	PRG	Item	Setting
Advanced Items + VoIP + General Settings + IP Addressing + IPL VoIP Resource IP Addressing	84-26-01	VoIP Gateway IP Address	Set according to customers network SV9100 & SL2100 requirements. This requires 1 x static IP address for the DSP resources

SV9100

Slot	Item	Value
001	VOIPDB DSP IP Address	192.168.103.20
001	RTP Port	10020
001	RTCP Port	10021
001	Video RTP Port	20020
001	Video RTCP Port	20021
001	IP Address for Browser Phone Comm...	0.0.0.0

SL2100

Slot	Item	Value
000	VOIPDB ...	192.168.103.20
000	RTP Port	10020

Advanced Edit	PRG	Item	Setting
Advanced Items + VoIP + Extensions + SIP Extensions + SIP Device Setup	10-33-01	Registration Expiry Time	Leave set as default to 3600 . The 2N Verso will re-register every 120s set by device.
	10-33-02	Authentication Mode	Set to Enabled for Authentication using User name and Password.
	84-20-01	Registration Port	Default Port is 5070

SV9100

The screenshot shows the 'Easy edit' window for SV9100 [SV9100 CP20 EMEA V10.5]. The interface includes a ribbon menu with options like 'Apply', 'Copy', 'Paste', 'Fill', 'Default cell', 'Group by', 'Column chooser', 'Filter bar', 'Expand/Contract', and 'Page view'. The main area displays a list of settings:

Registration Expiry Time	3600
Authentication Mode	<input checked="" type="checkbox"/>
Registrar/Proxy Domain Name	
Registrar/Proxy Host Name	
Registrar/Proxy Port	5070
Session Time	180
Minimum Session Time	180
Called Party Info	Request URI
Receiving Invite Message Expiry ...	180
Sending Invite Message Expiry TL...	180
SIP out of range timer	4

The left sidebar shows a tree view of 'Programming Level' categories, including Voicemail, Night Service, Eco Mode, ARS, LCR, F-Route, Additional Devices, Advanced Items, ACD, Tie Lines, Hotel, and VoIP (General Settings, QoS Settings, Extensions, DT900, SIP Extensions, SIP Device Setup, SIP Terminal Settings).

SL2100

The screenshot shows the 'Easy edit' window for SL2100 [SL2100 EMEA V...]. The interface is similar to the SV9100 version. The main area displays a list of settings:

Registration Expiry Time	3600
Authentication Mode	<input checked="" type="checkbox"/>
Registrar/Proxy Domain Name	
Registrar/Proxy Host Name	
Registrar/Proxy Port	5070
Session Time	180
Minimum Session Time	180
Called Party Info	Request ...
Receiving Invite Message Expiry Time	180
Sending Invite Message Expiry Time	180
TLS Registrar/Proxy Port	0
SIP out of range timer	4

The left sidebar shows a tree view of 'Programming Level' categories, including F-Route, Additional Devices, Advanced Items, ACD, Hotel, VoIP (General Settings, QoS Settings, Extensions, IP MLT Setup, SIP Extensions, SIP Basic Information Setup, SIP Extension Setup).

Advanced Edit	PRG	Item	Setting
Advanced Items + VoIP + Extensions + SIP Extensions + SIP Terminal Settings	11-02	Extension Number	SIP Extension must register to an un-carded Extension Port.
	15-01-01	Extension Name	Enter the Name of the device.
	15-03-03	Terminal Type	Set to Special - Receive DTMF tones after the initial call is setup
	15-05-15	Codec Type	Default setting is Type 1.
	15-05-16	Authentication Password	Enter an 8 digit Password that will be used for Authentication when the device registers.
	15-05-43	Video Mode	Set to Enabled

SV9100

SV9100 Test [SV9100 CP20 EMEA V10.5] - PCPro

Station Port	Extension	Name	Peer to Peer Mode	Terminal Type	Terminal Type	Select Special Terminal Ty	Terminal MAC Address	Nickname	Using IP Address	Codec Type	Authentication Password	IP duplication allow mo	Video Mode	Additional Information
040	239	EXT 239	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
041	240	EXT 240	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
042	241	EXT 241	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
043	242	EXT 242	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
044	243	EXT 243	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
045	244	EXT 244	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
046	245	EXT 245	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
047	246	EXT 246	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
048	247	EXT 247	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
049	248	EXT 248	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
050	249	EXT 249	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
051	250	EXT 250	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
052	251	GT890	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Enable		
053	252	EXT 252	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Enable		
054	253	DoorPhone 2N	On	None	Special - Receive DT	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Enable		
055	254	EXT 254	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
056	255	EXT 255	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
057	256	EXT 256	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
058	257	EXT 257	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
059	258	EXT 258	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
060	259	EXT 259	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		
061	260	EXT 260	On	None	Normal - Ignore DTMF	Fax	00-00-00-00-00-00	0.0.0.0	Type 1	Disable	Disable		

SL2100

SL2100 [SL2100 EMEA V3.00] - PCPro

Station Port	Extension	Name	Peer to Peer Mode	Terminal Type	Terminal Type	Select Special Terminal Ty	Terminal MAC Address	Using IP Address	Codec Type	Authentication Password	IP duplication allow mo	Additional Information	Using Router	Video Mode	DTMF Playb
043	Ext242			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-60-E9-192.168.55.	192.168.55.	Type 1			0.0.0.0	Disable	
044	Ext243			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	192.168.55.	Type 1			0.0.0.0	Disable	
045	Ext244			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	0.0.0.0	Type 1			0.0.0.0	Disable	
046	Ext245			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	0.0.0.0	Type 1			0.0.0.0	Disable	
047	Ext246			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	0.0.0.0	Type 1			0.0.0.0	Disable	
048	Ext247			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	0.0.0.0	Type 1			0.0.0.0	Disable	
049	Ext248			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-60-E9-192.168.55.	192.168.55.	Type 1			0.0.0.0	Disable	
050	Ext249			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	0.0.0.0	Type 1			0.0.0.0	Disable	
051	Ext250			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	192.168.1.	Type 1			0.0.0.0	Disable	
052	251	GT890.1		SIP	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	192.168.10.	Type 1			0.0.0.0	Enable	
053	252	GT890.2		SIP	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	192.168.10.	Type 1			0.0.0.0	Enable	
054	253	Door Phone		SIP	Special - Receive DTMF tones after the initial call is setup	Fax	00-00-00-0.	192.168.10.	Type 1			0.0.0.0	Enable	
055	254	ST500 Moby		SIP	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	192.168.10.	Type 1			0.0.0.0	Enable	
056	Ext256			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	0.0.0.0	Type 1			0.0.0.0	Disable	
057	Ext257			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	0.0.0.0	Type 1			0.0.0.0	Disable	
059	Ext259			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	0.0.0.0	Type 1			0.0.0.0	Disable	
060	Ext258			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	0.0.0.0	Type 1			0.0.0.0	Disable	
061	Ext260			None	Normal - Ignore DTMF tones after the initial call is setup	Fax	00-00-00-0.	0.0.0.0	Type 1			0.0.0.0	Disable	

Advanced Edit	PRG	Item	Setting
Advanced Items + VoIP + Extensions + SIP Extensions + SIP Terminal Codec Settings	84-19-28	Audio Capability Priority	Enter the Codec priority as required. Default value is G.711PT. The 2N Verso supports G.711, G.729 & G.722.

SV9100

The screenshot shows the EasyEdit software interface for SV9100. The main window displays a table of settings for 'Audio Capability Priority'. The table has columns for 'Type 1', 'Type 2', 'Type 3', 'Type 4', and 'Type 5'. The 'Audio Capability Priority' setting is set to 'G.711_PT'. Other settings include G.711 Maximum Audio Frame Size (20ms), G.711 Voice Activity Detection (checkbox), G.711 Type (A-law), G.711 Minimum Jitter Buffer Size (20), G.711 Average Jitter Buffer Size (40), G.711 Maximum Jitter Buffer Size (80), G.729 Maximum Audio Frame Size (20ms), G.729 Voice Activity Detection (checkbox), G.729 Minimum Jitter Buffer Size (20), G.729 Average Jitter Buffer Size (40), G.729 Maximum Jitter Buffer Size (80), Jitter Buffer Mode (Self adjusting), Voice Activity Detection Threshold (20), G.722 Maximum Audio Frame Size (30ms), G.722 Minimum Jitter Buffer Size (30), G.722 Average Jitter Buffer Size (60), G.722 Maximum Jitter Buffer Size (120), G.726 Maximum Audio Frame Size (30ms), G.726 Voice Activity Detection (checkbox), G.726 Minimum Jitter Buffer Size (30), and G.726 Average Jitter Buffer Size (60).

SL2100

The screenshot shows the EasyEdit software interface for SL2100. The main window displays a table of settings for 'Audio Capability Priority'. The table has columns for 'Type 1', 'Type 2', 'Type 3', 'Type 4', and 'Type 5'. The 'Audio Capability Priority' setting is set to 'G.711_PT'. Other settings include G.711 Maximum Audio Frame Size (30ms), G.711 Voice Activity Detection (checkbox), G.711 Type (A-law), G.711 Minimum Jitter Buffer Size (20), G.711 Average Jitter Buffer Size (40), G.711 Maximum Jitter Buffer Size (80), G.729 Maximum Audio Frame Size (30ms), G.729 Voice Activity Detection (checkbox), G.729 Minimum Jitter Buffer Size (20), G.729 Average Jitter Buffer Size (40), G.729 Maximum Jitter Buffer Size (80), Jitter Buffer Mode (Self adjusting), VAD threshold (20), G.722 Maximum Audio Frame Size (30ms), G.722 Minimum Jitter Buffer Size (30), G.722 Average Jitter Buffer Size (60), G.722 Maximum Jitter Buffer Size (120), G.726 Maximum Audio Frame Size (30ms), G.726 Voice Activity Detection (checkbox), G.726 Minimum Jitter Buffer Size (30), G.726 Average Jitter Buffer Size (60), and G.726 Maximum Jitter Buffer Size (120).

Advanced Edit	PRG	Item	Setting
Advanced Items + VoIP + Extensions + SIP Extensions + SIP Terminal DTMF Settings	84-34-01	DTMF Relay Mode	Set to RFC2833
	84-34-02	DTMF Payload Number	Set Payload to 110

SV9100

SL2100

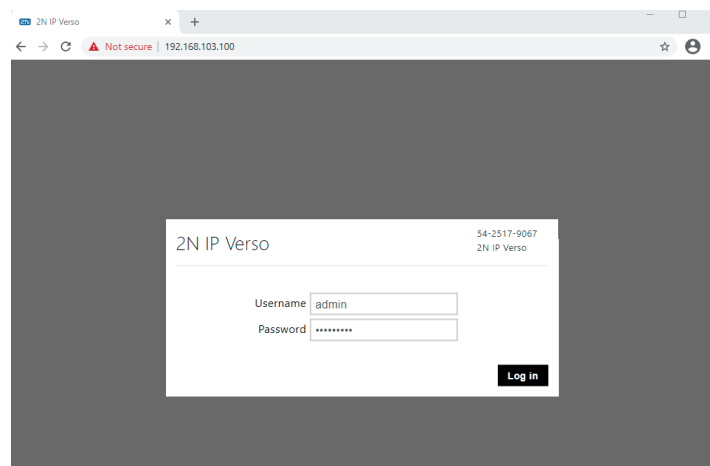
2N Verso SIP Video Door Phone configuration for connection to SV9100/SL2100 Platforms

LAN Connection and Logon Information

LAN Connection Setting You have to know the intercom configuration interface address to connect to the LAN successfully. Automatic IP address retrieval from the DHCP server is set by default in the 2N IP intercoms. Thus, if connected to a network in which a DHCP server configured to assign IP addresses to all new devices is available, the intercom will obtain an IP address from the DHCP server. The intercom IP address can be found in the DHCP server status (according to the MAC address given on the production plate), or will be communicated to you by the intercom voice function; refer to the Installation Manual of your intercom model. If there is no DHCP server in your LAN, use the intercom buttons to set the static IP address mode, refer to the Installation Manual of your intercom model. Your intercom address will then be **192.168.1.100**. Use it for the first login and then change it if necessary.

Now enter the intercom IP address into your favourite browser. We recommend you to use the latest Chrome, Firefox or Internet Explorer 9+ versions 2N IP intercom is not fully compatible with earlier browser versions.

Default User name is admin and password 2n (i.e. default reset password) for your first login to the configuration interface. The intercom requires a password change upon the first login. Strong passwords are only accepted - eight characters at least including one capital letter, one small letter and one digit.



Once you have logged in successfully, you should now be at the Home Screen



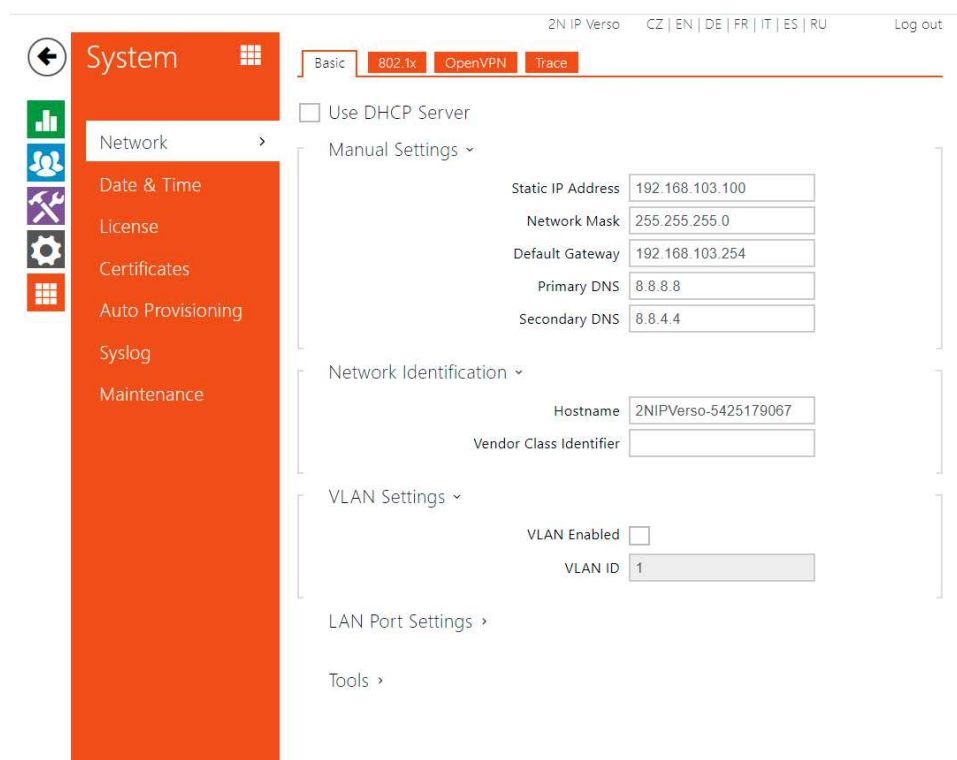
From the Home Page you need to configure as follows for the 2N Verso SIP Video Door phone to register to the SV9100/SL2100 Platforms.

To Set the Network settings for the device you need to click on the System button from the Home Screen.



Enter the new IP Address, Network Address and Default Gateway for the new device that must be in the range of the SV9100/SL2100 PBX.

Disable DHCP if not required. Then Click Save to confirm changes.



Click on the Services button from the Home Screen to enter the credentials to register to the SV9100/SL2100 Platform.



For SIP 1 enter the extension log in details required.

Intercom Identity:

- Display Name
- Phone Number (ID) – This is the extension number assigned
- Domain – This is the registration IP Address of the SV9100/SL2100 (10-12-09)

Authentication:

- Use Authentication – Set to enable (10-33-02)
- Authentication ID – Enter the extension number assigned
- Password – Enter the 8 digit password as set in 15-05-16

SIP Proxy:

- Proxy Address - IP Address of the SV9100/SL2100 (10-12-09)
- Proxy Port – Enter as 5070 (84-20-01)

SIP Registrar:

- Registration Enabled – Tick
- Registration Address - IP Address of the SV9100/SL2100 (10-12-09)
- Registration Port - Enter as 5070 (84-20-01)

Click **Save** when finished.

2N IP Verso CZ | EN | DE | FR | IT | ES | RU Log out

SIP 1 SIP 2 Calls Audio Video Local Calls

Intercom Identity

Display Name: IP Verso

Phone Number (ID): 253

Domain: 192.168.103.10

Test Call

Authentication

Use Authentication ID:

Authentication ID: 253

Password:

SIP Proxy

Proxy Address: 192.168.103.10

Proxy Port: 5070

Backup Proxy Address:

Backup Proxy Port: 5060

SIP Registrar

Registration Enabled:

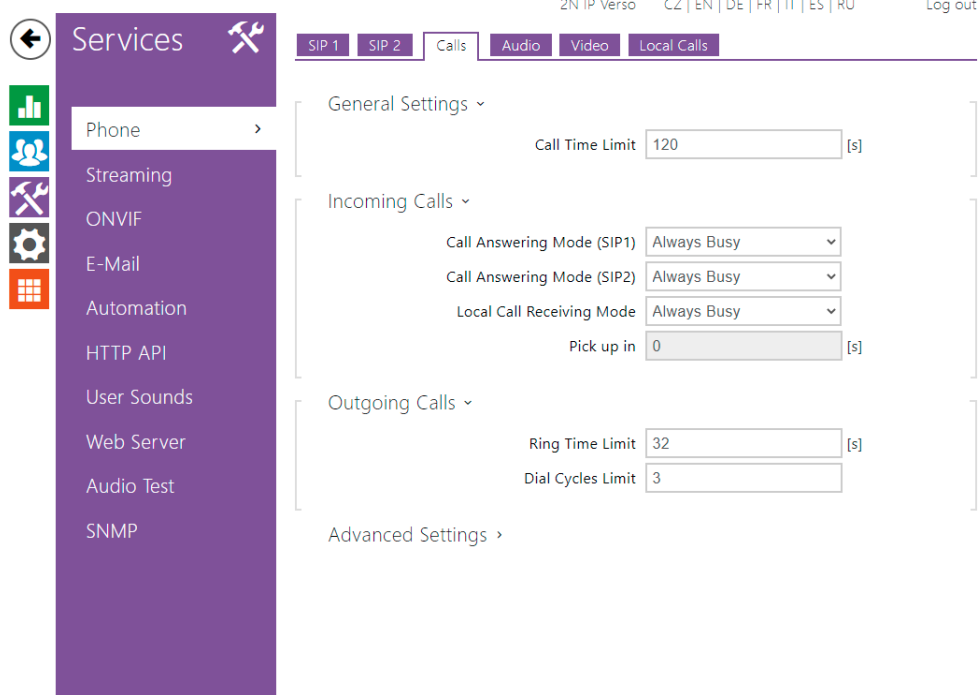
Registrar Address: 192.168.103.10

Registrar Port: 5070

Backup Registrar Address:

Save

In the **Call** page set the Maximum Call Time Limit required. Default value is 120s.



2N IP Verso CZ | EN | DE | FR | IT | ES | RU Log out

SIP 1 SIP 2 Calls Audio Video Local Calls

Services

- Phone
- Streaming
- ONVIF
- E-Mail
- Automation
- HTTP API
- User Sounds
- Web Server
- Audio Test
- SNMP

General Settings

Call Time Limit [s]

Incoming Calls

Call Answering Mode (SIP1)

Call Answering Mode (SIP2)

Local Call Receiving Mode

Pick up in [s]

Outgoing Calls

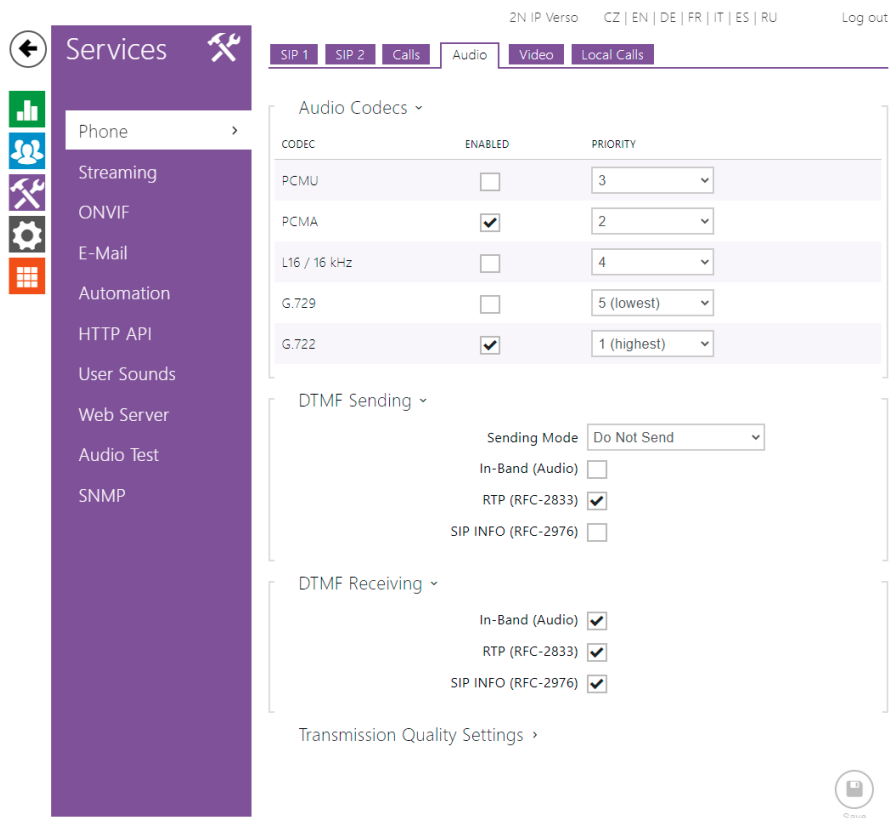
Ring Time Limit [s]

Dial Cycles Limit

Advanced Settings

In the **Audio Page** you need to set the supported Codec priorities. Tests performed using G.722 as priority 1 and PCMA (G.711) as Priority 2.

G.722 is the default Codec used with the GT890 while G.711 will be negotiated by the SV9100/SL2100 by default for calls to other devices.



2N IP Verso CZ | EN | DE | FR | IT | ES | RU Log out

SIP 1 SIP 2 Calls Audio Video Local Calls

Services

- Phone
- Streaming
- ONVIF
- E-Mail
- Automation
- HTTP API
- User Sounds
- Web Server
- Audio Test
- SNMP

Audio Codecs

CODEC	ENABLED	PRIORITY
PCMU	<input type="checkbox"/>	3
PCMA	<input checked="" type="checkbox"/>	2
L16 / 16 kHz	<input type="checkbox"/>	4
G.729	<input type="checkbox"/>	5 (lowest)
G.722	<input checked="" type="checkbox"/>	1 (highest)

DTMF Sending

Sending Mode

In-Band (Audio)

RTP (RFC-2833)

SIP INFO (RFC-2976)

DTMF Receiving

In-Band (Audio)

RTP (RFC-2833)

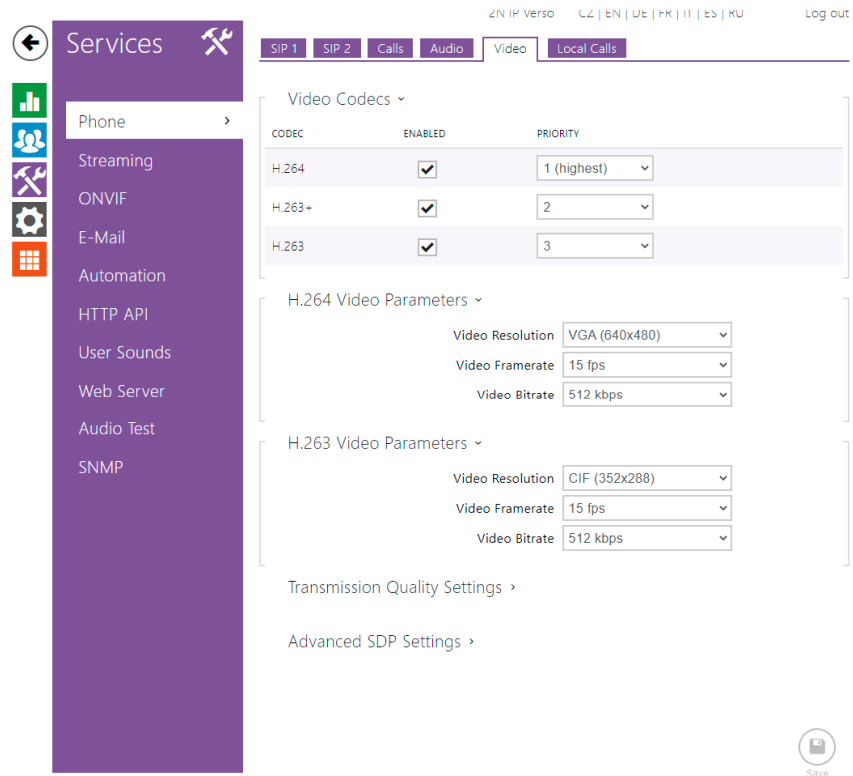
SIP INFO (RFC-2976)

Transmission Quality Settings

Save

In the **Video Page** you will find the Video Codecs supported.

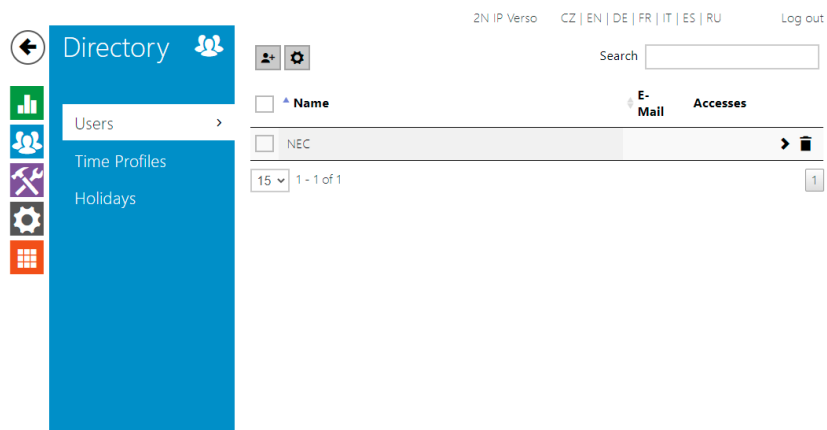
Video is only supported with calls to the GT890 Terminal using the ST500 Application. Supported video Codec is H.264 (Default Setting).



Return the **Home Screen** after saving settings and select the **Directory Button**



Within **Users** you need to create a New User, give this user a name. For this example the User is called NEC.



Click on the new user NEC and then you will be in the programming area below.

In User Basic Information, enter the name that appears on the 2N Device when the Door Entry Button is activated. This is also used in the Directory if multiple entry Buttons supplied.

In User Phone Numbers, enter the destination number to be called when the Door Entry Button is pressed. This can be any internal Directory number that can be dialled on the SV9100/SL2100 as an Extension, Virtual Extension.

2N IP Verso CZ | EN | DE | FR | IT | ES | RU Log out

Directory

Back to List

User Basic Information

Name NEC

Photo

E-Mail

Virtual Number

Add to Display

Position in directory

Calling group

User Phone Numbers

Number 1

Phone Number 251

Time Profile [not used]

2N IP Eye Address

Group call to next number

Number 2

Phone Number

Time Profile [not used]

2N IP Eye Address

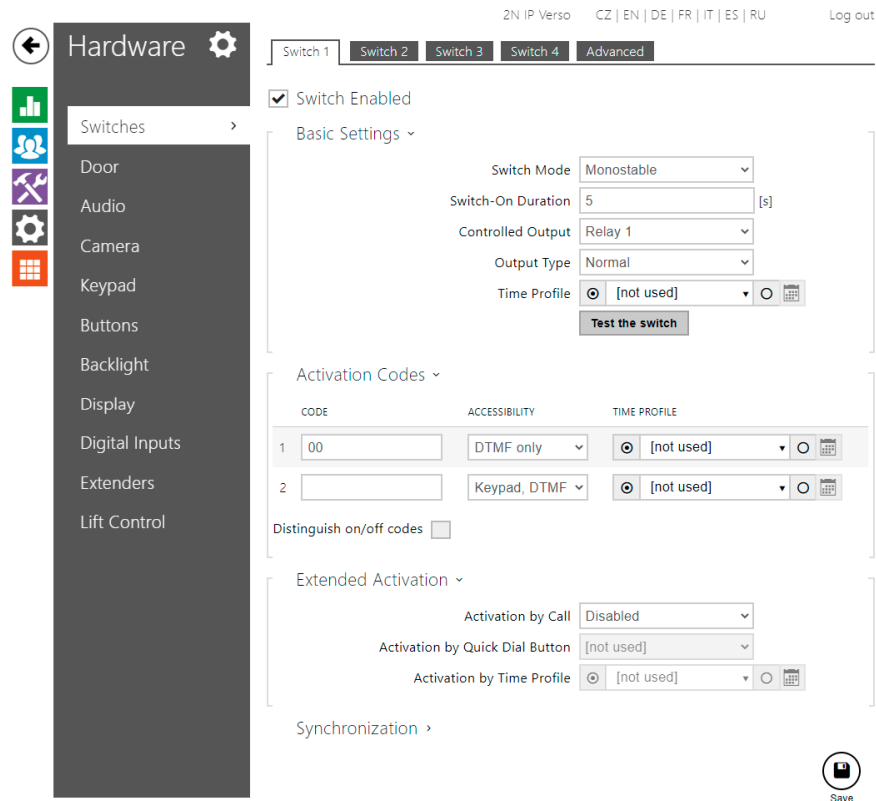
Save

Click Save when ready and return to the Home Page

Finally we need to select the **Hardware Button** from the Home Screen



In the **Switches area**, Tick Switch Enabled and then enter the Activation Code required to open the door. Set Accessibility to DTMF Only. The Code set in this example is 00 so to open the Relay the extension that answered the call would enter 00*



Click Save and return back to the Home Page. Then check if the device is registered as per the example for SIP 1 Number below. If it is then you are ready to make a test call.

Device Status

Status

SERIAL NUMBER	54-2517-9067
FIRMWARE	2.31.0.40.5
UP TIME	3h 34m 26s
SIP 1 NUMBER	REGISTERED 253
SIP 2 NUMBER	NOT REGISTERED 111

Known Limitations/Comments

- Video mode is only supported with peer-to-peer enabled.
- Video support with GT890 ST500 Application Only.
- Device is not capable of receiving incoming calls.
- Configuration notes provided are to set up a single 2N Verso SIP Door Entry Unit to the SV9100/SL2100 Platforms only.
- For further configuration information on the 2N Verso SIP Door Entry Unit then please view the installation Guide.

Compatibility Overview

	Compatible/Incompatible
Registration - No Authentication	
Successful Registration	Compatible
Maintain Registration	Compatible
Registration Refresh	Compatible
Registration Failure	Compatible
De-registration	Not Applicable
Registration with Authentication	
Successful Registration	Compatible
Maintain Registration	Compatible
Registration Refresh	Compatible
Registration Failure	Compatible
De-registration	Compatible
Registration Failure 480	Compatible
Call to PSTN/ISDN	
Call Setup	Not Applicable
Call Continuance	Not Applicable
Disconnect by Calling Party	Not Applicable
Disconnect by Called Party	Not Applicable
Call Cancel	Not Applicable
Call internal to Key telephone TDM/ GT890 / DT900	
Call Setup	Compatible
Call Continuance	120 seconds
Disconnect by Calling Party	Compatible
Disconnect by Called Party	Compatible
Call Cancel	Compatible
Incoming call from PSTN/ISDN	
Call Setup	Not Applicable
Call Continuance	Not Applicable
Disconnect by Calling Party	Not Applicable
Disconnect by Called Party	Not Applicable
Call Cancel	Not Applicable
Incoming call from Key telephone	
Call Setup	Not Compatible
Call Continuance	Not Compatible
Disconnect by Calling Party	Not Compatible
Disconnect by Called Party	Not Compatible
Call Cancel	Not Compatible
CODECs Disabling and Reordering	
Disabled CODEC outgoing call	Not Compatible
Disabled CODEC incoming call	Not Compatible
CODEC order outgoing call	Compatible
CODEC order incoming call	Compatible
Sending digits during ringing	
Sending digits during ringing	Not Applicable
Sending digits during conversation	

Sending digits during conversation	Compatible
Sending digits during ringing (DTMF Relay)	
Sending digits during ringing	Not Applicable
Sending digits during conversation (DTMF Relay)	
Sending digits during conversation	Not Applicable
Flash hook/recall	
Hold & retrieve	Not Applicable
Consult Transfer (Flash hook/recall)	
Hold & Transfer (REFER)	Not Applicable
Blind Transfer (Flash hook/recall)	
Hold & Blind Transfer (REFER)	Not Applicable
Call to PSTN/ISDN Peer to Peer enabled	
Call Setup	Not Applicable
Call Continuance	Not Applicable
Disconnect by Calling Party	Not Applicable
Disconnect by Called Party	Not Applicable
Call Cancel	Not Applicable
Call internal to Key telephone Peer to Peer enabled (GT890)	
Call Setup	Compatible
Call Continuance	Compatible
Disconnect by Calling Party	Compatible
Disconnect by Called Party	Compatible
Call Cancel	Compatible
Sending digits during ringing Peer to Peer enabled	
Sending digits during ringing	Not Applicable
Sending digits during conversation Peer to Peer enabled	
Sending digits during conversation	Compatible
Sending digits during ringing Peer to Peer enabled (DTMF Relay)	
Sending digits during ringing	Not Compatible
Sending digits during conversation Peer to Peer enabled (DTMF Relay)	
Sending digits during conversation	Compatible
Flash hook/recall Peer to Peer enabled	
Hold & retrieve	Not Compatible
Consult Transfer (Flash hook/recall) Peer to Peer enabled	
Hold & Transfer (REFER)	Not Compatible
Blind Transfer (Flash hook/recall) Peer to Peer enabled	
Hold & Blind Transfer (REFER)	Not Compatible
Power out/kill process	
Unexpected SIP extension termination (Idle)	Compatible
Unexpected SIP extension termination (In conversation)	Compatible
Network Issues	
Disconnect the Ethernet cable X3 (During idle)	Compatible
Disconnect the Ethernet cable X3 (During conversation)	Compatible
Disconnect the Ethernet cable X4 (During idle)	Compatible
Disconnect the Ethernet cable X4 (During conversation)	Compatible
Disconnect the Ethernet cable X2 (RTP stream) (During conversation)	Compatible

Audio CODEC Overview – See section 7 for detailed test results of CODEC compatibility.	
G.711	Compatible
G.729	Compatible
G.722	Compatible
G.726	Incompatible

Video Overview – Supported using GT890 Terminal with ST500 Application	
H.264	Compatible
H.263	Incompatible
H.262	Incompatible

Further Support

Actual product support must be obtained from the supplier or manufacturer of the third party device.

Document History

Version	Date	Description
1.0	02/03/2021	Initial Release

TO THE EXTENT NOT PROHIBITED BY LAW, IN NO EVENT SHALL NEC ENTERPRISE SOLUTIONS BE LIABLE FOR PERSONAL INJURY OR ANY INCIDENTAL, SPECIAL, DIRECT, INDIRECT OR CONSEQUENTIAL DAMAGES WHATSOEVER, INCLUDING WITHOUT LIMITATION, DAMAGES FOR LOSS OF PROFITS, LOSS OF DATA, BUSINESS INTERRUPTION OR ANY OTHER COMMERCIAL DAMAGES OR LOSSES ARISING OUT OF OR RELATED TO YOUR USE OF SIP COMPATIBILITY REPORTS, HOWEVER CAUSED REGARDLESS OF THE THEORY OF LIABILITY (CONTRACT, TORT OR OTHERWISE) AND EVEN IF NEC ENTERPRISE SOLUTIONS HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES, DUE TO THE POSSIBILITY OF PROVIDERS MAKING SYSTEM CHANGES WHICH PROCEED THE COMPATIBILITY TESTING DETAILED IN THIS REPORT.

IN NO EVENT SHALL NEC ENTERPRISE SOLUTIONS TOTAL LIABILITY TO YOU FOR ALL DAMAGES (OTHER THAN AS MAY BE REQUIRED BY APPLICABLE LAW IN CASES INVOLVING PERSONAL INJURY) EXCEED THE AMOUNT OF SEVENTY FIVE POUNDS (£75).

THE FOREGOING LIMITATIONS WILL APPLY EVEN IF THE ABOVE STATED REMEDY FAILS OF ITS ESSENTIAL PURPOSE.